DTS-HD Master Audio Suite™
User Guide

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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, 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 Express™ encoded files using the Encoder module, to verify and validate the encoded material using the DTS-HD StreamPlayer 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 up to 7.1 channels with sample rates up to 96kHz and up to 5.1 channels with a sample rate of 192 kHz.
- ❐ For 192 kHz material, DTS-HD Master Audio 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, with up to 7.1 channels from source files with sampling rates up to 96 kHz.
- ❐ Blu-Ray Secondary Audio Low Bit Rate (LBR) using DTS-HD Express encoding 1.0, 2.0, and 5.1 channels at bit rates from 24 kbps to 256 kbps.

  ☑ **Note**: DTS-HD Master Audio (lossless) and DTS-HD High Resolution Audio 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 mono or stereo in '.wav' '.aif' or '.aiff' LPCM format.

The DTS-HD StreamPlayer is a software 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.

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

1.2 DTS Surround Audio Suite

This software package consists of an encoder that is only capable of creating DTS Digital Surround encoded streams for DVD-V and DTS Music Disc (DTS CD) media types. It supports up to 6.1 discrete channels at 48kHz (96 kHz sample rate for DVD at bit rates up to 1.5 Mbps). The user interface behaves identically to that of DTS-HD Master Audio Suite with the noted exceptions found in the sections that follow.

2. System Requirements

2.1 Operating System Requirements

Microsoft Windows XP 32-bit Operating System or Windows 7 Professional 32-bit,
Greater than 2.0GHz Dual-Core processor
USB port for iLok dongle (only needed for DTS-HD StreamPlayer)
7200 RPM Non-system drive or
External firewire drive for DTS-HD StreamPlayer playback
Sun Java ™ 2 Standard Edition Runtime Environment, Version 6.
Internet Explorer 8

Apple Macintosh Intel running OS X (10.4.11 : 10.5.8 : 10.6.2)
Greater than 2.0GHz Intel Core 2 Duo processor
USB port for iLok dongle (only needed for DTS-HD StreamPlayer)
7200 RPM Non-system drive or
External firewire drive for DTS-HD StreamPlayer playback
Sun Java ™ 2 Standard Edition Runtime Environment, 32-bit Version of 6 or 5.
Safari 4

2.2 Memory Requirements

Microsoft Windows XP/7 32-bit Operating System: minimum 2 GB RAM Required, 4 GB RAM recommended

Apple Macintosh Intel running OS X: minimum 2 GB RAM Required, 4 GB 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. For StreamPlayer playback, a 7200 RPM Non-system drive or External firewire drive is required. The capacity of the hard drive depends on the number of encoded files that the user will save. Nominally, a lossless encoded file consisting of 8-channels of input material at 24 bit/48kHz resolution for a duration of 2 hours will occupy approximately 4GB 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 DTS-HD StreamPlayer, for optimum playback performance, the audio and video files should be located on separate non-OS drives. Playback performance may be significantly impacted if audio playback occurs from files that are resident on the system drive.

2.4 iLok Usage

The DTS-HD StreamPlayer is an iLok-enabled software product that utilizes an iLok Smart Key. The Smart Key contains the license and authorization necessary to activate and run StreamPlayer. Simply insert the Smart Key into the USB slot and the system is ready for use. It is strongly recommended that once StreamPlayer is running, the iLok Smart Key remain in the USB slot. Removing the Smart Key while StreamPlayer is running may result in the run-time application errors. The license 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. Encoder / Tools Installation Instructions

The DTS-HD Master Audio and DTS Surround Audio Suites encoder and DTS StreamTools installation packages contain 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
Setup.exe	**DTSInstaller.dmg**
1) Double Click on Setup.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 6 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) The Java run-time environment comes pre-installed on all Apple OS's. Please run System Update to ensure the latest version of Java is installed. MAS requires default settings of Java Application run-time environment. Custom settings can result in MAS not booting.

Upgrades from Surround Audio Suite to Master Audio Suite may be purchased through the DTS online store at http://www.dts.com.

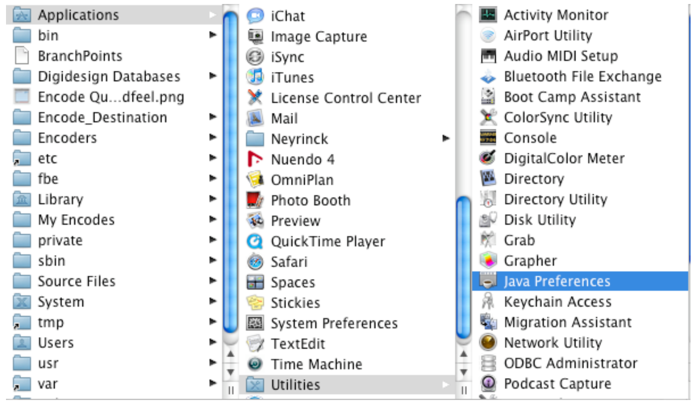
If you are experiencing problems installing or running the software, please contact DTS customer service at: proaudioinfo@dts.com

The DTS-HD StreamPlayer package contains 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

Mac OSX 10.5 Install Java Preference Requirement:

To run MAS properly on Mac a Java preference must be set to default.

