John Sayers Productions

ABN 85 588 065 521

A guide for the home recording artist

Contents

Instruments

Drums & Percussion

Introduction

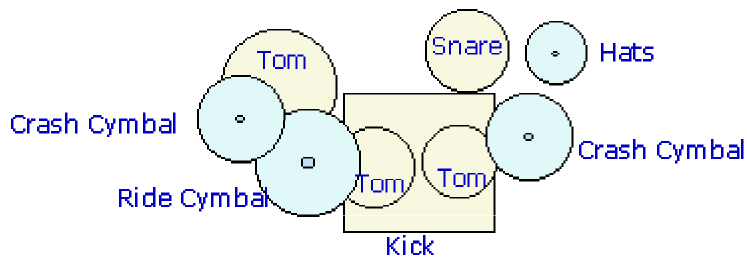
Have you ever noticed how engineers all seem to start with the drums when recording a band or mixing a PA. This is probably because the drums are one of the hardest instruments to record, yet the drum sound can dictate the whole sound of the band or the PA Mix. If you are learning, recording drums can be a real challenge as they are the most complicated instrument - or I should say - group of instruments.

Every engineer has his or her system of recording drums and they are all probably as similar as they are different. What I intend to do here is to establish a **method** for recording drums because without a starting method you can spend hours mucking around not really getting anywhere. I started recording drums in the mid sixties here in Australia and I really had no teachers (as opposed to you in the US who had a rich tradition of recording engineers). I think I was one of the first to record drums in stereo onto 2 tracks of an eight track with the bass mixed in. Therefore I had to develop a technique of my own, I used to spend hours in the studio late at night with drummer friends and play around trying every mic placement possible until I developed a system that we could apply in all circumstances. At the time we were recording commercials and television show bands (Brass, strings saxes etc all in) continually day in day out.- I hope that this method assists your recording techniques.

Drums and Percussion – Mic Placement

MICROPHONE PLACEMENT

Let's take a look at a standard drum kit.



Looking from the top this is the standard layout of right handed drummer. What we have to do now is Mike it! So let's start by putting two mikes over the top (generally called the "**Overheads**"). But where? We have to start thinking in terms of a **stereo image** right from the start.

Stereo Image

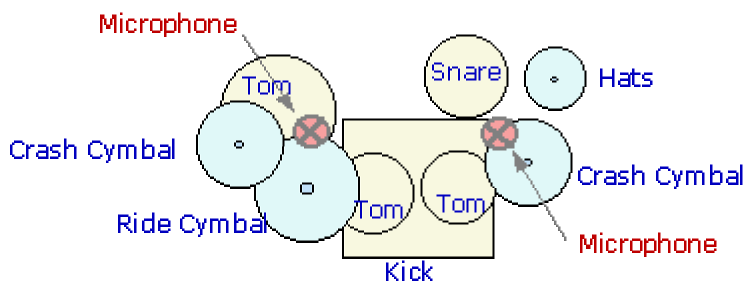
The **stereo Image** is the picture of sound created between the two speakers. When a signal is placed equally in each speaker the sound appears to come from a phantom centre speaker, this is what we call the **stereo centre** All balancing is based around this point - e.g. the bass, kick and snare are normally panned in the centre whilst the hihats, toms and cymbals are spread across the speakers from left to right. The easiest way to experience the stereo spread is to listen through headphones where the effect is more obvious.

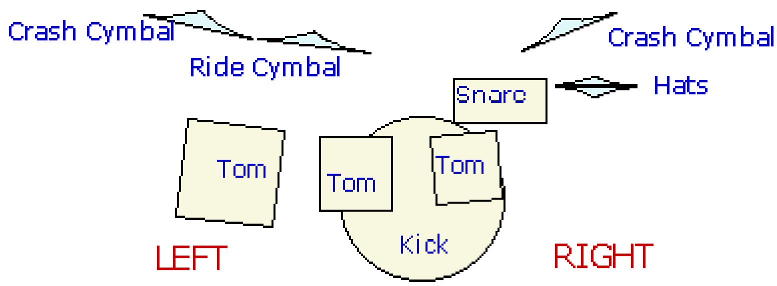
The standard panning setup in drum recordings is:

- Kick - Centre
- Snare Centre
- Hats - Half right/right
- Cymbals - left - right

- Toms - left - centre - right.

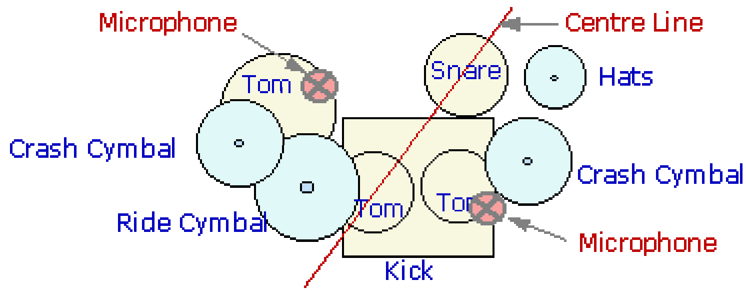
But if you look at a kit it isn't really setup like it should sound. The snare is to the right, the toms have no spread etc. In fact if you were to put up two overheads left and right the stereo image would have the kick centre and the snare to the right and the toms going from centre to left.

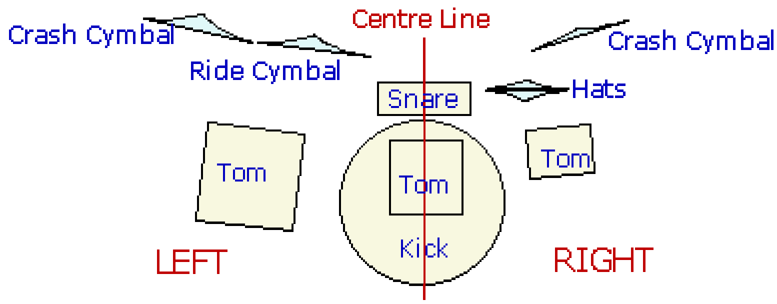




Sound Picture Created

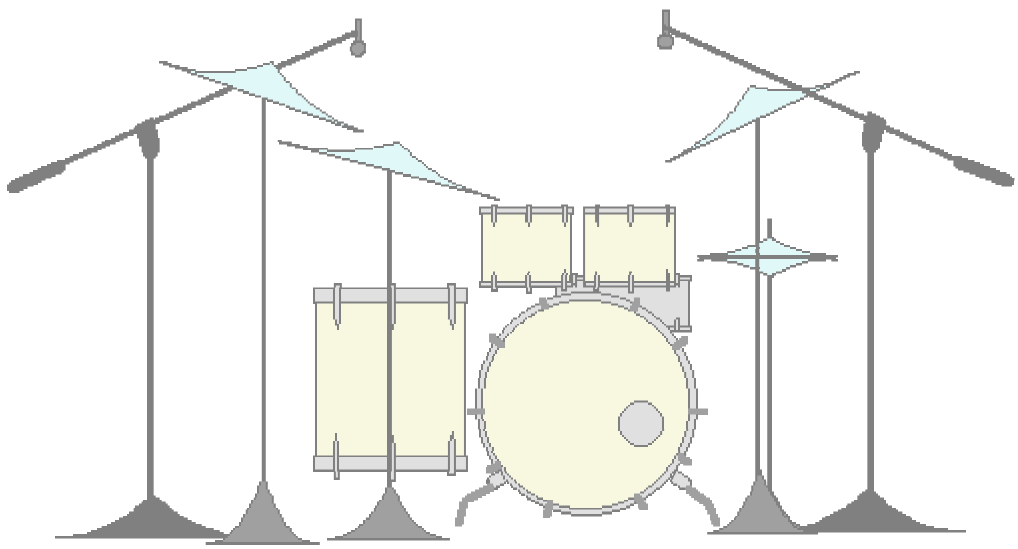
Yet on recordings if you imagine looking at the kit from the front it looks (sounds) like the snare is centre and the toms spread from left to right. (You will notice that I refer to the imaging as a picture - well that's what it is!) So can we get the overheads to paint a picture like this? Have a look at this setup which is based around drawing an imaginary line through the kit which lines up the kick and snare etc.

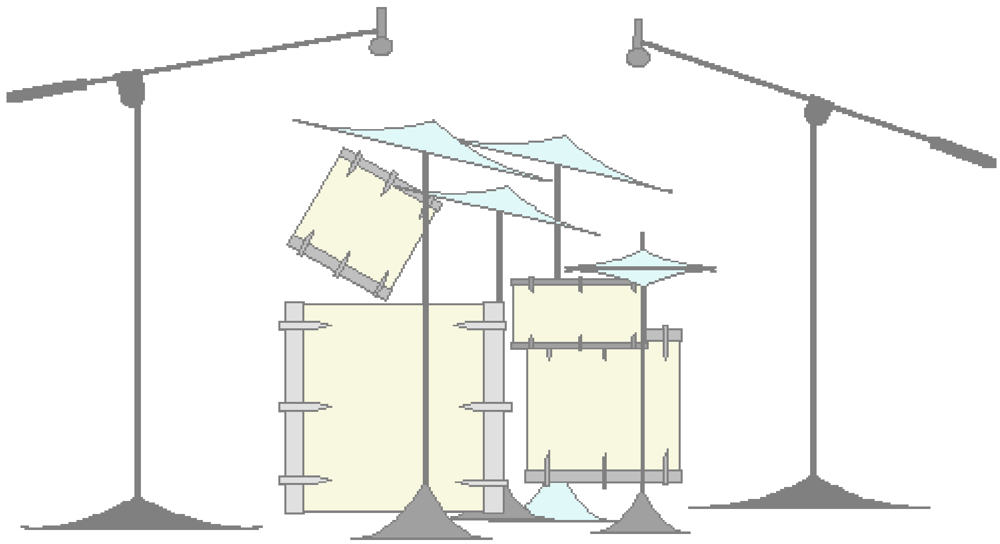




The microphone placement places a stereo image similar to where you want to go as far as the placement of the different components.

The mic placement will look like this"





You can now start to hear a stereo image of the kit as you will want to hear it in the mix. If you were to put the mikes together in a stereo pair but aimed each side of the dividing line you would get a stereo image but a narrow one. The width increases as you move the two mikes away from each other. The placement in the drawings above are about normal with enough spread to make the kit have some width. With practice and careful placement of these two mikes you should be able to get a good balance of the kit. If you added a kick drum you would have a real, open sound of the kit.

The next step is to mike the individual components so that their position and individual sound can be emphasised.

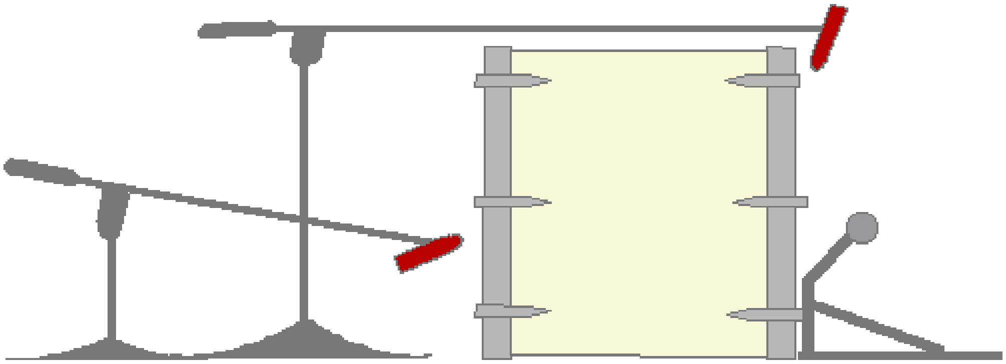
KICK DRUM

There are three ways of setting up a kick drum

- Front and rear skins on.
- Front skin with rear skin with hole in it
- Front skin only.

These three set-ups create three differing sounds. First you must tune the kick as per the directions in the **tuning drums** page.

The first setup with both skins is a thick, solid, round sound with a decay as the drum decays. I believe the best way to mike up this setup is to use two mikes. One over the pedal and one at the other end like this.



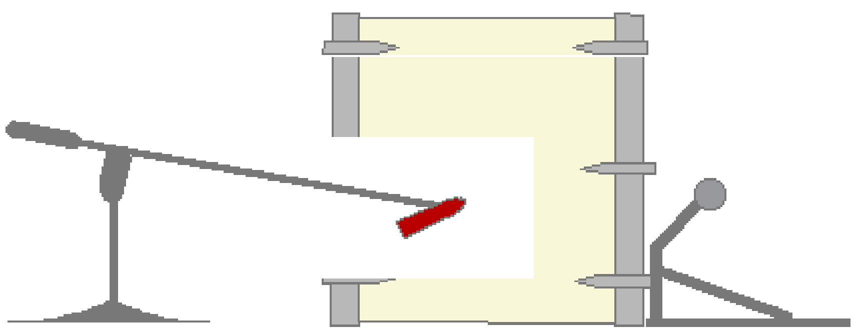
This setup allows you to balance the attack sound of the beater with the decay of the front skin. This miking setup also brings up and important factor in recording:

Microphone Phase Relationships

So which mike should you phase reverse??. If you look at the microphone over the beater it is pointing downwards like all the other microphones on the kit will do whereas the microphone on the front skin of the kick drum is facing the opposite way. Therefore the front skin mike should have the phase reversal. As you can see it is a good idea to reverse the phase of your kick mike even when you are not using two of them as the normal kick mike setup places the kick mike out of phase to the rest of the kit mikes.

Similarly, when we get into miking toms and snares top and bottom the bottom mike will require a phase reversal.

The next setup is where the kick drum has a front skin on with a hole in it. Because of the hole you can access the front skin - thus the attack sound - without having to use a beater mike.



Here the mike is placed inside the drum pointing to where the beater hits so as to get the full impact of the beater. Note that the mike is still out of phase to all the downward facing mikes on the kit so a phase reversal is preferred. The mike is also placed off centre within the shell.

Another factor effecting the kick sound is the beater the drummer uses. Beaters vary from soft to hard. Hard beaters (usually wood) have more impact sound than the softer beaters. Experiment with each and you will hear the difference. How close to the centre of the skin the beater is placed also varies the sound. Similarly the size of the drum sticks the drummer uses will also effect the sound - thin sticks aren't going to go boof! no matter how much you EQ them.

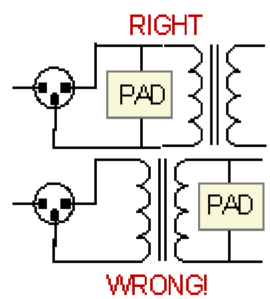
Sound Pressure Level:

It should be noted here that the SPL (Sound Pressure Level) created by drums is extreme so you must **select a microphone** that can handle high SPL and even then it will output a high voltage into the console. Therefore a Microphone PAD should be inserted in the console to prevent the front end of the microphone preamplifier distorting. If your console doesn't have a mike pad switch you should insert one in the microphone lead. Like the phase reversal plug you can purchase mike pad plugs from your local dealer. A pad of anywhere from 10db - 20db will be required.

A note here about mike pads.

Mike Pads

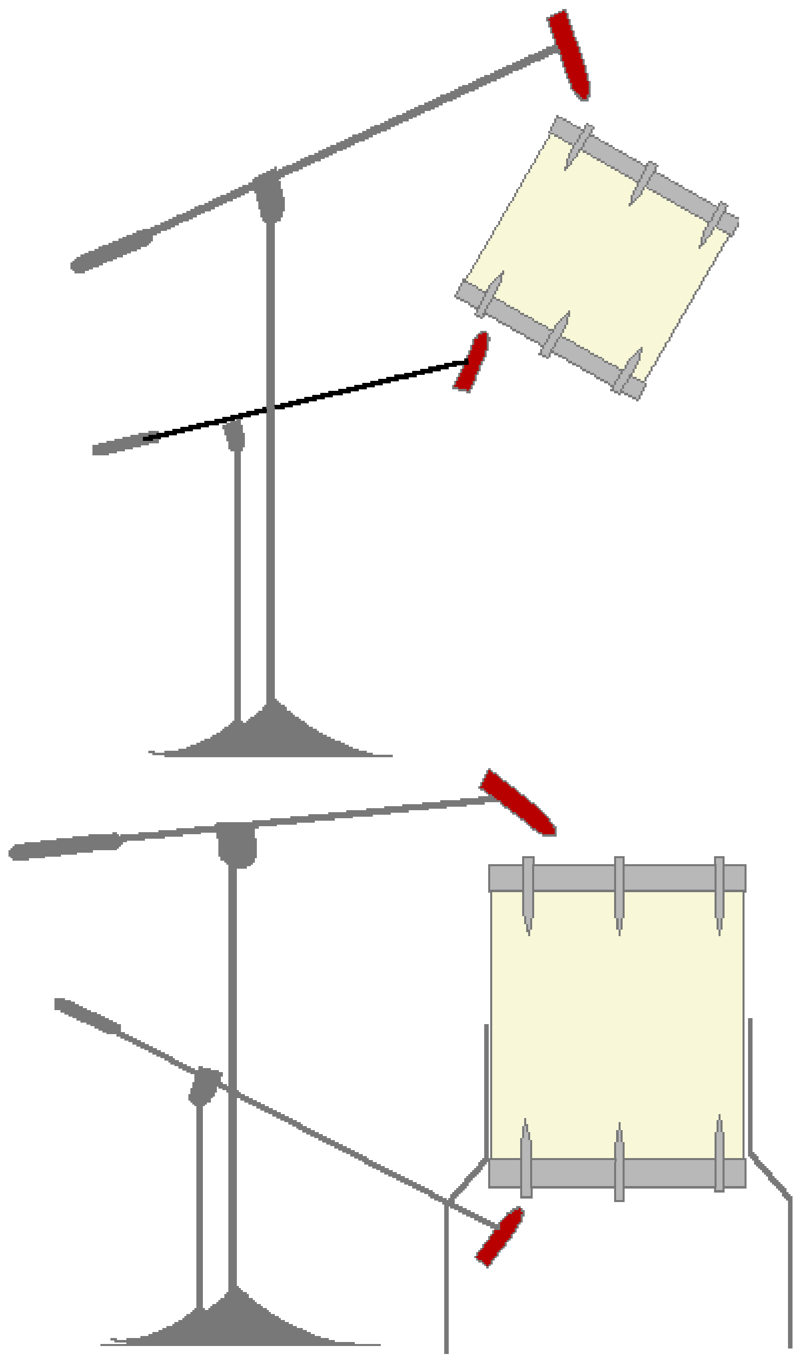
When purchasing a console check that the microphone pad is in the proper section of the circuit. Some manufacturers put the pad after the transformer and before the preamp. Unless the transformer is an extremely high quality one it will distort so make sure the pad is before it in the circuit.



Here's a circuit diagram for all you techo freaks who understand circuit diagrams. Nowadays all the front ends are electronically balanced circuits that are capable of handling the high voltages that modern mikes produce but if you are into **retro equipment** its worth checking your old console out because a lot were made with the pad in the wrong place.

TOMS

The toms are similar to the full kick drum miking in that there is a mike on the impact skin that gets the full attack of the stick when it hits the drum plus you can also add another optional bottom mike to get the hang of the the drum. You must again remember the phase relationships here. If you wish to add a bottom mike to the toms you must reverse it's phase.



RACK **FLOOR**

TOMS TOMS

If your drummer doesn't have a bottom skin on the toms you can use either a top mike or both mikes or you can opt for just one **under mike** with a phase reversal naturally. The advantage here is that the under mike is inside the tom which isolates the mike from the other drum sounds and improves separation.

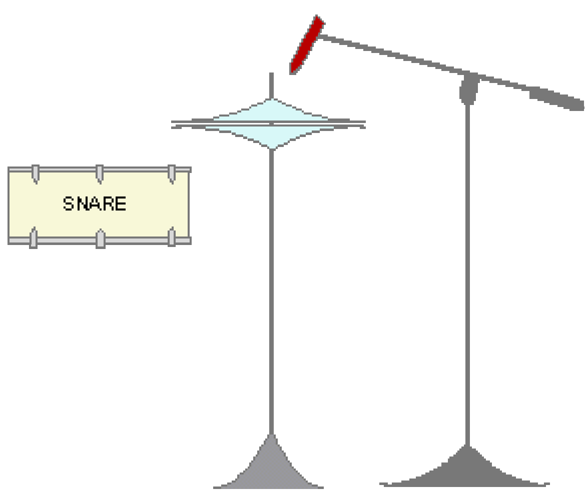
CYMBALS AND HIHATS

The Cymbals

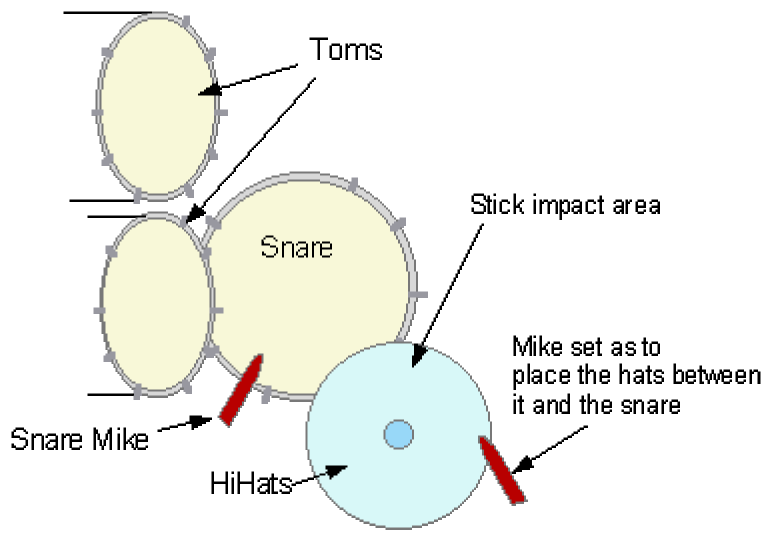
These are basically covered by the overheads but you might find that the ride cymbal needs a mike of it's own if the drummer rides it a lot through the chorus. Basically you want the crash cymbals to have a loose sound yet the ride often is the main drive as it replaces the hihat for the 8 a 16 feels. You must consider this factor when setting up the overheads. Drummers also accent using the bell of the ride cymbal that can be extremely loud so beware of miking too close to the bell of the ride cymbal or it will dominate the sound field. Some engineers mike the ride from underneath. In a complex drum setup with lots of splash and crash cymbals you might like to spot mike certain cymbals but I reckon that if you've setup your overheads correctly they should cover the full cymbal range.

The Hihats

Like the overheads the hihat also requires a mike with a clean top end so it's usually a condensor mike. I like to hide the hihat mike from the snare by placing it in a position that is pointed at where the drummer impacts it with his stick but the hihat is physically between the hihat mike and the snare.



Good separation between the hihat and the snare is desirable so consider the snare when you place the the hihat mike. Another factor of the hihat is the sound made when they are snapped together. I like to aim the mike so it is pointing at a point that gets the stick impact as well as the pointing at the edge of the hats as that is where the closing sound emanates. N.B. If you get too close you will get wind distortion from the hats as they close.

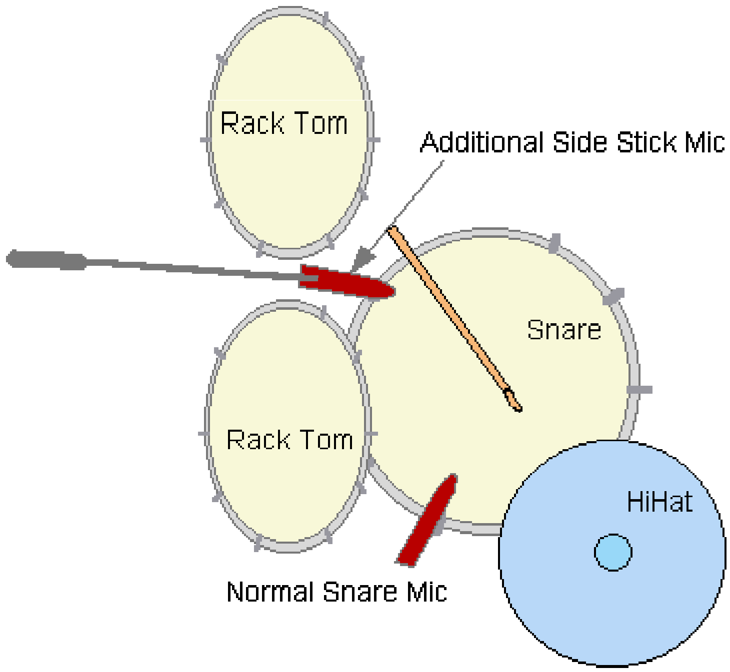


One of the problems you can get is where the drummer has the hihat low to the snare and the toms also low to the snare. This creates separation problems as well as making it hard to isolate the snare from the tom. There's not much you can do other than ask the drummer to change. This is not as awesome as it sounds, some drummers have never considered this aspect of their kit layout and on making the change actually say it's OK and find they easily got used to it and now prefer it. The same problem can occur with the ride cymbal - some drummers have their ride cymbal almost touching the floor tom which makes separation hard - I recently had a drummer like that and when I mentioned it he agreed to change. After the session he remarked that he actually liked the change and would do it in future. Moral of this story? - don't be afraid to ask!!

SNARE DRUM

Once again, the snare can be miked from the top and the bottom, in fact it is one most often double miked. The bottom mike on a snare can give the snare more depth but it also gives you control over how much snare crack sound is in the overall sound. (The **snare** is actually the stretched wires across the bottom skin and gives the snare it's sound - otherwise it's just another tom). The snare mike is normally squeezed in between the hihat and the first rack tom and like the tom mikes is aimed at the main impact area in the centre of the snare.

Side Stick: Often you have a drummer playing a lot of side stick. I have used a separate mike specifically for the side stick. The side stick action is for the drum stick to hit the rim of the snare drum and the main impact is on the right side of the snare. As your normal snare mike is placed on the left side it doesn't always pick up the side stick clearly. Not only does this give you a mike closer to the side stick action it also allows for different EQ and effects for the side stick sound. You can either track it to a different recording track or you can watch the drummer and switch mikes during record.



AMBIENCE MIKES

Drums miked close-up don't actually sound very real as their real sound is a combination of various factors. You actually have to get away from them to get the full sound. A close mike on a snare doesn't really sound like a snare drum (thus the importance of the overheads) so some engineers add Ambience mikes to allow the freedom to add the distance sound of the kit when mixing. Naturally the drums must be in their own room for this system to be used or the ambience mikes will pick up everyone else in the room. Basically ambience mikes are a stereo pair of mikes placed at a distance (room size limited) from the kit. They can be setup as a crossed pair or moved apart to gain a more ambient spread. You might like to try using a **MS Stereo** mike setup. Ambience mikes can also be **Gated** - so they only open when the snare is hit for example- and you must have plenty of recording tracks to allow for another stereo pair. Ambience mikes effect all the kit and push the drum kit back in the sound field so if you want a round tight kick sound and an ambient snare sound you have to gate them so they are closed for the kick and open for the snare. An engineer I know used to hang a very directional shotgun mike high above the kit aimed at the snare and use the under snare mike to trigger a gate that opened it whenever the snare was hit. He would then mix it in with the snare sound and it gave the snare a natural ambience and was extremely effective.

So now we have set up all the mikes we are ready to start **balancing and equalising them**

Microphones for drums

- **The Kick**. What are we looking for in a kick drum mike? Firstly and most importantly it must be capable of withstanding high sound pressure levels!! When a mike is only inches away from a kick drum beater the sound pressure levels are extremely high at low frequencies. The kick drum mike must be capable of handling the extreme transients involved. Secondly it must be capable of reproducing very low frequencies.
  The two most popular kick mikes are - The AKG D12 and the Beyer M88. The M88 is my favourite. Both these mikes have an extended bottom end response and can handle the high sound pressure levels associated with kick drums. On the other hand if the drummer is not hitting too hard you can't beat the Neuman U87 or 49, which are high quality condenser and have large diaphragms (good for low frequencies) and smooth low end response. Other mikes are the Shure SM57/58 and the old RE20 which are both capable of withstanding the load.
- **The Snare**. Here we are looking for a mike that will withstand extreme high end transients and has a tight pattern so as to keep out the high hats and the adjacent toms. The most common snare mikes would have to be the Shure SM57 and the Sennheiser MD421, followed by the Neuman U87/89 and the AKG 414EB. Others are the Sennheiser MD441 or the Neuman KM84. I'm always amazed at how many engineers still use the Shure SM57 even though there are lots of other mikes around. The main advantage of the SM57 is that it's a tight mike with a tight pattern that keeps out the spill from the hi/hat and the toms. They are also extremely reliable and don't mind being hit by a wayward drummer. I should note here that the difference between the SM57 and the SM 58 is that the SM58 has a permanent wind shield - the microphone section is identical. You can buy a wind shield for the SM57 (Note the two microphones next time you see a press conference from the White House.)
- **The Toms**. The two main mikes used for toms are the Sennheiser 421 and the Shure SM57. In the studio I like to use Neuman U87's as they have a beautiful warm bottom end. The Shure SM 57's don't have a lot of bottoms but if you're tight miked the proximity effect compensates for it and as with the snare their tight pattern helps.
- **Overheads**. Good condensor mikes make the best overheads. There are three main overhead mikes, the Neuman U87 for warmth, the AKG 414EB and the AKG 451 for crystal clarity. The AKG C1000 and the Roden are also a good budget condensor overhead mike except I find that both have a slightly tinny top end compared with the more expensive models. I would say the AKG 451 with a CK1 capsule and 10db pad is the most popular overhead mike.
- **Hihats.** Condensor mikes with a tight pattern make the best Hihat mikes like the AKG 451 or the Neuman KM84. Both have a 10db pad option which is handy as the high end transients from a hihat are extreme.
- **Ambience Microphones**. Usually high quality condensor mikes are used here.

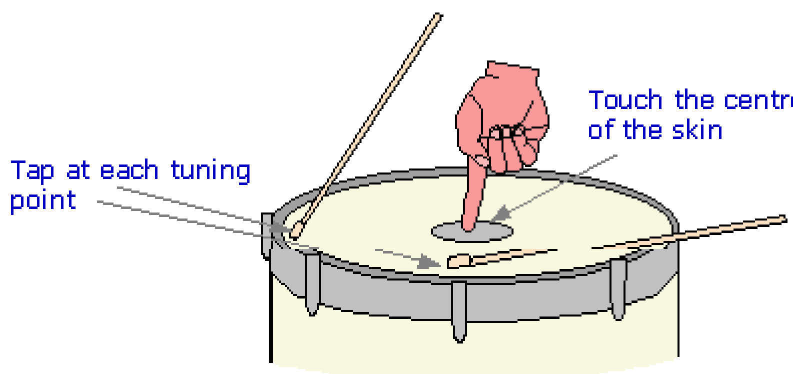
Drums & Percussion – Tuning Drums

" How do you get a great drum sound?" -- "Get a good drummer!" Like every other instrument, drums must be properly tuned and good drummers know how to tune their drums. They also know how to hit them consistently on the right spot so their sound is true. You can tell a good drummer by looking at the skins. If they are worn in a small circle in the middle - they are good - whereas if there are stickmarks all over the shop they're not consistent.

The kick, toms and the snare are all tuned the same way. I recommend you take the drums off the kit to tune them.

Note: When you start with new skins it is a good idea to stretch the skin once it is on the drum. A drummer suggested to me that he stands on the drum (especially kick drums) and lets his weight stretch the skin!! He swears by it.

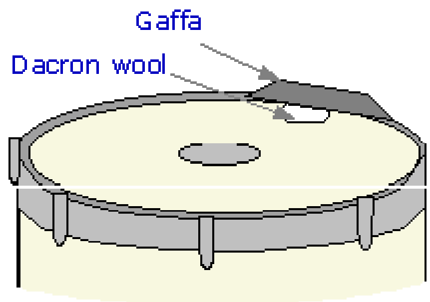
First, make sure the bottom skin is nice and tight. The bottom skin is usually tuned higher than the top - about a fourth up - every drummer has his own tuning, the main thing is to make sure that it is even. To tune a skin evenly you must put your finger in the centre of the skin and tap the outer part near each tuning point.



You will notice that each point produces a slightly different note so you go round tuning each point so that they all produce the same note. I usually work on the pins opposite each other because as you tighten one the opposite pin is effected. You do this procedure for both top and bottom skins. You must also tune each drum relative to the other drum so that the high toms progress down to the low toms.
Once you've tuned each drum mount them back onto the kit. Now if you hit each tom a pure note will sound. If you now take **one** of the lower tuning pegs (the one closest to you) and start to lower (unscrew) it as you keep hitting the drum you will find a point where the skin hangs out and even appears to drop in pitch. A straight tom sounds like **doom** whereas one with one pin detuned sounds like dooommmmmmmm with the mmmmmmm dropping in pitch and the whole note lasts longer. That's how you get the t-tooo t- tooo t-tooooo fill sound because the toooos drop in pitch. (Still with me??)

The snare is the same except that you don't want to detune one of the pins. Consistency of pitch at all the tuning points is essential on both top and bottom skins.

Damping: I try and avoid dampening toms. The old system of a piece of Gaffa Tape all over the toms doesn't produce a very good sound. If you need to dampen the toms or the snare I suggest a piece of dacron (polyester wool) with a piece of gaffa stretched over from rim to rim but with the gaffa **not touching the skin,** only the wool.



This way the skin is not choked too much and you can apply small amounts of dampening. The toms are going to ring - but so what. You can always gate them out later or **better still** automate them out of the mix with fader/mute automation. If you dampen them down so they don't ring they will sound like cardboard boxes like Ringo used to play. There are now available small squares of some kind of synthetic rubber that feels like a jelly baby which you place on the skin. They work very well for dampening because like the dacron they don't choke the skin.

