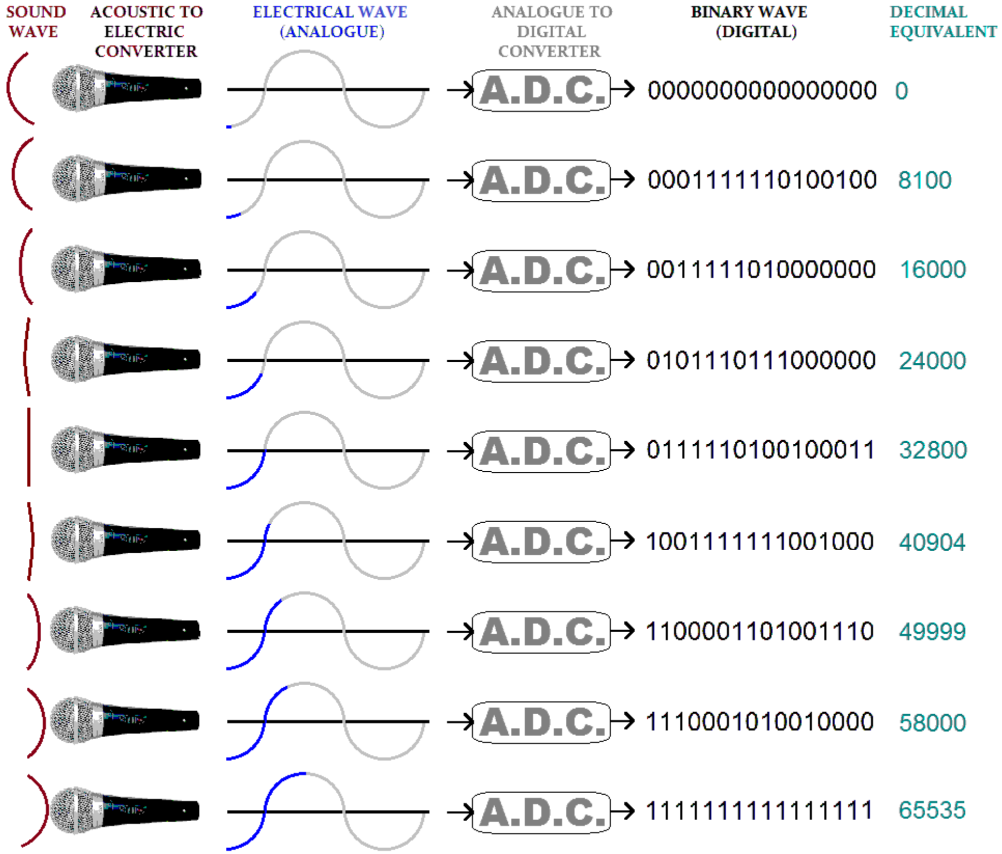
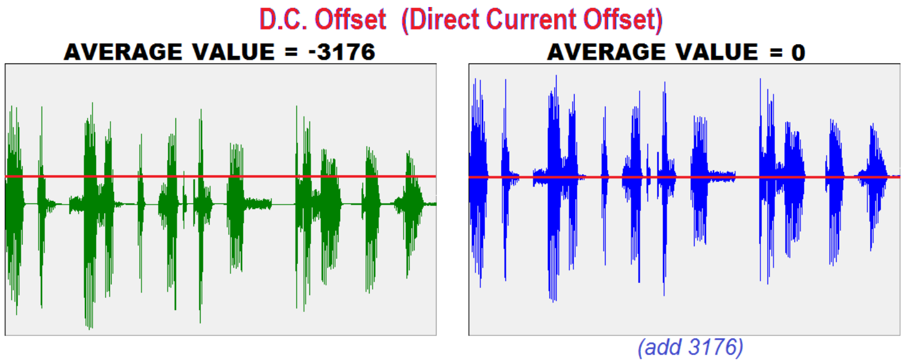

Multi-track Recording
✞ Church Audio ✞

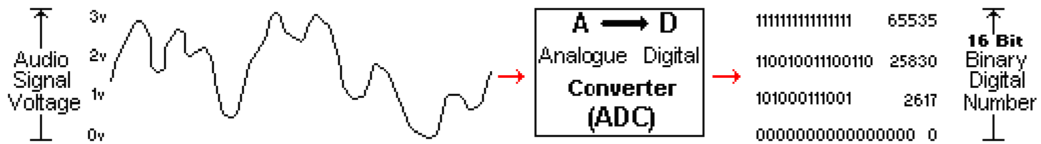
First I am going to discuss basic digital recording theory. This may be a little heavy going, don't worry, it's just to give you some reference notes so that if your recording software should ask you questions about how you want to set your sampling frequency, bitrate, float, anti-alias filter, DC offset, dither, jitter and a touch of noise-shaping, over-sampling etc you can at least make an informed decision.



