

Week 2 - 3:

Introduction to Online Radio - The Tools (Part 1)

Interview techniques, script-writing and basic recording will be covered in these lessons. Students will be shown how to insert sound bites and sound effects into a script, effective use of dialogue, drama etc. The students will also be shown how to use recording equipment, how to choose equipment and how to take care of their equipment.

Outline

1. Recording Equipment
 - choosing a recorder
 - choosing a microphone
 - different cables
 - connectors

Recording equipment

How do they work?

Basically there are two types of recorders: digital and analogue. Within these two categories, obviously, there are many different types. So what's the difference between the two.

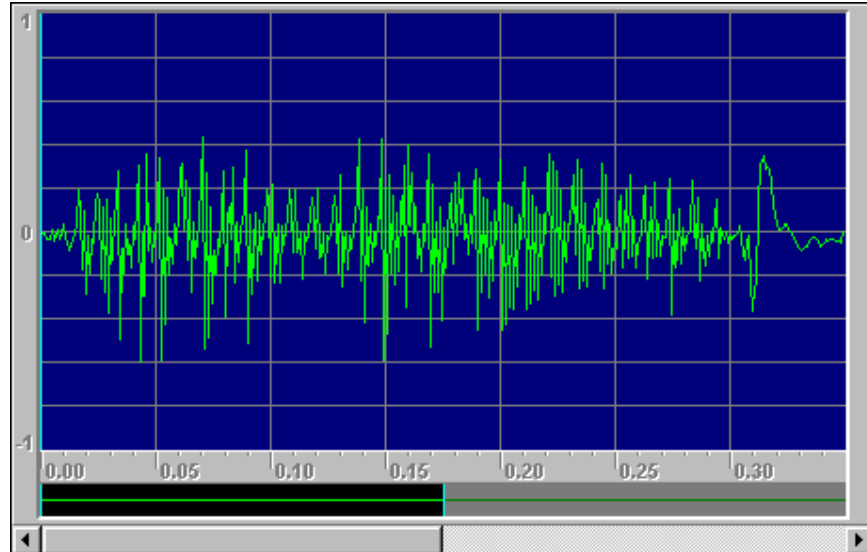
Recording started out with analogue, and with Thomas Edison – the guy who thought there was no future for broadcasting voice. He is credited with creating the first device for recording and playing back sounds in 1877. His approach used a very simple mechanism to store an analog wave mechanically. In Edison's original phonograph, a diaphragm directly controlled a needle, and the needle scratched an analog signal onto a tinfoil cylinder:

You spoke into Edison's device while rotating a cylinder, and the needle "recorded" what you said onto the tin. That is, as the diaphragm vibrated, so did the needle, and those vibrations impressed themselves onto the tin. To play the sound back, the needle moved over the groove scratched during recording. During playback, the vibrations pressed into the tin caused the needle to vibrate, causing the diaphragm to vibrate and play the sound.

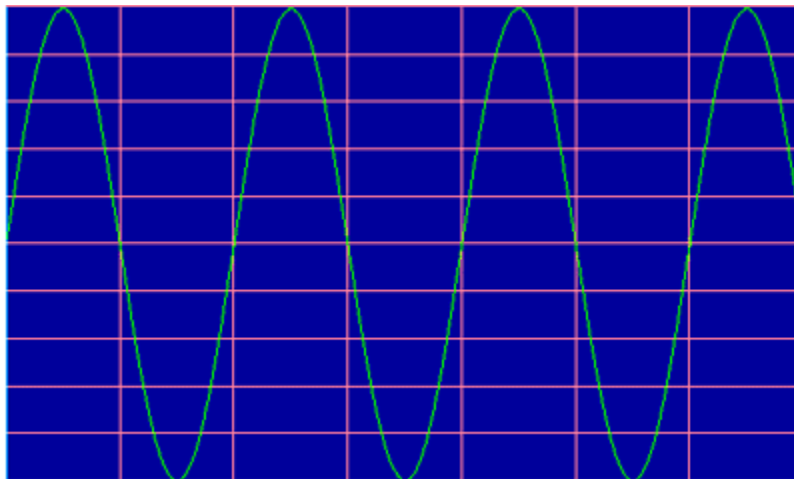
This system was improved by Emil Berliner in 1887 to produce the gramophone, which is also a purely mechanical device using a needle and diaphragm. The gramophone's major improvement was the use of flat records with a spiral groove, making mass production of the records easy. The modern phonograph works the same way, but the signals read by the needle are amplified electronically rather than directly vibrating a mechanical diaphragm.

