Setting Up Audio Equipment

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Excerpt from Apple/ Final Cut Pro manual that explains various aspects of audio connections. A great resource!

The built-in audio interface in your computer can be acceptable for rough editing, but for a professional sound mix, an external audio monitoring system is essential.

This chapter covers the following:

Choosing External Audio Monitoring Components

This section describes the basic equipment necessary for a professional audio monitoring system and explains how to set up Final Cut Pro to work with this equipment.

An external audio monitoring system requires:

- An audio interface to connect audio devices to your computer
- An audio amplifier (one for each speaker—this is usually a single stereo amplifier)
- A pair of professional speakers (for stereo monitoring)
- Proper placement of speakers and acoustic treatment of your editing suite
- A pair of low-quality speakers, or a television monitor (for listening to your audio as it will sound to most viewers—optional)
- Headphones for critical listening to audio, especially for low-level noise
- An external audio mixer for routing and controlling levels of multiple audio channels (optional)

Choosing an Audio Interface

An audio interface provides high-quality audio input and output between your computer and audio equipment. This includes analog-to-digital (A-to-D) and digital-to-analog (D-to-A) converters, a stable digital audio clock, and input connectors that are compatible with your equipment, such as 1/4" phone (tip-ring-sleeve) and XLR connectors.

When you select an audio interface, make sure it has the following:

- Connectors that match your audio equipment, such as XLR, 1/4" TRS, RCA, or TOSLINK (optical connector)
- Support for audio signal formats that your audio equipment uses, such as AES/EBU, S/PDIF, or ADAT Lightpipe
- Enough audio inputs and outputs to connect your equipment
- Sample rate and bit depth at least as high as your audio equipment. For example, if you have an audio device with a sample rate of 96 kHz and 24 bits, your audio interface should at least match this.

Some PCI card audio interfaces include a breakout box. A breakout box allows you to extend the interface's audio connectors away from the back of the computer, such as on a desk or even in an equipment rack.

The most common kinds of audio interfaces are described below.

Built-in Audio

For basic mixing, you can connect the built-in audio output on your computer to a pair of external speakers. This gives you two output channels, which can be configured for dual mono or stereo playback.

DV (FireWire)

If your sequence uses a DV codec, you can output audio via the FireWire port on your computer. This allows you to use a DV deck, camcorder, or DV-to-analog converter as an audio interface. In this case, you connect your FireWire cable to your DV device, and then connect the audio outputs of the DV device to external speakers or a television monitor.

Third-Party Audio Interfaces

Third-party audio interfaces support more audio channels than your computer's built-in interface, and they often have professional connectors such as XLR or 1/4" phone (tip-ring-sleeve). Most video interfaces, such as the AJA Io, have both video and audio connectors, so in some situations your video interface can handle audio input and output.

Important: If you are considering purchasing an interface, make sure it supports Mac OS X Core Audio.

Choosing Speakers and an Amplifier for Monitoring

Professional audio engineers mix by listening, so they have to be able to trust the sound coming from their speakers. When you mix your audio, you need audio monitors that can handle the full range of audio intensities and frequencies. Ideally, your monitors will have a flat frequency response from 20 Hz to 20,000 Hz (or 20 kHz). This means that they neither attenuate nor amplify any frequencies. Flat frequency response is important for critical listening because the speakers themselves are not coloring the sound.

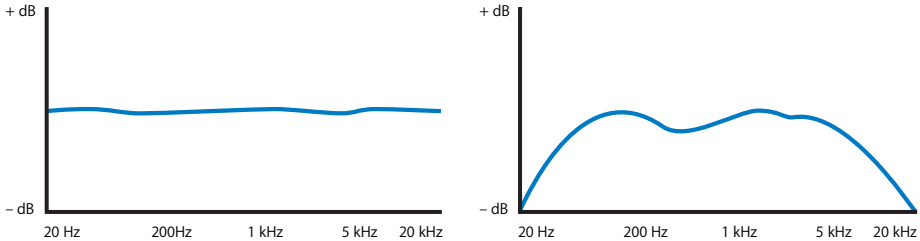
In addition to the speaker quality itself, additional factors affect your audio monitoring environment:

- Size and materials of the room
- Placement of the speakers within the room, such as distance from walls and angle of speakers
- Listener position between speakers

Frequency Response and Dynamic Range

Speaker quality varies greatly depending on their purpose as well as their price. For example, speakers in a boombox or television are designed to play audio that has already been mastered by a mixing engineer. Mastered audio such as audio CD, radio, television, and movie sound has a compressed dynamic range (meaning levels are fairly consistent and loud).

