Pro Tools®

Sync & Surround Concepts

Version 8.0

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chapter 1

Synchronization Concepts

This section is designed to give you a basic understanding of synchronization as it relates to Pro Tools®.

For information on synchronizing Pro Tools software to other devices, refer to the Pro Tools Reference Guide.

If you are using a device that sends or receives time code (such as a video tape recorder or MIDI interface), refer that device's documentation for information on how it generates or receives time code and how to configure its options.

Synchronization Requirements

Synchronization requirements and support vary among different Pro Tools hardware systems. This guide notes whenever a feature requires a synchronization peripheral (such as a Digidesign SYNC HD™, SYNC I/O™, or equivalent). See the guides for these peripherals for details on their installation and configuration.

We strongly recommend that you research your synchronization requirements thoroughly. For example, if you are using Pro Tools for audio post production work for video, consult your video engineer or editor to determine what time code format will be used. Also, there may be additional time code issues that affect how you use synchronization (such as pull-down). Your Digidesign dealer may be able to offer suggestions about working with synchronization in your studio.

Synchronization Peripherals

Synchronization peripherals include a wide variety of devices, ranging from simple MTC-capable MIDI interfaces, to high precision time code synchronizers capable of handling many professional time code and clock standards used in the music, television, and film industries.

Though many devices are capable of providing synchronization, only the Digidesign SYNC and USD™ peripherals deliver frame-edge accuracy when synchronizing Pro Tools|HD® systems to external time code sources.

Aspects of Synchronization

Synchronization in a digital audio workstation has two concepts that need to be independently considered:

◆ "Where are we?" This is called the Positional Reference.

◆ "How fast are we going?" This is called the Clock Reference.

To synchronize Pro Tools to another device (such as a tape machine or VTR) accurately over an extended period of time, Pro Tools needs to know where the device is and at what speed it is running. Some peripherals can provide only one of these references; for example, a black burst generator provides only a clock reference. Some peripherals, such as the Digidesign SYNC HD, can provide both.

Synchronizing Pro Tools

Pro Tools|HD systems require a clock reference, in addition to time code, to *maintain* correct synchronization once it has been achieved. In addition, transfers (recording) between digital devices *must* be performed to a resolved clock source.

You can resolve Pro Tools to an external clock reference without locking it to a positional reference. For example, you can use any Pro Tools|HD audio interface to resolve the Pro Tools sample clock to an incoming digital signal (S/PDIF, or Word Clock, for example) without synchronizing Pro Tools to external time code.

In the following example, Pro Tools uses time code for positional information, and a clock reference to maintain synchronization. Pro Tools is slaved, using the SYNC peripheral, to a video tape recorder, with Pro Tools and the VTR referenced to the same house video reference source (house sync).

Example: Pro Tools slaved, through a SYNC I/O or USD, to a video tape recorder

In this example, house sync is provided to the SYNC peripheral, which provides the master Loop Sync signal to the Pro Tools|HD system through a Pro Tools|HD audio interface. House sync is also provided to the VTR.

1 When you start the video tape, time code (LTC or VITC) is read from the tape and routed to the SYNC or USD. The SYNC (or USD) reads the time code position and continuously forwards the *positional reference* information down the SYNC serial connection to the Pro Tools card.

2 Pro Tools takes the first time code address it receives, and calculates the sample location in the session that corresponds to the address. Though you can work with Pro Tools using any standard time code format, it calculates internally in sample numbers.

3 Assuming that the time code address corresponds to a sample number that is within the Pro Tools session, Pro Tools converts the time code address to a sample number within the session, and begins playing from that point. The point from which playback starts is the *trigger point*. The clock reference is used by Pro Tools and the VTR to keep them running at the same speed.

4 At this point, if the video tape is stopped, rewound, and started again, the process repeats, based upon a newly calculated trigger point.

About Positional References

Time code is positional information in an analog signal or a digital streaming signal that can be recorded on magnetic video or audio tape. Time code can be used as a positional reference, to synchronize the playback and recording of your Pro Tools system with another machine's time code signal, such as an analog multitrack tape machine or a video tape recorder (VTR).

SMPTE and ISO/EBU Time Code

The "Where are we?" question refers to relative position. To describe position, many professional audio, video, and multimedia devices and programs use SMPTE (Society of Motion Picture & Television Engineers) time code. In Europe, a standard called ISO (International Standards Organization) time code, formerly called EBU (European Broadcasters Union) time code, is generally used. These time code types are almost exactly the same in terms of how they are represented electronically. However, ISO/EBU time code works at a fixed rate of 25 fps (frames per second) and SMPTE includes provisions for several different frame rates. For the purposes of explanation, this chapter will generally refer to frame-measured timing information as "SMPTE time code." This description should be understood to include SMPTE and ISO/EBU time code.

SMPTE Time Code Methods

There are two basic techniques used to record SMPTE time code onto magnetic tape: LTC (Linear Time Code) and VITC (Vertical Interval Time Code). LTC is recorded or generated on an audio channel or a dedicated time code track of the audio or video device. VITC is recorded within the video signal in the video "blanking area" of each video frame. VITC cannot be recorded on audio tracks, so it has no application when working with audio tape recorders, but it does offer features for post production (such as still frame and slow speed time code address reading).

There is also a non-SMPTE form of time code called MIDI Time Code (MTC) that some devices use to send timing information.

SMPTE Time Code Units

Time code describes locations in terms of hours, minutes, seconds, frames, and subframes (1/100th of a frame). The frame is used as a unit of time measurement due to SMPTE time code's origin in film and video applications (see also "Frame Count and Frame Rate" on page 3.) Depending on the SMPTE frame rate, one frame is equal to 1/24th, 1/25th, 1/29.97th, or 1/30th of a second. For example, a video tape time code reading of "01:12:27:15" would tell us that we were at a position of one hour, twelve minutes, twenty-seven seconds, and fifteen frames. However, this time address alone does not tell us frame rate information.

