The P.A. Workshop
by
Colin Owen

ABOUT THE AUTHOR.

Colin Owen has spent almost a whole lifetime being involved in music, some 36 years in all. Starting at the age of eight, the list of instruments he has played at some time or other throughout those years includes, mouth organ, ukulele, piano, guitar, bass guitar, synthesiser, flute, and a little cello.

Colin played ukulele in church between the ages of eight and ten, but began his career in music at the tender age of 13, when he first played guitar in the local working mens club. Being born in Liverpool, and living there at the time of the Beatles, it is no surprise that he followed a somewhat varied career in the pop business. Playing in many different kinds of groups. He travelled to Germany at the age of eighteen to the then famous Top Ten club in Hamburg, and from that time on became a professional musician, giving up his job working in a piano shop.

After ten years of playing music for a living, Colin gave up his travelling to become the resident sound engineer in a small studio in Worcester. Having always loved tape recorders even from early childhood, the work in a studio was ideal. It was here in the studio that the recording and PA system knowledge came together with the musical knowledge gained over the previous fifteen years, and as the studio grew and expanded, so did Colins experience and know how.

Working in the studio was a great training ground, and the many hours spent writing, recording, and singing radio jingles has proved to be of great use. Colin also had a small taste of success in the mid seventies with two records entering the top twenty, but this was short lived as he says someone ran off with all the money.

Colin was born again at the age of twelve, but sadly the pull of the pop music scene was too strong and so he left God behind to follow his own way. It wasnt until Colin turned forty that he came back to God in a real way. Having met up with and married Carol, also a singer, God became the central focus in his life once again.

Colin and Carol are currently worship leaders, and have a world wide tape ministry.

The Worship Workshop

Having travelled to many churches both in England and on the Continent, the need for this teaching, aimed at all those who are involved in praise and worship in the church in any way, be it musicians, singers, technicians, or church leaders, has been recognised. So this is what the Worship Workshop is all about.

Part 1 is for musicians, and deals with instruments.

Part 2 is for church technicians, and deals with many aspects of PA, and things like cables, and speakers etc.

It has been my pleasure to embark on writing this teaching course. My prayer is that our precious Lord will take of me, and give to you, I have been blessed with so many years experience in music and recording I want to share it with you

MICS.

There are many makes of mic on the market, and they range in price from 10 or so to something over 2000. I hope in this section to give you enough information to allow you to make the right choice when buying a mic, and also to help you get the best out of the mic that you may already have.

TYPES OF MIC.

There are two main types of mic, the dynamic, and the condenser. The condenser mic is really for recording so Ill talk here about the dynamic, as this is almost certainly the type of mic you will buy and or use in church.

THE DYNAMIC MIC.

Price ranges for dynamic mics are from 10 for a piece of rubbish, to 200 for a really good one. The sensible price range to look at for church use is around 40 to 120. Within this range you can get a very good vocal mic, which is all you need.

PICKUP PATTERN.

There are two types of pickup pattern in common use, the pickup pattern can be described as the area around the mic that picks up best. The simplest to explain is called omni, and is shown on the mic as a circle. Yes you guessed it, the omni mic picks up equally well from all sides. This is not a good mic to use for vocals as it picks up all sorts of things that you dont want, like the drums or the guitar amps, the congregation etc. Also they are a lot more prone to feedback. This type of mic is quite rare, and you would have to go and ask for one specially in order to get one, so there is not much chance of buying one by mistake, unless you buy one second hand or from a shop that doesnt know the difference.

There are two advantages to this pickup pattern, one is that they are a bit cheaper to produce, and also they dont suffer from proximity effect. I know, what in thunder is the proximity effect? Well its quite easy to explain, its the increase in bass that happens when you get right up to the mic. Neither of these things are of any real advantage to you the user when compared to the disadvantage I mentioned earlier. The most common pickup pattern is called cardioid, and is shown on the mic body as a heart shape. (sometimes shown upside down). The best way to understand this pattern is to imagine the mic head in the dimple of the heart.

It should be obvious that these mics dont pick up much from the rear, which is very useful in practice. They feedback a lot less, and they do suffer from the affore mentioned proximity effect, but this can be used to great effect by a skilled mic user.

There is one other pickup pattern that deserves a mention, and that is the figure of eight. Guess how they represent that on the mic body! This pattern is really useful for picking up two voices or instruments simultaneously. It behaves like two cardioid mics set back to back and can be used in the middle of a bunch of singers where some have louder voices than others. The ones with the loud voices stand to the side of the mic, whilst the quieter voices stand either to the front or back, thus the mic can go a long way to solving this kind of problem for you. In essence you only find these mics in the studio, as the cost is somewhat prohibitive when it comes to PA work, and the usefulness of this type of mic for PA purposes is quite limited.

IMPEDANCE.

Most people have heard the terms high or low impedance, what it actually means is not important so long as you know the difference it makes to your system. A mic, like all electronic equipment is made to work into a specific load, that is they need to be matched to the input of the amplifier. There are two types, high impedance matches to about 50000 ohms (an ohm is a unit of resistance), and low impedance is normally between 200 and 600 ohms. This only really applies to mics, as all guitars/keyboards etc run at 50000 ohms or more. Keyboards should ideally be matched to 1 meg (1,000,000) ohms, this is high impedance. It makes no difference to the system if the mic is 250 ohms or 600 ohms both are low impedance and will work equally well.

As a rule most mics will be low impedance or low-z as it is called, this is because of the benefits of this type of mic. Low-z mics will work with long cables, hi-z mics can only really work with cables of 10 or 12 feet before they start to loose treble, but low-z mics can work with cables up to several hundred yards, so long as they are balanced. Low-z mics also sound better than hi-z mics, the frequency response is much better.

You are probably puzzled about the word balanced, yes thought you might be. A balanced signal requires 3 wires. Two for the signal and one for screen. The screen wire is wrapped or braided around the two signal wires to prevent any RF or radio frequencies getting into the signal wires. The signal wires are connected to each end of a transformer coil. The screen wire is connected to the centre or mid point of the coil (called Ov, or O volt). So there is the resistance of the transformer coil between the earth connection and both of the signal wires. Also the transformer coils are wound in opposite directions, so the signal wires are anti phase, this helps to cancel hum pickup in the length of the cable. The screen is always earthed so that any stray radio signals are earthed out before they can enter the cable. That is why the screen is wrapped or braided around the inner cores of the mic cable, to give an unbroken shield along the cables length. This screen or ground as it is called is different to the signal earth, it connects the body of the mic to the chassis of the mixer which is in turn connected to the mains earth, or literally the ground.

The input of the amplifier must also have a similar transformer to accept the incoming signal. With this kind of balanced system you can safely run very long cables without much fear of RF or hum pickup, or signal loss.

It is possible to use unbalanced mics with balanced inputs, but you would need to connect one of the signal wires to the screen wire to unbalance the system. This works fine but you lose some of the benefits just talked about, and also it can introduce hum into the system.

As a rule balanced inputs on mixers or PA amps will be XLR type three pin sockets. While unbalanced inputs will always be on standard jacks. Some PA amps and cheaper mixers use stereo jacks for their balanced inputs, this is a good compromise as you can plug in an unbalanced or normal guitar jack and the barrel of the jack does the unbalancing for you by shorting the screen and the ring connections together.

If you have the option to run balanced or unbalanced, always run balanced, its much safer.

RADIO MICS.

Radio mics have come a long way since their introduction some years ago. Most of the problems of the earlier models have been sorted out, but you still need to know what to look for to get a good system. As a rule hand held radio mics are less troublesome than tie clip types, but you couldnt sing through a tie clip mic so that limits your choice somewhat. Look at the receiver first, it should have a choice of outputs to suit your requirements. Balanced XLR for connection to a mic input, and jack for connection to a line input. Beware of any system that has phono plugs fitted, this is a bad cost cutting exercise on behalf of the manufacturer, and means that there are probably other hidden things of a similar not so good nature inside.

Has it got two aerials? This is most important, a single aerial is much more likely to suffer from signal bounce, or reflection problems. That simply means the signal may die away as the mic is moved around. The resulting radio static noise is most annoying. A diversity system is much better as this has in effect two receivers in one box, with a sensor to select the strongest signal. This is a much more reliable system and is to be recommended. It uses two aerials set apart so that one aerial at least will be receiving a good clear signal.

The hand set is fairly fool proof. Make sure it is not too sensitive to contact noise through the body. Otherwise they are all pretty good whatever the price.

With regard to the tie clip type, they also are all pretty good, but some use the mic cable as the transmitter aerial. This in itself is OK, but you must remember not to coil or kink the cable as this does have an affect on the transmission quality, and distance. Its favourite to stuff the cable into the pocket, avoid this if you can. If you should run into any noise or loss of signal problems, move the receiver closer to the transmitter. Try putting it on the side of the platform instead of having it at the back by the mixer. This will probably help quite a bit, especially with the cheaper systems. Remember to change the batteries regularly.

