CONNECTIONS & GOOD PRACTICE

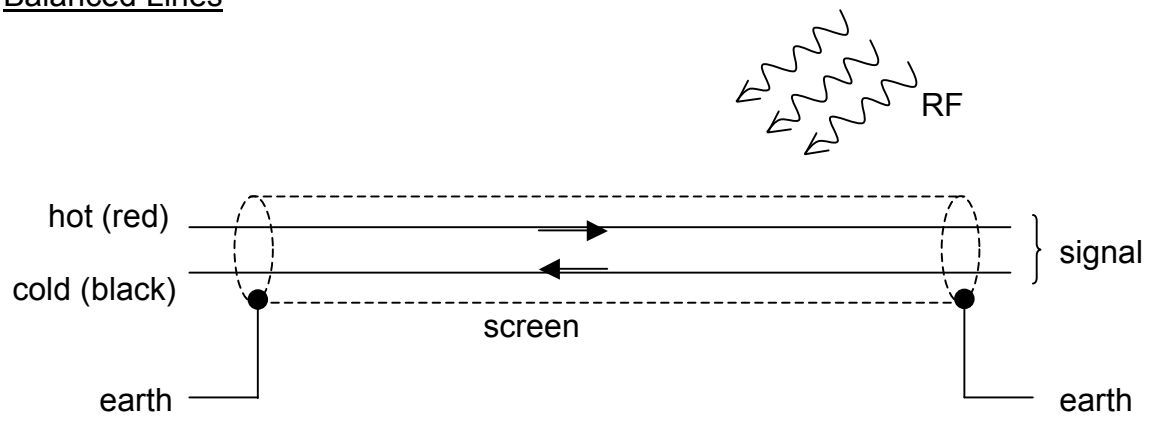
Interconnections

What we are talking about here is literally how one device connects to another, taking into consideration the input/output characteristics of the devices themselves, the cable and the connector.

Every connected device presents an impedance, a complex signal resistance, to the other (including the cable but it is often ignored as being the least significant component).
Studio equipment is designed to either present the same impedance to its connected partner (older 'vintage' style) or to expect a low impedance feeding a high impedance (modern style, not to present a 'load' to the incoming signal).
The point is to feed the signal between devices with the absolute minimum losses and alteration (distortion). Doing this incorrectly can lead to level loss, undesired overload distortion and/or frequency loss.

Of course the best quality cable is advisable for the best quality audio, 'oxygen free' is a popular modern choice for analogue audio (also the appropriate impedance co-axial for digital is worth mentioning, though here we are considering primarily analogue signal paths).

Balanced Lines



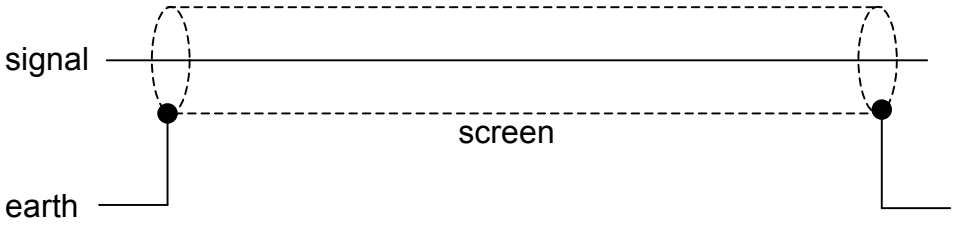
These connections and cables have two signal wires, hot (in phase) and cold (anti-phase or opposing polarity), and an earth (or chassis, 0 volts) connection.

When the cable picks up a spurious RF signal (radio frequency e.g. fluorescent light hum or a fridge's thermostat clicking on), all the conductors pick it up equally.

Now the hot and cold wires are effectively subtracted at the receiving end of the cable, cancelling the RF components, and the earth's component is shorted to 0 volts.

This system is used as a matter of course in most high end equipment, but is especially essential for low level signals (mic cables) or long cable runs.

Unbalanced Lines

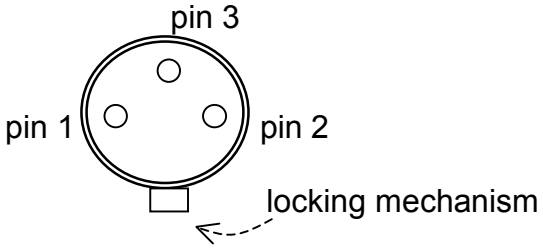


Consists of one signal wire and an earth connection.
These are fine for short runs of line-level signals (ref. mixers).

