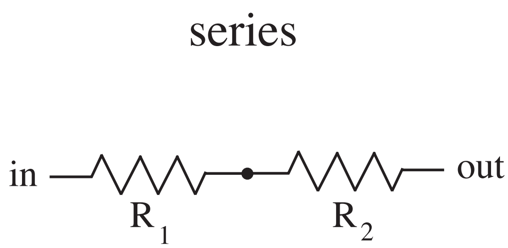
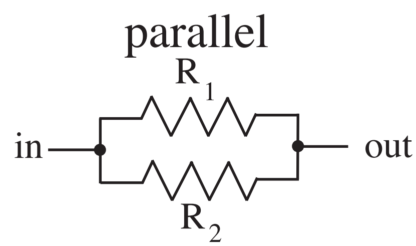
Electronic Circuits

We have already looked at one simple circuit, the voltage divider. In order to understand the operation of audio equipment, we need to look a little deeper in to electronic circuits. The arrangement of simple circuit elements determines the behavior of the complex audio circuits used for equalization, dynamics, and mixing. With a basic understanding of these circuits we may begin to understand what goes on inside those expensive “black boxes” that make up the recording studio.

The simplest kind of circuit that is of use is the voltage divider, which we have already encountered. This circuit is essentially two elements in series. With two-terminal circuit elements like resistors, capacitors and simple inductors, there are two ways to connect them together: end-to end (series) and both ends together (parallel).





In the series case, the resistance adds so that the overall resistance is the sum of the two individual resistances:

$$R_{total} = R_1 + R_2$$

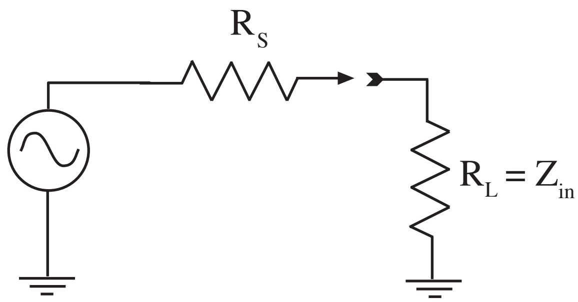
In the parallel case, the total impedance is more complicated to calculate:

$$R_{total} = \frac{R_1 R_2}{R_1 + R_2}$$

With series-connected elements, the total impedance is always larger than either element alone. In parallel, the circuit total impedance is always lower than either element alone. While we have shown these circuits for resistors, they apply to any impedance as long as we consider the frequency-dependent nature of the impedances as well. Using these basic circuit concepts, we can begin to examine the way devices connect together to make a functional studio.

Impedance matching

In order to transfer a signal from one device to another, their relative impedances may be optimized either to transfer voltage or power, the product of voltage and current. The absolute impedance levels will affect the amount of noise added to the signal, both from sources inherent in the circuits as well as from externally coupled or induced sources. The relative impedances will determine how much of the output signal is captured by the input of the next device in the signal chain. To maximize voltage transfer, the input impedance should be much higher than the output impedance, while to transfer power the impedances should be equal. This makes connecting devices together something of a "balancing" act.



Even very complicated circuits can be modeled as a simple circuit consisting of a voltage source and series resistance, known as a Thevenin equivalent circuit. In essence, this is what a device input would "see" when connected to the output. R_S represents the output impedance (series resistance, in this case), with the signal represented by the voltage source. The voltage seen by the input R_L (device input impedance appearing as the load resistance on the output circuit) depends on the ratio $R_L/(R_S+R_L)$. The tables below demonstrate the effect of varying the input impedance connected to a fixed output impedance as it relates to both voltage and power transfer.