NON RADIO TIE CLIP MICS.

Many churches dont have radio mics, but do use tie clips. This is quite OK as long as you dont mind the cable. You can pay all sorts of prices for these mics, but Ive found that the Tandy one works very well and it only costs 20 or so. If it should break in some way then just buy a new one. The cable needs to be lengthened but that is a simple job.

BALANCING AN UNBALANCED MIC.

I want to pass on a tip here that I got from a magazine called Home and Studio Recording, it concerns balancing the Tandy, or any other unbalanced mic for that matter. Weve already discussed the advantages of using balanced mics, so this trick is well worth the effort and the small cost. First check that your mixer/PA amp has balanced inputs, and also that the cable you are using from the platform to the mixer/amp has two inner cores and a screen, if so then what we need to do is trick the amp into thinking that the mic is balanced when in fact it isnt. Remember I said earlier that the balanced input has a transformer, the transformer has two wires connected to it and each wire should have a resistance of 600 ohms with regard to earth. The mic which is unbalanced has only one wire carrying the signal and a screen, nearly always fitted with a standard guitar jack, so at the socket one wire is shorted to earth ie no resistance. We need to create 600ohms resistance on the second wire that has been shorted. This is quite simple to do.

At the platform end of your mic/multicore cable. Assuming that the Tandy is plugged into a socket somewhere on the platform, you need to change the mono jack socket for a stereo one. If the socket on the platform is already a stereo jack, or an XLR, then thats fine, leave it alone. So you have three wires to connect to three terminals on the socket. Lets look at jack sockets first.

Connect the screen to the tag nearest the mounting nut, now connect one of the inner cores to the next tag via a 600 ohm resistor (Tandy sell them). Lastly connect the last wire to the last tag and thats it.

Now when you plug the mono jack plug from the mic into this socket, both of the inner cores of the cable to your mixer will have a 600 ohm load, one side the mic, and the other the resistor thus fooling the mixer into thinking that the mic on the other end is balanced. Look inside the XLR end of the lead that came with the mic (if there was one), if this has three separate wires then it is a balanced mic to start with. You need only change the mono jack plug on the other end for a stereo one, and separate the 2 inner wires so that one goes to each tag in the jack plug, and the job is done. In this case you wont need the resistor on the socket, a stereo socket will do.

If the socket on the platform is an XLR, then dont alter it, rather alter the XLR plug on the mic cable you are using. Once altered mark it so that it will only be used with unbalanced mics.

Connect the screen to pin 1, the resistor to pin 3 and pin 1. The other wire goes to pin 2. Pins 2 and 3 may need to be swapped depending on the way the mics XLR socket is wired. If it doesnt work at all then swap the wires.

ELECTRET MICS.

The Tandy is an electret condenser, that means it runs on a battery which will need to be replaced every 6 months or so. Electret condenser mics sound very good considering their price. You can get hand held electrets from 15 to 150, but the cheap ones work very well indeed and should be considered when buying vocal mics. A 30 electret mic will as a rule sound much better than a 100 dynamic mic, especially at the top end, so bear this in mind.

One point of warning about electrets is that they have a switch fitted to save the battery life, if you switch the mic on or off whilst the PA is on you will get an almighty thump through the speakers. So dont!

MIC TECHNIQUE.

Pop is more than a fizzy drink, to a PA system (and people) it can be a real pain. Many mic users dont know what pops are, let alone how to avoid them. So for those who dont know.....

A pop is what you get when you blow into the mic (dont do it). But you say I dont blow into mics, well sorry but you do. Want proof? Try this little test. Put your hand to your mouth palm inwards, and about 2 inches away from the said orifice, now say after me, Peter Piper put his peanuts in his pocket! Praise God for poetry!! Try ooh as opposed to ah.

The breath you feel hitting your hand is what the mic translates into pops. Now practice saying the rhyme without your breath hitting your hand, it can be done. This is the way you must sing or speak into a mic, learn how to do it and you will sound much more professional, and also you will not distract the congregation or damage your speakers, or mics. Use a pop shield. Pop shields have a two fold ministry to mics. They help to filter out pops ! But they also stop the damp from your breath getting into the very delicate coil and diaphragm just inside the head of the mic. Damp is bad for mics as is blowing into them to see if theyre on. The diaphragm is made of very light material and can be easily torn or blown out of shape by a blast of air. Never blow into a mic. Tell the pastor and the deacons and anyone else who needs to know.

Most singers are aware of the need to back off the mic when going for a loud note, and coming in on the soft ones. If you dont do this the sound operator will turn the mic down because of the loud notes, which means that the softer notes will be lost, this is a very common problem in church. Dont shout into the mic. Make it work for you, thats what its there for. You need never strain your voice if you know how to use the mic properly.
Remember also that you sing at a much higher level than you speak, this is more true of the ladies who can have quite a delicate speaking voice, but crack the chandelier when they sing. So its IN to talk and OUT to sing every time, get into the habit.

Be aware of these things, they matter, and given consideration they will enhance any performance. Pop shields should be washed regularly, they can very soon become quite offensive to the nose!!. If the foam gets clogged for some reason then the treble response will suffer. Dont use soap of any kind as you will never be able to get it all out, just a good soak in fairly hot water will do the trick.

Dont be tempted to wash the end of the mic. If it becomes necessary to clean it use a tooth brush dipped in carbontetrachloride, or isapropylalchahol, you can usually get these from your local chemist. Hold the mic upside down so that none of the fluid goes inside. Use circular movements, and MIND YOUR EYES.

TO SUMMARISE.

Choose your mic carefully, look after it, and it will last you a long time.
Put mics away after use. The case will protect the mic from damp and dust, both abound in most churches. A locked cupboard protects against thieves.

If you should go for the electret mic remember the battery, it should be changed every 6 months or so depending on how much you use the mic. Dont leave the battery in for ages it might leak and do damage to the mic. You may need to clean the switch with a cleaning spray from time to time as dust does get in and can cause nasty noises to come from the speakers. You can also get the odd problem with loose batteries. These are all minor niggles and shouldnt put you off these very good, value for money mics.

Maplins do a nice range of mics that are not too expensive. They also sell some electret condenser mics. Its well worth having their catalogue around.

PA SYSTEMS.

This is another area where church folk seem to be clueless. Why is it that church PA systems seem to cost 5000 and sound awful? The speakers are usually skinny little boxes that cant reproduce any bass at all. Modern church buildings seem to be much more with it with regards to PA systems than the more traditional ones. If you are a travelling music ministry you cannot rely on any church having a good PA system. Some do but the majority are next to awful. With this in view you should look to buying your own system. Even if you are a solo singer or duo as my wife Carol and I are, it is always better to have your own PA than to rely on the house PA.

For one thing you can control your own PA and not have to rely on someone else to do it for you. Good PA operators are even more rare than good PAs. Well look at that later. So what do you need to buy to make up a reasonable PA? First examine your needs, IE solo singers only need one mic, and maybe an input for backing tapes (more in workshop 3), so you wont need a 10 input amp or mixer. On the other hand, a group will need 4 or 5 mics plus inputs for keyboards, and maybe even mics for the drums. So know what you need before you look. Keep it as small as you can. A solo singer doesnt need great bass speakers, but any band that mics up the drums will. Consider transport and ease of set up its important.

MIXERS MIXER AMPS AND PA AMPS.

PA amps that have quite limited facilities are quite cheap to buy, but will only be in mono (if thats no problem). You will always find treble and bass controls, and usually a cheap reverb system. This type of amp, may have up to 6 inputs, tape input, and an output of up to 100/150 watts. They work fine and are your cheapest option. For a small setup a mixer amp may be preferable to a separate mixer and amp. Mixer amps come in many shapes and sizes, from 4 input mono, to 12 input stereo. Ive never seen a mixer amp with more than 12 inputs, so if you need more than 12 you will have to go for a mixer and separate amp.

A true mixer amp will have the appearance of a mixer, with many more facilities than the humble PA amp. Top, mid, and bass controls, faders instead of rotary knobs, meters etc. They are virtually mixers with built in power amps. This is a convenient system and quite a neat way of doing it, but they are expensive. Also you would have to run your speaker leads from the back of the room, or where ever the mixer was. You cant run the output from a power amp down a multicore!. Mixer amps may suit you if you do your own mixing from on stage though! Otherwise the best option is the separate mixer and power amp. Mixers are quite cheap these days, you could expect to get 12 channels for between 300 to 400. Add to this a power amp for something like 300, a 90 by 90 watt mosfet amp would be ample in most church setups, (custom sound make one), two good speaker cabs and youve got a PA. The advantage of a separate system like this is its flexibility, you can change or add to any of the component parts. Need more channels? Add another mixer, note it will often be cheaper to buy two 12 channel mixers than one 24 channel. To link them together costs you two channels, so you end up with 22 (assuming you are using stereo) not bad for 700 or so. This is a great way to sub-mix the drums and control the whole drum kit with just two faders. Need more power? Just add more amps and speakers, simple, flexible, efficient.