Note when connecting a balanced line to unbalanced, thus making the total interconnection unbalanced, the cold wire is usually connected to the earth (though some variation and experimentation may be necessary with some vintage gear to avoid hum and/or a degraded signal).

Connectors

XLR:



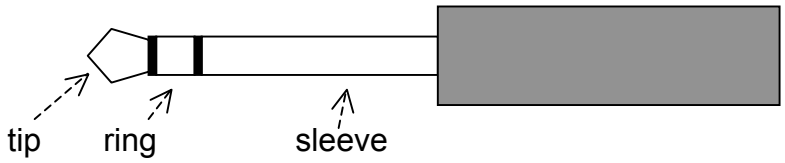
These connectors provide balanced interfacing between devices, particularly necessary for mics and the mic pres either on mixers or outboard units.
They naturally come in male (plug) and female (socket) varieties.
The most common pin allocation is:

pin 1	–	earth	
pin 2	–	hot	(red)
pin 3	–	cold	(black)

(though some vintage gear is wired with pins 2 and 3 interchanged, so care has to be taken).

JACK: Jacks, most often of the 1/4” diameter, are widely used for many applications in the studio.

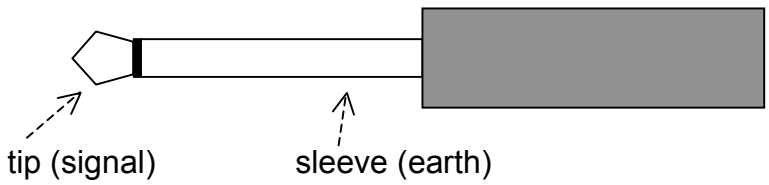
‘TRS’



These can be used in the following configurations:

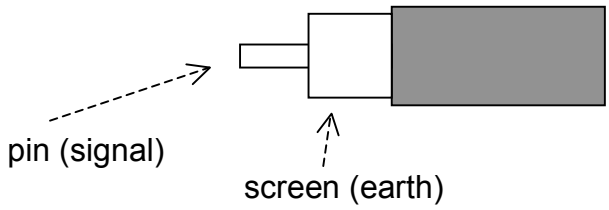
Balanced	tip	–	hot
	ring	–	cold
	sleeve	–	earth
Stereo	tip	–	left channel
	ring	–	right channel
	sleeve	–	common earth
Insert	tip	–	send
	ring	–	return
	sleeve	–	common earth

‘TS’



TS jacks get used for unbalanced line-level connections (interestingly if plugged into a TRS socket, these make the correct unbalancing connections), and are also the standard for electric guitars or basses.

PHONO:



Phonos tend to be used for unbalanced line-level connections where space and cost are significant issues, whilst not compromising quality.
Also used widely in the hi-fi industry.

MULTICORE CONNECTORS:

These come in a variety of sizes and configurations, where space and convenience dictate. A common connector is the ‘D sub’ type, so called because of its cross-sectional shape.

D.I. Boxes

The type of input circuitry traditionally used in studios is ‘Low Impedance’, however electric instrument pick-ups are traditionally ‘High Impedance’. If we try to connect these directly there is a mis-match, the low impedance tends to suck the energy from the pick-up and compromises the sound transferred.

So we use a Direct Inject Box (or D.I.) to:
[a] match the impedance
[b] balance the line
then connect it to a mic input.

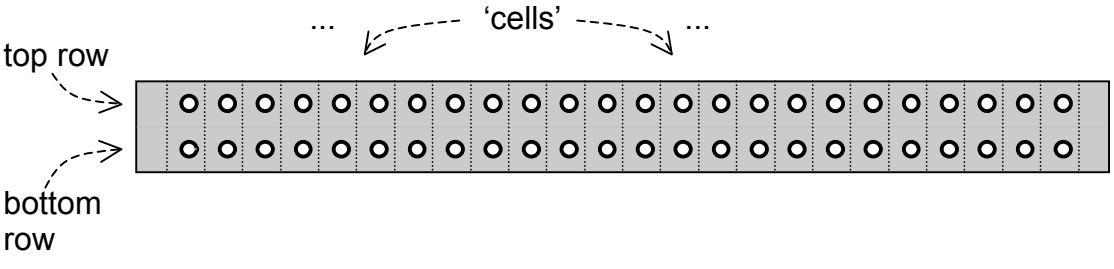
The original designs used a simple transformer to do this (passive units), but it is also possible to use balanced amplification electronics (active units).