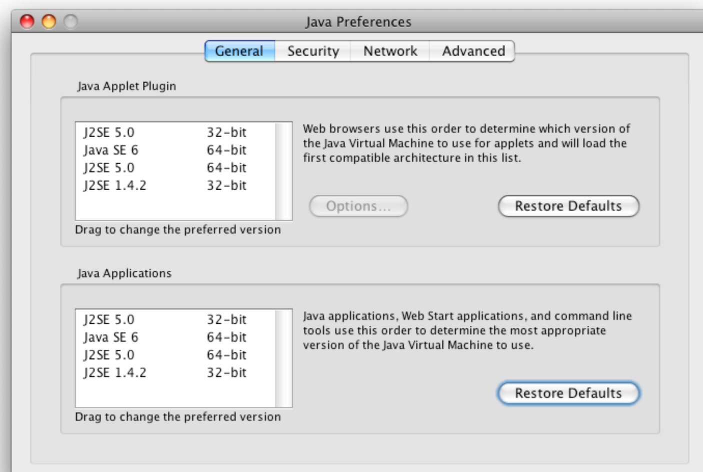
1) Open a Finder window
2) Navigate to Applications → Utilities → Java



3) Double click on the 'Java Preferences' application to open it

4) Click on the 'General' tab

5) Under the bottom 'Java Application versions' click the 'Restore Defaults' button which will place the 'J2SE 5.0 32-bit' entry to the top of the list



6) Quit the Java Preferences application

☑ **Warning: Do not move or modify the install path on Mac Intel**

International Character Support

The MAS Suite supports international characters for Input and Output files. While some dialogs will also display International characters properly not all will as MAS has not been internationalized from a graphics standpoint.

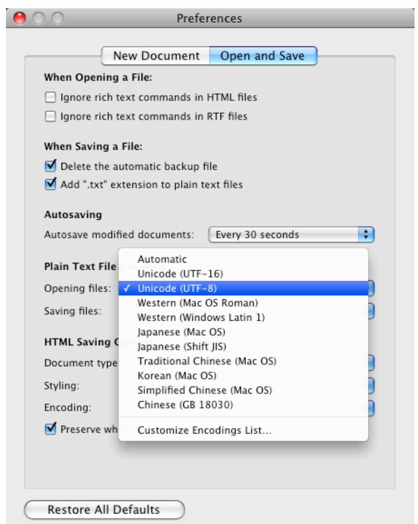
☑ **Warning: Only Standard English characters are supported for Channel ID's used in Auto import and Folder Based Encode functions. Non-English characters are NOT supported and 2 byte English characters are not supported. If Standard English characters are not used for the Channel ID of each channel, the function will not work!**

Mac Users additional steps for International Character Support:

Mac users that wish to properly view Log files should make the following changes.

Mac International Character Support Changes

1) Open Text Edit application in Applications directory
2) Select Menu item 'TextEdit → Preferences'
3) In the Preferences window select the 'Open and Save' tab
4) Under the 'Plain Text File Encoding – Opening Files' dropdown box select 'Unicode (UTF-8)'



5) Quit TextEdit
6) Open Safari browser application (if you do not use Safari please make this change in whatever browser you are using)
7) Select Menu item 'Safari → Preferences'
8) In the Preferences window select the 'Appearance' tab
9) In the 'Default Encoding:' dropdown box select 'Unicode (UTF-8)'

10) Quit Safari

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.

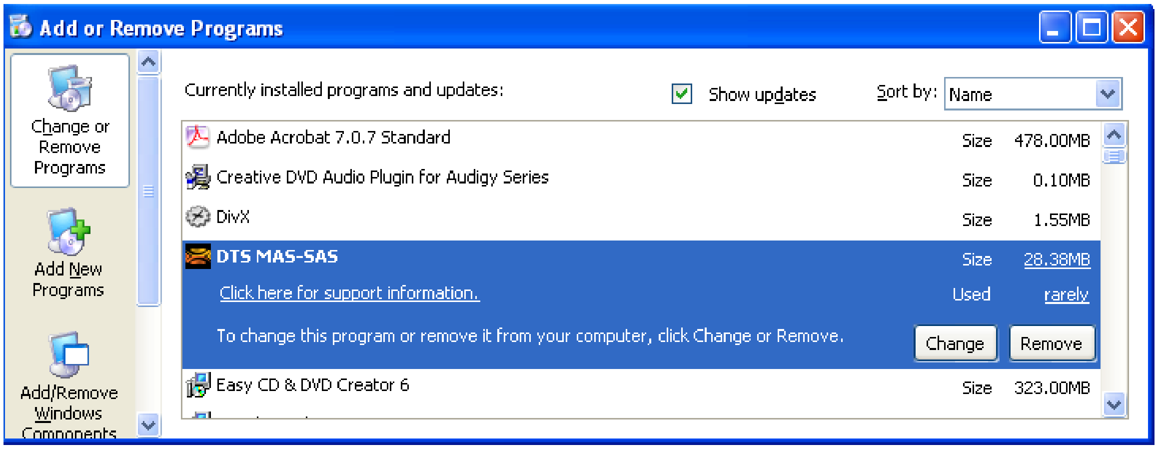


Figure 4-1 Windows Add/Remove Programs

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.