THE POWER AMP.

In workshop 1 I talked about wattage with regard to amplifiers, to recap just a little, the wattage of an amp relates to the heat the amp generates across the voice coil of the speaker, not volume. There is obviously a correlation between the two, but the scale is not a linear one IE 50 watts is not twice as loud as 25 watts, its twice as hot. Thats why you see things like air cooled, or fluid cooled, associated with loud speakers. A good cooling system allows the speaker to achieve a higher wattage. If you ever look inside a blown speaker you will see heat damage, burned out wire etc. You will hardly ever come across a speaker that has been damaged by too much volume. Volume damage is caused by the cone of the speaker moving further than it was designed to, tearing the cone, or the surround in some way. Transit causes this kind of damage more than anything, some sort of foreign body poked through. The roadies knee for instance, thats why cases are so important.

A MYTH BITES THE DUST.

If asked about such things, most people would say you have to have a 200 watt speaker for a 100 watt amp, because a 100 watt amp can generate spikes of 200 watts. True it can, but only for short durations. What the speaker needs to know is the RMS wattage, or the average wattage of the amp, and the average wattage will always be less than the stated maximum. Remember we are talking about heat, not volume. A far safer idea is to have a 500 watt amp with your 100 watt speaker! Why? Well a 500 watt amp running at half throttle will generate clean waveforms, as opposed to a 100 watt amp driven flat out. The distorted waveforms from the 100 watt amp will generate more heat in the speaker coil than clean ones from the 500 watt amp. Liken it to a mini travelling down the motorway at 80 miles an hour. The engine is being run at close to its maximum so it is struggling and therefore generating a lot of heat. Now do the same speed in a Merc and theres no problem, the engine is just ticking over so theres no struggle and nowhere near the same amount of heat being generated. For our purposes in this illustration heat = distortion. If you have a big amp and a small speaker, you will probably never blow your speaker. Thats the principal used in all recording studios, and big PA systems. Amps of 1 thousand watts driving speakers of 1 to 600 hundred watts. It sounds crazy, but it works. So if you are serious about your PA get a big amp. Remember that distorted waveforms generate heat in peoples ears as well as speakers, and they complain. But clean waveforms dont hurt your ears, even at 120 db (which is bloomin loud), so bear this in mind when buying.

If youve a mind to you can build your own amp quite easily (I know cos Ive done it several times) Good old Maplins catalogue has all the parts you need to build a stereo or mono amp with easy to follow instructions. Its cheaper than buying, but takes a bit of work. Why not take a look.

EARS.

It is probably prudent at this point to talk about ears and hearing. There are some things you should know as they will help you to keep from damaging your hearing which is precious. Lets start with the donts. Dont ever stand in a direct line with a loud sound source, IE speakers, trumpets etc. The SPL or sound pressure level is greatest in a direct line. Always stand to the side where the high frequency content of the sound is reduced. A trumpet can develop frightening SPL levels at the bell so keep your distance. As a rule of thumb sound will decrease in level by 3 db over 1 meter. What that means in lay speak is that sound will loose half of its energy for every meter it travels.

With hearing damage, the frequency range that goes first is around 4 thousand cycles per second (4K for short). Once this area has been damaged it cannot be repaired (except through prayer). Its called speech deafness, and is very common in industry where people work in noisy factories. Noise which is a constant drone is far more dangerous than amplified music which is a series of short but loud peaks. Dont ever get into a situation where your ears zing afterwards, this is a warning sign. If you constantly get into loud situations your ears will turn down to protect themselves, and eventually they will be unable to turn back up again. Most of us will have experienced this after a loud concert. Its a warning, ignore it at your peril. Dont have headphones too loud. This is an increasing danger area because of walkman type devices that are so popular these days. Walkmans are great, they sound really good, but all that treble going directly into your ears will eventually cause damage if you play your music too loud. Watch for the tell tale signs, note the number on the volume control at the beginning of the day, and see how it creeps up as you listen more and more, as you need more volume to keep the kick of the music, your ears are actually trying to protect themselves and so are turning down. They are your ears so its up to you to look after them.

Do keep away from any noise that hurts your ears, pain is always a message that something is wrong. If you work in a noisy environment wear ear protectors no matter what they look like. If you have trouble whilst playing because of the guitar amp or drums being too loud then move away from them, or turn sideways on. Be careful of having one ear blasted all the time. Experiment with the way you setup on stage, try to find the best layout. If you are happy with what you can hear then you will play better. Do think of the people out front, your guitar amp, on the floor of a raised platform, may not sound very loud to you if you are stood right over it, but to someone sat down in the congregation who happens to be in a direct line with the speaker, it may well be painful. Think of these things they are important. Tilt the amps up towards you, or get an amp stand that raises the amp up nearer to your ears. This will cause you to hear better and so turn down. Dont think they wont hear you at the back, they will. See the section on sound directivety.

SPEAKERS.

The size and type of speaker you will need depends on the demands you will make of it. A solo singer with backing tapes can get away with almost any good make of speaker, even quite small ones, as taped music doesnt have anything like the dynamic range of live music. Try putting a mic near a floor tom, and watch how far the speaker cones move when the drummer hits it. This wont happen from a drum machine, because most of the high dynamics were taken out before the sound was stored in memory. If you have faith for it then Bose, Cellestion, Ohm, Sherman, all make very good small PA speakers. Electrovoice is another name that springs to mind. Any of these makes will sound and work very well. The rub is that they are expensive. A pair of Bose 802s with the equaliser will set you back almost 1500. So too the small electrovoice, but they are light weight and sound miles bigger than they look. Bose also make Bass bins which can be added to the 802s. This is the system we used at Roffey Place for out going missions, it works very well, even with drum mics in operation. The bins make such a difference to the overall sound, they add that bottom end that you can feel, and also they take the strain off the 802s by dealing with the deep bass end, leaving the 802s to cope quite happily with the low mid through to the hi top. Again this makes for a flexible system. If I only have speech to amplify then the bass bins stay in the van. A pair of 802s can easily fill a 1000 seat auditorium with speech, Im quite fond of them as you can probably tell.

A word about efficiency. The speaker makes I have mentioned are all very efficient. That means you get a lot of volume from a moderate amp. I ran the whole system on a single Quad 520, which delivers something like 250 watts into 4 ohms. I can honestly say that it has never been within 20db of clipping (thats distorting in lay speak). The output meters on the mixer have never moved far off the end stop as far as I can remember, and yet we have filled some pretty big auditoriums over the years.

If you cant get your faith up for a 1500 pair of speakers then there are many makes of PA speaker that are perfectly adequate, but you will have to compromise somewhere along the way. Lets look at some cheaper options. You should be able to find a 12 inch and horn set up for under 150. Laney, Custom Sound, Peavey, are just some of the better known makes that spring to mind. All of these companies make a whole range of PA speakers from very small to quite huge, you only need to look.

MONITORS.

One of the main problems for music groups is hearing yourself above the overall din. This is more of a problem for those instruments that are plugged direct into the PA and dont have an amp of their own. There are two ways to tackle this one. One is to have wedge monitors on the floor near you, and the other is to have a small personal monitor on a stand quite close to you. Drummers can use headphones. The best approach for the church band is to use the small monitors. These dont put out as much volume as the wedges, and also they dont create as much of a mushy bass sound on the platform. They are small and easy to carry.

It is quite a common thing to be able to turn the main PA off and still hear everything loud and clear from the monitors, and I mean from out front. The small monitors get around this noise problem to a great extent. The lack of volume from them is made up for by being closer to them, as is the lack of bass. They are worth trying out, and it will make life easier for the sound operator. You can have more of them about the place too and get a better distribution. Try Canford Audio BA400s, these are all metal cased 100 watt speakers that sound miles bigger than they look. They are very small (6x4x4) and weigh hardly anything. They can be mounted on mic stands with an optional adaptor which costs 8!, The speakers themselves cost around 37 each. I've used them, and I was quite pleased with them.

Monitors for the drummer are always a problem because of the noise level of the kit. Headphones are by far the best answer to this if you can get him to wear them. The achievable level in headphones is sufficient to blow the head off even the most deaf drummer without offending anyone else at all. Perfect answer. Monitors can be a menace, or is it the people who constantly want them louder who are the menace? I dont know why it is, but for years singers can sing without monitors at all without complaint. Then they get monitors and suddenly they cant hear themselves! Monitors are not there for your personal enjoyment, or to have your voice so loud that your ego is satisfied. They are there for you to be able to keep in tune, and to make sure you hear the leader so you know when and where to start and stop.

Wedge monitors tend to add boominess to the overall sound if they are too loud. Try standing the wedges on pads. This may help to reduce the boom, especially if the platform is made of wood with a gap under.

Speaker cabs are very easy to make and again Maplins sell everything you need to build your own.

WIRING UP.

In workshop 1 I talked about 2 speaker cabs on one amp. I think a little re-cap is in order here. Most amps will run into loads of between 4 and 16 ohms, usually 8.