So basically, analog is a PHYSICAL means of recording sound – sound is directly etched into a physical medium.

What is it that the needle in Edison's phonograph is scratching onto the tin cylinder? It is an analog wave representing the vibrations created by your voice. For example, here is a graph showing the analog wave created by saying the word "hello":



This waveform was recorded electronically rather than on tinfoil, but the principle is the same. What this graph is showing is, essentially, the position of the microphone's diaphragm (**Y axis**) over time (**X axis**). The vibrations are very quick -- the diaphragm is vibrating on the order of **1,000 oscillations per second**. This is the sort of wave scratched onto the tinfoil in Edison's device. Notice that the waveform for the word "hello" is fairly complex. A pure tone is simply a sine wave vibrating at a certain frequency, like this 500-hertz wave (500 hertz = 500 oscillations per second):



You can see that the storage and playback of an analog wave can be very simple -- scratching onto tin is certainly a direct and straightforward approach. The problem with the simple approach is that the **fidelity** is not very good. For example, when you use Edison's phonograph, there is a lot of scratchy noise stored with the intended signal, and the signal is distorted in several different ways. Also, if you play a phonograph repeatedly, eventually it will wear out -- when the needle passes over the groove it changes it slightly (and eventually erases it).

Note that this leads to 'generations' -- a first generation recording is the original recording. Second generation is one recorded from this recording. The second generation recording is of lower quality than the first.

What are the consequences of this for YOU as a journalist or broadcaster?

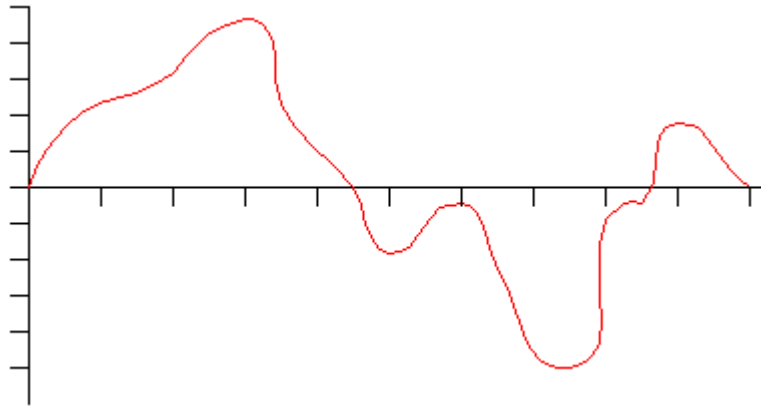
Digital Data

In a CD (and any other digital recording technology), the goal is to create a recording with very **high fidelity** (very high similarity between the original signal and the reproduced signal) and **perfect reproduction** (the recording sounds the same every single time you play it no matter how many times you play it).

To accomplish these two goals, digital recording converts the analog wave into a stream of numbers and records the numbers instead of the wave. The conversion is done by a device called an **analog-to-digital converter** (ADC). To play back the music, the stream of numbers is converted back to an analog wave by a **digital-to-analog converter** (DAC). The analog wave produced by the DAC is amplified and fed to the speakers to produce the sound.

The analog wave produced by the DAC will be the same every time, as long as the numbers are not corrupted. The analog wave produced by the DAC will also be very similar to the original analog wave if the analog-to-digital converter sampled at a high rate and produced accurate numbers.