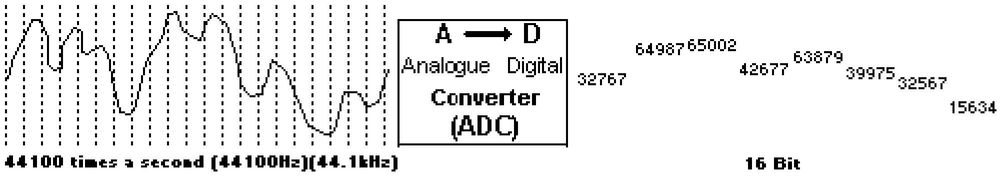
Digital recording involves converting the incoming Analogue signal from the microphone into Digital numbers using an Analog-to-Digital converter. Once in the digital domain modern software offers us tremendous opportunities to manipulate the sound without any noticeable loss of quality.



A Digital Recorder, e.g. Analogue to Digital Converter (A.D.C.), spreads the incoming sinewave across a central line. This is the zero (silence) line which should be at half way. It is also called the point of D.C. (Direct Current). For various electronic reasons the A.D.C. may sometimes have a zero line in a slightly different place, and when the digitised wave is subsequently displayed on the Recording Software you will see that the entire wave seems to be above or below the line.
To demonstrate this, if you look at the wave on the left, you can see it is offset from the D.C. line.
While DC offset isn't audible, it isn't desirable because it will cause the software to calculate things wrongly. Recording software always offers you the possibility of correcting D.C. offset. *Sometimes recorders even have a feature allowing you can choose to have it done automatically at the moment of recording.*



For a professional sound quality, we set our Recorder (A.D.C.) to at least 16 bits. With a **Bit Rate** of 16 bits the Recorder will divide the incoming signal voltage into **65536** parts (2^16 =65536). This is the professional minimum. The higher the Bit-rate you choose, the greater the resolution and quality of the recording.



Because you measure (sample) the incoming signal voltage at a constant rate (Sampling rate) (Sampling frequency) there is a side-effect. Tones (frequencies) are generated by your constant sampling speed, which will mix in with the original audio. They are called **Alias Frequencies** and show up mostly as inter-modulation distortion. In the 1920's ***Harry Nyquist*** of Bell Laboratories discovered that sampling at twice the frequency of the highest frequency that you desire pushes the Alias frequencies out of the audio area, so they don't bother anyone.
The highest frequency we normally use in audio is around 22kHz, which is the maximum of human hearing. So… if we make 22kHz our ***halfway (Nyquist)*** *frequency* we would select a **Sampling Frequency** (**Sample Rate**) of 44kHz (double what we need).
Computers actually use 44.1 kHz (44100 Hz) as the international CD quality sampling rate (44100 is an easy number for micro-processors).
Unfortunately, simply doubling our sampling frequency still allows a little inter-modulation distortion to sneak through… until 1948 when Claude Shannon, also of Bell Laboratories, demonstrated that if you filter out the high frequencies (using a low-pass filter) above your desired maximum frequency then no errors due to aliasing will result. Your Digital Recorder will automatically apply an **anti-alias filter** at half the sampling frequency you choose to record at.

Improving your recording:
As nothing is perfect, not even digital timers, and not every sample is taken at exactly the same interval every single time. This tiny amount that the Recorder slips back and forth is called **Jitter** (*Timing Instability*). An additional problem is that the Bit-rate you choose will be a whole number (16, 24, 32) and not a fraction (8.1, 16.7, 24.2, 32.9). The computer subsequently calculates using whole numbers (integers). Unfortunately, your incoming audio signal voltage isn't going to be that simple and will nearly always be in fractions. This means the computer will 'round-off' to the nearest number that it can work with. This is called **Quantizing** which means '*deciding between two values*'. For example, if the signal voltage was 3.15v then the computer would round off to 3v. We now have a quantizing error at that point of 0.15v, which is a distortion of 4.7%. Quantising error creates a random noise that mixes in with our audio. We call this *quantizing noise*. It is most easily heard when the signal is at very low levels, such as fading out, or on the tail of a Reverb. It is a *hissing* type of sound.
*Quantisation Solution #1- When selecting your Bit-Rate on the Recording software, always prefer a bit-rate that says **FLOAT**. This is your best option! Floating numbers work in fractions, reducing quantisation noise dramatically!*
*Quantisation Solution #2- Check the Recording software where you select the Recording format to make sure the **DITHER** is on. Dither (low-level random noise) is how we fight fire with fire. Curiously, by adding the tiniest bit of random noise to the sound, we increase the computer's ability to read very small values, in fact, smaller than our smallest bit. Dither will also diffuse any alias frequencies that might have slipped though the net.*
*Quantisation Solution #3- Another solution we use to fight quantisation is to **OVER-SAMPLE**. This means that you sample at a frequency far higher than you require. This moves the halfway (Nyquist) frequency a lot further up. Signal-to-noise is greatly improved as well, and Jitter, Alias Frequencies and Quantisation Noise* are *reduced dramatically. Increasing your **BIT-RATE** beyond what you need contributes even further.*
*Quantisation Solution #4- The Quantisation Noise can (optionally) be filtered using a special feedback filter system. By **Noise-Shaping** we can filter out the quantization noise more strongly at the point where the ear is most sensitive.*