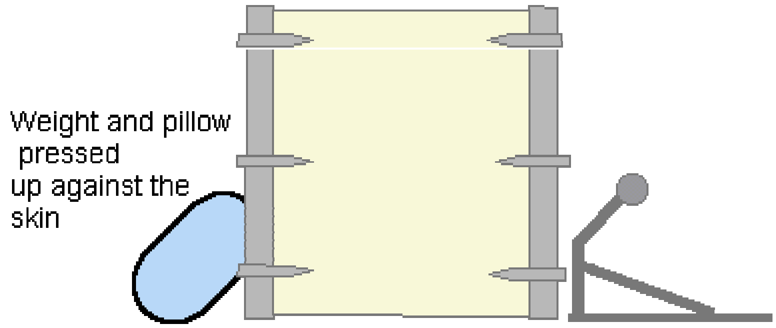
You can also dampen Cymbals by sticking a small strip of gaffa tape to the underside of the cymbal. This is only necessary when the cymbal (usually the ride) is too ringy and lacks definition.

KICK DRUMS

Kick drums are another story. There are three ways a kick drum can be setup depending on whether the drummer uses one or two skins or has two skins with a hole in the front skin. Some drummers actually line the inside of their kick drum with a layer of foam that acts as a permanent dampener.

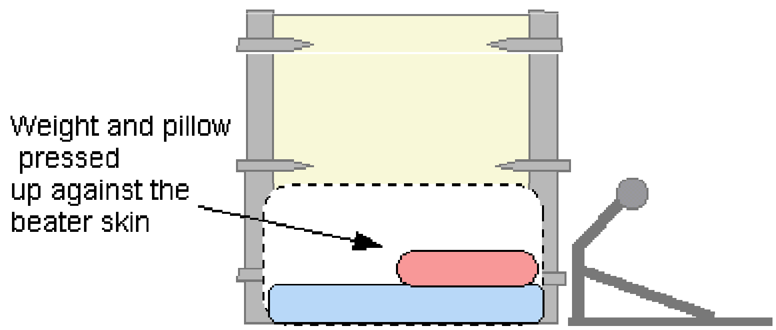
Both skins on

This is the traditional kick drum setup. Having tuned the drum using the previously suggested method you must next determine whether it needs dampening. If you feel it does (typically) you can use a pillow or a blanket pressed up against the front skin and held in place with a sand bag, brick, mike stand base, or anything with weight lying around the studio.



Both skins but with a hole in the front skin

In this situation you have access through the hole to place dampening inside the drum. Once again a blanket or pillow is placed on the base of the drum and held in place with a sand bag or weight.

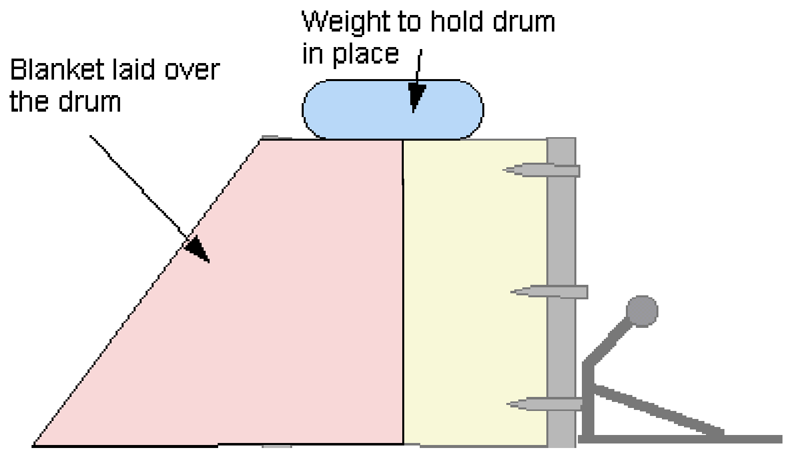


Beater skin on only.

Here the front skin has been removed allowing dampening to be placed in the drum as in the previous example. This is the most typical system yet I notice nowadays that there are a lot of drummers opting for the more traditional sound of using both skins and going for a more "natural" kick sound as opposed to the clicky percussive sound used throughout the last few decades.

Additional Option

Now that the kick drum has been dampened to your likening may I suggest you can dampen the whole drum by placing a blanket (or better still a sleeping bag) over the whole kick drum. This helps gain isolation of the microphone in the drum from the rest of the kit. (a sort of acoustic gate) Additionally a weight (sand bag) can be placed on the kick drum to make it rigid and dampen the shell vibrations.


Weight to hold drum
in place
Blanket laid over
the drum

Drums & Percussion – Tracking and Balance

Before we start on this subject let me tell you a story: I was working at a studio called the Music Farm and there was an English engineer working there called David Tickle. He went on to produce Tony Childs . One evening a muso asked him "How do you become a good engineer?" - to which he replied:

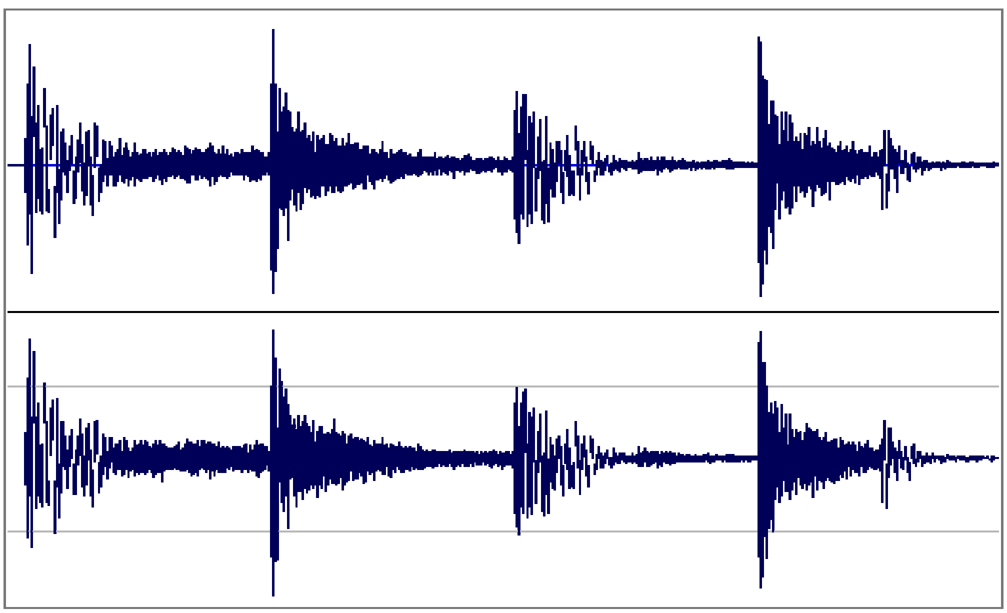
"You go into the studio and ask the drummer to hit the kick drum and you grimace. You put a mike on it and go into the control room and ask the drummer to hit it again. Again you grimace. If you can make the kick drum sound like the one you hear in your head - you are a good engineer! It doesn't matter how you do it but a good engineer can"

I can't really tell you how to do it either but this page will give you a starting point.

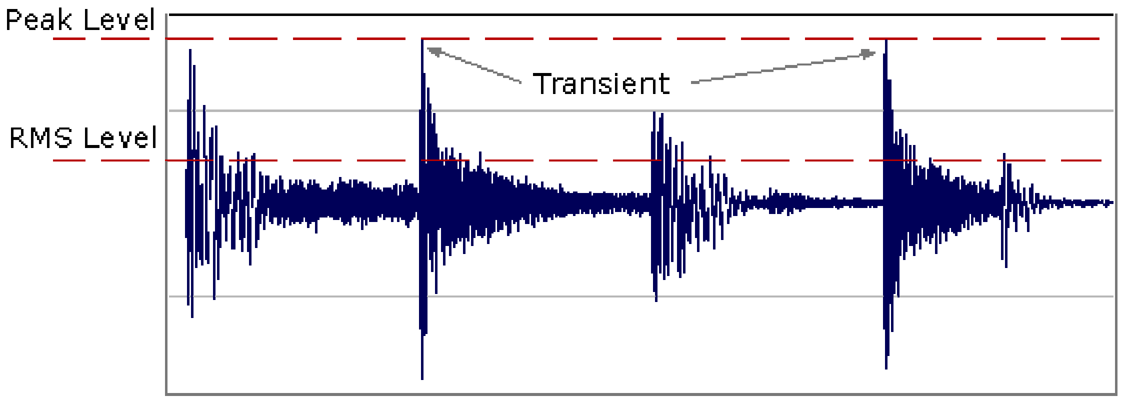
METERS

Before we do anything we must talk about meters and what they read.

Take a look at this image of a drum sample: Click image to hear sample.



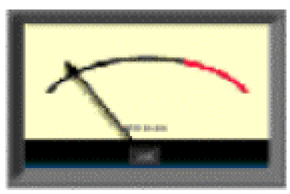
You will notice from the shapes that the waveform rises sharply at the start and then tapers off as it fades. This initial rise to the highest point at the front is called a **transient** and the highest point is called the **peak** of the waveform.



There are two ways to meter this waveform:

1. **The VU or Volume Unit meter**
2. **The Peak Meter.**

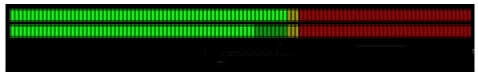
The VU Meter



These meters read **volume units** , hence VU, and do not give an idea of the peak material. They indicate the **RMS** (root mean square) value. Some VUs might have a peak LED in the upper corner. NOTE: If you peaked this meter to Zero on the drum waveform the actual peak transient would in fact be 10 - 20db or so higher than the meter was reading and can give you a wrong impression of the actual peak content. If you peaked this meter to about what it's reading now the peaks would be in the red section. So if you are working with this type of meter don't peak transient sounding instruments like kick, snare, percussion, toms etc. to zero because you will distort all the transients. In the late seventies the console manufacturers started adding peak meters. So if you are into retro gear with VU meters watch your transient levels!!

The technique of pushing magnetic tape by saturating the tape with high level was really just a method of compressing the transients by using the tape as a compressor thus the new trendy term **Tape Saturation**. Actually the tape played back the transients but only for a few days - after a week or so the transients were well and truly eliminated and engineers complained that it sounded great when we recorded it but somehow the tape has lost it!!! (the tape also suffered from extreme print through which is a condition where the magnetic flux on a tape is so high that it printed through onto the tape layer adjacent to it when it was rewound and was left sitting on a shelf.)

The Peak Meter



These are the ones used today. They often incorporate a peak hold function that lets you see what your peaks are doing and are therefore easier to use and are a truer indication of what is actually happening. Don't try saturating your tape with one of these meters as digital doesn't saturate - it just distorts severely!!

TRACKING

Before we can balance the mikes we must bring them up into the console and set their levels to tape. How the mikes are tracked to tape is really dependent on how many tracks you can afford for the drums. I usually allow 7 or 8 tracks for drums. This might seem extravagant to some and not enough to others. I'm conditioned to the old 24 tracks of recording and therefore think in those terms but nowadays with multitrack digital recorders and multitrack computers anything goes but with 8 tracks you can assign your drums as:

1. Kick
2. Snare
3. Hats
4. Tom 1
5. Tom 2
6. Tom 3
7. Overhead left
8. Overhead Right

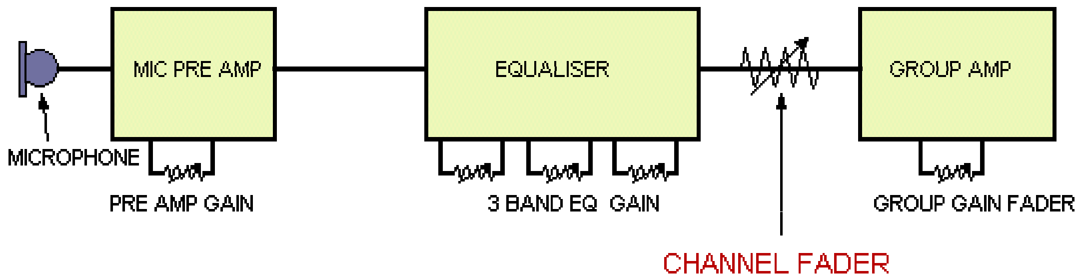
Alternatively you can premix your toms and put the snare under mike on it's own track or you can mix your toms and put the bass on the 8th track. (Or in my case the 4th track because I think in pairs!!) so it goes:

1. Kick
2. Snare
3. Hihat
4. Bass
5. Toms L
6. Toms R
7. O/Head L
8. O/Head R

This setup will leave you 16 tracks for the rest of the instruments and vocals etc. I'm not going to go into how to assign your tracks, I assume you will know this.

I would make a special note here:

You must always be aware of the gain structure surrounding a console. Look at the gain structure and how it works.



The microphone signal is amplified by 40 - 50 db by the mike preamp to bring it up to the operating level of the rest of the console. Any EQ is additional gain or reduction. Lets say you open the channel fader 1/4 the way up and then increase the gain by increasing the mic preamp gain. It is very likely that the higher output from the mic preamp will distort when it enters the input of the next stage, the EQ. You will then EQ it adding more gain and more distortion. Similarly with the group output fader if you have one. If you run this fader low all the previous stages will be driven to overload.

Always start with your channel and group faders at Zero and adjust your gain at the mic preamp

You'd be amazed at the number of times I glance at a PA console and see the group faders at 1/4 and the channel faders at 1/4 and the mic preamps turned up!! and a worried engineer wondering why the PA sounds awful.

EQUALISATION

EQ can enhance a sound but it can't fix it. Too many engineers try to EQ their way out of trouble and believe me - it doesn't work. Adding highs might make it sound better but if you play it through a speaker that doesn't have the great highs your studio speakers have it will still sound awful so you really must aim to get the **sound correct its source**.

OK so we open up the kick mike and ask the drummer to hit the kick. Bring up the preamp level so that the kick peaks to just short of Zero in your meter. (I guarantee that the drummer will hit it louder during the track) Now if you are going to EQ it you can do it now or later. I prefer to do a basic EQ now because it makes the kit sound better while you're tracking the other instruments etc. but don't make it extreme.

The traditional EQ for a kick drum is to add some mids (say +4db) at around 3.5K to bring out the attack, and pull out some low mids (-4db) around 300Hz. (The fundamental frequency of a 22" Kick drum!) I wouldn't advise adding any low end at this stage but you might be tempted to add some 80Hz, say +3db.

If it still sounds awful you must fix it by either:

1. Using a different mike.
2. Changing the dampening.
3. Moving the mike
4. Retuning the Drum.

If none of these work you've got problems.

Moving on: now let's open the snare mike and as with the kick bring your fader to zero and increase the level of the mic gain. Now you have the option of adding the under mike if you've put one up. (Remember the phase reversal) By balancing these together you can vary the amount of the metal snarey sound in the mix.

The standard EQ for a snare drum is basically add some tops around 7Khz. Once again if it sounds awful you must change it at the source.

Now the overheads:

Ask the drummer to hit the snare drum and then open the overhead mikes and balance them so that the snare drum sounds centre. Then ask the drummer to play the whole kit with the toms and the cymbals and listen to the stereo spread listening for the balance of the cymbals. If say the left crash is too low go into the studio and lower the overhead over that crash. Repeat the process starting with the snare again. What you're doing here is making sure that the overhead sound is a true stereo image of the kit. You will hear the toms spread left to right, the kick will be centre. as will the snare. The Hihat will be off to 1/2 right. If you wish to EQ it maybe add some 10Khz tops to it but if your mikes are good you shouldn't need anything else.

Now you go through the toms one by one. Basically toms need a bit of tops and they often need some upper low mids(400Hz) taken out. You may also wish to add some lows like +4db at 80Hz. I suggest you balance the floor tom higher than the first rack tom. Drummers invariably loose energy as they go around the toms - sorry guys. Now that you've got all the toms peaking to zero in the meters open up the overheads. Suddenly they will start to come to life as the overheads will give them space and transient attack.

Finally the HiHats. Remember these are extremely transient so watch the level carefully. For EQ I usually roll off a lot of the bottom end quite severely as well as adding highs at around 10Khz to give them some sizzle.

Gates: Gates can be used when you record drums and the most common method is to gate the kick and the snare and if you have enough gates to gate the toms individually. Personally I don't gate at the record stage mainly because if you haven't set them up correctly you can't redo it later because you've already lost the signal. I recommend that you gate at the mix stage where you have more time and freedom to get the gate tuned exactly for each track. Quite often you will find that the drum sound for one track works great with gates yet another track sounds better ungated. For toms I prefer to automate the tom tracks with mutes instead of using gates. For more on the operation of gates refer to the **Effect Pages.**

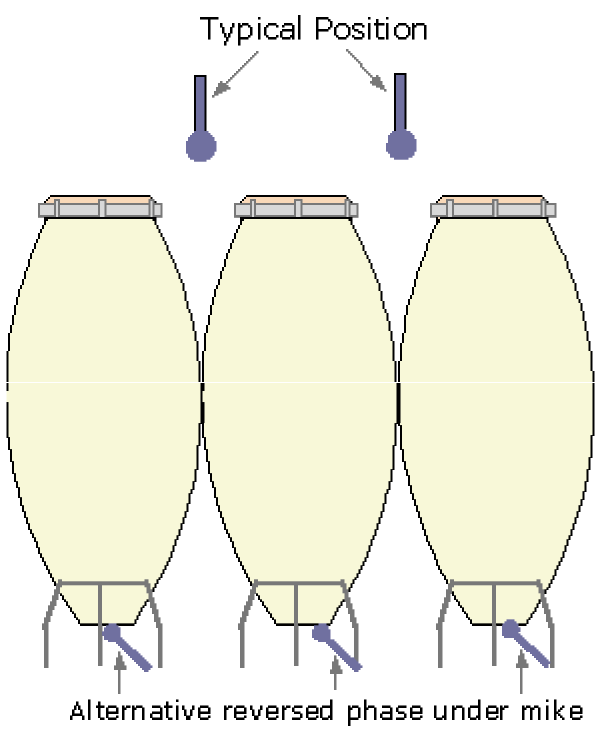
Reverberation. I don't advise adding reverb to your drum tracks at the record stage however you can add reverb to your monitoring by adding reverb to the monitoring controls of your drum tracks, This allows you to have reverb on your monitors but the reverb is not on tape. N.B. you can't undo reverb!!

Drums & Percussion – Recording Percussion

The main concern with percussion instruments is the **extreme transients** they produce. Tambourines and shakers all have extreme transients so beware of getting tambourines too close to a mike and make sure you meter the **peak content** in the signal.

Congas

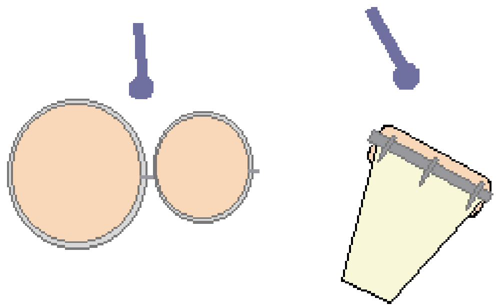
Congas are usually miked with one mike between each conga thus:



The under mikes are an option that can add depth and body to the sound of congas. Like undermiking toms the mikes must be phase reversed relative to the overhead mikes.

Bongos

Bongos are similar to the congas:



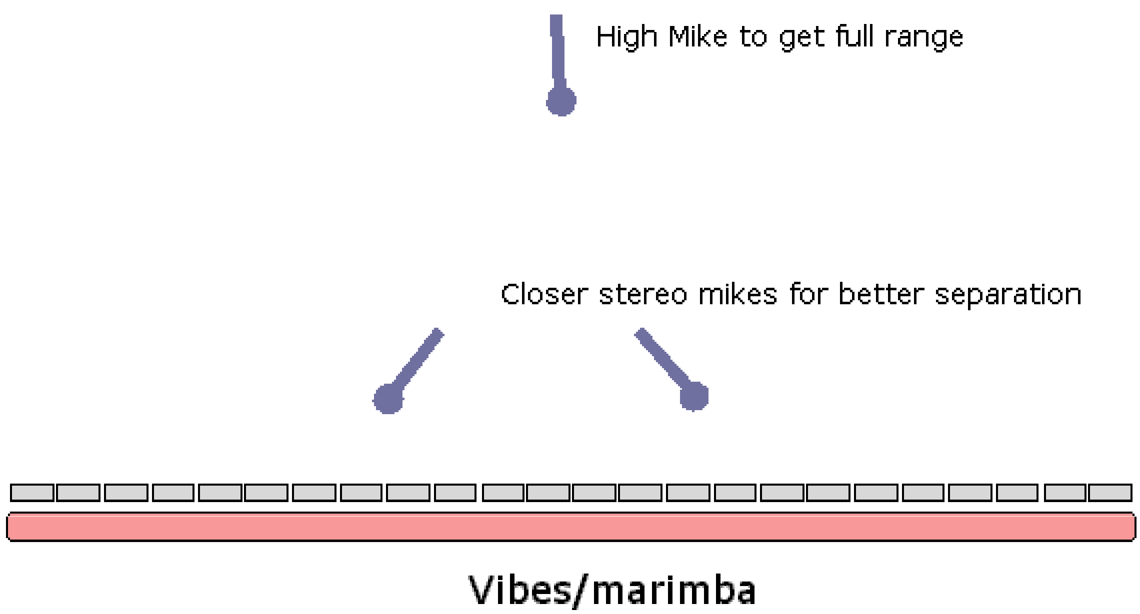
Here once again watch the transients. You can put one mike per drum if you want to spread the stereo sound for effect and you can also mike them from underneath of you want separation in a studio situation.

Tambourines, Shakers, Maracas , Bell Tree etc.

Tambourines, shakers etc. produce extreme transients so use a mike capable of handling high Sound Pressure Levels if you want to close mike them. For shakers and tambourines I like to have the player stand 3 - 4 feet from the mike so some of the room ambience creates a space around them.

Vibes and Marimba

Vibes and Marimbas can be either stereo miked or single miked. A single mike need to be higher than stereo miking to capture the full range of the notes.



The Main thing with percussion is **WATCH THE TRANSIENTS**

Recording Vocals & Harmonies

I suppose the recording of vocals is the one area of recording that the nerves start to automatically step in. Deep down everyone knows that the recording will sink or swim on the ability of the vocal to sell the song. Sure the other musicians have had their own stresses and strains throughout the recording process but everyone knows that what the public listen firstly to the song and the vocal but before we get into that we must decide what vocal mike to use and then get into it.

The Vocal Mike

The choice of vocal mike is really dependent on what mikes you have in your mike cupboard. I believe it's a good practice to setup your best mikes in a row and have the singer sing into each one and then compare them all. What worked yesterday for one singer mightn't work with another. So make sure you start off with the best mike for the situation. Let's face it, any of your best mikes will do the trick. I've heard great vocals recorded on a SM58.

Wind Shields

The use of wind shields or pop filters as they are sometimes called is a good idea. If you want the singer to be in your head they have to be close to a mike, and I mean close, like 2.5 - 5cm(1" - 2") away. Therefore a wind shield is a good bet. P's, B's, C's, all produce a jet of air and it's that shot of air that causes the diaphragm to bottom out as it were and produce a pop sound. Wouldn't you be pissed off if you got the great performance but there was a big POP in the middle. (Sure you can whip it into soundforge and EQ it out but that's another story) The problem with windshields is that they effect the top end response but if you use an appropriate material it shouldn't effect the sound too much. You can purchase windshields but they are easy to make. Make a wire hoop out of an old coathanger and tape a strip of nylon stocking over it and gaffa it to a mike stand. There you are - a great wind shield! Incidentally I have found that PZM mikes don't pop even when extremely close and they don't have **proximity effect.** Check out this factor if you haven't already.

Presence

Presence on a vocal is important. A big factor is the room the singer is in. If it's live there will be a lot of room sound in the sound but if they are in a dead room it will be lower. I prefer a dead room for vocals.

Headphones

The singer can use either headphones or speakers to sing too but the usual way is to use headphones. The most important factor here is to make sure that the sound and balance is right. Please make sure that the headphones are in stereo!!!! It's a good idea to spend the time to get a good stereo mix of the track with reverb and effects in the singers cans. Secondly make sure the singer can hear themselves clearly above the bandtrack and give them some nice reverb, it makes so much difference. Some singers like to sing with one can off (tucked up against their head) so that they can hear themselves properly. If your headphone balance is correct this should not be necessary.

Speakers

Alternatively, you can get the singer to sing to a pair of speakers, either in the studio or in the control room. In this case their will be bandtrack spill into the vocal track but if you position the mike between the singer and the speakers with the speakers off axis (180 degrees) to the mike it shouldn't be too bad. Personally I prefer headphones.

Compression

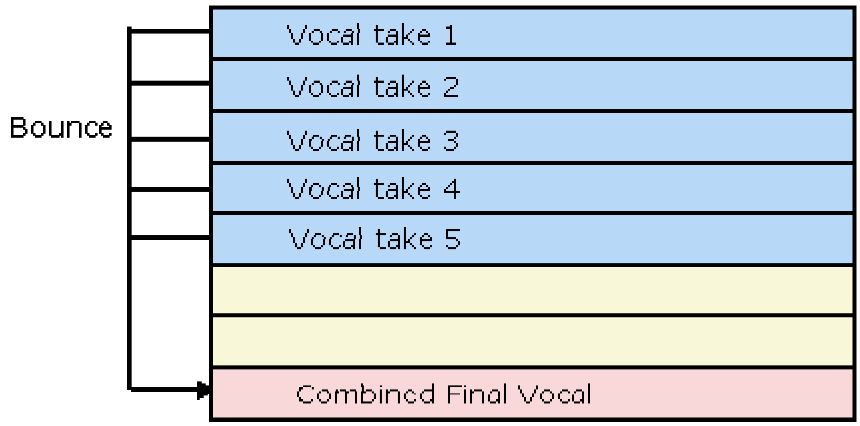
You should consider using a compressor when recording vocals. I like to, not heavily but enough to keep the dynamic range under control. It's a bit like the windshield, it's a protection. Say around 3db on peaks at 6-1 to 10 - 1 ratio. Check out the **compression** page.

Multi-tracking Vocals

Multi-tracking vocal takes is a good idea. Get the singer to record four or five takes on different tracks. Then you can go through the each take and select the best performance of each take. That's cheating!!! I hear you say. Well, if it is, every major singing artist in the world is a cheat because they all do it. Bring up all the vocal takes on your console and set them all at the same level and assigned to the track you want your final vocal to land up on. Now go write out all the lyrics on a sheet and mark each take of each line that's OK like this:

Lyrics	Take 1	Take 2	Take 3	Take 4	Take 5
Blah blah blah	✗	✓	✗	✗	✗
Blah Blah	✗	✗	✓	✗	✓✓
Blah Blah Blah etc	✗	✗	✓	✗	✗

Now, playing the track back mute and unmute each take that's OK and bounce them down to the final track like so: You will note how some get a double tick - I do that when I really like a delivery.



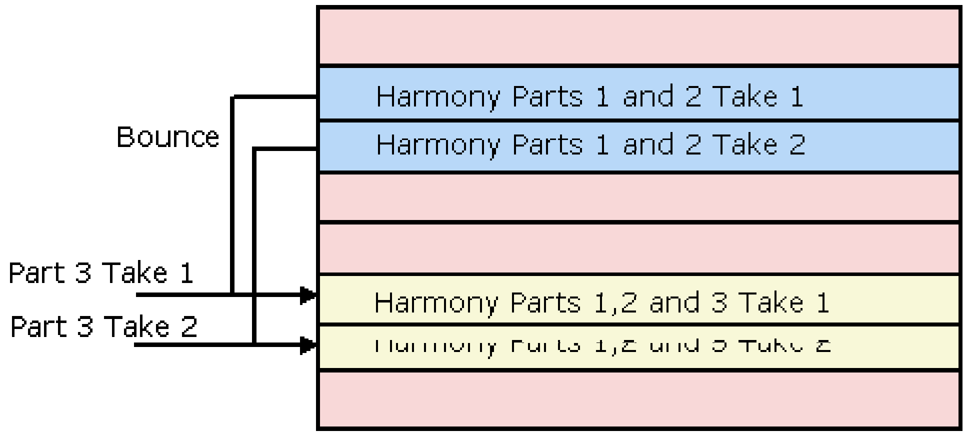
This could take some time as working out each line, line by line, is tedious but believe me the singer will love you for it (providing their ego is in tact) because you will have captured the best they can do and that is what recording is about!! Incidentally, I do all this on a computer now because it's so much easier, then I archive all the out takes in case I need to revert back to them later.

The Harmony Vocals

Recording harmony vocals really depends on how many harmony parts there are. You can put two or three singers on one mike, or two singers on one mike in a figure 8 position with one one each side or you can give each singer their own mike. It's up to you really, so long as the outcome is properly balanced parts. Headphone balance with a three part vocal group can be a problem so I recommend that if one can't hear themselves tell them to remove one can.

Multi-tracking Harmony Vocals

Tracking harmony vocals can be a problem - do I put each part on a separate track or mix them together etc. It really depends on how many tracks you can afford and if you are going to double track them. I usually double track harmony vocals as it creates a blend of parts. So I'll mix the parts straight away onto one track and then double it again onto another. If we then decide that there is another harmony to add and I don't have lots of tracks free I'll mix the first track with the new harmony and record it onto another track. Then I'll do the same with the double track and end up with three parts on one track and three parts on another. Then I'll wipe the first two tracks to free them up for other things.



Alternatively you can use one singer and put all the different parts onto different tracks and mix it all down later. Using one singer gives the harmony vocals a character of their own. The blend one singer has with themselves is great.

So who sings what

Let's say you've got three singers. You can either:

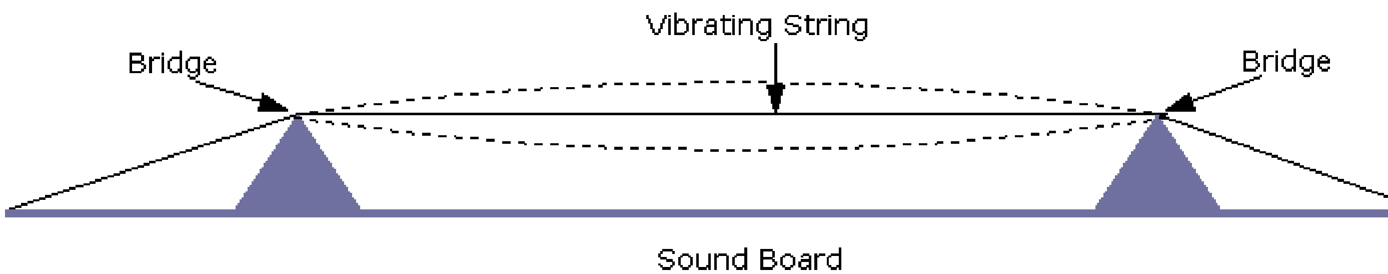
- Give each singer a part each and record three part harmony straight out.
- Get all three singers to sing one part, then all three sing the second etc.

It really depends on the ability of the singers. The first is the hardest because the singers must be good to hold their own part, the second way is really good when you have a band where the backup singers aren't that good plus it's easier to balance.

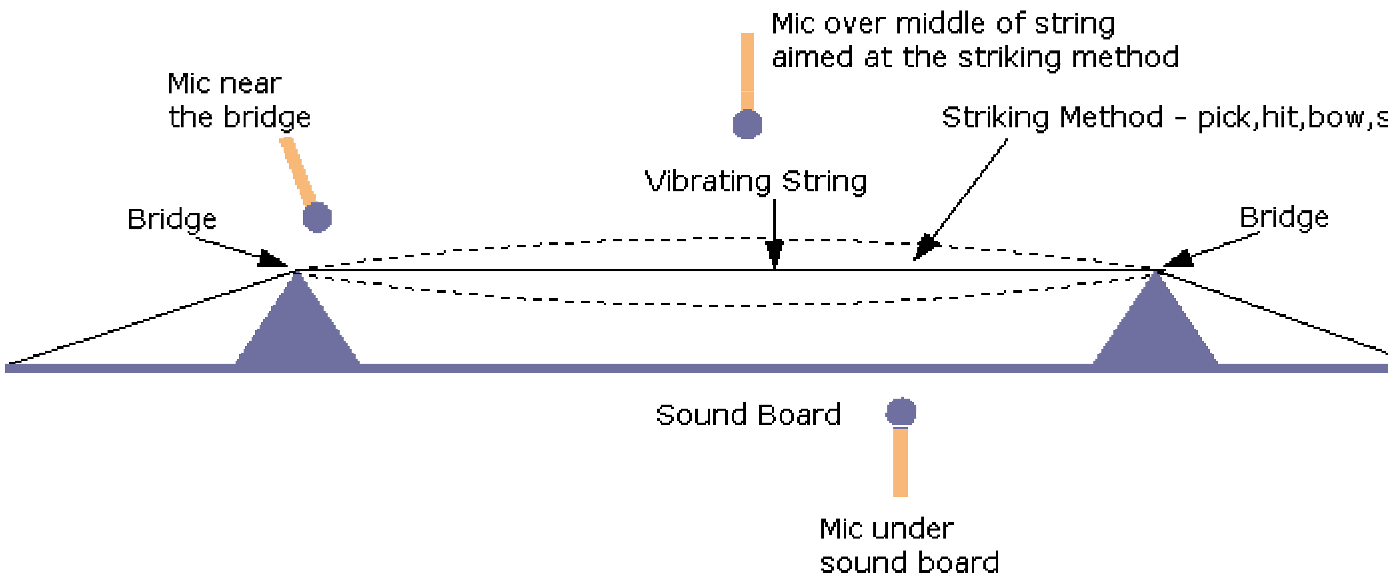
Stringed Instruments

Introduction

So what is specificlly similar about all stringed instruments? Basically all stringed instruments work on the same principle of a tightened string between two points thus:



If you check out a guitar and a piano they are both the same. Both have a tunable string stretched between two points. In the guitar the sound board is the flat front of the guitar where the bridge is mounted, in a piano there is a sound board below the strings serving the same purpose. The sound actually varies depending at which point you place the microphone, so that the sound near the bridge end is different from the sound in the middle of the string:



These three different mike positions all present a different aspect of the sound.

- The microphone over the bridge is harder sounding and has more of the mid frequencies than the low frequencies.
- The microphone over the middle of the string has a full version of the sound and when placed near the striking method area has the transient sound of the string and the strike.
- The microphone near the soundboard has no string impact sound but has all the body of the sound.