Valve amps are less sensitive to mismatch in this area than transistor amps, but they will always work best if they are properly matched to the speakers. Transistor amps will be damaged if they are run into less than their stated minimum load, but will simply loose power if you run them into too high a load. Look at the back of the amp to see what the stated optimum load is. Almost certainly it will be between 4 and 8 ohms. Dont exceed the limit at either end and the amp will work satisfactorily. Less than 4 ohms will cause the amp to over heat drastically, more than 8 will loose power and you will be dissatisfied with the sound. If you are using 2 cabs with your guitar/bass amp and it sounds a bit naf then its probably mismatched. Valve amps have an output transformer that needs to be set to the right impedance, but transistor amps dont have this buffer, and so are much more sensitive to mismatch. The formula for working out speaker impedance is quite simple. Speakers wired in parallel half the impedance, and speakers wired in series add together the impedance. So two 8ohm cabs wired in parallel will result in 4ohms, and if wired in series they will result in 16ohms.

The simplest way to check your speakers is to buy a multi meter with a one ohm setting and measure the resistance on the meter. If your amp has two speaker sockets on the back they will be wired in parallel, this also applies to speaker cabs with two sockets fitted. Wiring speakers in series is only done, as a rule, inside speaker cabs. An example of this would be the 4x12 cab, which has four 8ohm speakers inside and an overall impedance of 8ohms.

This kind of wiring is called series parallel. The bottom two speakers are wired in series with each other giving 16ohms, as are the top two, also giving 16ohms. The two pairs are then wired in parallel thus halving the two sets of 16ohms to 8.

A point to remember is that speakers wired in series add together the wattage of each speaker, so two 100 watt speakers in series = 200 watts. This is not so for speakers wired in parallel, in this case the wattage remains the same.

If you dont need two great big cabs, its better to buy four smaller ones, why? Well you can get much better distribution of sound from four cabs than from two.

SOUND DIRECTIVETY.

Always remember that sound coming from a speaker cab tends to be directional. That is, sound will be loudest and have the most treble in a straight line from the cab, and as you move away from this straight line to the side, or over or under the speaker, the volume decreases as does the treble. So much so that at 90o the sound is almost unrecognisable.

This is where four cabs are better than two, you can angle the cabs in such a way that the sound is fairly even all round.

Let me pass on a tip here to those who may need to use big bass bins. Bass has the ability to travel in every direction equally regardless of which way the cab is facing, so the problem mentioned above doesnt really apply to bass. This opens up the option for you to have one pair of bass speakers and several pairs of smaller speakers for better distribution of the mid and top frequencies. With this in mind you can tailor the system to your own personal needs.

This kind of setup will be miles more efficient if it is BI amped. What that means is to use separate amps for bass and mid/top. This requires the use of a crossover of some sort to split the frequencies. There is a cheat way of doing this that saves quite a bit of money as it only uses one amp, but is not quite as effective. In this system the crossover we talked about is known as passive, and is fitted inside the bass bin. Some makes of bass bin provide this kind of crossover, Bose and Peavey do to my knowledge, but there may well be others. What happens is that you simply feed the output from the power amp into the bass bin, and take a modified output supplied by the bass bin up to the mid/top speaker. The crossover in the bass bin sucks the deep bass out of the signal and feeds the remaining frequencies out to the smaller speakers. The bass bin receives the full frequency range but can only deliver the very deep bass. So you have a cheap way of splitting the signal.

The small speakers will work quite happily at volume levels way beyond their normal limits, because of the lack of bass being sent to them. Having said that, beware of the horns at the top end before you get carried away. This type of passive split system does work very well and is worthy of investigation. You can also add this type of bass bin to your existing system and greatly enhance the performance of even a very modest PA without too much expense.

So thats the cheap way what about the proper way? Well this involves an electronic crossover placed before the power amps. It splits the signal from the mixer before it is amplified and is very much more efficient than the passive variety. Usually there is some degree of control built into the crossover so that the whole system becomes very tuneable. This can be of immense importance to the travelling band as it allows for various room acoustics to be dealt with quite quickly. It requires two power amps as stated and so the overall wattage goes up which makes for cleaner sound. There are other advantages too, but they are beyond this type of teaching and lie more in the realm of acoustics. Suffice to say that a BI amped system delivers a lot of high quality sound for the size of the system. The electronic crossover and the extra power amp add to the expense but are well worth it in the end.

CABLES.

Cables can be a bit of a mystery to church folk. There are three main types in use (4 if you count mains cable). They are:

1. Speaker cable (2 core mains cable).
2. Guitar cable (single core screened).
3. Mic cable (twin core screened).

Speaker cable is totally different to mic cable, and the two are not interchangeable. Speaker cable is akin to two core mains cable and should be rated at at least 5 amps if not twice this amount. Always make your speaker leads equal in length.

Mic cable is not designed to carry high voltage and should never be used as speaker cable. On the other hand speaker cable is not screened and will hum or buzz away quite nicely if it is used as mic cable, so make sure you use the right cable for the job. Guitars use single core screened cable, and mics usually use two core screened cable (single core for unbalanced mics).

Mic cable is usually of a much better quality than guitar cable due to the need to reduce the amount on noise generated by the cable itself. Mics are amplified to a much greater extent than guitars, because the mic doesnt generate the same output level as a guitar. Its this extra amplification that causes the problem, as even tiny stray signals picked up by the cable and sent to the amplifier will be heard.

CABLE PHASE.

Phase is probably a new term to most of you, but it simply means getting the positive and negative wires the right way round at each end of the cable. A speaker will still work if the cable is out of phase, but there will be audio problems that would take me a month of Sundays to explain, so I wont. Lets just agree to check all cables for phase and have done with it. If you use 2 core mains cable for your speakers, then use the brown wire for positive and the blue for negative. Brown would be connected to the centre or tip connection on a standard jack plug, and blue to the outer or sleeve connection. If it is the same at both ends then all is well. If not then change the end where the blue is connected to the centre connection. For XLR plugs, the blue always goes to pin 1, and the brown to either pin 2 or 3, but usually 2. Some manufacturers wire the brown to both 1 and 2.

This kind of wiring fault is very common in home made/repaired cables. So are shorts, so please always check your work thoroughly. A short on a speaker cable can have a devastating effect on your amplifier. The quick way to check your cables is to use the affore mentioned multi meter. Set it to any resistance setting and touch the probes on the tips of both jacks. If you get a reading then all is well, phase wise.

Set the meter to its highest resistance range and touch the tip and sleeve on just one jack with the meters probes. If you get a reading of any sort then there is a short on the lead. A reading of Zero ohms means a dead short, and the lead wont work at all. But a reading of more than a few kohms means that the insulation is breaking down, and to some degree the lead will work, but the signal will be degraded to some extent. This is quite common in cheaper guitar cable. Always buy good leads.

Any heat in the speaker wires is a sign of cable in trouble. Either from too much current being passed through it, ie using 2 amp lighting flex on the output of a 500 watt amp, or a partial short caused by a break down in the cables insulation. This second fault normally follows the first one.

It is much harder to get a guitar lead out of phase because the screen is usually wound around the inner core, so you have a visual representation of how to wire it. If You should get it wrong the guitar will still work, but there will be an almighty buzz from the amp. Balanced mics cant really be wired out of phase so there is no problem here. Unbalanced mics will not work at all if they are wired out of phase, because the wire that is connected to ground will simply short the whole thing out.

To check the phase of your speakers take a 9 volt battery, a PP3 is adequate. Plug a known to be in phase speaker lead into the cabinet, and touch the terminals on the battery with the jack on the other end of the cable, make sure you touch the positive terminal with the tip of the jack.

Watch the speaker cones! They (or it) should jump forwards. If not then there is a problem on the inside of the cabinet. It is fairly common for speakers to have their voice coil wound the wrong way. Most speakers have a mark to show which is the in phase connection. If need be connect the battery directly to the speaker, again positive terminal to the marked connection and see which way it moves. If it pulls back then the mark is on the wrong tag on the speaker. If the speaker goes forward then the jack socket on the back of the cab is wired wrong.

If you have more than one speaker in the cab then all speakers should go in the same direction, if not check each speaker in turn to find out which is right, and swap the wires on any that are wrong. Youll find that after doing this that your speakers will come to life and sound cleaner.

THE DI BOX.

It is quite common for acoustic guitars and sometimes keyboards to be plugged straight into the PA without having any amp of their own. Both Carol and I do this when we go out on mission. It saves carrying extra amps with all the attendant wires and things. You cannot plug a guitar straight into a mic input on the PA, even if the input socket on the PA is a jack. The reason is that there will be a horrible mismatch resulting in loads of distortion, and a naf sound. Its even worse with a keyboard, and may possibly damage the input stage of the mixer. The answer is to use a direct injection box, DI for short. This will match the high impedance output of the instrument to the low impedance input of the PA. It will also balance the signal, which is a good thing, especially if you have a long multicore cable between the platform and the mixer. The same is true of the line or preamp out of your amplifier. As a rule, if the socket on the amp is a jack it needs to be plugged into a DI box. If the socket is an XLR then the signal from it is probably already low-z and balanced. A lot of the better makes of amp provide this facility these days. Never connect the speaker output from your amp to anything other than a speaker cab. Some DI boxes will permit this type of connection but not all of them. If you are in any doubt dont risk it, the mixer will die if you make a mistake with this one.

