The Ins and Outs of Gozintas and Gozoutas

Think of a mixer as the heart of your system. Just about every device in your studio or sound reinforcement system is connected to the mixer at some point, many starting and ending there. Plugs, jacks, inputs, and outputs come in all different shapes and sizes, and it's important that all the devices that you interconnect play together nicely.

Three things of concern (or not – you'll find out more later) when interfacing analog electronics are impedance, operating level, and topology (balanced or unbalanced). Some of these tend to go hand in hand, but there's no reason for that other than convention.

In this article, we'll explore these issues and fill in the blanks.

Quick Summary – Balanced to Unbalanced Connections

In most systems, there'll be a mix of balanced and unbalanced inputs and outputs among the various pieces of equipment. This usually will not present problems when making connections if you understand how inputs and outputs are wired.

A *balanced* input or output consists of two signal terminals, called *high* (or *hot*) and *low* (or *cold*). It's the voltage between those two terminals that does the work.

There is almost always a shield over the signal-carrying conductors of the cable. The shield carries no signal current, and it's usually connected to the chassis (ground) of the device. Examples are an XLR (microphone) connector or TRS (tip-ring-sleeve) jack.

An *unbalanced* input or output also consists of two leads, but what makes it unbalanced is that one terminal of the connector, and hence one of those leads, is always at ground potential (*and impedance*). The other lead carries the signal, and the signal voltage is between the signal (high or hot) terminal and ground. Examples are TS (tip-sleeve or "mono") plugs, or RCA jacks.

When connecting a balanced output to an unbalanced input, be sure the high signal connections at each end are wired to corresponding plug terminals, and that the balanced low signal is wired to the ground terminal of the unbalanced input. In most cases, the balanced output ground lead will also be connected to the unbalanced input's ground through the cable shield.

When connecting an unbalanced output to a balanced input, be sure that the high terminals on the source and destination ends are connected together. The ground terminal on the unbalanced end should be connected to *both* the low signal and the ground lead from the balanced input.

If you have hum or RF interference problems, you may have created a ground loop between the two units. In some cases this can be fixed by disconnecting the connection between the cable shield and ground at one end of the cable or the other.

Sometimes you will have to make up special adapters to interconnect your equipment. For example, you may need a balanced XLR female connected to an unbalanced 1/4" TS phone plug. You can find some of them off-the-shelf, but there are so many possibilities that you may not find exactly what you need at the store. Learning to solder is an extremely valuable skill, but be careful not to burn your pickin' fingers.

When You're Hot, You're Pin 2

"Is Pin 2 Hot?". Sometimes it matters, sometimes it doesn't, but people sure worry about it a lot.

What's Hot?

A voltage is always measured between two terminals, one of which is the reference for the other. An audio signal constantly alternates from positive to negative and back again with respect to that reference terminal. We call the reference the "low" or "cold" terminal, and we call the one that's changing relative to it "hot". Since we're only looking for a difference in voltage between two terminals, it's arbitrary which one we designate as "hot".

So, why the fuss? Two reasons: In certain wiring configurations, one terminal is always at zero voltage - it can never be "hot". Also, we want to be able to maintain (or at least be cognizant of) the actual polarity of the signal throughout our chain.

When the diaphragm of the microphone picking up the sound we want to reproduce moves inward as a response to increased air pressure, we want the loudspeaker at the other end of the chain to move in the direction which creates a similar increase in air pressure at our ears. (that's outward)

Pin 2?

History lesson coming up! The Pin 2 we're talking about is Pin 2 of the three-pin XLR-type connector commonly used on audio equipment. Back when every connection was balanced, on an XLR and fed from a transformer, life was simple. Pin 1 was where you connected the cable shield. The signal was connected between pins 2 and 3.

Attention to details of design and wiring assured that when Pin 2 was positive with respect to Pin 3 at the input, the same would be true at the output. Any voltage measured between Pin 1 and either Pins 2 or 3 was purely incidental, due to electrical leakage or capacitive coupling between the transformer windings and ground – neither side of the audio signal was ever intentionally connected to ground.