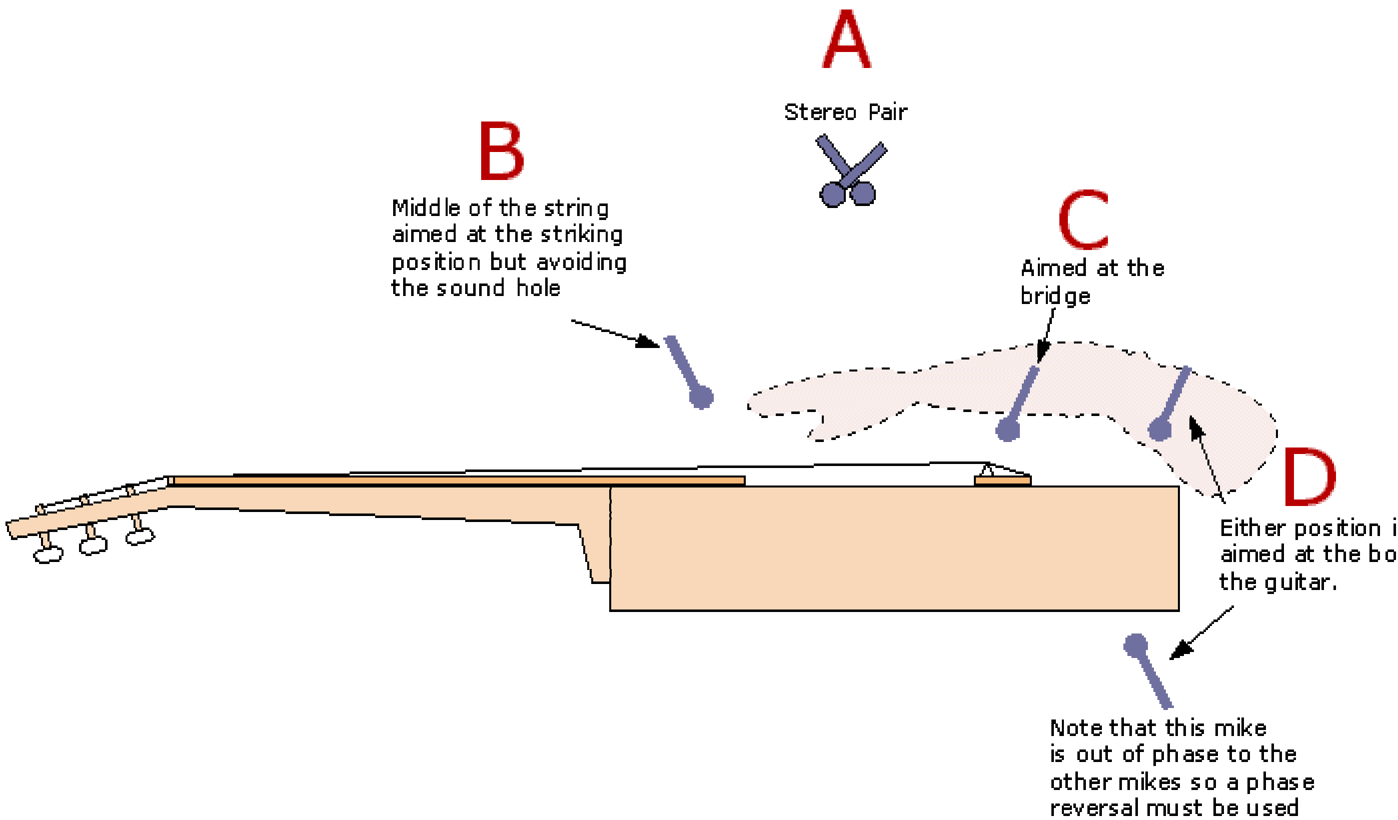
The way the string is vibrated or striking method - also determines how it will sound.

- You can **strum** it as in a guitar.
- You can **hit** it as in a piano.
- You can **bow** it as in a violin.
- You can **pluck** it as in a harp.

Either way, the three positions of miking it will always apply.

Stringed Instrumenrs – Acoustic Guitars

The acoustic guitars is the classic stretched string instrument as it has all the positions where the sound varies. These are the main positions that can be used to record an acoustic guitar.



Lets look at the different positions.

- **Position A**

This is the standard crossed pair stereo miking position. This gives an overall sound of a guitar, it is not a tight presence sound like position B but has the combination of the striking sound, the bridge sound and the body sound. The stereo effect is not very wide but if you want the ambience of the room (like in a full strum rhythm track) it can be appropriate. This can also be a position for a single mike.

- **Position B**

Position B is the most popular mike position and the one I recommend for normal acoustic guitar recording. The mike is placed about 15cm(6") from the guitar pointing at the end of the finger board but not directly at the sound hole. The pick sound is emphasised in this position giving a nice clean attack to the guitar. This position also has the mike pointing away from the fretboard so finger noise is reduced.

- ### Position C

This mike is aimed directly at the bridge and is close around 10cm(4"). The sound here is harder sounding as it has less low frequencies and the mid range sounds are emphasised. If I wish to record a stereo guitar I usually use positions B and C and pan one mike left and the other right. In these positions the stereo spread is emphasised because

1. The mikes are about a 30cm(1 foot) apart therefore increasing the difference thus the spread.
2. The sounds are very different because each mike is recording a different aspect of the guitar sound.

- ### Position D

Position D can be used as an alternative to position C for a warmer stereo sound as it doesn't have the hardness of the bridge sound yet emphasises the warmer body sound. You must note that the mike position at the rear of the guitar causes the mike to be 180 degrees out of phase to the mikes in the other positions therefore a phase reversal must be used.

- ### All Positions!

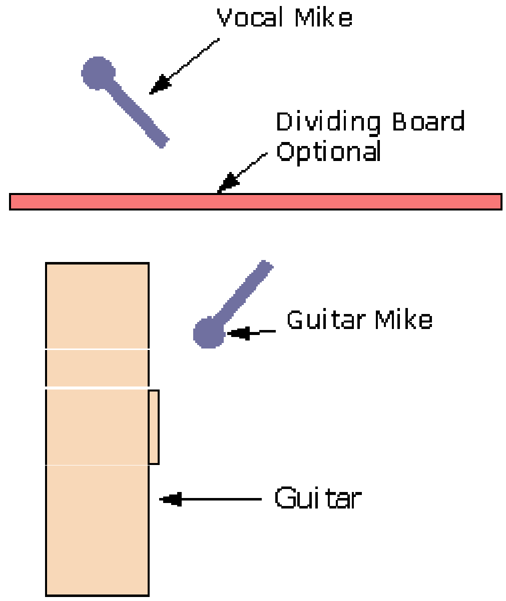
Why not use all positions? If you are about to record acoustic guitar tracks why not set up mikes in all positions and play with the balance of each mike to gain the benefits of each. You might have a great stereo spread between positions B and C yet adding some of position A will add fullness and body, or adding position D panned centre to do the same. Play around, don't just limit yourself to one position only.

Tracking Techniques.

How we wish the guitar to sound in the track determines how we track it to tape and how many tracks we use. Lets look at the various ways acoustic guitars are used.

Solo guitar as in folk singer.

Here you can either use position B and have the track in mono or you can create a stereo track using position B and C. The thing about folk singers is that they sing and play at the same time!! so the guitar mikes are going to pickup the vocal as well therefore any EQ, Reverb etc. that you put on the guitar will also affect the vocal. To get the minimum spill of the vocal into the guitar mike I recommend you use position B and raise the mike so it points down at the guitar at about a 45 degree angle but still in the B position. This tends to put the vocalist off mike to the acoustic guitar mike. (You can also do the same with the vocal mike by having it pointing up at the singer and away from the guitar.) Another method I've seen is to place a soft covered sheet of cardboard or timber horizontally above the guitar that divides the spaces between the guitar and the vocal, but your guitarist must be able to play without seeing their hands!! but it does work.



If the singer is going to overdub the vocal later then you can afford to make more of the guitar sound by recording it in stereo but the singer must play the guitar track without singing.

Strummed Rhythm Guitar.

In this situation you may wish to have a single stereo/mono guitar track or you may wish to multitrack the acoustic. I often double track an acoustic rhythm guitar with one panned left and one right. Another good method is to get the guitarist to play one part through then to put on a capo and play the same chords but in a different position on the guitar. This expands the guitar sound and sounds really good. Some people call it "adding a high strung" You play the first part in say the standard C position and then play the part capoed up to the third fret but play it in the A position. You can go even further , as I have often, and record two tracks in the C position and then do two tracks in the higher capoed position. The effect is a wall of acoustic guitars. You can pan the two high strungs left and right and pan the low strungs half left and right.

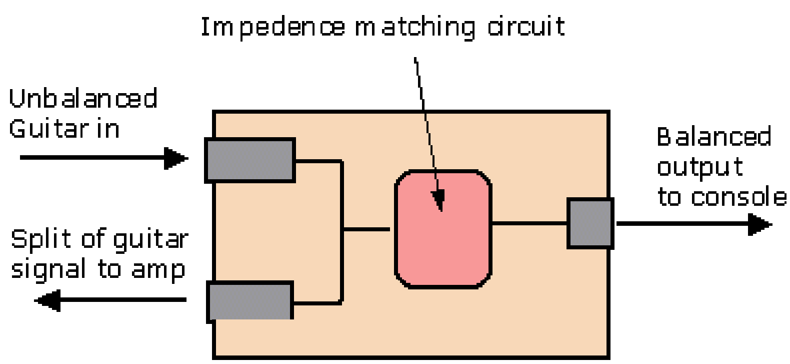
Stringed Instruments – Electric Guitar & Bass

Electric guitars lend themselves to multiple recording techniques. You can put a mike on and amp and leave it at that or you can try all sorts of things. The following options are available:

1. Direct feed from the guitar.
2. Close mike on the amp
3. Ambience mike on the amp.
4. Direct feed from the effects units
5. Room Ambience mike.
6. Second Guitar amp.

Direct Feed.

If you are fortunate enough to have a real-time analyser you will find it interesting to plug your guitar straight into it and look at the frequency response a guitar puts out. The standard Fender Strat peaks at around 7kHz and rolls steeply off from there up whereas the old classic Les Paul peaks at around 4kHz and falls off quickly from there. It's worth noting that factor when listening to the direct sound from a guitar. If you are going to plug the guitar directly into the console you will need a **direct box.**

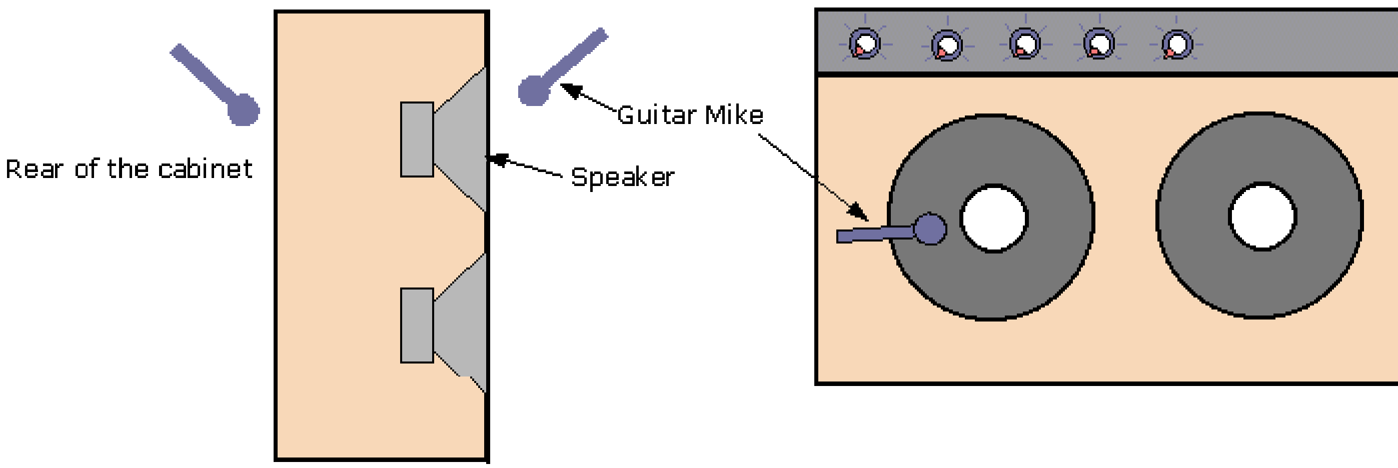


This is a box that matches the impedance of the console and the guitar. A guitar is designed to plug into an amplifier that has a high impedance input whereas a console mike input is designed for low impedance inputs thus the direct box. The impedance matching circuit can be either a transformer - **passive** - or a circuit - **active**. If your unit is an active one it will require power which can be supplied either by an internal battery or by Phantom Power fed from the console mike input. Once plugged into the console have a listen to the sound. You will find immediately that the sound is dull and has no real bite in the top end like we are used to in a guitar so quite a large amount of high end EQ is required to brighten up the sound. You can put the direct feed through some effects units and compress it and it will probably sound better but it won't sound like an electric guitar as we know it . On the other hand a small amount of the equalised/compressed direct signal added to the amp sound can add a soft presence to the sound that is nice in certain circumstances like a soft chorus guitar playing chords etc. To get the full grunt of a guitar you will need an amplifier.

Miking an amplifier.

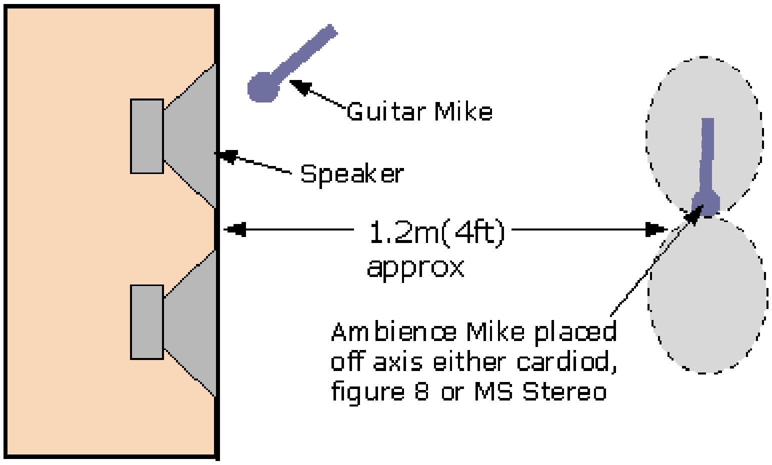
The thing about guitar amplifiers is that they have a huge amount of upper-mid and high end equalisation in the first stage, which is called the **pre-amp**, to compensate for the lack of high end in the original signal. Guitar amps also have addition equalisation on the front panel as an option. This equalised signal is then fed to the power amp and the speakers. Some amplifiers allow you access to the signal after the preamp and before the power amp. It is then possible to take a split of the signal after the preamp , with all the additional EQ, and feed it into a direct box and then straight to the console.

The standard mike technique for recording an amp is to place a mike 10cm(4") from the speaker at an angle.



You will note the mike at the rear of the cabinet. This mike has a boxier sound than the front mike and is 180 degrees out of phase to the front mike so a phase reveral is required. Remember when setting the sound of an amplifier to put your head where the microphone is. The front of a standard amp is directional and if you stand above the amp you won't get the true sound coming from the speaker. The microphone used must be capable of handling high sound pressure levels.

Adding an ambience mike.



An ambience mike will add another dimension to the sound. It can be another cardiod mike or you can us a U87 in a figure 8 pattern. (Very popular) This puts the direct sound off axis to the ambience mike and it also picks up the room ambience. This extra mike can be mixed with the other mike onto one track or it can be tracked to another track allowing you to adjust the balance at the mixing stage. It can also be panned differently than the close mike which gives the guitar sound a stereo sound with more breadth and makes the guitar sound bigger. Alternatively you can use a **MS Stereo** setup.

Using effect boxes.

Most guitarists these days have a bank of effect units setup between the guitar and the amp. You can intercept them by plugging them into the direct box before the amp or you can use your own effects. You must remember that the sound coming out of the DI box will not be the same as the one coming out of the amp because the amp adds all its EQ etc. but a feed from the units can contribute to the sound. Some of the effect units such as the multipedal floor units also operate in stereo and can give you a stereo feed of the signal with stereo effects.

But what about my own effects I hear you say - why should I use that cheap $150 delay stomp box when I've got a $2000 delay unit. This question is a matter of choice - the guitarist might like the cheap effect unit , is used to it and has created a sound around it - on the other hand you may be able to produce a much more diverse delay effect. Remember the guitarist's effects are going through the amp whereas yours aren't. This is where you and the guitarist must play around and try different things. If both of you are into getting the best sound you will get it but if you are both into maintaining your respective egos all hell could break loose.

For more info regarding using effects units go to the pages on **Using Effect Units**.

Adding a room ambience mike.

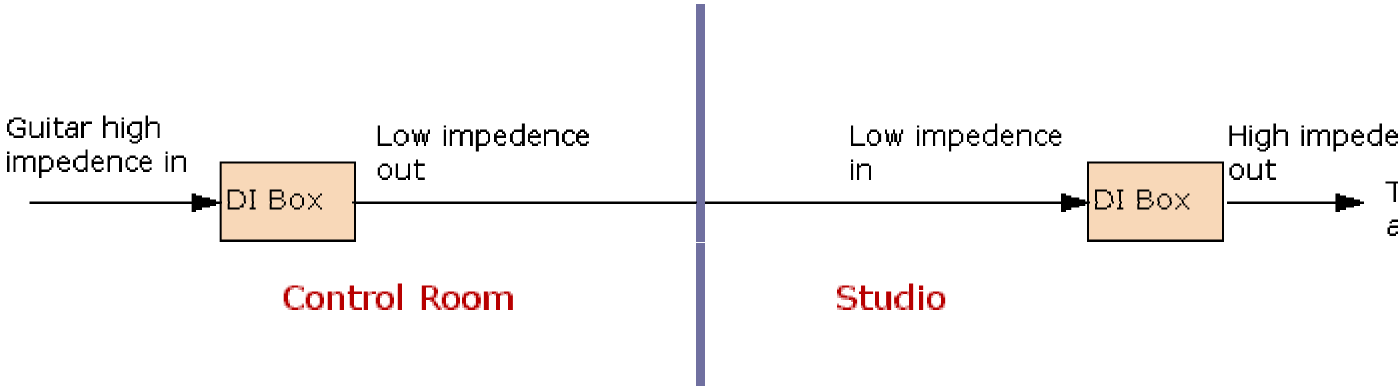
You can go one step further than the close ambience mike and add a room mike (or two). This can give that large grunge guitar an extra beef and for extra effect can be gated so it stops short when the guitar stops. It can be a mike like a U87 with a figure 8 setting or you can use a shotgun mike and aim it at the amp. Considering that sound travels at around 30cm(1ft) per millisecond a mike at 15ft is going to be delayed by 15ms. This could be great or it could be awful - experiment!!

Adding a second amplifier.

You can also add another amp and split the guitar feed into each. If you have a stereo effect system you can split it left and right, mike each amp and put a stereo ambience mike between both amps. You can set each amp up differently, or use two different amps. If miked separately you can achieve a perfect double track as each amp will sound different but have the same signal.

Playing in the Control Room.

Most guitarists like to play in the control room even though their amp is in the studio. This allows them to hear the guitar as it would in the track on your speakers and with any effects that you've added. To enable this you must run a long guitar lead through to the amp. It is worth considering having a plug in the wall that they can plug into that can be picked up on the other side of the wall and plugged into the amp. Alternatively you can run a long lead via the doors - unfortunately guitar leads don't like being long as they loose high frequencies when travelling long distances. One way to stop the loss is to use two passive transformer based direct boxes. You plug the guitarist into one in the control room and then take the low impedance feed out and run that into the studio. In the studio you plug in the other DI box and come out of the guitar input and plug it into the amp. What we are doing here is using low impedance to travel the distance and bring it back up to high impedance to plug into the amp.



You will need a sex change plug from male to female XLR to get back into the second DI.

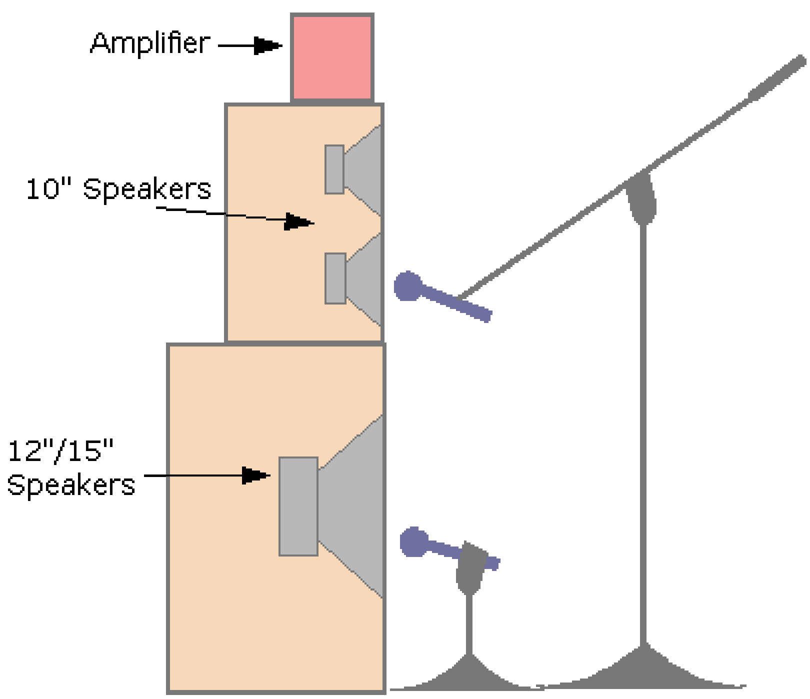
Additional Factors.

There are a few additional factors that must be considered here. I'm sorry but a great engineer can't make a bad guitarist sound great!! There are a few things that can seriously effect the sound a guitarist makes. Firstly, is the guitar setup correctly? Apart from the pickups, model etc. which are set, the variables are - correct alignment of the neck so that the strings are not too low. If they are too low you will experience string distortion caused by the string hitting the adjacent fret, which tends to muddy the sound as the string is not free to vibrate evenly. Secondly the strings used can be too light. A guitar strung with light gauge strings will not sound fat and grungey. A very good guitarist friend of mine says that most people can't play his guitar because it is strung so high and the strings are heavy gauge, but believe me his sound is great. From a musical point of view the guitar might not have the harmonics in tune so that when the guitarist plays up high on the frets the guitar is flat or sharp. All these factors affect a guitar sound but you can't beat the truism that if you want a great guitar sound get a good guitarist.

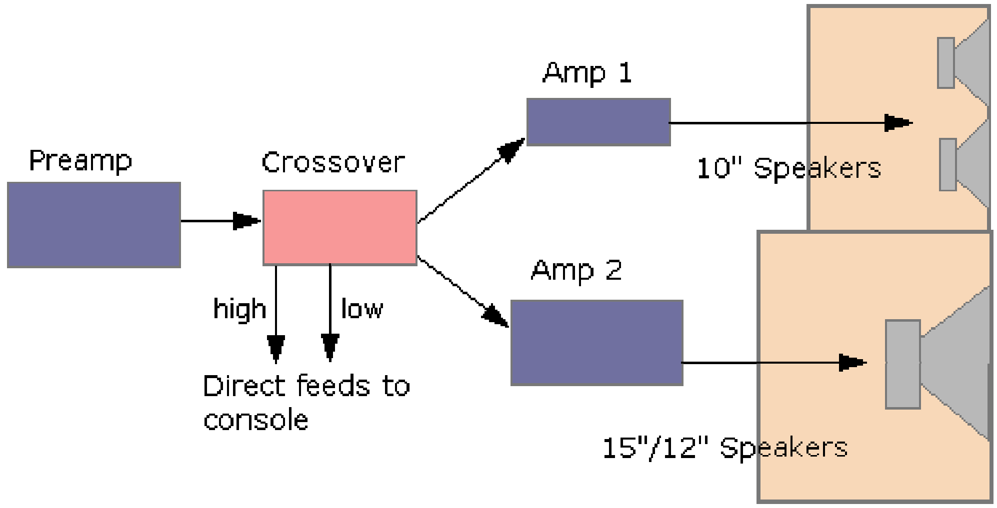
There are many ways of approaching recording the electric guitar. The main thing I believe is to give yourself as many options as you can. Experimentation is the call here, as with the acoustic guitars take the time to put up all the mikes and experiment with the different combinations. I can remember when I was recording an OZ band called Mondo Rock and we wanted a close sounding power chord in a song called "Come said the Boy". The sound we wanted was a Marshall wound up to 11 but recorded close. We tried every mike we had but they all distorted when put so close to the amp, even an SM57 fell over. That day a rep from Neuman came to the studio to try to flog us the new TLM mike. I was reading the specs and it said that it would handle up to 139spl so we asked him if he could leave the mike with us and we'd assess it. When he'd gone we quickly stuck it on the amp and bingo! it worked. The song went on to sit at number two on the charts for about eleven weeks constantly stopped from going number 1 by John Lennon's Imagine. Them the breaks!!

BASS GUITAR

The electric bass guitar differs from the electric guitar in that the direct signal from the instrument does not need special EQ so direct feed via **direct box** is the normal way of recording a bass guitar. Typically most bass amps offer an extensive EQ section and some offer a valve preamp but the bass amplifier is just a dirty big power amp which is required to move the cones of the large heavy speakers. Often a bass amp setup will have two boxes, one with a set of 10" speakers and another with a heavier 12" or 15" speaker. In this setup you can mike each box individually



The bass guitar also lends itself to **bi-amping** where a crossover circuit divides the signal into two or three frequency bands and uses a separate amplifier and speaker for each band.



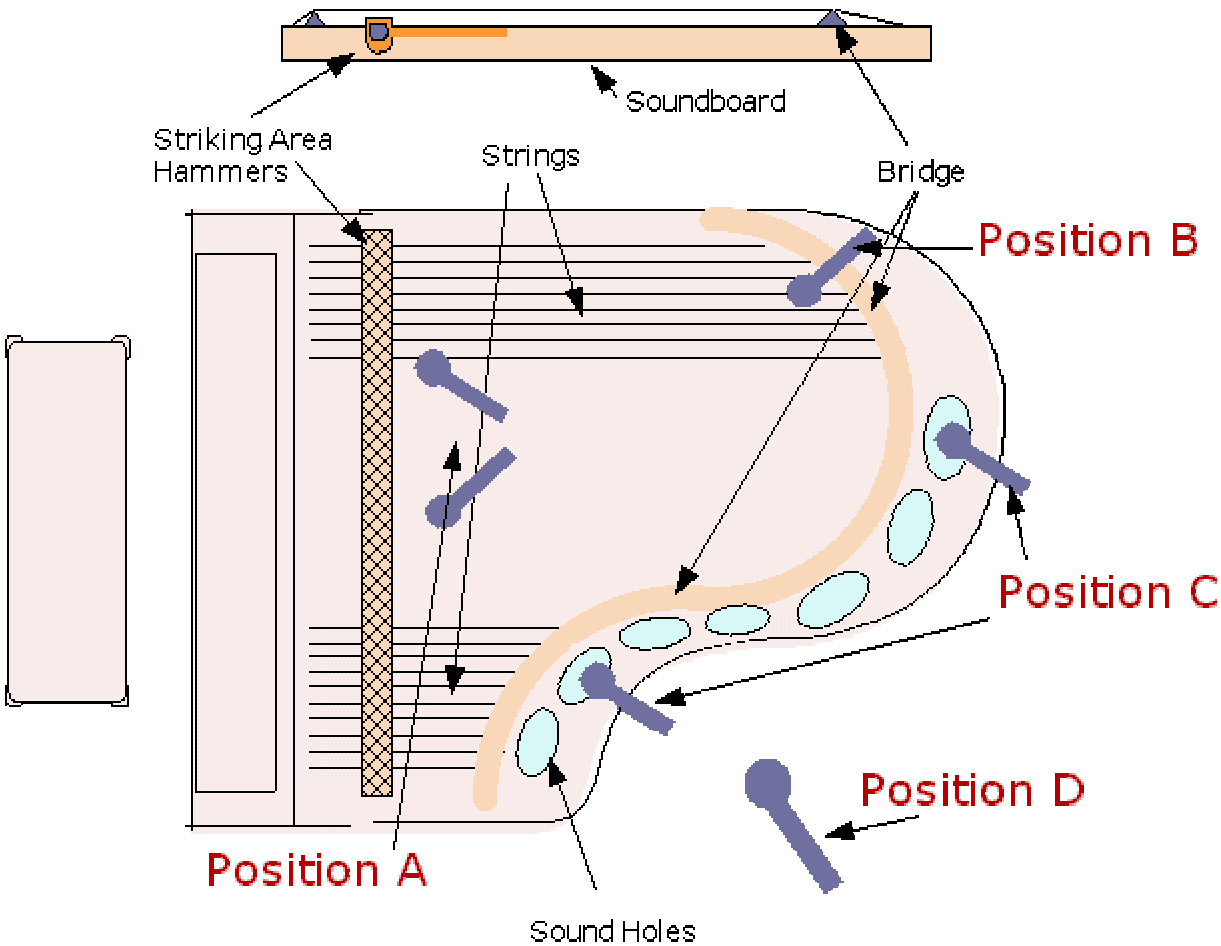
The split from each frequency band is sometimes available as a console feed from the rear of the amp so you can take two/three direct feeds into your console. This allows you to compress and EQ each band separately and assign them to different recording tracks for full control later in the mix. The crossover frequency is selectable in most amps with the crossover frequency usually at around 100 - 150Hz with the 10" speakers handling the high section and the larger 12"/15" speakers handling the powerful lows.

I often feed the bass straight into a DI box and have the player in the control room which helps separation. The bass guitar lends itself to **compression**. The low frequencies it produces contain a lot of energy and containment with compression is recommended.

__Stringed Instruments – Piano__

The Grand Piano

The piano is really just a guitar (or more accurately a harp) lying on its side. It has strings stretched between two bridges, a striking area where it is hit with a soft hammer, and a sound board below.



The drawing above shows the main areas of concern when recording a piano. The mikes can be placed in any of the position A - D as well as underneath. The sound holes give you access to the sound board below the strings. So lets look at each position.

Position A

Position A is the most typical mike position used in studio music recording. It's a stereo pair that is about 150cm(6") apart, placed over the hammers with one pointing to the lower strings and the other directed toward the high strings. They should be about 150cm(6") above the strings. They are placed just behind the music stand. If there is no music involved the music stand can be removed giving a cleaner access to the strings. If these two mikes are placed correctly you can achieve a really good stereo image where the low strings appear from the left and the notes follow to the high notes on the left. Because the mikes are over the hammers the notes are bright and have a nice attack.

Position B

Position B utilises the hardness of the bridge and can be used to emphasise the low strings. I often add a small amount of it to the left of the image to accentuate the bass strings. Great if you have a 7 or 9 foot grand!!

Position D

Position D is the traditional Classical way of recording a grand piano and is still used today when recording grand pianos with an orchestra. It can also be used to add body and warm to a position A setup.

Position C

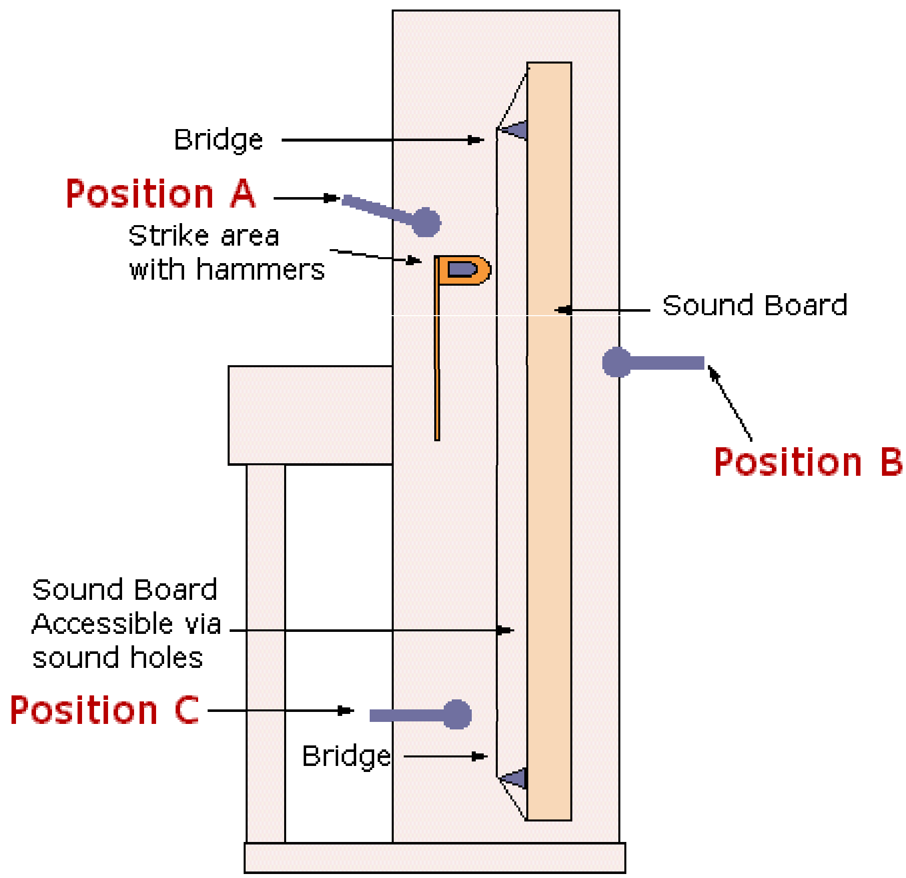
Position C is a position that accesses the sound board. The level coming off the sound board is quite high so it is a good position if you are caught having to record a grand piano in a studio with other instruments and you want separation from the other instruments. Two mikes places in the sound hole s allow you to lower the piano lid to the lower stand and with a couple of blankets or sleeping bags thrown over the lot you will get good separation yet a clean sound that will sound even better with a bit of high shelving added. Another way of accessing the sound board is to place a mike under the piano pointing straight up. This is often used in TV where they don't want the mikes to show.