THE MULTICORE.

The best way to setup a PA system is to have the mixer somewhere out in the auditorium. Not necessarily at the back, the middle is usually more accurate for sound especially in the bass region. Remember what I said about bass travelling in every direction equally, well it also travels further than top or mid and so will be more prominent at the back than the top or mid frequencies which get absorbed by almost everything in the room, including bodies which are excellent sound absorbers. The multicore is a vital piece of PA gear as it is the umbilical chord between the mixer and the platform, and all signals travel along it both to and from the mixer. If theres one area that is prone to problems its the multicore and its connections.

The multi pin plugs on either end can be easily damaged so take special care when rolling out or winding back the multicore. Take it slowly. As the cable whips along the floor the pins in the plugs can be bent, or connections strained, and believe me the last thing you need is to have to go into one of those multi pin plugs with a soldering iron. Watch out for the stage box too, as this can suffer bent pin damage very easily whilst packed into a wire case. Intermittent connections are a right pain to find and fix, and a right pain when something goes off for no apparent reason. If the power amp feeds lose the earth connection you can look out for some real loud cracks and bangs, and probably say goodbye to a few speakers along the way.

Only use twin screen cable for mics, and always run balanced. Never buy any other type of multicore, it may be cheaper but its asking for trouble. Always use individually screened cable. This will lessen the chance of signal interaction within the multicore. What can happen with cheaper types of multicore is, the line level output from the mixer can be induced into the mic lines and fed back into the mixer resulting in a whining sound or even worse a squeal. If this happens there is no real cure, only to turn the mixer down, which may be no cure at all. This wont happen with individually screened balanced cables.

MAINS CABLES.

Most people will know how to wire a mains plug, even a plug board with four or more sockets on. But how many of you know about three phase supplies, and how does it relate to the travelling band? Well in your average church you wont find a three phase mains supply so there wont be any problem, but on occasion you may play in a theatre or some other public building where there will be a three phase setup. Just what is a three phase supply? Well it is in simple terms three separate (well almost) supplies fed to the building. It is done so that within the building the mains supply can be zoned as it were. IE one phase for the lights, one for the stage, and one for the auditorium. This cuts down on interference, and is generally a good thing, but there is a problem, its called getting across phases. It happens when something is plugged into one phase, say a guitar amp on the stage, and is connected to something plugged into another phase, say the mixer out in the auditorium. Should there be a fault at some point there can be 440! volts flying round the system (each phase adds to the other hence 2x220=440 volts). This is not very good for anything that gets on the end of it, amps, mixers, singers and the like, so precautions need to be taken. Make it a rule to take all your power from the same point. That is, on stage. Power for amps, keyboards etc and the mixer. You will need a long mains lead to reach out to the mixer, but its worth it to be safe. Never take chances with mains cables. Check them regularly. Tighten loose screws etc. Never take the earth off anything mains wise, if you have earth loop troubles then take the earth off one of the signal wires, or connect the earth via a small resistor.

Make special leads and carry them with you, to connect several pieces of gear together, this is far safer than taking earths off. Always check inside mains plugs for good tight connections. If the earth wire comes off inside the plug and manages to touch the live wire you are as good as dead. Not very funny that. Loose screw connections inside the plug can SPARK which generates heat in the cable, and also loud barking sounds from your speakers.

Make frequent checks for any mains cables that are getting warm. This is a sign of overload and probable cable failure. Replace any that you find immediately. Always use higher rated cable than the load requires, this will avoid having to replace cable in the future. You will probably add to your equipment from time to time, and each addition will add to your mains power load. Think ahead.

Also, use the correct size of fuse in the plugs. Dont use 13 amp where 3 amp is required. This will save cables from almighty surges of power should a fault occur. A fuse will usually blow at 3 times its rating. Never fit a bigger fuse to your amps than stated, if the fuse blows there is something wrong.

MIXERS.

This item of PA gear could take up a whole series on its own, so for the purposes of this course I will keep it simple. I have to assume that you the interested reader already have some knowledge of mixers in order to avoid absolute basics. What mixers are about is getting the best signal to noise ratio that can be achieved. This is done by optimising the channel input gain called mic gain or level, so that the channel is driven to a safe amount below clipping but not less than this, about 10 db below should be somewhere near. Remember that EQ controls will add to or subtract from the overall channel level quite dramatically. Excessive use of the EQ can overload the channel quite easily. If there is an led overload indicator fitted keep an eye on it. If it blips on then back off the channel gain a bit until it remains off. If your mixer has a PFL (pre fade listen) system with a meter then press the pfl switch on the channel you are working on, and monitor the level going through the channel on the pfl meter. Anything under the red is totally safe. Having set the individual channel gain controls you need to set the channel faders. If they are calibrated in dbs and have a +10 db setting then set the fader to 0db as a standard. If 0db is the maximum setting then set the faders around the -10db mark.

If there are only numbers 1 to 10 on the faders, then set the fader so as to allow you 30% of the faders travel upwards. You need to be able to boost as well as cut volume on each channel. The main output faders should be set to something similar. Always allow for boost. As a rule of thumb always have your power amps up full. Then use your output faders to control the overall volume. Dont let the channel faders get too far above the output faders, if this happens you may overload the mix amps. Within 10 db is fine. The mixers output meters should ideally read around the 0db mark for optimum noise figures, if you find that you cant get to this mark without taking the heads off all and sundry with volume, then back off the power amps and increase the output fader levels until you have a good average reading. This is only really practical on fixed systems, but you should be able to setup a good average on the mixer that will work in most venues.

If you can, always work in stereo. A stereo PA system will always fill a hall better than a mono one. Even just a small turn on the pan pots to the left or right will make a difference to the sound.

Try never to play a stereo tape back in mono. The azimuth of the tape machines playback head will, if not spot on, make the treble fizz. This generally spoils the sound quality.

EQUALISATION.

Tone controls is the lay speak for the EQ section on the mixer. EQ can be as simple as treble and bass, or as complex as 6 band parametric. Graphic equalisers can go as far as 31 bands for 3rd octave control, (that means you get 3 faders per octave).

Most PA mixers of affordable quality will be fitted with a 3 band equaliser. Top, mid, and bottom, or treble, middle, and bass, or Hf, MF, and LF, to give them their names.

The variance in makes will be in the middle band. Some makers give you only a fixed mid frequency control, and some give you a sweepable mid section. Always go for a swept mid if you can.

There are 2 main types of EQ around. The more common one found on PA mixers is called BELL, because of the shape of the control curves achievable with this kind of EQ.

The other is called parametric, and is much more flexible than the humble bell, but is more costly to make and so doesnt find its way into your average church PA system.

Bells with a swept mid are fairly common now so well take a look at these first.

The treble control (HF from now on) will be fixed somewhere between 8khz and 12khz, nearly always at 10khz (k= thousand). Its range of cut and boost will vary depending on the maker, but 10db would be the minimum, and 15db a maximum. The bass control (LF from now on) will be fixed somewhere between 30 and 180 hertz, but usually at 100Hz (hertz is another name for cycles per second). Cut and boost will always be the same as for the HF control.

The mid range control (MF from now on) is sweepable, meaning that the control is continuously variable over a range of frequencies from say 1khz, to 5khz. The frequency range covered is usually printed onto the mixer.

Also associated with this frequency sweeper is a cut and boost control. This control is capable of amplifying or attenuating whatever frequency has been selected on the sweep control, usually in the same order as the other controls.

PARAMETRIC EQ.

The word parametric has been attached to EQ systems for quite some time now. It started out in life as a specialised tool used only for really complex problem solving, but has since become so popular that some manufacturers now include it as standard mixer EQ.

So what is it ? I hear you ask, well in principal it is similar to our 3 band equaliser as just discussed, but with one almighty difference. For the purpose of this illustration Ill talk about the MF section of the EQ.

The sweepable control is the same and has the same range, but there is now an extra control. This extra control can take the form of a switch in the cheaper systems, or a rotary control in the more expensive ones.

What this control does is to alter the Q of the equaliser, simple isnt it ? In ordinary English what the control does is this.

Imagine the range of our sweepable control as a straight line, with the lower frequencies to the left, and the higher frequencies to the right.

Now imagine our sweeper control to be a pyramid, the base of our pyramid sits on the straight line the cut and boost control represents the height of the pyramid, if the control were set to minimum, or full attenuation, the pyramid would hang underneath our straight line, but for now well have it full up, clear so far? Right. As we rotate the sweeper control the pyramid moves from left to right and we hear the change in sound accordingly. This would be a pictorial representation of our MF sweep control, but with a parametric equaliser we have this extra control which gives us control over the width of the base of our pyramid, (ah so that's it).