Speakers and amplifiers that are designed for mastered audio often intentionally emphasize certain frequencies, as is done with the bass enhancement feature found on many systems. This may make an audio CD sound better but it is not recommended for mixing production sound because you get a false impression of the audio signal. For example, if your speakers overemphasize frequencies around 2 kHz, you may compensate during mixing by reducing the intensity of audio around 2 kHz. If you then play your mix on a different set of speakers with a flat frequency response, the frequencies around 2 kHz will sound too muffled.



Amplifiers and Signal Levels

Audio speakers require signals with higher voltage than consumer and professional equipment can provide directly. Speakers require speaker level audio signals, while audio devices such as tape recorders and audio mixers usually provide line level signals. An audio amplifier boosts line level signals to speaker levels to properly drive speakers. Wide gauge speaker cables that can handle the higher electrical strength of speaker levels are used to connect the amplifier to speakers. For more information about audio signal levels, see "Microphone, Instrument, and Line Level" on page 30.

Self-Powered Versus Passive Speakers

Speakers powered by an external amplifier are called *passive speakers*. When you use separate amplifiers and passive speakers, complex factors such as impedance matching and cable length affect the overall frequency response and quality of your audio. Instead of using a separate amplifier and speakers, a simpler option is to use *self-powered speakers* (speakers with built-in amplifiers). These have become increasingly popular, especially for studio monitoring and video editing.

Self-powered speakers deliver more consistent performance because both components are designed to work together and are housed in a single enclosure. For video editing systems, self-powered speakers are a good, easy-to-use solution. Self-powered speakers accept line level inputs, so it's fairly easy to connect them to your audio interface.

Matching Your Mixing and Screening Environments

It's critical that you monitor your mix in an environment that closely matches the final viewing environment. A movie destined for a theater should be mixed on an audio system that matches the theater sound system. Likewise, a movie destined for DVD release for home viewing should be mixed on a system that resembles a home viewing environment.

Setting Up a Proper Audio Monitoring Environment

Room shape and material are just as important as the quality of the speakers themselves. Every surface in a room potentially reflects sound, and these reflections mix together with the sound originating from the speakers. Rooms with parallel walls can create *standing waves*, which are mostly low-frequency sound waves that reinforce and cancel each other as they bounce back and forth.

Standing waves cause some frequencies to be emphasized or attenuated more than others, depending on your listening position. When you mix in a room that creates standing waves, you may adjust certain frequencies more than necessary. However, you may not notice until you play back your audio in a different listening environment, in which those frequencies may sound overbearing or nonexistent.

Tip: A much cheaper alternative to building new walls is to mount angled pieces of material to the existing walls to eliminate parallel surfaces.

If the material in a room is very reflective, the room sounds "brighter" because high frequencies are easily reflected. Mounting absorbing material (such as acoustic foam) on the walls can reduce the brightness of a room. A "dead room" is one that has very little reflection (or *reverberation*). Try to cover any reflective surfaces in your monitoring environment.

Speaker Placement and Listening Position

Most video editing suites use nearfield monitors, which are speakers designed to be listened to at fairly close range. Speakers should be at least a foot or two away from any walls to prevent early reflections of sound which combine with and muddy up the original sound.

Position the speakers as far from your listening position as they are from each other (forming an equilateral triangle). For example, if the distance between the speakers is six feet, you should place yourself six feet from each speaker. The apparent width of the sound stage, or stereo image, increases as the distance between the speakers increases. However, if the two speakers get too far apart, sound information appearing in the center (between both speakers) starts to disappear.

Using Headphones

Many people use headphones as an alternative to critical monitoring speakers. Headphones can help you identify sounds that may not be obvious in your speakers. However, you shouldn't rely solely on headphones for mixing because many viewers don't listen to movies with headphones.

Audio Cables, Connectors, and Signal Levels

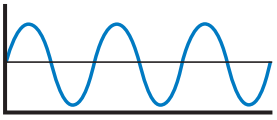
When connecting audio devices, you use cables with the appropriate connector on each end. Audio cables can be either balanced or unbalanced, depending on their intended use.

About Balanced Audio Signals

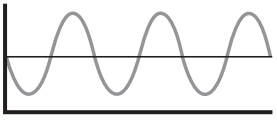
For long cable runs, especially when using relatively low microphone levels, a three-wire balanced audio circuit reduces noise. Balanced audio cables use the principle of phase cancellation to eliminate noise while maintaining the original audio signal. See "Phase" on page 38 for more information.

Here's how it works:

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

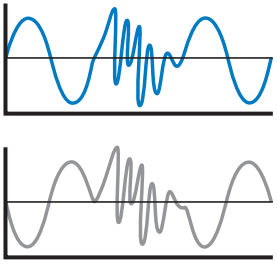


Original signal



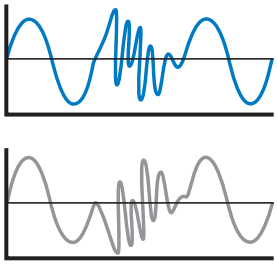
Inverted signal (reverse phase)

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



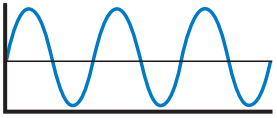
Noise on line
(affects both signals)

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 applications.