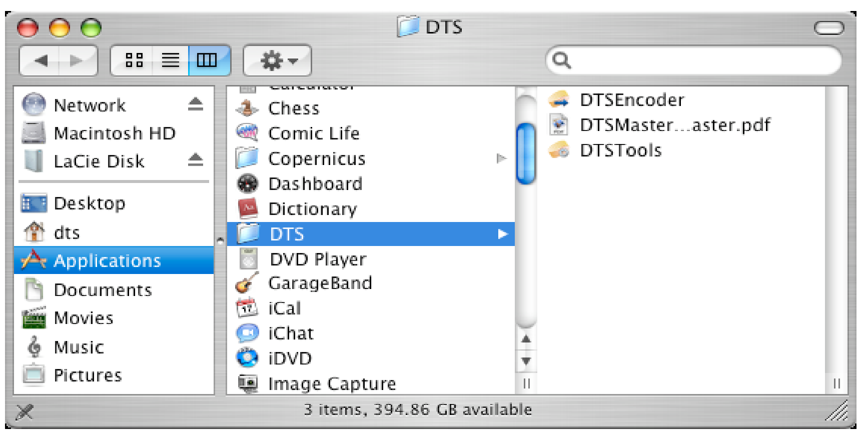


Figure 4-2 Macintosh Remove

5. Common Terms and Abbreviations

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

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,:
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	DTS-HD Express - Low Bit Rate technology used with Secondary and Sub-Audio
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

Table 5-1 Common Terms and Abbreviations

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.

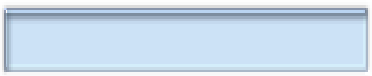
User Interface Element	Description
	Input text field; may have double-click capability to access a file browser window for selecting input files. ☑ **Note**: Tabbing out of an input text field may not 'commit' the text that is changed. It is highly recommend to use the 'Enter' key on the keyboard to 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.

Table 6-1 User Interface Element Descriptions

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. With all of the required data on one screen, and with drag-and-drop support for loading the input source files, the user will be able to efficiently create a DTS encoded stream.

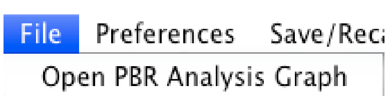
The  in the upper right-hand side of the interface, when pressed, will launch a browser window displaying the on-line User's Guide where all of the details about the functionality of this product can be found.

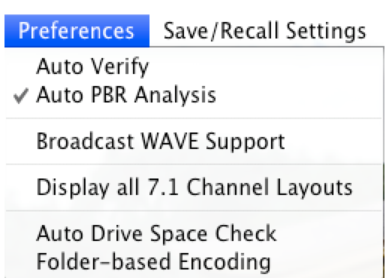
The About Button in the upper right-hand corner, when pressed will display a window showing specific information about the encoder such as build version and revision number, build date, and email contact information for customer support. Figure 7-3 provides an example of the splash screen

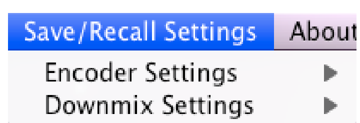
Figure 7-3 About Splash Screen

7.1 Main Menus

The Main 'File' menu holds the open and exit options. The Options menu contains preference settings for operations that will automatically perform after an encode has completed. The Save/Recall Settings menu houses various save and recall settings that the encoder offers.







7.1.1 Open PBR Analysis Graph

When the Open PBR Analysis Graph File menu item is selected, an open dialog box will appear. Navigate to the desired directory, select a '.dtspbr' file, and click 'Open.' For more details about the Peak Bit Rate tools, please look at Section 10.8 Peak Bit Rate Analysis.

7.1.2 Auto Verify

When the Auto Verify preference is checked ✓ Auto Verify, the encoder will automatically add a verify job directly following the actual encode. If left untouched, the Verify job will begin as soon as the encode job has completed. When this function is turned off, the encoder will only perform a MD5 validation check on the encode. For more information about Auto Verify jobs and MD5 validation checks please look at Section 7.1.22 Verify Jobs and Section 7.1.24.1 MD5 Validation.