Because SMPTE stores an absolute time reference on the tape in the form of time code, any location on that tape can be precisely located by devices that read time code. Once the time code has been recorded or *striped* on a tape, it provides a permanent positional reference that allows Pro Tools to link the playback of an event to an exact tape location. For example, with time code synchronization, a gun shot sound effect can be played at the precise instant that the gun's flash appears on-screen.

Frame Count and Frame Rate

It is important to remember the difference between frame count and frame rate.

Frame count is the amount of frames that the time code counts before ascending to the next second count. Standard frame counts are:

Frame counts per frame rate

Frame Count	fps
frames 0–23	24
frames 0–24	25
frames 0–29	30

"Frame rate" is the rate of speed that the time code is ascending through the frames. For example, when someone refers to a time code of 29.97, they usually are referring to "using a frame count of 30 frames, but counting each frame at the speed of 29.97 frames per second."

LTC (Longitudinal or Linear Time Code)

LTC is time code that is recorded and played back, or generated, in the form of an analog audio signal. LTC is supported by many audio and video tape recorders.

LTC Speed Usage

LTC can be read at high tape shuttle speeds, allowing a machine's time code reader to communicate with synchronizers at rewind or fast forward speeds exceeding 50 times playback speed (provided the tape recorder is able to reproduce the time code at this speed). However, LTC cannot be read at very slow shuttle speeds (such as when you are "crawling" the tape frame by frame) or when the machine is paused. With LTC, the VTR must be running (usually at a minimum speed of about 1/10th normal playback speed) in order to capture a SMPTE time address.

VITC (Vertical Interval Time Code)

VITC is a type of time code that is recorded and played as an invisible part of a video signal. VITC is commonly used in professional video editing and audio-for-picture applications. Because VITC is recorded as part of each video frame, it must be recorded at the same time as the video signal—it cannot be added later as LTC can. Since VITC cannot be recorded on audio tracks, it is never used to synchronize audio-only recorders. Instead, LTC is most often used in audio-only applications.

VITC Speed Usage

VITC's ability to be read when moving a VTR transport at slow speeds or when the VTR is paused makes it more useful than LTC in these situations. When VITC is used, Pro Tools can capture the current SMPTE time from the VTR when it is paused or in "crawl" mode. However, if you are using additional external transport synchronizers in your setup, most synchronizers cannot read VITC at speeds exceeding approximately 10 times playback speed, preventing slaved machines from maintaining synchronization during rewind and fast forward.

LTC/VITC Auto-Switching

Many synchronizers and devices support automatic switching between LTC and VITC, depending on the speed, to get the best of both worlds (both the SYNC I/O and USD support auto-switching). For example, VITC might be used when a VTR is paused, or crawling frame-by-frame, while the synchronizer might automatically switch to LTC when fast-forwarding.

Bi-Phase/Tach

This electronic pulse stream is used by film mag recorders, film editing stations, and film projectors. You can use this format to synchronize Pro Tools if you have a SYNC (or USD). Unlike time code, Bi-Phase/Tach doesn't actually contain absolute location information. It simply supplies speed (based upon the frequency of the pulses) and direction, and therefore, relative position. Since the SYNC unit can "count" both the speed and direction of the stream of pulses, it can use a Bi-Phase/Tach source to deduce positional information from a starting "address point." The difference between Bi-Phase and Tach formats is that Bi-Phase encodes rate and direction on a pair of signals using a format called phase-quadrature, while Tach encodes rate on one signal and direction on the other. For more information on Bi-Phase/Tach, see the *SYNC HD Guide* or *SYNC I/O Guide*.

SMPTE Frame Formats

Several different formats of SMPTE time code exist, and Pro Tools can synchronize to all common formats with a compatible synchronization peripheral.

When you work with NTSC video (the standard in North America and Japan), you will generally work with the NTSC color video standard: either 29.97 fps Non-Drop or 29.97 fps Drop frame. If you are working with PAL, your frame rate is 25 fps.

Pro Tools supports the following SMPTE frame rates:

30 fps Frame Format

This is the original SMPTE format developed for monochrome (black & white) video, and is commonly used in audio-only applications. This format is often referred to as 30 Non-Drop frame format.

30 fps Drop Frame Format

Some field film recordings are done at 30df so when they get pulled down after the telecine transfer, they will end up as 29.97df.

29.97 Non-Drop Frame Format

This format is used with NTSC color video. It runs at a rate of 29.97 fps.

29.97 Drop Frame Format

NTSC color video has an actual frame rate of 29.97 fps, so an hour's worth of frames (108,000) running at 29.97 fps Non-Drop will take slightly longer than one hour of real time to play. This makes calculating the actual length of a program difficult when using 29.97 Non-Drop time code. A program that spans one hour of 29.97 Non-Drop time code addresses (for example, from 1:00:00:00 to 2:00:00:00) is actually 60 minutes, 3 seconds and 18 frames long.

To make working with 29.97 time code easier for broadcasters, the SMPTE committee created 29.97 Drop Frame time code, which runs at exactly the same speed as 29.97 Non-Drop (non-drop frame) time code, but compensates for the slower speed by "dropping" (omitting) two frames at the top of each minute, with the exception of every 10th minute. For example, the time code address of 1:01:00:00 does not exist in drop frame code because it has been skipped.

Note that even though time code addresses are skipped in drop frame format, actual frames of video material are not dropped.

At the end of a program that spans precisely one hour of drop frame time code (for example, 1:00:00;00 to 2:00:00;00), exactly one hour of real time has elapsed.

Although it sounds complicated, drop frame time code allows broadcasters to rely on time code values when calculating the true length of programs, facilitating accurate program scheduling.