<u>Voltage transfer</u>: $V_o = V_i(R_L/(R_L + R_S))$ [assuming V_i = 1 V]

R_S (ohms)	R_L (ohms)	V_O (volts)
100	1	0.0099
100	10	0.091
100	100	0.5
100	1,000	0.91
100	10,000	0.99

Power transfer: $P_o = V_o^2 / R_L = (V_i (R_L / (R_L + R_S)))^2 / R_L$

R_S(ohms)	R_L(ohms)	P_{out}(mW)	P_{total}(mW)	P_{out} / P_{total}(%)
100	1	0.1	9.9	1
100	10	0.81	9.1	8.9
100	100	2.5	5.0	50
100	1000	0.83	0.9	92

Below are some real device impedances, as they are specified and as they are actually measured:

Device type	Specified Z_{in} (ohms)	Real Z_{in} (ohms)	Specified Z_s (ohms)	Real Z_{out} (ohms)
microphone	-----	------	150	110
mic preamp	600	3000	600	110
Low Z line amp	600	2800	600	110
Hi Z line amp	5000	5000	100	47
power amp	600	2800	8	0.1

Low impedance transmission systems have the following advantages:

1. Reduced inherent thermal noise due to lower resistances. Thermal noise voltage (rms) can be calculated from the equation:

$$v_n = \sqrt{kTRB}$$

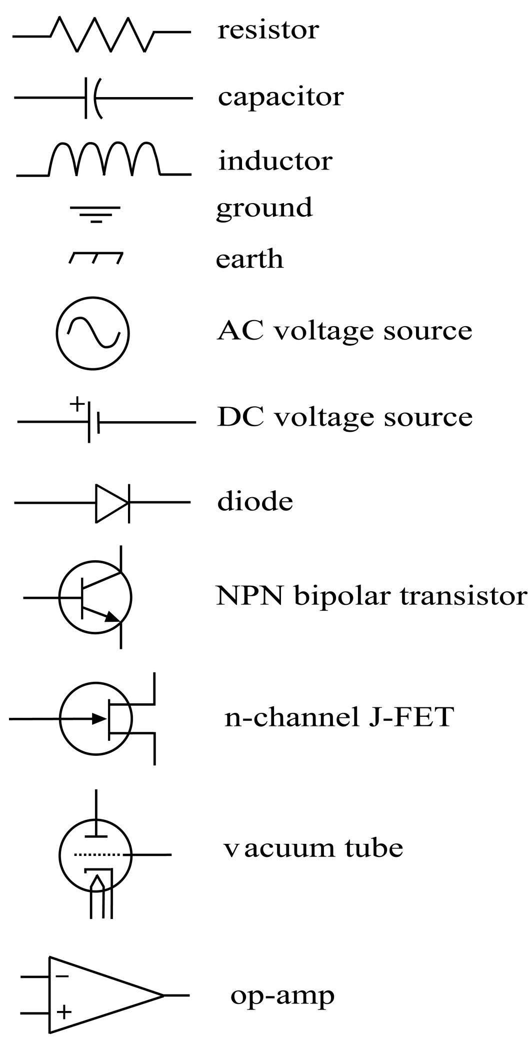
where k = Boltzmann's constant (1.38x 10^{-23} joule/°K), T = temperature (°K), R = resistance (Ω), and B is bandwidth ($f_{max} - f_{min}$) in Hz. At room temperature, a 1 kΩ circuit has a minimum of about 0.3 μv of noise while a 1 MΩ circuit generates about 9 μv of noise. While this may seem like a very small amount of noise, when multiplied by 60 dB (1000x) of gain in a microphone preamp, it becomes considerable.

2. Reduced susceptibility to electromagnetic coupling of external signals.

Balanced (differential) low impedance transmission systems have an additional advantage:

1. Reduced coupling of radiated noise due to rejection of common mode signals from electromagnetically coupled sources.

Electronic Circuit Symbols and Schematics



Electronic circuits used in audio devices are documented in drawings known as schematic diagrams or simply as schematics. These pictures are descriptions of the interconnection of the basic electronic circuit elements that comprise the audio device. Each element is depicted by a symbol, often including a number indicating the actual value of the component. While the schematic is usually intended to provide information to a service person should the device require repair, the schematic offers a quick way of studying the device: it shows the signal path and what the device is doing to the signal, if we know what to look for. We will discuss the basic symbols and what they describe. With a bit of knowledge about the function of the basic circuit elements, even complicated schematics can be understood in a block-function way.