Your Recording interface (A.D.C.) will have a number of inputs (**8, 12, 16** etc). You are limited to this number of inputs (at any one time). You can however **pre-mix** together some channels at the mixer before passing to the Recorder, which gets you more inputs.
Although the purpose is to record the live performance as it is, remember that you can add singers or musicians at a later moment by playing the recording back to them through headphones, and they perform in time with the recorded music. This is called **Overdubbing** (or **Tracking**) and even with Live recordings it can sometimes be the case to overdub things.

MULTI-TRACK RECORDING SESSION (*for the church CD*)

You may choose to record **all-at-once** which will give a 'live' sound with everyone particularly in-time with each other, causing what we call a '**tight**' feel. If someone makes a mistake, then everyone has to play it again. If someone's performance is border-line, then you may have to accept the "less than perfect" performance because everyone else played well. Recording all-at-once is preferred by Jazz musicians who care more about feel and live sound than everything being perfect. For some church music teams this is the simplest method. Record the same song a few times (Take 1, Take 2, Take 3 etc) then pick the best.

For those who want a more perfect performance on their recording they can record in phases (**over-dubs**). Typically, the drums, bass and a rhythm guitar and/or rhythm piano are recorded first These are what we call the **rhythm tracks**. It is helpful to have the lead singer singing into their headphones, so they get the right feel (a *guide vocal* track) so that when the singer sings it properly later the music will be following their singing style. The Rhythm Tracks are crucial, they generate the heartbeat (the groove) for the song and really need to be well played. Work with these musicians until you have a great rhythmic and harmonic background on which to build the rest of the song. Record again and again until all interested parties are happy, then we move on to the next layer of overdubs until the song is complete. This is the Recording Studio approach and permits you to get a very polished finish.
When using a Software Recorder there will be a slight lag (called **latency**) heard in the headphones. This is the delay in time between the sound already recorded being sent back to your headphones, and your new recording going in to digital, and then coming back to you. Some software lets you adjust the *latency*, so it isn't distracting (*it can be a little weird*).

Consider the 'Line Up' carefully: What are the instruments? Who is playing accompaniment and who are the soloists? Any Singers? Are there backing singers? You need to understand the Line Up so that you can begin to plan the order in which you will record them. What will you record 'in direct' (initial tracks) and what will you overdub in subsequent phases? Do you have suitable Microphones, DI's, Headphones and Headphone Amp? Do you need to borrow/hire? Correct Cables and Adaptors? Separation Panels (Gobos)? Do you need a Sound Booth, a Side-room?

Track Setup:
Create a new Session folder on your Recorder. Set the Sampling Frequency and Bit-rate for the session (CD Quality would be 44.1kHz 16bit).
Create your Audio tracks. Set them to be single Mono tracks, or Stereo should the case arise.
Name each track the way you will best remember it. Don't leave them as track 1 track 2 etc as it's a recipe for confusion later.
Select Audio Devices (Hardware) on each track: *Input Source (where you are recording from e.g. Interface). Output (so you can hear the recording in headphones).*
Arm the track(s) you will be recording, monitor and adjust the incoming level using the Level Meters.
A reverb effect that is just in the headphones (but not recorded) for the singers will usually improve their performance as their voice will sound more pleasant.
Check that they **tune** their instruments, it will have a big effect on the sound.

RECORDING (TRACKING):

When recording the task is to get:

- Good electrical levels with the peaks staying just under the zero (this is digital, not analogue, going over zero is immediate distortion).
- Clean sound with no background noises.

At the moment of recording: no electronic device (or light switches) can be turned on or off, cell-phones turned off, and no audio connectors plugged in or taken out while recording as these all cause clicks and buzzes in the recording. Random clicks occurring in a recording can occasionally be caused by a water heater thermostat, a soft-drink dispenser machine, or refrigerator, or washing machine… anything that switches on and off and doesn't have the appropriate capacitor on its power cable can cause a click in the power and can enter the recording. Only devices necessary for the recording should be switched on and in the recording chain. If there is a background noise and you can't locate it then you may need to disconnect everything that is plugged in and systematically reconnect one at a time waiting for the reappearance of the noise. We always trace our unwanted noises by a process of elimination and substitution.

Don't use equalisation when recording, unless there is a very real need. Concentrate on getting a good clean sound using the right microphone in the right place, at a healthy electrical level. The recording phase is not the place to be equalising sounds.
Don't record any effects, like reverb, mixed in with the sound. If you need to (perhaps it's part of a performance, or singer particularly likes the reverb sound) then record the effect on a different track. You are in no place to decide on the correct amount of effect until you have all the sounds done later in the mix.