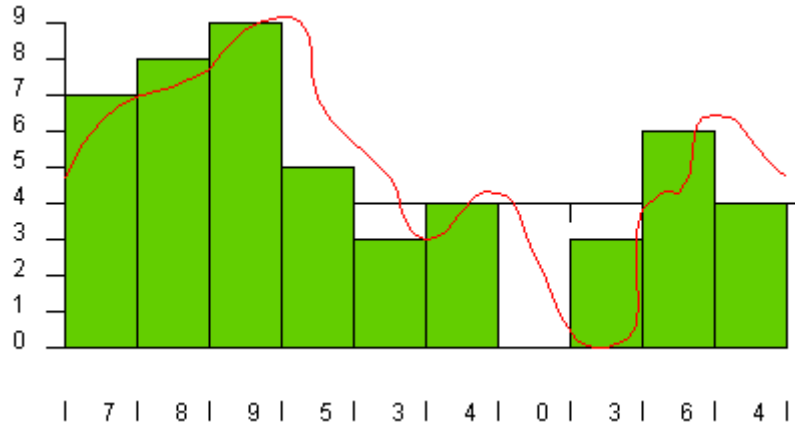
You can understand why CDs have such high fidelity if you understand the analog-to-digital conversion process better. Let's say you have a sound wave, and you wish to sample it with an ADC. Here is a typical wave (assume here that each tick on the horizontal axis represents one-thousandth of a second):



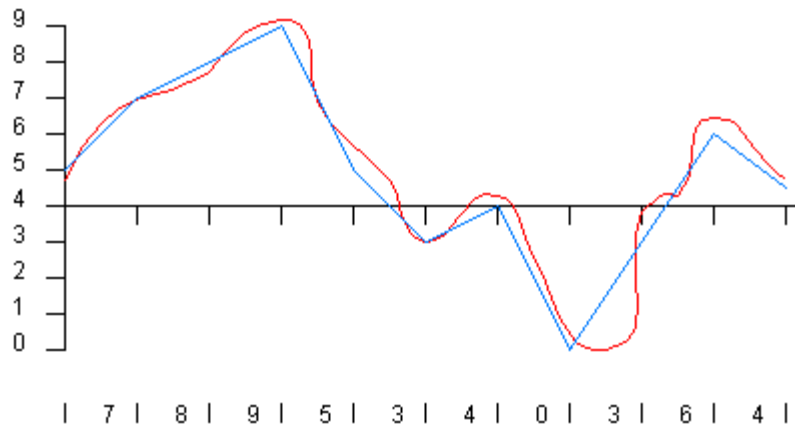
When you sample the wave with an analog-to-digital converter, you have control over two variables:

- The **sampling rate** - Controls how many samples are taken per second
- The **sampling precision** - Controls how many different gradations (quantization levels) are possible when taking the sample

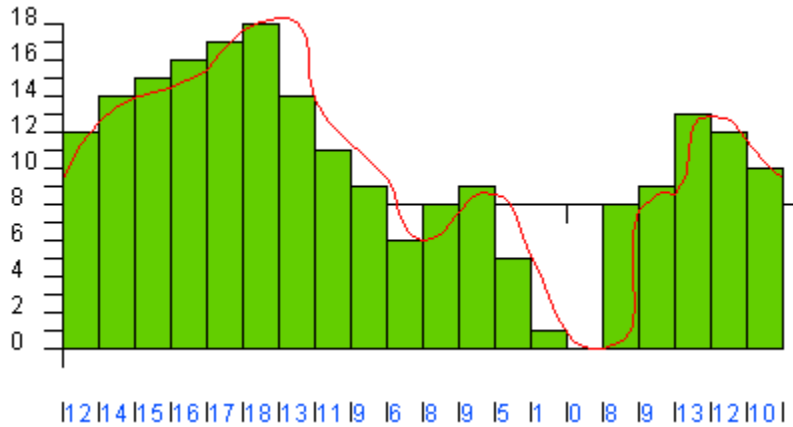
In the following figure, let's assume that the sampling rate is 1,000 per second and the precision is 10:



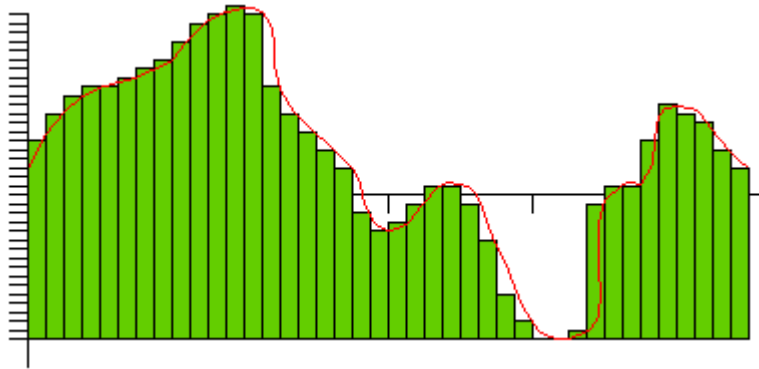
The green rectangles represent samples. Every one-thousandth of a second, the ADC looks at the wave and picks the closest number between 0 and 9. The number chosen is shown along the bottom of the figure. These numbers are a digital representation of the original wave. When the DAC recreates the wave from these numbers, you get the blue line shown in the following figure:



You can see that the blue line lost quite a bit of the detail originally found in the red line, and that means the fidelity of the reproduced wave is not very good. This is the **sampling error**. You reduce sampling error by increasing both the sampling rate and the precision. In the following figure, both the rate and the precision have been improved by a factor of 2 (20 gradations at a rate of 2,000 samples per second):



In the following figure, the rate and the precision have been doubled again (40 gradations at 4,000 samples per second):



You can see that as the rate and precision increase, the fidelity (the similarity between the original wave and the DAC's output) improves. In the case of CD sound, fidelity is an important goal, so the sampling rate is 44,100 samples per second and the number of gradations is 65,536. At this level, the output of the DAC so closely matches the original waveform that the sound is essentially "perfect" to most human ears.

Tape recorders

That's an analogue recorder.



What are you looking for in a tape recorder?

- a microphone socket (preferably XLR, but we can't have everything!)
- portability
- ease of use
- noise level indicator (peak indicator)

Other stuff that's nice:

- doesn't eat batteries too quickly
- half speed recording
- phone jack
- noise reduction for non-microphone inputs

Advantages

- it works, and can be fixed
- easy to buy tapes
- low maintenance
- easy to learn how to use

Digital recorders

Digital recorders are becoming more and more common. They come in a variety of flavours, such as DAT, minidisc, solid state PC card, CD, and hard disk. What distinguishes them from one another are the recording medium (what it records onto) and the recording format (the way the information is saved). Ideally, one should use digital recorders that capture sound in an uncompressed, PCM (pulse code modulation) format, but these recorders are REALLY expensive, so most journalists use minidiscs, which involve some compression. However, the sound quality is still high enough for broadcast.

At the heart of a digital recorder is the Analog-to-Digital converter (ADC). In digital systems, the analog audio signal must first be converted to digital form before it can be further processed. This entails sampling the signal in very short bursts and converting the value of the sample to a binary number representing the amplitude of the waveform at that instant. The output of the ADC is a series of digital "words", typically at a rate of 44,100 words per second (sample rate). Each word contains a certain number (typically 16) of bits (binary digits). The sample rate and bit depth are the most important factors determining the accuracy of the digital representation of the analog waveform.

The most typical digitization settings (so-called "CD quality") of a 44,100 Hz sample rate and a 16-bit word length render a recording that has a broad frequency response (about 21,000 Hz) and an impressive dynamic range of 96 dB. Many modern ADCs are capable of sampling at the rate of 96 kHz and a 24-bit resolution, which produces a highly accurate signal with negligible quantization noise. The main problem with this higher quality is the amount of memory it takes up, either on the disc or in your editing programme afterwards.