7.1.3 Auto PBR Analysis

When the 'Auto PBR Analysis' preference is checked ✓ Auto PBR Analysis, the encoder will automatically create a Peak Bit Rate file as the Encode is processing. (This process generally does not cause a notable change in encode time on most machines) The Peak Bit Rate file will share the same file name as its related user specified encode file name but will have a '.dtspbr' extension. It will be located in the same destination directory as its related encode. Once the encode is complete, a user can view the graph by selecting Open PBR Analysis Graph from the 'File' menu of the encoder. The graph can also be opened in the StreamTools application. For more details about the Peak Bit Rate tool and use, please look at Section 10.8 Peak Bit Rate Analysis.

7.1.4 Broadcast Wave Support

When the 'Broadcast Wave Support' preference is checked ✓ Broadcast WAVE Support, if a Broadcast Wave file is loaded into the encoder, the encoder will automatically update the Timecode fields based on the timecode setting within the loaded file. For more details about this please look at Section 7.1.13.1 Broadcast Wave File Timecode Extraction

7.1.5 Display all 7.1 Channel Layouts

The Encoder will default to only show the 2 most popular 7.1 Channel Layouts. When the 'Display all 7.1 Channel Layouts' preference is checked ✓ Display all 7.1 Channel Layouts, the encoder will display all available 7.1 Channel Layouts.

7.1.6 Auto Drive Space Check

When the ✓ Auto Drive Space Check preference is checked, the encoder will check the available disc space for the destination directory against the estimated disc space require for the resultant encode. If not enough free disc space is detected a warning message will display informing the user about the lack of free space and will present an option to either continue with the encode and risk running out of space or to abort the encode, free up space on the drive, and try again. If 'Auto Drive Space Check' is not turned on the encoder will not check for available disc space when the encode button is pressed.

7.1.7 Folder-Based Encoding

When the ✓ Folder-based Encoding is checked, the function of the input and output file directories will change from a single file input output model to a source and destination folder model.
First, a confirmation pop-up message will display after clicking the Encode button warning the user that Folder-based Encoding is enabled. If the user clicks CANCEL, no encoding will occur.
Once the user clicks OK, the encoder will search the Destination Directory identifying valid mono audio file groups in the same fashion that the 'Auto Import' feature loads a group of files.
All valid audio file groups will be encoded with the same settings (e.g. format, stream type, channel count, sampling rate, bit-depth, bitrate, downmix(es), frame rate, start time, etc.) used for the initial encode; the only differences will be the input files and the duration (end time), which may differ depending on the length of the input files. The longest source file of each file group will determine each individual encode length.

Please read the (15. Folder Based Encoding Supplement) at the bottom of this manual for details on this function.

- ☑ **Auto Drive Space Check will disable when Folder-Based Encoding is enabled**
- ☑ **Auto Verify will disable when Folder-Based Encoding is enabled. MD5 validation checks are still done in triplicate during each files encoding process.**
- ☑ **Folder-Based Encoding ONLY works for mono audio files (single channel). Stereo (and above) is not supported.**
- ☑ **The Start time of the first encode (the setting used when the Encode button was pressed) will be used for ALL encodes. Broadcast Wave File's will have no effect on changing the Start Time of subsequent Folder-Based Encodes.**

7.1.8 Encoder Settings

These menu items, found under the 'Save/Recall Settings' menu, allow all encoder settings to be saved to a preferences file, which can later be recalled. The Save Current Encoder Settings menu option will open a Save dialog box for the user to choose where the '.dtshdsettings' file will be created. This file will save all user settings available in the encoder with the single exception of the input files section.
The Load Encoder Settings menu option will display an Open dialog box which allows a saved '.dtshdsettings' file to be recalled. Once recalled, all parameters will be restored to the saved state.

☑ **Note:** Recalling a saved settings file with source material different from the original saved file might result in un-desired timecode field settings.

7.1.9 Downmix Settings

These menu items, found under the ‘Save/Recall Settings’ menu, allow all downmix parameters to be saved to a preferences file, which can later be recalled. The Save Downmix Settings menu option will open a Save dialog box for the user to choose where the ‘.dtshddmx’ file will be created. This file will save all user settings available on the Downmix panel. The Load Downmix Settings menu option will display an Open dialog box which allows a saved ‘.dtshddmx’ file to be recalled. Once recalled, all downmix parameters will be restored to the saved state.

7.2 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 AUDIO tab 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, recommended by DTS)
7. Select the desired Bit Rate. Note that for DTS-HD Master Audio encodes that the Bit Rates selector specifies the Core (substream) backwards compatible constant bit rate. DTS recommends using 1509 kbps. In the DTS-HD High Resolution case, the Bit Rates selector specifies the total bit rate for this stream type with a 1509 kbps @ 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.

To enter Program Information or customize Secondary Audio parameters, select the BITSTREAM tab. Default Secondary Audio parameters are automatically selected prior to encode. Should default values require alteration prior to encode, the user may change these settings by activating the bitstream control page.

If downmix processing is desired, selecting the DOWNMIX tab 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.

7.2.1 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 Suite, only DVD and DTS Music Disc Destination Formats will be selectable.