25 fps Frame Format

This format is used with the European PAL video standard, which runs at a 25 fps frame rate. This format is also called the EBU (European Broadcast Union) format because it's used by broadcasters throughout most of Europe.

24 fps Frame Format

This format is used for high-definition video and film applications. Film is typically photographed and projected at a 24 fps frame rate, so this SMPTE format is useful when one time code frame should equal one film frame

23.976 fps Frame Format

This format is used for high-definition digital video production using NTSC video equipment, and film applications.

Working with Film-Originated Material

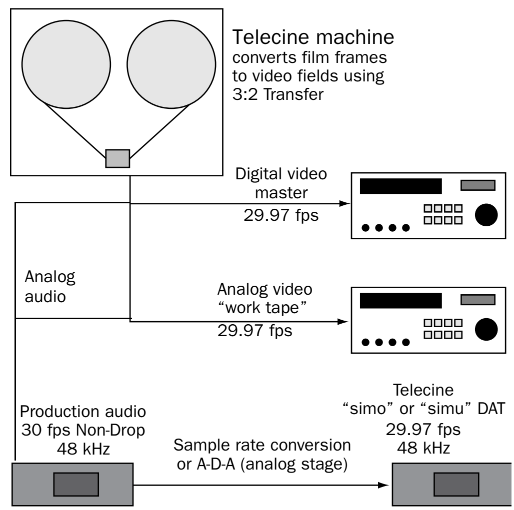
When you do post production work in Pro Tools, you will usually work with video material. However, it is possible that the video you are working on was shot on film.

Film footage and production sound go through separate conversion processes before they reach video, and the audio post production stage. The film is transferred to video using a process called Telecine, using a method called 3:2 Pulldown. Audio can also be pulled down during the transfer, or you might end up working with audio that has not been adjusted (production sound).

Typically, during the Telecine process, a master digital video tape is created, along with a work copy on Betacam or 3/4-inch analog video tape for the picture editor to use. At the same time, a new audio master may be created by slowing down, or *pulling down* by 0.1%, the production sound to compensate for the change in speed from film to NTSC video. (See Figure 1 on page 8.)

Guide Tracks and Conforming

In the Video editing process, the audio track produced by the video editor (the "guide track") is rough and needs to be enhanced and improved by the audio engineer. For this reason, the audio engineer will need to re-assemble the original sound elements in a process known as *conforming*.

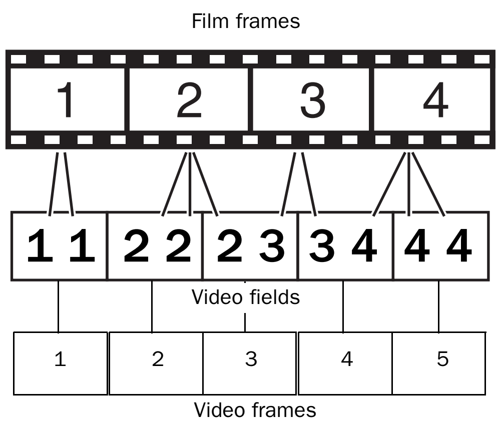


The Telecine stage of video post production

3:2 Pulldown

A film clip that lasts 1000 seconds consists of 24,000 film frames (pictures). If you want to transfer that film to 1000 seconds worth of NTSC color video, you have to fit 24,000 film frames into 29,970.02997 video frames.

If we use the black and white NTSC video standard (30 fps) instead of 29.97 fps, the process of converting film frames to video frames is greatly simplified. Now instead of any fractional frames, we have 24,000 film frames going into 30,000 video frames (60,000 video fields). In the Telecine process (for NTSC color video), each odd film frame is copied to two video fields, and each even film frame is copied to three video fields, creating what is called a 3:2 Pulldown. The speed of the film is also “pulled down” to 23.976 fps in order to accommodate the slower speed of NTSC color video compared to NTSC black and white video (29.97 fps compared to 30 fps).



How film frames translate to video fields and frames in a Telecine transfer

Film Speed Differs from NTSC Video Speed

When spotting audio to video that was transferred from film to NTSC video, there are two important terms to keep in mind: film speed and video speed.

Film Speed Film speed refers to audio that was recorded and plays back in synchronization with the original film material. This audio often comes from production reels recorded on a Nagra® recorder or a field DAT recorder, and is usually striped with 30 Non-Drop time code. Film must be pulled down 0.1% when being transferred to 29.97 NTSC. Film must be pulled up by 4.16667% when film is being transferred to PAL.

Video Speed Video speed refers to audio that is running at the NTSC color standard of 29.97 fps. Video speed is 0.1% slower than film speed, so audio that is still at film speed will be out of sync with the video.

In Figure 1, note the following:

- The vertical arrow at the left (“faster” “slower”) represents program speed or rate.
- Telecine is a frame count conversion (for example, 24 to 30).
- The speed you play the result back is a frame rate conversion (for example, 30 to 29.97).

The following table lists the applications for each of the frame rates and counts shown in Figure 1 on page 8.

Frame Rates and their Applications

Frame Rate	Applications
25 fps	PAL
24 fps	Film, and 24P HD
23.976 fps	24P pull down, for NTSC broadcast, and HD digital video
30 fps	30 frame music production, and 30 frame recording for film
29.97 fps	Broadcast NTSC video

Frame Rates and Relative Playback Speeds

The following diagram illustrates the relative playback speeds of SMPTE formats.