The point of the pyramid is the actual frequency selected, and receives the most treatment but as you slide down the sides of the pyramid you affect a gradually wider chunk of the spectrum, so with an ordinary equaliser you affect by degree frequencies both above and below the selected frequency, and as the level decreases so the width or Q of the equaliser widens. That's where our extra control comes in, by narrowing the Q (base of the pyramid) we affect less of the frequencies surrounding the frequency we have selected, and so have more precise control over what we alter. The Q control can make our pyramid into a pencil.

If we make the Q as narrow as we can, what we end up with is a NOTCH FILTER. This allows very precise control over the frequencies that we alter, most useful for getting rid of amplifier buzz, or any other type of narrow band noise that may get into our gear.

Some DEESSERS work on this principal, having a fixed frequency of 5khz, but the more sophisticated ones have the ability to sweep through the frequency spectrum until they find the offending S before NOTCHING it down in level.

A GOLDEN RULE.

The reason EQ was invented to correct duff sound. Duff sound usually comes from there being too much of something rather than too little. So golden rule number 1, always cut before you boost.
If something sounds mushy dont just add top, take away the mush. Find it on the EQ and turn it down. Thats where sweep EQ comes in so handy.

Remember that taking away some frequencies makes the others sound louder, so to make a bass drum brighter lose some mid. Around 500hz should do it. Try it for yourself.

The same principal can be applied to the voice. Try losing a few dbs at 1.5khz for that nasty nasal sound that some people have. Or 5khz for those who suffer from false teeth whistles!

Remember that excessive boost in the EQ can make your amps and speakers work very hard, especially in the bass.

METERS.

All mixers are fitted with some kind of metering, whether it is the VU type or the PEAK reading type, or just a simple peak indicating LED. The meter is there to be used as a guide (only) as to how much level is passing through the mixer.

In general the peak reading type is better, as it shows you the loudest signals as they pass through, and yet all the expensive tape machines have VU meters fitted, why is this ? Well VU stands for volume unit, and the dial is calibrated in dbs, or decibels.

What you really get from a vu meter is an average level reading. It could be said that this type of meter is better for recording as it allows you to record at a higher average level than a PPM (peak programme meter), but this is only true at low and midrange frequencies, at hi frequencies, or with sounds of short duration a vu meter simply cant react fast enough to give you a useful reading.

As a rule of thumb remember that a 6 db change in level as shown on the meter indicates either twice as loud +6db or half as loud - 6db. It will not sound this way to your ears, but we are dealing with voltage, not sound.

Some useful points: When recording a short duration sound such as a bass drum, the maximum reading on a vu meter would be around - 10db. Any higher than this would result in some degree of distortion, and also a noticeable loss of treble, and punch. To your ears the sound would become soggy and dull.

Another notorious example would be tambourine. A maximum vu setting for this would be just above the -20 mark. Two factors come into play here: 1 is the vu meters inability to react quickly enough to short duration sounds, and, 2 the mainly high frequency content of the sound of the tambourine.

Both of these factors are weak points in the vu meters character.

With regard to a bass drum the vu meter will register approx 12 db lower than the actual level.

Remember this when making your recordings. The figure would be 15db or so for very high frequency sounds such as small bells, triangle, tambourine etc.

Also note that the vu meter has been designed to mimic the human ear, so what you are seeing on the vu is approximately what you are hearing.

Your ears are also less sensitive to short duration sounds, which is a blessing, else a hand clap which is the loudest sound a human body can create, would take your head off!

PHANTOM POWER.

Some of the better mixers will have phantom power available. Phantom as it is called is 48 volts DC applied to the mic socket on the mixer. It enables you to use the more professional condenser mics. These mics will give you a much better frequency response and miles better noise figures. They have built in pre amps and so have a much higher output level. This means that your channel gain can be set a lot lower. Also as you are sending 48volts down the multicore there is virtually no chance of any stray pickup in the cable. Non phantom dynamic mics wont come to any harm by having 48 volts sent to them, but some mics like electret condensers, may object by giving out some noise or other.

Keyboards could be damaged if connected directly to phantom power, guitars also, so always use a DI box for any direct instrument connections.

DI boxes are immune, in fact some of the more expensive DI boxes use this power instead of batteries. Some DI boxes are passive, that is, they dont need any power to operate. As a rule, if you dont need the phantom switch it off.

THE SOUND MAN/WOMAN.

To be honest in my opinion the sound man, or woman is the most vital link in the chain. This single body can make or totally break the whole shabang. So I say to those who feel they would like to have a go, make sure you are called of God before you dare to take into your hands this very responsible job. You, dear sound person need to be in the Spirit as much as anyone in the meeting. You need to have a sensitivity to what is happening on the platform. You need to keep your eyes OPEN! If someone has a word to bring and the mic isnt on and youve got your eyes shut, the moment will be lost, and that could be quite serious for someone. No-one likes looking like a fish, talking into a dead mic. Be aware, be ready, be attentive, its important. If you cant hack it then do everyone a favour and quit. The chances are that no one else will come along for the job whilst you are stood there. Now thats tough talk, but I want to give you a sense of how important the role of PA operator really is. There are too many cowboys who just havent got a clue, they kill peoples ears with treble, or have it so quiet that everyone has to strain to catch whats being said. Or they are guitar fanatics so all you can hear is guitar, even though the keyboard is playing some tremendous brass lines!!!

Contrary to popular belief being a HIFI enthusiast is not what it takes to make a sound man. Knowing about woofers and tweeters is one thing, but being able to operate a mixer is something else. It takes skill and a lot of experience to master.

So, how do you learn? You learn by doing. You practice in the same way a musician practices, in your front room. Take some gear home with you and set it up, use headphones if the neighbours are chicken. Play tapes through the mixer mess about with the controls and reverb effects. Find out what every control does, learn how to use it and abuse it, some of the better effects have come about by breaking the rules, try it. If you cant take gear home then sit in the church and practice. No matter whats on the telly!

Practice with the musicians. Become part of them, and whilst theyre practising on stage you be out front practising your effects, its the only way to become proficient.

A FEW HINTS.

Make it a rule to plug speaker leads in first, signal leads in second, and mains leads in third.

switch the mixer and all the effects/tape machines on first, then the power amps.

Reverse the order of the above for switching off and unplugging. If you get into this habit you will avoid accidental speaker and amplifier damage.

May the Lord inspire you as you learn.

THE SOUND BOOTH.

I must admit this makes me smile. Churches seem to think that shutting the PA mixer and operator away in a little cupboard is the height of sophistication. Why would you want to put the person whos doing the PA balance into a different room than the PA that hes trying to balance? Balance can only be achieved by a skilled pair of ears, and the said ears need to be able to HEAR what is happening in the room where the PA speakers are. How else can the right amount of volume or treble or bass be set? Close the ears in a box, and the ears will automatically make adjustments according to the box! Crazy! But what about the recording you ask? Studios have the mixer in a separate room!

Well studios are not PA systems, and they record music. In church you only really record the sermon, which is one mic direct to a cassette machine. Hardly hi tech is it, and doesnt require any quality reference monitoring as it does in the studio. At least nothing that a pair of headphones cant handle. If you are thinking about having a booth, DONT!

There is one good thing you can say about sound booths. You can lock them, and keep thieving, unwanted, chocolate covered fingers off your gear. Amen.

RECORDING

If you want to record from the mixer try to take a feed that wont be affected by any movements you make on the main output faders. Some mixers have a dedicated tape output, if so then great, but if not you can always use an aux send to feed the tape, or even the headphone socket. If your mixer has neither of these then you can make up a split lead by soldering a 10k resistor to the centre connection of the output jack (2 for stereo), and feed the tape machine via the resistors.

The resistors isolate the mixer output from the tape machine so that any adjustments to the tapes record level wont affect the PA volume, dont forget the earth connection. Let the meters go into the red on the tape machine. Dont be afraid to record at high levels, most tape machines are setup at the factory in such a way that it is almost impossible to distort a recording so long as the meters dont stay at the end stops for any length of time. Short duration spikes that distort a bit are much less of a pain than constant tape noise.

SOUND EFFECTS.

In the concert type of situation sound effects can be very effective when used correctly. The mistake many people make is to use too much or none. I think there is probably nothing worse than someone's voice being drenched in reverb, especially the cheap spring type of reverb that is fitted to amps as a gimmick. I call it the club singer syndrome yuk. There are many useful effects to consider for PA use so Ill try to explain a few in this section. Reverb is the most common, and can be used to great effect even in rooms that already have some natural reverb built in. Points to remember with reverb 1 the amount, and 2 the length. If the room/hall has a long reverb, then you use a short one. If the room/hall has a short zingy reverb then you use a longer smoother one. Dont try to fight the natural reverb, rather use it to your advantage.