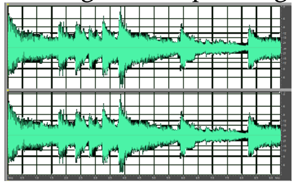
Always watch the Level meters, they tell you a lot about what is being recorded (including the presence of background noise). Concentrate on full levels 'without peaking'. Listen intently for background noises while you are recording, and then later while you are re-listening. In a recording bad levels and background noises are the Recording Engineer's problem, and responsibility, regardless of who caused them.

Take #1 *The first attempt at Recording will usually be a bit rough and untidy, but it will sound more 'Live' than subsequent recordings.*
Take #2 *and* ***Take #3*** *are usually the best (they are still fresh, but the execution has improved).*
Take #4 *onwards and the performance will slowly get more precise and technically superior, but audibly less 'live-human' than earlier recordings.*
You decide! A CD for church members to hear over and over shouldn't have musician errors, and you should be able to hear every word in the songs clearly (to sing along to) and all the instruments should sound clear and present. This can really only be done by over-dubbing.

Distortion 99% of the time involves excess volume arriving at one of the audio devices in the chain. Distortion can happen even with the Gain right down, and no sign from the red Peak LED which means it is distorting *before* the mixer. Microphone capsules can distort if too much pressure (volume) is arriving at the capsule… back off or turn the mic to receive a glancing sound rather than a direct sound. Reduce all electrical levels along the recording chain until you find it. Oxide on old connectors that have been rarely used (or rarely moved) can cause distortion. Flattening batteries will cause the electronics to start failing, generating distorted signals or high-pitched noises.

SAVING YOUR RECORDING AS AUDIO FILES:

Digital Recordings use a Sampling frequency, and a Bit rate. We call this method Pulse-Code Modulation (**PCM**). Recording software displays the PCM it records on to the screen using the RAW format. This is simply the value of every sample, laid out side by side, exactly like it arrived at the interface.
This is the format that all Wave Recorders use, even when they are effecting or manipulating the sound.



Whenever you save your recording the software will convert it to any audio format you desire. When you reload an audio file from whatever format it was saved as, it will always convert back to the RAW format, because this is the easiest to display and to manipulate mathematically (for effect work).

Recording Software may let you save in **.raw** format or they may call it **.pcm**. but be aware that you must always remember the bit-rate, sampling frequency, and if it was stereo or mono because **.raw** files only save the pure recorded audio data and contain no other information (whenever you load a .pcm or .raw file it will ask you for the information it was recorded with, as it won't know).
Because RAW / PCM files don't contain the rest of the information to be a portable stand-alone wave file, we always prefer saving our recordings in a format that automatically adds the bit-rate, sampling frequency, and if it was stereo or mono information inside the audio file.
Professional formats use **iff (Interchange Format File)** to do this. IFF formats keep the PCM data exactly as it was recorded, but adds all relevant recording information plus lyrics, text notes etc. if you wish.
Apple developed their version of the IFF (called AIFF) and save as **.aif .** IBM and Microsoft developed their version of the IFF (called RIFF) and save as **.wav**

Over time software developers have attempted to apply mathematics to wave files to **compress** them and take up less space on the hard-drive. Some compressed formats are of decent quality though the majority of them inevitably throw away some of the audio detail to reduce size and the result becomes semi-professional or even amateur sounding.
There are plenty of compressed formats: .sam .mod .au .snd .mp3 .mp4 .mpga .flac .ogg .smp .vmf .ra .ram .vox .bwf .wma .dwd .gsm .ac3 .asf .sd .voc .svx
As mentioned earlier, whenever we open an audio file, the software will always convert it back to a RAW format. Even though a compressed file may now have been restored to a RAW file, whatever sound information was lost during the initial conversion into a compressed file is not coming back, and this is a wave of lower quality now (if you save that back to the compressed format again later, you throw a little more frequency content out).

APPENDIX: *THINGS THAT MIGHT HELP WHEN RECORDING SPECIFIC INSTRUMENTS AND SINGERS*

PIANO:

The task is to mic-up the piano so that the pianist can run their fingers up the keyboard from the lowest to the highest notes and you can hear all the notes at the same volume, with a nice ivory-key sound. The sound is generated by the soundboard (the large flat surface under the strings) which is stimulated by the vibrating strings stretched across them. The wooden body (shell)(resonant chamber) of the piano resonates and amplifies the sound coming off the soundboard, and the lid reflects the sound out (if it is open). Some technicians remove the lid to get a more modern (less resonant) piano sound.