The schematic is intended to convey very detailed information to a service person with no prior knowledge of the specific device. While a user can often interpret the schematic diagram, a simpler picture of the device operation is frequently provided: the block diagram. The block diagram shows the device as a series of interconnected functional blocks that depict the device in a less detailed manner. The block diagram is most useful to the user because it simplifies the information in a way that conveys the electronic processes as functions like gain, mixing, filtering, etc. With a little knowledge, however, it is possible to read a schematic and mentally convert it to a block diagram in order to fully understand the functions of a device. (Or you could read the operator’s manual...)

The way to convert a schematic into a more useful block diagram is to become familiar with the common circuit topologies encountered in audio devices. Fortunately, there are relatively few circuit types which are combined in analog audio devices: amplifiers, filters, mixers, and the occasional dynamic range processing block (which is really a special kind of amplifier.) Once we get used to recognizing these circuits, we can rapidly understand the signal path as it flows through the device. Since most modern audio devices are constructed from op-amp circuits, we can easily recognize the function of each block by knowing the way op-amp circuits function.

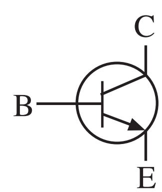
Active Electronic Elements

Active elements require outside power supply to function, unlike passive components. They use an input signal to modulate a supplied voltage (or current) to amplify the input signal. The main gain element in audio circuits is the transistor, which is able to amplify signals by using a small current to control a larger current derived from a voltage provided by a power supply. Amplifiers can be made of separate transistors (discrete) or may be fabricated multi-transistor devices called integrated circuits (ICs). The standard audio IC is the operational amplifier, or op-amp. Using negative feedback (some of the output signal inverted and routed back to combine with the input) and tailoring the feedback component network, most audio circuits, including amplifiers, mixers, and filters, may be implemented using op-amps alone. While audio op-amps and other audio integrated circuits have been improved greatly in recent years, there are still very high performance circuits, notably microphone preamps, which benefit from discrete construction. The individual transistors and other components can be hand-selected to deliver the best possible performance.

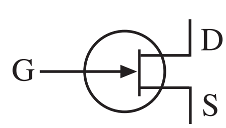
There is much interest recently in the vacuum tube as a gain element, although the tube largely fell out of favor when the transistor was developed. Basically, a filament in a vacuum is heated electrically and placed near a cathode, which allows electrons to be released by the cathode. Another element, the plate, is made positive relative to the cathode so it attracts electrons. By placing a grid in between, close to the cathode, a signal imposed on the grid controls the flow of electrons from the cathode to the plate, thereby producing amplification. Since tubes must operate at high voltages (up to several hundred volts), they are unsuitable for battery power and, hence, portable use. They produce a lot of heat, since the filament must be heated to incandescence in order to create current flow in the device. Nevertheless, tubes are making a comeback, in part due to the harmonic content of their distortion, which is perceived as more "musical" (or less harsh) than that generated by transistor circuits.

There is a type of transistor, called field-effect (or FET), which operates more like a vacuum tube than a regular transistor. They produce similar distortion spectra to vacuum tubes and are popular in power amplifiers and preamps. Some op-amps used in audio circuits use FET input stages.

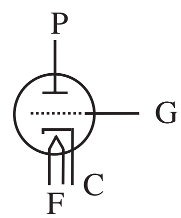
While we won't go deeply into circuit design, a basic understanding of active devices will allow the engineer to understand basic signal flow in schematic diagrams, often simplifying the job of understanding complicated devices like mixers. It should also make troubleshooting problems in equipment less mysterious.



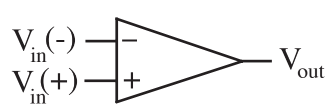
Bipolar Transistors: signal current into the base terminal (B) controls the current in the collector terminal (C), which is amplified by the current gain (β or h_{fe}) of the transistor. The output voltage depends on the resistors used in the circuit (Ohm's Law again!). If a resistor is placed in the collector lead (C), the circuit is an inverting voltage amplifier. If it is placed in the emitter lead (E), the circuit gives current gain but no voltage gain: it functions like an impedance converter, or buffer. The transistor can also be connected as a current gain device as it often appears in power supply circuits where the base current is regulated to control a larger collector current.