Minidisc recorders

Minidisc recorders are quite popular among reporters, particularly due to their small size and ease of use. MDs are usually shipped with small, inexpensive lavalier condenser microphones whose frequency response and dynamic range are typically rather poor – those are the small clip-on mics. (I suggest you bin them!) In addition, there is virtually no control over the incoming signal – this can be a real pain, but can be overcome with a reasonable microphone.

The most popular choice of MD are the portable consumer recorders, ie not those designed specifically for broadcast recording. They are inexpensive, tiny and easy to use, and while there is some data-compression in the recording process, the sound quality is still very good.

These machines do still have some down-sides: you can't find blank discs in the local convenience store; the microphone inputs are less-than-ideal minijacks; the built-in mic pre amps are often not strong enough to record loud, clean signals from dynamic microphones; and there's a risk of losing your audio if the unit loses power before a table-of-contents is written. There are techniques for recovering a lost table-of-contents, but it's not always simple (and personally, I don't know what they are!).

DAT recorders

DAT recorders use magnetic tape as a storage medium. Typical DAT tapes allow the storage of 120 minutes of uncompressed, high-quality mono recording at 48,000 Hz/16-bit. The migration of digital audio from DAT to a personal computer hard drive for analysis is a lossless process, provided the correct digital transfer interface is used. The DAT tape is as fragile as a regular analog cassette tape, which, again, raises the issue of long-term preservation. Most digital-born recordings stored on DAT tapes will eventually have to be migrated to either spinning disk or optical storage.

The main problems with DAT recorders are that they are VERY expensive, that it is practically impossible to buy tape for them, and that they are high maintenance.

VISIT www.transom.org and www.minidisc.org

Microphones

THIS IS YOUR MOST IMPORTANT PIECE OF EQUIPMENT.

There are lots of different kinds of microphone types: dynamic, condenser, ribbon, boundary, binaural, M-S and more. There are a myriad of pick-up patterns, different-sized diaphragms, variations in frequency response, sensitivity, self-noise, susceptibility to handling noise, wind or plosives. The possibilities can boggle the mind.

The types you are most likely to come across are:

- **Dynamic microphones** - A dynamic microphone takes advantage of electromagnet effects. When a magnet moves past a wire (or coil of wire), the magnet induces current to flow in the wire. In a dynamic microphone, the diaphragm moves either a magnet or a coil when sound waves hit the diaphragm, and the movement creates a small current.
- **Condenser microphones** - A condenser microphone is essentially a capacitor, with one plate of the capacitor moving in response to sound waves. The movement changes the capacitance of the capacitor, and these changes are amplified to create a measurable signal. Condenser microphones usually need a small battery to provide a voltage across the capacitor.
- **Lavalier microphones** – These are the small clip-on mics, they tend to be omnidirectional and extremely sensitive. The only reason they're included here is because they tend to come with MDs, but they are of little if any practical use.

As you can see, just about every technology imaginable has been harnessed to convert sound waves into electrical signals. The one thing they all have in common is the **diaphragm**, which

collects the sound waves and creates movement in whatever technology is being used to create the signal.

Any decent music store or catalog will have a large selection of mics, but most of them will be designed for fairly specific purposes or for recording of musical instruments. Some of these mics can also be useful for collecting sound in the field, or recording voice tracks in the studio, but conducting interviews and tracking narration have some special requirements.

The most basic distinctions we are concerned with are between Dynamic vs. Condenser, Directional vs. Omnidirectional, and Large Diaphragm vs. Small Diaphragm. None of these choices is universally superior, each has strong and weak points.