Horizontal Grand Piano "*Classical*" Sound:
For an orchestral piano sound don't put the microphone(s) right inside the Piano; ideally stay out at the wooden curve, pointing in for a 'natural' classical sound, which requires the sound of the wood. The high frequencies (the freshness and clarity of the piano sound) are reflected horizontally by the lid. The higher the frequency the more directional and the microphone 'sweet spot' for the Horizontal Grand Piano is therefore quite narrow and you need to search for it (somewhere on the pianists' right near the shorter strings).
A single microphone 1 metre out from the wooden curve gives a truly warm classical sound, and mic'ing out a little like this avoids the peaky transients that a piano generates when keys are struck hard and avoids the inevitable mechanical noises from pedals and dampers. Unfortunately, at this distance the microphone will pick up other instruments around it (unless you screen it off).
The Piano is not a 'present, fresh' sounding instrument and so its harmonics are not as high as many other instruments.
If the Piano is too dynamic (the distance between soft and loud is too great) then use a Limiter to clip the peaks, rather than using a Compressor.

Horizontal Grand Piano "*Modern*" Sound:
Place the microphones inside the piano, closer to the strings, but don't position them too close or you will over-emphasise section of notes directly in front of the mic, and the neighbouring notes will not be heard as well when the pianist moves up and down.

The Upright Piano
We need to mic up close because it does not radiate sound in a helpful way.
The standard is to lift the lid and put a microphone over the bass strings, and another over the treble strings. Since it is the soundboard that you are trying to capture, some technicians remove the back of the piano to reveal the soundboard, and mic it up directly. You will hear less mechanical noise, especially on old piano's doing it this way.

ACOUSTIC GUITAR

Consider the best type and gauge of guitar string according to the kind of sound you are after. Ensure that the guitar's action (string height above the frets) is set correctly so that it plays without buzzing. Accurate tuning is paramount so check the tuning using an electronic tuner.
Acoustic guitars thrive in live (reflective) ambients where there is a short fresh sounding reverb. Try to position the acoustic guitarist so that the instrument is played close to some reflective surfaces. If everything is carpet or dampened environment… place a sheet of wood on the floor beneath the instrument or make a vertical screen to make a more reflective 'live' sound.
If the guitar is picked, or does ornamental or melodic work, use a fresh sounding microphone and aim at the point where the neck joins to the body.
Filter out some of the bass and mid frequencies of acoustic guitars that are strumming chords, or they just sound like a drone, and muddy the other sounds.
If the acoustic guitar is strumming chords, it may sound insignificant, and get a bit suffocated in the mix by the other instruments around it. A good solution is to record it twice (physically play it twice, by over-dubbing)). The musician will never play exactly the same and these subtle differences make a great stereo guitar sound (pan the two tracks L & R). You can further enhance the effect (if you wish) by using a different guitar the second time or using the same guitar once with open chords and the second time with a capo or barrè chords. A subtle chorus on the 2nd guitar track can enhance it further.

INSTRUMENT LOUD-SPEAKER CABINETS

Mic'ing close to the centre of a loudspeaker cone gives the brightest sound. Really close mic'ing is normally done using a dynamic mic. If you use a condenser mic up close then angle it so that it doesn't take the direct blast.
Moving the mic towards one edge of a loudspeaker cone produces a more mellow tone, further back and it will become warmer. Further back is less 'dynamic' and you can use a better quality condenser mic without hurting it.
Do you have Multiple Speaker Cones in the Cabinet? Mic back far enough so that the combination of all the cones can be heard, the sound will be fuller.

If your mic is up close, you can place a second mic further out, to capture the ambience. An ambience mic will always add a nice dimension to a Speaker Cabinet sound. Some place a PZM microphone on the floor in front of the Cabinet.
If you have a Stereo Effect pedal, then you can feed the left and right into two different Cabinets. Using two Speaker Cabinets with different models of microphone in different positions offers you a professional stereo track and you only played it once.

ELECTRIC GUITAR:

As well as putting a microphone in front of the Loudspeaker Cabinet it is good practice to record a clean D.I. signal from the Guitar (or Amp) on to a second track. Sometimes you arrive at the Mix-down phase and find that the effect that is on the guitar isn't as suitable as you originally thought. You simply play the extra (clean) D.I. guitar track you recorded through a software Guitar-Cabinet simulator for an awesome guitar sound. You can even send the D.I. signal back through a Guitar Effects pedal and real Loudspeaker Cabinet, place a microphone and re-record it.

Use new fresh strings and ensure the intonation is accurate. Ask the guitarist to turn around slowly to find the angle of least hum between the Guitar Pickups and the Loudspeaker Cabinet. Use distinctively different guitar sounds or even different guitars when recording two or more guitar parts so they stand out better in the mix because if you always use the same guitar sound it will quickly cause a confused drone sound.