The variety is quite wide. So how about echo. Again the amount is important, and equally so the length of delay. Try timed echo. How about mixing the two. These kind of effects can make your band sound mega if they are done right.

COMPRESSION.

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 dont have a clue about them. However the compressor can be a very useful tool to have in the PA system. 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 youve got it, it can really work for you. More later

NOISE GATES.

The noise gate is another useful tool left over from the recording studio. They are great for cleaning up noisy guitar pedals and sloppy bass players. You see its worth getting to know about these things and incorporating them into your own setup. Every little helps to make you sound just that bit better. More later.

MICING UP THE KIT.

It may not seem necessary to mic up such a loud sound source as a drum kit, especially in church, but there is one area where it can be a positive advantage to do this. Have you noticed at concerts how the sound seems to be delayed compared to what you see, especially if you are near the back? This is because of the relatively slow speed of sound, (750mph). Light, and therefore vision, travels at 186,000mps, yes miles per second, which is knocking on a bit. Thats why you have this effect. Now in the church, especially the bigger church, you can have quite a considerable delay from front to back, and this just adds to the confusion when people are trying to clap or sing along. The best answer to this problem is to mic up the drums so that they can be fed into the auditorium at the same start point in time as the other instruments and voices IE through the PA speakers. If you are using a PA system that has speakers around the room then this is doubly effective. The drums dont need to be loud, just there. People like to lock onto a good rhythm.

PLACING THE MICS.

Putting the mics in the right place is half the battle when it comes to recording, or PA for that matter. The other half is having the right mics. You can only work with the sound that the mic picks up, so if the mic is miss placed in relation to the sound source, you cannot expect the best results.

As a rule, if you are having to use a lot of EQ to get the sound right, then chances are the mic needs to be moved, or changed.

Any mic picks up best from front dead centre, even omni mics do this. Maximum treble and bass are captured at this position. Turning the mic sideways will lessen the mics sensitivity to both treble and bass frequencies to some extent, depending on how good the mic is, having the effect of boosting the mid range.

The best way to find the right position for the mic is to use your ears. Move around the sound source till you hear the best sound, and place the mic at that point.

BASS DRUM

You cant very well stick your head inside a bass drum neither should you try it, so the following will help.

1. Think of the sound as a whole, that is, start, and decay. This is important as these both come from different places. The sound start or attack is the beater hitting the front head. The decay is both the head and shell vibrating after the hit.

2. Know what you want the mic to hear. Put the mic close to the head for more attack/click, and back from the head for more thud/bass.

3. Always start at dead centre. Top to bottom, left to right, front to back. This is the most common place for a good bass drum sound.

mics to use: AKG D12 - Electrovoice RE20 - Shure SM57. AKG now do a replacement for the D12, and Beyer also do a very interesting bass drum mic. You should find out about these.

SNARE DRUM

The mic should go under the Hi Hat (if possible) about one and a half inches above the head, and pointing down towards the middle of the head. Have the head of the mic an inch or so over the rim of the drum. Try to keep it away from the drummers sticks!

Mics to use: Sure SM57 - AKG 451 with CK1 capsule and -20db pad.

HI HAT

Never point the mic at the opening of the hat as this will get a blast of wind each time the hat closes. Again think of the start and end of the sound. For best definition place the mic over the hi hat so that it picks up the stick hitting the metal. If you want a heavier sound then move the mic more towards the edge of the hi hat, but pointing down from above it.

Mics to use: AKG C451 EB - Senheiser 421 or 441, or almost any reasonable mic.

TOMS

Place the mics over the toms for the best brightness. Use the same rules as for the snare drum. Over the rim, pointing down towards the centre of the head. Really watch for the sticks. Use a mic for each tom if you can, or one between two.

You can mic toms from below if you want, the sound is totally different from underneath, a more boxy middley sound can be obtained. There is less cymbal spill as well which may be useful. If there are no bottom heads on the toms you can actually put the mics up inside, this makes for better separation, but the EQ needed may be excessive.

Mics to use: Senheiser 421 - Electrovoice RE20 - Sure SM57.

CYMBALS

Cymbal or overhead mics should be about three feet above the cymbals, not to the side. This avoids the vibrato effect as the cymbal swings up and down. Aim the mic at the centre of the cymbal if you can. You can usually get all the cymbals with just two mics, more than this starts to bring phase troubles.

Mics to use: AKG C451 EB - AKG C112 or 114 - or any really good vocal mic Shure SM57/58.

AMBIENCE MICS

In the studio, using ambience mics on the kit has become common place. If your budget can run to such things then the U87 is by far the best mic for this job. Set your ambience mics as high as you can get them, but watch out for the lights! Some lamps make a whining noise when on and the 87 will pick this up as sweet as you like. If you cant get very much height then go for distance. Remember that sound travels at a constant speed, so setting one mic further away than the other one will introduce time delay effects into the ambience. This may well enhance the stereo effect better than simply placing one mic left and the other right. Experiment with this one.

GUITAR AND BASS AMPS

Put the mic right in the middle of the speaker and about 4 inches back from the cabinet for the brightest sound. Or anywhere between the centre and the edge of the speaker for a more meaty sound. Be aware that bass guitar speakers can move a lot of air. Consider the poor mic when placing it.

Mics to use: Just about anything good.

ACOUSTIC GUITARS

The needs of the acoustic guitar mic differ greatly between the studio and the live PA. For the PA the mic has to be as close as possible to the guitar, and usually pointing towards the sound hole. Try to catch the plectrum noise if you can as this will add definition to the sound of the guitar in the PA mix. You will need to role off some bass at the mixer to lose the boominess. In the studio the mic is used to just pickup the guitar, unlike in the live PA situation where the mic has to not pickup everything else. So the position of the mic can be drastically different.

I recommend you use the ear trick to find your own best sound, but by far the best that Ive found is to place the mic low down pointing at the body of the guitar behind the bridge.

Mics to use: AKG C451 EB. For PA use any good vocal mic, Shure SM58.

BRASS AND WOODWIND INSTRUMENTS

Trumpet is a notorious instrument for giving mics trouble. The SPL, or sound pressure level at the bell can reach quite staggering proportions. Also, because the bell is always pointing away from the player, the player will often be unaware of the harshness and spittyness of the sound actually being produced. This leads to frequent problems between engineers and players when a recording is played back. Experienced players will often be seen playing against a wall or some other reflective surface, now you know why. Trombone is just as bad, as is French Horn, Tuba etc

In placing the mic you must consider the affore mentioned problems, and balance that with the noise if any from the mechanical valves. Add to this the need to poke the mic through or around a music stand at what is usually a moving player and you get an idea of what trouble really is. The only mic that can be guaranteed to work every time is the Electrovoice RE20. These are very expensive dynamic mics and look horrible, but they work a treat with any loud sound source, even bass drums.

Try not to have two brass instruments blowing into the same mic, as this will create what is known as harmonic distortion, which sounds terrible. Ive known this distortion create a third harmony when only two instruments were playing, no it didnt sound good! This principal applies to any loud sound source, such as vocals. It is not so bad when both players are playing the same note, it happens most when two different notes are being played simultaneously. Mics can be placed anywhere from one to four or five feet away from a brass instrument with no problem.

Woodwind instruments on the other hand are much more gentle and not so loud, but they present a different set of problems altogether. The principal used by woodwind instruments to create sound is to split the breath of the player by some means. A reed in most cases, two reeds in some, and metal mouth pieces in others. The body of the instrument is used for the pitch and tone of the sound like any other instrument, but woodwind instruments have lots of keys and little pads and springs, and almost everything makes a noise. The instrument itself dictates the positioning of the mic. A noisy key means the mic has to go at the bell end. Silent keys mean the mic can be placed more along the body where the tone is usually sweeter. Almost any decent mic will work with woodwind so theres some compensation for you.

PIANO

Micing up a grand piano is the easiest thing in the world. You simply lift the lid, get the player to play, stick your head inside the piano and listen. Your ears will automatically find the right places to put the mics. Put the mics where the good sound is coming from. Always mic up a grand piano in stereo, in mono there is no life, but in stereo the sound comes to life. This is not true of upright pianos which sound flat no matter what you do! With uprights you dont put the mic inside, because all you get is the action noise, no, put the mic around the back. Your ears will tell you where.

USING TWO MICS TOGETHER.

Sometimes you can get a more realistic sound from the guitar amp if you use two mics. If the amp has an open back try placing a mic at the rear. The sound from the rear will be somewhat dark and heavy, but when mixed with the sound from the front mic it can add that certain something that cant be found from just the front mic alone.

You can also use two mics on the front of the amp, one set further away than the other. This introduces a kind of variable phase control. The distance between the mics is the control. Try this one and experiment with the second mics distance, its a very useful technique.

An area of recording where two mics can be used together is vocals. Many vocalists like to either hold or HOG the mic whilst they are singing. This can be a right pain for the sound engineer. The cure is to set up a dummy mic that the singer can abuse whilst the recording is taken from the second mic. This works even better if the singer doesnt know what is going on!

