About Analog Audio Levels

There are six basic kinds of analog audio levels found on most equipment:

- *Microphone level:* Around 50 or 60 dB less than line level. When you use a microphone, the level is very low, requiring a preamplifier to raise the signal to line level before it can be recorded or processed. Most audio mixers, cameras, and professional portable recording devices have built-in preamplifiers.
- *Instrument level:* Between microphone and line level, around –20 dBV or so. Guitars and keyboards usually output at instrument level.
- *Line level (consumer):* Consumer line level is output at –10 dBV.
- *Line level (professional):* Professional line level is output at +4 dBu (or dBm in older equipment).
- *Speaker level:* This signal varies considerably depending on the amplifier used, but it is very strong compared to the others because it is used to drive speakers.
- *Headphone level:* This signal is like speaker level, but much lower. The sole purpose of this signal is to drive stereo headphones.

About Units of Analog Audio Measurement

Professional audio equipment typically uses higher voltage levels than consumer equipment, and it also measures audio on a different scale. Keep the following points in mind when using consumer and professional audio equipment together:

- Professional analog devices measure audio using dBu (or dBm in older equipment). 0 dB on the audio meter is usually set to +4 dBu, which means optimal levels are 4 dB greater than 0 dBu (.775 V), or 1.23 V.

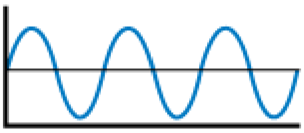
- Consumer audio equipment measures audio using dBV. The optimal recording level on a consumer device is –10 dBV, which means the levels are 10 dB less than 0 dBV (1 V), or 0.316 V.

Therefore, the difference between an optimal professional level (+4 dBu) and consumer level (–10 dBV) is not 14 dB, because they are using different signals. This is not necessarily a problem, but you need to be aware of these level differences when connecting consumer and professional audio equipment together.

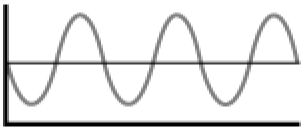
About Balanced Audio Signals

Audio cables can be either *balanced* or *unbalanced*, depending on their intended use. For long cable runs, especially when using relatively low microphone levels, a three-wire balanced audio circuit reduces noise. Balanced audio cables use the principle of phase cancelation to eliminate noise while maintaining the original audio signal.

A balanced audio cable sends the same audio signal on two wires, but inverts the *phase* of one signal by 180 degrees.

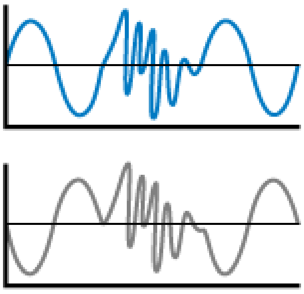


Original signal



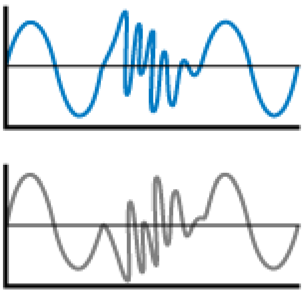
Inverted signal
(reverse phase)

When noise is introduced into the cable, it is introduced equally to both the original and the inverted signal.



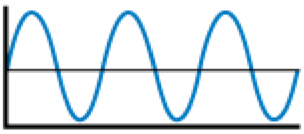
Noise on line
(affects both signals)

When the signal arrives at its destination, the inverted signal is put back in phase and both signals are combined. This puts the original and inverted signals back in phase, but it causes the noise signals on each line to be out of phase.



Inverted signal
(inverted again)

Now, both audio signals are in phase, but the noise is inverted, causing the noise to be canceled. At the same time, the original signal gets a little stronger because it is sent on two wires and combined. This helps compensate for the reduction in signal strength that occurs naturally on a long cable run.



Combined signals
(noise eliminated)

Any noise introduced into the cable across its long run is almost completely eliminated by this process.

Note: Unbalanced cables have no way of eliminating noise and are therefore not as robust for long-distance cable runs, microphone signals, and other professional uses.