Patchbays

When dealing with all the inputs/outputs of audio interfaces and mixers, hooking up hardware outboard equipment AND wanting to be able to configure the connections freely and creatively ... in professional environments it soon becomes attractive to have some sort of patchbay.

In commercial studios these are usually designed to be exhaustive i.e. all devices inputs/outputs/etc. appear on the patchbay allowing virtually any permutation of connections.. But also, in smaller setups, fewer ‘patch strips’ can be used to simply provide necessary common alternative connections.

In most serious studio setups where a patchbay is deemed necessary, a balanced jack arrangement is usually favoured where each ‘strip’ has two rows of sockets. The back of the panel can either be also jacks or solder connections.



To prevent a tangle of patch cables always having to be present for essential connections to be made, a system known as **normalisation** is used.
i.e. if no patch cable is plugged in, the top row jack is internally connected to the bottom jack of the ‘cell’; but if a patch cable is inserted in the bottom row jack (only), that internal connection is broken.
This configuration is the most useful, known as **semi-normalisation**.
These ‘cells’ can also be rigged to be **isolated**, so that no internal connection exists.

So when designing the layout of the patchbay, it can be decided where to use semi-normalised cells (e.g. equipment outputs to mixer line inputs, or insert sends to insert returns) or isolated cells (e.g. outboard gear inputs and outputs).

The type of jacks used can be of the 1/4” variety or, as is used on the higher end mixing desks, the smaller bantam (or ‘TT’) jacks.

Good Practice

Monitoring

Ideally monitor systems need to be as accurate as possible, in order to reveal all the information contained in the recording, thus allowing the recordist/engineer to make wise decisions concerning mic'ing/balancing/processing/etc.
This is true of the amps and speakers (as well as other associated hardware) employed, as well as the acoustic nature of the room in which they are installed. It's worth noting that the best gear choices perform (and can be appreciated) best in a reasonably accurate listening environment.

In practice compromises have to be made depending on the budgets available to set up these systems, not many of us can afford high-end super-accurate monitors and a professional acoustically designed space to put them.
But great results can be obtained with informed and sensible choices, in addition to (and most importantly!) the recordist being thoroughly familiar with what the available system is capable of revealing. A very useful method to become intimate with a system is to use well-recorded (and admired) commercially produced recordings as a reference.

Priorities in the Recording Process

So we already have lots of options to consider and with which to make (hopefully) good choices, but perhaps it would be a good idea to have some guidelines to help with those choices.
Some useful information would be how experienced professionals view the relative importance of the available tools.

Assuming we are aiming to make the highest quality recording possible, there are of course the technical considerations (level setting, noise, the best use of digital resolution, etc.). But also there are so many options and opinions (if you talk to anyone with some experience) that, as a novice, it can feel daunting and confusing that you are handling all the aspects correctly ... AND you've got to listen as well!

So some sort of guidelines can be helpful:

HIGH PRIORITY

1] **The Source of the Sound**

2] **Microphone Choice and Placement**

3] **Processing**

LOW PRIORITY

What this means is if the source of the sound doesn't sound good in the room itself, then no amount of expensive microphones and/or corrective processing is going to make the recording the best possible. Similarly no processing will compensate for the wrong mic choice or technique if, again, we are aiming for the best possible.

Of course every studio engineer will tell you they have to apply corrective techniques at some point, but if you end up relying on this the chances are that you'll also be asking yourself why your recordings don't sound as good as you think they ought to.

The source of the sound includes (starting right at the beginning):

- the musician (a seasoned player can SOUND very different from a beginner, quite apart from what they play)
- the quality and condition of the instrument (e.g. new strings for guitars usually sound better)
- the acoustic environment (does the nature of the room suit, preferably enhance, the sound being recorded?)

Once the source sounds good (hopefully inspiring!), then the real skill of the recording engineer is to choose and place the microphone(s) to CAPTURE that good sound.
If a 'direct' signal (no mic) is being used, then the interfacing takes prominence.

Now the decision to process can be taken wisely.
If a natural sound is required and the equipment being used of sufficient quality (top engineers go out of their way to use their favourite kit) then little or no processing may be necessary.
But if, at the other end of the scale, the style of music calls for some radical abuse then you are in a strong position.

Note if things are not sounding right at any stage make sure the highest priority element is as good as it can be first and work downwards, this principle should keep you on the right track (though precisely how this is applied is between you and your ears!).