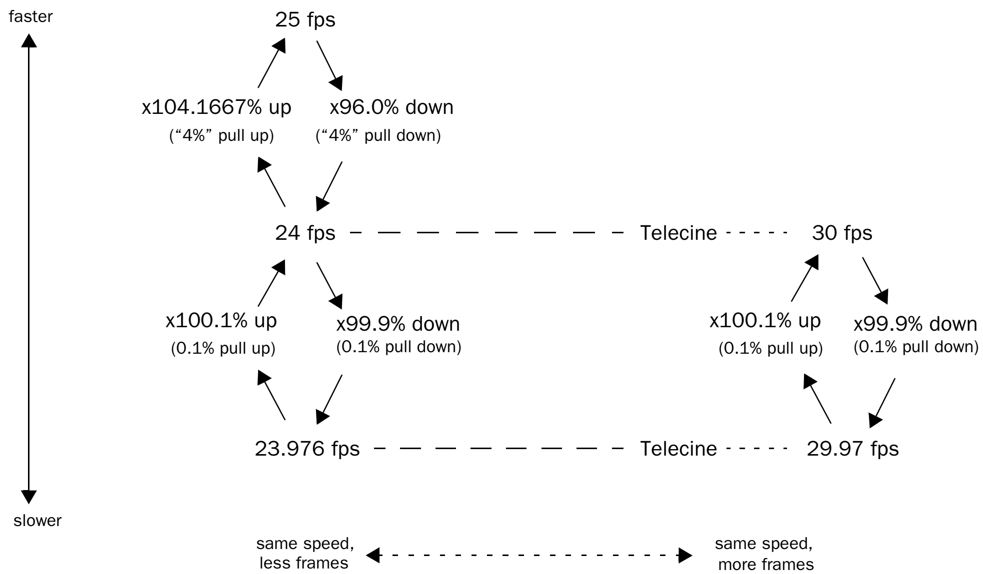


Figure 1. Frame rates, telecine transfers, and relative playback speeds

Pull Up and Pull Down

Pull Up and *Pull Down* are terms used to refer to the deliberate recalibration of the audio sample rate clock (speed, or musical pitch) in order to compensate for a speed change. Pro Tools can be used to pull down or pull up audio or video playback rates. Pro Tools 5.3.1 and higher support pull rates known as "4% factors" (4.0% down, or 4.167% up to be exact), applicable to audio and video playback, to support PAL/film/NTSC conversion requirements.

Pull Down allows you to play back film-originated material at video speed (–0.1%). Pull Up allows you to play back video speed material at film speed (+0.1%).

Using Pro Tools in Pull Up or Pull Down modes requires a SYNC HD, SYNC I/O, USD, or third party synchronizer.

When to Pull Up or Pull Down

There are many ways to get audio into Pro Tools for post production. Consider your source audio and your final destination format carefully. In some cases, audio will already be pulled down for you. In other cases, audio will have to be temporarily pulled down. In still other cases, you may choose to pull down your audio source, like a DAT deck, then use a a D-A-D (digital-to-analog-to-digital) process, or the Sample Rate Conversion option on the 192 I/O, to record the audio into Pro Tools at the proper sample rate. Or you may choose only to pull Pro Tools up or down on delivery of the audio.

Because mistakes in pull factors can be expensive, good communication with other participants in the project is critical. Find out as soon as possible how your task relates to the rest of the project.

Final Audio Destination: Film

If your final destination is film, your source audio is at film speed, and your goal is to edit and mix audio in Pro Tools and then lay back to a device that runs at film speed (such as mag or time code DAT), you can temporarily pull down the audio in Pro Tools for NTSC video work, then return the audio back to film speed when you're finished (by disengaging pull down).

For example, film speed audio from a field audio recorder that is referenced to 30 fps time code is recorded into your Pro Tools system at a sample rate of 44.1 or 48 kHz. Keeping in mind that film speed is faster than video speed, select 30 fps in your Session Setup window in Pro Tools, and record in your audio online and referenced to the time code on the field audio recorder.

⚠ *Most "simul-dats" or "simo-dats" already have their audio pulled down, and should not be pulled down further.*

Once all the audio has been recorded, and you are locked to a video work print (at video speed), enable Pull Down. If you are using a Digidesign SYNC HD or SYNC I/O, select Pull Down in the Session Setup window. If your synchronizer is not a Digidesign product, select Pull Down on the front of your synchronizer, then enable Pull Down in the Pro Tools Session Setup window. At this point, it is highly recommended that you verify whether the video you're working with is striped with 29.97 Drop Frame or Non-Drop Frame time code. While in Pull Down mode, you can work with your reference video and everything will remain synchronized and run at the proper speed (assuming your system is completely resolved).

Once you are ready to lay back your completed project to an audio device running at film speed, deselect Pull Down in the Session Setup window, and from your synchronizer if it is not a SYNC HD or SYNC I/O. Then change your time code frame rate in the Pro Tools session back to 30 fps. Once Pull Down has been deselected, the audio played back from Pro Tools will synchronize perfectly with the edited film.

Alternatively, you can pull down the source audio deck while recording audio into Pro Tools, work at 29.97 fps with no Pull-Down selected in Pro Tools, and then switch to 30 Non-Drop frame format, and select Pull Up during the delivery stage. You will have to perform a sample rate conversion on the audio either digitally or by using an analog stage (D-A-D).

Only the inputs on the Digital I/O card (on the 192 I/O) feature real-time sample rate conversion.

⚠ *You will also need to select the Audio Rate Pull Up/Down option in the Session Setup window.*

Final Audio Destination: Video

If you are working with video that was transferred from film, your audio source is at film speed, and the final layback destination is NTSC video (or television), and you would like to provide a digital transfer to your clients, you will need to alter the above recipe slightly. Keep in mind that when you are working in Pull Down mode, your active sample rate is 44.056 kHz (if the audio was recorded at 44.1 kHz) or 47.952 kHz (if the audio was recorded at 48 kHz).

Pull Down the Audio Source

Some professional DAT machines will let you pull down the sample rate to 44.056 kHz (and 47.952 kHz). You can record this audio into Pro Tools using a D-A-D (digital-to-analog-to-digital) process, or Digidesign's 192 I/O Sample Rate Conversion option. Then your audio will be at the correct speed for the remainder of the project, since the final destination is video, and no Pull Down or Pull Up is necessary.

