DTS-HD Master Audio Suite User Guide

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Record of Changes

Manual Version / Date	**Description**
V1.00 Rev 1/August 2006	**Initial Document Release**
V1.00 Rev 2/October 2006	**Document Cleanup**
V1.00 Rev 3/November 2006	**Edits for input files and general clean-up**
V1.00 Rev 4/Novemver 2006	**Added Uninstall section, revised Log File discussion, clarified Join/Append discussion**
V1.1 Rev 1/February 2007	**Added Encode Queue (section 7.4). New Main Window screen shot**

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1. Introduction

This user guide explains the use and operation of the DTS-HD Master Audio Suite and DTS Surround Audio Encoder.

1.1. DTS-HD Master Audio Suite

This suite consists of the DTS-HD Master Audio Suite Encoder, DTS-HD StreamPlayer ProTools plug-in and DTS-HD StreamTools This software suite is designed for the professional audio user to easily create DTS-HD, DTS Digital Surround and DTS-HD LBR encoded files using the Encoder module, to verify and validate the encoded material using the DTS-HD StreamPlayer plug-in and to perform various stream edits, for those cases where necessary, using a series of DTS-HD StreamTools.

The DTS-HD Master Audio Encoder is a software product capable of encoding the following:

DTS-HD encoded streams:

- ❒ DTS-HD Master Audio - variable bit rate, bit-for-bit (lossless) encoded streams with up to 7.1 channels from source files with sample rates up to 96kHz and in stereo from source files with a sample rate of 192 kHz
- ❒ For 192 kHz source material, HD DVD supports up to 2.0 channels and Blu-ray supports up to 5.1 channels.
- ❒ DTS-HD High Resolution Audio - constant bit rate (lossy) encoded streams at bit rates up to 5.766 Mbps for Blu-ray Disc and up to 3.018 Mbps for HD DVD, with up to 7.1 channels from source files with sampling rates up to 96 kHz.
- ❒ HD DVD Sub-Audio Coherent Acoustics (CA) using DTS Digital Surround encoding with up to 2.0 channels at bit rates from 126 kbps to 510 kbps
- ❒ HD DVD Sub-Audio Low Bit Rate (LBR) using DTS-HD Low Bit Rate encoding with up to 2.0 channels at bit rates from 24 kbps to 192 kbps.

  ☑ **Note**: DTS-HD Master Audio (lossless) and DTS-HD High Resolution encoded streams each contain a core substream of type 'DTS Digital Surround' that is backward compatible with all DTS Digital Surround decoding technologies in the market.

- ❒ DTS Digital Surround® encoded streams for DVD-V, up to 6.1 discrete channels at 48 or up to 6.1 channels with matrixed center surround (Cs) at 96 kHz sample rates, at bit rates up to 1.5 Mbps

☑ **Notes**:

1. The input source material may be 16-bit or 24-bit resolution.
2. Although the DTS-HD Master Audio Encoder module is capable of creating secondary audio streams for Blu-ray Disc format, at the time of this encoder software release, the required audio mixing metadata is not supported. As such, this encoding option is not selectable in this release.

The DTS-HD StreamPlayer is a software plug-in for Digidesign's ProTools application capable of playback of all DTS-HD and DTS Digital Surround audio streams as well as synchronizing the playback of audio to a video clip using a QuickTime movie placed into the ProTools session timeline along with a DTS-HD encoded audio file.

The DTS StreamTools is a series of software products capable of verifying and editing DTS-HD encoded streams as desired.

1.2. DTS Surround Audio Suite

This software package consists of an encoder that is only capable for creating DTS Digital Surround encoded streams for DVD-V, up to 6.1 discrete channels at 48 or up to 6.1 channels with matrixed center surround (Cs) at 96 kHz sample rates, at bit rates up to 1.5 Mbps. The user interface behaves identically to that of DTS-HD Master Audio with the noted exceptions found in the sections that follow.

2. System Requirements

2.1. Operating System Requirements

Microsoft Windows XP Operating System, minimum P4 running at 2.4 GHz

Apple Macintosh running OS X, G5 recommended.

Sun Java™ 2 Standard Edition Runtime Environment, Version 5 or later.

2.2. Memory Requirements

Microsoft Windows XP Operating System: minimum 512 MB RAM; 1GB RAM recommended

Apple Macintosh running OS X: minimum 512 MB RAM; 1GB RAM recommended.

2.3. Hard Disk Requirements

Hard drive performance depends on factors including system configuration, number of encodes, sample rate and bit rates. Locally mounted hard disk(s) are highly recommended to optimize file I/O processing. The capacity of the hard drive depends on the number of encoded files that the user will save. Nominally, an encoded file consisting of 8-channels of input material at 16 bit/48kHz resolution for a duration of 2 ½ hours will occupy approximately 3.5 GB of storage space. Given that the user will likely have video files and other material stored on a local hard drive, a minimum storage capacity of 50GB is highly recommended.

Network attached drives can also be used for storing the encoded streams. Disk I/O times will vary according to network parameters and settings. Please consult your network administrators for information related to network attached drives.

SCSI Hard Drives
For maximum encoding and playback performance, DTS recommends qualified SCSI hard drives and a qualified SCSI host bus adapter (HBA) card or (on Windows systems) a qualified build-in SCSI HBA connector on the motherboard.

FireWire Hard Drives
DTS recommends qualified FireWire drives and a qualified FireWire host adapter.

NOTE: DTS highly recommends that when using the Neyrinck Sound Code DTS-HD StreamPlayer ProTools plug-in, for optimum playback performance, the audio files should be located on a FireWire drive. Playback performance may be significantly impacted if audio playback occurs from files that are resident on the system drive.

2.4. iLok Usage

DTS-HD Master Audio Suite is an iLok-enabled software product that ships with an iLok Smart Key. The Smart Key contains the license and tokens necessary to activate and run the software you purchased. Simply insert the Smart Key into the USB slot and the system is ready for use. It is strongly recommended that once the DTS-HD Master Audio Suite applications are running, that the iLok Smart Key remain in the USB slot. Removing the Smart Key while the applications are running may result in the run-time application errors. The tokens may be transferred from one iLok Smart Key to another. Please refer to the "Manage Your iLok" on the iLok website located at http://www.ilok.com.

2.5. Port Assignments

The DTS-HD Master Audio Suite utilizes specific ports for communication between the Java interface and underlying applications. These ports are allocated as follows:

DTS-HD Master Audio Suite and DTS Surround Audio Suite
- client listens on port 4445
- application framework listens on 4444

DTS-HD StreamTools
- client listens on port 4448
- application framework listens on 4442

3. Installation Instructions

The DTS-HD Master Audio and DTS Surround Audio Suites encoder and DTS StreamTools applications ship on a CD-ROM located in this package. This CD-ROM should contain the following installation packages with the instructions on how to install the software and any other modules list in Table 3-1:

Table 3-1 Installation Instructions

Windows	Mac OS
DTSInstaller.exe	**DTSInstaller.dmg**
1) Double Click on DTSInstaller.exe	1) Double-click on DTSInstaller.dmg then Double-click on the DTSEncoder.pkg
2) Follow the instructions on the Wizard	2) Follow the instructions on the Wizard
3) If Java 5.0 run-time environment is not loaded, visit the Sun website at: http://www.java.com/en/download/windows_ie.jsp Locate: Select the Begin Download button and install the modules on your computer.	3) If Java 5.0 run-time environment is not loaded, visit the Apple website at: http://www.apple.com/support/downloads/j2se50release4ppc.html Locate: Select the 53MB button and install the modules on your computer.
4) Make sure that the iLok dongle is inserted into the USB port prior to initial launch of the DTS-HD Master Audio Encoder Suite. If the iLok Smart Key drivers are not installed or an older version of iLok drivers are installed, the drivers must be upgraded for proper DTS-HD Master Audio Encoder Suite functionality. The driver is located on the distribution disc in the "iLokDriver" folder. Remove all iLok/Pace USB dongles from the computer before installing/upgrading the iLok drivers. Double-click on the ***setup.exe*** file located in this folder and follow the instructions. At the conclusion of the iLok drivers installation, **REBOOT THE MACHINE.**	4) Make sure that the iLok dongle is inserted into the USB port prior to initial launch of the DTS-HD Master Audio Encoder Suite. If the iLok Smart Key drivers are not installed or an older version of iLok drivers are installed, the drivers must be upgraded for proper DTS-HD Master Audio Encoder Suite functionality. The driver is located on the distribution disc in the "iLokDriver" folder. Remove all iLok/Pace USB dongles from the computer before installing/upgrading the iLok drivers. Double-click on the ***InterLok Extensions Install*** file located in this folder and follow the instructions. At the conclusion of the iLok drivers installation, **REBOOT THE MACHINE.**

While the entire set of products contained in the installer will be copied to the local disk, access to the products is strictly controlled by the tokens located on the iLok Smart Key. Upgrades may be purchased by contacting your local dealer or visiting the DTS on-line store at http://www.dts.com.

If the product contains incomplete components or you are experiencing problems installing or running the software, please contact DTS customer service at: 1-800-959-4109 or email us at techsupport@dts.com

The Neyrinck DTS-HD StreamPlayer plug-in for ProTools ships on a single CD ROM that includes the installer and user guide for this module. Follow the instructions for installation and iLok authorization shown in the installer wizard. For more information contact DTS at proaudioinfo@dts.com

4. Uninstall Instructions

4.1. Windows Operating Systems

To uninstall the applications, please follow the standard process for removing applications by selecting the **Add/Remove Programs** from the Windows Control Panel. The Control Panel can be activated by selecting the Windows Start→ Settings button found on the taskbar

On the **Add/Remove Programs** page, search for the DTS-HD Master Audio Suite-DTS Surround Audio Suite application as shown in Figure 4-1. Select this icon and click on the "Remove" button. To ensure that the application has been removed from the system, check C:/Program Files/DTS/MAS-SAS for any residual files. If any exist, simply delete C:/Program Files/DTS/MAS-SAS from the system.

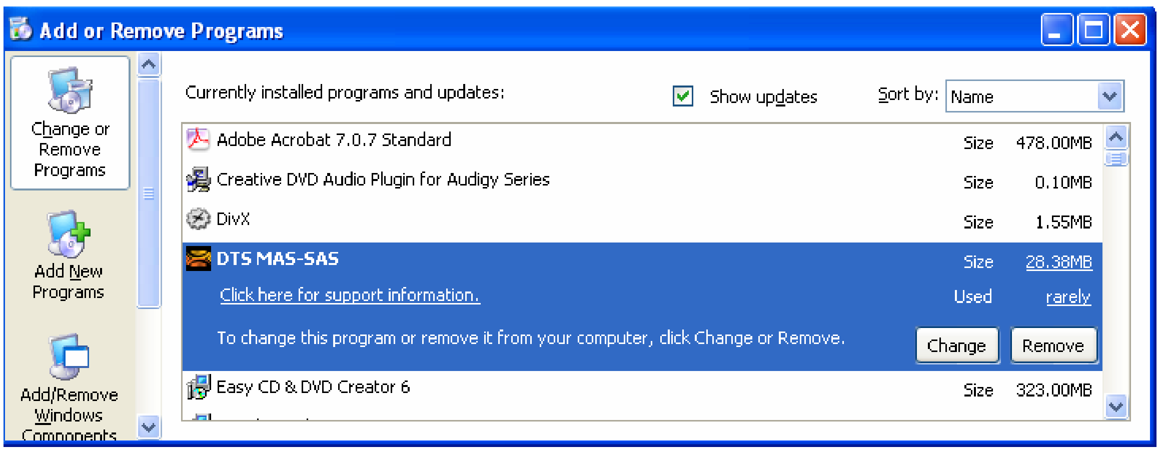


Figure 4-1 Windows Add/Remove Programs

4.2. Macintosh Operating Systems

To uninstall the DTS-HD Master Audio Suite-DTS Surround Audio Suite applications, simply drag the DTS folder located in the Applications directory into the trash as shown Figure 4-2.

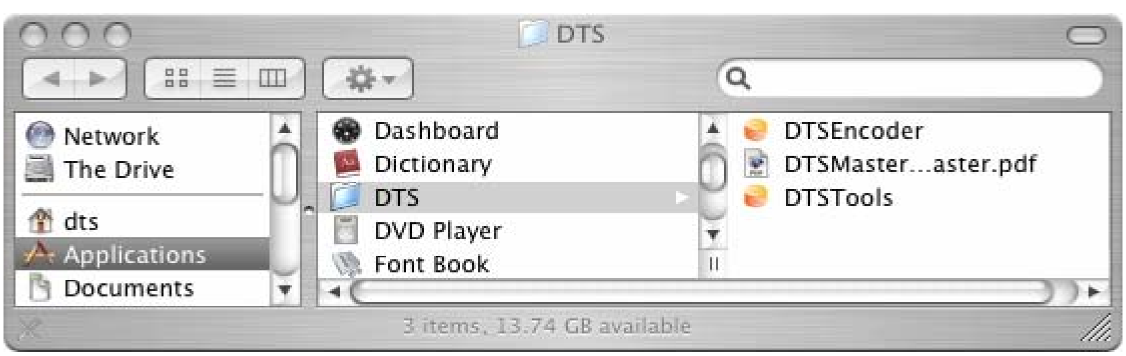


Figure 4-2 Macintosh Remove

5. Common Terms and Abbreviations

The terms and abbreviations found in Table 5-1 are commonly used throughout this manual.

Table 5-1 Common Terms and Abbreviations

Term or Abbreviation	Meaning
CA	Coherent Acoustics
CBR	Constant Bit Rate - used in the DTS-HD core substream, maximum value 1.5 Mbps
Core Substream	The portion of the DTS-HD stream that contains the backwards compatible component that can be decoded by legacy DTS decoders.
Cs	Center Surround
Dialog Normalization	Sometimes known as the 'reference offset' value. Instructs the decoder to reduce the level of its outputs by the amount specified within the bit stream.
dBFS	"**D**eci**B**els [relative to] **F**ull **S**cale,:
DTS-HD LBR	DTS-HD Low Bit Rate technology used with Secondary and Sub-Audio
ES	Extended Surround used in 6.x ES Matrix channel layouts
Extension Substream	The portion of the DTS-HD stream that contains the VBR lossless extension or the CBR High Resolution extension
Fs	Sampling Frequency - typically in Hz or kHz
GUI	Graphical User Interface
HD	High Definition
INF	Infinity
kbps	Bit rate in kilobits per second
LBR	Low Bit Rate
LPCM	Linear Pulse Code Modulation
LeqA	**L**evel **EQ**uivalent **A** [-weighted]
Ls	Left Surround
MAS	DTS-HD Master Audio Suite
Mbps	Bit rate in megabits per second
Rs	Right Surround
SAS	DTS Surround Audio Suite
TC	Timecode
VBR	Variable bit rate – used in the DTS-HD extension substream

6. Graphical User Interface Overview

The DTS-HD Master Audio Suite software provides a graphical user interface (GUI) designed from the ground up for ease of use with dynamic parameter settings. Throughout the user interface, certain controls will be restricted, activated or de-activated depending on the decisions and selections that are made in the various drop-down menu items. The chart in Table 6-1 shows the behavioral characteristics of the user interface controls.

Table 6-1 User Interface Element Descriptions

User Interface Element	Description
	Input text field; may have double-click capability to access a file browser window for selecting input files. ☑ **Note**: Tabbing out of an input text field may not 'commit' the text that is changed. It is highly recommend to use the 'Enter' key on the keyboard to ensure that the text that is entered is saved.
	Read-only text field loaded with text.
 or  or 	Radio or push button selected.
 or  or 	Radio or push button not selected but selectable.
	Push button not selected and not selectable.

7. DTS-HD Master Audio Suite Encoder

When the DTS-HD Master Audio Suite encoder application is launched, the splash screen in Figure 7-1, will be displayed while the application is being initialized. At the completion of the initialization phase, the main window will be displayed.

Figure 7-1 DTS Digital Entertainment Splash Screen

The main window consists of an easy-to-use interface with all of the required selector menus and input fields for creating an encoded stream in a single session. The user interface is depicted in Figure 7-2

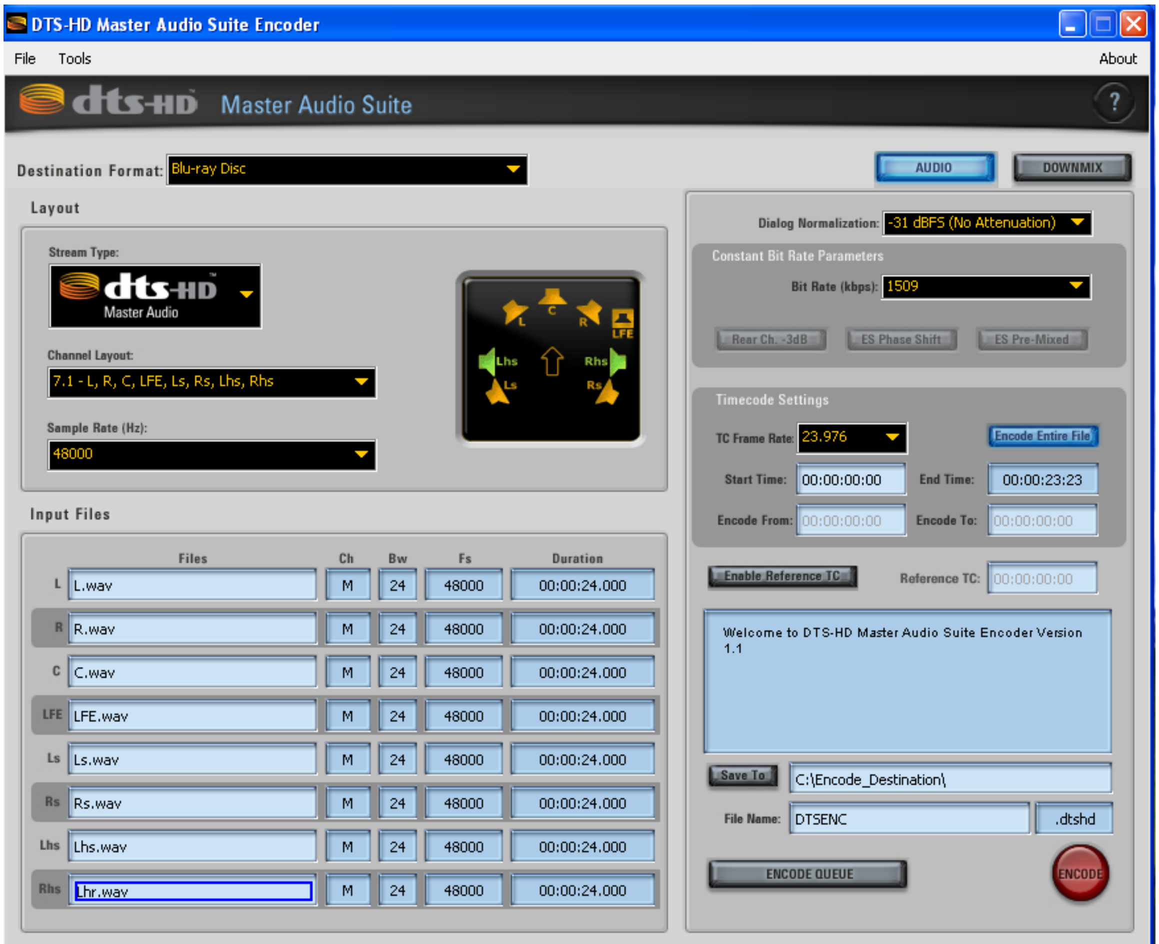


Figure 7-2. Master Audio Panel

Through a series of drop-down menus on the audio panel, the user can quickly specify the Destination Format, Stream Type, Channel Layout, Sample Rate and Bit Rate. Having all of the required data on one screen, and with drag-and-drop support for loading the input source files, will enable the user to efficiently create a DTS encoded stream.