PZM Mikes

There is one more way of recording a grand and that is to use 2 x PZM mikes fixed to the lid and then the lid closed. This produces a beautiful clean sound and is also great if you have spill problems, hence there use on stage shows. Give me two good Neumans or AKGs and I'll go with them anyday though. The AKG 451 is my favourite.

The Upright Piano

Unfortunately most home studio owners don't have a grand piano but lots of you have an upright. So what's the best here - well - treat it like a grand. It has all the same spots.



Here we have a typical upright piano with the typical three positions. To access some of these positions you may have to pull the piano apart. The front panel above the keys can easily be removed as can the panel below the keys. The easiest and simplest is to simply drop two mikes on boom arms through the top and set them up as a stereo pair as in the grand piano over the hammers and about 15cm(6") apart and pointing left right. This placement is easier if you can remove the front panel.

Position A

This is the standard position as described above and can be supplemented with either a rear soundboard mike (out of phase) or a lower mike in a sound hole under the keys. In this case the lower front panel must be removed. Try and avoid getting the lower mike too close to the pedals as their sound will become annoying.

Position B

Position B is the one used on the old TV shows where they didn't want you to see the mike but it also has a lot of body and warm in the sound so when incorporated with position A it can be helpful.

Position B

Position C is a variation of position B except that it can also incorporate the harder bridge sound.

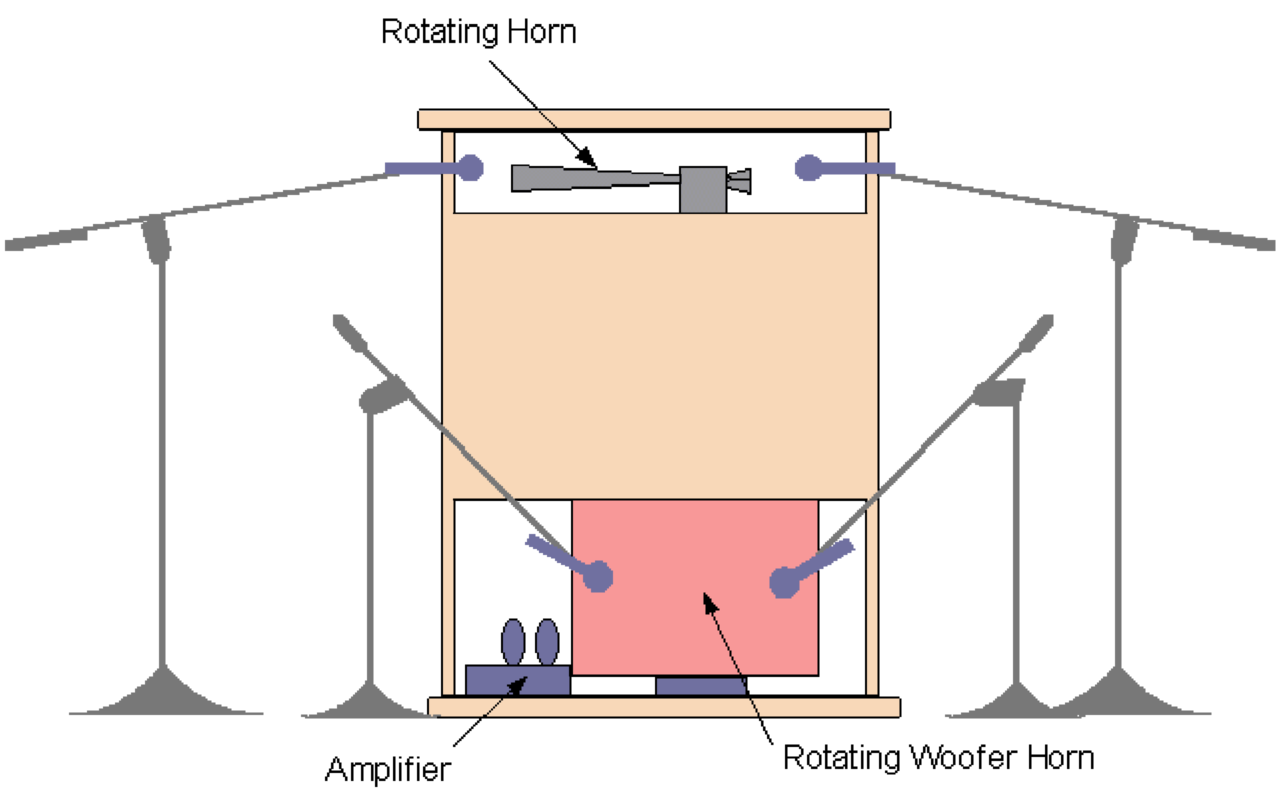
Personally I would go for position A every time and would only use the other positions to supplement the sound or because I can't get into the piano and can't take off the front panel.

PZM Mikes

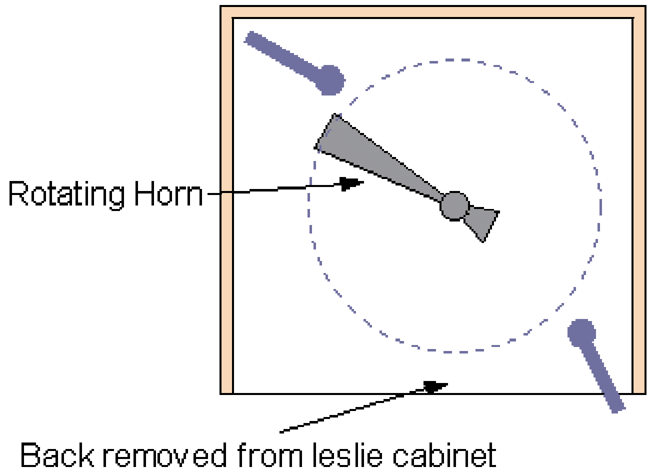
Once again two PZM mikes strapped to the front panel at the height of the hammers will work very nicely indeed.

The Organ

The **Hammond Organ** is another beast altogether and although it's not a stringed instrument I'll include it here. The Leslie box consists of a divided cabinet. In the top section is a rotating horn covering the high frequencies from around 800Hz up while below is a woofer cabinet covering the lows. The woofer also has a wooden horn shape that rotates. This is how I like to mike a Hammond Leslie Box.



The rear of the Leslie cabinet will need to be removed, its only a few screws. The microphones are basically two stereo pairs which you pan L/R. If you wish to be really mad you can carefully put one of the high frequency mikes into the cabinet like this:



This really gives a great stereo effect and the Leslie rotates in your head with headphones. The top mikes are totally 180 degrees out of phase but who cares, the effect is great.

Incidentally, if you want that incredible Emerson Lake and Palmer growl from the Leslie, remove **one** of the large output valves that I've drawn in the picture above. Some people have modified their Leslie cabinets so you can plug a guitar directly into the valve amp so you can get a real Leslie effect on a guitar. If you also remove the output valve you'll get the wildest guitar grunge!!

Stringed Instruments – The Rest

The Harp

The harp is the ultimate stringed instrument and I sympathise with all the harp players as it is the devil of an instrument to tune. They say mischievously that harp players spend half the time tuning up and the other half playing out of tune! (sorry harp players)

There are basically two ways to record a harp. If it's on its own as an overdub a simple high quality Condensor mike 30 - 60cm(1 -2 ft) from the instrument aimed at the striking point (hands) will cover it fully. You can also use two mikes as a stereo pair similar to the position A in pianos aimed at the top and bottom strings respectively and as close as the player finds comfortable.

I used to often have a harp within a big band/strings situation where it was so quiet relative to the rest of the brass etc. that a mike in this position was 80% spill so I had to find a better way. What I found is the other way of recording a harp using the sound board as the source. Like the holes in the frame of a grand piano the harp has a series of sound holes down the back. If you get a quality mike and wrap it in a cloth you can jam it into the centre hole. The cloth will hold it tight and stop any handling noise and you will get a good level signal that with a little top EQ will work very well and have a lot less spill and you'll be able to mix those beautiful glissandos over the brass. A combination of both mike positions will give a fuller richer sound if used together.

The Banjo and Mandolin

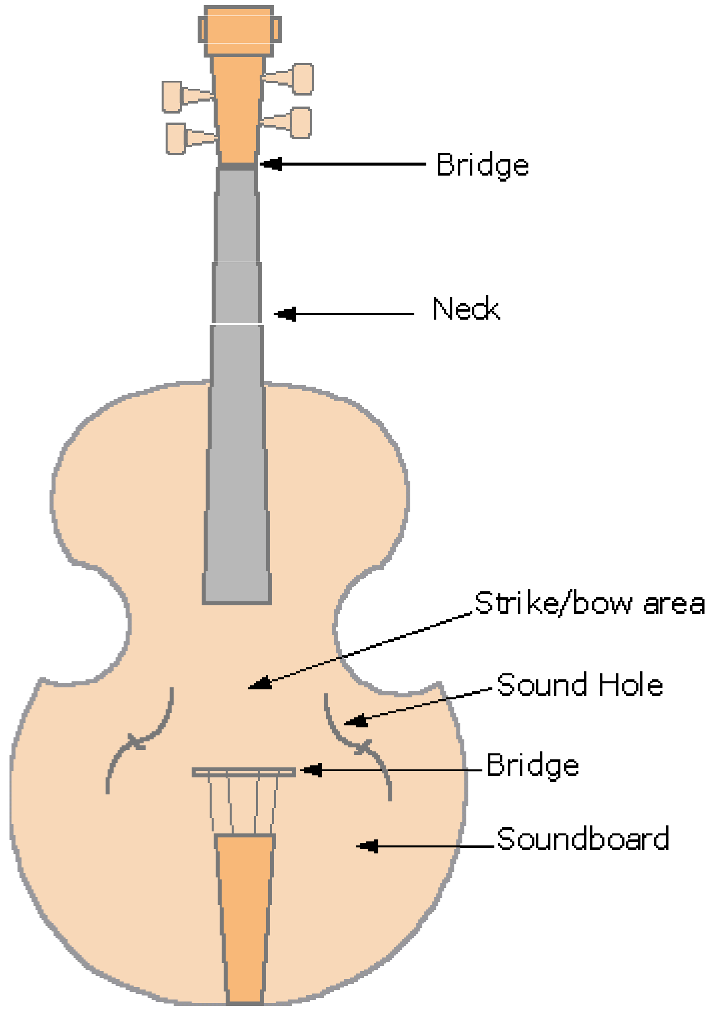
The banjo and mandolin are similar to the acoustic guitar and both have the same points - strike area, bridge and sound board - thus the same mike positions apply. I don't like to get too close to a mandolin or banjo because their sound doesn't fully develop until around a foot or two away.

The Dobro

The Dobro and lap steel both have a recording problem mainly the string noise as the slider moves up and down the strings. If you think of them both as guitars on their backs and place the mike in the acoustic guitar position B where the mike points back towards the striking position you can put the fingerboard off axis to the mike and thus reduce the slide noise. You can also mike the sound board and bridge as per the acoustic guitar.

The Violin Family

The violin, cello viola etc. are all the same except different sizes.



They all have a strike point (bow area), a bridge and a sound hole and soundboard. For the violin and viola the typical miking is to place a quality mike 30 - 60cm(1ft - 2ft) above the instrument pointing down aimed at the strike area. This gives a balance between the bow, bridge and soundboard sounds. One technique I have tried is to mike the violin from underneath as well as overhead with the bottom mike in the opposite position to the overhead mike and phase reversed. By adding a little of the under mike you can add body and warmth to the sound because you are adding more of the soundboard sound. It can also be said that a violin player should be on a reflective floor as opposed to carpet because the sound emanating from the soundboard will reflect back off the floor and add to the fullness of the sound.

The cello is the same with the mike out in front of the instrument pointing at the bow area. Additional close mikes near the bridge and the sound board/hole can be used for effect if required. Again a reflective floor is recommended.

The Acoustic Bass

I have singled out the acoustic bass because it is one of the hardest instruments to record in my opinion. Being a classic stringed instrument it has all the sound areas - bow area, bridge, and soundboard and soundhole. It depends on the style of music as to how you mike it but the hardest is the straight plucked jazz bass. The traditional technique is to use a good mike (preferably with a large capsule like a U87 or U49 and put it about 5 - 10cm(2" - 4") from the bridge. This will emphasise the attack of the fingers with the added hardness that the bridge sound has. Another mike can also be added that is aimed at the sound hole which will emphasise the warmth and lower frequencies. A mix of these two should cover it nicely. Many bass players have an electric pickup on their bass and a combination of direct pickup and mike works well as the pickup adds presence.

Be very careful about the low end of the sound. It may sound silly but quite often to get a good bass sound you have to remove bass from the signal. A low end rolloff from around 80 - 100Hz can stop the bass from sounding muddy or a dip around the low mids at 200 - 300Hz will also work. There is a lot of energy in the low end of an acoustic bass and reasonable compression can help to contain it.

Effects

Introduction
Delays
Reverberation
Equalisation
Compression

Introduction

Without doubt the biggest influence on recording styles over the past 25 years has had to be the introduction of digital effect units. When I started there were only four effects available - tape delay, reverb chamber, reverb plate and spring reverb.

I remember when the track Itchycoo Park by The Small Faces (I think!) came out and had tape phasing!! Wow I remember desperately trying to figure out how they did it. It wasn't till I joined Armstrong's Studios in Melbourne under the great Roger Savage that I learnt how to achieve it and I used it on the now classic Australian track "The Real Thing" by Russell Morris. Now you can dial it up as an option on almost any effect unit and you can even get it in a stomp pedal!! but it's not the same.

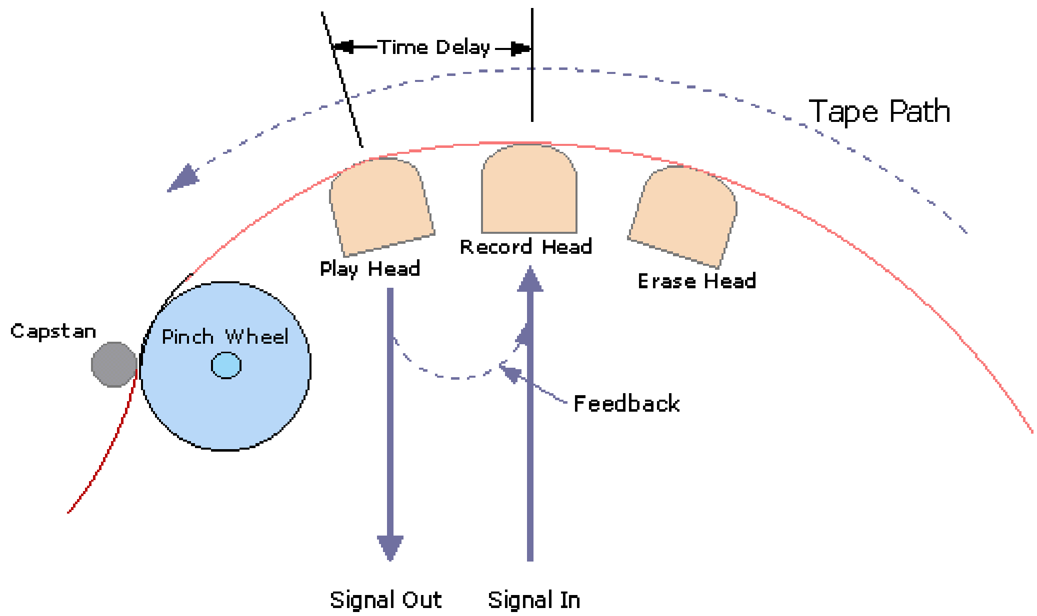
Tape delay was limited to how fast the tape machine would go so it was usually limited to quarter and eighth note delays around the two speeds of 7-1/2 (which gave 16th delays around 120bpm) and 3-3/4 ips speeds (which gave 8th delays) which were standard speedsn on the tape machines then. It wasn't until the introduction of the digital delay that suddenly the whole gamut of sound effects that work in milliseconds appeared and brought us phasing, flanging, chorus.

Then came the first of the digital reverbs and a whole new world opened up. Suddenly you could change the reverberant field around any instrument quickly and easily. To explain how all the effects work and how to use them I will look at it from the perspective of how these effects came about because a lot of the terms relate back to those days. For example the term flanging came from the technique of slowing down a tape machine to create phasing effects by holding onto the flange of the tape reel. Silly isn't it!

Effects – Delays

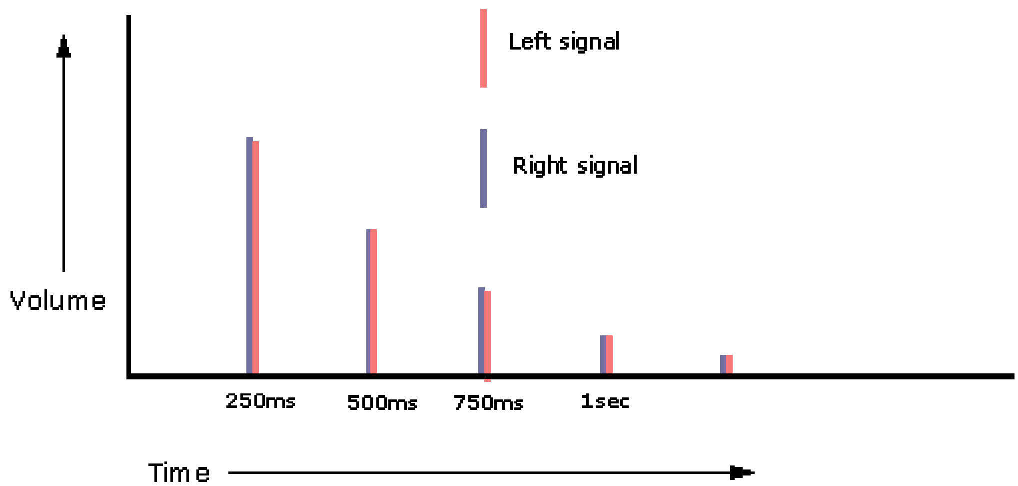
Tape Delay

The first delay was created using a tape machine. The following is a drawing of the tape path in a typical tape recorder.



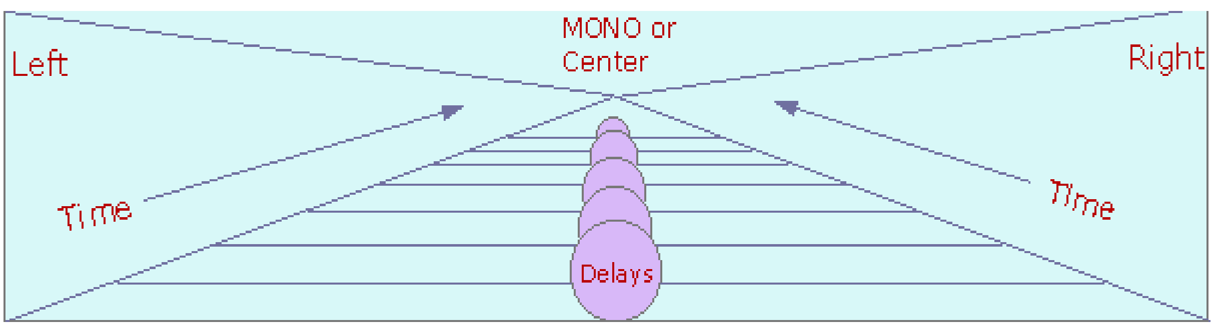
The tape first passes the erase head that wipes the tape. Then it passes the record head where the signal is put onto the tape. Finally it passes the play head where it is played back. The time taken for the tape to pass from the record head to the play head determines the delay. If the tape recorder is going fast the delay will be short, if slow the delay will be long. The speed of the tape is determined by the speed the capstan motor is turning and in the early seventies the tape machine manufacturers started to add variable speed motors with a control called **varispeed**. Using this you could set the delay (by ear) to the appropriate speed. If you took the output from the play head and fed it back into the input of the recorder you would get repeat after repeat as it cycled around. As only a small amount of the first delay was fed back it would continue round the loop dropping in level each time and the classic delay was created. This control was called **Feedback**.

If you were using a stereo tape recorder you could have stereo delay. If you put in a mono signal you got out a mono delay because each side of the delay was the same. Thus:



Which sounded like this visually!

Click on the Image to hear the sound

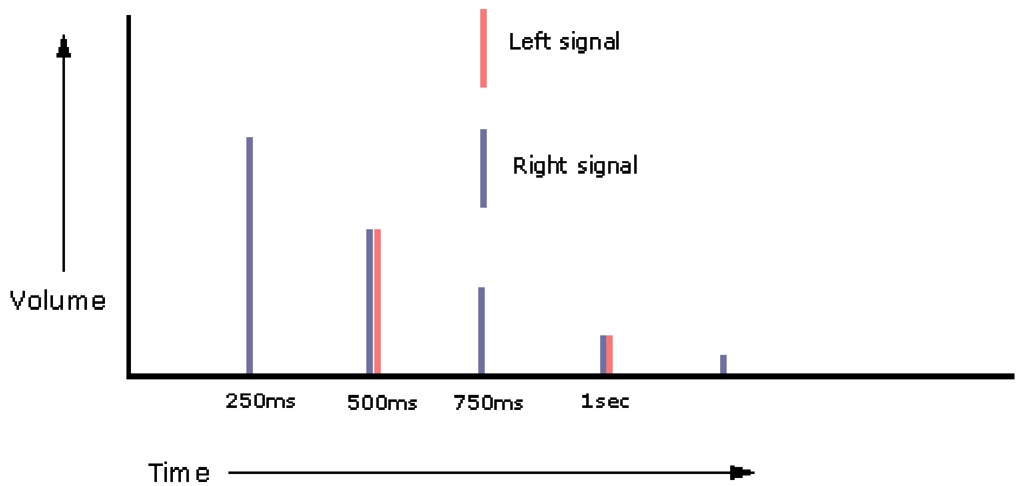


Because both sides have the same signal the delays sound mono in the centre.

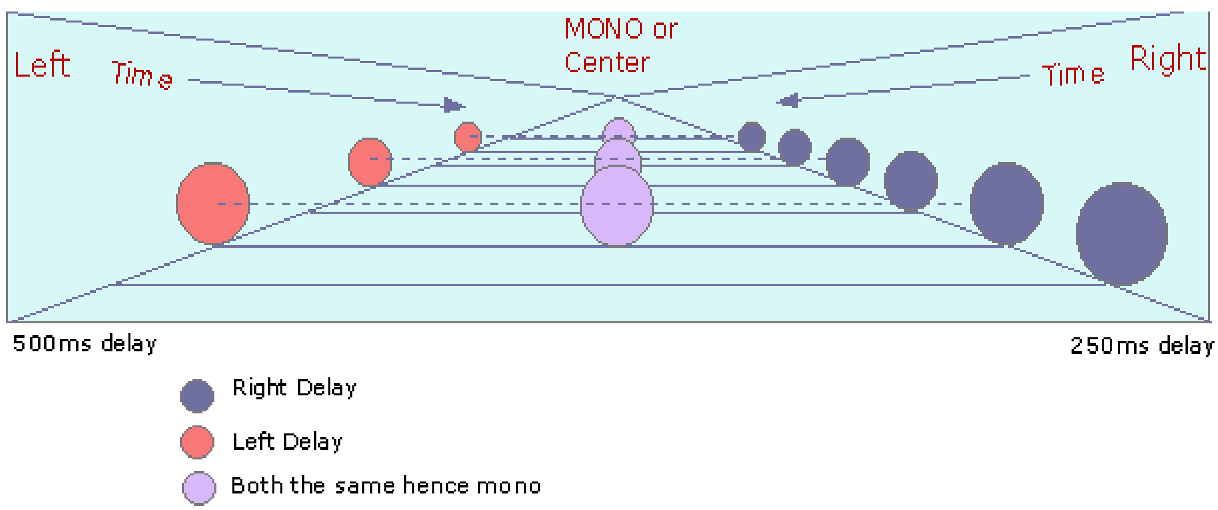
Enter the Digital Delay Unit.

Digital Delay

The digital delay unit changed everything. Firstly you could dial up the delay you wanted but more importantly you could vary the delay of the left and right sides. So if you put in a mono signal and set the left side to 500ms and the right side to 250ms, applied some feedback and you ended up with a delay in which the left side was different to the right side and true stereo delay was starting to be a reality.

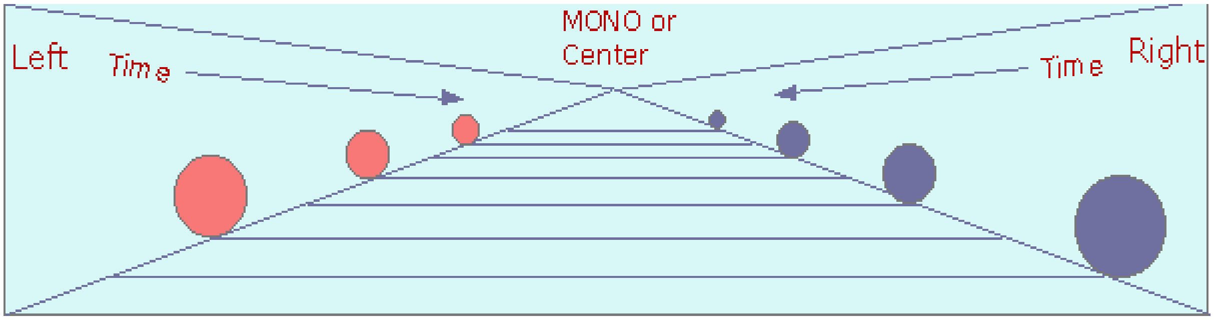


Example 1



I hope you are following me here - what I'm trying to do is to get you to picture the delays as they would be heard. On the left side we have a 500ms delay while the right is at 250ms. Every 500ms the left and right delays are the same - therefore the sound comes from the middle. What you think is stereo really isn't, what you are wanting is something like this:

Example 2

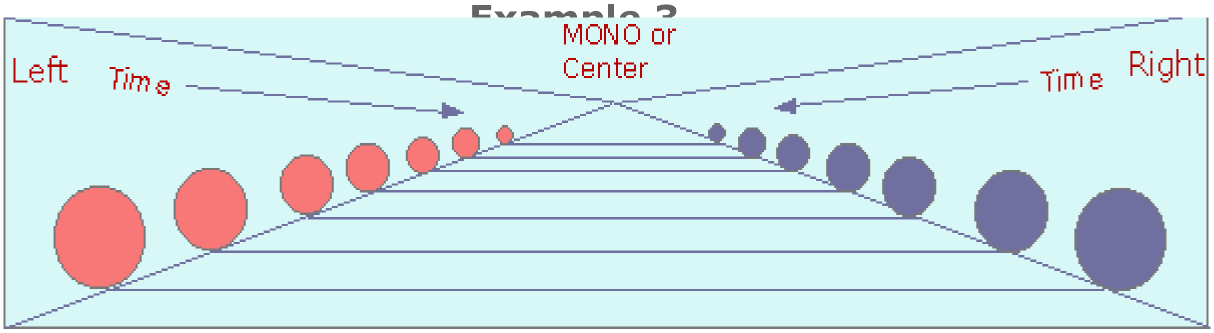


Right Delay

Left Delay

Both the same hence mono

Example 3



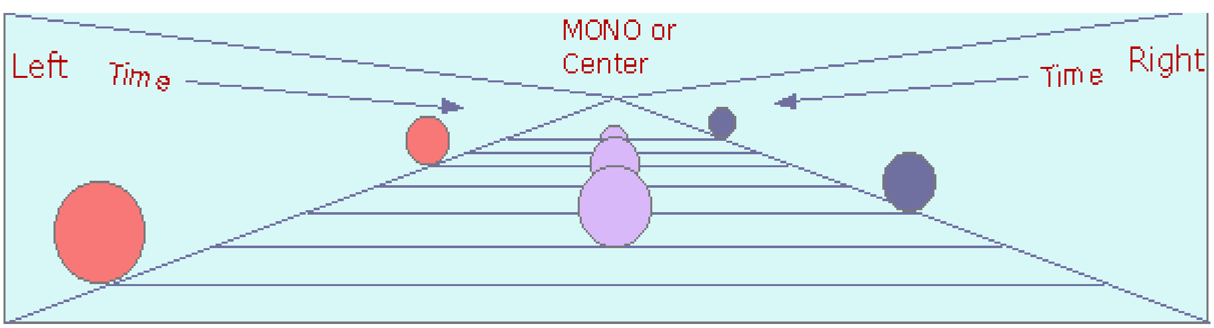
Right Delay

Left Delay

Both the same hence mono

or this!

Example



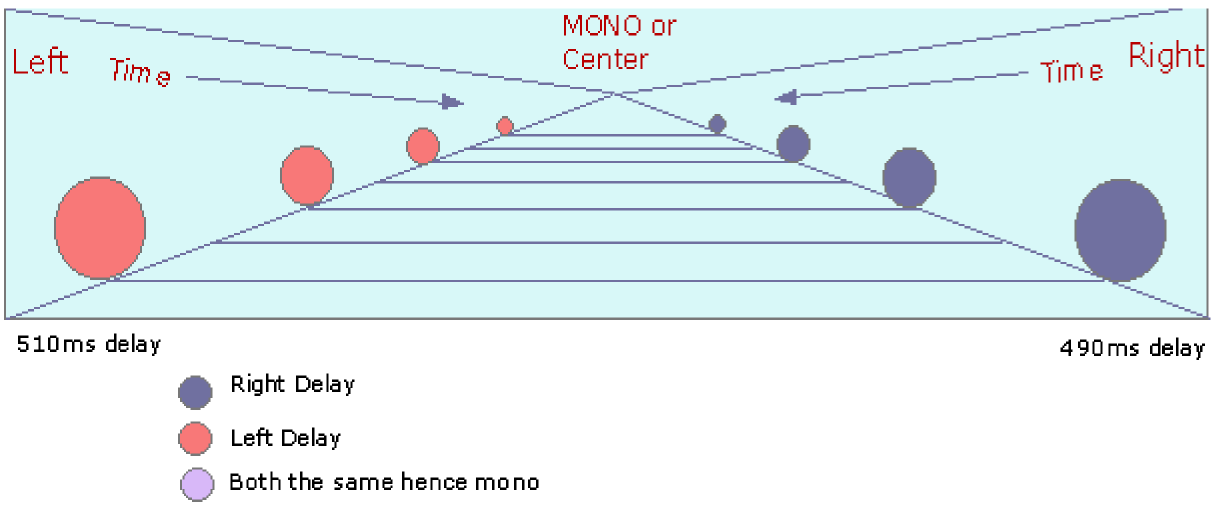
Right Delay

Left Delay

Both the same hence mono

This is where the understanding of what makes things stereo and what is mono is extremely important. The thing that makes a sound come from the left can be more than just it being a mono signal coming from the left. It can be a stereo signal, where the left is different from the right, yet appears to come from the left. If we were to set the delay so that the left delay is 510ms and the right is 490ms the delays would be 20ms apart, right? If two sounds are 20ms apart or greater they sound like two different signals left and right - so if two delays are 20ms apart then they should sound as though they come from left and right. Thus:

Example 5 Click on the Image to hear the sound



Sure the left delay will be 20ms later every delay but in 4 delays that's only 80ms out of 510ms and remember relative to the beat one delay starts 10ms ahead whilst the other is only 10ms behind. Try it with a track as you'll see what I mean.

Another way to make each side dissimilar is to change the **pitch** of one side or both sides. I created **Example 3** by doing just that - I changed the pitch of one of the sides so even though they are in time they appear as stereo because they are dissimilar.

I created these delay sounds using a **Multi-Tap delay**. Instead of using feedback to create the repeats as with tape and straight digital delay, in a Multi-Tap Delay you can control each delay. If you imagine that each delay is a Tap, you can set what each delay will be at each tap - even where it is panned. This is much more extensive a control of the delays than using straight feed back where each delay is just a repeat of itself.

You can see how I've played around with aural visualisation and that's what the guys and gals who make all those incredible delay programs do all day. Next time you get one out try looking at the sound it makes and **picture** what's going on instead of just listening to it.

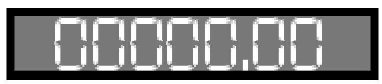
sound has depth,height,breadth

Setting the Delay time

Setting the delay time depends on the **Tempo** of the track you're recording. If the tempo is 120 beats per minute there are 120 beats in 60 seconds or 120 beats per 60,000 milli seconds which is one beat every 500 milli seconds. So with 4 beats in a bar, quarter beats are 500ms, eighths are 250ms, sixteenths are 125ms etc. So how do you find out what the delays are if you know the tempo? There are some computer programs like Beat Calc that will automatically work it out for you, some of the new delay units and programs have a **tap** function that allows you to tap in a tempo and the device will work it out and you can get charts with it printed out. They will not only tell you what the 1/2, 1/4, and 1/8th beats etc. are but will also tell you the dotted note delays and the triplet delays. I've created a chart for you called the **Tempo Chart**

Quick Delay Calc

There is a quick way you can work out the delays of a track using a stopwatch that reads 100th of a second.



Play the track and start counting the quarter beats. Then start the stop watch on the beat and count ten quarter beats and stop the clock on the eleventh beat. You will get a reading like:

Thus a quarter note beat will be 460ms, an eighth will be 230ms and sixteenth will be 115ms etc. This is a handy technique if you are a PA mixer and you want to put a delay on the vocal and you quickly need to work out the tempo the band is playing at.

ENTER THE MODULATORS

The Phaser/Flanger

When a jet flies over, you hear it coming and its sound is going up in pitch - then when it goes away its pitch drops. This effect is called the **Doppler Effect.** If you had a very tight delay of 10ms but could change it using a **modulator** so it varied from 0ms through to 10ms and back to 0ms etc. sweeping forward and back relative to the original signal. When the delay is increasing the phase shift is increasing and the doppler effect will cause the sound to lower in pitch like the jet flying away and when its modulating back and shortening the delay the phase shift will cause the pitch will rise like the jet coming towards you.