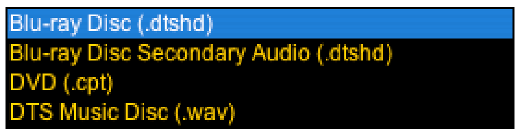


Figure 7-4 Destination Formats

7.2.2 Layout

This section of the Audio panel, (as shown in Figure 7-4), 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.

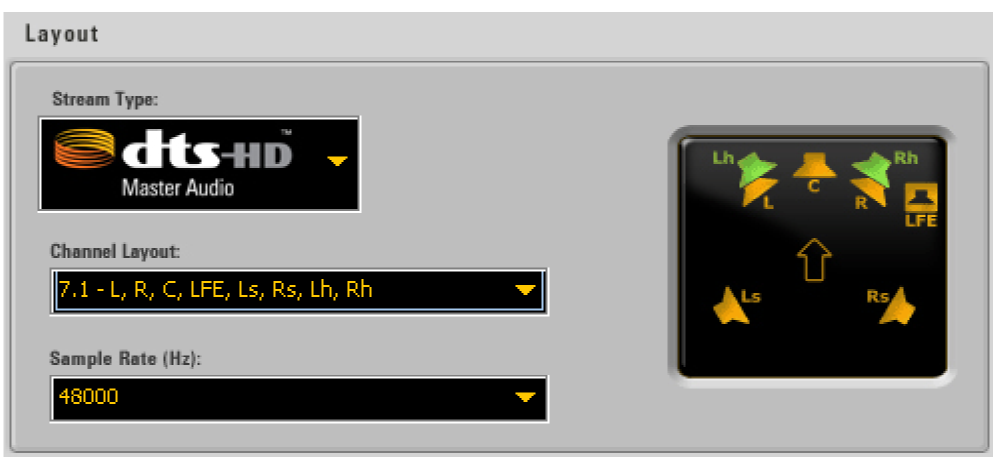


Figure 7-4 Layout Section

7.2.3 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 destination format. For 192 kHz source material, supports 2.0 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. Only available in Blu-ray Disc destination format.
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
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 for Blu-Ray and DVD destination formats.
dts™ Digital Surround	DTS Digital Surround up to 5.1 channels, at bit rates up to 1.5 Mbps for DVD. Available with all destination formats except for Blu-Ray Secondary Audio.
dts™ Express	DTS-HD Express (Low Bit Rate). For Blu-Ray Secondary Audio up to 5.1 channels.

☑ **Note:** For DTS Surround Audio Suite, only **DTS Digital Surround | ES**, **DTS Digital Surround | 96/24** and **DTS Digital Surround** will be selectable.

7.2.4 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.

Table 7-2 Channel Layouts

Primary	Secondary Audio DTS Express
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 Discrete – L, R, C, LFE, Ls, Rs, Cs	
6.0 Discrete – L, R, C, Ls, Rs, Cs	
6.1 Matrix – L, R, C, LFE, Ls, Rs, Cs	
6.0 Matrix– L, R, C, Ls, Rs, Cs	
5.1 – L, R, C, LFE, Ls, Rs	5.1 – (Express)
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 (Express)
2.0 – Lt/Rt	
1.1 – C, LFE (DVD Only)	
1.0– C (Mono)	1.0 – C (Express)

☑ **Note:** DVD destination format is limited to a maximum of 6.1 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

7.2.5 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 Express	48 kHz

7.2.6 Input Files

The Input Files that are required for the encoding process must be loaded into this section of the user interface (see Figure 7-5 and Figure 7-6). 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, Broadcast LPCM wave (.wav) files, and LPCM Audio Interchange File Format (.aif).

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. If you need to clear all of the input files, the “Clear All” button will clear all of the input files in one step.

☑ **Note:** Input files may have varying durations. The encode process will encode up to the longest file duration. For example, in Figure 7-6, the input files have duration of 00:00:10.000 and 00:00:30.030. The process will create an encoded file up to 00:00:30.030 in length. The shorter files will be padded out with 0’s (-INFdB).

☑ **Note**: The input source material may be 16-bit or 24-bit resolution mono or stereo in ‘.wav’ ‘.aif’ or ‘.aiff’ format only.

7.2.7 Auto Import

Upon import of the Left (L) channel audio file, the MAS Encoder will populate the remainder of input fields with audio files that follow the guidelines outlined below. This process occurs automatically and dynamically according to the currently selected channel layout. Refer to Table 7-4-1 Supported Channel IDs and Definitions for supported channel IDs.

1) The Encoder will only search for matching files within the same directory as the "L" channel audio file. Parent and sub-directories are ignored.

2) Excluding the channel ID, all input file names must be identical for each encode.
   Example: Audio_L.wav, Audio_R.wav, Audio_C.wav, etc.