Field Effect Transistors: FETs are physically different in construction and operation from the bipolar transistor. Whereas bipolar transistors are basically current devices, FETs are voltage devices: while the base current into the bipolar transistor determines the collector current, the gate voltage to the FET determines the source to drain current. As we'll see, this more resembles the operation of a vacuum tube than a bipolar transistor. FETs have much higher input impedances than do bipolar transistors, so they are quite simple to use in audio applications. Since the FET acts much like a voltage-controlled resistance, it can be used as an analog switch as well as an amplifier.



Vacuum tubes: Before there were transistors, there were vacuum tubes. Thermionic devices like vacuum tubes operate at high temperatures in a vacuum. The filament is heated electrically until it glows red-hot. This heats a cathode until it sheds electrons that are attracted to the plate by a high positive voltage bias. A control grid is placed between the cathode and plate and the signal voltage applied to the grid controls the current flow in the plate electrode. This describes a simple triode; more complicated tubes with additional electrodes are also common. Due to the vacuum separating the electrodes, tubes have very high input impedances so they are easy to employ in audio circuits. The requirement for high voltages, however, make their circuitry depend on high voltage capacitors and can be hazardous to the casual experimenter. Although the relationship between temperature, impedance, and inherent noise would tend to make vacuum tubes noisy, they can be low noise devices if properly designed and constructed. Vacuum tube audio circuitry has enjoyed a renaissance, partly induced by the explosion of digital recording techniques which no longer introduces the tape compression and head bump effects of analog magnetic recorders which tend to make an audio signal sound "warmer" by altering the harmonic content of the reproduced signals.



Op-amps: Op-amps are integrated circuits, meaning many transistors and resistors are combined in a functional circuit where terminals are provided to connect external components and power. The output voltage is equal to the difference between the voltages at the V(+) and V(-) inputs multiplied by the device gain, which is very high. In real use, some of the output voltage is fed back into the negative input, stabilizing the device and allowing precise amplification and other operations to be implemented as a function of the external circuit. The input impedance is very high and the output impedance is very low. It is nearly a "perfect" amplifier. The external circuit can either be inverting or non-inverting, depending on the desired function. It can also be a filter and a summing amplifier, hence the popularity of the op-amp in audio devices. Another important use for the op-amp is the differential amplifier: it amplifies the signal difference between the inputs, but not any voltage common to both inputs (so-called common mode signal). Since induced hum and noise are usually common mode signals, they are rejected by the op-amp input. Such differential lines became known as balanced lines in audio terminology, because in addition to being differential, the impedances at both ends were equal, or balanced. This is no longer necessarily the case, but the name stuck.

So what gives it "that sound"?

A lot of what we perceive as the "character" of an audio system is due to subtle imperfections in the behavior of the electronic elements. The descriptions of the devices given so far are idealized and ignore the small deviations caused by effects like stray capacitances and inductances because their effects are relatively small. But when a large collection of these devices are combined in complicated circuitry, the imperfections all add together in unpredictable ways and can generate a sonic signature that gives each device its own special sound. Sometimes these effects are desirable, but they are always due to deviations from the ideal behavior of the components.

Passive devices are usually thought of as behaving exactly like their models: capacitors have only capacitance and not any resistance or inductance. In reality, there are small (sometimes not so small) inductances and capacitances associated with resistors; so when extreme conditions exist in terms of signal frequency, these reactances can change the effective value of the resistance from what we see if we measure the resistance with a DC resistance meter. And carbon-composition resistors will differ from metal film resistors and wire-wound resistors in their behavior and sound in some applications. Capacitors have a large but finite resistance in parallel with the capacitance and will eventually discharge even if completely disconnected from any external circuitry. Transformers will saturate when a signal is too large and saturates the metal core with magnetic flux. All of these effects may contribute to the sound of a circuit under extreme conditions.

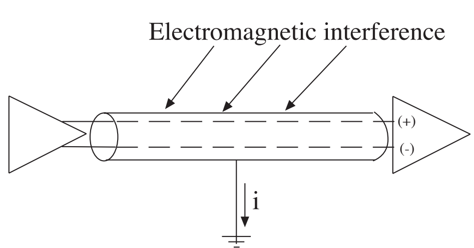
While passive elements contribute somewhat to the deviation from the ideal, active devices are prone to exhibit more significant limitations. Amplifiers have problems with signals that change very rapidly and have very wide dynamic ranges. There is a limit to how fast a device like an op-amp can change its output voltage, for example. This is called slew-rate limiting. Although the small signal bandwidth of the amplifier might be more than sufficient, the output stage cannot produce big instantaneous voltage swings and consequently the amplified signal cannot exactly follow the input. When you pass through several such stages in a mixer, you will begin to hear the result.