Pull Up Pro Tools While Recording

If you cannot pull down your source, you can accomplish the same thing by Pulling Up Pro Tools, and setting your SMPTE frame format to 30 fps, before recording in the production sound. First select 30 fps as the frame format in the Session Setup window. Then select an Audio Rate Pull Up option in the Session Setup window (and on your synchronizer if it isn't a SYNC HD or SYNC I/O) before you record in the production audio. In this case, while the production audio is running at 44.1 or 48 kHz, Pro Tools is running (and recording) at a rate of 44.144 or 48.048 kHz. After all the production audio has been recorded into Pro Tools, reset the Audio Rate Pull to none in the Session Setup window, and on your hardware synchronizer if it is not a SYNC I/O. After you deselect Pull Up, the recorded audio will play back 0.1% slower, synchronized with the video, while achieving a true playback sample rate of 44.1 or 48 kHz. Note that this process is designed for a final destination of video; to bring this audio back up to film speed you would have to pull up Pro Tools and record to a destination that is not Pulled Up.

Note on Sample Rate Conversion

In many cases, you have to perform a sample rate conversion at some point, either digitally, or by recording in audio using an analog stage (D-A-D). The only situation where sample rate conversion never has to be performed is when you are working with film speed audio and your final destination format is film. Then you can simply pull down Pro Tools while you work with the video, then deselect Pull Down to set the audio back to film speed. (On the 192 I/O and 192 Digital I/O, the inputs on the Digital I/O card feature real-time sample rate conversion.)

Using Digital Input

If working with Pull Up or Pull Down, do not "resolve" to any digital inputs that are used as audio sources in Pro Tools. This would override the use of the SYNC HD or SYNC I/O as the clock reference. Any equipment providing digital audio sources to Pro Tools should be synchronized externally.

⚠ *You will also need to select the Audio Rate Pull Up/Down option in the Session Setup window.*

chapter 2

Surround Concepts

Pro Tools|HD systems and Pro Tools LE® systems with Complete Production Toolkit support multichannel mixing for surround sound.

If you are new to surround mixing, read this chapter for an introduction to surround terminology and concepts.

For information on using Pro Tools surround features, refer to the Pro Tools Reference Guide.

Mixing Formats and Surround Formats

When running Pro Tools with the Surround Mixer plug-in, you can mix in 3- to 8-channel formats, in addition to standard mono and stereo.

Supported mixing formats include Mono, Stereo, LCR, Quad, LCRS, 5.0, 5.1, 6.0, 6.1, 7.0, and 7.1.

Surround formats include Dolby Surround (Pro Logic), Dolby Digital, DTS, and SDDS. Pro Tools does not provide its own surround format processing, and requires appropriate plug-ins (such as Dolby Surround Tools™) or hardware to provide Dolby surround encoding and decoding.

Pro Tools Mixing Formats

Table 3. Multichannel Mixing and Surround Formats

Speaker Channels	Multichannel Mixing Format	Surround Format	Channels and Track Layout
1	Mono		C
2	Stereo		L R
3	LCR	Cinema Stereo	L C R
4	Quad	Quadraphonic	L R Lr Rr
4	LCRS	Dolby Surround (Pro Logic)	L C R S
6	5.1	Film (Pro Tools default), for Dolby Digital	L C R Ls Rs LFE
6	5.1	SMPTE/ITU (Control|24)	L R C LFE Ls Rs
6	5.1	DTS (ProControl)	L R Ls Rs C LFE
7	6.1	Dolby Surround EX	L C R Ls Cs Rs LFE
8	7.1	SDDS	L Lc C Rc R Ls Rs LFE
Legend: L = Left; R = Right; C = Center, S = surround (mono); Ls = Left Surround; Rs = Right Surround; Lc = Left Center; Rc = Right Center; Cs = Center Surround; Lr = Left Rear; Rr = Right Rear; LFE = Low Frequency Effects (handled by a sub-woofer or bass management systems)			

Speaker Layouts

Figure 2 on page 13 illustrates the speaker arrangements of each surround format. Speaker placement for each format is approximate. For proper placement, alignment, and calibration of surround monitoring systems, consult the documentation that came with your speakers and other monitoring equipment. Placement of speakers is crucial to accurate monitoring of any mix, but this is especially true with multichannel mixing for surround sound.