3) Punctuation: a period, dash, underscore, or space should be used to separate the channel ID, which can exist at any point within the file name.
   Examples: 'AudioChannel_L.wav', 'Audio_L_Channel.wav', 'AudioChannel.L.wav', 'AudioChannel L.wav', 'Audio_L-Channel.wav', or 'Audio L Channel.wav'

4) If the Encoder cannot locate a file, it will leave the field empty and continue to the next active field.

Table 7-4-1 Supported Channel IDs and Definitions

Channel ID	Definitions
L	Left Speaker
R	Right Speaker
Lt	Left Total
Rt	Right Total
C	Center Speaker
LFE, LF, or SW	Low Frequency Effects / Subwoofer
Ls	Left Surround Speaker
Rs	Right Surround Speaker
Lsr	Left Surround Rear Speaker
Rsr	Right Surround Rear Speaker
Lss	Left Surround Side Speaker
Rss	Right Surround Side Speaker
Lw	Left Wide Speaker
Rw	Right Wide Speaker
Lh	Left Height Speaker
Rh	Right Height Speaker
Lhs	Left Height Side Speaker
Rhs	Right Height Side Speaker
Cs, S	Center Surround Speaker
Ch	Center Height Speaker
Oh	Overhead Speaker

7.2.8 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 7-5 this is shown as L, Lss, C, and Lsr) 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 7-5 this is shown as R, Rss, LFE, and Rsr) 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.

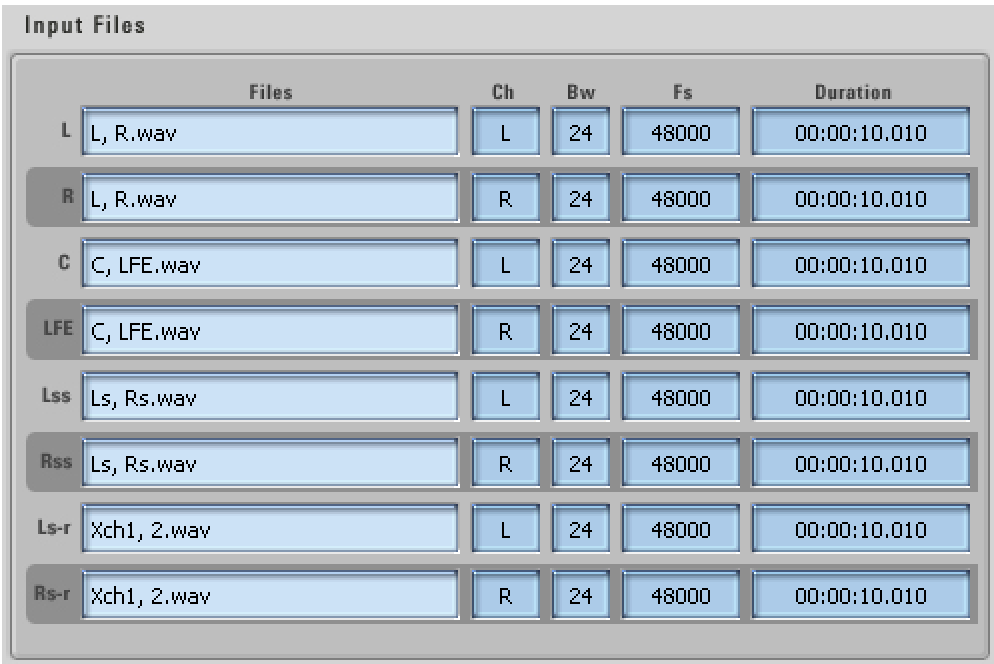


Figure 7-5 Input Files Section (Stereo Case)

b) 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 7-5 above), then in the singleton cases, only mono input files will be allowed. For example, in Figure 7-6, 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.