That is the classic phaser sound. The effect was originally created on short wave radios where a receiver was picking up a signal that had come around the world one way as well as another longer way and when the two were added together at the receiver they added and subtracted from each other causing the **phase shift** or **Comb Filter Effect** that creates the sweeping pitch effect we now associate with phasing, and which is also why they are called phasers. **Phasers** shift the phase relationships within the sound using phase shift circuits. **Flangers** are the same sort of thing, except that flangers use **time shift** circuits to obtain the effect. The modulator has controls like

- **Delay** which is how much delay do you want,
- **Depth** which is how much do you want the modulator to control the delay,
- **Rate** which is how fast or slow do you want the modulator to oscillate,
- **Feedback** which is like the tape feedback, sending the signal back into itself and
- **Shape** which is how do you want to drive the modulator, i.e. with a sine wave shape or a triangle wave shape etc.

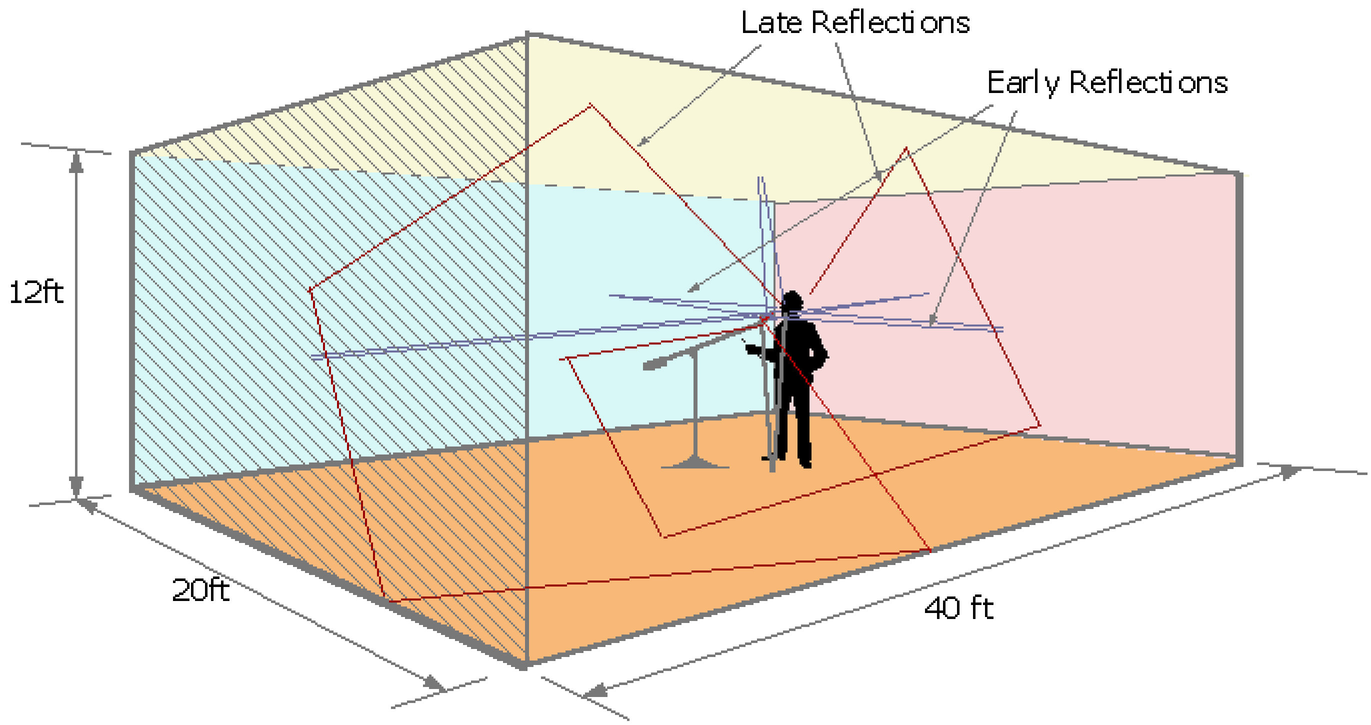
The Chorus Unit

If you increase the delays from the 0 - 10 ms area and go out to the 60ms - 80ms delays but still modulate the delays the effect changes to the **Chorus** effect.

The guitar in this track has the classic chorus guitar sound most of which was created using the classic Roland Dimension D, in fact there were two of them. The modulation rates are usually faster than phasing but the depth is a lot less so there is only subtle change going on. If you look at the controls on a chorus unit you will find the same controls as in phaser units, Delay - Depth - Rate - Feedback - Shape. You can set the delay on a chorus to be in time with the track so if the tempo is 120bpm the 16ths are at 125ms and the 32nds are at 62.5ms - or you could try 62.5ms and 31.25ms as well. It really makes a difference.

Effects – Reverberation

Imagine someone singing in a large room with painted concrete floor, walls and ceiling. Where is the sound going and what is the microphone hearing?



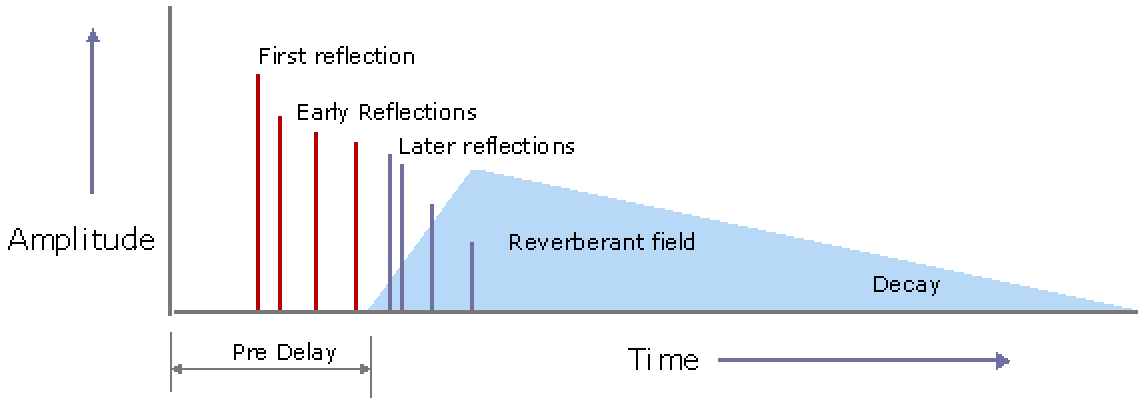
Every sound that leaves the singer reflects off the walls, floor and ceiling. Initially the sound from the singer will reach the microphone first - followed by the first reflections. In this instance the first reflection would be from the floor followed by the ceiling as they are the closest, then the walls on either side followed finally by the reflections from the walls in front and back of the singer. These reflections wouldn't stop there, they would go on and on. Then would come the longer reflections where the sound has bounced off the ceiling, hit a wall then the floor and back to the mike. The time taken for the first reflections to arrive back at the microphone is proportionate to the size of the room. Sound travels at 30cm(1ft) per millisecond so if the singer was equidistant from the side walls and they were 20ft apart the first reflection from those walls would be delayed by 20ms. If the singer was 20ft from the end wall those reflections would arrive at the mike 40ms later. Then the late reflections would start arriving but by them the reflections would have built up and up until a reverberant field was established where none of the reflections were distinguishable and true reverberation has occurred.

Because the room is made of painted concrete there would be a good reverberation of around 2.21 seconds (According to the **reverb calculator**). The walls of this room are flat and reflective so there is nothing on their surface (or the floor or ceiling) that would scatter the sound around like if they would if they were made of river rocks, or were covered in triangles and boxes etc. So the reflections within the reverb are slowly bouncing around and decaying. If the walls were made of rocks the reflections would be going all over the place and there would be a mass of differing reflections.

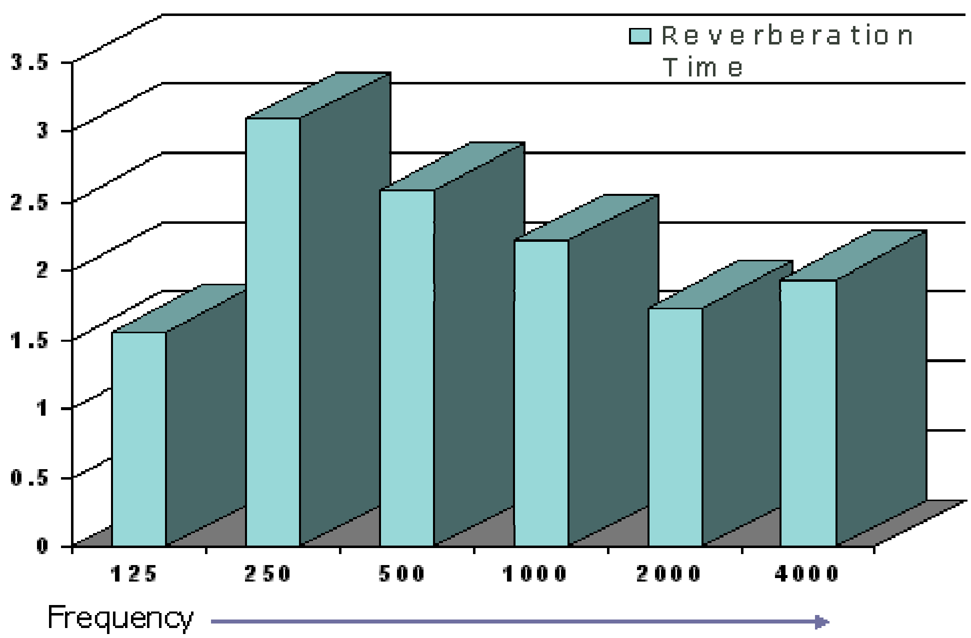
The reflections would be more dense or diffuse and we would say that there was more diffusion.

So what does the microphone hear?

A **Singer** followed by **Reverb** created in a **Room** (not a hall) of a particular **Size** and made up of a **First Reflection** followed by the **Early Reflections** and the **Later Reflections** and finally by the **Reverberant Field** that arrives after it's **Pre Delay** and has low **Diffusion** but creates a **Reverberation Time** of around **2.21 seconds**



But there's more. From the reverb calculator you can work out the reverb time at different frequencies. The room above comes out like this:



The 2.21 reverberation time noted before was at 1000Hz whereas at 250Hz it's 3.09 seconds. In other words the decay at 250Hz is longer than the decay at 1000Hz.

Some of the reverberation units and programs give you individual control over the reverb time of the high and the low end. Others allow you to EQ the reverb to roll off or boost the highs or lows.

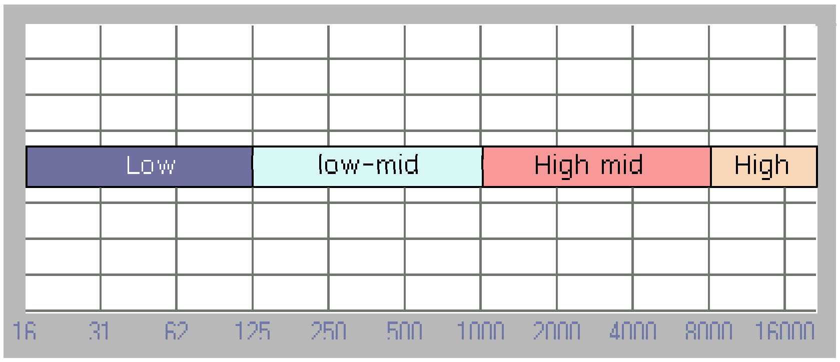
Anything more and you are in the hands of the programmers who write the reverb, hall, bathroom programs, or are you ?

At 120 bmp one bar lasts 2 seconds. 32nds are 62.5ms, 16ths are 125ms, 8's are 250ms and quarterbeats are 500ms. Sound like good predelay and early reflection times to me. Set the reverb at 1 second and it'll last half a bar. At least you can get it in time.

Before there were plates, halls, rooms etc. there were reverberation chambers. These were rooms specially build for their reverb. They were build under the studios like Abbey Road in London and Capitol Studios in LA. They were (or is it are?) large rooms designed to create a uniform reverberant field. A speaker (or two) were placed in them and a microphone (or two) was setup to pick up the sound. You used it like you would a reverb unit today. You sent a feed to the speaker and mixed the return with your track. But any area can act as a reverb chamber, a stairwell is pretty good, a garage, the local hall, a concrete water tank is a beautiful reverb chamber, a large concrete pipe with a speaker at one end and a mike at the other. If you want to experiment there's lots of ways of creating reverb and now that all the recording gear is so portable, why not take the drummer out to the local hall one day. The reverb you get will at least be distinctive and with good ambience mikes you can control most of it.

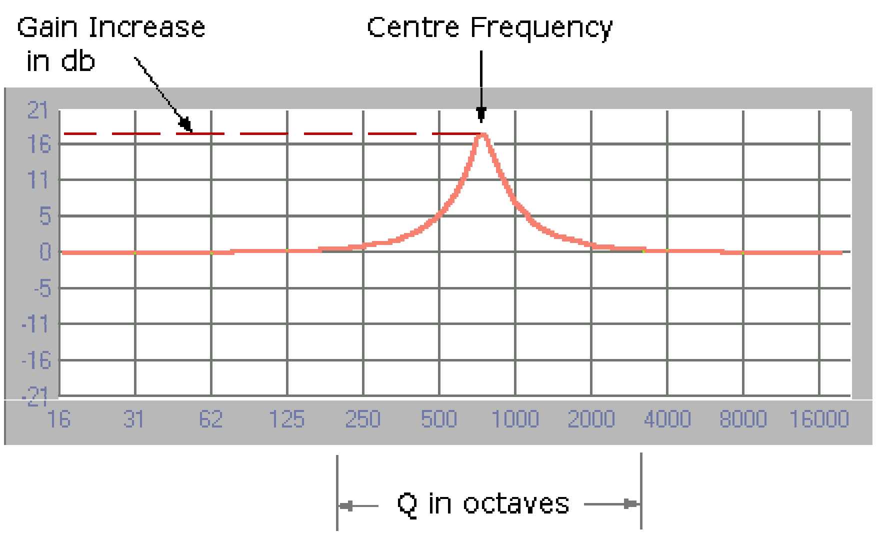
Effects – Equalisation

Whilst compression effects dynamic range, equalisation (EQ) controls **frequency range**. The frequency range of sound is shown in the chart below and is divided up into four **bands**.



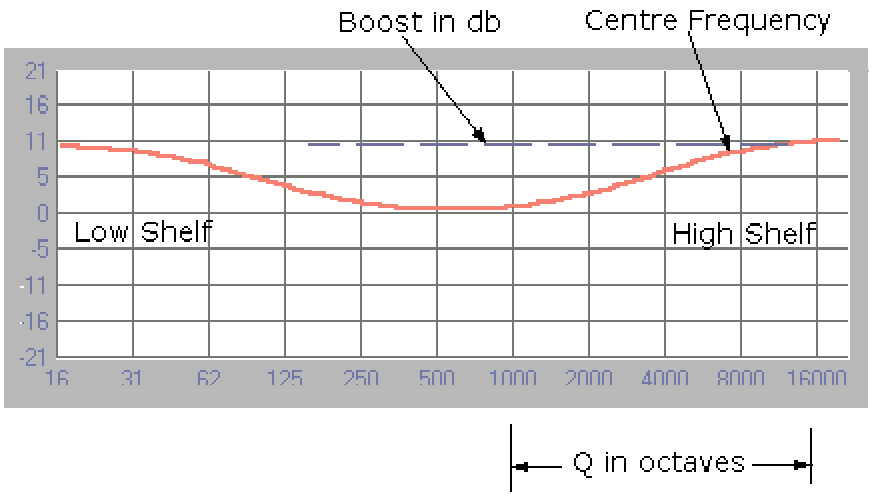
The scale along the bottom of the chart shows the frequencies from 16Hz to above 16kHz. When engineers talk about the high mids they are referring to the frequency range from 1kHz to 8Khz, roughly.

The following drawing shows a typical EQ **Peak Curve** based around a **Centre Frequency**.



The centre frequency is around 750Hz and the **Gain Increase** (boost or cut) is around 18db. The **Q Factor** is the width of the frequencies effected by the boost and is measured in octaves. A high Q is narrow and a low Q is wide.

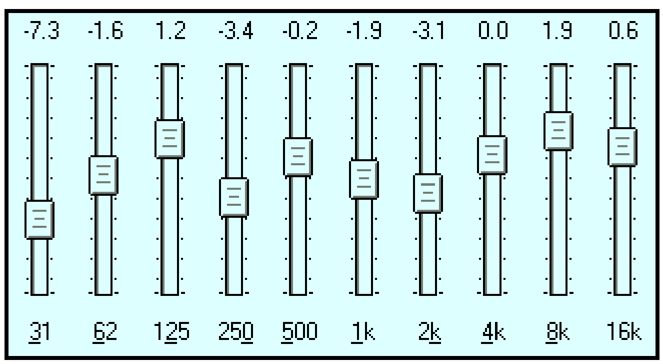
The following is is a drawing of a **Shelf Curve** where the frequencies above or below the centre frequency are all boosted or cut.



There are two kinds of equalisers, **Parametric** and **Graphic** and each can control a number of bands.

The Graphic Equaliser

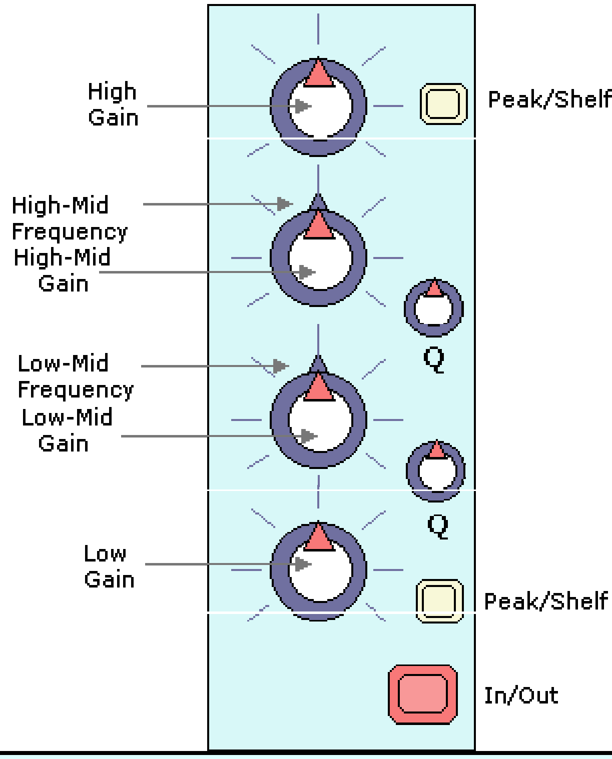
Here is a drawing of a typical **10 band** graphic equaliser.

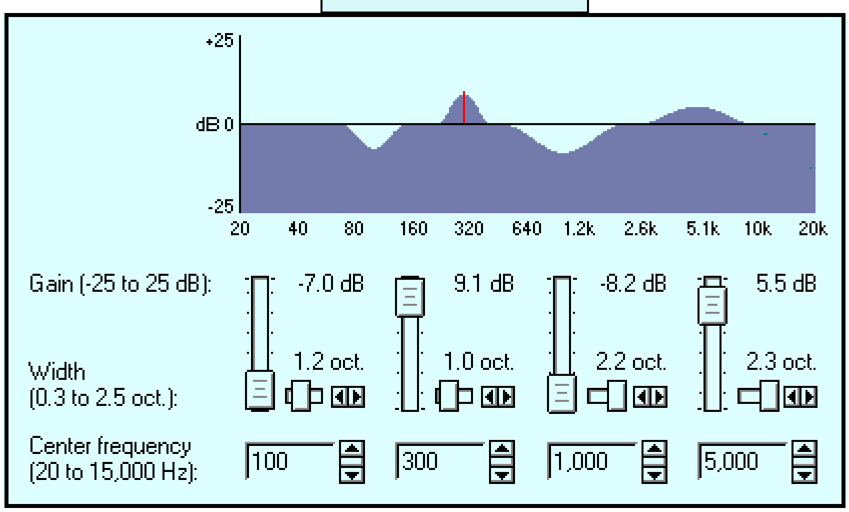


You will note that there are slider controls for each frequency and the scale along the base shows which frequency. The scale along the top states how many db change has been made at each frequency and it can be positive or negative (boost or cut). A typical graphic equaliser does not have any controls over the Q factor of each boost, it is normally pre-set.

The Parametric Equaliser

For an equaliser to be called a parametric equaliser it must have a variable Q factor and a variable centre frequency. Below is an example of a parametric equaliser:



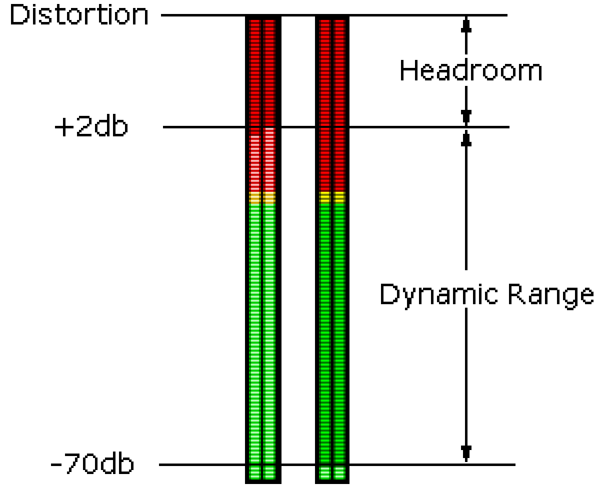


The left unit is a typical high end console analogue equaliser whereas the right one is a new generation computer program digital ones. The left one has a switchable peak/shelf High frequency control. It has two sweepable mid bands with variable Q and a peak/shelf low frequency control. The computer version has 4 Bands each with it's own centre frequency, Q width and gain. The resultant EQ curve is displayed as well. (It's a digital EQ) The mid bands of the analogue version are usually divided into two sweepable bands the the low - mid covering 100Hz - 4Khz with the other covering 600Hz - 15Khz (typically - it varies from console to console) You will note that the digital unit is sweepable from 20Hz to 20KHz in all bands.

Effects – Compression

Dynamic Range

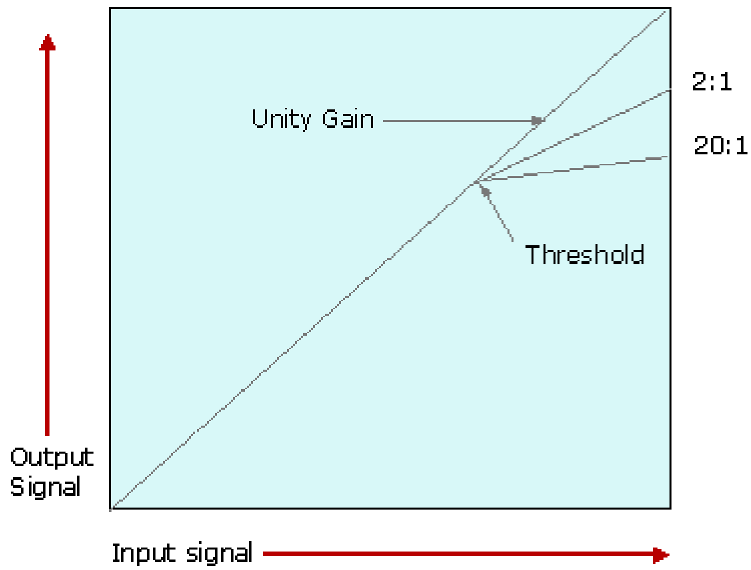
Before we look at compressors and limiters we must understand the term **Dynamic Range**. The Dynamic Range of a sound is the range between the quietest section and it's loudest section or in the case of a recorder the range between the noise floor and the point of distortion. You know how loud a Symphony Orchestra can get yet you also know how quiet it can get. An Orchestra has a wide Dynamic Range.



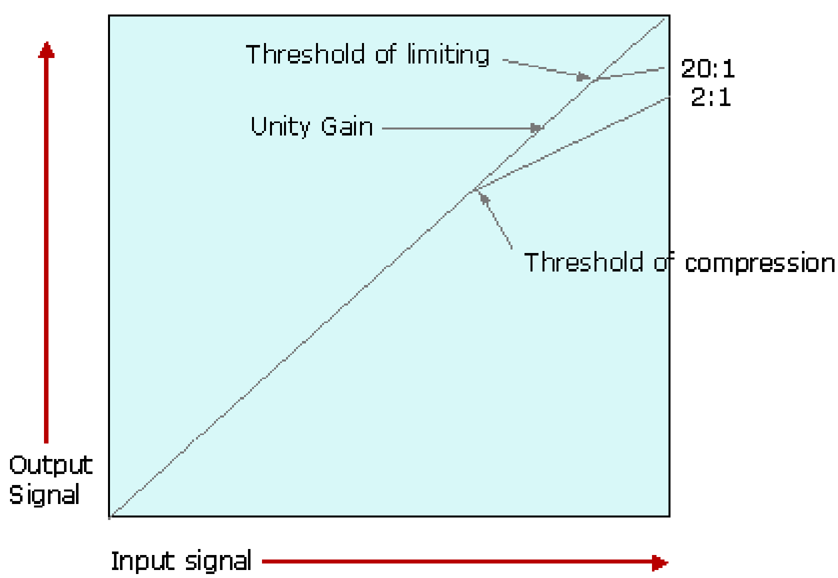
The meters above show a dynamic range of 72db. On a home cassette recorder the quiet section in this track would be below the noise of the tape recorder and all you would hear through the quiet passage would be tape hiss. The distance from the loudest section to the point of distortion is called the **Headroom**. If distortion is reached at +6db then we currently have a 4db headroom. To reduce the dynamic range you could ride the whole track with a fader and turn it up when it's too low and pull it back when too high **or** your can use a **compressor**.

Compressors

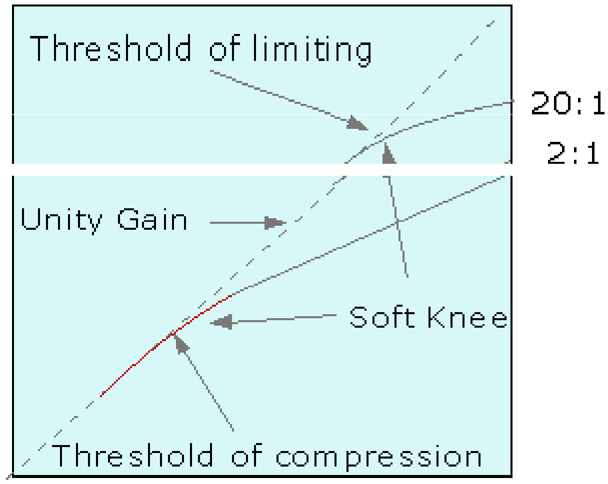
A Compressor can change the input signal to output signal ratio.



In the diagram above unity gain means that what you put in you get out. In the 2:1 ratio example When the signal is above the threshold the signal output is reduced in a ratio of 2db in will give 1db out when the **compression ratio** is set at 2:1, so you have saved 1 db off the top of your dynamic range and you can turn it all up by 1 db without effecting the headroom. In a more severe case like the 20:1, which is more commonly called **limiting**, for a 20 db rise in signal only 1 db comes out. The compressor and limiter can be used together in one unit where the compressor works in the 2 - 15:1 range whilst the limiter stops the extreme transient peaks in the signal in the 15 - 20:1 ratios which is why it is often called a **Peak Limiter.**



In the above graph the threshold of the limiter has been raised so that the main program material will be compressed above the threshold of compression at 2:1 and above the limiting threshold it will be 20:1. A compressor is a **gain reduction** device, therefore all compressors have a **make up gain** control so that if you are using 3db of gain reduction you can turn the output by that amount and still retain the same headroom.



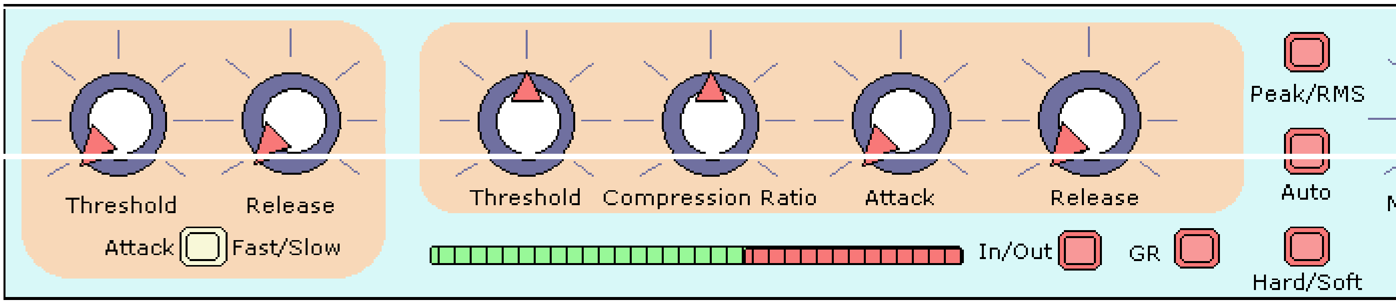
In the diagram above the transition from unity gain to compression at the threshold of compression/limiting is gradual instead of a straight line. This is called a **Soft Knee** threshold and is much smoother.

The **Mete**r on a compressor can usually be switched to read either the input level, output level or the amount of gain reduction. It is advisable to check that the input level is correct before you start adjusting the threshold and setting the compression ratio etc.

The **attack time** determines how quickly the the compressor reacts to signals above the threshold. Signals have short sharp peaks called **Transients** that can easily trigger a compressor to act. The attack time determines how long the peak should be above the threshold before compression takes place. These short transients are important in the **clarity** of a sound but don't effect the **loudness** of the sound. The aim of compression is to make the instrument sound louder, to squeeze the dynamic range, therefore you may wish to lengthen the attack time and let the transients through (to be dealt with by a limiter if necessary) and the compressor will then be working on **sustained levels** above the threshold.

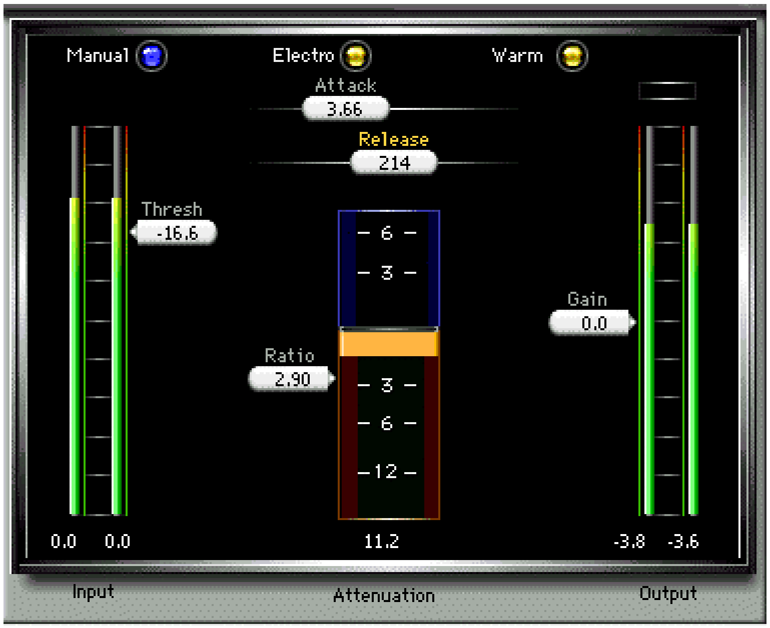
The **release time** determines how quickly the compressor lets go, or restores normal gain. If the release is too fast for the amount of gain reduction applied then the return to normal gain over and over as the signal moves above and below the threshold can cause what is known as **pumping** because the gain structure is changing rapidly. It is advisable to ask the player to play sustained notes and set the release so the change of gain is smooth. Instruments that have long sustaining notes like bass guitars should tend to use a slower release times than sharp percussive instruments like percussion. Most of the new generation compressors now have an **Auto** button that leaves it to the compressor to work it out, and they usually do it fine.

Take a look at a typical compressor and its controls:



The left section is the **Noise Gate** section. It has controls for the threshold at which the gate opens, the release time variable and a switchable fast/slow attack control. The centre section is the compression section with the standard controls over threshold, ratio, attack and release. The **Peak/RMS** switch determines how the compressor will track the signal i.e. its peak content or its RMS content. The **Auto** button is often an option where the compressor works out the attack and release times itself by analysing the program material. The **hard/soft** switch determines the **Knee** setting. The **meter** can read input or output levels plus it can read the amount of gain reduction. Finally there is the **makeup gain** control (Often just labelled output level) The **link** button is there if there are two compressors in the unit . Stereo Compressors have a link facility that makes one of the two compressors a **master**. (Usually the left compressor). All the controls on the master effect the slave compressor, so they both operate together. If the compressors weren't linked any strong signal on the right would be gain reduced and the stereo image would move because centre panned instruments would vary in their left to right balance so when compressing a stereo signal make sure the compressors are linked.

Similarly take a look at this image of a Computer program Compressor by Waves. All the controls are there.