The  in the upper right-hand side of the interface, when pressed, will launch a browser window displaying the on-line User's Guide where all of the details about the functionality of this product can be found.

The About Button in the upper right-hand corner, when pressed will display a window showing specific information about the encoder such as build version and revision number, build date, and email contact information for customer support. Figure 7-3 provides an example of the splash screen

Figure 7-3 About Splash Screen

7.1. Audio Panel

The Audio Panel (see Figure 7-2) was designed to facilitate a top-down, left-to-right workflow.

When the Audio Panel is active, as indicated by the 

control being highlighted, the user has the option of selecting the appropriate settings from various drop-down menus to create the desired encode.

The typical encoding process workflow would be as follows:

1. Select the Destination Format
2. Select the Stream Type
3. Select the Channel Layout
4. Select the Sample Rate
5. Input, either by drag-and-drop or by File Browser selection, the required source material in the Input Files Section
6. Specify the dialog normalization setting (defaulted to -31 dBFS which is no attenuation)
7. Select the desired Bit Rate. Note that for DTS-HD Master Audioencodes that the Bit Rates selector specifies the Core (substream) backwards compatible constant bit rate. In the DTS-HD High Resolution case, the Bit Rates selector specifies the total bit rate for this stream type with a 1.5 Mbps @ 48 kHz Core (substream).
8. Specify any frame rate, start/end reference time information if applicable.

9. Specify the destination folder and optional filename (default filename is DTSENC)
10. Press Encode button.

If downmix processing is desired, selecting the 

control will enable the user to modify the default downmix parameters for the selected encode. Default downmix parameters for 5.x, 6.x and 7.x channel layouts are automatically selected prior to encode. If the default values require alteration prior to encode, the user may change these settings by activating the downmix control page.

The following sections describe the input parameters for each section of the Audio Panel.

7.1.1. Destination Format

The Destination Format menu allows the user to select a specific media type for an encode. Figure 7-4 identifies the destination formats supported in the DTS-HD Master Audio Suite Encoder. When the destination format is selected, other menu items on the Audio Panel are updated to reflect the specific options that can be further selected.

☑ **Note:** For DTS Surround Audio, only DVD Destination Format will be selectable.

Figure 7-4 Destination Formats

7.1.2. Layout

This section of the Audio panel, (as shown in Figure 7-5), provides the user with several menus to specify the stream type, channel layout and sample rate. The speaker layout diagram on the right hand side of the Layout section changes dynamically to reflect the selected channel layout. The arrow at the center of the speaker layout diagram indicates the listener's head orientation in the output sound field. Speakers annotated in green indicate that the speaker location is in a different vertical plane, typically in an overhead or height speaker placement.

Figure 7-5 Layout Section

7.1.2.1. Stream Type

The Stream Types drop-down menu contains a list of all of the DTS stream types. The respective product logos for each stream type help guide the user for easy selection. The stream types are tightly coupled to the selected destination format, thus preventing the selection of DTS stream types that are not supported by the respective specifications of the destination format. For example, whenever a destination format is selected (e.g. DVD), the stream type menu will be updated dynamically to indicate only the stream type(s) which the destination format is able to support. Table 7-1 defines all of the DTS stream types that are currently supported.

Table 7-1 Descriptions

Stream Type (logo)	Product Description
dts-HD Master Audio	DTS-HD Master Audio, up to 7.1 channels of variable bit rate lossless compression, sampling rates of 48 kHz, 96 kHz and 192 kHz, with backwards compatible, constant bit rate substream. Only available for Blu-ray Disc or HD DVD destination formats. For 192 kHz source material, HD DVD supports up to 2.0 channels and Blu-ray supports up to 5.1 channels.
dts-HD High Resolution Audio	DTS-HD High Resolution Audio, up to 7.1 channels lossy compression, bit rates up to 6 Mbps for Blu-ray and up to 3 Mbps for HD DVD. Only available in Blu-ray Disc or HD DVD destination formats.
dts Digital Surround | ES	DTS-ES, up to 6.1 discrete channels at bit rates up to 1.5 Mbps. Available in all destination formats except for Blu-ray Disc Secondary Audio and HD DVD Sub-Audio
dts Digital Surround | 96/24	DTS 96/24, up to 6.1 ES Matrixed channels, at bit rates up to 1.5 Mbps from source files with a sample rate of 96kHz. Available in all destination formats except for HD DVD Sub-Audio
dts Digital Surround	DTS Digital Surround up to 6.1 ES Matrixed channels, at bit rates up to 1.5 Mbps for DVD. Available with all destination formats. When used with HD DVD Sub-Audio only valid at up to 2.0 channels.
dts-HD Low Bit Rate	DTS-HD Low Bit Rate. For HD DVD Sub-Audio up to 2.0 channels.

☑ **Note:** For DTS Surround Audio, only **DTS Digital Surround | ES**, **DTS Digital Surround | 9624** and **DTS Digital Surround** will be selectable.

7.1.2.2. Channel Layout

The Channel Layout drop-down menu will change depending on the destination format and stream type selections. This allows the user to access only those channel layouts which can be properly encoded under the current selections. The list of all possible channel layouts is given in Table 7-2. If the destination format and stream type are changed such that a selected channel layout is not valid for that particular combination, the first valid channel layout found in the list corresponding to the selected destination format and stream type combination will be selected. Whenever a channel layout is selected, the speaker layout diagram is updated with the visual representation of the selected layout.

Note: At the time of this release, channel layouts for Blu-ray Secondary Audio are not supported

Table 7-2 Channel Layouts

Primary	Sub-Audio Coherent Acoustics	Sub-Audio (LBR)
7.1 – L, R, C, LFE, Lss, Rss, Lsr, Rsr		
7.1 – L, R, C, LFE, Ls, Rs, Lh, Rh		
7.1 – L, R, C, LFE, Ls, Rs, Lhs, Rhs		
7.1 – L, R, C, LFE, Ls, Rs, Lsr, Rsr		
7.1 – L, R, C, LFE, Ls, Rs, Cs, Ch		
7.1 – L, R, C, LFE, Ls, Rs, Cs, Oh		
7.1 – L, R, C, LFE, Ls, Rs, Lw, Rw		
7.0 – L, R, C, Lss, Rss, Lsr, Rsr		
6.1 ES Discrete – L, R, C, LFE, Ls, Rs, Cs		
6.1 ES Matrix – L, R, C, LFE, Ls, Rs, Cs		
6.0 ES Matrix– L, R, C, Ls, Rs, Cs		
5.1 – L, R, C, LFE, Ls, Rs		
5.0 – L, R, C, Ls, Rs		
4.1 – L, C, R, S, LFE		
4.1 – L, R, Ls, Rs, LFE		
4.0 – L, R, Ls, Rs		
4.0 – L, C, R, S		
3.1 – L, C, R, LFE		
3.1 – L, R, S, LFE		
3.0 – L, C, R		
3.0 – L, R, S		
2.1 – L, R, LFE		
2.0 – L, R	2.0 – Stereo (CA)	2.0 – Stereo (LBR)
2.0 – Lt/Rt	2.0 – Lt/Rt (CA)	
1.0– C (Mono)	1.0 – C (CA)	1.0 – C (LBR)

☑ **Note:** DVD destination format is limited to a maximum of 6.1 channels. HD DVD Sub Audio (both DTS Digital Surround® and DTS-HD LBR) are limited to 2.0 channels.

The abbreviations used in Table 7-2 are defined in Table 7-3 below.

Table 7-3 Channel Abbreviations

Abbreviation	Description
L	Left
R	Right
C	Center
LFE	Low Frequency Effects
Ls	Left Surround
Rs	Right Surround
Cs	Center Surround
ss	Surround Side
sr	Surround Rear
h	Height
S	Surround
Oh	Overhead Channel
w	Surround wide (+/-60^0 from center)
Lt/Rt	Left Total/Right Total

7.1.2.3. Sample Rate

The Sample Rate drop-down menu contains all of the available sampling rates in Hz based on the selected stream type. These values vary according to these selections as depicted in Table 7-4

Table 7-4 Stream Type Sample Rates

Stream Type (logo)	Available Sample Rates
dts-HD Master Audio	48, 96 and 192 kHz
dts-HD High Resolution Audio	48 and 96 kHz
dts Digital Surround | ES	48 kHz
dts Digital Surround | 96/24	96 kHz only
dts Digital Surround	48 kHz
dts-HD Low Bit Rate	12, 24, and 48 kHz

7.1.3. Input Files

The Input Files that are required for the encoding process must be loaded into this section of the user interface (see Figure 7-6 and Figure 7-7). Input files may be loaded either by drag-and-drop into the appropriate text field or by double-clicking on the text field and selecting the source files from the file browser window. All active channels should be populated with mono or stereo audio input source files. The selected channel layout determines which audio input fields must be populated. Supported audio input file types include LPCM wave and Broadcast LPCM wave (.wav) files

The labels running vertically on the left side of the Input Files window pane from top to bottom correspond to the active channels of the selected channel layout. If a channel is not active, then that row will not be selectable. The labels will change dynamically to reflect the channel layout that is selected.

The column to the right of the Files text boxes labeled 'Ch' specifies which channel of the input material will be used in the encoding processing. If a stereo input file has been entered, Left ('L') or Right ('R') will appear in the column. An 'M' will be displayed if the input source material is a mono file.

The column labeled 'Bw' specifies the bit-width of the source file. Typically this value is 16 or 24 bits.

The column labeled 'Fs' specifies the sample rate (F = frequency, s = sample) of the input material. The column labeled 'Duration' denotes the length of the input file in HH:MM:SS.MSEC where HH denotes hours, MM denotes minutes, SS denotes seconds and MSEC denotes milliseconds.

If a desired input file is incorrect, positioning the mouse over the file and clicking the right mouse button, will display a pop-up menu showing a "CLEAR" button. Selecting this button will clear out all of the values in columns (i.e. Bw, Fs, and Duration) associated with the selected file.

☑ **Note:** Input files may have varying durations. The encode process will only encode up to the shortest file duration. For example, in Figure 7-7, the input files have duration of 00:00:10.000 and 00:00:30.030. The process will create an encoded file up to 00:00:10.000 in length.

Audio Channels (Ch)

The audio channels input fields accept either mono or stereo wav files. It is highly recommended that mono file be used whenever possible. However, when stereo files are used the following rules are in effect:

a) If a stereo file is entered in first, third, fifth, or seventh input file locations (in Figure 7-6 this is shown as L, Lss, C, and Ls-r) the file's left channel will be used as the source for the selected channel while the file's right channel will be used as the source for the next channel **OR** if a stereo file is entered in second, forth, sixth, or eighth input file locations (in Figure 7-6 this is shown as R, Rss, LFE, and Rs-r) the file's right channel will be used as the source for the selected channel while the file's left channel will be used as the source for the previous channel.

Figure 7-6 Input Files Section (Stereo Case)

a) If a selected channel layout does not contain adjacent paired channels (i.e L/R, C/LFE, Lss/Rss, Ls-r/Rs-r in Figure 7-6 above), then in the singleton cases, only mono input files will be allowed. For example, in Figure 7-7, the selected channel layout (4.0 – L, C, R, S), a stereo file is valid for the L and R channels, but only mono files will be valid for C and Cs.

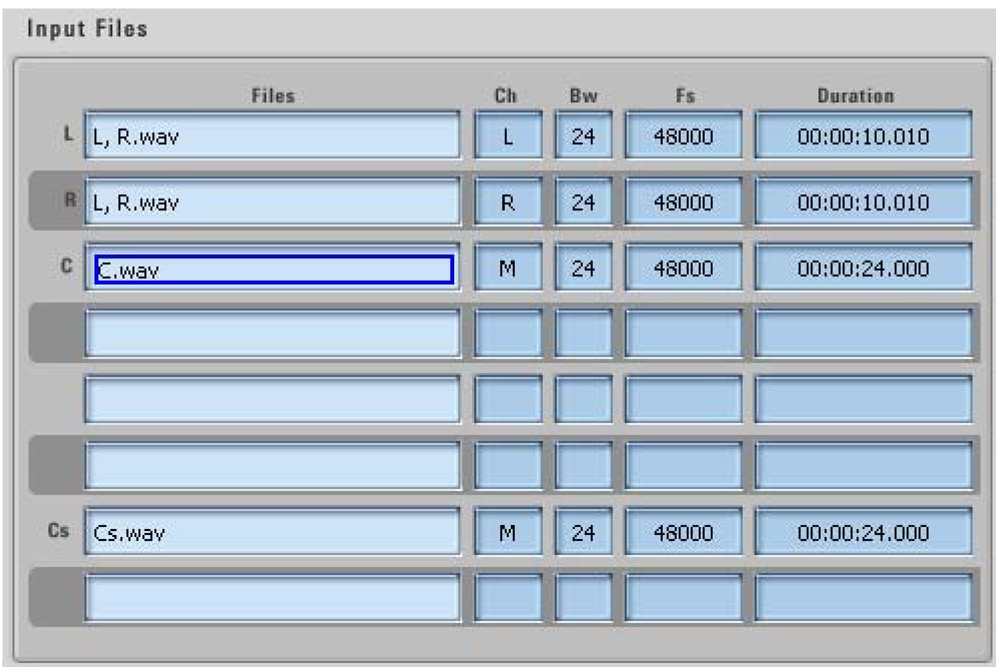


Figure 7-7 Input Files Section (Stereo and Mono Case)

Bw and Fs – Bit width Resolution and Sampling Frequency of Audio Channels

All audio files entered for any channel layout must have the same bit-width resolution (Bw) and sampling frequency (Fs).

☑ **Note:** If the sample rate of the input material does not match the selected sample rate in the Sample Rates drop-down menu, when the 'Encode" button is selected, an error message will be displayed indicating that there is a mismatch. Unless the error is corrected, the user will not be able to create an encoded file.

7.1.4. Dialog Normalization

On the upper right side of the Audio panel, a Dialog Normalization menu allows a value to be selected for the specified encode (as shown in Figure 7-8). These values range from -1 to -31 dBFS LeqA. The default value is -31 dBFS LeqA, which corresponds to no attenuation, in effect turning dialog normalization OFF. Dialog Normalization is a post-process operation performed by the DTS decoder (for a detailed discussion on Dialog Normalization, refer to Section 13).

Figure 7-8 Dialog Normalization

7.1.5. Constant Bit Rate Parameters

The Constant Bit Rate Parameters section allow the user to specify the bit rate to be used in the core substream, to select Rear Ch -3 dB attenuation, whether to perform ES phase shifting and whether the input material for a 6.1 ES Matrixed channel layout has been pre-mixed (indicated by selecting the ES Pre-Mixed button). Figure 7-9 depicts the interface for these settings.

For DTS-HD Master Audio and DTS-HD High Resolution Audio stream types, the encoded file that is created consists of a core substream and an extension substream. The core substream is that part of the DTS-HD stream that can be decoded by all current DTS Digital Surround decoders. Whenever a DTS-HD Master Audio stream is created, the selectable bit rate on the user interface applies to the core substream. For DTS-HD High Resolution Audio streams, internal to the encoder, the core substream bit rate will be set to the maximum value (typically 1.5 Mbps but varies by channel count) and the extension substream bit rate encoded will be the difference between the core substream bit rate and the value specified in the user interface.

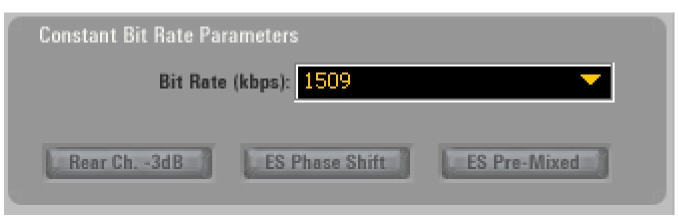


Figure 7-9 Constant Bit Rate Parameters

7.1.5.1. Supported Bit Rates

Depending on the parameters and spectral content of the input source material, DTS-HD Master Audio utilizes variable bit rate coding (VBR) for its extension substream component. As such, the bit rates referred to in this table for DTS-HD Master Audio applies only to the DTS Digital Surround® constant bit rate (CBR) core substream.

For the selectable destination format, stream type, channel layout and sample rate (Fs), the supported bit rates are identified in Table 7-5 below. The user interface will always default to the highest available bit rate for the selected destination format, stream type, channel layout and sample rate such that the default settings can create the highest quality DTS audio stream.

Table 7-5 Bit Rates for Destination Formats and Stream Types

Destination Format	Stream Type (Logo)	Bit Rates (kbps)
Blu-ray Disc Primary Audio **or** **HD DVD Main Audio**		Allows only for a variable bit rate (VBR) stream to be encoded but that stream also includes a constant bit rate DTS Digital Surround® backward-compatible core data substream. **DTS Digital Surround Core bit rates in kbps (Mono through 5.1 channels)** 192, 255, 318, 384, 447, 570, 639, 768, 960, 1152, 1344, 1509 ☑ **Note:** For channel configurations 5.1 and above, the core bit rate is limited from 768 through 1509 kbps (inclusive)

Destination Format	Stream Type (Logo)	Bit Rates (kbps)
Blu-ray Disc Primary Audio **or** **HD DVD Main Audio**	High Resolution Audio	Allows for a constant bit rate stream to be encoded which includes a 1509 kbps constant bit rate DTS Digital Surround backward-compatible core substream. **For Stereo through 7.1 ch** Bit Rates in the range of 2046 through 3018 kbps for HD DVD Bit rates in the range of 2046 through 5760 kbps for Blu-ray Disc ☑ **Note**: the 5760 kbps option is only available when files with a sampling rate of 96 kHz or higher are used as input files
Blu-ray Disc Primary Audio **or** **HD DVD Main Audio**	Digital Surround | ES	For 6.1 channels, supported constant bit rates are: 768, 960, 1152, 1344, 1509
DVD-V	Digital Surround | ES	For 6.1 discrete channels, supported constant bit rates are: 754 or 1509
DVD –V	Digital Surround | 96/24	For mono through 6.1 ES Matrix channels, supported constant bit of 1509
DVD-V	Digital Surround	Stereo through 5.1 channels layouts that may be encoded at 754 or 1509 kbps
HD DVD Sub Audio	Digital Surround	**Stereo** 192, 255, 318, 384, 447, 510 kbps **Mono** 126, 192, 255, 318, 384, 447, 510 kbps
HD DVD Sub Audio	Low Bit Rate	**Stereo** 48, 64, 96, 128 and 192 kbps **Mono** 24, 32, 48,64, 96 kbps

7.1.5.2. Rear Ch -3dB Attenuation

This option is available for 5.0 and 5.1 channel layouts for all destination formats except for Blu-ray Disc and HD DVD when using a DTS-HD Master Audio stream type. When this button is illuminated, Rear Ch -3dB the audio provided for the left surround Ls and right surround Rs channels will be attenuated by 3 dB prior to the encoding process.