But then a devilish deception came along - an XLR connector that was actually wired to an unbalanced input or output. With an unbalanced connection, the voltage reference point is the equipment ground. To this end, one of the two signal pins is connected, along with the cable shield, to Pin 1. This causes the signal voltage to appear between pins 2 and 3 just as it would on a balanced XLR connector, however there is also signal voltage appearing between one of those two pins and Pin 1.

Which signal pin? The one that's not tied to Pin 1, of course. Now, suddenly, we have a significance to one signal pin being "hot". Ampex usually gets the blame (or the credit) for initially establishing an industry standard of Pin 3 hot. Many of their units were supplied unbalanced, with an optional plug-in transformer available to make it balanced. It was their decision, based on a certain amount of logic, to make Pin 3 hot.

But you know what they say about standards - gotta love 'em because we have so many. Since this wasn't really a published standard, other companies, notably those in Europe, wired their unbalanced products with Pin 2 hot with respect to ground rather than Pin 3.

Enter Mr. Mike

The point where most signals enter the recording chain is the microphone. Fairly early on, American microphone manufacturers adopted a wiring convention: Positive pressure on the microphone diaphragm caused the voltage on the mic's Pin 2 to go positive with respect to Pin 3.

While there were some disagreement (or simply careless assembly) early on, eventually, microphone manufacturers worldwide settled on the XLR Pin 2 positive convention. Balanced or unbalanced, if a system is wired correctly, sound which pushes air against a mic diaphragm will cause the speakers to push air against the listener.

The Balanced and Unbalanced Mix

Where the Pin 2 Hot issue becomes significant today is when interconnecting balanced and unbalanced equipment. While we usually expect an XLR connector to be balanced (some old broadcast equipment notwithstanding), the 1/4" phone jack in common use on audio equipment today is available with either one or two signal-carrying contacts, so it can accommodate either balanced or unbalanced connections. The terminals on a ¼" plug are called the Tip, Ring (only on balanced plugs), and Sleeve for reasons which are pretty obvious when you look at it.

In an unbalanced configuration, the tip contact is hot and the sleeve acts as both the low signal terminal and shield connection. In a balanced configuration, the tip and ring of the jack carry the signal, with the sleeve being used to connect the shield.

When an unbalanced plug is inserted into a balanced jack, the jack's ring contact touches the plug's sleeve, which electrically connects the ring (usually the low signal) terminal of the jack to ground. This is the equivalent of tying Pin i of an XLR connector to one of the signal pins. Which signal pin? Another standard to the rescue!

AES14-1992 established today's wiring convention of Pin 2 Hot. Since the tip of a 1/4" jack is always hot, it's the equivalent to Pin 2 of the XLR, with the ring (if it's a TRS) being the equivalent of Pin 3. Most modern equipment adheres to this convention, making it straightforward to interconnect balanced and unbalanced equipment without losing the signal or inverting polarity.

Why the Fuss About Plus?

Most engineers, having given little thought to the issue of absolute audio polarity, don't expect that it makes any difference to the listener. Most of us have heard the hollow out-of phase sound which results when the leads on one loudspeaker are swapped, but few people worry about which wire on the speaker goes to which terminal on the amplifier as long as both speakers are connected the same.

Consider the sound of a kick drum as heard by the drummer and by the audience. They're each listening on opposite sides of the drum head. The audience hears the beater attack as air that is initially compressed (air pressure increasing momentarily), while the drummer hears the attack as air that is initially rarefied (decreasing air pressure). Our ears perceive these two different changes in air pressure differently. The same effect is true for anything that makes a sound, but the reason (and the effect) is most apparent for a simple vibrating membrane like a drum head.

It's easy to blame non-linear components for this phenomenon, but we can't easily replace the suspension mechanism of our eardrums or swap out op-amps in our brain. If we want our recording to sound to the listener the same as it sounded in the studio, we must preserve polarity, at least to the extent that we can.