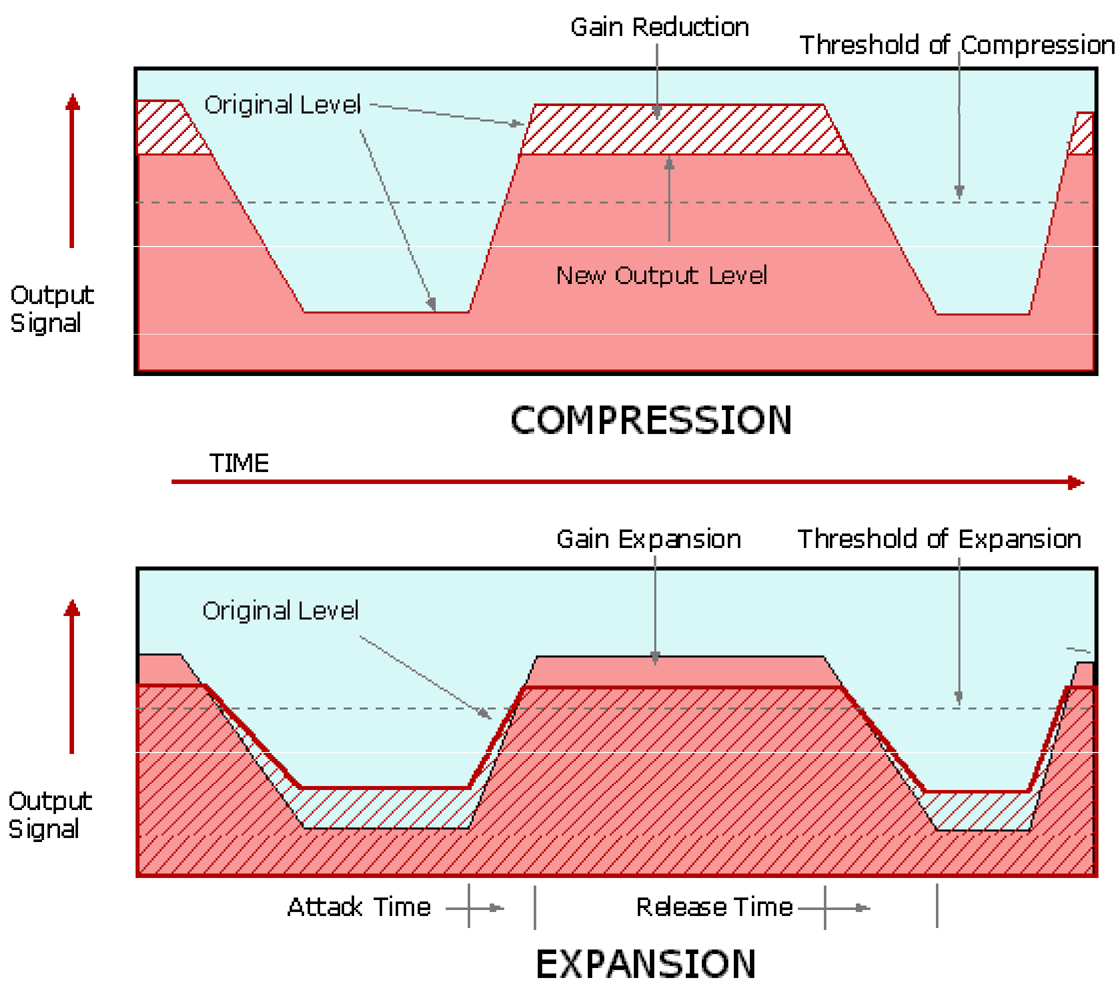


The Electro and Warm options are computer additions not found in a stand alone analogue version. As you can see the threshold is below the peak signal so gain

reduction is taking place as indicated in the attenuation meter. The ratio is set at 2.90:1 and there has been no make up gain applied.
The attack time is set to 3.66ms and the release is at 214ms and the control on them is manual (not auto).

Expanders

The expander is a compressor in reverse. There are two types of expander. In some, signals above the threshold remain at unity gain whereas signals below the threshold are reduced in gain, whereas in others the signal above the threshold also has the gain increased. Therefore you can use an expander as a noise reduction unit. Set the threshold to be just below the level of the player when playing. When the player stops the signal will fall below this threshold and the signal is reduced in gain thus reducing the noise or spill.



The drawing above shows the different actions of compressors and expanders. The expander in the drawing is increasing gain above the threshold and reducing gain below the threshold.

Most recording in popular music today has had heavy compression. Recording are loud and in your face! As well as most of the components of a track being individually compressed the whole mix overall has been compressed and limited before going to CD. I don't think that's a bad thing.

Limiters

A limiter is just a severe compressor where the compression ratios are high. On some units like the DBX 160 and the Aleisis compressors an additional Peak limiter control with a LED that flashes is supplied, but units like the Aphex Dominator are pure limiters and are very sophisticated in how they attack and control peaks and you can get some pretty hot "brick wall" mixes through them.

De-Esser

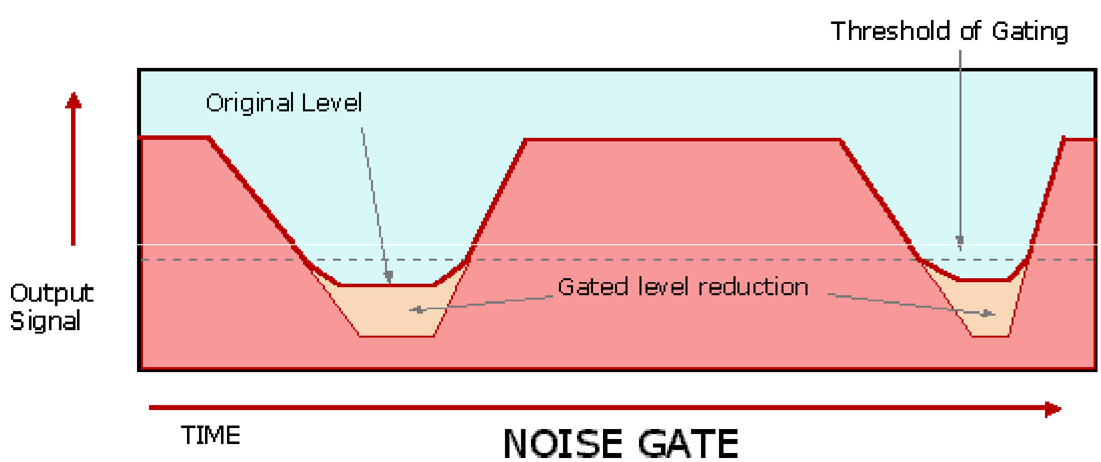
A De-esser is a frequency selective compressor/limiter that compresses only at a predetermined frequency. If set to the frequencies around the sibilance area of a vocal (4kHz - 8Khz, it varies between men and women,) the vocal will be compressed only at those frequencies which will reduce the sibilance. Sibilance is the peaks of high frequencies created by 'S', 'T', 'C's etc.

The new generation

The new generation compressors, expanders gates etc. in the new computer programs are worth a mention here. These compressors have one outstanding advantage over the stand alone compressor. They can read the signal ahead of time by extracting the signal from the hard disk ahead of time, analysing it and then outputting it in real-time. They know what is going to happen next which gives them a distinct advantage in maintaining smooth control over the signal.

Noise Gates

Noise gates are units that let a signal pass if it's above a certain level and shut it down if it's below that threshold.



The diagram above shows how a gate works on level. When the signal falls below the threshold the gate reduces the level to the specified reduction level. The attack time here determines how quickly the gate will open and the release time determines how fast it will close. Some gates have a **Hold** function that allows you to tell a gate to hold open for a set time once it is open and then the release time can take over and close the gate. This facility can stop the gate opening and closing quickly due to peaks. It can also be used as an effect, especially if it is put over the return from a reverb unit. If you have some reverb on say a snare, and you put a gate over the reverb return signal, you can get the hold function to hold the reverb open for a period set by the hold function and then to quickly close it by using a fast release. This effect is called **Gated Reverb** and is now a standard program in most reverb units.

A gate can also be set to be triggered by something else via a **side chain**. For example, if you put a gate over a room ambience mike you could use the snare mike to trigger it to open when the snare was hit and to close when the snare stopped. This is called **Gated Ambience.** Another effect is to put a hihat feel into the side chain and modulate the gate to open and close on a synth sound. The effect is a modulating synth with the attack and release times controlling the modulation.

Gates can also so **linked** so that one controls the other and when one opens the other opens as well. (Like the compressor) This is used in stereo gate situations like over stereo toms.

Equipment

Introduction

Since posting the Recording Studio Design site on the web I've been asked by numerous e-mails to suggest what gear the writer should purchase to establish their studio. What an ask! In this section I will cover the three main pieces of equipment a studio will need. Understanding what each unit does and what to expect of it can quickly assist decision making.

Microphones are your front line in the studio. If you have the wrong type of microphone on an instrument you are starting on the wrong foot right away. Understanding the different types and how they work can help you decide what you need.

Consoles come in all shapes and sizes so I've described the workings of a console from the mike input to the monitor output. You probably can't afford the super console but if you know what each section does and how it is applied you can work out what your minimum requirement and go looking for that or you've already got a console that mightn't have everything but this section will help you understand how to use what you've got properly.

The **Recorders** section describes the different types of recorders and data storage systems available as of now which includes analogue tape, digital tape, and digital hard disk - tomorrow could be a different story. I've also covered Digital Sound. We are entering a totally new era in sound recording since the advent of digital sound, it's not perfect yet, but I believe it very soon will be.

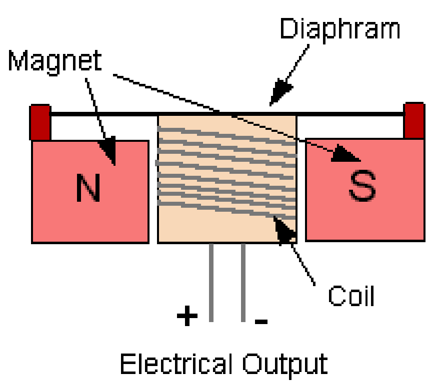
Equipment – Microphones

With the plethora of microphones around you'd be surprised at how engineers all over the world seem to use the same mikes. Go surfing to all the studios and you'll find the same mikes in their mike list. I've got to state here that I'm not pushing any particular brand or type - I am not sponsored - so I'm only stating what I've observed over the years.

So how do they work? Basically all microphones have a **diaphragm** that vibrates when hit by sound waves. The vibration of the diaphragm is translated into an electrical signal that corresponds to the variation in the sound wave. That is why it is necessary to clean the diaphragms in your mikes on a regular basis as a build-up of dust, spit etc. will impede the vibration of the diaphragm and thus distort or colour the sound.

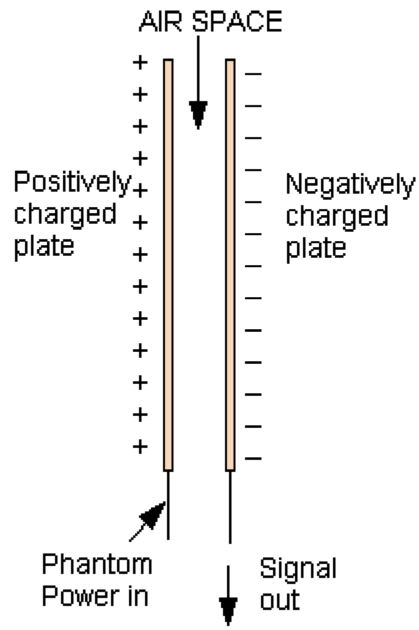
The Dynamic

In a **Dynamic microphone**, also referred to as a **moving coil** microphone, the capsule is rather like a speaker in reverse. The cone is the diaphragm and it has a coil attached that is suspended in a magnetic field. When the diaphragm vibrates the coil creates an electrical current. This is an entirely **passive circuit** as the magnet can be a permanent one so **no external power** is required.



The Condensor

On the other hand the **Condensor microphone** has two plates, one fixed and one moveable, that are each charged with a polarising voltage that creates a capacitor. The vibration of the plates creates a change in the distance between them which changes the capacitance and thus the sound wave is converted into an electrical current. In this case **external power is** required as there is an electrical circuit required to produce the polarising voltage. Because the current obtained is so small an amplifier circuit is also included.



Thus when using Condensor mikes an external power supply is required. This can be either a stand alone power supply for one or more mikes or it can be fed to the microphone from the console down the microphone cable and is commonly referred to as **Phantom Power** and is now standard at 48 Volts and all new consoles have that facility and usually consists of an on and off switch on the rear of the console or is an on/off option on each module. Incidentally, don't worry about sending phantom power to a dynamic microphone, it won't blow it up as the circuit is inactive in a dynamic mic situation.

The Electret

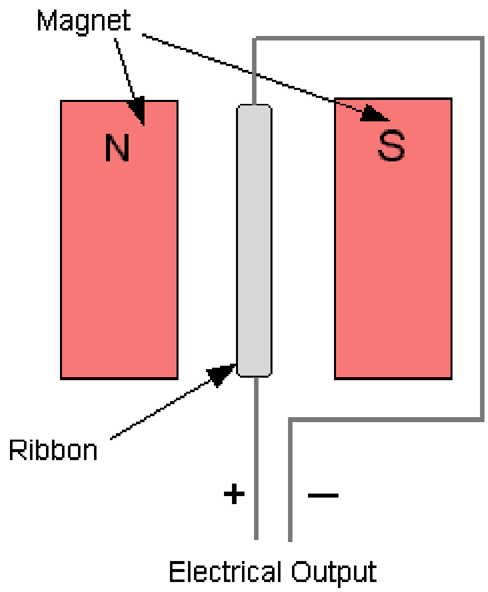
An **Electret Microphone** is also a Condensor microphone except that the charge on the plates is created by a permanent electrostatic charge. Therefore an external polarising voltage is not required but once again the voltage obtained is small so an amplifier is usually built in and powered by an internal battery. Electrets are often thought of as the cheap cousin to the condensor mike because the material required to hold the charge on the diaphragm is heavier but good electrets can sound fine.

The Pressure Zone

The **PZM** or **Pressure Zone Microphone** is also an Electret microphone except that it is mounted in a special housing near the pressure zone on the surface of a plate. This plate can be mounted on a flat surface like on the wall, floor or the lid of a piano. I have found that PZM mikes are not prone to popping and appear to have no proximity effect. They are typically used for pianos in concert situations where the lid can be closed to reduce spill and are also ideal as floor mikes in stage show productions.

The Ribbon

The **Ribbon Microphone** consists of a thin metal ribbon that is placed in a magnetic field. The vibration of the ribbon within the magnetic field induces a current that is proportional to the variation in the sound wave. This is also a passive circuit as the magnetic field can be created by a permanent magnet.



The Valve Microphones

Finally I must say something here about **Valve Microphones**. As mentioned before, the signal from the diaphragm in a Condensor microphone is small and must be amplified before it reaches the console where again it is amplified further. It is within this area that signal deterioration can easily occur and therefore the quality of the microphone must also be judged by the quality of the first stages of amplification. In a valve microphone the Condensor stage is a standard condensor system but the amplifier section uses a valve circuit to amplify the current as opposed to a transistor circuit used in later models. When I first started as an engineer in 1966 all the Condensor microphones were valve and each had its own power supply. The introduction of the **transistor** microphone eliminated the need for power supplies because phantom power was invented for the purpose.

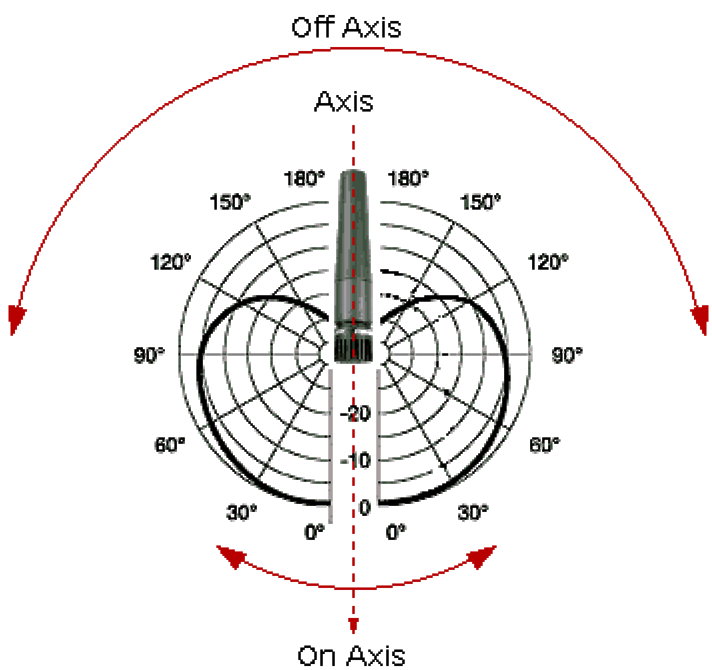
The other major factor in those days was signal to noise. The average tape recorder had a signal to noise ratio of around 58db as opposed to the 70+ with today's analogue recorders.(Mainly due to the improvement in the surfacing of tape.) With such a low signal to noise ratio we were always careful about the high end of our recordings because if you had to add it later you sacrificed your noise and increased hiss. So when the transistor microphone came out we all remarked "Far out!" (it was the 60's) listen to that top end!!" and immediately used them instead because records were getting brighter then. What we were hearing was the difference in distortion between a valve and a transistor. A valve distorts in the 2nd harmonic first whilst a transistor distorts in the 3rd harmonic. The 2nd harmonic distortion is smooth, we can handle it but 3rd harmonic distortion is hard and harsh to our ear hence the difference between the two. The valve appears warmer like a valve Marshall does compared with a transistor version.

Today, on the other hand, the top end and noise is not a problem as modern analogue tape recorders have good signal to noise ratios and our mike preamps are also quiet yet from another aspect it is. The top end of digital is extremely bright compared with analogue tape due to the inherent distortion of frequencies above 7kHz created by the slow sampling frequency of 44.1kHz which in reality produces close to a square wave above 10kHz I find it produces what I call digital fatigue. Rupert Neve was recently reported as saying that we will need to sample at 24 bit/192kHz to equal analogue. (We will eventually) Meanwhile the warmth of the valve acts with the harshness of digital and produces a great compromise, hence one of the reasons for the popularity of valve mikes today.

Alternatively engineers today will put a mike through a valve preamp which is the second stage of amplifying a mic signal. Once again it is the soft clipping of the high end that produces that warm sound. What a lot of manufactures do today is the put a valve within a transistor circuit thus obtaining the soft clipping of the valve with the improved signal to noise of the transistor circuits. I've even seen an ad for a CD player that has a valve circuit in it!!

Polar Patterns

A dynamic microphone has a set sensitivity pattern called **Cardiod Pattern** or "heart shaped" or "Kidney shaped" pattern and the response looks something like this.



Please note that this is not the response curve of a SM57, a SM57 is might tighter than this, it is only a demonstration. The line through the centre of the mike is called the **Axis** and when standing directly in front you are said the be **On Axis** as opposed to being **Off Axis** at the side and rear. In this example at 0 degrees there is full sensitivity, at 90 degrees the signal is reduced - 5db, at 120 degrees by 10, at 150 degrees by 20 db etc.

Condensor microphones have the added advantage of being able to alter their pattern from the standard cardiod and produce either a **Figure 8** pattern or an **All-round** pattern.

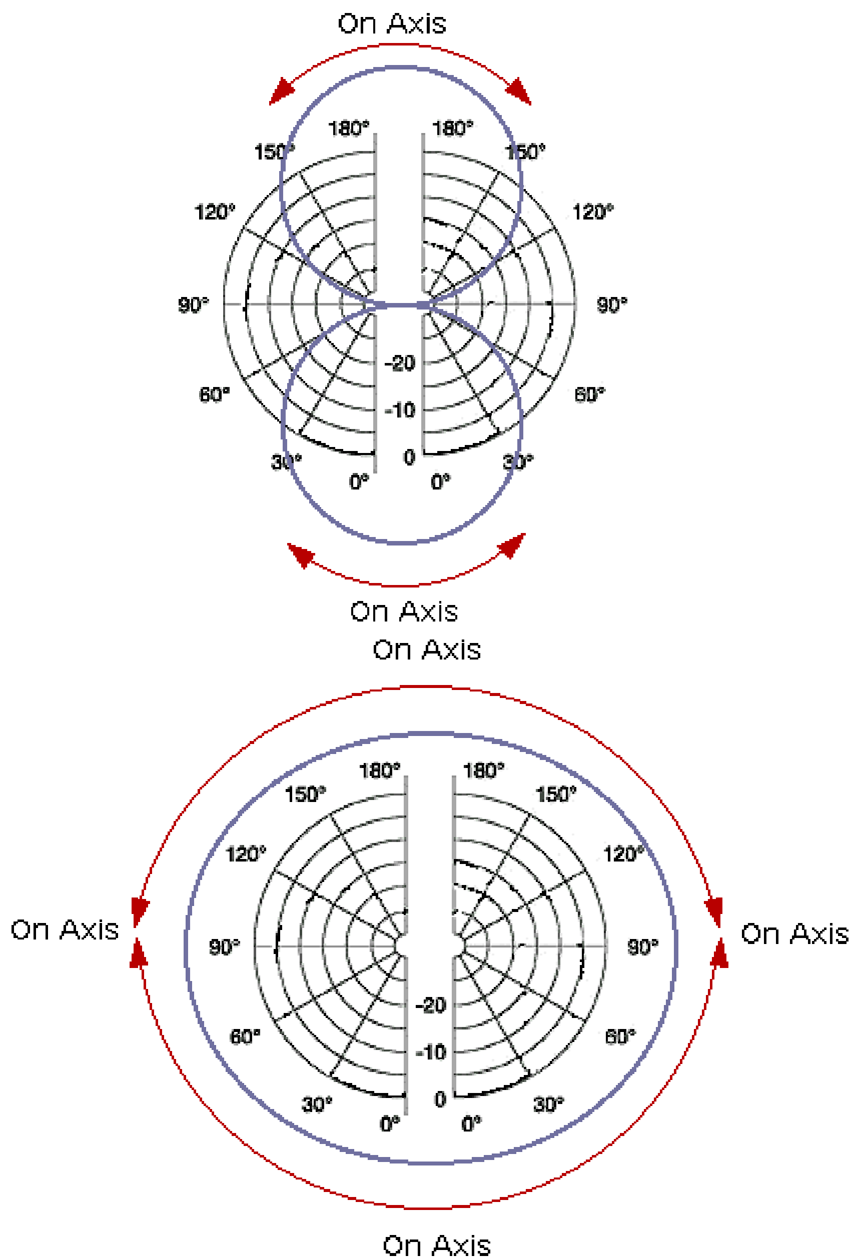
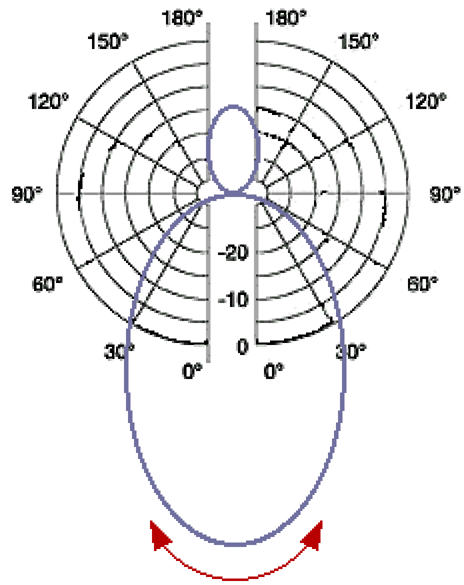


FIGURE 8 and ALL-ROUND PATTERN

When using a Fig 8 mike you can place an instrument or singer on either side of the mike. With the all-round pattern you can place anyone anywhere as the pattern picks up through 360 degrees. Incidentally the all-round pattern does not exhibit proximity effect.

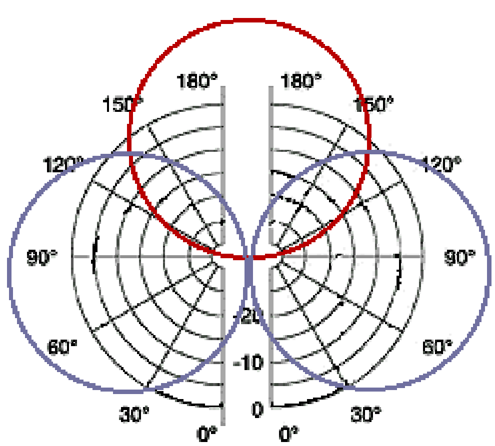
Hypercardiod

Finally there is the Hypercardiod pattern. This is like a cardiod pattern but tighter.

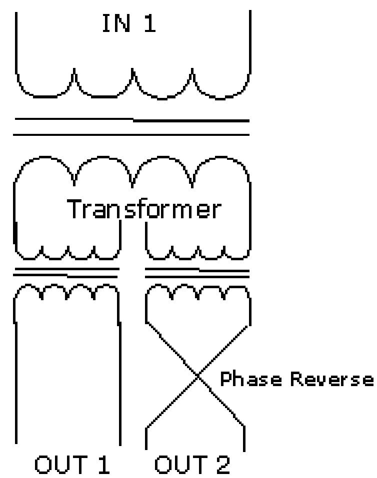


MS Stereo

MS stereo is short for **Mid Side** miking. It is recognised as being the truest form of stereo miking because it is not subject to centre lift in mono. When you join a stereo signal into mono the instruments panned to the centre (i.e. equal left/right) lift in the balance and is referred to as **Centre Lift**. MS stereo recordings don't have that tendency. You can buy MS Stereo microphones but if you've got a cardiod quality mike and condensor that will produce a Figure 8 pattern you are in business. Set them up like this.

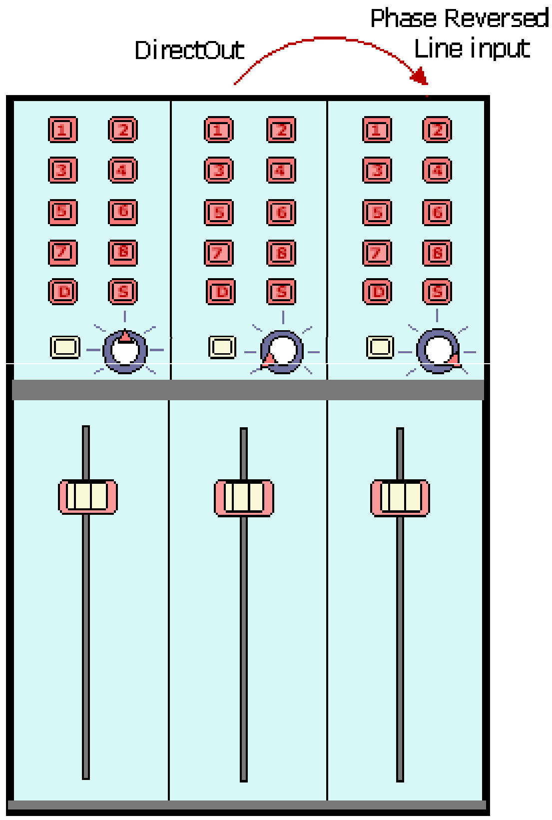


The signal from the Figure 8 mike will need a mike splitter that splits into two mike inputs. This is the tricky part, To have a figure 8 mike it must be a condensor with phantom powering and if you split it and phase reverse, it will cut off the phantom power. You can purchase a transformer box like this:



The other way is to bring the Fig 8 mike up into a console and then take a feed from the direct out of that channel and bring it back in via a line input on another channel and phase reverse it.

Bring up the cardiod mike and pan it centre, now take the two splits of the figure 8 mike and pan one left and one right. Now reverse the phase of one of the splits. If you now have the cardiod mike pointing at what you are recording and you slowly add the fig 8 mikes you will hear the sound change from mono to wider and wider stereo as you add more of the fig 8 mike. The Cardiod mike is called the **mid mike** and the fig 8 is called the **side mike**.



When you mono this signal the left and right signal cancel each other and you are left with the mono centre signal which is a true mono. You can use MS Stereo for all sorts of things like overheads on a drum kit, ambience room mikes, stereo ACC guitars and pianos etc. You can always buy a MS Stereo mike but they are very expensive.

Proximity Effect

Anyone using microphones must understand **proximity effect**. When you get close to a microphone there is a rise in the low frequencies called the proximity effect. This low end boost can be 20+db boost at 50Hz!! A vocal mike like the Shure SM58 has a built in roll off to compensate for this because live performers like to sing close to the mike, but if you stand back from the mike it will start to sound thin, in other words if you want the SM58 to sound flat you must be close to it. Most mikes will have proximity effect so a **low cut filter** option is often supplied to compensate for it.

MICROPHONE PHASE RELATIONSHIPS

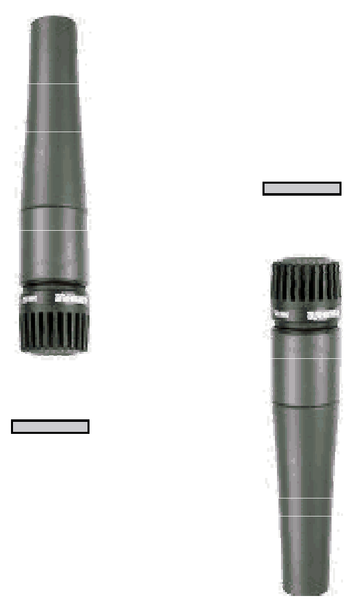
Before you record anything it is imperative that you check all your microphones for phase.

Two diaphragms in phase

Here the two diaphragms are moving in the same direction so they are in phase. Imagine them as two overhead mikes and they will both receive the signal from the drums in the same phase.

Two diaphragms out of phase

Here the two microphones are pointing opposite to each other yet their diaphragms are receiving the same signal. When the left mike's diaphragm moves in the other mike's diaphragm moves out.



As a result the two mikes are said to be out of phase and a phase reversal must be inserted or the two microphones will cancel each other. Officially they should cancel totally but they don't entirely in practice because each has a slightly different signal because of it's different position in the sound field. It will be most noticeable in the low frequencies so if you top and bottom mike a snare and don't use a phase reversal the sound will

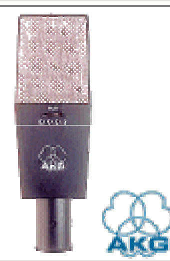
be thin and lack low
frequencies.

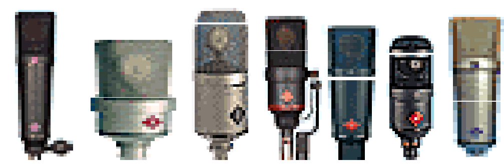
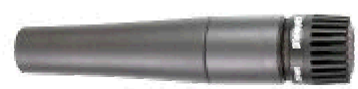
Similarly, miking toms top and bottom the bottom mike will require a phase reversal. If your console doesn't have a phase reversal switch on it (funnily a lot don't) you should build some phase reversal plugs of your own. This can be done by simply making a male to female mike lead with pins 2 and 3 reversed. It's a good idea to paint them red or something so you know that they are phase reversal cables. You can also purchase pre-made phase reversal plugs from some retailers. Some people simply connect a male and a female cannon plug together with the leads reversed, paint them red and insert them into the mike lead before the mike patch point.

Checking your phase

A small note here - before you start recording it is a good idea to check the phase of all your microphones and cables. You can purchase small phase check boxes where you plug each end of the cable into it and if all three lights light up the cable is OK. At some stage it is worthwhile setting up all your mikes, select one mike as a reference, and getting someone in the studio to speak into your reference mike and each mike in turn to check that each mike is in the same phase and that all your cables are correct. You will notice immediately if one of your mikes is out of phase.

THE MOST COMMON MICROPHONES

The famous D112 from AKG - a standard kick drum microphone.	
The classic AKG 414 EB This is a great overhead - hihat mike (it's also a great kick mike if you're prepared to put such an expensive mike on the kick).	
The AKG 451 is a beautiful all purpose quality Condensor microphone. It comes with various alternative diaphragm capsules with different pickup patterns.	

The classic Sennheiser MD421 tom microphone. (John Laws has a gold one!! for you OZreaders)	
Sennheiser MD441 is another great snare mike and can be used on toms.	
The fantastic range of microphones from Neumann Germany. Unfortunately they are now so expensive that the average home studio owner can't afford them. You can probably buy 10 SM57/58s for one budget Neumann!! But they do sound **extremely good** and are one of the best!! If you can afford at least one, preferably a pair, you'll never regret it.	
The classic Shure SM57. Probably the best value microphone available. You can use it on drums, guitars, vocals, whatever.	

You can check all the mikes out at the websites of the manufacturers:

- **NEUMANN**
- **SENNHEISER**
- **AKG ACOUSTICS**
- **SHURE**
- **BEYER**

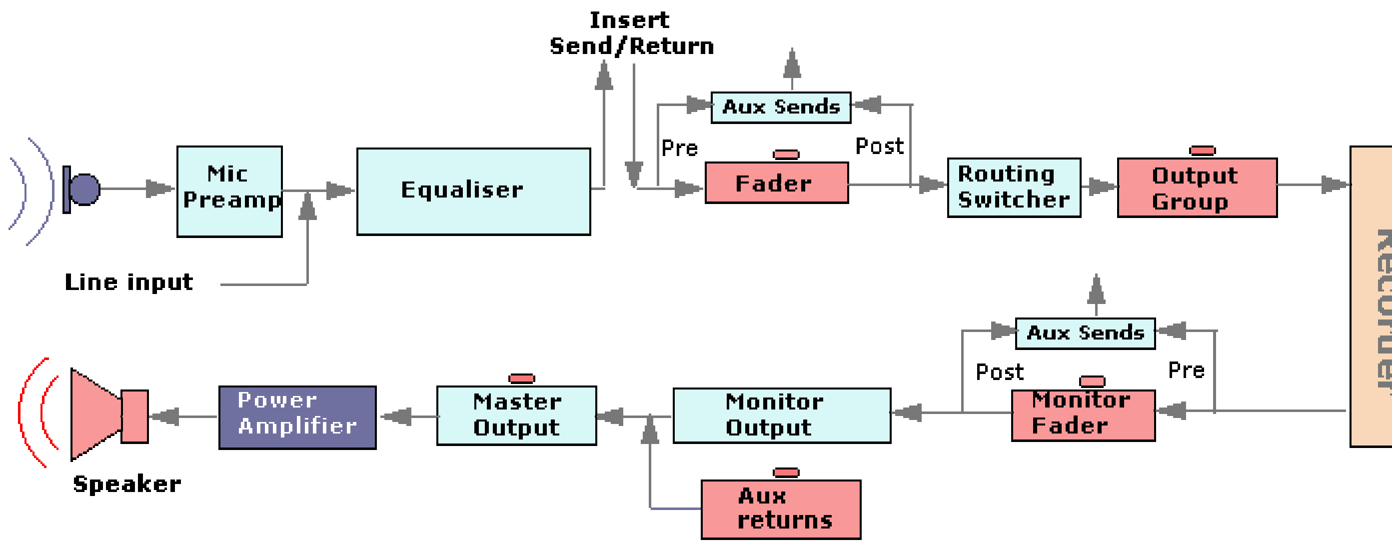
Equipment – Consoles

Recording Consoles are simply a set of component parts all put into one box which can be configured into two modes

- **Record/Overdub**
- **Mixdown**

RECORD/OVERDUB MODE

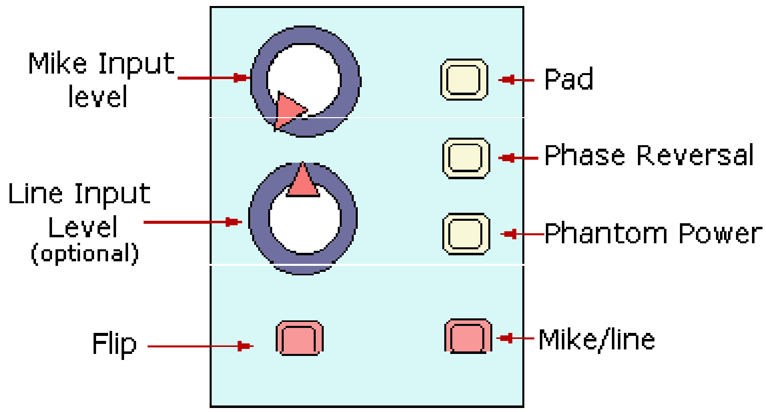
The components in the Record Mode are configured thus:



The recorder and the speaker and power amp are not part of the console but are part of the chain. So let's go through each component and look at what it contributes to the chain.