7.1.5.3. ES Phase Shift

ES refers to the Extended Surround or center surround Cs channel capability of DTS Digital Surround technology. This option is only available for 6.x channel layouts. Selecting ES Phase Shift ES Phase Shift performs phase shifting of left Ls and right Rs surround channels by +/- 45 degrees prior to the addition of the center surround channel. The ES Phase Shift option can only be selected for stream types other than DTS-HD Master Audio.

- ☑ **Note:** DTS recommends enabling ES Phase Shift for 6.1 ES Discrete encodes.
- ☑ **Note:** DTS recommends disabling ES Phase Shift for 6.0 and 6.1 ES Matrix encodes.

7.1.5.4. ES Pre-Mixed

This option is only available for 6.0 or 6.1 ES Matrix channel layouts. Selecting this option ES Pre-Mixed tells the encoder that the extended surround flag should be activated in the decoder when the stream is decoded by a DTS-ES compatible decoder. It strictly means that the left surround Ls and right surround Rs input channels already contain the center surround pre-mixed signal and that the encoder shall not perform any mixing during the encoding process.

- ☑ **Note:** ES Phase Shift and ES Pre-Mixed cannot be used simultaneously.

7.1.6. Timecode Settings Section

The Timecode Settings section allows the user to specify a SMPTE (Society of Motion Picture and Television Engineers) timecode frame rate and the time frame to encode (see Figure 7-10). The default setting is to encode the entire source file. For DTS-HD encodings, the timecode settings are stored in the encoded output file.

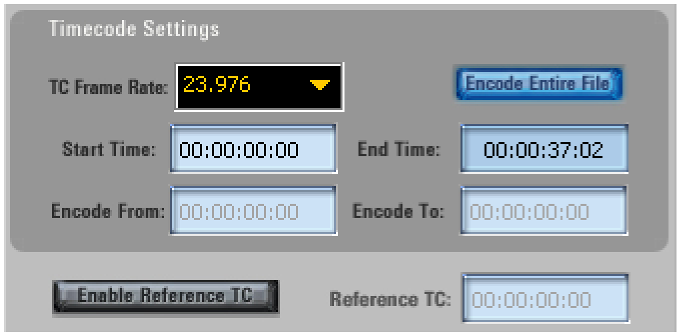


Figure 7-10 Timecode Settings Section

7.1.6.1. TC Frame Rate

The TC Frame Rate drop-down menu makes it possible to select a frame rate in frames per second (FPS). The supported frame rates include: 23.976, 24, 25, 29.97 Non-Drop, 29.97 DROP, 30 and 30 DROP.

7.1.6.2. Encode Entire File

When illuminated (default setting) 

, the input files will be encoded in their entirety and the **Encode From** and **Encode To** text fields will be disabled.

When not illuminated, 

the **Encode From** and **Encode To** input text fields will be enabled allowing the user to specify the desired timecode ranges to encode. If these fields are modified, encoding will only be performed for the time span specified in these fields.

7.1.6.3. Start Time

The **Start Time** input text field allows the user to specify the start time in hours, minutes, seconds, and frames.

7.1.6.4. End Time

The **End Time** is read-only field and specifies the end time relative to the **Start Time** duration of the input files. The value shown represents the end time of the shortest source file of the input files specified.

7.1.6.5. Encode From

The **Encode From** input text field allows the user to specify a time interval to encode, constrained by the start/end time of the source material. This field is active only when the Encode Entire File is in the off state 

. This allows the user to create a smaller encode relative to the entire source file length. The input time must be in hours, minutes, seconds, and frames. Additionally, this input time must be <u>greater than</u> the start time and <u>less than</u> both the End Time and the Encode To time.

7.1.6.6. Encode To

The **Encode To** input text field allows the user to specify a time interval to encode, constrained by the start/end time of the source material. This field is active only when the Encode Entire File is in the off state 

. This allows the user to create a smaller encode relative to the entire source file length. The input Encode To time must be in hours, minutes, seconds, and frames. Additionally, this input time must be greater than the Start Time and greater than the Encode From time but less than the End Time.

7.1.6.7. Retain Residual DTS Frame Data

The encoder application provides options for creating either an encode that ends exactly on a SMPTE timecode frame, or an encode that retains all of the extra samples from the source audio that extends past the last specified SMPTE timecode frame. These samples are contained in DTS Frames. With  enabled, these extra samples contained within the DTS Frames (i.e. the Residual DTS Frame Data) are automatically included/retained in the output '.dtshd' file ensuring that the entire source audio file is encoded. If a user only wishes to encode a portion of the source material they can by de-activating the  function giving them the options of **Encode From** and **Encode To**. **Encode To** will halt the encoding exactly at the specified SMPTE timecode frame. No residual DTS frames would be retained. However, if the **Encode To** timecode is EQUAL to the **End Time,** when the Encode button is clicked the application will display a prompt, as shown in Figure 7-11 Residual Frame Data Acknowledge Prompt, asking if residual data should be retained. Select "Yes" to retain residual data. Select "No" to discard the residual data and stop encoding at the specified SMPTE timecode frame.

Figure 7-11 Residual Frame Data Acknowledge Prompt

☑ **Note:** This option is closely linked with the DTS-HD StreamTools **Join** Function. For more information about reasons for using **Retain Residual DTS Frame Data** and DTS Frames, refer to Join Tool in section 10.2.

7.1.6.8. Enable Reference Time

The encoder application supports a mechanism by which two or more encoded files can be joined to create a single encoded file while maintaining audio to timecode synchronization. In order for this to occur, all encoded files that will be joined must have identical reference times (i.e. the DTS frames from the subsequent streams are aligned relative to the first). The Enable Reference Time feature provides the user with a method of creating a custom reference time in order to join one file to another using the Join Tool in the DTS StreamTools Application Suite while maintaining audio to timecode synchronization.

When the Enable Reference TC button is activated,  the Reference TC text field becomes editable allowing the user to enter a custom reference timecode. The specified value must match the reference time of the encode to which it will be joined. The Join Tool will only allow encodes to be joined if their reference times are identical. By default, Enable Reference TC is not activated. Creating encodes without a user specified reference time will render the encode's reference time equal to its start time. *(see* Tools *section* Join Tool *for more information on Reference TC)*

☑ **Caution:** DTS HD encoded streams who's reference time **do not** equal it's start time may have a delay that is inappropriate for disc authoring. Encoding with a user specified reference time is only necessary for use with the Join operation.

7.1.6.9. Timecode error processing

The user interface contains rudimentary error processing whenever and invalid timecode is specified. Only numeric values can be entered into the timecode field. Any character other than a numeric value will result in the cursor staying at its current location.

If the 'NumLock' key on the keyboard is in the locked state and the keyboard has a numeric keypad, entering the timecode start time and/or timecode end time can be done quite easily. The timecode interface is structured such that the cursor will 'jump' over the colon or semi-colon place holders. These characters cannot be deleted. If a timecode value that is entered results in an error condition, the timecode field will change to red and the keyboard will beep. Error processing takes place at the expected levels (i.e. hours > 23, minutes > 59 and seconds > 59). Timecode validation occurs when the encode button is selected.

Figure 7-12 and Figure 7-13 show examples where the hours and minutes field contained an erroneous input.

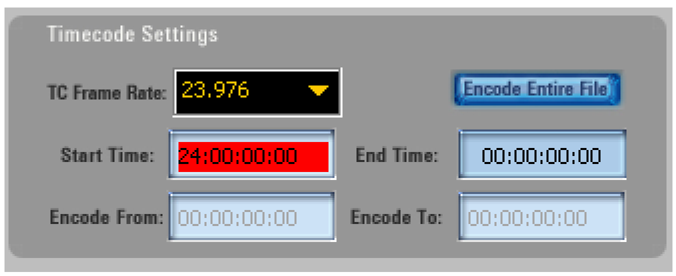


Figure 7-12 Timecode Error (hours)

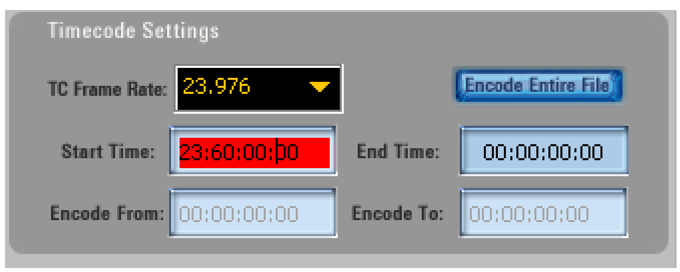


Figure 7-13 Timecode Error (minutes)

7.1.7. Diagnostics and Output Section

The lower right portion of the Audio panel, as shown Figure 7-14, provides feedback on the operation of the user interface as well as any error messages that are generated by the encoder. It also provides the control mechanism to submit an encode job and to launch the encode queue to determine the status of each submitted job.

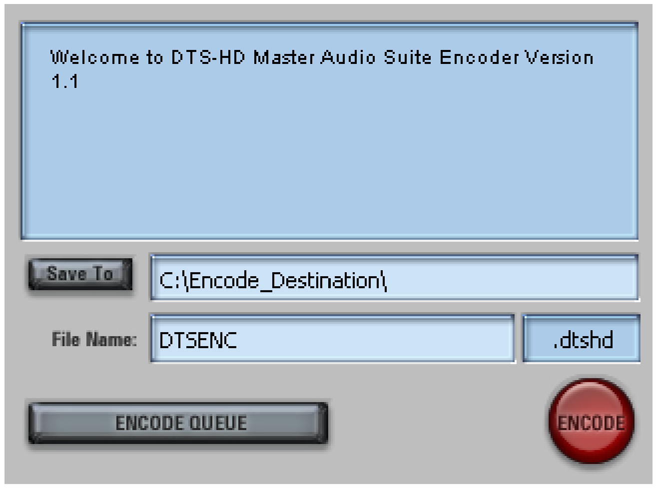


Figure 7-14 Diagnostics and Output

The Save To button, when selected will display a file browser window allowing the user to specify a directory for the new encoded file to be saved. A filename may be specified in the file name text field. The default filename is “DTSENC” if a filename is not supplied. The extension to the right of the file name text field will be appended to the specified file. If the extension that is selected appears in the saved file name, the user interface will automatically remove it and will automatically append the extension displayed in the extension field. The user will be warned whenever an existing encoded file will be overwritten prior to encoding. If an output filename is specified, the bit-stream generated by the encoder will be stored in this file and it will be written to disk with the “.dtshd” filename extension for media types other than DVD Video. For the DVD destination format the filename extension will be “.cpt”.

The encode button will be in the inactive state ENCODE until all of the required input files have been inserted. For example, all of the **active channels** (i.e. those that contain labels corresponding to the selected channel layout) must have a specified input file. When all of the input files have been entered, the encode button will change to the active state ENCODE. When pressed, if a path or filename is not included in the Save To text field, an error message will be displayed. Otherwise, if all the parameter on the Audio panel are set appropriately, the job will be sent to the encode queue (see section 7.4). If the encode queue window is not active, when the ENCODE button is pressed, the encode queue window will be automatically displayed. Alternatively, he Encode Queue window may be activated by pressing the ENCODE QUEUE button.

If there is an error with the encoder, a message will appear in the status window (shown circled in red in Figure 7-15). If the error message contains a numerical (hexadecimal) value, section 11 (Encoder Error Codes) contains the list of error codes explaining the error and information on how to further isolate the problem with the requested encode. In this example, the configuration file was in error or a license file was not found.

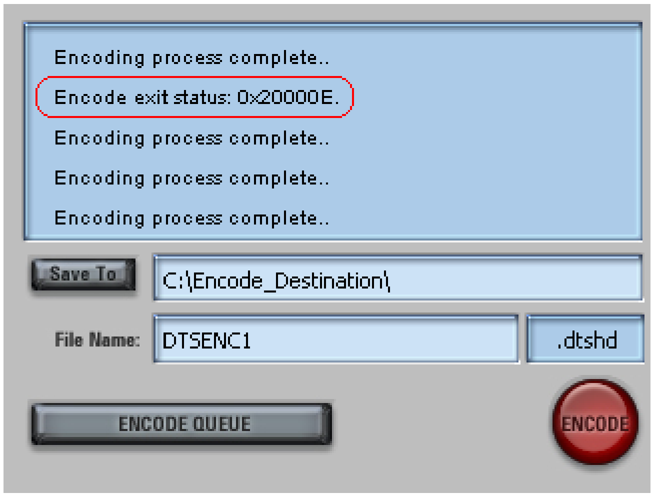


Figure 7-15 Encode Error Condition

Each job that is submitted will generate a unique log file that can be viewed with any text editor installed on the computer. This ASCII file will contain key parameters used for the selected encode. The file name of the document will be the same name as the encoded file with ".log" as its filename extension. The parameters will include File Start Time, File End Time, Sample Rate, Channel Layout, ES Flag, Rear Channel Attenuation, Dialog Normalization Setting, Reference Timecode, and Downmix Coefficients. Section 14 depicts an example of the log file output.

7.2. Downmix Panel

When the Audio panel is displayed, selecting the 

button will activate the Downmix Panel as shown in Figure 7-16, only when the selected channel layout is one of 5.x, 6.x or 7.x channels. There are two levels of downmixes on this panel. The top section is reserved for those cases where a downmix to 5.1 channels is desired while the lower section is reserved for those cases where a downmix to stereo (2.0 channels) is desired. Downmixes to 5.1 channels can only be performed when a 7.x channel layout is selected (6.1 ES Discrete will use the built-in Legacy downmix). Downmixes to stereo can only be performed when the selected channel layouts contain at least 5.1-channels.

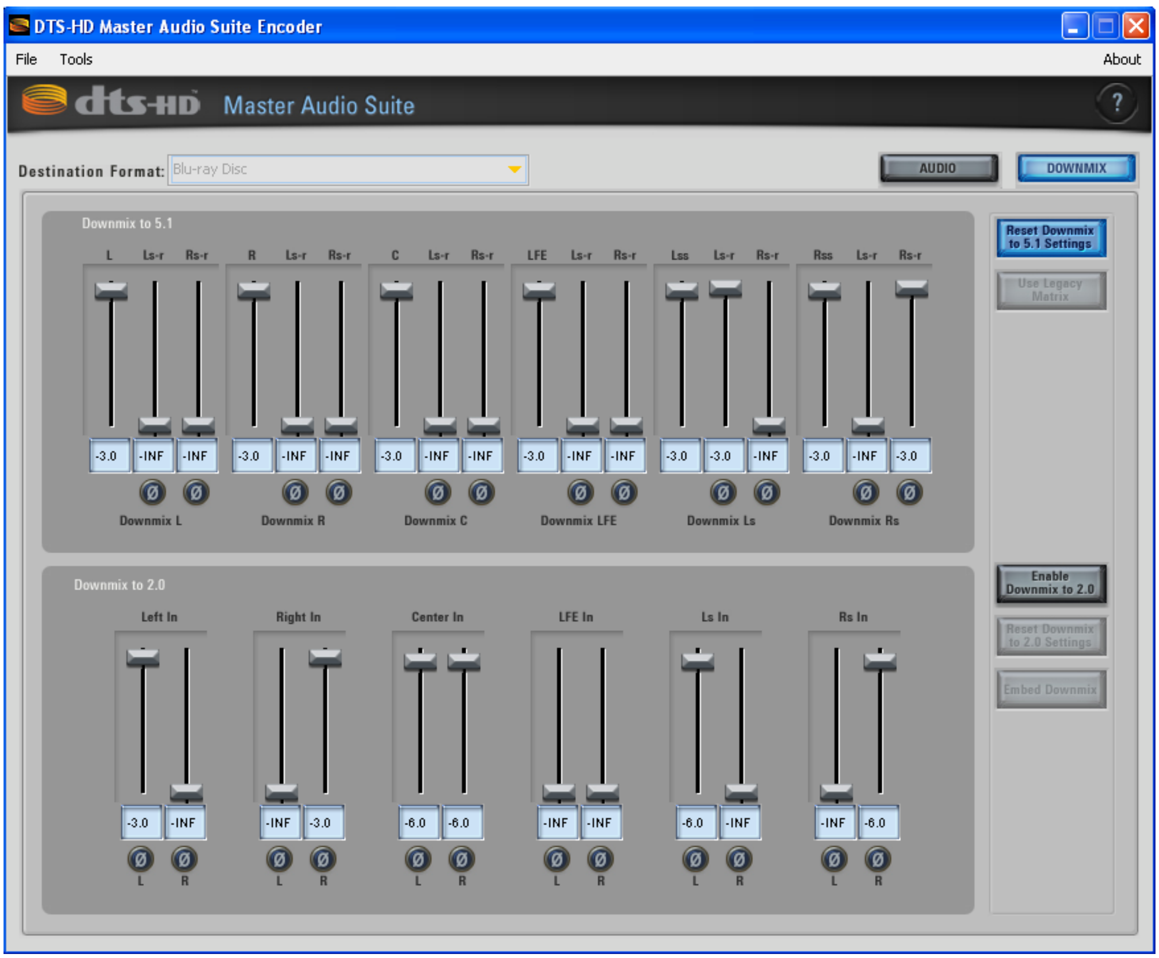


Figure 7-16.Downmix Panel

7.2.1. Downmix to 5.1

The Downmix to 5.1 section of this panel allows the user to specify the appropriate parameters for the dowmnix as shown in Figure 7-17. This section will only be active if the selected channel layouts have 7.0 or 7.1 channels.