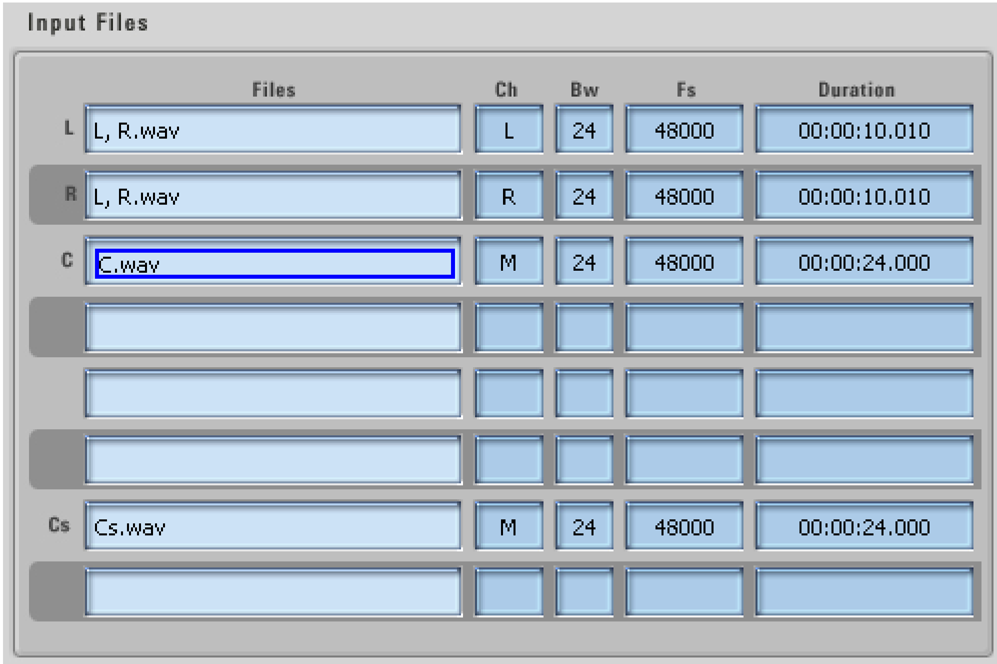


Figure 7-6 Input Files Section (Stereo and Mono Case)

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.

Clear All

Located below the Input Files section of the GUI, the 

button will unload all active input files from their related channel inputs when clicked.

7.2.9 Dialog Normalization

Located in the upper right side of the Audio panel, the Dialog Normalization menu allows a value to be selected for the specified encode (as shown in Figure 7-7). 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. At current, all 6.x Discrete channel layouts require a -31 dBFS Dialog Normalization setting. Dialog Normalization is a post-process operation performed by the DTS decoder. DTS recommends leaving the Dialog Normalization setting at -31.

Figure 7-7 Dialog Normalization

7.2.10 Constant Bit Rate Parameters

The Constant Bit Rate Parameters section allows the user: to specify the bit rate to be used in the core substream, to select Rear Ch -3 dB attenuation, to perform ES phase shifting and to inform the encoder that the input material for a 6.1 ES Matrixed channel layout has been pre-mixed (indicated by selecting the ES Pre-Mixed button). Figure 7-8 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.

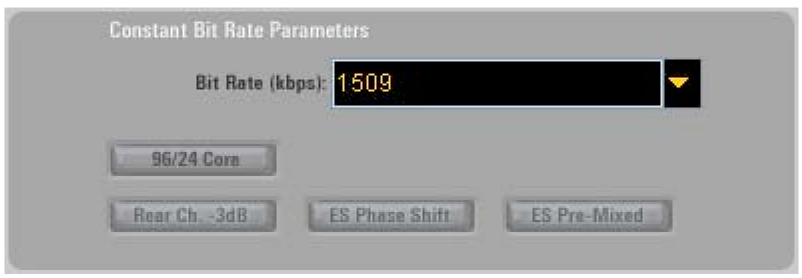


Figure 7-8 Constant Bit Rate Parameters

7.2.10.1 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	dts-HD™ Master 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** can range from 192 to 1509 kbps. (Channel layout dependent) ☑ **Note:** For channel configurations 5.1 and above, the core bit rate is limited from 768 through 1509 kbps (inclusive)
Blu-ray Disc Primary Audio	dts-HD™ 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. Bit rates can range from 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 are used as input files
Blu-ray Disc Primary Audio **DVD-V** **DTS Music Disc**	dts™ Digital Surround | ES	For 6.x discrete and Matrix channels, supported constant bit rates can range from 754 to 1509 kbps.
Blu-ray Disc Primary Audio **DVD–V**	dts™ Digital Surround | 96/24	Channel layouts can range from 1.0 to 6.1 ES Matrix. With constant bit ranging from 384 to 1509 kbps.
Blu-ray Disc Primary Audio **DVD-V** **DTS Music Disc**	dts™ Digital Surround	Stereo through 5.1 channels layouts that may be encoded at bit rates ranging from 255 to 1509 kbps
Blu-Ray Secondary Audio **HD DVD Sub Audio**	dts™ Express	Mono, Stereo, and 5.1 channel layouts. Constant bit rates can range from 24 to 256 kbps.

☑ **Bit Rate ranges** given will all be further limited based on the specific channel layout chosen. In general, the small channel layouts will use the smaller bit rates and the larger channel layouts will use the larger bit rates within the given range.

7.2.10.2 Rear Ch -3dB Attenuation

This option is available for 6.x Discrete, 6.x Matrix (Pre-Mixed), and 5.x channel layouts in all Stream Types except DTS-HD Master Audio. When this button is illuminated, Rear Ch. -3dB the audio provided for the left surround Ls and right surround Rs channels (and center surround Cs when applicable) will be attenuated by 3 dB prior to the encoding process.

7.2.10.3 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 and cannot be used in conjunction with ES Pre-Mixed. 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 and 6.0 Discrete encodes.

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

7.2.11 ES Pre-Mixed

This option is only available for 6.1 or 6.0 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 of the Center Surround Cs during the encoding process.