The Microphone Preamp

The microphone preamp is possibly one of the most important component in the whole recording chain apart from the microphone itself. Between them, these two units bring the minuscule voltage coming from the diaphragm up to the 1+volts that we work with. From that point on everything is operating at normal levels and the signal can be recorded, EQ'd etc. without problems with noise and distortion. The typical mike preamp can have the following controls.



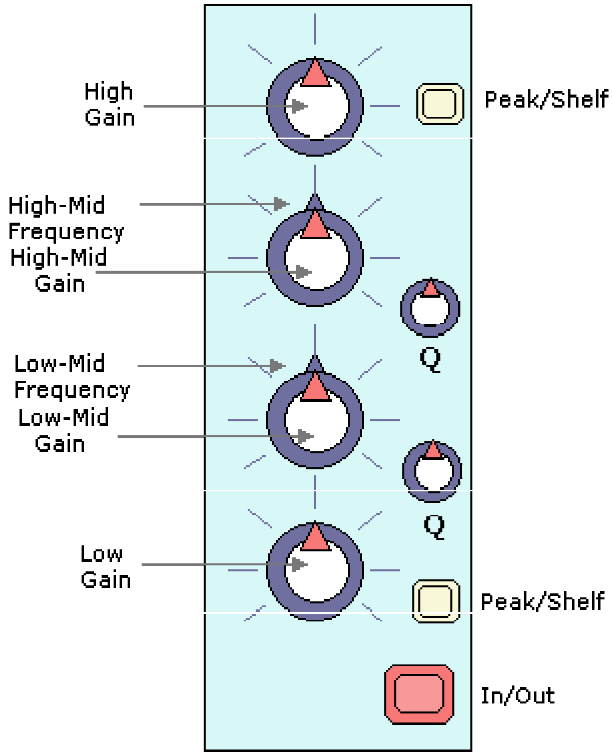
I said "can have" because not all do. First there is the **mike input level** knob. This is the gain control for the preamp.

A special note here - Before you start to set the mike input level you must set the console up for **unity gain**. This involves first setting the console output faders to Zero, then the channel fader to zero. If you are going out a group put that fader to zero. This first step is vitally important because a console is capable of increased noise and distortion if not setup with correct gain structures. If you have a little Mackie or something which doesn't have a separate control over monitoring turn your amp and speakers down. Basically if you run the output faders low you have to get the gain from somewhere so you turn up the mike preamp which is capable of adding noise and distortion.

If you find that you are fully counterclockwise and still have too much signal you must insert the Pad. The pad drops the level by 10 - 20db (depending in the console) and stops the preamp from overloading. Some consoles will include a **phase reversal** button which is a very handy option to have. There is also a button for **Phantom Power** which will supply power to your mikes if required. (See **microphones**) It often comes as a single on/off switch on the rear of the console. There is often an optional line input knob with an associated **mike/line** switch. This allows you to trim the level of the line inputs. Finally you will probably find a mike/line button that allows you to adjust the level of the line input individually. The **flip** button is only on certain consoles. It swaps the two main faders over (line and monitor), More about this option later when we look at monitoring..

The Equaliser

The equaliser is the next stage within a console. This are the controls on a quality console equaliser.

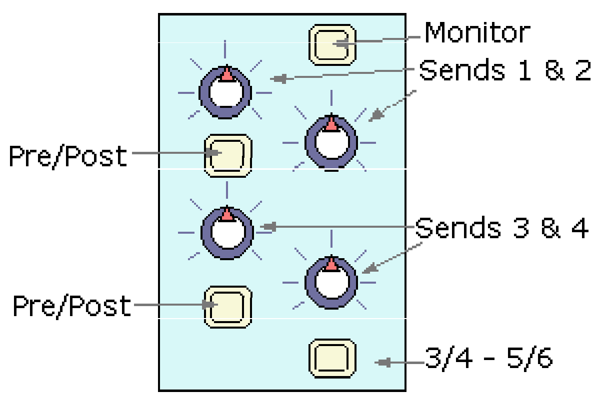


This equaliser is a true **4 band, parametric equaliser.** It incorporates four separate bands with the low-mid and high-mid bands fully frequency sweepable with variable Q. Go to the full page on **equalisation** for more about EQ.

Insert Send/Return

The insert send/return appears as an output and return on the rear of the console typically as a stereo jack with the tip being the send and the ring the return and the sleeve as a common ground. This facility allows you to insert a compressor or effect unit into the signal chain.

The Auxiliary Sends



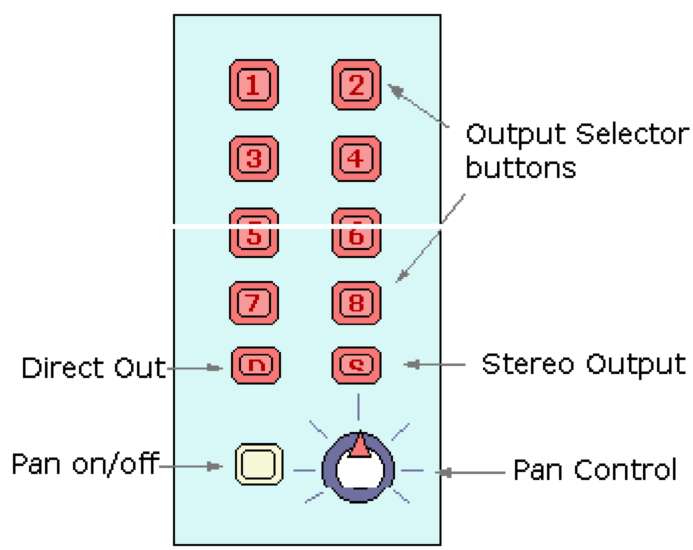
The **auxiliary sends** are split feeds of the track signal that can be sent to other units such as reverb and effects units. It can also send a separate mix to the **headphones** for the musicians. The **send** knobs are self explanatory, the **pre/post** switch determines whether the signal it sends comes from before or after (pre/post) the fader. If it is set to **post fader** as you turn the fader up and down the signal going out the aux send follows and also goes up and down. The **pre fader** position means that changes you make to the fader won't effect the aux send output and its level will remain constant. This is important if you are sending a mix to the musicians which is normally done with pre fader sends so that your mucking around with the faders doesn't change their balance. The monitor button will switch the input to Aux 1 and Aux 2 from the channel signal to the monitor signal on that module.

The Channel Fader

The channel fader gives you control over the level of the channel. As mentioned earlier In recording it should be used as near to 0 as possible. It also becomes your mix fader.

The Routing Switcher/Output Groups

The output group is the amplifier that joins all the separate channels together and routes them to their destination.



The **Routing Selector** allows you to send the signal from a channel to any of your **subgroup outputs**. I say sub groups because you will also have a **stereo main** output. The Pan control allows a stereo output between two selected channels typically 1 & 2 , 3 & 4 etc. The **pan control** will have a pan additional **pan on/off** switch. The **direct out** button allows you to send the channel directly out of the console via a plug on the back of the console without going through a group amplifier. The stereo output selector sends the signal directly to the master output and is used when in mix mode.

The Recorder

The signal from the group outputs will now appear at the **recorder inputs** and when monitoring through the recorder or playing back, it's output will appear back at the console at either the **line input** or more typically the **tape return** input depending on console type.

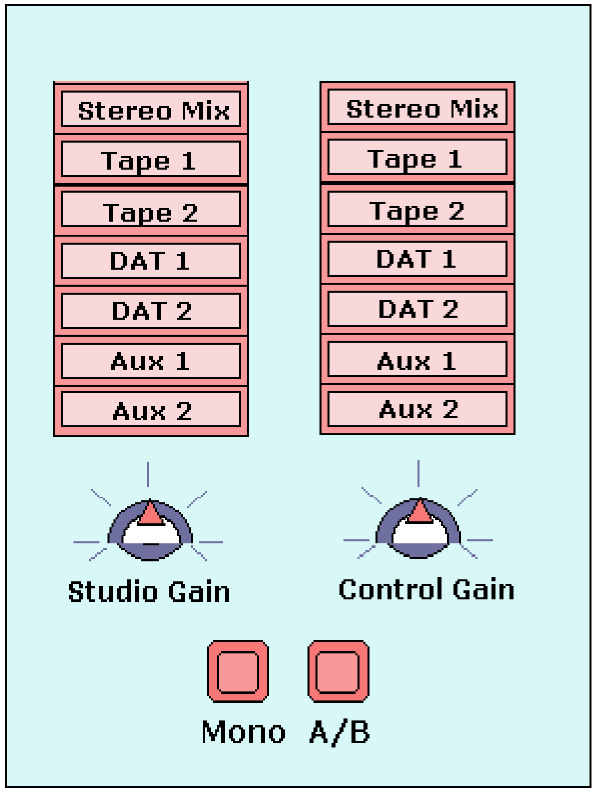
Monitoring

You must be wondering how you are going to hear what you are recording. The key line here is "what you are recording". You must follow the sound through the above outputs etc. so that when you monitor the sound it is "what you are recording". Now we have all the channels going to the recorder **and back**. You must now listen to the signal through the **Monitor control**. This is usually a knob (or fader on the more expensive consoles) below the auxiliary send section and has a pan control associated with it so that each **monitoring return** can be panned left or right in the **monitor speakers** as the output of the **monitoring mixer** goes directly to the monitoring output of the console and to the speakers. You can change the balance, switch instruments on and off in the monitoring section without affecting the signal you are recording. The monitoring section also has auxiliary sends or it has access to them via the **monitor** switch in the auxiliary send section as mentioned earlier. **This is where you should send your headphone sends for the musicians.** You must also send them prefader mixes so your changes doesn't effect their balance. I have emphasised this because it's very important:

- If you have added effects to an instrument and inserted it between the console and the recorder the musician will not hear it if you have sent his headphone send directly from the input channel aux send.
- When you have recorded a take you can immediately play it back with the same balance you had when you recorded it and the balance in the headphones doesn't change for the musicians.
- Also if you sent their feed from the channel they will hear nothing in the headphones when you play the track back.

The Monitoring Selector Section

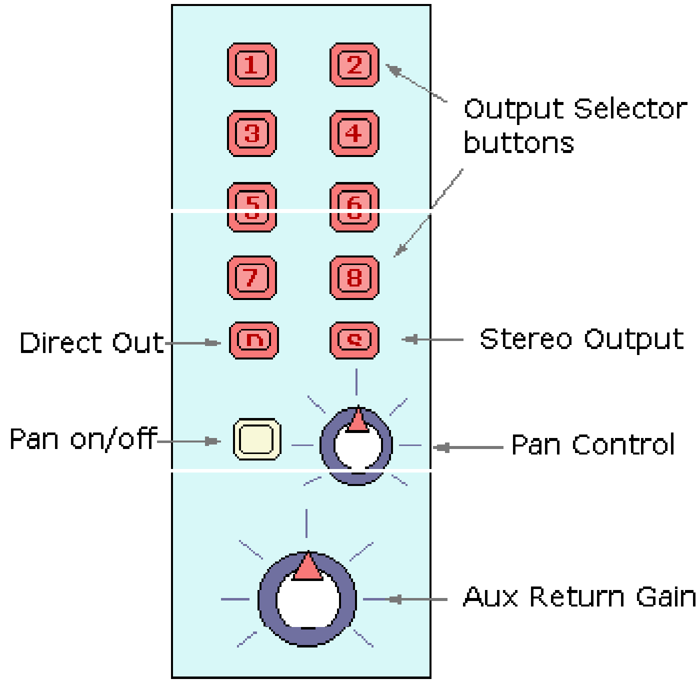
The monitoring output goes to the power amplifier and thus to the speakers. Some better consoles have a monitoring selector section where you can control what signal goes to the speakers.



There are two outputs from the **monitoring switcher**, one goes to the **studio speakers** (if you have them) and the other to your **control room speakers**. The selection buttons allow you to select the source for your speakers The control Mix selection puts the . The **mono** button allows you to check the signal in mono and the **A/B** button allows you to have two sets of monitor speakers (main and nearfield) and switch between them.

Auxiliary Returns

The **aux returns** (or effects returns) are the return channels for all your effect units like reverbs etc.



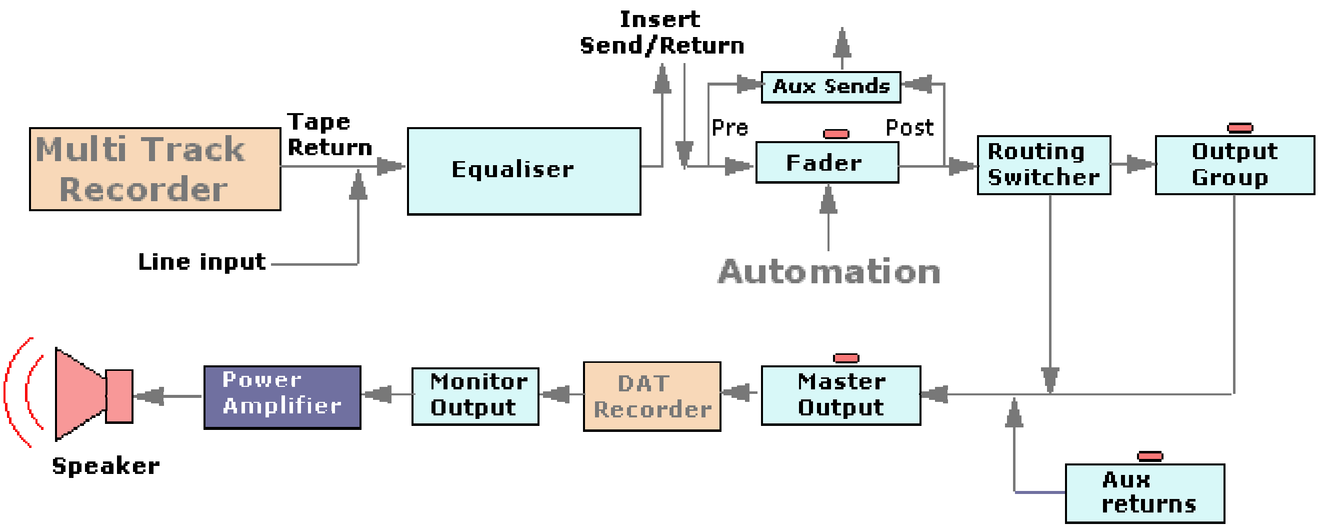
The aux returns are where you bring the returns from your effect units back into the console. They are usually supplied with a routing selection output just like the group outputs so they can be sent to the group outputs as well as the master outputs. If sent to the group outputs the effects can be sent to the recorder with the original signal from the channel and recorded.

Overdubbing

In the overdubbing situation nothing need change as the console is already in that mode. Tracks from the recorder are played back through the monitoring mixer and new instruments processed through the channels and group outputs to the recorder. Please make sure you send the musician a good headphone balance, it makes it so much easier for them to play well.

MIXDOWN MODE

To mixdown your creation the console needs to be reconfigured thus.



You will note that what was the mike channel has now become the tape return channel and follows the same route through the equaliser, inserts, fader and aux sends to the routing switcher. Here it can either go direct to the master output or several channels can be grouped together via the output group and then on to the master output. The aux returns are as before. The signal now goes through the **DAT** and back to the monitor output and to the speakers. Once again we have a "what you are recording" situation where you are listening to what is going on tape.

Automation

I have included an extra in the mixdown stage called automation. Console automation makes use of a unit called a **voltage controlled amplifier** or **VCA.** An amplifier is normally in full gain mode and you change the gain of a signal by putting more or less into it. A VCA on the other hand is an amplifier where you can adjust its gain via a change in an external voltage. In an automated console the amplifier at the fader stage is a VCA and the fader adjusts the voltage of the amplifier thus its gain. When you turn the fader down you decrease the voltage and visa versa when you increase the gain. There are three important advantages here:

- The fader **doesn't carry the audio signal** only the voltage to the VCA therefore if its cheap or dirty the audio signal is not effected.
- The variation in this voltage can be read and recorded externally and **played back**.
- When you turn the voltage off you in fact **mute the VCA** so you can also automate mutes.

The variations in the voltage during a mix are recorded as data in the automation module and when the mix is played back the VCA changes in time with the track and follows your changes. These fader changes (and mute changes) can be altered progressively until you are happy with the mix. I Love Automation!!! For the automation to stay in sync with the track it will need a timebase reference which is usually **Time Code** from the recorder.

Some companies like Mackie produce an automation package that you can plug into any console so long as you have insert send/return plugs on the back. Here the VCA becomes like an external effect and is plugged into the channel via the Inserts. You place all your faders to Zero, turn on all your mutes and proceed to set your fader moves using the supplied remote control. All the changes in the VCAs are recorded within the unit and can be played back. I've done heaps of mixes using this system and I highly recommend it as a simple, cheap, and reliable automation system. (Hey I'm not sponsored by anyone so I can say what I think!)

Summary

We are now entering an era where the console is coming back but in a different form. Companies like DigiDesign with ProTools and Yamaha with the O2R are producing consoles designed to be a user interface to a computer based hard disk recording system where there is a friendly 'hands on' interface to a sophisticated hard disk recorder setup. I dream of the Virtual Reality studio where the home recording artist can put on a set of Virtual Reality glasses and be in any control room they like with touch sensitive virtual consoles and effects. Your friends can be included with selectable identities and even though you are only in your 8' x 10' bedroom in virtual reality you are in the control room of a major studio. Any smart software writers out there?

Equipment – Recorders

The recorders are data storage systems and fall into three categories:

- **Analogue Tape**
- **Digital Tape**
- **Digital Hard Disk.**

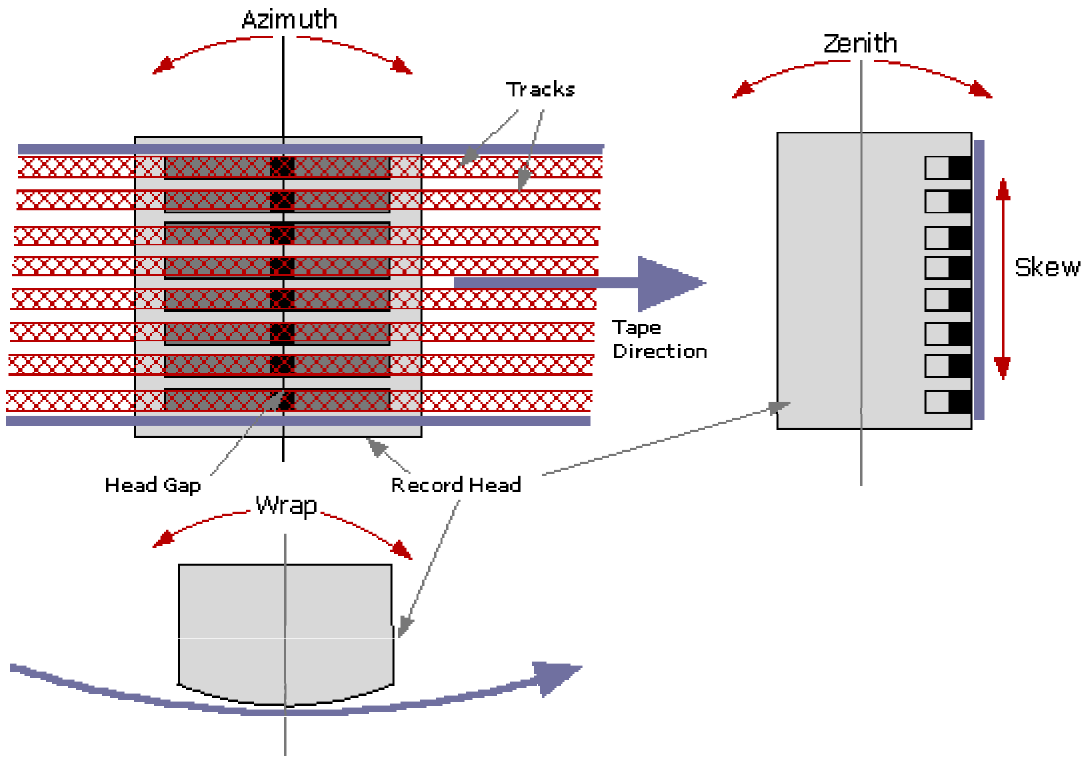
Analogue Recorders

Despite the onslaught of the digital recorders over the past decade the analogue recorders are still hanging in there primarily because engineers still believe that the analogue sound has not been surpassed by the digital medium, and quite rightly so. The top of the line 2" analogue recorders are still being used and sold but primarily by the perfectionists and dedicated audiophiles who will probably have a digital system with the analogue being used for bass, electric guitars and drums. Vocals and the rest being covered by digital. Some engineers still insist on mixing down to 1/2" stereo analogue masters and some mastering studios actually transfer their digital masters via an analogue recorder to soften the harshness of the digital top end etc.

Meanwhile the home recording enthusiast is very likely to have a 1/2" sixteen track or a 1" twenty four track etc. The important aspect of the analogue recorders is their need for regular **maintenance and servicing**. If you are an owner of a analogue recorder you must have the **alignment tape** required for correct alignment of the transport and electronics. This is a tape with pre-recorded signals that determine:

- Head alignment
- Transport alignment
- Frequency response and level structure

So let's look at the Head alignment factors:



The above are the variables in head alignment in an 8 track analogue recorder. The head alignment is executed in three dimensions through the variables:

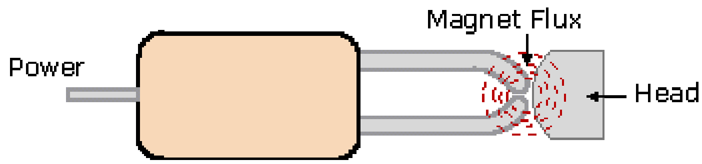
- **Azimuth**
- **Zenith**
- **Wrap**
- **Skew**

All variables are really self-explanatory from the diagram and the head block on which the heads sit has the appropriate adjustment screws to line the head up. What is required is the correct **alignment tape** for the **tape format** and **speed**. These tapes can be purchased from their manufacturers.

Before you let your precious alignment tape near your recorder you must degauss it:

Degaussing

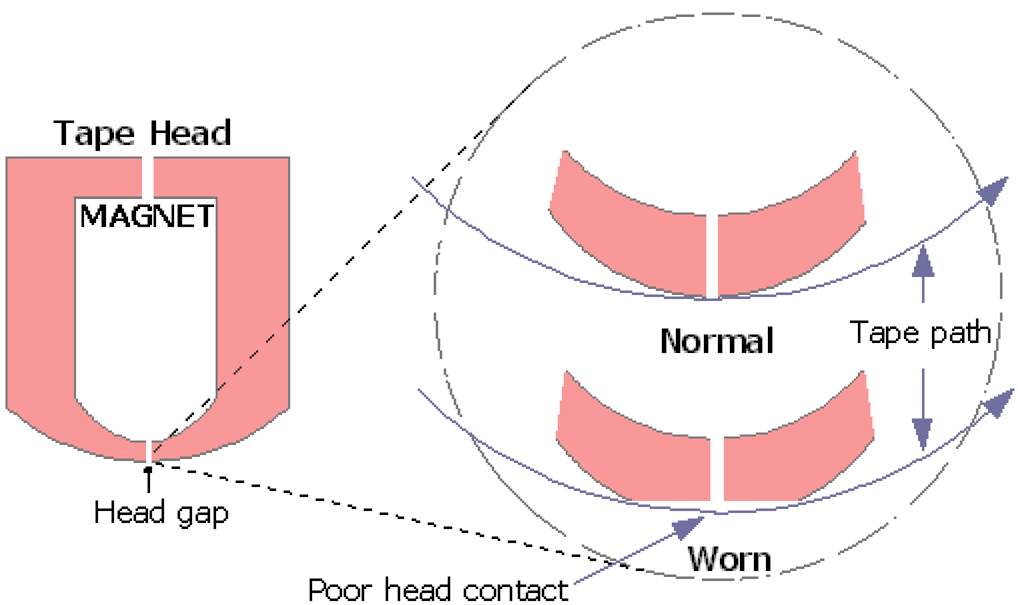
As the magnetised tape travels through the tape path the magnetic flux on the tape progressively is transferred to the metal parts such as the tape guides and heads. This build-up if left unattended will induce magnetism onto these parts and they will progressively erase the high frequencies on the tape as it passes. Therefore regular degaussing of these parts is a necessary maintenance procedure. A degaussing tool is required for this operation.



It is common practice to wrap a layer of insulation tape around the head of the degausser so that the metal doesn't damage your heads if you accidentally touch them. The technique here is to first **turn off the tape recorder**. If you don't you'll blow out all your circuits!! Then switch on the degausser with the head 1m(3ft) away from the recorder. Slowly bring the degausser head up to the play head and move the degausser slowly down and back up the head then slowly draw it away around a foot, then do the same to the next head and the next if there is one. Then using the same technique do the tape guides, rollers etc. Finally draw the degausser away from the machine around 1m(1ft) and switch the degausser off. Your machine is now degaussed.

Cleanliness.

The heads on a recorder acquire a build up of tape oxide after constant use so it is necessary to clean the heads and the tape guides regularly, like twice a day. Cotton buds and Isopropyl Alcohol (available from most chemists and drug stores) is the most common method. The main thing is to regularly check your heads to make sure that there is not too much build up. If there is you should recheck your head alignment or transport alignment. Too much tension on the tape can cause oxide build-up because of the increased pressure on the head. Another cause is worn heads so if you suspect worn heads get your machine checked by an expert. This is an exaggerated version of what happens:



Alignment Tapes

Alignment tapes come in three types according to their calibration. They can be either **CCIR, NAB or IEC** alignment. CCIR is recognised as being the British or European standard whereas the US is NAB. IEC only applies to the speed of 30ips. When you purchase an alignment tape you must know what your machine is calibrated to. Secondly you must know what speed your machine uses and if more that one speed is available you will need to purchase an alignment tape for each speed. These standards apply to the pre equalisation curves that a recorders have. Basically they boost the high frequency onto the tape and then reduce it back again on playback. This improves the signal to noise ratio. Each of the types, CCIR, NAB or IEC, have different EQ curves.

Secondly alignment tapes are recorded at a specific **flux level** measured in nanowebers. The first ampex alignment tapes were at 185 nw which meant that if you aligned your playback head to zero and then your record head to zero you would be recording a flux density of 185 nanowebers. Tapes then came out at 250 and then 300. You must check what level your alignment tape is at and what the capability of your machine is. The new tapes are capable of recording a higher flux than the old and the level has been going up an up over the years as tape manufacturing improves.

Bias

Simply the **bias** is a high frequency (typically between 150kHz and 280kHz depending on recorder type and model) signal that is added to the record signal to compensate for the irregularities in the ability of tape to hold flux uniformly. It is an adjustment in the record side calibration.

Alignment

You will need an **oscillator** capable of producing frequencies from 50Hz to 16kHZ for this procedure. Some consoles have a tone generator built in. Your machine will also have 1,2 or 3 cards for the **record, play and bias** adjustment.

OK so the machine has been cleaned and degaussed, now you can put your precious alignment tape on the machine.

The first thing to do is to check the tape path and confirm that the tape moves freely within the guides etc. Now you are ready to check the playback alignment. You must first check the azimuth. Take a playback signal from your two outer tracks, 1 - 8, 1 - 16, 1 - 24, and bring them up on your console at equal level panned centre. Now playback the 10 or 16kHz section on the tape and adjust the azimuth on the head so you get the highest reading. This assures that the head azimuth is correct and that the phase relationship between your outer tracks is the same.

Now play back the 1kHz tone and adjust the playback level to zero. If you wish to add more flux to tape because you have the latest tape you may wish to put 3db more level onto tape. In this case you should line up the playback reference to -3db. Now play back 10kHz and adjust the playback highEQ on your record card to read as close to zero as you can get or -3db if adding higher flux. Now playback the low frequency tones like 100Hz and line these up to zero (or -3db) or as close as you can get. Now recheck your 1kHz tone and do all the other tracks the same. You are now ready to align the record head. Remove the alignment tape and store it in a safe place away from anything magnetic like speakers etc.

Now put on a new roll of tape which you have labelled **Record Test Tape.** Keep this tape with your alignment tape for further alignment sessions. Put the tape on and put all the tracks into record. Roll tape and hit record and select playback on all of the tracks. You must now adjust the record level so that you get around 0db level played back. Now you are ready to check the **Bias level**. The adjustment for this will be on your bias card. Record a 10kHz tone and switch the machine to playback and adjust the bias level control. You will notice that as the signal rises it reaches a peak and starts to drop again. If not you must find this area. Line it up to the peak and then keep increasing it until it drops by 3db. This is called **overbiasing** by 3db. Do this for each track.

Now you must check the azimuth of the record head which is done by recording 10kHz onto your two outer tracks and playing them back through the console like we did before adjusting the azimuth so that you get the highest reading.

Now that the bias and azimuth are calibrated you can start the frequency response calibration starting once again with 1kHz followed by the high and low frequencies and adjusting the record card controls. A good machine should give you a flat response + or - 3db from 50Hz to 15Khz.

Digital Recorders

If you have moved to the new digital recorders and have just read the previous rave you must be breathing a sigh of relief as the digital recorders don't have frequency response or bias and azimuth adjustment. All they do is record 0s and 1s, albeit really fast at 48kHz/sec.

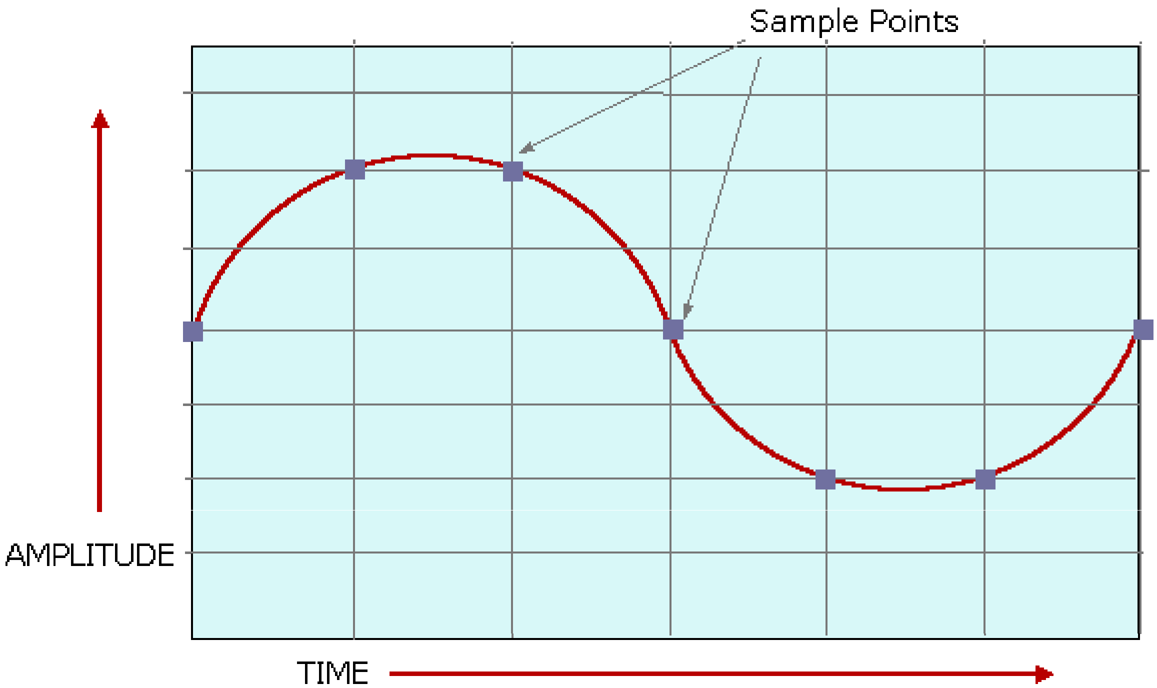
So lets start with some basic knowledge of digital sound. What does it mean when they say that a track is recorded in 16bit digital at 44.1K.

Bit Rate

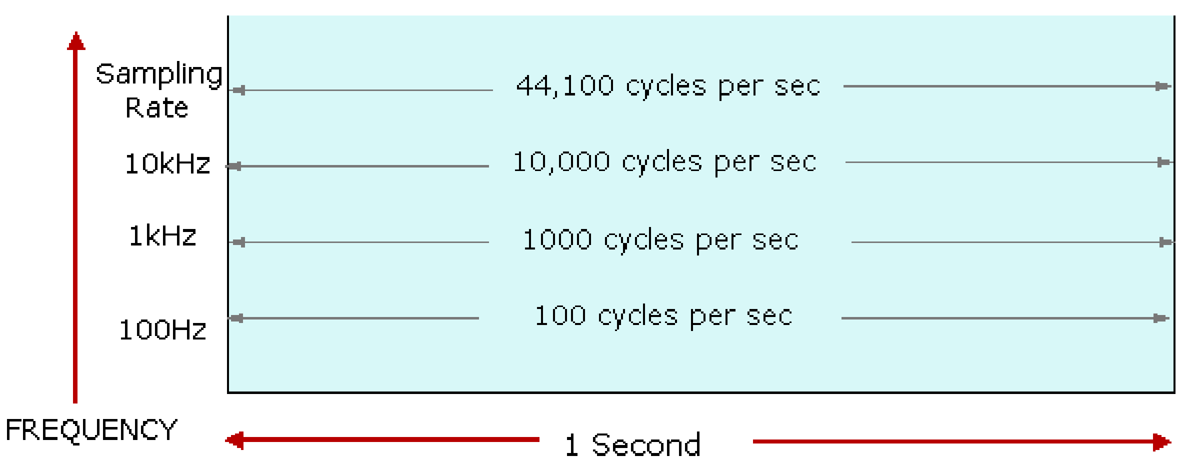
Digital sound is made up of **words** of 0's and 1's and 00, 11, 01, 10 are the four possibilities in a **two bit** word. A three bit word can be made up with 000, 111, 001, 010, 100, 101, 011, 110, which means there are eight possibilities. You see - 2 bit gives 4, 3bit gives 8, 4 bit gives 16, 5 bit gives 32, and so on. Now if we were to use the bit words to express volume with a four bit word we could give 16 different values for volume. So the higher the bit rate the more accurate the resolution be it volume, digital pictures or digital sound. So 24 bit digital sound has more resolution and accuracy than 16bit digital sound.