Microphone, Instrument, and Line Level

Audio equipment can output line level at –10 dBV (consumer level), +4 dBm/dBu (professional level), or microphone level, which is 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 is between microphone and line level, around –20 dBV or so. Guitars and keyboards usually output at instrument level.

Signal Differences Between Pro and Consumer Equipment

Professional audio equipment typically uses higher voltage levels than consumer equipment, and also measures audio on a different scale.

- 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.

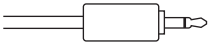
Therefore, the difference between an optimal professional level (+4 dBu) and consumer level (–10 dBV) is not 14 dB, because they are referencing different signals. This is not necessarily a problem, but you need to be aware of these level differences when connecting consumer and professional audio equipment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

Audio Connectors

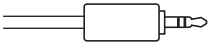
Different audio connectors are suited for different purposes. Audio connectors are often indicative of the kind of signal they transmit. However, there are enough exceptions that it's important to know what kind of audio signal you are connecting, in addition to the connector type. An important distinction is whether an audio connector carries a balanced or an unbalanced signal.

1/8" Mini Connectors

These are very small, unbalanced audio connectors. Many computers have 1/8" mini inputs and outputs at –10 dBV line level, and many portable audio devices such as CD players, Walkmans, and MP3 players use these connectors for headphone outputs. Portable MiniDisc and DAT recorders often use 1/8" mini connectors for connecting microphones.



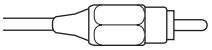
Mono miniplug connector



Stereo miniplug connector

RCA Connectors

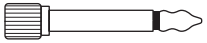
Most consumer equipment uses RCA connectors, which are unbalanced connectors that usually handle –10 dbV (consumer) line levels.



RCA connector

1/4" Tip-Ring (TR) Connectors

1/4" connectors with a tip and a ring are unbalanced connectors often used for musical instruments like electric guitars, keyboards, amplifiers, and so on.



1/4-inch Tip-Ring (TR) connector

1/4" Tip-Ring-Sleeve (TRS) Connectors

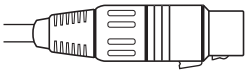
Professional equipment often uses 1/4" TRS (tip-ring-sleeve) audio connectors with +4 dBu line level. TRS connectors connect to three wires in an audio cable: hot, neutral, and ground, and usually carry a balanced audio signal. In some situations, the three wires may be used to send left and right (stereo) signals, making the signals unbalanced.

1/4-inch Tip-Ring Sleeve (TRS) connector

Note: Tip-ring and tip-ring-sleeve connectors (also called *phone* connectors) look almost identical. Some audio equipment (especially mixers) accept a TR connector in a TRS jack, but you should always check the equipment documentation to be sure. Remember that most 1/4" tip-ring connectors connect to –10 dBV line level equipment, while 1/4" tip-ring-sleeve connectors usually expect a +4 dBu line level.

XLR Connectors

These are the most common professional audio connectors. They almost always carry a balanced signal. Many cables use an XLR connector on one end and a 1/4" TRS connector on the other. The signal may be microphone level (when using a microphone) or +4 dBu/dBm (professional) line level.



XLR connector

Configuring External Audio Monitors

The following section describes how to connect external audio speakers to your editing system, how to select an audio interface for output, and how to make audio volume adjustments in Final Cut Pro and Mac OS X.

Connecting Speakers to Your Editing System

When you add audio speakers to your editing system, you need to make sure that the speakers are properly connected to your audio interface or built-in computer audio output, and that the interface is properly configured in Final Cut Pro.

To connect self-powered speakers to your computer:

- Connect the main left audio output of your audio interface to the left speaker, and connect the main right audio output of your audio interface to the right speaker.

  For more information about types of audio connectors and adapters, see "Audio Connectors" on page 30.

To choose an audio interface to monitor your audio:

1. Choose Final Cut Pro > Audio/Video Settings, then click the A/V Devices tab.

2 Choose an audio interface from the Audio pop-up menu. This device is used for playing audio from the Viewer, Canvas, and Timeline.

- *Default:* This option uses the output device you have selected in the Output tab of the Sound pane of Mac OS X System Preferences (or in the Audio MIDI Setup utility). If you want to control the audio output of Final Cut Pro from these locations, choose this option. This is the default setting for audio output.
- *Built-in Audio:* This is your computer's built-in audio interface.
- *FireWire DV:* If you connect a DV camcorder or deck to your computer, this option becomes the default setting.
- *Other available audio interfaces:* If you have a third-party video or audio interface installed, it appears in this list.