Since you can't test every home playback system and rewire it if necessary, there will always be someone who will hear your recording with incorrect polarity, but as a responsible engineer, it doesn't hurt to get it right when you record it. It's not difficult to do, either. Check your mics and cables and fix any that are miswired. Check for polarity inversions in your equipment and for correct wiring when going between balanced and unbalanced connectors. Awareness is the biggest step. You don't have to worry about your Mackie – we took care to get it right, just make sure your cables are wired correctly.

Balanced Connections

It's well known that balanced connections are superior to unbalanced connections, all other things being equal. They provide better rejection of stray noise, (*common mode* rejection), they hum less, and they help prevent tooth decay.

The reason why a balanced connection is quieter is because the *input* is differential. The input stage amplifies voltage *difference* between the two input terminals. If the same voltage relative to ground is applied to both terminals simultaneously, as would be the case with a stray radio signal or hum from a power transformer, the difference between the two identical voltages is zero – the interference gets cancelled out.

Textbooks define a balanced connection as one with a source that has the same *impedance* between each of the differential inputs and ground. It turns out that this is a sufficient (and necessary) condition for a balanced input to reject noise that's picked up by both conductors connected to its input. So, it takes two to tango. You need both a balanced output and a differential input in order to have any of the advantages of a balanced connection.

Balancing Methods

The original balancing device for both outputs and inputs is a transformer. Many people believe this is still the best way to do the job, even though high quality, low distortion transformers are expensive. In the digital age, transformers are often put in a signal path, not just to provide balanced connections, but to add some (usually) euphoric distortion to make things sound "less digital."

Transformerless balanced outputs and inputs have become popular because of their lower cost and physical weight. With good design, they can perform very well.

Differential Outputs

For a long time, balanced outputs were, by nature (whether electronically or transformer balanced) were differential – during the portion of the AC cycle when Pin 2 goes positive with respect to ground, Pin 3 goes negative with respect to ground by exactly the same amount. The impedance to ground of both output terminals is identical, so we've fulfilled our requirement for a balanced source. In addition, the differential output can provide twice the maximum voltage of a similarly designed single-ended output – Twice as hot or twice the headroom.

Impedance Balanced Outputs

It's fairly common today for an output to be balanced, but not differential. A term that I use for this (I'm not sure if I invented it or not) is *impedance balanced*, to differentiate them from differential balanced outputs. With this configuration, only the tip of the connector carries the signal voltage. The ring is always at zero (ground) potential. Rather than being connected directly to ground (which would make it a Plain Jane unbalanced output), it's connected to ground through a resistor that's equal in value to the impedance of the hot output. The ring terminal, therefore, presents the same impedance *when looking back into the output* as the amplifier driving the tip (signal) terminal.

Most balanced inputs will think that they're being driven by a balanced source, and will happily reject common mode noise. Since there's no active circuitry driving the low side of the balanced output, you don't have to worry about shorting out anything out when connecting it to an unbalanced input.

Not all balanced inputs are designed alike. Certain input circuit topologies *require* a true differential source in order to achieve lowest distortion and maximum headroom. With all the cool gear out there, you might occasionally run across one of these inputs. It'll appear that your mixer output isn't "hot enough" to drive the unit. This isn't a fault of the mixer, it's just an incompatible design. The easiest fix it to put a transformer between the mixer's output and the unit's input.

Impedance – To Match or Not

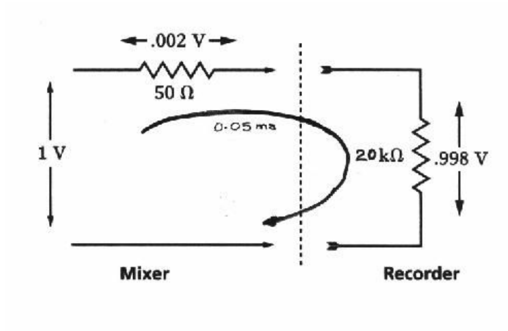
Impedance is the resistance to the passage of an alternating current. It's abbreviated with the letter Z, and the practical unit of measure is the ohm, just like DC resistance.