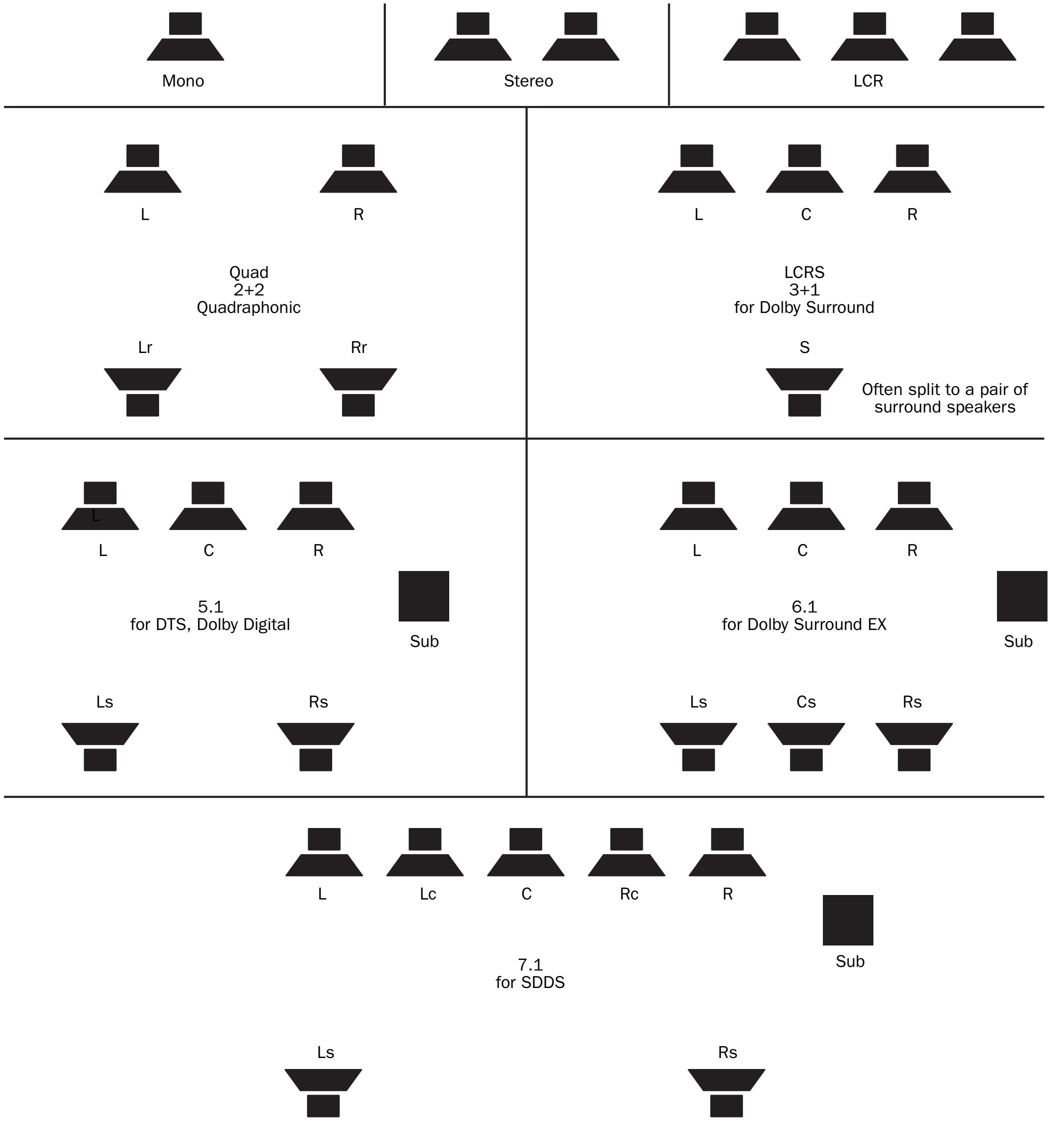


Figure 2. Speaker arrangement of surround formats

Surround Monitoring

In order to monitor your current multichannel mix in a given surround format (such as DTS or Dolby Digital), encoding and decoding equipment for that format is required. Pro Tools does not provide direct support for surround format processing, but encoding/decoding hardware or software is available from Digidesign and third-party manufacturers. Contact Digidesign or your dealer for information.

Proper speaker placement, studio calibration, additional surround processors, monitors, and a properly calibrated system can lessen the variables that your mix will be subjected to when the audience eventually hears it in the theater, in their home, or elsewhere.

The Importance of Speaker Placement

It is very important that your surround monitor system be installed and configured correctly. Proper speaker placement, angling, and level calibration are necessities for surround mixing, so consult the manufacturer of your monitor system. Several surround formats (especially all of those from Dolby as well as DTS) have very specific speaker and monitoring recommendations, so take the time to locate this information and adhere to the suggestions they provide.

For an example of a 5.1 monitor system, see Figure 3.

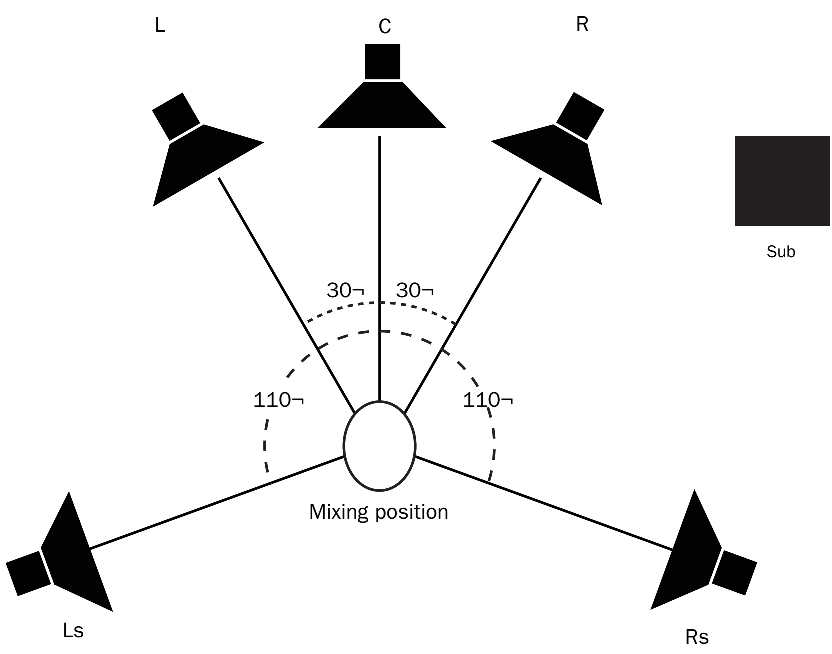


Figure 3. Example of speaker arrangement for 5.1 mixing and monitoring (SMPTE specification). Sub placement is variable, while all other speakers should be as close to the suggested angles as possible.

Calibrated Surround Monitoring

One of the most important things you can do is to calibrate not only Pro Tools, but also your monitoring system and other equipment.

For example, many post production facilities calibrate their Pro Tools audio interfaces for –20 dB = 0 VU. Once Pro Tools hardware has been calibrated to unity gain, the monitoring system can be adjusted to account for anomalies in the room acoustics, speaker placement, and other variables.