Dynamic Mics	Condenser Mics
Need no external power. Durable. Handle high volumes well.	Require Phantom Power Fragile. Louder output.
Common Examples: Shure SM-58, SM-57, SM-7 Electrovoice RE-50, RE-20, 635A Sennheiser 421	Common Examples: AKG 414, C-3000, C-1000 Neumann U-87, KM-series modular Sennheiser K6-series modular

Condenser mics are often a little more expensive, usually have a louder, more detailed output, but one should not overlook the benefits of dynamic mics, especially in the field. Dynamics are MUCH more forgiving of rough treatment, and do not require external power. Condenser mics break more easily if dropped, and require phantom power to operate, which must come from the recorder (draining batteries faster) a mixer, a preamp, or a separate power source. And even in the studio as an announce mic, Large Diaphragm dynamics such as the Electrovoice RE-20, Shure SM-7 and Sennheiser 421 produce very high-quality results. In the field, the overall durability and lack of phantom-power issues make dynamic mics very attractive.

There is often confusion between phantom power and preamplification. Although many preamps provide phantom power, they are separate power streams. All microphones require a pre-amp stage to raise levels approximately 60dB, but condensers, and some other types of mics, also require a low-level current to charge the diaphragm of the mic, setting up the electrical differential between the diaphragm and the backplate of the mic. Most modern mics want to see 48volt phantom power.

Note that it is because the mics need pre-amplification that their sound signal is different from 'line-in' signals – less noisy and less loud. This means that if you're recording from anything OTHER than a mic you either need to use a 'line-in' option somehow, or to cut down the sound coming into your recorder. This is more of a problem with analog recorders than digital. (Don't worry if you don't understand this right now, you'll get it when you use it!).

It is good practice to only turn on phantom power after all mics and cables are securely plugged in, and to turn off the phantom power at least a minute before unplugging the mics.

Many condenser mics, especially those designed to be hand-held, can accept a battery to provide the phantom power. This eliminates the need for an external box if your recorder or mixer will not provide power, but also adds another battery that can drain at the least opportune moment.

Condenser mics almost always provide a louder output, reducing the often noisy preamplification stage. And condensers often give a brighter, more detailed sound.

Pick-up Lines

Either type of mic can have various pick-up patterns. Some mics can be switched to change patterns, or capsules can be screwed on and off, giving greater flexibility. Of course, these mics are usually more expensive than fixed-pattern mics, so if you're on a tight budget, you might want to choose a fixed-pattern mic that best suits your needs.

The most common directional pattern is called "cardioid" for its heart-shaped lobe of maximum sensitivity in front of the mic. Sounds to the sides and especially to the rear of the mic are largely rejected, or at least attenuated. Hypercardioids have an even smaller, more focused pattern, the most extreme example of which is the "shotgun" mic, which is meant to be used at greater distances. It's important to remember that even the most directional mics do not completely reject sound outside of the pick-up pattern. "Off axis" sounds are reduced in volume, but also often acquire a phasey or muffled character, sometimes making extraneous sounds seem rather odd. Sometimes, what's most effective for one purpose, such as reducing feedback on a noisy rock concert stage, is not as pleasing for another, such as an intimate commentary in a quiet room.

The next most common pick-up pattern is called "omnidirectional" for its ability to pick up sounds in all directions equally. Because of the mic's design, it has less of a focus on a particular direction, but also, off-axis sounds are picked up more accurately and naturally, lending a more realistic ambience. Some people mistakenly believe that omni mics will pick up close and distant sounds equally, making the background too loud compared to the primary source. But these mics must still obey the laws of physics, and focus on the subject can be achieved simply by getting the mic in close.

Both human perception of audio and the physical behavior of sound waves are measured according to the inverse square law, which is to say that in ideal conditions, sound intensity drops 6 dB (half the volume) every time the distance is doubled, and conversely, the volume doubles as the distance is halved. This gets more complicated based on the sound source and the actual acoustical properties of the space, but it remains a good rule of thumb. What this means for mic technique is that even an omnidirectional microphone can focus on the close sound, because the level of distant sounds decreases logarithmically with distance. And conversely, in most cases, in order to get good sound on tape, one must get the microphone close to the source because its sound pressure levels are dropping rapidly as you lengthen your distance.