One last area where two mics may be of use is on the snare drum. One over and one under. The sound from under the snare drum is extremely nasty and harsh but this can be rolled out with EQ to leave a fizzy top end that will brighten the snare drum sound. The trick is to only have as much as you need of the under head mic. Too much sounds yuky.

CASSETTES.

Always use the Dolby noise reduction system fitted to your cassette deck it has two advantages.

1. it keeps the noise down to a bearable level

2. it can be switched off whilst playing back the recording to gain more treble. This can be very useful.

Chrome tape is better than ferric. It gives a better signal to noise ratio, better top end, and a higher record level without distortion. Chrome tapes cost a bit more but they are worth the extra. Use them to record your masters, you can then use ferric to make your saleable copies.

Make sure you set the machine to the right setting for the type of tape you are using. The normal position is for ferric tape, the cro2 position is for chrome. You cannot make a satisfactory recording if the machine is set to the wrong position. However, once you have made the recording you can then use the wrong position to your advantage. Playing a chrome tape back on the normal position will enhance the top end quite nicely, and playing a ferric tape back on the chrome position will reduce excessive tape noise (and top end).

Some machines have an automatic tape select system, this prevents you from making any mistakes, but also prevents you from abusing the system to your own advantage. If you are in doubt as to which tape type you have, you can always tell by the tapes colour. Ferric tape is always a tan brown, whilst chrome tape is always a blue black. Metal tape is always a grey black.

If you are thinking of buying a cassette machine for use in the church, I would recommend an auto reverse deck. In particular you can get a twin auto reverse deck that serves a dual purpose. You can simultaneously record 2 cassettes at once, each with auto reverse, and you can also use it to make good copies also with auto reverse. We always record two cassettes at the same time. One a C60, and the other a C90. You never know just how long the sermon is going to be. The auto reverse facility gets rid of the need to turn the tape over at the end of side one. This saves having a gap in the sermon, and also prevents the clatter created by ejecting and reloading the cassette. Techniques do a nice one, so do a company called Kenwood. Both sell for about 250 which is very reasonable for what is in effect two cassette decks in one box. Denon also make very professional machines, but they do cost a bit more.

Lastly a look at fast copiers. These are very expensive, and quite troublesome beasties. I've used Sonny machines and these are quite simply the best. They run at 16 times normal speed so they are fast, and they are certainly the most reliable. They come in two forms, master and slave. The master has three copy bays, and the slave has four copy bays. The master can run any amount of slaves, they simply daisy chain together, and the master can be switched to become a slave should you so desire. The only problem is the price, some 3500 each!!! This is for stereo versions, but if you are only copying speech you can use mono, which is around 1000 cheaper.

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 dont 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 2dbs 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 6dbs to allow just 1db through. Ratios 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. Its 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.

ABOUT 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, Ill 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 thats it. Its 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 its 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 gates 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 its 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.

VOLUME.

A quick word about volume, as you will inevitably come across this problem in your travels. How loud do you have the PA? Who controls the volume? What do you do when they tell you to turn down? What about the old folks?

These are all good questions, and Ive come across all of these at various times, so, just how loud do you have the PA system?

You have it loud enough to fill the room that you are in. By fill I mean that the music should have power and excitement, this takes just the right amount of volume. Too little and things get weak, too much and you lose the clarity. A room has a volume limit, going beyond this limit causes the sound to merge into one messy mass. Dont go this loud as it is counter productive and usually painful to the listeners.

You have it loud enough so that vocals and speech etc can be heard over the level of the musicians. If the band are playing so loud that the PA is struggling to match it, then the band MUST turn down.

WHO CONTROLS THE VOLUME?

The person appointed to do the job! No one should be appointed to do a job in the church unless the leadership are confident that the person is both able to do it, and more importantly anointed to do it.

If as a music team you have a sound man (or woman) who does the mixing for you, then you must put your trust in that persons ability and anointing to do the job for you, or get someone else.

Having found such a person, if you travel to another church and they moan about the volume, you must support your sound mixer, and if need be pack up and go home! Be careful as a band not to play to loudly. Have a system of signs by which the person sat at the mixer can tell you if anything needs to be altered on the platform, and be obedient to the signs. Even if YOU dont agree with them. Remember, sound is funny stuff, and changes drastically as it travels through the air. What you hear on the platform will not be the same as what they hear off the platform.

(SPIRITUAL STUFF)

There are those people who seek to control. The church is full of them, and music/praise seems to attract them like moths to a flame. They will flap and panic about noise before the system is even switched on, you know the kind of thing, my mother is coming shes 87 you wont have it too loud will you!. Another favourite one is Gods not deaf you know, and the ever popular one they cant hear themselves singing. Praise the Lord I always reply to that one. Dont ever listen to any of this tripe.

During the praise time they moan to their neighbour, usually with faces like thunder, (fingers in ears etc) and why? Because its not the way THEY want it, or its not the way they have always done it. God is not in the business of following us!

I say to you in the name of Jesus, dont ever let any such person influence you in any way with regard to the music you play, or the volume you play it at. They will try to drag you down to their level.
Never let Gods praise be defiled, or influenced, or controlled by anyone who isnt CALLED, ANOINTED, and APPOINTED to do the job. Remember there is an enemy at work and he hates the praises of God, stand firm against any attempt at interference. Only the shepherd of the flock has authority over you. Obey him, this is Gods order of things.

WHAT DO YOU DO WHEN THEY TELL YOU TO TURN DOWN?

Well, if you, the sound mixer are in the Spirit, and in tune with what God is doing, then by instinct you will be at the right volume. Gods volume. You will get a witness in your spirit about the request. If the request grieves you in spirit then refuse to turn down, and I mean refuse. Dont make some wimpy excuse as to why not etc. Simply tell the person involved that you WILL NOT. Use your authority. If the witness you get is that you are too loud, then simply say thanks to the person for their help and turn down.

Assuming that the person making the request is not the shepherd, tell them to go and complain to him. This will put most moaners in their place. If they do go to him, and he comes and says turn down then you are faced with a decision. Is he simply a man pleaser? or is he right in making the request? You must decide.

If the man is a man pleaser, by that I mean, if the volume is really not offensive, and hes just asking you to turn down to keep the peace, then resist him. If he insists, then dont turn down, turn off.

Paul says in Galatians 1:10 Am I now trying to win the approval of men, or of God? Or am I trying to please men? If I were still trying to please men, I would not be a servant of Christ. (NIV) Neither is anyone who compromises their faith to please people. Never come under the authority of a wrong or bad spirit. And never cast your pearls before swine. Rather take your anointing and go home.

Jesus said, if they do not receive you, shake the dust off your feet and leave.

How will the church be set free, if you go to them and come under their bondage? No, my friends, take freedom and truth with you when you go, and never compromise.

Lets face it, theres nothing quiet going on in Gods presence, He doesnt seem to mind a bit of noise. Look in your Bible.

WHAT ABOUT THE OLD FOLKS?

Ah! the age old problem (forgive the pun). This is easy to answer. How many old folks do you know that have good hearing? Well if they are two parts deaf how can they be upset about it being too loud? Its daft when you think about it, you usually have to shout to get them to hear you anyway. No, the problem is not the old folks, but the not so old folks who want to run their lives for them. Its the middle age group who are the real problem. They think granny will be upset by the nasty noise. Rubbish, in my experience the old folks get on quite well if theyre left alone. Theres a lot more life in some of THEM than in some of the younger ones.

As a general rule its not the volume thats the problem for the older folks, its the style that upsets them. You sing To God Be The Glory with the organ blasting out and theyll join in with great gusto, and not turn a hair at the volume. But put an electric guitar in front of them or a drum kit and suddenly its of the devil or fleshly. Tradition is the enemy of true praise. Clinging to the past is not Gods way, neither should it be ours.

TO SUMMARISE.

In any average congregation about 30% will have some kind of hearing deficiency. Few if any will be deaf, most will suffer to a greater or lesser extent from speech deafness caused by years of working in a noisy environment, or just plain old age. Ears wear out just like any other part of the body.

The problem is that the 30% never complain, some dont like to admit that they are hard of hearing, and others just sit there and put up with it not knowing the difference.

How many complaints have you had that it was too quiet? Yet one complaint about noise over rides the very real need for a lot of the congregation, who are missing parts of the message, or the spoken instructions from the worship leader, because of the very small minority who think it should be quieter.

Those who have more sensitive hearing are very VERY rare. No matter what people say to bolster their complaint. If you get this one thrown at you, suggest cotton wool. A big wadge of it stuffed in the mouth. (I repent).

FINALLY.

Let common sense prevail. The Holy Spirit will always guide you in these matters, but you must be obedient to Him. And dont ever use volume as an ego trip like they do at rock concerts. Jesus is Lord, and we do all in His name, and for His glory. Its not to show off the band, or the songs, or our prowess as musicians, or even the mega quality of the PA system! Its for Jesus.

May the Lord richly bless you as you serve Him.

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