Calibrating your Pro Tools system involves attenuating the input and output stages of your system. Settings and reference level vary according to the applicable standards of the current project.

Instructions for calibrating a Pro Tools 192 I/O audio interface are included with the 192 I/O documentation.

Calibrating Your Studio The first task for surround mixing should be getting your monitor speakers in proper position, alignment and calibration for accurate listening and mixing.

Speaker position and alignment will vary according to each installation, and is very specific for different speakers, manufacturers, and for each surround format. Instructions for optimal speaker location should have been provided with the speaker and monitor system, so consult the documentation provided by its manufacturer.

Proper speaker calibration ensures that your system plays back at accurately balanced levels. This lets you compensate for any variances in speaker position, alignment, or performance.

Formats and Terminology

Some multichannel mixing formats are referred to numerically (such as 5.1 and 7.1). These numbers describe the number and type of output (speaker) channels required for that format. Other formats are referred to by their track layout (such as LCRS and LCR).

Surround formats are referred to by their developer and product name, such as Dolby Surround (or Pro Logic), Dolby Digital, DTS, Dolby Surround EX, and others. Each surround format requires you to mix in a compatible mixing format. For example, both Dolby Digital and DTS are "5.1 surround formats."

The standards for each format include more than just the number of speakers. Formats also may require specific track layout and filtering for the various channels. For example, the surround channel in Dolby Surround is band limited from 100 Hz to 7 kHz.

Surround formats also increase the importance of phase, balance, and sound placement due to the various ways in which multichannel mixes can be encoded, decoded, compressed, matrixed, or downmixed. Many publications are available on these subjects. For a listing of several sources of surround format information, see "Where to Get More Information on Surround Technology" on page 18.

LCRS for Dolby Surround

Dolby Surround (Pro Logic) is a four-channel format that is one of the most widely installed consumer surround formats.

This format, which requires Dolby encoding and decoding technology, provides three full-range front speakers left, center, and right, plus a mono surround channel (often split and monitored through pairs of satellite speakers, but a single "rear" channel nonetheless).

Surround Format Example

The goal of Dolby Surround encoding is to make a four-channel mix (LCRS) more portable, by transforming it into an encoded two-channel mix referred to as Lt/Rt. Lt and Rt represent Left Total and Right Total, respectively. Whenever you see Lt/Rt, it refers specifically to material that has been processed by a Dolby Surround Encoder (such as the Dolby SEU4 or Dolby Surround Tools plug-in).

The Lt/Rt mix can then be transported, transmitted, or played as a stereo mix, making it compatible with most home and theater systems. Alternately, the Lt/Rt pair can be decoded back into an LCRS mix in home theater systems where the required decoding hardware is available.

⚠ *For professional results, always use professional-level products (such as the Dolby Model SDU4 Decoder, Model DP564 Decoder, or Dolby Surround Tools plug-in) when mix monitoring. Pro Logic-equipped consumer products have auto-balancing features which, if used for mix monitoring, may contribute to inappropriate mixing decisions*

5.1 for Dolby Digital and DTS

Both Dolby Digital and DTS formats are six-channel, 5.1 formats. Both provide five full-range speakers and a sub. These two surround formats use different track layouts, filtering, and compression as part of their specifications.

.1 Formats

The ".1" in any multichannel mix format indicates an LFE (Low Frequency Effects) track in the mix. Whenever a decimal is part of a surround format, it indicates a discrete *Sub* channel in the surround playback system that is intended for playback through a subwoofer speaker. For example, Dolby Digital is a 5.1 format intended for playback through systems with five full-range speakers and one subwoofer speaker. Dolby Surround EX is a 6.1 format, with six full-range speakers and one subwoofer speaker. SDDS is a 7.1 format, with seven full-range speakers and one subwoofer speaker.

Sub content will include the LFE channel of a ".1" format mix, as well as the effects of any bass management in the playback system. See "LFE" on page 17.

.0 Formats

The ".0" formats (5.0, 6.0, 7.0), while not associated with any specific surround formats, are useful in many mixes as sub-paths. For example, not all tracks will have content that needs to be routed to the LFE channel (the ".1" channel). Assigning such tracks to a 5.0 path provides a 360° panner without LFE controls.

LFE

When a decimal such as ".1" is present in the name of a surround format, it indicates the presence of an LFE channel. The LFE channel (for Low Frequency Effects, also known as Low Frequency Enhancement) refers most often to a specific track used in *production* (not during playback or decoding). LFE is the ".1" component in mix formats such as 5.1 and 7.1.

The reason the LFE component is referred to as ".1" is that unlike the other five or seven speakers, LFE is not full bandwidth (or full-range). The LFE signal provides a discrete path for low end to the subwoofer, unaffected by bass management (if any). Low frequency effects include such things as thunder, explosions, and other bass effects.

LFE is generally used to enhance sound effects in films. There are no set rules about whether or not you must employ an LFE channel.

⚠ *Pro Tools applies no filtering to LFE signals. Some delivery requirements may require filtering for the LFE track. See "Mixing Formats and Surround Formats" on page 11 for more information.*

Sub Content

When surround mixes are played back on systems that include one or more subs, subwoofer content comes from either or both of the following sources:

- All bass from all channels that is below the threshold of the bass management filter cutoff (if any)

  – and –

- Audio in the LFE channel (if any)

A surround sub will play back all the lowest-frequency sounds from all of the other speaker channels, below a fixed frequency threshold. The threshold varies among different surround formats. See "Where to Get More Information on Surround Technology" on page 18 for a list of surround specification resources.

In addition to the frequency cutoffs and other specifications of each surround format, playback system variables can affect sub content. See "Surround Playback System Variables" on page 19 for more information.