While we might be tempted to consider the shortcomings of analog circuits to be a problem, we have become quite used to the sound of transformers and vacuum tubes as applied to music recording since we heard popular music that way for many decades. Design engineers might like to have extremely linear and quiet circuits, while many recording engineers want the sound that tubes and inductors have conferred on music recording, which we have come to define as the "warm" sound of the "oldies". With the advent of digital recording and processing, we now have the ability to simulate the sound of the traditional analog recording system in computer-based recording programs. Among the most popular plug-ins for computer recording programs are analog simulation and digital models of classic tube/transformer audio compressors and equalizers.

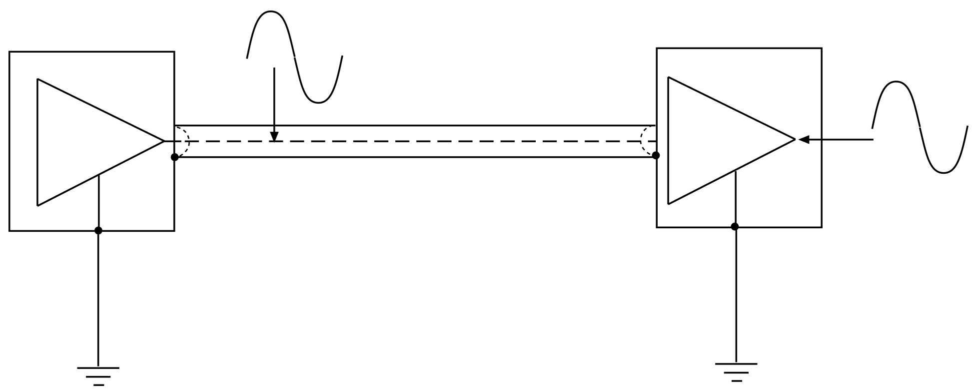
Device Interconnection

An important, if less than glamorous, aspect of audio signal handling is the connection of one device to another. Of course, a primary concern is the matching of signal levels and impedances between the devices, but there are additional considerations. Not only are wire and connectors of differing types involved, the grounding of a device is potentially affected by all the other equipment to which it is connected. Interconnection practices are important because all signals must be transmitted from device to device and noise added in the process can be very difficult to eliminate once it becomes part of the signal. By using properly shielded and grounded connections, a high quality audio signal can be preserved, even in very large systems. While this discussion relates to analog connections, there are similar concerns in digital systems that we will address later.

Shielding: Noise sources couple into wires by induction (like a transformer) and by capacitive coupling. These noise signals are produced by the 60 Hz power-lines, by radio transmitters, computers, electric motors, and many other electrical sources present in the environment. Shielding is a method of protecting audio signals from radiated noise by providing a low impedance pathway to ground for the unwanted electromagnetic signals. Audio signals are always transmitted on shielded cable (except for high power, high level, low impedance signals like power amplifier connections to speakers), thereby reducing or eliminating the problem of noise coupling.