ELECTRIC BASS:

New strings help to bring 'edge and detail' to the bass line (older strings sound dull in their attack).
Often the best Bass Guitar sound comes from a mixture of a D.I. signal and a microphone in front of the Loudspeaker Cabinet. As with the electric guitar, recording a clean D.I. bass track allows the use of software Bass-Speaker Simulators in the mix... *an excellent result!*

THE MODERN DRUM-KIT

Have some oil around in case of mechanical squeaks from the kick drum, high hat, or drum throne.
Once all the mics are in place, it's time to check the relative ***phase***. The kick drum, being the loudest drum with the lowest fundamental frequency, is the best phase reference because it will tend to spill into all the other mics. Ask the drummer to play a bit of everything on the kit. Raise the kick drum fader and leave it up, and then bring up each drum mic one at a time listening to the mix of each drum with the kick drum mic. Flip the polarity switch on the drum channel so that you hear the fullest sound. Now all drum mics are *'in-phase'*.
Noise-Gates and Expanders are great on drum-kits, but they should never be used when recording... do it later because if it cuts out a bit of the sound while recording (which is inevitable) you can never get it back.

KICK (BASS) DRUM:

Many recording studios remove the front skin to dampen the boom which sounds nice 'live' but not-so-nice in recordings. As a rule, we place a woollen blanket, pillow or sandbag inside the kick-drum resting lightly against the batter-head. A more permanent method is a strip of foam glued to the wooden shell where the shell and the skin meet. The foam presses lightly against the skin and will dampen skin vibrations without changing the tone of the instrument.
You cannot use ordinary microphones on a kick-drum, the mic must have a good bass frequency response. If you want to place the microphone up close, or inside the shell of the drum, then the mic must be robust (eg a dynamic microphone). If you are further back, then you can use a large diaphragm condenser mic. In general, we aim our microphone off-centre to avoid the direct air blast and to pick up more of the upper harmonics as these have the higher frequency component that makes a good kick drum sound.

The Kick is a mixture of two sounds: a Boom underneath and a Thud (attack) on top. The thud sound can sound more like a click when hit very hard. If you have too much click, then turn the mic so it isn't facing directly at the beater skin. To get more attack you can replace the beater with a wooden head (if it isn't wood already). Adhesive stickers can be purchased and placed where the beater hits the skin for a brighter hit.

SNARE DRUM:
To dampen the inevitable resonance/ringing (which sounds nice 'live' but rarely sounds nice in recordings) we use screw-on dampers, circumference rings or we tape cotton wool or foam sponge to the top or bottom head. A snare has a flat sound in the centre, and gradually gets more resonant towards the rim.
The Snare is a mixture of two sounds: a boom from the drum itself, and a metallic sound from the vibrating wires. A good snare sound is rich in the dry wire-sound and weak in the basses. If you detach the wires you should hear a gunshot when hit hard. The drummer should have tuned it (or else get it tuned nicely).

HI-HAT CYMBAL:
Aim for the most natural sound possible. The sound is a combination of the hit from the stick and the sizzle of the inverted cymbals rubbing together. Small (miniature) condenser microphones make a great sound. If the drummer is a hard hitter, be sure to angle the microphone so it is picking up the least amount of the snare drum in the Hi-Hat track as possible (then a bass-cut filter during mixdown will reduce the snare spillage even further).

TOMS:
The Toms should be tuned so that when you pass around them you hear nice melodic steps, evenly spaced.

Unlike the snare drum a slight resonation in the toms is OK. If they 'sing' too much, then dampen the skin so that it decays faster by taping small pieces of foam sponge to the top or bottom head. Some recording technicians take off the bottom skins of the hanging Toms for studio work. This gives them a dull thud (instead of singing). By placing the mics inside each hanging Toms, you get greater separation, though there will be less attack (a softer hit). Experiment!

Be sure to use present, fresh sounding microphones. The Toms are a mixture of two sounds: a Boom underneath, and a Note on top.

You can try double mic'ing each individual tom with one mic on the top skin and the other mic placed underneath (you should phase-reverse one of them). In this way you will have a larger, fuller three-dimensional sounding tom.

CYMBALS:
Technicians often prefer two overhead microphones over the drum kit to capture the cymbals plus the toms and perhaps the hi-hat. This gives a spacious stereo image of the drum-kit. If you are aiming microphones directly at cymbals, then aim the mics towards the outer edge of each cymbal so that the sound doesn't waver when the cymbal starts swinging. Be sure to use a 'present' mic to get a nice fresh sound. A-B or X-Y microphone configurations are great.
The cymbals should have a clean metal sound (*if they ring too long then they can be muffled slightly by tightening the clamps and foam washers that hold them on to the stands or apply masking tape in radial strips from bell to rim*).

Although some drummers hate them, plastic beaded sticks can give a better 'ting' sound on a ride cymbal especially on a fast ride pattern and are often better suited for a recording.

VOCALISTS:

A professional voice recording relies on three things:

- good vocal technique from the singer.
- good microphone technique controlled by the engineer.
- good interpersonal skills from the engineer/producer to make the singer feel as confident and focused as possible.

In the case of the singer singing later (overdubbing) it is extremely important that you use every opportunity to communicate with them, encouraging and telling them what you are talking about in the Control Room (keep the singer 'connected' to the session). Eye-contact and smiles between the Control Room and the Vocalist are very important. The stress and expectation on a singer in a recording session is very high and psychology plays a big part in obtaining a good vocal track.