Depending on your hardware, additional options may be available to configure your audio interface.

3 Click Options, then make choices for the following:

- *Channels:* Choose the number of channels your audio interface is capable of outputting. You can assign sequence tracks to these channels by creating an audio output preset.
- *Bit Depth:* Choose a supported bit depth for the audio. 16-bit audio is compatible with most DV equipment. For high-resolution output and export, 24-bit audio is preferred.
- *Sample Rate (Hz):* Choose a standard sample rate from the pop-up menu. For professional audio applications, 48 kHz is typical, though 96 kHz is becoming more popular.

Note: If Final Cut Pro considers the selected interfaces in the Video and Audio pop-up menus to be paired together (or part of the same interface), the Audio pop-up menu is controlled by the Video pop-up menu. This means that changing the selected video interface automatically switches the audio to the same interface. This behavior is known as *audio follows video*. If the selected audio interface is different from the selected video interface, changing the video interface has no effect on your audio output.

4 Enable or disable the following alert message options:

- *Do not show External A/V Device Warning when device not found on launch:* Select this option to turn off the alert message when selected devices cannot be found during application startup.
- *Do not show warning when audio outputs are greater than Audio device channels:* When selected, this option disables the following message: "The selected external audio device does not support N outputs. Unsupported outputs will be ignored during playback on this machine." This message appears when you assign more audio output channels to a sequence than the currently selected audio interface in the Audio/Video Settings window can support. For more information about setting audio outputs, see "Working With Multiple Audio Output Channels" on page 61.

To enable "audio follows video" switching for video and audio outputs:

1 Choose View > Video Playback, then select a video output from the submenu.

2 Choose View > Audio Playback > Audio Follows Video, and make sure that the option is enabled (indicated by a checkmark).

With the Audio Follows Video option enabled, you can now select a video interface and the corresponding audio interface or device is selected automatically.

Tip: If you don't see your audio interface appear in the list of audio outputs, choose View > Refresh A/V Devices, then try to select your output again.

Setting Monitoring Levels and Muting System Sound Effects

When you mix your audio, it's important to monitor using a consistent volume setting. If a sound is too loud in the mix, you should adjust the level of the audio in Final Cut Pro, not the volume on the speakers themselves. Once you set up your audio monitoring levels, you should not need to adjust the overall volume of your audio very often.

If all of your audio is consistently too quiet or too loud, you should probably change the overall volume setting for your speakers, and then keep it at this new level. There are a few different places to adjust the volume, including the volume knob on the speakers themselves.

If you are using the built-in audio output of your computer, you can adjust its volume in the Sound pane of Mac OS X System Preferences or by using the volume control keys on the keyboard.

To adjust the built-in volume of your computer using the volume slider in the menu bar:

1. Open System Preferences by choosing Apple menu > System Preferences, then click Sound.
2. In the Sound pane of System Preferences, make sure the "Show volume in menu bar" checkbox is selected.

   When the checkbox is selected, a volume icon appears in the menu bar.
3. Adjust the volume in the menu bar.

   You can also adjust the volume in the Sound pane of System Preferences.

To mute all alert and Mac OS X user interface sound effects:

1. Choose Apple menu > System Preferences, then click Sound.
2. Click the Sound Effects button.
3. Deselect the "Play user interface sound effects" checkbox.
4. Deselect the "Play feedback when volume keys are pressed" checkbox.
5. Slide the Alert volume slider all the way to the left.

If you are using an audio interface other than the built-in audio, you can route the alert sound effects to the built-in speakers, but monitor Final Cut Pro audio from your audio interface.

To route Mac OS X alerts and sound effects through your computer's built-in speakers:

1. Choose Apple menu > System Preferences, then click Sound.
2. Click the Sound Effects button.
3. Choose "Built-in Audio: Internal speakers" from the "Play alerts and sound effects through" pop-up menu.

While monitoring the audio of your program, avoid changing the volume of your speakers unless it is absolutely necessary. A consistent monitoring level allows you to get used to the average loudness you're establishing for your mix, so that you can better judge how well the louder and softer sections of your mix are working together.

To adjust the volume of your speakers, try playing a signal that represents the average volume you want to monitor. Avoid setting speaker volume so high that it fatigues your ears or distorts in the speakers.

Some people use the 1 kHz tone of the Bars and Tone generator to set the volume of their speakers. However, you may find that the 1 kHz tone causes you to turn down your speakers down lower than you would for normal audio because the tone is so incessant and your ears are particularly sensitive to this frequency. Generally, 1 kHz tones are useful for setting levels from device to device when looking at meters, but not as helpful for setting average listening levels.