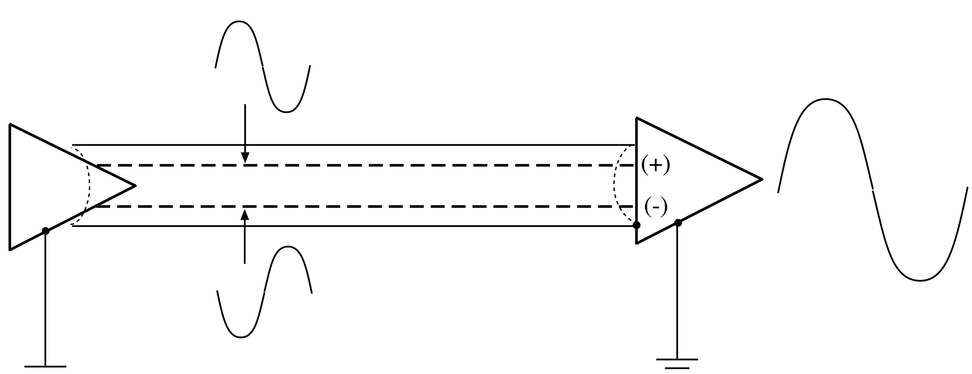
The shield is usually a braid of copper wire (or metal coated plastic) entirely surrounding the wire or wires that conduct the audio signal. The signal return path must be conducted by the shield in two-conductor cables (unbalanced circuits.) In three-conductor cables (balanced circuits), the signal is carried on two wires and the surrounding shield is separate. In order for shielding to provide protection against noise pickup, it must be connected to ground via a low impedance connection. Since noise induced from magnetic fields creates currents in the shield that must be carried to ground, any impedance in the shield connection to ground will result in a voltage drop (Ohm's Law) causing the shield to carry a non-zero voltage which may become part of the transmitted signal, especially in unbalanced circuits.



Unbalanced circuit

Unbalanced connections: When a signal is carried as a voltage on a single wire, the circuit is said to be unbalanced. The shield functions as a ground reference connection as well as a shield, which can lead to noise contamination if the connections are not solid at both ends of the cable. This circuit is used in consumer stereo equipment and many lower-cost audio devices: while this type of circuit is cheaper to produce, it is only suitable for short cable lengths and relatively simple setups in which the equipment is not grounded through the power line connection.

Because unbalanced circuits usually have high input impedances, the length of cable permitted is limited due to the capacitance of the cable. As the length increases, the cable capacitance shunts the input impedance to form a low-pass filter that reduces the high frequency response of the circuit.



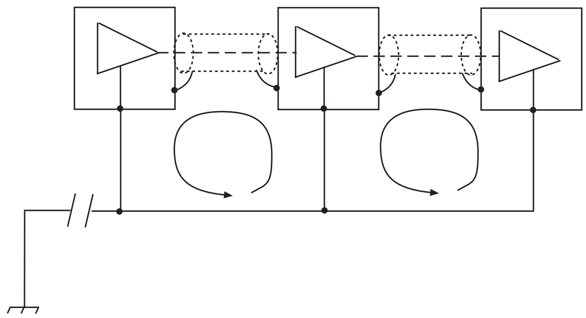
Balanced circuit

Balanced connections: Most professional audio gear includes balanced inputs and outputs. These connections send signals on two separate wires by splitting the signal into inverted and non-inverted signals and sending these on separate wires. The receiving input then subtracts the inverted signal from the non-inverted signal, producing a signal of twice the amplitude. This type of input (a differential amplifier) has the benefit of subtracting out any signal common to both conductors (called common-mode signals), which is typical of most radiated noise signals. Because of the noise rejection and the fact that balanced circuits are low impedance circuits, very long cables can be used without signal degradation.

Balanced amplifier inputs may be created from differential amplifiers like op-amps or with transformers. Each type has its strengths and weaknesses. Transformers have excellent common mode rejection (CMRR) and provide complete physical isolation for the two circuits, reducing potential ground loops. They have imperfect frequency response and the best quality transformers are expensive. Active electronic differential inputs have excellent frequency response and are inexpensive, but their CMRRs are not as good and there's no physical isolation. In situations where there is a lot of radiated noise (like venues with dimmers and lots of lighting equipment), transformers may be preferred, while in studios with good grounding and shielding, electronic amplifiers may be more desirable.