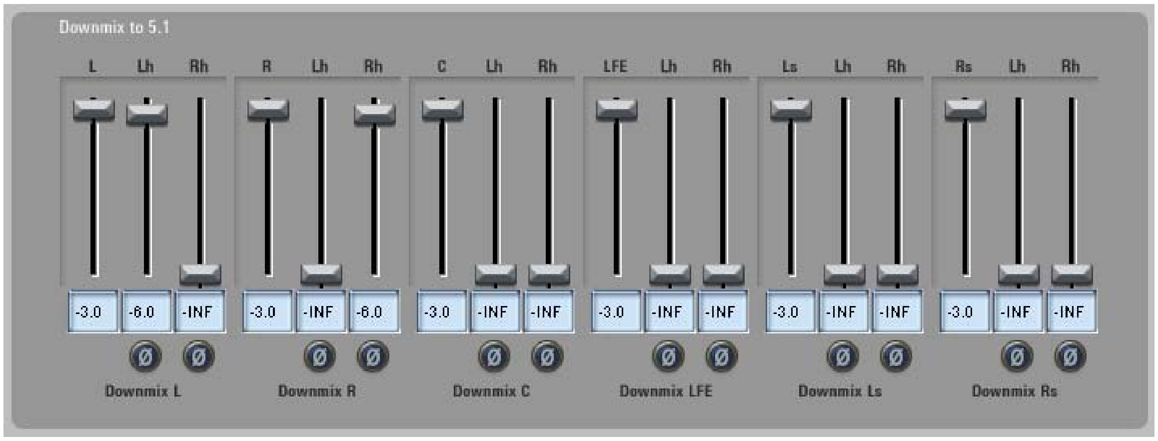


Figure 7-17 Downmix to 5.1

The channel layout selected on the Audio panel will dynamically update the headers in the columns of the Downmix to 5.1 faders. The first fader (i.e. L, R, C, LFE, Ls and Rs) in each control set, known as Scaling Coefficients, shows the scaling value for each of those channels. The values in these fields, at the bottom of each column, specify the channel's contribution to the output mix. The values range from 0.0 to 40.0 (0 to -40 dBFS) inclusive. The mixing coefficients for the extra channels (i.e. Lh and Rh in Figure 7-17), known as Downmix Coefficients, range from 0 to -60 (0 to -60 dBFS) inclusive or INF. INF implies that there is no contribution to the channel (INF = Infinity). The labels on the extra channels will update dynamically depending on the channel layout that is selected on the Audio panel.

The other two columns in each control area are the surround channels (i.e. the two extra channels in the 7.1 channel layout case). The values in these fields specify the supplemental channels' contributions to the output mix of the L, R, C, LFE, Ls and Rs channels).

When the phase shift button is activated,  the corresponding channel above it will be phase-reversed (180 degree phase inversion) before the channel is mixed into the output mix. The downmix levels can be updated either by clicking and dragging on their vertical faders or by entering the desired downmix value in the bottom input text fields. If the entered value is outside the minimum or maximum values, the fader will be positioned to the nearest value. For example, if the entered value is greater than 0.0, the fader would be positioned to 0.0 and the text field would be updated to 0.0. If the entered value is less than -40.0, the fader position would be located at the minimum value and the text field would indicate "INF".

☑ **Note:** If the faders in the Downmix to 5.1 sections are altered and the user decides to change the selected channel layout on the Audio panel, when the Downmix panel is subsequently displayed, the faders will be changed to the default values for the selected channel layout. The faders will also change to the default setting for the selected channel layout when the user selects the 

button.

When a 6.x channel layout is selected, the second extra channel (i.e. the third fader in each group) will not be selectable as it is not required for the selected channel layout. Furthermore, when using the 6.1 ES Discrete channel layout, only the 

is valid. If this button is not activated for the 6.1 ES Discrete channel layout, an error message will be displayed informing the user to activate this button.

- ☑ **Note:** When selecting 5.1-channel downmix values for all stream types except DTS-HD Master Audio, the L, R, C, LFE, Ls and Rs channels must have the same scaling value. In the DTS-HD Master Audiostream type case, for a 5.1 channel downmix, the scaling value for these channels can be different.
- ☑ **Note:** If the selected channel layout is 5.0, 6.0 or 7.0, the faders under LFE will be deactivated as this channel is not included in the selected channel layout

7.2.2. Downmix to 2.0

The Downmix to Stereo section of this panel allows the user to specify the appropriate parameters for the downmix as shown in Figure 7-18. This section will only be active if the selected channel layout has 5.1 channels.

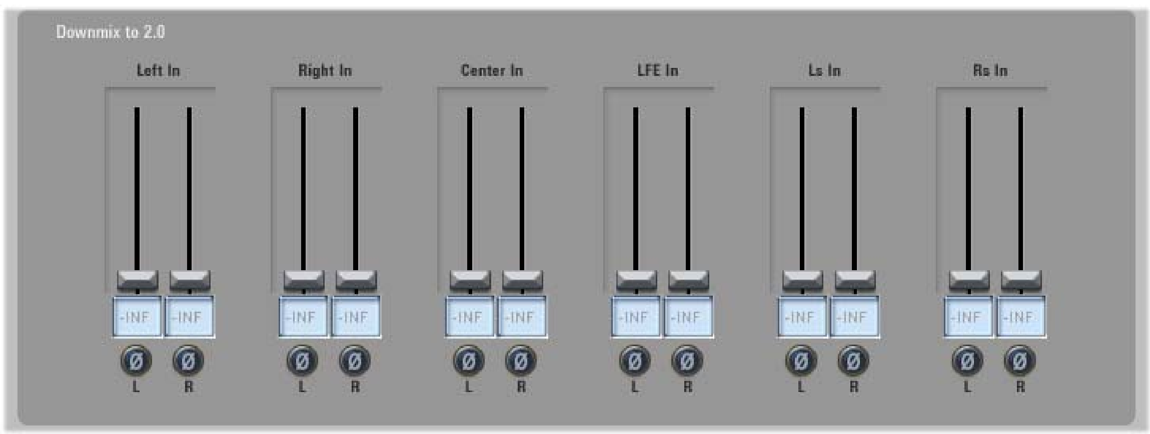


Figure 7-18.Downmix to Stereo Panel

The values specified for the L, R, C, LFE, Ls and Rs denote their respective contributions to the output mix for the Left and Right stereo channels.

When the phase shift button is activated,  the corresponding channel above it will be phase-reversed (180 degree phase inversion) before the channel is mixed into the output mix. The downmix levels can be update either by clicking and dragging on their faders or by entering the desired downmix value in the bottom input text fields. If the entered value is outside the minimum or maximum values, the fader will be positioned to the nearest value. For example, if the entered value is greater than 0.0, the fader would be positioned to 0.0 and the text field would be updated to 0.0. If the entered value is less than -40.0, the fader position would be located at the minimum value and the text field would indicate “INF”.

- ☑ **Note:** If the selected channel layout is 5.0, 6.0 or 7.0, the faders under “LFE In” will not be deactivated as this channel is not included in the selected channel layout

7.2.3. Downmix Processing Options

The processing options for downmixing are located to the right of the faders and are controlled through a series of buttons as described in Table 7-6. If the faders for a particular channel layout are modified, if the channel layout is subsequently changed, the faders will remain in their current positions. The faders will not reset simply because a different channel layout was specified. In the initial state, the downmix settings for the 5.1 channel downmix correspond to the L, R, C, LFE, Lss, Rss, Lsr, Rsr 7.1 channel layout. If this is not the desired layout, then pressing the 

will reset the faders to the default downmix settings for the selected channel layout.

Table 7-6 Downmix Buttons

Downmix Button States	Description
	When this button is pressed, the settings for the selected 7.x or 6.x channel layout will reset to their initial factory default states.
	If this button is illuminated, then the encoder will use the built-in legacy downmix coefficients at encode time.
	If this button is illuminated, the user can update the parameters in the fields on the Downmix to 2.0 portion of the Downmix Panel.
	If this button is not illuminated, it is not possible to access the Downmix to 2.0 fields or to specify values for the faders.
	If this button is illuminated, then the Left and Right channels are replaced by a mixing of all the original 5.1 channels based on the coefficients supplied in the interface.
	If this button is not illuminated, then all 5.1 channels remain unaltered. Note: It is not possible to illuminate this button unless the Enable Downmix to 2.0 button is illuminated Embedded downmix is useful for playback systems that lack the processing power necessary to perform a downmix. It is suggested to leave this option unchecked.
	When this button is pressed, the settings for the Downmix to 2.0 section will be reset to their initial factory default states.

7.3. Discussion on 6.1 ES Matrix Processing

DTS-HD Master AudioEncoder supports the ability to encode a sixth ‘phantom’ channel from a 5.1 channel layout. This is accomplished by selecting the 6.0 ES Matrix or 6.1 ES Matrix channel layout from the Channel Layout menu. When this occurs, the speaker illustration will update with a speaker configuration showing the phantom channel as shown in Figure 7-19.

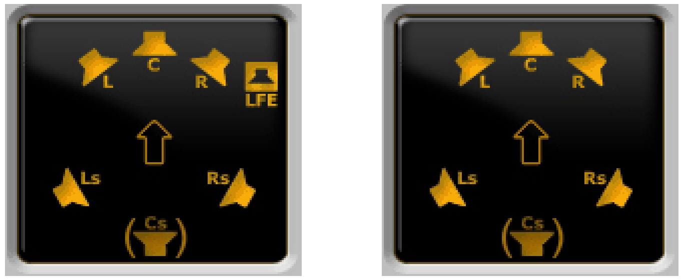


Figure 7-19 6.0 and 6.1 ES Matrix Phantom Speaker Layout

The speaker displayed as , defines the 'phantom' speaker that is matrixed by the combined signals from the left surround  and right surround  signals. This is simply a visual representation to aide the user to better understand DTS 6.1 ES Matrix processing.

When the DTS-HD Master Audio stream type and 6.x ES Matrix channel layouts are selected, the audio panel user interface will automatically enable the 

button. Under this scenario, to ensure bit-exact losslessness, the encoder is prohibited from performing the 6.x matrix processing.

Note: DTS recommends enabling ES Phase Shift for 6.1 ES Discrete encodes.

Note: DTS recommends disabling ES Phase Shift for 6.1 ES Matrix encodes.

Note: Due to an error in this version of the encoder library, when selecting a 6.x ES Matrix channel layout, it is not possible to enable the 2.0 channel downmix. This means that no user selectable stereo downmix coefficients can be embedded in the encoded stream. The decoder will automatically invoke the pre-defined downmix coefficients in this situation.

7.4. Encode Queue

The primary function of the encode queue is to allow the user to control and manage all of the jobs that have been submitted through the encoder user interface. The encode queue operates on the local machine only allowing the user to control the jobs they have created. In the case where multiple users have access to the same machine, with unique login accounts, the encode queue will behave as if all of the jobs were submitted from the same user account.

A job can have one of four states: **In-Progress, Pending, Completed,** or **Canceled.** Figure 7-20 depicts an example of the Encode Queue user interface with encode jobs in various states. The encode queue has been implemented as a first-in-first-out (FIFO) task manager and has a limit of 99 jobs.

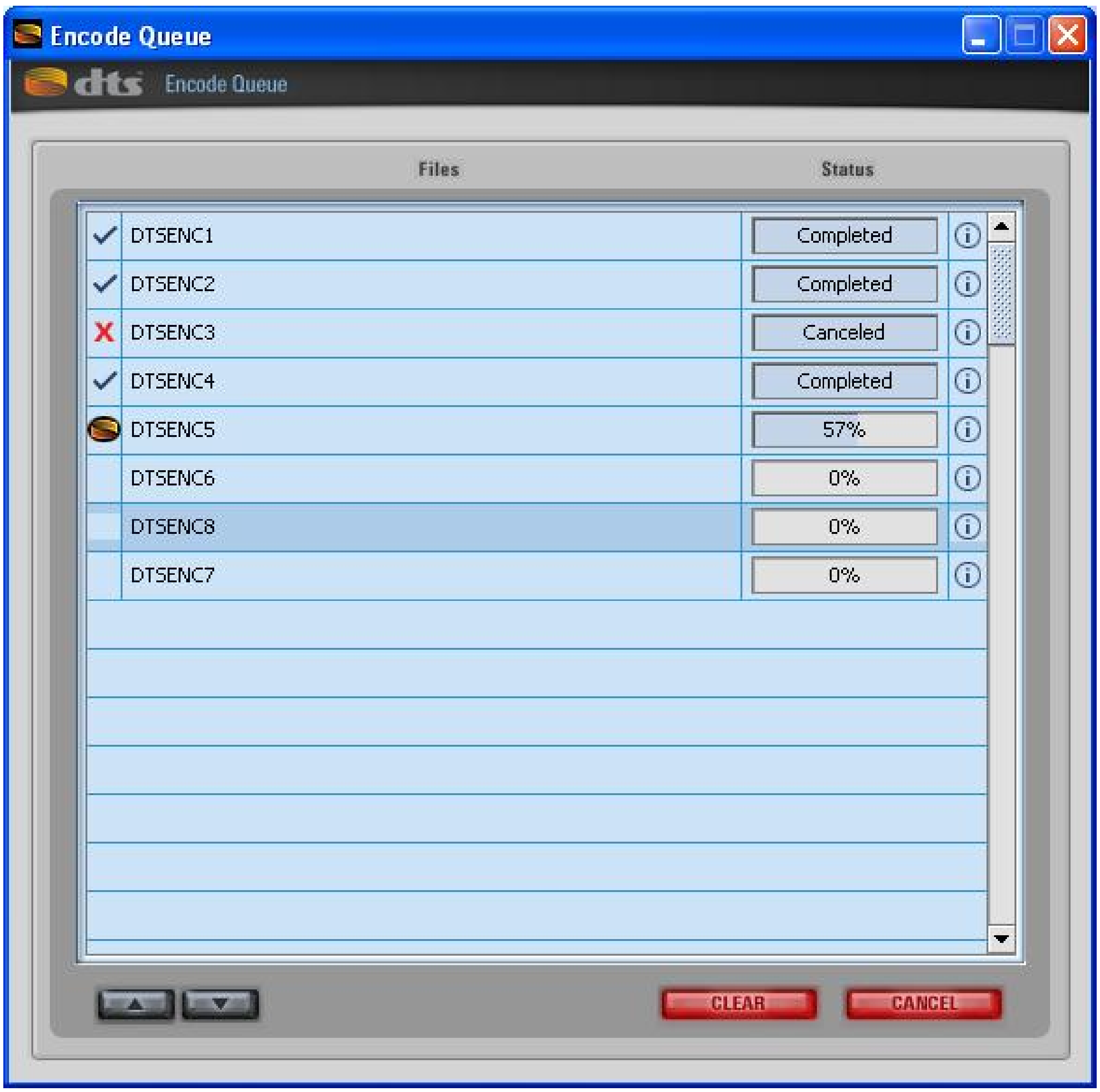
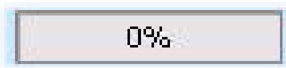


Figure 7-20 Encoder Queue

The encode queue has several user interface components that serve as key indicators for the state of any job in the queue. Table 7-7 describes each user interface component and its corresponding meaning.

Table 7-7 Encode Queue Components

User Interface Component	Description
	This symbol specifies that the job has completed
	This symbol indicates that the user has canceled the job
	This symbol, when pressed, will display the log file associated with the selected job (see Section 14 Encoder Log File Output Example).
	This symbol indicates that the job is currently being encoded with the status bar indicating the percentage complete.
	The status bar shows the progress of an active job or the status of a completed/canceled job.

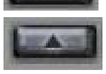
7.4.1. Encode Queue Control Buttons

The control buttons at the bottom of the page (see Figure 7-21) allow the user to manage jobs present in the Encode Queue.

Figure 7-21 Encoder Queue Control Buttons

The  buttons at the bottom of the page (see Figure 7-21) allow the user to alter the priority of any job currently in the queue that is in a '**Pending'** state.

The  button, allows the user to lower the selected jobs priority by moving it "down" in the queue.

The  button, allows the user to raise the selected jobs priority by moving it "up" in the queue. A job that is currently running (i.e. the progress bar for the job has a value greater than zero and less than 100% as seen in Figure 7-20 Encoder Queue) is a job that is currently "**In-Progress"**. The priority of jobs that are "In Progress" cannot be changed. Moving pending jobs ahead of an 'in-progress' job is not permitted. Once a job completes (or is canceled), its priority cannot be altered (i.e. it cannot be moved).

When the currently "**In-Progress"** job is selected and the CANCEL button is pressed, the job that is currently "**In-Progress"** will halt. The job that is next in the queue will start immediately.

To remove any job that has Completed, has been Canceled, or is Pending, either select the appropriate job and click the CLEAR button, or position the mouse over the appropriate job and click the right mouse button to display a pop-up menu showing Remove this Job from Queue menu item. Both methods will remove the job from the queue and will NOT remove the corresponding files stored on disc (i.e. log file, encoded file, etc.).

8. DTS Surround Audio Suite Encoder

The DTS Surround Audio Suite encoder application behaves exactly like the DTS-HD Master Audio Suite. All of the processing capabilities described in Section 6 pertain to the DTS Surround Audio Suite Encoder with the exception that the only available destination formats is DVD. Features that are not available in the DTS Surround Audio Suite include Downmix, and Timecode Settings. The reason this is the case is that the legacy encoder does not contain a stereo downmix and timecode processing is not required in the stream. The DTS-HD StreamTools cannot operate on a DVD encoded stream.

The main window consists of an easy-to-use interface with all of the required selector menus and input fields for creating an encoded stream in a single session. The user interface is depicted in Figure 8-1.

Contact your local dealer, visit http://www.dts.com, contact customer support at 1-800-959-4109 or email us at techsupport@dts.comfor details on how to upgrade to DTS-HD Master Audio Suite.