Print up multiple copies of the lyrics, including all repeated lines written out in full, so that everyone concerned has a copy and can discuss the treatment of specific parts of the song as the session progresses. Any lines you want to do again can be marked, and the best version of each line (if the voice-track was recorded more than once) can be noted correctly for the final mix.

If you are recording an album of songs then schedule the vocals to be recorded throughout the recording sessions, not just on the last day(s). Even experienced vocalists can usually only be at their best for three hours a day. You need a few sessions on separate days in case the singer is having an off-day. The voice will quickly betray you if you are not relaxed and free of concerns. Before each singing session arrange for somewhere private for the vocalist to warm up their voice, dong vocal exercises and practicing using rough mixes (in mp3 etc) of the intended songs.

Correct the lighting and room temperature to suit whatever the singer wants. Pamper them!! Unnecessary onlookers (and even necessary ones!) can be intimidating to singers, so ask them if they prefer to turn the microphone around so they can't see anyone in the Control Room. I often ask them if they would like us to turn off the lights in the Control Room, or in the Studio, to reduce problems with self-consciousness.
Give the singer a bottle of warm water (not iced), a glass, tissues and a rubbish bin. The engineer needs to take a moment to physically listen through the singers' headphones; never just trust the singer to say it's ok. It is crucial that the singer hears correctly.

Record every take (every practice) even if you know the singer is just warming up. The first takes are often the best (when people don't think we're recording, they give more stress-free natural performances).
If there are particularly high notes that are difficult for the singer, then get them to come back and sing late in the evening.
If there are particularly low notes that are difficult for the singer, then get them to come back and sing early in the morning.

We tend to use a non-reflective room or place Gobo's around the microphone to reduce room ambience when recording vocal tracks. You can even make a temporary, more contained, vocal booth using heavy drapes suspended on a frame; this gives great results.
Avoid placing the microphone in the exact centre of a room as the room standing-waves will all be in phase at that point and strange tones will add to the vocal track.

Many singers like to sing with one headphone off (tucked up against their head) so that they can hear their own voice with their free ear. The problem that we technicians face here is that the microphone picks up the voice and also the music coming out of the free headphone which often brings in phasing problems.

Solutions include…
Switch off the headphone not being worn (pan the mix 100% to the headphone in use).
Invert the phase of the headphones so the music will cancel itself.

For some singers (especially those with little studio experience) the best recording method is to sing the song in its entirety many times while you make a note of lines that were good. When they have sung all lines well at least a couple of times then stop there. Now you re-listen to all the 'takes' then copy and paste the best lines over onto one nice vocal track.
For some singers a better method is to sing the song in its entirety a few times until you get a decent track. Now start from the beginning, discussing what you have line by line, and punch-in where you wish to redo a line, phrase or word (depending on the ability of the engineer). When you are punching in it is important that the singer sings at least a line beforehand to keep the same energy at the point of re-recording.

Tube mics and Ribbon Mics are particularly suited for vocals as they impart a warmth to the sound.
An omni-directional mic may be suited for a single vocalist if the singer is in a room with a nice ambience.
A super-cardioid mic may be suited if the singer is singing in the control room, or other music is playing at the same time that can spill into the vocal mic.

Plosives:
Having problems with a singer 'popping' whenever they say 'P'? The microphone must be in line with the wind blasts coming from their mouth. Move the mic above or below their mouth-height and aim it at the mouth. A pop screen (and *not* a foam sock) eliminates pops. Remember that PZM condenser microphones won't pop even when extremely close as they don't have a proximity effect and you can't blow air directly against their diaphragm.

Sibilance:
Pointing the microphone slightly above or below the singer's mouth sometimes helps reduce the 'S'. You might also need to avoid a fresh present sounding microphone. A pop screen helps here too.

BACKING VOCALS (BV) / HARMONY SINGERS:
Harmony Singers are there to 'support' the Lead Singer, not compete, which is why Backing Vocals should never sound like a row of Soloists singing. We aim for a smooth 'group' sound that supports, but doesn't invade, the Lead Singer's space. To create the sound of a fat backing vocal section you can get them all to sing close to the microphones, then step back a couple of paces and sing the same thing again. This gives timing and timbre changes similar to large choir.
Another method is to get them to sing normally, and then turn their backs to the mic and sing again. One of the advantages of this distant singing means that the microphones pick up more room reverb which is often sounds far superior to any artificial reverb you will add later in the mix.

CHOIRS:
Consider stereo-mic'ing because MS or XY configurations are excellent for large group recording (then you can add some spot mics for any soloists etc). Mics suspended on string across balconies or light fittings is standard practice. Think of how wide the angle of pickup is for each microphone and, given that angle, how far back you will need to be to capture the block of singers you want to be in that particular microphone.
One problem I have always had with large groups of singers is capturing good articulation. You need to tell the Head of the Choir to work hard at this because many people all a little out of time result in a terrible confusion of consonants and vowels.