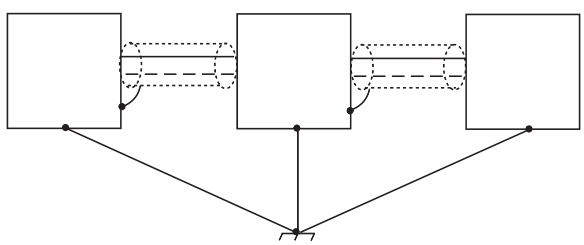
Grounding

In our analysis of circuits, the availability of a reliable voltage reference (0 volts) is assumed. Providing this reference may not to be as simple as it might seem. Since our audio devices are generally powered with AC electricity, we are tempted to make use of the power system's ground line as our audio ground reference. Power is delivered on a three-wire outlet: one wire carries the high voltage, one provides a neutral return, and another provides a safety connection to earth through the building wiring. (See the June 1995 issue of the J. Audio Eng. Soc. for an in-depth discussion of power and grounding.) Normally, current flows in the "hot" line and out the neutral. Both the neutral and ground are connected to a bus bar in the distribution box, which is connected to earth. The difference is that the service current flows in the neutral but not in the ground, which should only conduct current in the case of a major fault in equipment or a break in the neutral connection: the ground is simply seen as a safety connection and small noise currents present on the ground circuit are ignored. But these small currents can generate a voltage drop in the wiring, especially if the lines run hundreds of feet from the service entrance and through multiple sub-panels. We cannot assume the safety ground will always be an acceptable reference voltage, so to minimize noise coupling in a complex circuit like a recording studio special steps need to be taken to provide the necessary grounding.



Ground loops in unbalanced interconnect

The ground loop: Ideally, all devices should be connected to a single, common reference potential known as ground. Since the earth functions as a huge sink for electrons without it affecting its potential (voltage), a copper rod driven into the ground is often used as a ground reference for electronic signals. The voltage of the rod is known as "earth" for obvious reasons. A thick wire is then used to bring this reference voltage into the building and distribute it to the various circuits, where it is then referred to as "ground". The distinction is that the earth potential doesn't change, while the local ground potential is affected by voltage drops caused by any current that flows down the connecting wire to earth. If the wire has significant impedance, the currents cause a voltage drop that changes the voltage on the ground circuit.



Star ground with balanced interconnect

The ideal way of connecting devices to the ground reference is the so-called "star" ground, in which each device has a single direct connection to the reference point, forming a star-shaped distribution network. In this way, any currents flowing from a single device to ground affect only that device. In the real world, most equipment is grounded through the power line using a third wire separate from the actual power-handling wires. To assure user safety in the event of a massive device failure, the equipment case is attached to the ground connection. If we then use signal connections containing a ground connection and/or the devices are bolted into a metal rack, we have created multiple potential current paths between the devices. If any of the ground wires actually conducts a current, a voltage difference will be created (by Ohm' s Law) in the wire. The devices then see different reference voltages. This difference becomes part of the audio signal; the so-called ground loop has the effect of adding voltages caused by currents flowing in the ground connections to the signal. Since the currents are generated mainly by inductive leakage from transformers, the ground loop adds 60 Hz sine waves (and harmonics due to distortion) to the audio signal: the resulting hum is a ubiquitous contaminant of audio signals.

So how do we deal with potential ground loops in audio systems? We do so by carefully examining our studio setup in order to eliminate possible multiple ground connections. If we are using rack-mounted equipment, we have three such potential connections between devices: the metal rack-to-chassis connection, the power-line ground connection, and the signal wire ground connection. We must choose one of these and eliminate the others. The power line ground connection is the primary ground connection and is intended to provide electrical safety in case of a major electrical fault. We would ideally like to leave it connected! Unfortunately, it is also one of the easiest connections to break: a simple ground-lift adapter can be inserted between the grounded plug and the wall ground receptacle. It is very difficult to break the physical connection between the chassis and the metal rack unless the manufacturer thoughtfully provides a ground-lift switch or unless we go into the device and install our own ground-lift modification. (There are plastic tabs available that do this, but they must be installed on each piece of equipment in the rack.) Another easy connection to break is the one in the wire with which we connect our signal between devices. In order to decide which grounding connection we will leave and which we will eliminate, we must consider the actual physical setup with which we are dealing. The preferred way of dealing with potential ground loops and the safety concerns would be to use transformers for signal isolation (leaving the power line grounds connected), but they are expensive and may degrade the sound somewhat if not of the finest construction. In permanent installations, we may be able to run cabling with the shield connected at one end only, still providing shielding but interrupting the redundant ground connection.

<u>Connectors</u>: One of the most common problems in device interconnection is the issue of connector types. There are three main audio connector types: RCA/phono, phone plug, and XLR or Cannon.