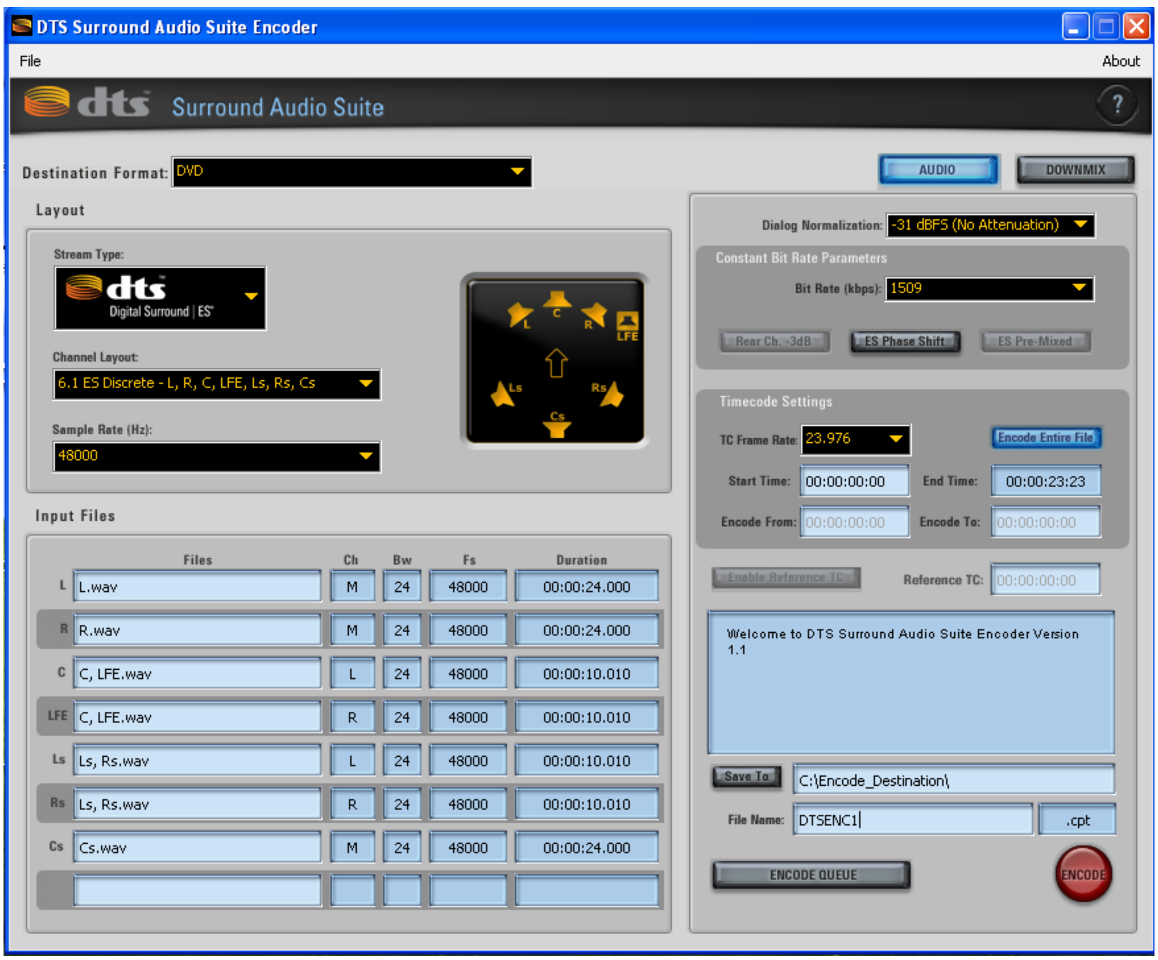


Figure 8-1. Surround Audio Panel

9. DTS-HD StreamPlayer

The DTS-HD Master Audio Suite utilizes the Neyrinck Audio SOUND.CODE DTS-HD StreamPlayer ProTools Plug-in for DTS-HD encoded file playback. Please consult the user manual included with this plug-in for operation details.

10. DTS-HD StreamTools

In addition to the DTS-HD Master Audio Suite Encoder, the DTS-HD Master Audio Suite contains DTS-HD StreamTools; a set of editing tools used to assist the user in performing specific modifications to the streams without the need for re-encoding. The tools can be launched from the DTS-HD Master Audio Suite Encoder main window (see Figure 7-2. Master Audio Panel) by clicking on the menu labeled "Stream Tools".

When the DTS StreamTools application is launched, the splash screen in Figure 10-1, is displayed while the application is being initialized.

Figure 10-1 Tools Applications Splash Screen

At the completion of the initialization phase, the main tools window will be displayed. Figure 10-3 Join Tool and Main Tools Screen shows the layout of the user interface for the DTS-HD StreamTools Application. The specified tool can be activated by selecting any one of the buttons at the top of the tools user interface.

Each tool consists of a series of independent applications that allow the user to manipulate DTS-HD audio streams. These application tools consist of:

- Join Tool
- Append Tool
- Trim Tool
- Split Tool
- Re-stripe Tool
- File Info Tool

❒ Verification Tool

Due to the non-synchronous relationship between DTS frames and SMPTE timecode, caution must be taken when using these tools to ensure:

a) Synchronization of audio to SMPTE timecode

b) Bit Exact audio edits (source audio as compared to the decoded audio) for Master Audio encodes.

c) Quality of audio transition (when using Join or Append)

All of the tools except for File Info and Verify may either alter the contents of a selected stream or will create a new stream based on the processing that is selected. In the sections that follow that describe the operational capability of DTS Tools, the legend shown in Figure 10-2 can be referenced to assist the user in a better understanding of the functional processing of the tools.

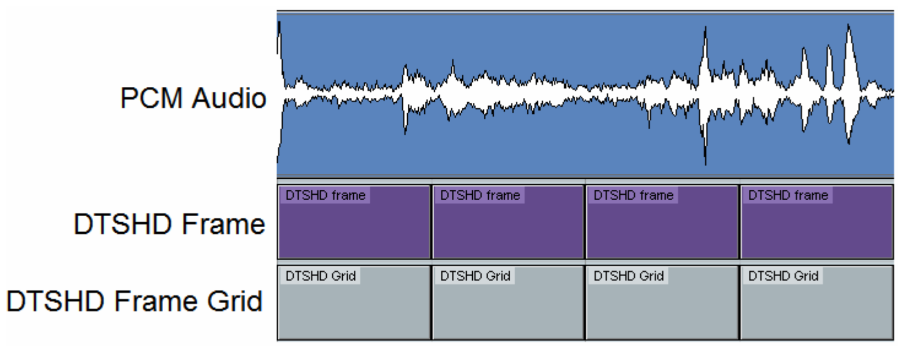


Figure 10-2 DTS Tools Legend

(not to scale)

10.1. Join Tool

The Join tool allows users to join DTS-HD encoded streams where audio to timecode synchronization is *mandatory*[1]. The Join Tool is activated by selecting the 

button, as shown in Figure 10-3 Join Tool and Main Tools Screen, and requires two DTS-HD encoded streams as its input and a destination file to save the results of the join operation. Double-click in each of the filename fields to load the files that are to be joined. The Timecode Start and Timecode End text fields show the start and end time of the input materials for each of the selected files. The Start Time and End Time text fields specify the user designated Start and End time of the DTS-HD file that will be created by the join process. Use the “Save To” button and the “File Name” fields to specify the output directory and filename of the resultant join operation. Pressing the “Process” button will initiate the join processing. Pressing the “Cancel” button will stop the running process.

[1] ***CAUTION***: *Restriping DTS-HD files will render the file’s reference time irrelevant. Consider audio to timecode synchronization lost once a file has been restriped. Audio to timecode synchronization cannot be maintained by the Join operation if the selected DTS-HD file has been restriped (see section 10.5).*

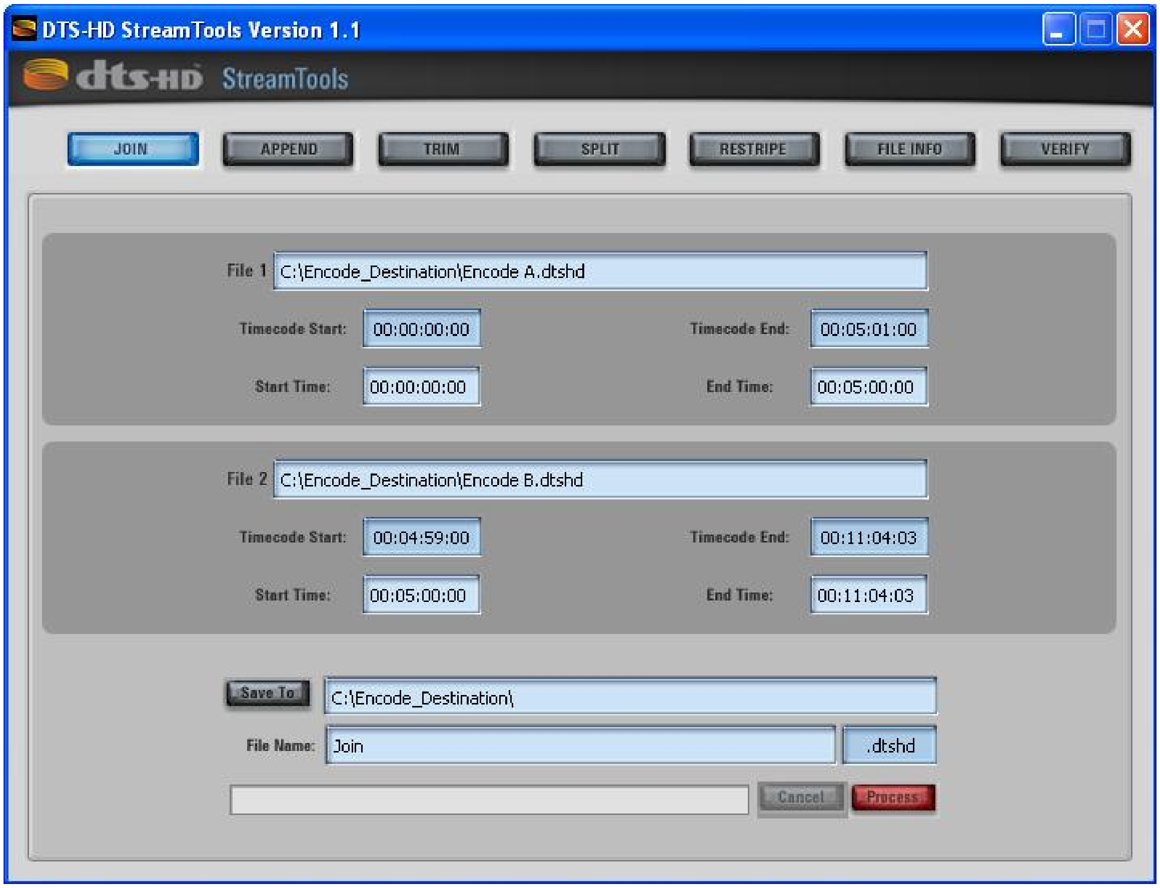


Figure 10-3 Join Tool and Main Tools Screen

There are several requirements that must be met before a join operation is permitted. These requirements are as follows:

When specifying a join time, the user specified end time of the initial DTS-HD encoded stream must equal the user specified start time of the subsequent DTS-HD encoded stream. This time is known as the "Join Time"

Join operations can be performed with or without overlapping audio regions. If overlaps are present, the audio contained in the overlapping region MUST be bit for bit identical. Overlapping regions must be at least 1 second in length and MUST be present before and after the intended join time in order for a join with overlaps to be permitted. **If the use of overlaps is not possible, it is recommended that the join time take place during a silent passage to avoid audio anomalies at the join point.**

> Overlaps are necessary for a bit-exact edit to be performed on DTS-HD Master Audioencoded file joins. The illustration in Figure 10-4 shows source audio aligned to a DTS frame grid with a user defined join time.

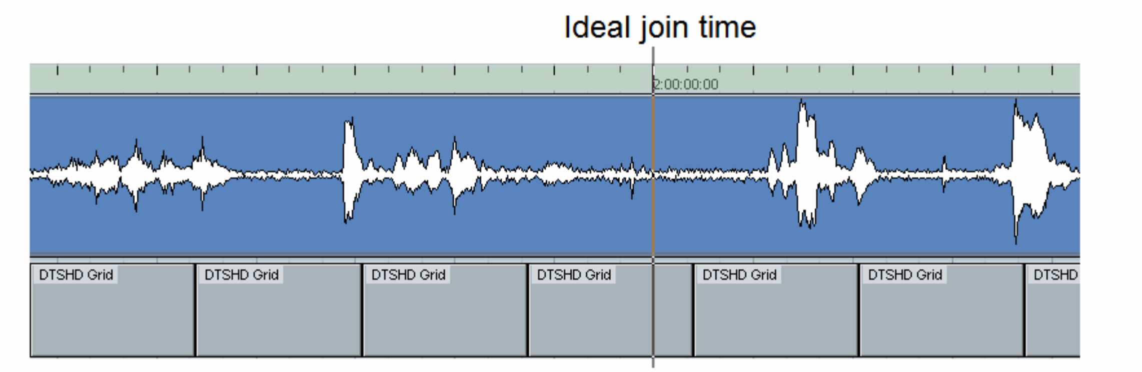


Figure 10-4 Audio Alignment to DTS Frame

If the use of overlaps is required, as shown in Figure 10-5, the source material should be divided into two pieces, using your preferred DAW (digital audio workstation), with pre- and post-roll (the overlaps described above) each containing the duplicate audio material of at least 1 second of more (seen below).

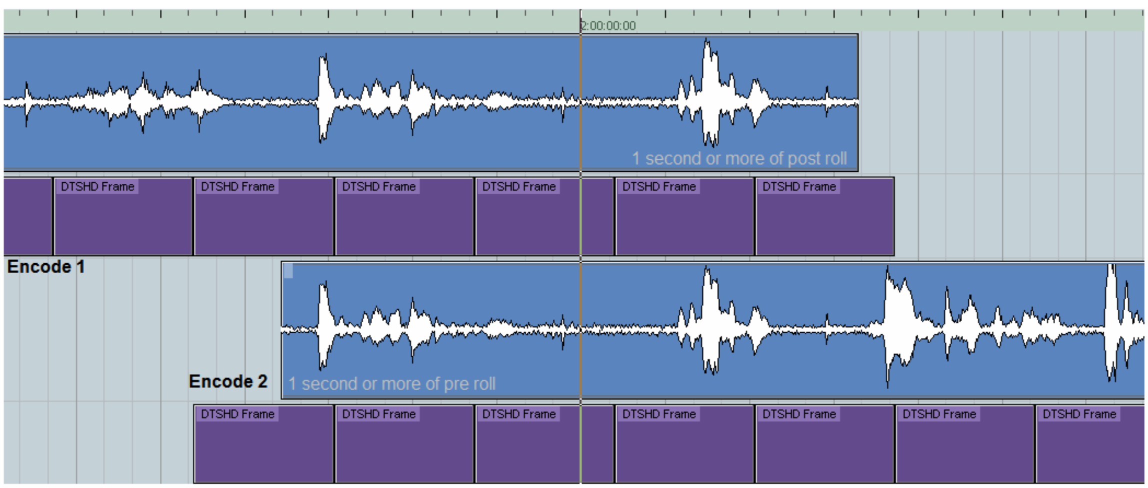


Figure 10-5 Join Overlap Diagram

Once the resulting audio segments are encoded with the same reference timecode (see section 7.1.6.8 for reference timecode explanation) the files can be joined at the user defined join time (02:00:00 as seen in Figure 10-4). **If the use of overlaps is not possible, it is recommended that the join time take place during a silent passage to avoid audio anomalies at the join point.**

- The subsequent DTS-HD encoded stream must contain a Reference Time equal to the Reference Time of the initial DTS-HD encoded stream. This will insure compatible DTS frame alignment between all encodes having the same reference time, thus maintaining audio to timecode synchronization.

☑ **Note:** Encoding with a Reference Time can only be performed with DTS-HD Pro Series Software Encoder version 0.97, DTS-HD Master Audio Suite-DTS Surround Audio Suite version 1.0 or later.

☑ **Note:** Encoding performed with the Encoder's "Reference Time" button disabled will result in a reference time that is equal to the file's start time.

☑ **Note:** DTS HD encoded streams whose reference time does not equal it's start time may have a delay that is inappropriate for disc authoring.

The illustration in Figure 10-6 depicts an attempted join of two DTS-HD encoded streams with *differing* reference times.

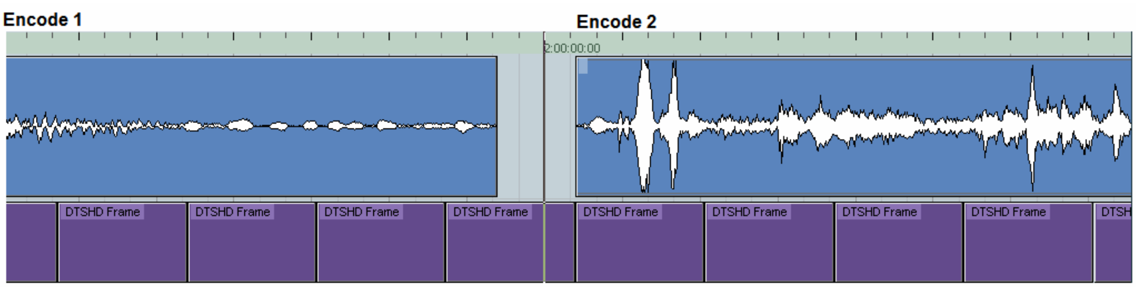


Figure 10-6 Join Example with Differing Reference Times

From the example in Figure 10-6, Encode 2 would have to be shifted prior to 02:00:00:00, it's intended start time, in order to be joined to Encode 1 between DTS frames. Encode 2 would then be out of sync after being joined to Encode 1.

The scenario in Figure 10-6 is not permitted using the Join operation. Encode 2 would need to be re-encoded with a reference time identical to Encode 1 (01:00:00:00) in order to be joined with Encode 1 using the Join tool.

If DTS-HD Master Audio (bit for bit exact) encoded files were used, simply decode Encode 2 and re-encode with the proper reference time (01:00:00:00).

If audio to timecode synchronization is not required, the Append operation can be used to join these DTS-HD encoded files as depicted, allowing encode 2 to shift in time, as depicted in Figure 10-7. *(see section 10.2* Append Tool*)*.

The example in Figure 10-7 shows how the start of Encode 2 would be lined up exactly to 02:00:00:00, if Encode 2 were encoded with a reference timecode of 01:00:00:00. Notice how each stream's DTS frames (in purple) line up exactly. Note each encode's DTS frames align exactly.

Figure 10-7 Join Example with the same Reference Times

- ❒ Audio data must exist in both encoded files at the DTS frame found at the specified join time. Gaps in audio data greater than one DTS frame are not permitted by the join operation. Pre and Post roll is required if it does not. In general, it is good practice to provide pre and post roll of 1 second or more on either side of the join time for ALL join operations. **If the use of pre and post roll overlap is not possible, it is recommended that the join time take place during a silent passage to avoid anomalies at the join time**.

☑ **Note:** The actual join will occur between the DTS frames nearest to the user specified join time.

- ❒ If audio material is encoded with "Encode Entire File" enabled, the end of the resulting encode may not land exactly on a timecode frame as shown in Figure 10-8. The "Encode Entire File" feature specifically retains any residual audio data (DTS frames / samples) following the last timecode frame.