Omnidirectional mics have the added benefit of being less susceptible to handling noise, and more tolerant of wind and plosives (popped "p" sounds and the like) and less "boominess" when close to the source, although, of course, not completely free from these problems.

Directional Mics	Omnidirectional Mics
Heightened focus on centered subject. More rejection of ambience. Needs precise mic placement, off-axis sources often sound bad. More susceptible to handling noise, wind and plosives.	Natural, "you are there" sound. Picks up more of surrounding ambience. More forgiving on mic placement, off-axis sounds are more natural. Less susceptible to handling noise, wind and plosives.
Common Examples: Shure SM-58, AKG C-1000, Sennheiser ME64 capsule, shotgun mics	Common Examples: Electrovoice RE-50, 635A, Sennheiser ME62 capsule, most Lavalieres

Does Size Really Matter?

Among the varieties of microphones covered above, there is yet another distinction based on the size of the diaphragm. Dynamic or Condenser, Cardioid, Omni, Figure-8 or other patterns, each of these can employ a large or small diaphragm. In general, handheld microphones will have small diaphragms, which are more durable, less susceptible to handling noise and air currents, and well...gee...they're smaller.

Large diaphragm mics are best suited for studio work in controlled environments, where they can be placed on a stand, preferably with a shock mount and a pop-filter. The larger, more flexible diaphragm gives a richer, more detailed sound with a higher sensitivity, and so is generally preferable for primary announcing duties. Even in the field, if circumstances allow conducting an interview while the subject is sitting in one place, it's worth considering bringing a large diaphragm mic, a stand and a pop filter, particularly if the interviewee is used to microphones (an experienced public speaker, a musician) because the overall sound quality will be better than with a smaller hand-held mic. But these large-diaphragm mics are very susceptible to extraneous noise and vibrations, and despite frequent claims by manufacturers of having integral pop-filters, they always need additional foam or mesh pop-reduction devices.

Small diaphragm mics can usually handle louder sources, and counter-intuitively, actually can have a larger frequency range, especially in higher frequencies, due primarily to the smaller diaphragm having less weight.

For more info:

www.dpamicrophones.com - go to the Microphone University!

Jacks and wires

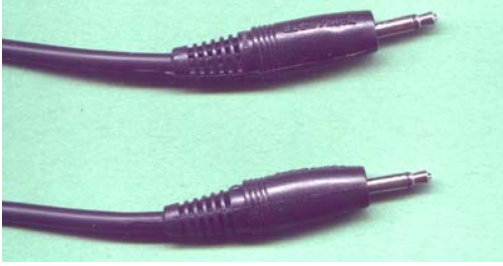
XLR

This is the best for recording purposes – designed for the transfer of sound from mics to machines. However, most MDs and recorders now have a mini-jack mic socket.



Mini-jack

These are the ones that everything seems to use currently. They are flimsy, so if you can get right-angled jacks, they seem to be a bit better. These come in stereo and mono. You'll see a small black band around the jack – if there's one, it's a mono jack, if two, it's stereo (the ones below are mono).



RCA jack

Often used when you're NOT using a mic. Also very useful if you want to tape off the TV, videos or a sound system. Usually you'll want a cable with two RCA sockets on one end, and a mini-jack on the other. Another useful piece of equipment is an RCA adaptor (also shown below) – two mini jacks plugging into an RCA socket.



RCA socket



RCA adaptor

Standard jack

As with the mini jacks, these come in stereo and mono – there's one of each in the picture below. And there are adaptors (very useful) that convert a mini jack into a standard jack (and vice-versa).



Taking care of your equipment

1. Hold your mics properly
2. Don't put any strain on flimsy sockets
3. Have SOMETHING that can hold them steady (if possible)
4. Ideally, don't pull the jacks in and out constantly. BUT if they are not being held steady, DO NOT travel with your cables connected.
5. Clean your tape heads regularly.
6. And read your manuals, so that if something goes wrong, you might be able to fix it.