Divergence

Divergence is a surround mixing control that lets you set panning "width." Full divergence results in discrete (or narrow) panning. Lower divergence settings result in progressively less discrete (or wider) panning.

With Full (100%) divergence, tracks can be panned exclusively, or discretely, to a single speaker. Sounds panned to a single speaker are only audible in that speaker.

When divergence is less than 100%, tracks will be audible in neighboring speakers even when panned directly to a single speaker in the grid. Lower divergence settings result in a progressively wider source signal.

Divergence Example

To understand divergence, it can be helpful to imagine the inside of a large movie theater. This is a good example because one of the most challenging playback variables one faces in multichannel mixing is the size of the intended listening environment.

Unlike the typical living room, movie theaters are large spaces with speakers placed widely apart. Due to distance, sounds panned discretely to the front right speaker, for example, might be inaudible in the opposite corner. To avoid this problem, variable divergence lets you control the panning width, in order to widen the sound source. This results in signals spreading into adjacent speakers, even when panned 100% to an individual speaker.

Divergence options in Pro Tools are explained in the Pro Tools Reference Guide.

Where to Get More Information on Surround Technology

The table below lists several sites that are good starting points for your research into the constantly evolving world of surround sound production.

Resources for surround specifications and information

Surround Specifications	Web Site
Dolby Surround/Pro Logic	www.dolby.com
Dolby Digital	www.dolby.com
Dolby Surround EX	www.dolby.com
DTS	www.dtsonline.com
SDDS	www.sony.com
THX	www.thx.com

Surround Mixing Concepts

As in stereo mixing and mastering, the goal in surround mixing is to provide the best sounding mix to the greatest number of potential listeners. Doing so for surround mixes requires many of the same techniques used for professional stereo production, plus several unique factors that are introduced in the following sections.

Surround Format Compatibility

Pro Tools lets you mix in surround and create multichannel masters, consisting of four, six, or more tracks that comprise an LCR, 5.1, or other format mix.

◆ Whenever necessary, multichannel mixes can be transferred as discrete, unencoded, multitrack masters. Track layout requirements vary by format.

◆ Consumer playback systems do not necessarily support every format. *Downmixing* occurs when a specific format mix has to be created from another. (A typical example of this is listening to a DVD's 5.1 Dolby Digital soundtrack downmixed to stereo.)

To anticipate the effects of surround encode and decode, as well as potential downmixing, professional surround mixes are monitored through appropriate encoding and decoding processors. Monitor controller systems let engineers hear their mix through different speakers and configurations for reference, as well as compare factors such as different bass management settings and their effect.

Surround Formats and Delivery Mediums

Surround mixes are tailored for their specific delivery medium. Most often, this requires format-specific encode and decode processing.

⚠ *Some delivery media on which surround formats are distributed may have additional audio constraints, which could influence your work flow in Pro Tools. Dolby Digital audio on a DVD-Video disc, for example, requires a 48 kHz sampling rate. If your Pro Tools session isn't at 48 kHz, you must sample rate convert the audio before encoding it with Dolby Digital for DVD.*

Encoding and Decoding

Many surround formats utilize some form of encoding and decoding to make it practical to deliver, broadcast, and transfer the multiple channels of full-bandwidth audio they require.

◆ *Encoding* is the process necessary to make multichannel mixes portable and playable. In many cases, this involves taking the four, five, six, or more discrete channels resulting from a multichannel mix and converting them into a two-channel stream for broadcast (still a predominantly 2-channel medium).

◆ *Decoding* is the process needed to reproduce (or unfold) the discrete surround channels from a 2-channel delivery medium.

Both encoding and decoding, no matter how refined, represent additional processing stages applied to your mix before it reaches its ultimate destination, the audience.

For example, because the Dolby Surround algorithm depends heavily on phase relationships, there is always a significant difference in a decoded LCRS output as compared to the original LCRS mix. To account for these anomalies, engineers mixing for Dolby Surround listen through hardware encode and decode processors for reference.

Professional mixing and mastering engineers use encoders and decoders to precisely audition the effect of the encoding and decoding process, and make any adjustments necessary.

Surround Processing and Pro Tools

Pro Tools requires additional software or hardware for surround encoding, decoding, and processing. For example, the Dolby Surround Tools™ plug-in lets you monitor and process completely within the Pro Tools environment.

Surround Tools and other surround processing solutions are available from Digidesign and third party manufacturers. Contact your Digidesign Dealer for more information.

Surround Playback System Variables

Different playback systems for surround sound introduce varying amounts and types of filtering, bass management, and other variables. These include the specifications for certain surround formats, as well as options to fine-tune a system for its particular installation.

In your own studio, you know what speakers you are listening to, what their qualities are, and what your control room sounds like. What is impossible to know, however, is what speakers are in the living room, theater, or concert hall where your mix will be heard by an audience.

Bass Management

Bass management is a function of the playback system. Bass management refers to a number of processes by which a playback system can control what will be heard through the Sub(woofer) and other channels. The purpose of bass management is to optimize low frequency reproduction and overall frequency response for your specific monitoring system. Bass management provides a way to tune bass response for variables, including room size and shape, the presence (or absence) of a subwoofer, and the frequency range of each speaker.

Filtering

In its simplest form, bass management applies high-pass filtering to the full-range speakers. Frequencies below the filter cutoff are summed with the LFE channel (if present) and then routed to the subwoofer. Different systems may offer additional bass management processes, including:

Bass Redirection Mutes or unmutes the low-frequency signals filtered out of the full-range channels (all signals that fall below the bass management filter cutoff frequency). This is useful when the full-range speakers are truly full-range, capable of accurately reproducing bass without distorting.

Bass Extension Allows the bass (however it is derived) to be routed back through the full-range speakers. Though this may seem like a contradiction, its main purpose is to let sub content emanate from all around, rather than only from the subwoofer.

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