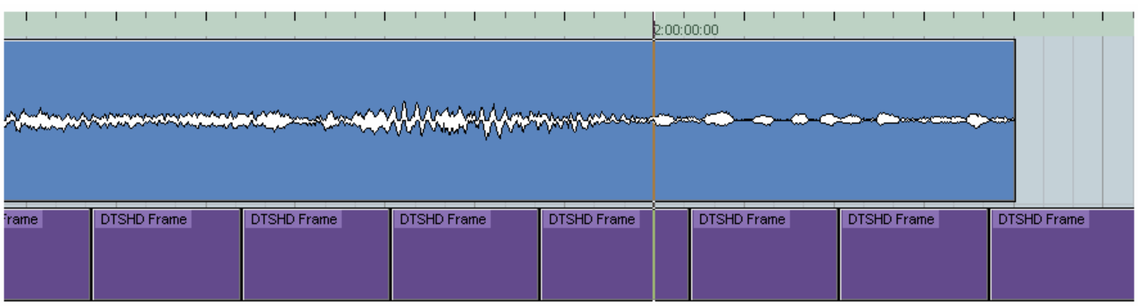


Figure 10-8 Join: Encode Entire File

In order to retain *all* audio data after a join operation is performed, the user specified end time for Encode 2 may be greater than that of Encode 2's actual end time (02:00:01 for the example shown in Figure 10-8). This will insure any residual audio data following the end time of Encode 2 is retained. Use the File Info tool to determine if an encode has residual audio data following it's end time (*see section 10.6*).

- In the rare event that Encode 1 is a few samples short of Encode 2's start time *yet* the number of DTS frame reaches the start time of Encode 2, the user may specify the start time of Encode 2 (01:00:00:00 in Figure 10-9) as the end time of Encode 1 even though that time is greater than the *actual* end time of encode 1 as shown in Figure 10-9. Gaps in audio data greater then one DTS frame are not permitted by the join operation.

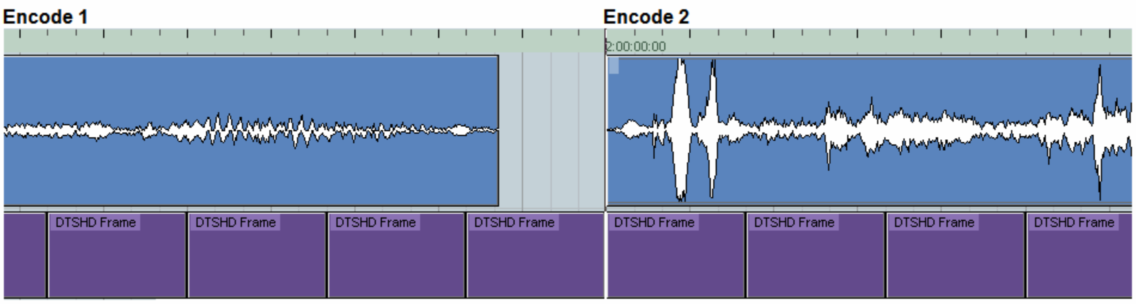


Figure 10-9 Join: Gap in Audio

- Only DTS-HD encoded streams with matching stream characteristics can be joined. The only items that may differ include:
  - File name
  - Start time
  - End time

Since it is not currently possible for the join or append applications to detect "-3dB Rear Channel Attenuation", "ES Phase Shift" and "Downmix Coefficient" discrepancies between encoded streams that are being joined, it is the responsibility of the user to avoid joining streams that do not match in this respect, DTS recommends that users **NOT** join encodes that differ in this fashion

- Audio to timecode synchronization cannot be maintained by the Join operation if the selected DTS-HD file has been restriped (see section 10.5 Restripe Tool).

In conclusion, the join operation has an advanced rule set that guarantees audio to timecode synchronization, as long as the encoded streams used have not been restriped.

Summary of JOIN requirements

- When specifying a join time, the end time of Encode 1 must equal the start time of Encode 2.

- If overlap is present, the audio contained in the overlapping region MUST be bit for bit identical. Overlapping regions must be at least 1 second in length and MUST be present before and after the intended join time in order for a join (with overlaps) to be permitted.

- Overlaps are necessary for a bit-exact edit to be performed on DTS-HD Master Audio encoded file joins.

- Encode 2 must be encoded with a Reference Time equal to the reference time of Encode 1.

- A DTS frame must exist in both encodes at the specified join time.

- If audio material is encoded with "Encode Entire File" enabled, the end of the resulting encode may not land exactly on a timecode frame. In order to retain *all* audio data following a join operation, the user specified end time for Encode 2 may be greater than that of Encode 2's actual end time.
- In the event that Encode 1 is a few samples short of Encode 2's start time *yet* the number of DTS frames reaches the start time of Encode 2, then the start time of Encode 2 may be specified as the end time of Encode 1 even though that time is greater than the *actual* end time of Encode 1.

- Only DTS-HD encoded streams with matching stream characteristics can be joined.

- Audio to timecode sync can not be maintained by the Join operation if the selected DTS-HD file has been restriped.

10.2. Append Tool

The Append Tool is activate by selecting the 

button, as shown in Figure 10-10 Append Tool, requires two DTS-HD encoded streams as its input and a destination file to save the results of the join operation. Double-click in each of the filename fields to load the files that are to be joined. The Timecode Start and Timecode End text fields show the start and end time of the input materials for each of selected files. Use the "Save To" button and the "File Name" field to specify the output directory and filename for the resultant append operation. Pressing the "Process" button will initiate the append processing. Pressing the "Cancel" button will stop the running process.

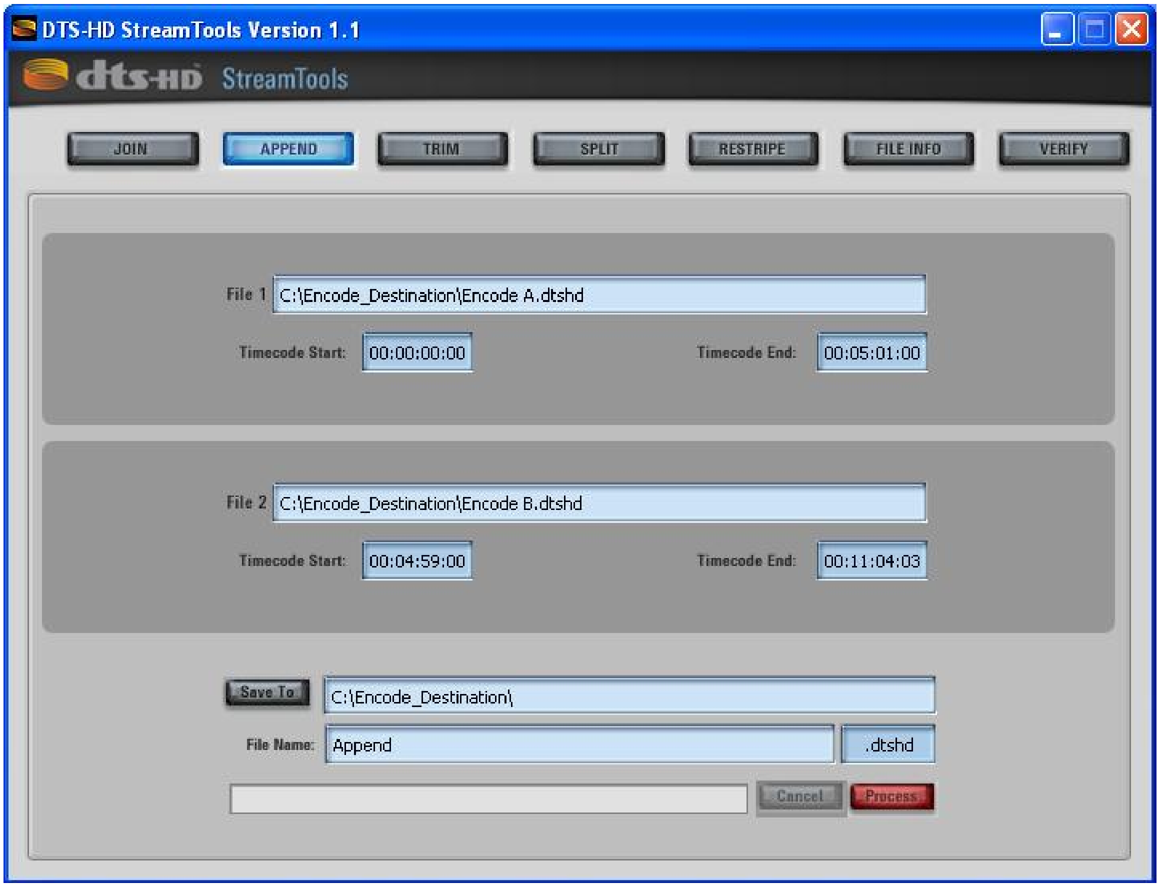


Figure 10-10 Append Tool

The Append tool allows the user to concatenate one DTS-HD encoded stream to another DTS-HD encoded stream with no regard for audio to timecode synchronization as is illustrated in Figure 10-11.

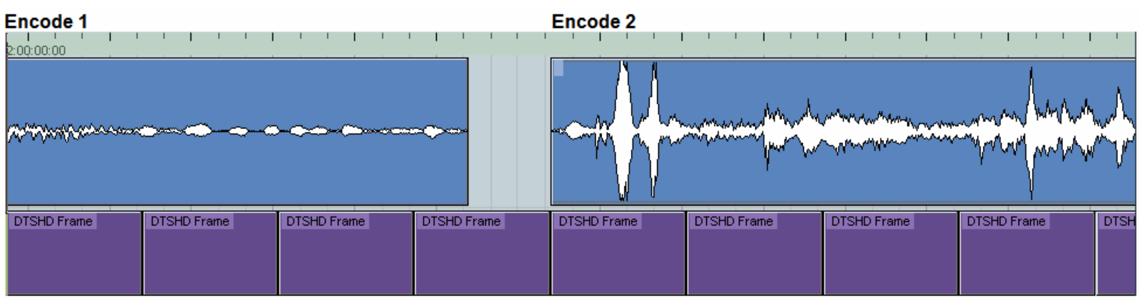


Figure 10-11 Append Tool Diagram

Audio to timecode synchronization for all encodes following the initial encode will be lost. As demonstrated in Figure 10-11, appending one Encode to another will cause Encode 2 to drift later in time, no greater then one DTS frame. This drift will increase with every Encode appended. The resulting DTS-HD encoded file will retain the start time of the initial encode.

☑ **Note:** Only DTS-HD encoded streams with matching stream characteristics can be joined with the append operation. The only items that may differ are File name, and timecode related information. Since it is not possible to detect "-3dB Rear Channel Attenuation", "ES Phase Shift" and "Downmix Coefficient" discrepancies between encoded streams that are being joined, DTS recommends that users **NOT** join encodes that differ in this fashion.

☑ **Note:** Proper audio editing technique dictates that all edits take place during a silent passage or a zero crossing in order to avoid any audio anomalies at the edit point

10.3. Trim Tool

The Trim Tool is activated by selecting the 

button, as shown in Figure 10-12 Trim Tool, requires a single DTS-HD encoded stream as its input and a destination file to save the results of the trim operation. Double-click in the filename input field to select the file that is to be trimmed. The Timecode Start and Timecode End text fields show the start and end time of the input material. The trim start and trim end time specifies the timecode where the trim operation will "cut" data from the DTS-HD encoded stream. Use the "Save To" button and the "File Name" field to specify the output directory and filename for the resultant trim operation. Pressing the "Process" button will initiate the trim processing. Pressing the "Cancel" button will stop the running process.

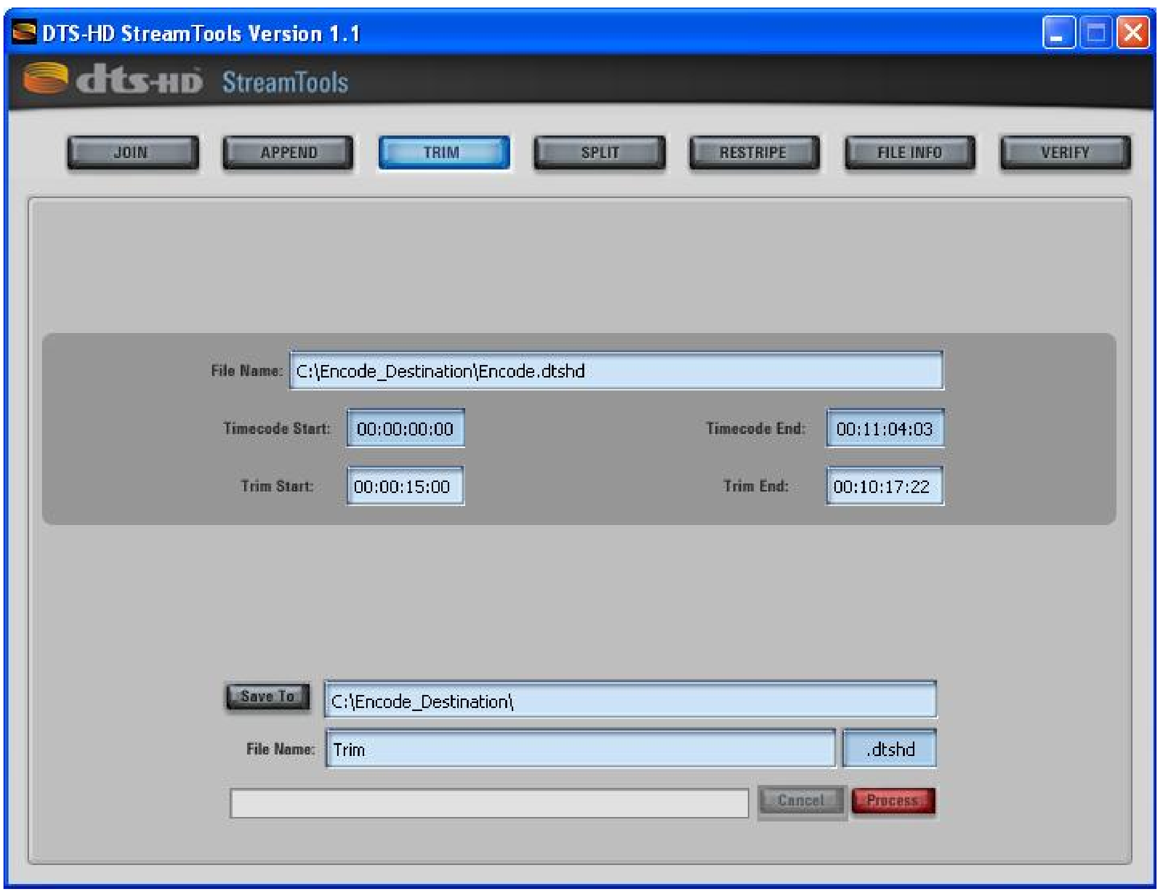


Figure 10-12 Trim Tool

The Trim tool operates by dropping DTS frames outside the boundary of the user specified start and end times. An encoded stream trimmed from 01:00:00:00 to 02:00:00:00 as shown in Figure 10-13.

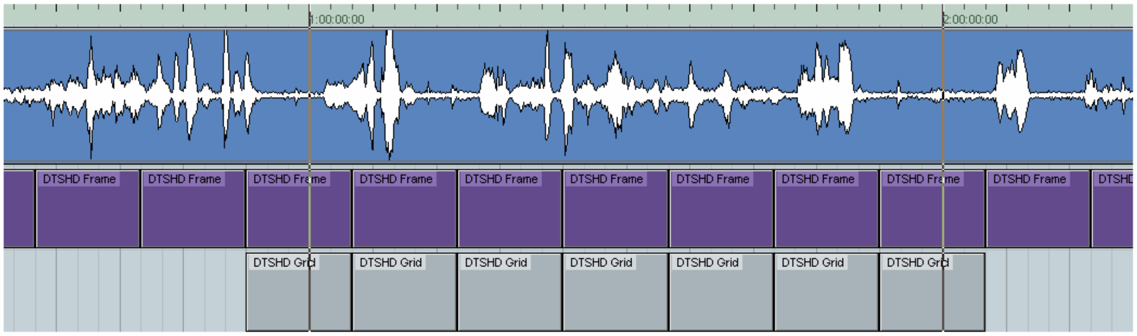


Figure 10-13 Trim Tool Diagram

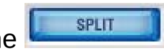
The DTS frames located at the specified Trim timecodes are retained in their entirety. However, the DTSHD header is modified accordingly. Upon decoding, only the audio between the specified Trim timecodes will be decoded.

☑ **Note:** Trimming the start time of a DTSHD file may make the file's delay inappropriate for disc authoring.

☑ **Note:** If audio material is encoded with "Encode Entire File" checked, the end of the resulting encode may not land exactly on a timecode frame. In order to retain *all* audio data (DTS frames / samples) after a trim operation, the user specified end time may be greater than that of the file's actual end time. This will insure any residual audio data following the end time is retained. Use the File Info tool to determine if an encode has residual audio data following its end time. (*see section 10.6).*

☑ **Note:** Proper audio editing technique dictates that all edits take place during a silent passage or a zero crossing in order to avoid any audio anomalies at the edit point.

10.4. Split Tool

The Split Tool is activated by selecting the 

button, as shown in Figure 10-14, requires a single DTS-HD encoded stream as its input and a two destination files to save the results of the split operation. Double-click in the filename fields to load the file that will be split. The Start Timecode and End Timecode text fields show the start and end time of the input material for the selected files. Use the "Save To File" buttons and the "File Name" fields to specify the output directories and filenames for the resultant split operation. Pressing the "Process" button will initiate the split processing. Pressing the "Cancel" button will stop the running process.