SPOKEN VOICE / NARRATOR:
The spoken voice needs to be fresh and present sounding. Background noise is always a risk when recording speech and it is good to turn up the mic sensitivity (gain) and listen for any noises that the microphone can pick up before starting. Large diaphragm Studio Condensers are often the most suitable because they capture the delicate high-frequency detail in the voice and also because they are quieter than Dynamic microphones (all microphones have background noise from their own electronics, but the Condenser has a particularly low noise floor).
Use a cloth (or foam rubber) on the table if the person speaking is sitting with many sheets of paper (get them to practice turning pages while you watch them). A table-covering also stops the table from becoming a sound reflector.
Get the speaker to surrender all jewellery and keys etc before going inside to ensure no jingling or rattling.
Most professional 'voice-over' or 'dubbing' studios (who record all the voices of the actors in a movie) use vocal booths that are practically anechoic (zero reverb and zero outside interference).

If the room is small and quite reflective then the recording might sound reverberant and boxed in. One solution is to make a box of absorbent foam that surrounds the microphone on 5 sides. Place the microphone well inside so that it can only hear the person speaking into the opening (and not the room reverberations). Be sure that the person doesn't have a reflective surface close behind them.

As with singers a Pop-Screen should always be used to catch any mouth pops ("P") and saliva sounds.

BRASS INSTRUMENTS

Instruments made of the Copper alloy called 'Brass' are rich in harmonics. They are typically Saxophone, Trumpet, Trombone, Tuba, and Horn.
Place a microphone close, slightly off to the side (off-axis) to avoid the wind blast. Aim at the bell (flare) of the instrument, this should give you a natural brass sound. Warm sounding large-diaphragm condenser microphones are ideal.

Trumpets and Trombones. The high harmonics tend to beam directly forward from the front of the bell (flare) while the lower frequencies radiate over a wider arc. Like all musical instruments you need to keep back a certain distance with your microphone to capture a proper balance of the various harmonics generated by the instrument. With powerful instruments like the Trumpet and Trombone the microphone is ideally positioned at least one metre.
The trumpet can sound quite mellow when played softly but will be bright when blown hard.
The trombone retains much of its rich and bright quality even during quieter passages.
A common recording position is to have the musician facing a wall or window whilst playing as this adds a full natural ambience to the sound. This method also provides great acoustic fold-back (wall reflection) for the musician to hear exactly what he/she is doing.
A pressure zone mic (PZM) taped to a wall at a suitable height does a good job of capturing a clean, detailed sound.

Saxophone

The Sax is a brass flare-shaped pipe and is capable of substantial volume. The bell (flare) is curved upwards to project the sound forwards. If we position a microphone directly in front of the bell, we will only pick up the high-frequency component of the instrument and a bright characterless sound is obtained. In the recording studio we adopt a similar approach to that of the woodwinds (Clarinets, Flutes) with the microphone aimed towards the middle of the instrument.
Positioning the mic between a half and one metre away will capture the direct sound from the body of the sax combined with the high harmonics from the bell.
Sax players are prone to movement as they play their instrument, and this can cause fluctuations in the volume of the recording. If this is the case, then you may need a clip-on mic attached to the flare.

The Flute

As with the reed instruments the horizontal Classical Flute radiates a full balanced sound off the sides of the tube. The sound at the open hole on the end of the tube has a very weak, thin and nasty sound. The best microphone position for a classical sound is around one metre away, aiming the mic at the centre of the keys. For a more modern breathy flute sound come closer and slightly nearer the mouth.

CREDITS

This material is offered freely to the Christian Churches; downloadable at Pietango.com

Text: *Original, by the Author, a Christian Recording Engineer.*
Images: *Designed by the Author. Some photographs were sourced from the Internet, then re-worked.*

Ever since the creation of the world, God's invisible attributes and divine nature have been evident. They are clearly understood through his workmanship, and all the wonderful things that he has made. Therefore, those who fail to believe and trust in him are without excuse, or defence. **Romans 1:20**

All of us have sinned and fallen short of God's glory, but God treats us much better than we deserve.
Because of Christ Jesus, he freely accepts us and sets us free from our sins. God sent Christ to be our sacrifice. Christ offered his life's blood, so that by faith in him we could come to God. **Romans 3:23**

If you declare with your mouth, "Jesus is lord," and believe in your heart that God raised him from the dead, you will be saved. For it is with your heart that you believe and are justified, and it is with your mouth that you profess your faith and are saved. **Romans 10:9**

For the Scripture (*Isaiah 28:16*) says, "Whoever believes in Him will not be disappointed." **Romans 10:11**

These things have been written so that you may believe that Jesus is the Christ, the son of God; and that by believing, and relying on him, you may have new life in his name. **John 20:31**