Ever notice, particularly in an older house, how the lights dim momentarily when the refrigerator or air conditioner switches on? If the wiring is old and thin enough, the lights stay a little dimmer all the time the refrigerator is running.

On a much smaller scale, the same thing happens when you interconnect two pieces of audio equipment. No, the house lights wont dim when you connect a recorder to your mixer's outputs, but if you were to measure the output level of the mixer with a very sensitive voltmeter before and after you connected the recorder, you'd see a very small voltage change.

Unmatched Impedances

A simple but effective model of an output is a voltage source with a resistor in series with it. This could represent the output of a mixer, a signal processor, a recorder, or even a hydroelectric power generation plant. An input can be modeled simply as a resistor connected between its input terminals. Undignified as it seems to represent your expensive vintage compressor by a single resistor, it allows us to complete our simple circuit model:



Most modern audio equipment makes liberal use of operational amplifiers the output of which, in the most common applications, appears as a voltage source with a series resistance of around 50 ohms.

The input impedance of an opamp circuit in most common audio applications is in the ballpark of about 20,000 ohms (20 kΩ).

Because we have a complete circuit here, the same current flows through both the 50 Ω source and the 20KΩ load. Applying Ohm's Law to our circuit, with 1 volt coming out of the mixer,

$$I = \frac{1V}{(50\Omega + 20000\Omega)}$$

we see that this current is about 0.00005 amperes (0.05 ma) - not much current, but we're just pushing electrons around, not loudspeaker voice coils, so the mixer doesn't have to do much work.

Now, juggling the terms of the Ohm's Law equation and solving for voltage (V=I x R), we see that our current causes a voltage of 0.998 volts to appear across our 20KΩ load, while the remaining .002 volts appears across our 50 Ω series source resistance. We've split the voltage between the source and the load, but managed to get nearly all of the voltage to go out of the mixer and into the recorder where we want it.

Older equipment worked by transferring power (voltage times current) from one unit to the next rather than voltage. 600 ohm inputs and outputs were common, and the outputs had enough power (real tubes, not tiny low power ICs) behind them to drive a 600 ohm load without breaking a sweat.

Modern studio gear is designed to work on the principle of maximum voltage transfer. This works fine unless you're driving very long cables (thousands of feet) and saves cost, weight, and power. For maximum voltage transfer, so you *don't* want to match the source and load impedances. You want source (output) impedances to be low and load (input) impedances to be high. A good rule of thumb is about a 50:1 ratio.

Since most equipment today is designed so that you'll have a much higher ratio than that, you can easily "mult" an output to several inputs without overloading the source. This is why you can double- or triple-bus subgroup outputs to two or three recorder inputs, or split a recorder output to two mixer input channels so you can set different equalization and effects in different parts of the song.

Matching Impedances

One instance where it's important to match, or nearly match source and load impedances is when connecting a loudspeaker to a power amplifier. Since a loudspeaker is a mechanical device and needs to do some hard work, (pushing air molecules around) it needs to receive high current as well as voltage from the amplifier.

Power is defined as voltage multiplied by current. Say you have an amplifier designed to work with an 8 ohm speaker. Juggle the numbers and you'll see what happens to the power when you vary the load impedance from above to below 8 ohms. You'll find that the speaker receives the maximum *power* from the amplifier when the source and load impedances are equal. We need less than a milliwatt of power to drive a recorder or a compressor, but we need several hundred watts to rattle the walls in the club and get the vocals loud enough to be heard over the drums.

The source impedance of a modern solid state power amplifier is typically quite a bit lower than the speaker for which it's rated, so in reality today's power amplifiers use "voltage transfer" to speakers rather that "power transfer," similar to our mixer-to-recorder example. The difference is that the impedances are much lower since we need to deliver more current to the speaker The lower the source impedance, for a given amount of current through the speaker, the less power will be lost across the amplifier's output impedance, the higher the efficiency, and the cooler the amp will run.