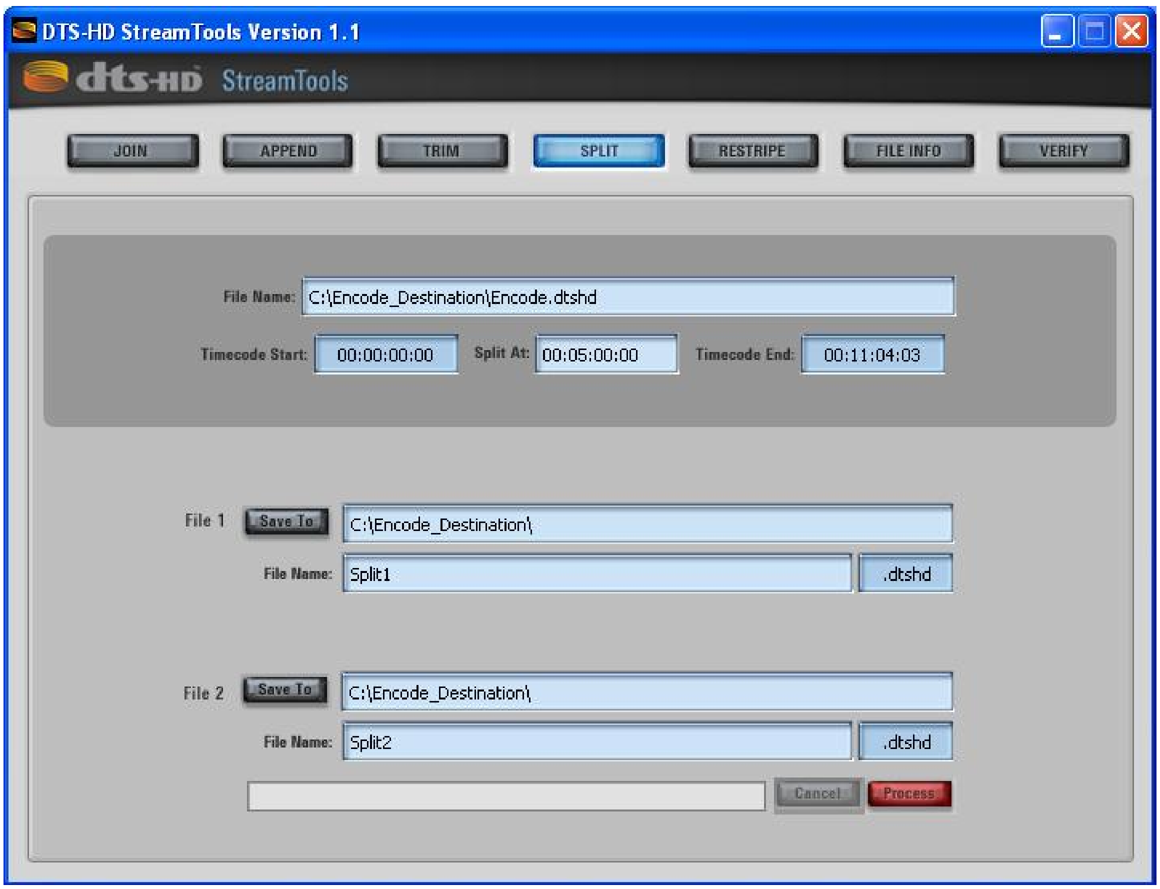


Figure 10-14 Split Tool

The Split tool operates by dividing an encoded stream at the DTS frame nearest to the user specified split time, resulting in the creation of two additional encodes as illustrated in Figure 10-15.

Figure 10-15 Split Tool Diagram

The DTS frame located at the specified split time is retained in both encodes. When the encoded streams are decoded, only the audio between the specified timecodes will be decoded.

- ☑ **Note:** The start time of the second DTS-HD stream created by the split operation may make the file's delay inappropriate for disc authoring.
- ☑ **Note:** Proper audio editing techniques dictates that all edits take place during a silent passage or a zero crossing in order to avoid any audio anomalies at the split point.

10.5. Restripe Tool

The Restripe Tool is activated by selecting the 

button, as shown in Figure 10-16 Restripe Tool, requires a single DTS-HD encoded stream as its input. Double-click in the filename fields to load the file that will be restriped. Specify the start timecode and the frame rate for the restripe operation. Pressing the “Process” button will initiate the restripe processing.

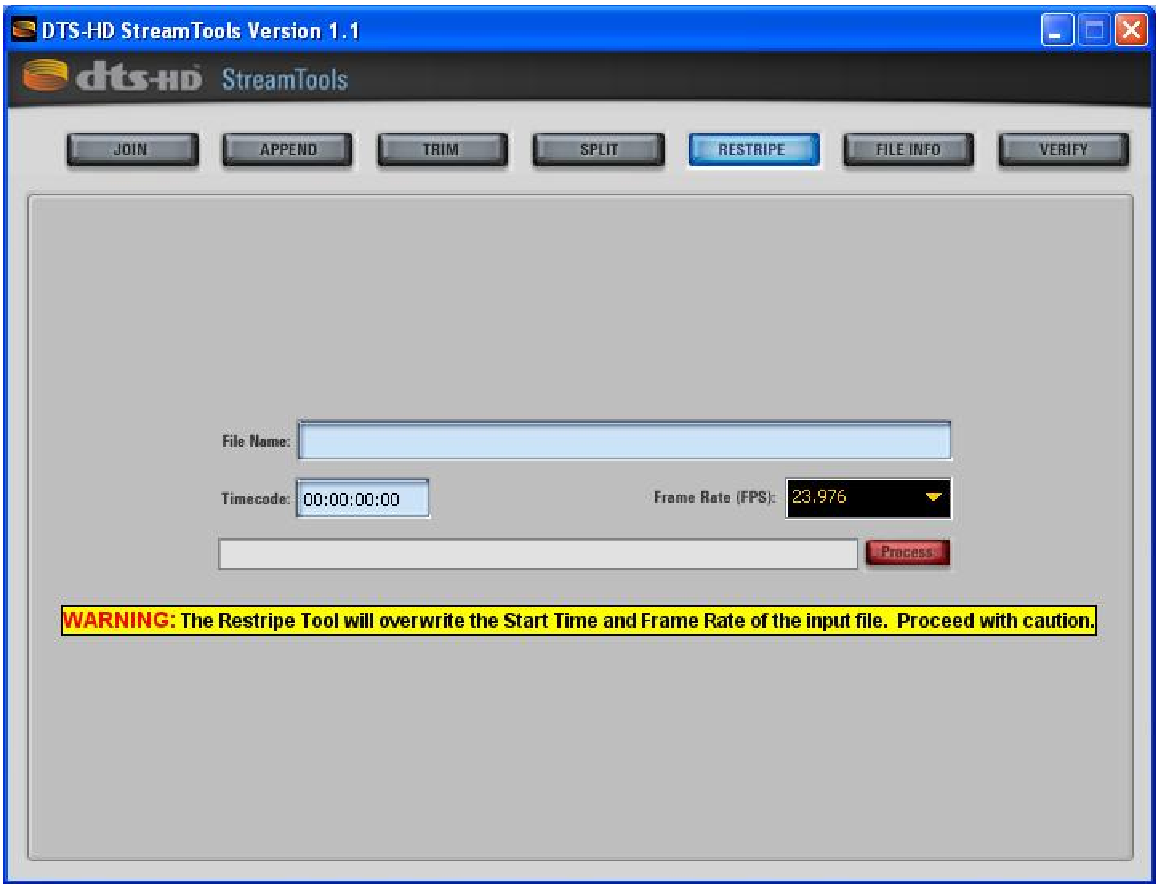


Figure 10-16 Restripe Tool

The restripe operation simply re-labels the start time and frame rate of the selected DTS-HD file. The file’s end time is a calculation of its start time.

> **CAUTION**: Consider audio to timecode synchronization lost once a file has been restriped. Audio to timecode sync cannot be maintained by the Join operation if the selected DTS-HD file has been restriped. Drift, no greater then one DTS-HD frame, may be experienced when a file is restriped.

☑ **Note:** Restriping an encoded file will adjust the file's Reference Time to a valid timecode per the new frame rate and frame count. The reference time will remain valid after a restripe is performed.

10.6. File Info Tool

The File Info Tool is activated by selecting the 

button, as shown in Figure 10-17.

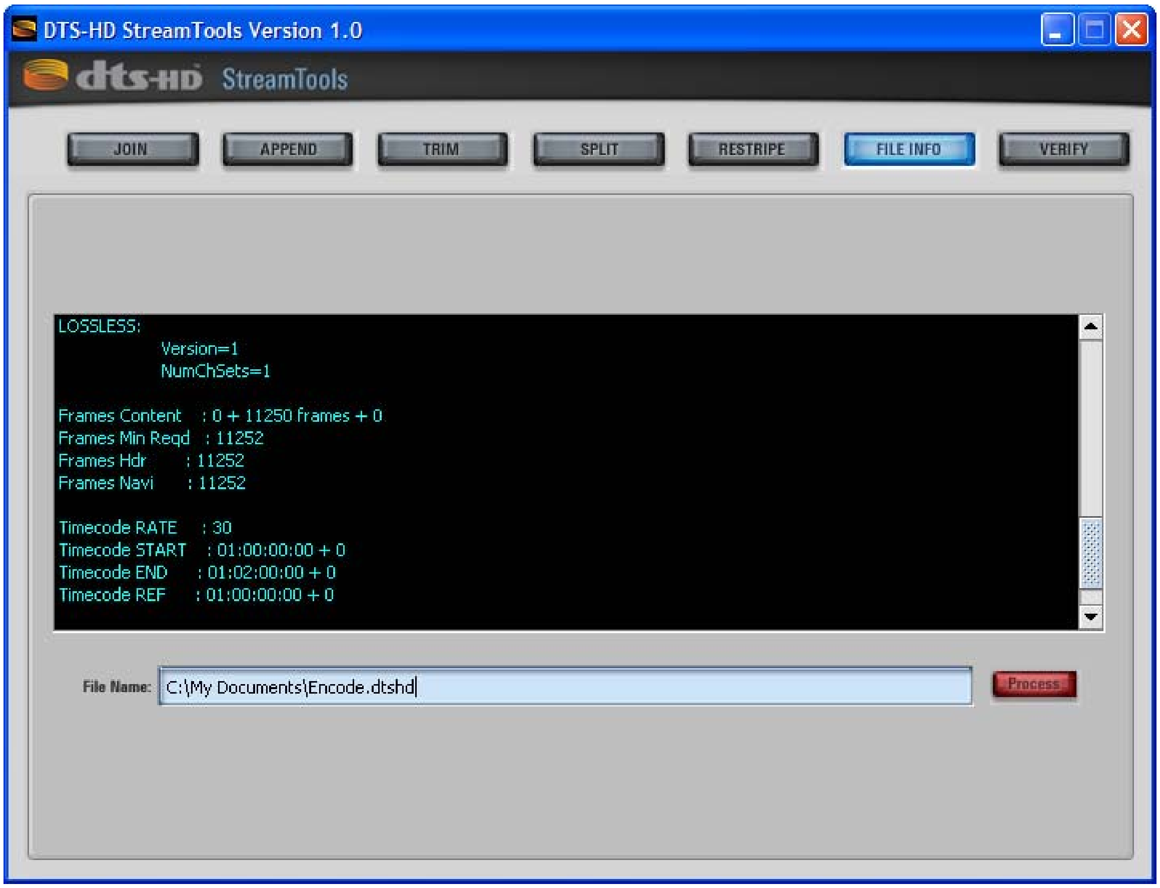


Figure 10-17 File Info Tool

This tool opens the selected DTS-HD file and reads the DTS-HD header extracting the information that is contained within the encoded stream. The tool does not decode or perform any validation. That is the function of the 'Verify Tool' described in section 10.7 The data is partitioned into several sections showing information found in the header, the core substream, the extension substream, the audio presentation data such as number of channels, bit rate, sample rate, etc., the size of the encoded audio stream, navigation data and the synchronization words that are pertinent to the stream. The scrolled area of the Verify tool can be selected by highlighting with the mouse and copy/paste to an external file for future use. This is accomplished by using Ctrl-C (Windows) or Apple-C (Macintosh) to copy from the scrolled pane and Ctrl-V (Windows) or Command-V (Macintosh) to paste into an open file.

Table 10-1 shows an example of the output generated from the File Info Tool.

Table 10-1 DTS File Info Application Output

```
File Information for C:\My Documents\DTSENC.dtshd:

DTSHDHDR:
        Header Version=0
        Num Audio Presentations=1
        cbr(YES), pbrs(NO), navi(YES)
        core(YES), ext(YES)

CORESS:
        Max_Sample_Rate_Hz=48000
        Bit_Rate_Kbps=1509
        Channel_Mask=00000002
        Frame_Payload_In_Bytes=2012

EXTSS:
        Ext_Ss_Avg_Bit_Rate_Kbps=0
        Ext_Ss_Peak_Bit_Rate_Kbps=0
        Pbr_Smooth_Buff_Size_Kb=128
        bcc(CORE), ll(YES), lbr(NO)

Source Samples    : 1783293
Sample Rate       : 48000Hz
Samples Per Frame : 512
Codec Delay       : 1024

LOSSY:
        FSize=2012
        Amode=2
        LFF=0
        SFreq=13
        PCMR=0
        DialNorm=0


LOSSLESS:
        Version=1
        NumChSets=1

Frames Content    : 0 + 3482 frames + 509
Frames Min Reqd   : 3485
Frames Hdr        : 3496
Frames Navi       : 3496

Timecode RATE     : 23.976
Timecode START    : 00:00:00:00 + 0
Timecode END      : 00:00:37:02 + 1513
Timecode REF      : 00:00:00:00 + 0
```

10.7. Verify Tool

The Verification Tool is activated by selecting the  button, as shown in Figure 10-18.

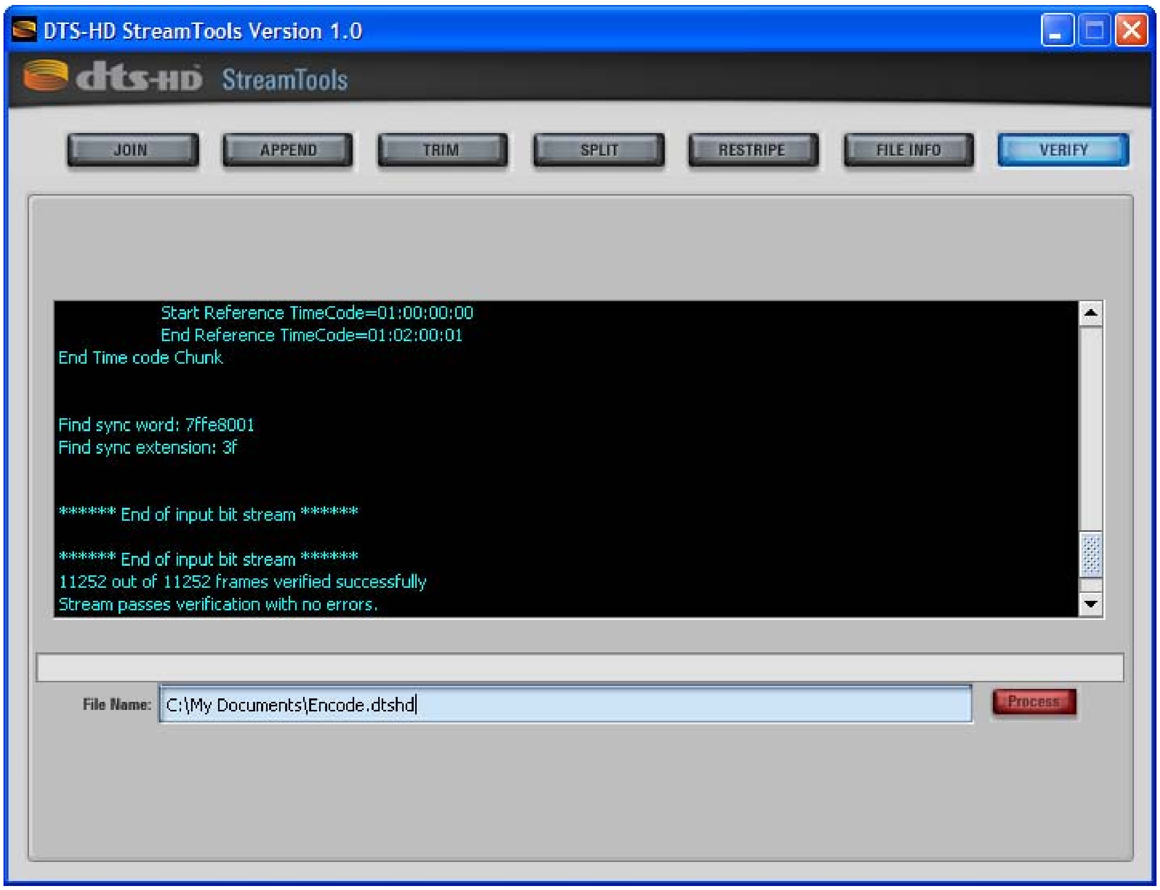


Figure 10-18 Verification Tool

This tool operates exactly like the File Info tool except that it performs a full decode of the selected DTS-HD file (.dtshd extension), however, no audio data is generated during a verify operation. The process of decoding the data results in validating that the audio components are in their proper locations. It does not perform any audio performance processing. The data is partitioned into several sections showing information found in the header, the core substream, the extension substream, the audio presentation data such as number of channels, bit rate, sample rate, etc., the size of the encoded audio stream, navigation data and the synchronization words that are pertinent to the stream. The status bar above the file name selection area provides feedback on the number of frames that have been processed. The scrolled area of the Verify tool can be selected by highlighting with the mouse and copy/paste to an external file for future use. This is accomplished by using Ctrl-C (Windows) or Apple-C (Macintosh) to copy from the scrolled pane and Ctrl-V (Windows) or Command-V (Macintosh) to paste into an open file.

Table 10-2 shows an example of the output generated from the Verification Tool.

Table 10-2 DTS Verification Application Output

```
----- Start Verify process -----
Verifying file:C:\My Documents\DTSENC.dtshd
****** DTSHD Verification Tool Version 300.de Aug 18 2006 09:11:13 ******
DTSHD Header Chunk
        TimeCode reference clock=1.0/48000.0
        TimeCode Frame Rate=23.976
        VBR mask enabled
        VBR Navigation Table mask enabled
        Core Substream mask enabled
        Extension Substream mask enabled
        Num of Audio Presentations=1
End DTSHD Header Chunk
Begin Core SubStream Chunk
        Core Sample Rate=48000
        Core Encode BitRate=1509
        Core ChMask=2059
        Core Channels: Ctr L R LFE LSideSrrd RSideSrrd
        Core Frame Payload(Bytes)=2012
End Core SubStream Chunk
Begin Ext SubStream Chunk
        Ext contains VBR component.
        PBR=n/a
        PBRBuffer=128
End Ext SubStream Chunk
Begin Audio Presentation Chunk
        Audio Presentation Index =0
        Max SampleRate=48000
        Total Number of Frames=3496
        Total Number of samples per frame=512
        Total Number of samples in original(@ Max SampleRate)=1783293
        Channel Mask=2123
        Channels: Ctr L R LFE LSrrdRear RSrrdRear LSideSrrd RSideSrrd
        Codec delay in samples=1024
End Audio Presentation Chunk
Begin Stream Data Chunk
        Encoded Stream Size=16828648 bytes
End Stream Data Chunk
Begin Navigation Table Chunk
        Num of Entries=3496
        Frame Interval=1
        Bytes Per Entry=4
        Skipping printing of Navigation entries.
End Navigation Table Chunk
Find sync word: 7ffe8001
Find sync extension: 3f
****** End of input bit stream ******
3496 out of 3496 frames verified successfully
Stream passes verification with no errors.

----- End Verify process -----
```

11. Encoder Error Codes

The following section is a detailed list of possible error return codes.