Phono connectors, also known as RCA connectors, are the typical consumer audio and video connectors. They consist of a concentric plug, with the signal on the center conductor and the shield on the outer conductor. Phono connectors can only be used on unbalanced signals and are not the most reliable of connectors, either mechanically or electrically.

Phone plugs (in both 1/4" and 1/8" diameter) are commonly known as “guitar plugs”, since they are most commonly used on electric guitars and amplifiers. They are somewhat more reliable than phono plugs and can be used for both balanced and unbalanced circuits.

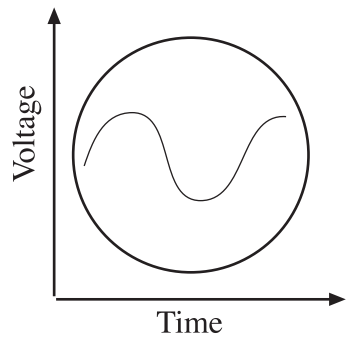
Two conductor types are known as T-S (tip-sleeve), with the signal connected to the tip and the shield connected to the sleeve.

Three conductor types are known as T-R-S (tip-ring-sleeve), with the hot (+) signal on the tip, the cold (-) signal on the ring and the shield connected to the sleeve. The smaller 1/8" diameter phone plugs are to be avoided in pro audio, since they are mechanically fragile and prone to poor connections. They are frequently used on miniature equipment like WalkMan™ systems.

The standard pro audio connector is the XLR type, which is found on low impedance microphones and most professional audio gear. These are the most expensive of the audio connectors, but also are the best in terms of reliability and mechanical strength. The audio type of XLR connector consists of three pins surrounded by a metal shell. Pin 1 is the ground pin, connected to the shield. Pin 2 carries the high (+) signal while pin 3 carries the cold (-) signal. Some older equipment was made with pins 2 and 3 reversed, so one must expressly check the manual to see what a specific piece of gear uses. To prevent ground loops, the XLR connectors allow disconnection (or “floating”) of the shield connection at one end of the cable. This should only be done for permanently installed equipment where the lifted shield can be properly identified to prevent a possible shock hazard. Note that the shield in the wire is connected to pin 1 of the connector, but generally not to the shell.

The Oscilloscope

One of the most useful of audio “tools” is the oscilloscope. It allows us to look at electronic signals. There is often information about signals that is more easily understood by visual examination than by simply listening. The oscilloscope displays signals as voltage versus time. We can select the time axis (or time base) with a knob that lets us determine the time it takes the electron beam to sweep from the left to the right of the cathode ray tube. The signal amplitude is displayed as vertical deflection of the beam and can be measured from a grid of lines printed on the display. The oscilloscope allows us to look at very fast signals, even if their frequency is too high to be audible.



Most oscilloscopes allow several input channels to be displayed at the same time. This allows us to compare two signals, input and output, for example. If we are interested in a stereo signal, it can be displayed in X-Y mode as well as by two separate beams. In X-Y mode one channel is fed to the vertical amplifier and the other is fed to the horizontal amplifier (instead of the time sweep). The result is a so-called vector display of the correlation of the signals. If both signals are of equal amplitude and phase, a straight line of slope 1 is displayed. As the two signals diverge, the display begins to “spread out”. In the case of two signals exactly 90° out of phase (and of the same frequency), the display is a circle, at 180° a straight line of slope -1. In fact, the relative frequencies of the two signals can be calculated from the display, at least for simple sinusoids. One of the more common uses of this system is in the physical alignment of tape recorder heads, which will be discussed later. It is a good idea for the student of audio to spend some time looking at the scope display while listening to music in order to get an appreciation of how audio signals “look and feel”. Of special interest is an examination of the ratio of the peak amplitudes to the average signal amplitude, known as the crest factor. Since short peaks can cause overloads of electronic circuits even when the average level indicated by a VU meter seems to be well within the “safe” range, an visual representation of the electronic signal variations is helpful.

Not only can the oscilloscope display several input signals simultaneously, but many can display the difference between two inputs. This allows the oscilloscope to duplicate the function of the differential (balanced) audio interconnect. While a common ground reference is required for display of unbalanced signals, the differential amplifier can display a difference signal without such a ground connection, provided the two signals under comparison share a common ground.