The reason an amplifier which may have an actual output impedance of a few tenths of an ohm is rated to work with an 8Ω speaker is because the amplifier can only supply a limited amount of *current* – that amount of current which, through the rated load impedance, will equal the amplifier's rated power. A lower impedance speaker will try to draw too much current from the amplifier. Doing that for too long will release the magic smoke from the amplifier chassis and it will cease to operate.

So, when we connect an 8 ohm speaker to a modern amplifier's 8 ohm output, we're not really matching impedances, we're making sure that the amplifier remains within it's rated parameters.

Operating Levels

There are two common operating line levels, -10 dBV and +4 dBu. There are historical reasons for those numbers.

The Telephone Company, where many of our audio principles originated, was connecting all of their equipment using the “maximum power transfer” principle, with all input and output impedances being 600Ω. They decided on a standard power level of 1 milliwatt and called it 0 dBm – no dB difference from a milliwatt.

Then along came someone with a voltmeter and said “Hey, at 0 dBm, there’s 0.775 volts coming out of this 600Ω output.” Since this guy was talking about volts, he got out his slide rule and said “. . hmmmm . . the ratio of .775 volts to 1 volt, log of that, times 20 is 2.2 dB.

$$dB(volts) = 20\log\frac{Voltage(measured)}{Voltage(reference)}$$

If dBm is milliwatts, then I guess we’ll have to call this voltage 2.2 dBV. Darn, I wanted a nice round number, too.”

To get that round number, he defined 0 dBu as the *voltage* required to pump 1 milliwatt of power into a 600Ω load. So far so good.

Then someone came along and invented the VU meter to indicate the program level being sent to a broadcast transmitter. The best meters they could make back then weren’t sensitive enough to move the pointer very far up scale with only 0.775 volts going into them, so they just cranked up the level until the meter reading looked .. well . . normal. It happens that this required boosting the 0 dBu signal level by 4 dB, and the standard operating level of +4 dBu was born.

This was back in the days when everything had tubes and transformers, and that combination was capable of putting out many more volts than the low power integrated circuits that are the heart of most of today’s equipment. Professional studio equipment has always been able to comply with the +4 dBu standard operating level with plenty of headroom, but that was expensive.

The musical instrument industry never really had to deal with professional studio levels in the early days, so they set their own standards to match lower cost circuitry which was happier working at lower signal levels. They picked 1 volt as their nice round number and made their standard operating level 10 dB below one volt, hence the “consumer” or *semi-pro* operating level of –10 dBV.

Today it’s pretty easy to find gear designed to work at +4 dBu, but -10 dBV designs are still common, so it’s important to understand the significance of the different levels. You need to take care of business if you’re trying to connect two pieces of equipment with different operating levels..

If you connect a +4 dBu recorder output to a -10 dBV console input, you'll run the risk of overdriving the input and introducing distortion. At best you'll wonder why you have to run your console faders or trim controls down at the bottom of their range to keep the console meters from hitting the pin. "My input is too hot" is the common expression of this interfacing problem.

Conversely, if you connect a -10 dBV mixer output to a +4 dBu powered speaker input, you'll get a fairly low out of the speaker unless you push the console to near its maximum output level. This is where you say "My mixer isn't hot enough."

There are converters which adapt between operating levels, and sometimes purchasing one is the best solution to matching up your equipment.

Interrelationships

There's no reason for impedance, level, and balanced/unbalanced characteristics to be related other than by manufacturing cost, but that's really important, so a number of de facto relationships have emerged.

As a rule, -10 dBV inputs and outputs are unbalanced. They used to have higher output impedances, but today that's rarely the case. As a rule, +4 dBu inputs and outputs tend to be balanced, though frequently not differential.

Operating levels used to be one of the things that distinguished pro and semi-pro gear. Today, that distinction is blurred, but the folklore of "-10 unbalanced, +4 balanced" remains, and so, still, does the requirement for most systems to be able to accommodate both operating levels.