Each error code entry is listed in the following format:

- Error code value in hexadecimal
- Symbolic constant
- Description of error (in programming terminology)
- If exists, output message displayed (if no message defined, "Encode exit status: 0xfoobar" is displayed)

0x10001
ERR_S_SERVER_INIT
Windows socket library - initialization failure "Error: Socket library initialization error."

0x10002
ERR_S_SERVER_S_LISTEN_CREATE
Windows socket library - unable to create server listening socket
"Error: Unable to create listen socket."

0x10003
ERR_S_SERVER_BIND
Windows socket library - server listening socket bind error
"Error: Socket bind error."

0x10004
ERR_S_SERVER_LISTEN
Windows socket library - server listening socket listen error
"Error: Socket listen error."

0x10005
ERR_S_SERVER_S_LISTEN_CLOSE
Windows socket library - server listening socket close error
"Error: Socket close error."

0x20001
ERR_ENCCMDLINE_NO_FILES
No encoding input files specified
"Error: No input files specified."

0x20002
deprecated - unused

0x20003
ERR_ENCCMDLINE_UNMATCHED_QUOTE
Unmatched quotation marks enclosing input file name
"Error: Unmatched quotation mark in file name."

0x40001
ERR_ENCINPUTFILES_FCNT
Insufficient number of input files listed for specified encode configuration
"Error: Insufficient number of file input channels for specified configuration."

0x40002
ERR_ENCINPUTFILES_FOPEN
Error opening input file
"Error: Unable to open input file "

0x40003
ERR_ENCINPUTFILES_FORMAT_CORECH
Sample rate or sample width of core input file does not match with encode configuration
"Error: Incorrect format for core channel audio file."

0x40004
ERR_ENCINPUTFILES_FORMAT_XCH
Sample rate or sample width of XCh input file does not match with encode configuration
"Error: Incorrect format for extended channel audio file."

0x40005
ERR_ENCINPUTFILES_HDR_INVALID
Error reading 'wav' file header
"Error: Invalid audio file header"

0x40006
ERR_ENCINPUTFILES_LBR_BITWIDTH
Input file for LBR encode has invalid sample width (only 16-bit or 24-bit allowed)
"Error: LBR encoding requires 16-bit or 24-bit audio."

0x80001
ERR_ENCCONFIG_NO_FILE_SPECIFIED
No configuration file specified
"Error: No configuration file specified."

0x80002
ERR_ENCCONFIG_FILE_OPEN
Error opening configuration file
"Error: Unable to open configuration file "

0x80003
ERR_ENCCONFIG_LBR_BITRATE_INVALID
Requested LBR bit rate is too low for specified configuration
"Error: LBR bit rate is too low for specified configuration."

0x80004
ERR_ENCCONFIG_LBR_FILE_CNT
Incorrect number of file input channels listed for 5.1 LBR encode
"Error: LBR 5.1 encode requires exactly 6 file input channels."

0x80005
ERR_ENCCONFIG_LBR_FILE_STEREO_RLS

Invalid stereo file input channel pairing for 5.1 LBR encode; Right and Left Surround
"Error: LBR 5.1 inputs for R + Ls may not be specified by a stereo file."

0x80006
ERR_ENCCONFIG_LBR_FILE_STEREO_RSC
Invalid stereo file input channel pairing for 5.1 LBR encode; Right Surround and Center
"Error: LBR 5.1 inputs for Rs + C may not be specified by a stereo file."

0x80007
ERR_ENCCONFIG_SMPTE_FORMAT_INVALID
Invalid SMPTE timecode string
"Error: SMPTE timecode format is invalid. Expecting 'hh:mm:ss:ff' or 'hh:mm:ss;ff'."

0x80008
deprecated - unused

0x80009
ERR_ENCCONFIG_MINUTES_INVALID
Number of minutes in SMPTE timecode string greater than 59
"Error: Specified SMPTE minute value is out of range."

0x8000A
ERR_ENCCONFIG_SECONDS_INVALID
Number of seconds in SMPTE timecode string greater than 59
"Error: Specified SMPTE second value is out of range."

0x8000B
ERR_ENCCONFIG_FRAMES_INVALID
Number of frames in SMPTE timecode string greater than max allowable at specified frame rate
"Error: Specified SMPTE frame value is out of range."

0x8000C
ERR_ENCCONFIG_TC_OBJECT_CREATION_ERR
Error creating SMPTE timecode object
"Error: Unable to create timecode object."

0x8000D
ERR_ENCCONFIG_TC_FRAMEREATE_INVALID
Unsupported frame rate specified
"Error: Timecode frame rate is invalid."

0x8000E
ERR_ENCCONFIG_TC_ENCFROM_INVALID
Specified 'Encode From' time is prior to the material start time or is greater than the 'Encode To' time
"Error: Specified encode from time is invalid."

0x8000F
ERR_ENCCONFIG_TC_ENCTO_INVALID
Specified 'Encode To' time is beyond the end of the material
"Error: Specified encode to time is invalid."

0x100001
ERR_ENCCTRLBLK_SAMPLE_BEGIN_INVALID

Specified encode 'Start Time' is beyond the time duration of the input files
"Error: Specified beginning sample index is invalid."

0x100002
ERR_ENCCTRLBLK_SAMPLE_END_INVALID
Specified encode 'End Time' is prior to the encode 'Start Time' or is beyond the time duration of the input files
"Error: Specified ending sample index is invalid."

0x200001
ENC_ERR_CATCH_INITIAL
Indicates C++ catch handling for encode state ENC_STATE_INITIAL which includes following encoder API calls:
DTSEnc_CreateEncoder
DTSEnc_ParseConfigStream
DTSEnc_ValidateConfigStream
DTSEnc_GetLastError

0x200002
ENC_ERR_CATCH_SET_IO_BUFFERS
Indicates C++ catch handling for encode state ENC_STATE_SET_IO_BUFFERS which includes following encoder API calls:
DTSEnc_GetDataIn
DTSEnc_IsDTSHDEncode
DTSEnc_GetDataOut

0x200003
ENC_ERR_CATCH_READ
Indicates C++ catch handling for encode state ENC_STATE_READ which consists mainly of method calls to the WAVEIO class

0x200004
ENC_ERR_CATCH_ENCODE
Indicates C++ catch handling for encode state ENC_STATE_ENCODE which includes following encoder API calls:
DTSEnc_EncodeFrame
DTSEnc_GetLastError

0x200005
ENC_ERR_CATCH_WRITE
Indicates C++ catch handling for encode state ENC_STATE_WRITE which consists mainly of method calls to the ofstream class

0x200006
ENC_ERR_CATCH_POST_ENCODE
Indicates C++ catch handling for encode state ENC_STATE_POST_ENCODE which includes following encoder API calls:
DTSEnc_EncodeFrame
DTSEnc_GetDTSHDAppendix
DTSEnc_GetDTSHDHeader

0x200007
deprecated - unused

0x200008
ENC_ERR_CATCH_FINAL
Indicates C++ catch handling for encode state ENC_STATE_FINAL which includes following encoder API calls:
DTSEnc_DestroyEncoder

0x200009
ENC_ERR_STATE_INVALID
Invalid encode state encountered

0x20000A
ENC_ERR_ENC_CREATE
Encoder API call to DTSEnc_CreateEncoder() returned NULL

0x20000B
ENC_ERR_CONFIG_FSTREAM
Error opening configuration file stream

0x20000C
ENC_ERR_CONFIG_ISTREAM
Error creating configuration file input stream

0x20000D
ENC_ERR_CONFIG_PARSE
Encoder API call to DTSEnc_ParseConfigStream failed

0x20000E
ENC_ERR_CONFIG_VALIDATE
Encoder API call to DTSEnc_ValidateConfigStream failed either due to invalid configuration file content or invalid DTS license file

0x20000F
ENC_ERR_OUTPUT_FSTREAM
Error opening output file stream

0x200010
ENC_ERR_ENCFRAME
Encoder API call to DTSEnc_EncodeFrame failed

0x200011
ENC_CANCELLED
Encode termination due to user cancellation request

0x400001
ENCLBR_ERR_ENC_CREATE
LBR encoder API call to LBRENC_Allocate() failed

0x400002
deprecated - unused

0x400003
ENCLBR_ERR_SUBSTRM_HDR_CREATE
Error creating new DTS substream header (class CDTSSubStrmFrm)

0x400004
ENCLBR_ERR_DTSHDFILE_CREATE
Error creating new DTS file wrapper (class CDTSHDFile)

0x400005
ENCLBR_ERR_INITIALIZE
LBR encoder API call to LBRENC_Initialize() failed
"LBRENC Initialize Error: Invalid bit rate - must be at least 24000 bits/sec per channel"
"LBRENC Initialize Error: Invalid number of channels"
"LBRENC Initialize Error: Invalid sample rate"
"LBRENC Initialize Error: Incompatible bit-stream version"
"LBRENC Initialize Error: Invalid data length"
"LBRENC Initialize Error: Out of memory"
"LBRENC Initialize Error: Invalid samples per frame"
"LBRENC Initialize Error: Unknown"

0x400006
ENCLBR_ERR_INPUT_MEM_ALLOC
Memory allocation error of either the LBR input file buffer or LBR input frame buffer

0x400007
ENCLBR_ERR_OUTPUT_MEM_ALLOC
Memory allocation error of the LBR output frame buffer

0x400008
ENCLBR_ERR_FSEEK
Error in file seek to specified byte offset

0x400009
ENCLBR_ERR_FREAD
Error reading input file

0x40000A
ENCLBR_ERR_SUBSTRM_HDR
Error in call to LBR support function SetupLBRSSHeader()

0x40000B
ENCLBR_ERR_ENCODE_FRAME
LBR encoder API call to LBRENC_EncodeFrame() failed

0x40000C
ENCLBR_ERR_OUTPUT_STREAM
Error opening output file

0x40000D
ENCLBR_ERR_FWRITE
Error writing to output file

0x40000E
ENCLBR_ERR_FWRITE_STRM_HDR
LBR encoder API call to WriteLBRStreamHeaderToFile() failed

0x40000F

ENCLBR_ERR_FWRITE_STRM_SYNC
LBR encoder API call to WriteLBRStreamSync() failed

0x400010
ENCLBR_ERR_STATE_INVALID
Invalid LBR encode state encountered

0x400011
ENCLBR_ERR_LICENSE_INVALID
LBR encoding not enabled in license file

0x400012
ENCLBR_CANCELLED
LBR encode termination due to user cancellation request

12. DTS Tools Error Codes

The following section describes the possible error codes that may be generated by the DTS Tools application.
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12.1. File Error Codes

FILE_VERSION_NOT_SUPPORTED_BY_TOOL - "This file version is not supported"
Only DTSHD files created with DTS-HD Pro Series Encoder 0.97 or later can be used with the Tools.

FILE_OPEN_FAILED "File could not be opened"
File does not exist or can not be created.

PBRS_FILE "This is a smoothed file"
Peak bit rate smoothing has been performed on specified file. No further editing is possible.

READ_READ_GARBAGE - "File structure is corrupt"
File should be re-encoded

FILE_EXTENSION_NOT_PRESENT - "File extension is not present"

FILE_EXTENSION_NOT_RECOGNIZED - "File extension is not .dtshd"

FILE_IS_EMPTY - "File is zero size"

READ_NO_HEADER_FOUND - "File is missing header"
File should be re-encoded

READ_REQUIRES_PADDING_BYTES - "READ_REQUIRES_PADDING_BYTES"

BAD_ERROR_NO - "Bad error number"

12.2. Timecode Error Codes

INVALID_DROP_FRAME_TIMECODE "Invalid drop frame timecode"
The SMPTE Drop Frame standard dictates every frame :00 & :01 are dropped for each minute except for minutes ending in 0 (00:, 10:, 20:, 30:, 40: & 50).

CHECK_TIMECODE_OUT_OF_FILE_RANGE "Timecode is out of file range"
The specified timecode is out of range of the selected file.

FRAMES_TOO_LARGE "Frame count too big"
Timecode value specified contains more frames then given frame count. (Ex: 01:00:00:45 at 30 frames per second).

CHECK_TIMECODE_FORMAT "Timecode format does not match with file"

Improper timecode format. Proper timecode format is 00:00:00:00 or 00:00:00;00 where each digit is a number from 0 to 9.

TIMECODE_IS_NONSENSE "The timecode format is unrecognized"

NOT_A_VALID_TIME "Not a valid time"

12.3. Join / Append Error Codes

OVERLAP_NOT_IDENTICAL - "Overlapping audio regions must be identical"
Identical, overlapping audio of at least 1 second in length MUST be present before and after the intended join time in order to join two lossless DTS-HD encodes.

FILES_DONOT_MATCH - "Files are not compatible for concatenation"
Joining non-matching encodes (ex: sample rate, bit depth, channel layout, core bit rate, reference time etc.) is not permitted.

GAP_AT_END_OF_FIRST_FILE - "Missing data at the end of the first file"
Audio data must exist in both encodes at the specified join time.

JOIN_NO_COMMON_TIMECODE - "No common timecode for join"
When specifying a join time, the user specified end time of the initial DTS-HD encoded stream must equal the user specified start time of the subsequent DTS-HD encoded.

13. Detailed Discussion on Dialog Normalization

Dialog Normalization is the use of metadata to control playback level. It has no effect on encoding or decoding. It is an instruction to playback equipment to adjust level post-decoding. A single DIALNORM value is applied to a given program: level is not adjusted within a program. There is no compression involved. All channels are affected equally. DIALNORM can only attenuate.

The DIALNORM value entered into the encoder is the subjective average dialogue level of the program, *not* the amount of playback attenuation. The subjective average dialogue level is typically approximated by LeqA measurement. Table 13-1 shows the correspondence between DIALNORM value and attenuation.

Table 13-1 DIALNORM and Attenuation Correspondence Values

DIALNORM value (dialogue level, dBFS LeqA)	**Attenuation** (dB)
-31	0
-30	1
-29	2
-28	3
-27	4
-26	5
-25	6
-24	7
-23	8
-22	9
-21	10
-20	11
-19	12
-18	13
-17	14
-16	15
-15	16
-14	17
-13	18
-12	19
-11	20
-10	21
-9	22
-8	23
-7	24
-6	25
-5	26
-4	27
-3	28
-2	29
-1	30

The purpose of DIALNORM is to achieve a consistent level of dialogue from one program or program source to the next. End-users set volume to achieve a pleasing and intelligible level of dialogue. However, dialogue level varies considerably across different program types. For example, in a feature motion picture, dialogue typically averages -27 dBFS LeqA in order to allow sufficient headroom for explosions and other effects. In contrast, in a news broadcast the announcer's voice may average -15 dB, closer to full scale; and various other programs will have other dialogue levels appropriate to their headroom needs. If there is no compensation, the end-user will have to turn up the feature film, turn down the news broadcast, etc. With dialogue normalization, a DIALNORM value of -27 is assigned to the movie, and a value of -15 is assigned to the newscast. The dialogue levels then match on playback and no user adjustment should be necessary.

For music, which contains no dialogue, DIALNORM may be used to set playback loudness at a level appropriate in the context of other program material, determined subjectively.

A DIALNORM value of -31 results in no DIALNORM attenuation.

For broadcast or disc authoring of a feature motion picture, if direct measurement has not indicated a different level, set the DIALNORM value to -27. This has been widely accepted as typical for features, and THX-certified playback equipment assumes a value of -27 in aligning levels between the home and the dubbing stage.

If more than one codec is being used for the same audio track (e. g. on DVD), DIALNORM value should be set the same on all codecs for a given program.

For further information on DIALNORM, including typical values for different types of programs and music, see:

Tomlinson Holman, 5.1 Surround Sound Up and Running. Boston: Focal Press, 2000. 163-167.

14. Encoder Log File Output Example

Due to the use of tabs and other white spaces contained within the log file, Table 14-1 depicts the editors that should be used for best on-screen viewing on the log file.

Table 14-1 Recommended Word Processors for Viewing Data

Windows Operating System	Macintosh Operating System
Notepad	Textedit
WordPad may be used but the data will not be formatted as desired.	

The following log file example was generated for a Blu-ray Disc, 7.1 channels (lossless) at 1509 kbps at 48 kHz with 5.1 downmix settings, and 2.0-channel embedded downmix.

```
*** DISCLAIMER ***
This file contains the parameter settings and selections for the specified encode. Be
advised that the encoder may use slightly different values for certain parameters to
complete the requested encode.  As such, for verification purposes, please use the
Verify or Info application found on the DTS-HD StreamTools User Interface to confirm
this encode.

****************************************

AUDIO INPUT SETTINGS
------------------------------------------------------
Media Type              = Blu-ray Disc
Product Type            = DTS-HD Master Audio
Bit Rate                = 1509 kbps
Channel Layout          = 7.1 - L, R, C, LFE, Lss, Rss, Lsr, Rsr
Bit Width               = 24
DialNorm                = -31 dBFS (No Attenuation)
Sample Rate             = 48 kHz
-3db Rear Attenuation   = false
ES Phase Shift          = false
ES Pre-Mixed            = false

INPUT FILES
------------------------------------------------------
Left                     C:\Documents and Settings\dtshduser\Desktop\48kTone.wav
Right                  = C:\Documents and Settings\dtshduser\Desktop\48kTone.wav
Center                 = C:\Documents and Settings\dtshduser\Desktop\48kTone.wav
LFE                    = C:\Documents and Settings\dtshduser\Desktop\48kTone.wav
Left  Surround         = C:\Documents and Settings\dtshduser\Desktop\48kTone.wav
Right Surround         = C:\Documents and Settings\dtshduser\Desktop\48kTone.wav
XCH1                   = C:\Documents and Settings\dtshduser\Desktop\48kTone.wav
XCH2                   = C:\Documents and Settings\dtshduser\Desktop\48kTone.wav

TIME CODE SETTINGS
------------------------------------------------------
Frame Rate             = 23.976
Encode Entire File     = true
Start Time             = 00:00:00:00
End   Time             = 00:00:30:00
Use Reference          = false

OUTPUT LOCATION
```

```
-------------------------------------------------------
Directory             = H:\My Documents\
Filename              = DTSENC.dtshd

DOWNMIX SETTINGS
-------------------------------------------------------
      5.1 Downmix Settings
      ---------------------------------------------------
                   Scale
      Input        Factor       XCH1         XCH2
      ------       ------       ---          ----
      Left         3.0          INF          INF
      Right        3.0          INF          INF
      Center       3.0          INF          INF
      LFE          3.0          INF          INF
      Ls           3.0          3.0          INF
      Rs           3.0          INF          3.0

      2.0 Downmix Settings
      ---------------------------------------------------
      2.0 Downmix Embedded
                   Scale
      Input        Factor       Rs           Ls           LFE          Center
      ------       ------       ----         ----         -----        ------
      Left         3.0          INF          6.0          INF          6.0
      Right        3.0          6.0          INF          INF          6.0
```