Sampling Rate

Digital sound is produced by sampling a sound (or should I say the electrical version of it) in real time and expressing it in bit words. Once you start sampling or recording digital sound a clock starts and progressive samples of what the sound is are taken. The rate at which the samples are taken is called the **sampling rate**.



The drawing above shows a wave of a sound being sampled. If the time in the drawing is 1 second, then there are 6 samples (the last one is the first in the next second) of the sound in one second or a sampling rate of 7. So obviously the higher the sample rate the more accurate the resolution. So when we say that the sound is 16bit, 44.1Khz it means that the sound is being sampled at 44.1 thousand times a second and it is being measures with 16 bit accuracy. In the above waveform the sampling volume levels given would be 0,2,2,0,-2,-2, Not a very accurate version of a simple waveform. But 44.1kHz, now that's fast, or is it? Lets look at sound in seconds.



In this chart you can see the relationship between the sampling rate and the waveforms it's sampling. 1kHz will have 44.1 samples taken of each of it's waveform as its oscillating at 10,000 waveforms a second. 100Hz will have 441 samples taken of each of its waveforms. But 10kHz will have 4.41 samples taken of each of it's waveforms. Now look at the first waveform we drew. In that drawing we took 6 samples of the waveform and got an amplitude reading saying 0,2,2,0,2,2. imagine how inaccurate 4.41 samples are of a complex waveform. That is why digital high frequencies sound harsh!! The industry has constantly denied this factor and even gone to the extent of saying the hear can't distinguish between a square wave and a sine wave above 7kHz. Pigs Bum.

At a sampling rate of 96kHz you get 9.6 samples of a 10kHz wave and believe me, you can hear it.

In an article by Rupert Neve, I read recently, he said that we should aim for 24bit resolution and 192kHz sampling rate if we want to equal the quality of high quality analogue recording. We will get there. DVD is already up to 24 bit 96kHz sampling so we are on the way. But if your 16bit, 44.1kHz CD sounds bright, consider what makes it bright and you will see that it's a false bright created by the high frequencies sounding like square waves!!

Why 44.1kHz Sampling Rate?

Why not 44, or a nice round number like 50. When the first engineers were inventing digital sound they had worked out the on/off, 0/1, idea and needed a way to represent it. The idea came to use white dots on a TV screen where a white dot was on and a black dot was off. Neat. So you record it like a video picture on a video recorder. That was fine, but the engineers had been caught out before. What about PAL (the European video standard) and NTSC? (the American and Japanese standard.) They weren't going to get caught up in that again, no way, so they configured a number that was compatible between the 528 line NTSC and 625line PAL and the number was 44.1kHz. Just a piece of useless info you might want one day!

What you can see from the above is how the digital recorders were developed. They were Beta Video Recorders with an external processor and digital audio had arrived. The beta video became the DAT and the DAT became the ADat and the D88 and they are all basically video recorder decks. The ADat used a SuperVHS deck while the D88 used a High8 deck. The basic SuperVHS deck was pretty awful and the ADat of today is a completely rebuilt deck. It's a shame the world chose to make the VHS deck the standard because the Beta Decks were far superior. Market forces don't always give the best outcome. Did you know that when an ADat or D88 records on a new track it plays the bit stream off the tape , mixes in the new track, and records it again. Now that's worth thinking about.

Hard Disk Recorders

Now we are entering a new era in recording with the advent of the computer. Now hard disk drives can store gigabytes of information and retrieve it at high speed. The Pentium 11, 350mHz computer I am writing this on will play back around 13 - 16 tracks of stereo digital audio in real-time at 16 bit 44.1kHz. It will also process it in real-time so I can add EQ, Compression, Reverb and Effects, I can cut and paste it, I can timestretch and pitch change it, I can even put it in tune. It cost under $US-

2,000 and will do my Internet, keep my tax records and play games as well. The hard drive can be backed up onto CD with my CD Burner. At the time of writing the new 1gig processors are being released and my local dealer is offering 27gig hard drives for around $300. As the telephone lines get better and the Internet gets faster and faster I will soon be able to play with a guitarist in London , a Bass player in the US, a drummer in Africa in real-time wow! where is it going to go?? Anyway back to reality.

The first units were stand alone units that had hard drives built in and all the companies were there with some kind of model. Meanwhile the programmers were busily rewriting their computer sequencer programs (Cubase and Notator) for the Macintosh platform and Cakewalk was writing theirs for Windows. Dididesign was developing ProTools and computer based hard disk recording was born. Now they all have computer based hard disk recording programs and due to market forces again they were all rewritten for Windows.

I won't go into the pros and cons of all the current programs, there are lots of qualified people able to discuss them all and I suggest you read the reviews suffice it to say that this is where the next generation of studios will be heading. In fact I would suggest that most of you reading this will have some kind of computerised hard disk recording already and are as excited about the future of this technology as I am. They are already offering 24bit 96kHz sampling and the new generation effects units are getting better an better. For the price of an ADat you can get a whole studio!!!!!

Mixing

Without doubt the hardest part of recording is mixing, yet it is also the most enjoyable as this is when everything starts to come together and all the hard work justifies itself. A good mixer paints a picture in sound that attracts the listener and conveys the song clearly and simply. I could sum up a good recording as a series of priorities which are:

- **The song**
- **The singer**
- **The feel or groove**
- **The fiddley bits**

The **song** is set from the start and a good producer will have chosen a song that has 'something to say' and a good mixer will convey that something to the listener.

The **singer** is the next most important aspect and a good mixer will allow the singer to be heard and the lyrics to be conveyed clearly but with style. There is nothing more annoying than hearing a track and not being able to distinguish the lyric amongst a babble of instrumentation. Fortunately you don't hear recordings like that on commercial radio as they just don't get a look in. Engineers are often guilty of cluttering up tracks with all sorts of tricks and garbage that distract from the song and the singer because they know the song so well after days in the studio that they think everyone hears it like they do. If the track is to have a chance of commercial success it must be understandable from the first hearing. Always **underestimate** the ability of the listener as they are not professional listeners like you.

The **feel or groove** is what catches the listeners attention initially and sets up the mood and emotion of the track. This is created by careful balancing of the rhythmic aspects of the track be it drums, percussion or a great guitar groove.

Finally there is the **fiddley bits** as I call them, they are the musical phrases linking lyrics, joining verses to choruses and filling solo sections etc. that are created by the guitar licks, the piano fills, the answering vocal phases etc.

So where to start?

Monitoring Speakers

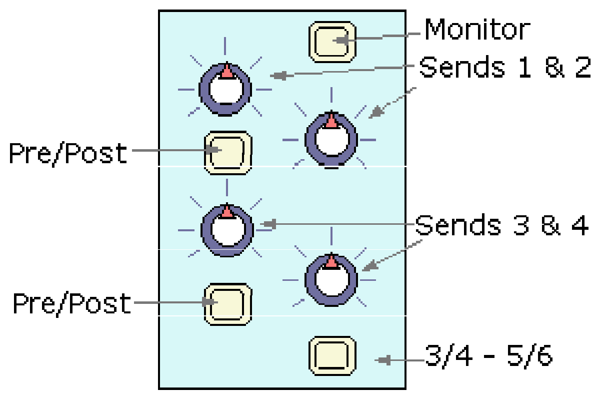
Monitoring speakers come in two types. Nearfield and Main. I like to use both. I work primarily on the nearfield to establish my balances etc. and then every now and then I will switch it up to the big speakers as they give a better idea of the low frequency balance, plus it sounds good eh! (I was a Yamaha NS10 freak for years but now I'm totally sold on the Event 20/20. Well done Event!) To me a big speaker system is like a magnifying glass, it blows the sound up and you can hear more but for a big system to be really good you have to flush mount them and have good speakers and a good amplifier system. Can I say here that I don't like equalised speaker systems. If they don't sound good flat, get another speaker!!

Level Structure

The first important procedure is to setup your console for mixing. The first requirement is to setup your levels to and from your master recorder, usually a DAT. If your console has an oscillator send tone to the DAT and balance left and right channels. Then check that the return to your console, which is what you'll monitor, is balanced correctly left and right. At this stage it is also recommended that you insert your master compressor either in the master stereo output inserts or inline between the console and the DAT and line up correct left/right balance. This procedure is very important as it effects your level structure from then on and if you don't do it now you can end up with your levels all over the shop later.

Aux Sends and Returns

Next you must establish your auxiliary sends and returns.



One of the best ways to get perspective and separation within your mix is to what I refer to as **"putting everyone in their own space"**. You can achieve this through the use of reverb and effects. I like to have one reverb unit dedicated to the drums. No other instruments are sent to this effect, only the drums which will put them in their space. The choice of reverb for drums depends entirely on the track but I start by putting reverb on the snare and going through the presets to find the one that works best for the track. I find it usually ends up with a bright reverb of shortish length around 1 - 1.2sec reverb time.

Note: A very fine producer in OZ was once quoted as saying "Give me a studio with 10 Midiverbs over a studio with one Lexicon 224XL" We all know what fantastic units the Lexicons are but if it's all you've got you are limited to only one perspective.

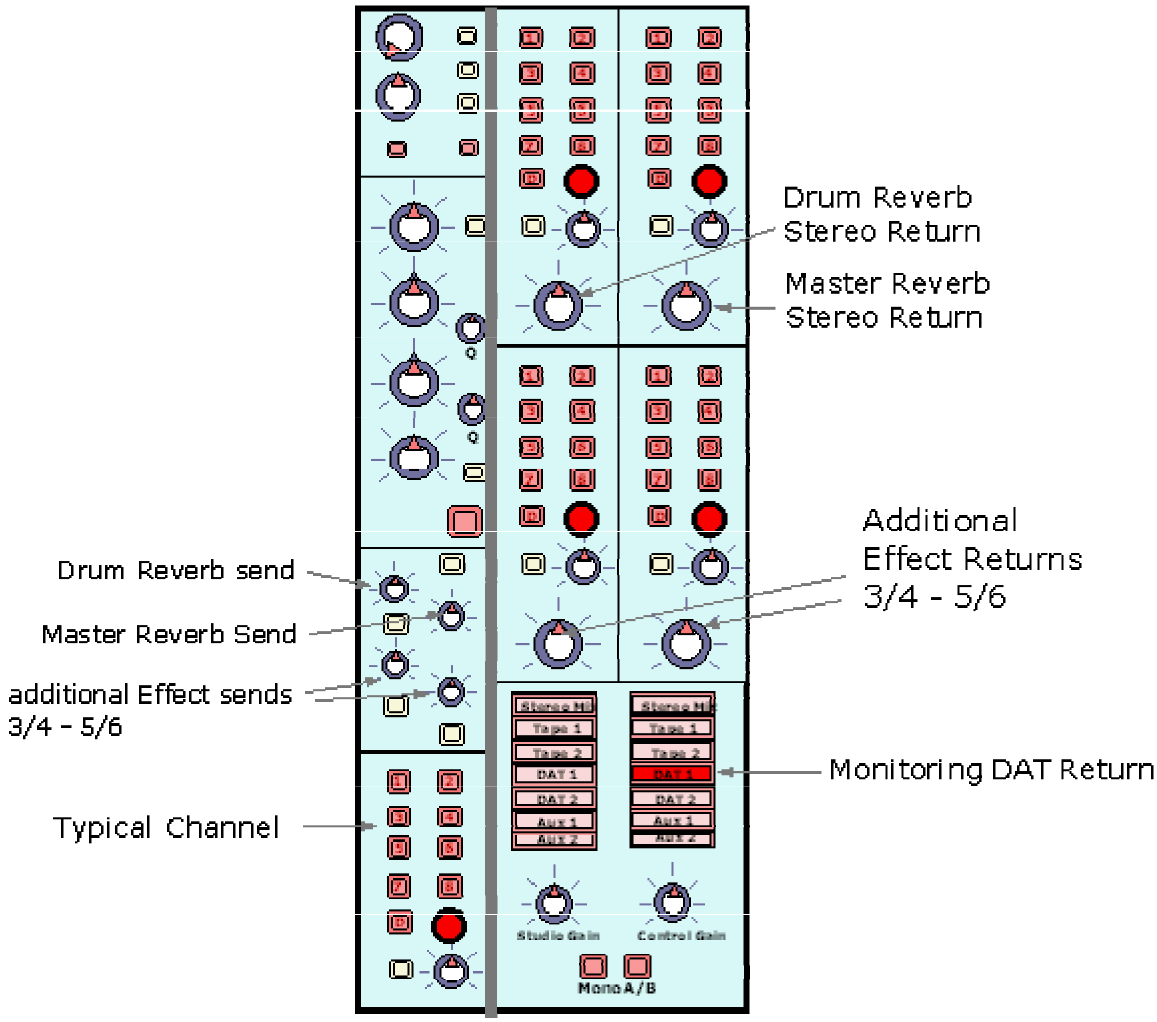
Next I'll dedicate a reverb unit to act as my overall reverb effect. I look for the best (not necessarily most expensive) unit in the studio for this will be my **master reverb** for vocals etc.

In the example above there are 6 sends with 5 & 6 being an option over 3 & 4. I therefore like to use 1 for my drums and 2 for my master verb. Then I can assign the others for effects. I do this so that I can always add master reverb as well as effects if necessary and if I had used say 3, I couldn't put master verb on channels where the effect was assigned to 5. Should I use a **stereo or mono send to the effects??** To be perfectly honest I don't think it matters. Most of the stereo input reverb units I find have a mock stereo input, not a true stereo. If you use two sends it really doesn't make a difference unless you are working with the more expensive units like the aforementioned Lexicon, and even then I question the validity of two inputs especially if you are limited in the number of sends.

I then assign the sends 3 - 6 to additional effects like **delay**, pitch change etc. to act as perspective enhancers. When establishing delays I set them to the track tempo. See **Tempo Chart.** The idea is to add these perspective effects so you only just hear them when in solo and they appear to disappear when mixed into the track. Bob Clearmountain - the world famous mixer - always had two delays going, one on eighths and the other on 16ths. It puts an air around instruments and if mixed in correctly you won't actually hear them, just sense them. **Pitch change** is another effect to consider with say the left channel set to -.008 cents and the right to +.008 cents. This effect is great on harmony vocals and it puts them in a different space form the lead vocal. Finally a **soft flange or chorus** is another effect I'll have as an option for guitars etc. See **Effects pages** for settings.

Make sure that all your effects are returned through the effect returns and assigned to the master stereo output. If you are fortunate enough to have spare channels on your desk you can return your delay and chorus type effects back through a console channel as this gives you the option of adding master reverb to them and using the channel EQ. Delays can soften if master reverb is added to their returns plus you can attain your feedback from the console instead of using the control on the effect unit. Say you are using send 3 to a delay unit you can feed back to the delay by sending the send 3 on the return back into the unit. N.B. Incidentally, make sure that the **dry/wet** or **mix** controls on your effect units are set to wet as you are only wanting the effect from the units and you won't need any dry sound. (If you are using the Alesis Quadraverb check this as all the default settings have 50% dry and 50% wet.) The returns from effects are usually panned full stereo L/R, but you may wish to bring the drum reverb back half L/R to separate the two.

Your console should now be setup like this



Mixing

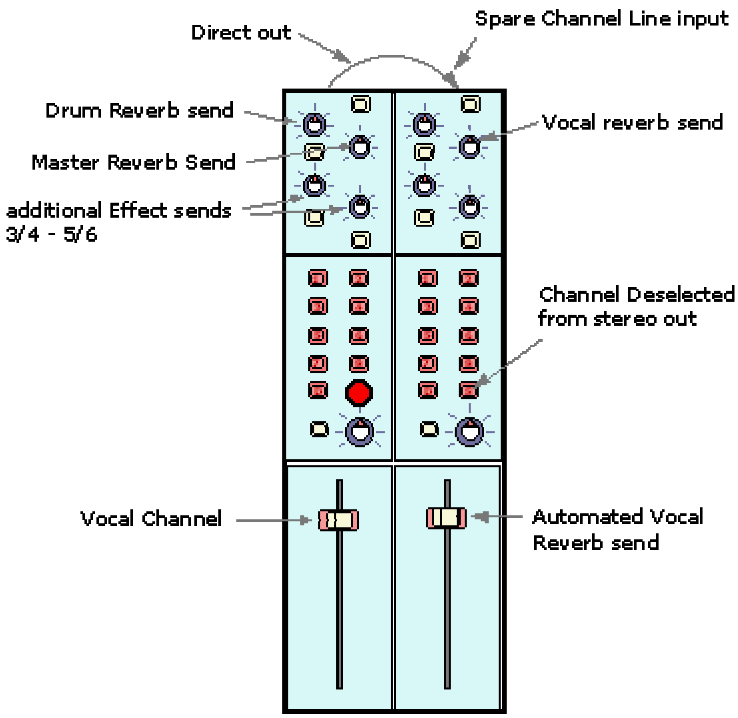
Some mixers start with the drums, others start with the vocal. I must admit I start with the drums as they convey the **dynamic** of a song. Hopefully you will have **automation** on your console, if not, you must now start setting up a series of moves and remember where and how they occur because, let's face it, the balance within a mix is not static, it varies continuously throughout a song. For example lets say the drummer plays a rimshot snare through the verses and full snare in the chorus. The EQ required on the rimshot snare sound is probably different from the chorus snare sound so I often split the snare return from the recorder into two console channels so I can EQ and effect each separately and automate the switch between the two. For example, the snare in the chorus will probably require more reverb than the rimshot so having a separate channel allows for that. Automation also allows for the tom mikes to be muted when not needed thus reducing the spill of the rest of the kit and cutting out the constant ringing of the toms which occurs with undampened toms. The overhead mikes also will need to be ridden throughout the track, I tend to lower the overhead mikes when the rimshot is playing to achieve a tighter sound, then I lift them in the chorus when the full snare comes in. Reverb on the overheads gives reverb on the cymbals but it also adds reverb to the snare in the chorus and lifts the whole ambient sound of the kit. This has the effect of changing the **perspective** of the drums in a mix. You can also change the perspective by putting master reverb on the overheads which blends with the drum reverb.

Once we have achieved a reasonable balance of the kit and the dynamics are set in place we can add the bass. The bass and the kick drum will determine the bottom end of the track so the balance between the kick and bass is critical. The kick will give the bass punch and attack when they hit together.

Note: I must say a few words here about bottom end. The big mistake in mixing is to make the bottom end sound too big by adding lots of bottom end EQ to the kick and the bass. You must bear in mind how the track will be played back by the listener. Nowadays everyone has a stereo system with bass boost as an option either as a **loudness** switch or as a **sub bass** control. Everyone who has this option has it switched on!! If you get out a few of your favourite recordings and listen to them on your mixing speakers you will find that they are relatively shy in the bottom end and yet when played through your average boom box sound tight and fat. You have to start to understand what a **flat response** really means and learn to mix that way. If you put a bass on a VU meter you will notice how much energy there is in the bottom end. A bass peaking to zero will have the same apparent loudness as a highhat peaking to -30db. That's because a hithat has no real bottom end compared with a bass so be careful with your low end EQ on basses and kick drums. I like to solo the two together and EQ them so that they are tight but not boomy.

Add the vocal

OK, so the bass and drums are now at their first mix level so next I will add the vocal and mix it sitting just above the bass and drums. This might mean an EQ change so they all sit tightly together. The vocal might need to be ridden with the automation and I'll probably **compress** it again to keep the dynamic range within the boundaries of the whole track. I often find that the reverb on the vocal will need to be ridden so that the screaming high notes need more reverb that the quiet intimate sections in the verse. Here I take a feed from the **direct out of the vocal channel** and bring it up on another channel on the console. I then deselect this channel from the stereo mix output so it goes nowhere but the aux sends are still working. By adding reverb to this channel I can use this channel to ride the reverb on the vocal as an automated send.



Adding the rest

Now we can start to add the fiddley bits like the rhythm guitar and keyboard pads etc. adjusting their balance to fit tightly but not overpowering the vocal. (Please understand I am not defaming guitars etc. by calling them fiddley bits, they are just as important as every other part) The track should now be starting to take shape. If the dynamics of the drums and vocal have been set correctly the placement of the additional instruments will fall into place easily. The vocal harmonies, and solo instruments can now be mixed into the track and we are nearing the completion of the first mixdown.

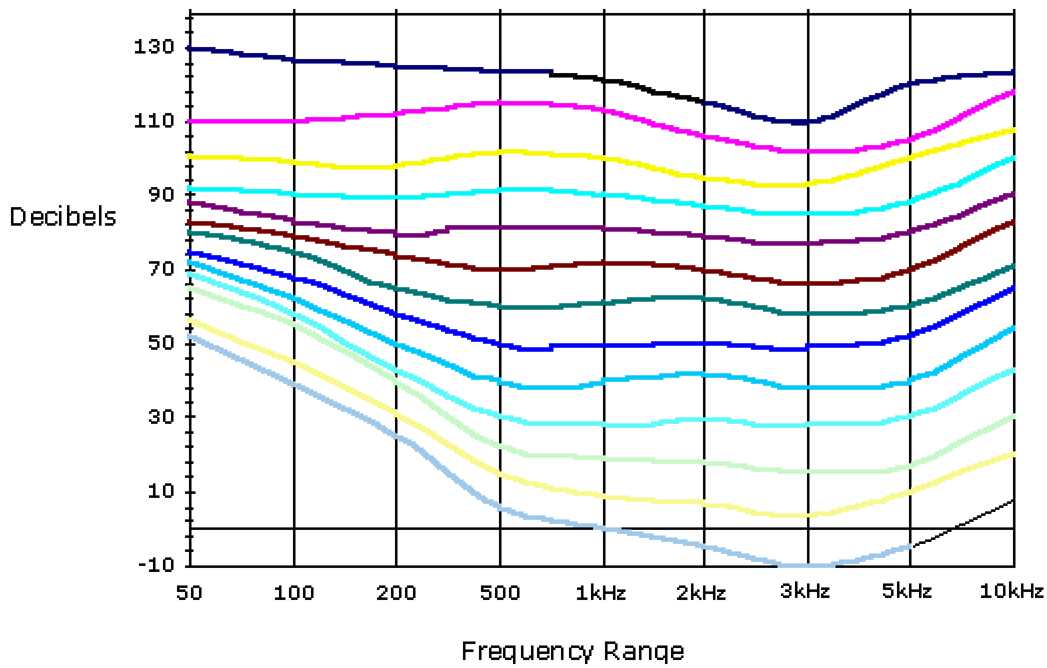
Note: It is important to keep checking your mix in **mono**. Unfortunately stereo and mono are not compatible. When you switch to mono, instruments that are panned centre are 3db higher than in stereo so your vocal, kick and snare, for example, will come up in the mix. Some engineers actually make two mixes of a track: One that is full wide stereo with full dynamic range for home listening and one where all the hard left and right signals are panned to the centre or half centre and compressed for radio. It's really hard because if you make a mix sound great on a good home hi-fi it won't have the tightness and punch a mix made for commercial radio will have where the dynamic range is low. It's common practice to make separate mixes of the **singles** from an album for radio whereas the remaining tracks are mixed totally for home hi-fi. I think you will find that most commercial records are mixed to sound great on FM Radio.

Rest and Recreation

It is important that you constantly give your ears a break during the mixing process as your ears have little compressors in them that will progressively shut your ears down. Have you noticed that when you've been in a loud club with a loud band when you go outside you can't hear as well. It's part of your ears protection system and a cup of coffee in another room watching TV or something will allow them to start opening back up. I like to "**mix from the kitchen**" as I call it. This means playing the automated mix and listening to it from an adjacent room with the control room door open, you'd be surprised how clearly you can hear the balance between instruments when you get away from the direct sound from your speakers. The relationship between the bass and kick, the balance within the harmonies, the clarity of the vocals etc. all become clearer when you relax and listen from another room.

Monitoring Level

Unfortunately the human ear is not flat at all levels. Some guys called Fletcher and Munson worked out what the response curve of the ear was and found that at low levels the ear missed out on the low frequencies and the high frequencies, whereas at loud levels it was the opposite.



From the above chart you can see that around 80 - 90db the ear is the flattest. The fact that we don't hear low frequencies and high frequencies at low levels created the **Loudness** switch on stereo systems which boosts the low and high frequencies to compensate for the ear. Unfortunately, Joe Public doesn't know this but knows that when it is switched in things sound fatter and brighter so they leave it in all the time. It is generally recognised that a level of 85db is where the ear is at it's flattest so don't mix too loud if you want a flat response.

The important thing about mixing is **apparent loudness**, or relative loudness. If I whisper into a mike and then I shout into a mike the shout will appear louder because I know that shouting is loud. It's the same with mixing. You create an illusion of loudness, everything is relative. You can't get bigger if you are already at your maximum. If I mix a soft acoustic guitar and vocal and peak to zero then bring in a full kit and grunge guitar also peaking to zero it will apparently get louder because I know that drums and guitar are loud. Mixing is the art of making signals that all peak to zero sound as if there is a dynamic range. Nowadays with the excellent compression systems we have most recordings are heavily compressed. I was told of a producer who hired a mixing engineer to mix an album. The guy turned up with racks and racks of compressors and set about compressing every track. He had one compressor for this and another for that etc. In the end the whole mix was pumping away and almost mixed itself. That album went on to sell millions of copies world wide. Those of you who have played with Waves Ultramaximiser will know what compression can do for a mix. If you watch most modern pop recordings on a VU meter the needle is almost static varying only a few db yet the tracks go from quiet intros to full on chorus and solo sections yet still there is only a small variation in level. So setting compression (and limiting) levels is important. I will always have a compressor across the output of my mixes as it helps control the peaks and brings up the loudness of the track but I may use individual compressors on separate channels.

Finally - do take the time to get a good mix. If you don't you have not given justice to all the effort you put into recording it in the first place. It may take a few remixes, so what - it's the final product that counts.

TEMPO DELAY TIMES

Thankyou BEATCALC - by FUali for the figures

TEMPO	1/4 Note	1/8 Note	1/16 Note	1/4 * Note	1/8 * Note	1/16 *Note	1/4 Triplet	1/8 Triplet	1/16 Triplet
60	1	.5	.25	1.5	.75	.375	.666	.333	.166
61	.983	.491	.245	1.475	.737	.369	.655	.327	.164
62	.967	.483	.241	1.451	.725	.363	.645	.322	.161
63	.952	.476	.238	1.428	.714	.357	.635	.317	.158
64	.937	.468	.234	1.406	.703	.351	.625	.312	.156
65	.923	.461	.230	1.384	.692	.346	.615	.307	.153
66	.909	.454	.227	1.363	.681	.341	.606	.302	.151
67	.895	.447	.223	1.343	.671	.336	.597	.298	.149
68	.882	.441	.220	1.323	.661	.331	.588	.294	.147
69	.869	.434	.217	1.304	.652	.326	.579	.289	.145
70	.857	.428	.214	1.285	.642	.321	.571	.285	.142
71	.845	.422	.211	1.267	.633	.317	.563	.281	.140
72	.833	.416	.208	1.250	.625	.312	.555	.277	.139
73	.821	.410	.205	1.232	.616	.308	.548	.273	.137
74	.810	.405	.202	1.216	.608	.304	.540	.270	.135
75	.800	.400	.200	1.200	.600	.300	.533	.266	.133
76	.789	.394	.197	1.184	.592	.296	.526	.263	.131
78	.769	.384	.192	1.153	.576	.288	.512	.256	.128
79	.759	.379	.189	1.139	.569	.264	.506	.253	.126
80	.750	.375	.187	1.125	.562	.281	.500	.250	.125
81	.740	.370	.185	1.111	.555	.277	.493	.246	.123
82	.731	.365	.182	1.097	.548	.274	.487	.243	.122
83	.722	.361	.180	1.084	.542	.271	.482	.240	.120

84	.714	.357	.178	1.071	.535	.268	.476	.238	.119
85	.750	.352	.176	1.058	.529	.264	.470	.235	.117
86	.697	.348	.174	1.046	.523	.261	.465	.232	.116
87	.689	.344	.172	1.034	.517	.258	.459	.229	.115
88	.681	.340	.170	1.022	.511	.255	.454	.227	.113
89	.674	.337	.168	1.011	.505	.252	.449	.224	.112
90	.666	.333	.166	1.000	.500	.250	.444	.222	.111
91	.659	.329	.164	.989	.494	.247	.439	.219	.109
92	.652	.326	.163	.978	.489	.244	.434	.217	.108
93	.645	.322	.161	.967	.483	.242	.430	.214	.107
94	.638	.319	.159	.957	.478	.239	.425	.212	.106
95	.631	.315	.157	.947	.473	.236	.421	.210	.105
96	.625	.312	.156	.937	.468	.234	.416	.208	.104
97	.618	.309	.154	.927	.463	.232	.412	.206	.103
98	.612	.306	.153	.918	.459	.229	.408	.204	.102
99	.606	.303	.151	.909	.454	.227	.404	.201	.101
100	.600	.300	.150	.900	.450	.225	.400	.199	.100
101	.594	.297	.148	.891	.445	.222	.396	.197	.099
102	.588	.294	.147	.882	.441	.220	.392	.196	.098
103	.582	.291	.145	.873	.436	.218	.388	.194	.097
104	.576	.288	.144	.865	.432	.216	.384	.192	.096
105	.571	.285	.142	.857	.428	.214	.381	.190	.090
106	.566	.283	.141	.849	.424	.212	.377	.188	.094
107	.560	.280	.140	.841	.420	.210	.373	.186	.093
108	.555	.277	.138	.833	.416	.208	.370	.185	.092
109	.550	.275	.137	.825	.412	.206	.367	.183	.091
110	.545	.272	.136	.818	.409	.204	.363	.181	.090
111	.540	.270	.135	.810	.405	.202	.360	.180	.090

112	.535	.267	.133	.803	.401	.201	.357	.178	.089
113	.530	.265	.132	.796	.398	.199	.354	.176	.088
114	.526	.263	.131	.789	.394	.197	.350	.175	.087
115	.521	.260	.130	.782	.391	.195	.347	.173	.087
116	.517	.258	.129	.775	.387	.194	.344	.172	.086
117	.512	.256	.128	.769	.384	.192	.341	.170	.085
118	.508	.252	.127	.762	.381	.190	.339	.169	.084
119	.504	.252	.126	.756	.378	.189	.336	.168	.084
120	.500	.250	.125	.750	.375	.187	.333	.166	.083
121	.495	.247	.123	.743	.371	.186	.333	.165	.082
122	.491	.245	.122	.737	.368	.184	.327	.163	.082
123	.487	.243	.121	.731	.365	.183	.325	.162	.081
124	.483	.241	.120	.725	.362	.181	.322	.161	.080
125	.480	.240	.120	.720	.360	.180	.320	.159	.080
126	.476	.238	.119	.714	.357	.178	.317	.158	.079
127	.472	.236	.118	.708	.354	.177	.315	.157	.078
128	.468	.234	.117	.703	.351	.175	.312	.156	.078
129	.465	.232	.116	.697	.348	.174	.310	.154	.077
130	.461	.230	.115	.692	.346	.173	.307	.153	.076
131	.458	.229	.114	.687	.343	.171	.305	.152	.076
132	.454	.227	.113	.681	.340	.170	.303	.151	.075
133	.451	.225	.112	.676	.338	.169	.300	.150	.075
134	.447	.223	.111	.671	.335	.168	.298	.149	.074
135	.444	.222	.111	.666	.333	.166	.296	.148	.074
136	.441	.222	.110	.661	.330	.165	.294	.147	.073
137	.437	.218	.109	.656	.328	.164	.292	.145	.073
138	.437	.218	.109	.656	.328	.164	.292	.145	.073
139	.431	.215	.107	.647	.323	.161	.287	.143	.072

140	.428	.214	.107	.642	.321	.160	.285	.142	.071
141	.425	.212	.106	.638	.319	.159	.283	.141	.070
142	.422	.211	.105	.633	.316	.158	.281	.140	.070
143	.419	.209	.104	.629	.314	.157	.279	.139	.069
144	.416	.208	.104	.625	.312	.156	.277	.138	.069
145	.413	.206	.103	.620	.310	.155	.275	.137	.069
146	.410	.205	.102	.616	.308	.154	.274	.136	.068
147	.408	.204	.102	.612	.306	.153	.272	.136	.068
148	.405	.202	.101	.608	.304	.152	.270	.135	.067
149	.402	.201	.100	.604	.302	.151	.268	.134	.067
150	.400	.200	.100	.600	.300	.150	.266	.133	.066
151	.397	.198	.099	.596	.298	.149	.264	.132	.066
152	.394	.197	.098	.592	.296	.148	.263	.131	.065
153	.392	.196	.098	.588	.294	.147	.261	.130	.065
154	.389	.194	.097	.584	.292	.146	.259	.129	.064
155	.387	.193	.096	.580	.290	.145	.258	.128	.064
156	.384	.192	.096	.576	.288	.144	.256	.128	.064
157	.382	.191	.095	.573	.286	.143	.254	.127	.063
158	.379	.189	.094	.569	.284	.142	.253	.126	.063
159	.377	.188	.094	.566	.283	.141	.251	.125	.062
160	.375	.187	.093	.562	.281	.140	.150	.124	.062