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

7.2.12 96/24 Core

This option is only available if the Stream Type is Master Audio, the sample rate is 96kHz or higher, the Channel layout is 6.1 Matrix, 5.1, 5.0, or 2.0, and the bit rate is 1509kbps. When these conditions are met the '96/24 Core' button will un-gray. Selecting this option 96/24 Core adds the 96/24 extension stream into the encoding in addition to the lossless Master Audio and lossy Digital Surround streams; thus, when playing back a stream that was encoded with this option on using a DTS Digital Surround decoder, the lossy 48kHz core portion of the stream will decode. When playing back the stream on a DTS 96/24 capable decoder, the lossy 96/24 extension core portion of the stream will decode. And when playing back the stream on a DTS Master Audio decoder, the lossless portion of the stream will decode. All three decode scenarios given play from the same encode.

7.2.13 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 7-9). The default setting is to encode the entire source file. For DTS-HD encodings, the timecode settings are stored in the encoded output file.

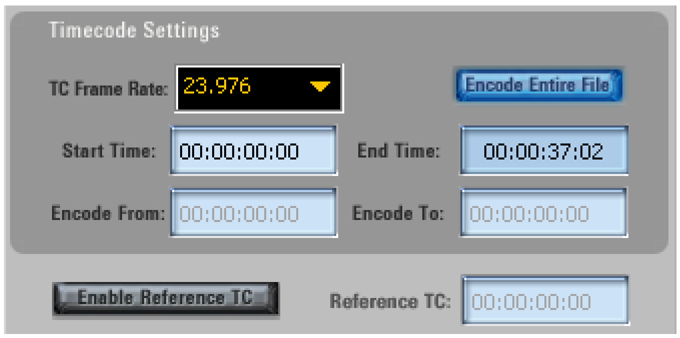


Figure 7-9 Timecode Settings Section

7.2.14 Broadcast Wave File Timecode Extraction

When the ✓ Broadcast WAVE Support option is selected in the Preferences menu the automatic timecode extraction feature is enabled. With this feature enabled, if a Broadcast Wave file is Imported into the file input section the MAS Start Time timecode field will automatically update with the 'original start time timecode stamp' that is contained within the Broadcast wave file.

☑ **Note:** Timecode Frame Rate **Must** match the original frame rate that was used when the file was created to re-produce the same Start Time.

7.2.15 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, 29.97 DROP, 30 and 30 DROP.

7.2.16 Encode Entire File

When illuminated (default setting) Encode Entire File**,** the input files will be encoded in their entirety and the **Encode From** and **Encode To** text fields will be disabled.

When not illuminated, Encode Entire File 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 by these fields.

7.2.17 Start Time

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

7.2.18 End Time

The **End Time** is read-only field and specifies the end time relative to the **Start Time** based on the duration of the input files. The value shown is based on the duration of the shortest used source file present in the input files section.

7.2.19 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** button is in the off state. 

This allows the user to create a smaller encode relative to the 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.

7.2.20 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** button is in the off state. 

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

7.2.21 Enable Reference Time

The encoder application supports a mechanism by which two or more encoded files can be losslessly 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 enabled, 

the Reference TC text field becomes active allowing the user to enter a custom reference timecode. The specified Reference TC 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 encodes reference time equal to its start time. *(see* Tools *section* Join Tool *for more information on Reference TC).* Reference time is only necessary for Join operations.

☑ **Note:** If 

is off, then the Reference time of the encode will equal the Start Time of the encode.

☑ **Caution:** A DTS-HD encoded stream whose reference time **do not** equal its 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.

7.2.22 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 placeholders. 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 7-10 and Figure 7-11 show examples where the hours and minutes field contained an erroneous input.

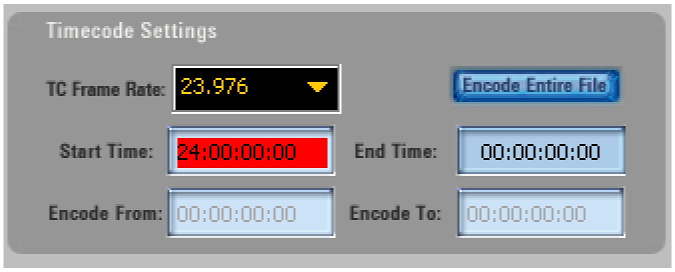


Figure 7-10 Timecode Error (hours)

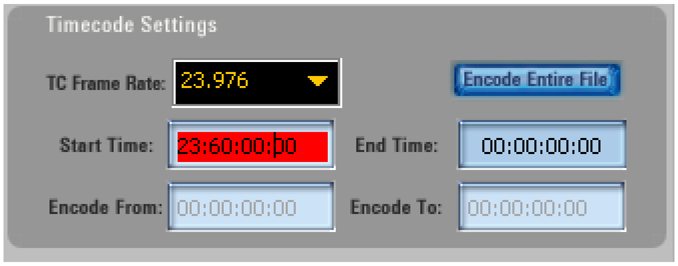


Figure 7-11 Timecode Error (minutes)

7.2.23 Diagnostics and Output Section

The lower right portion of the Audio panel, as shown Figure 7-12, 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.

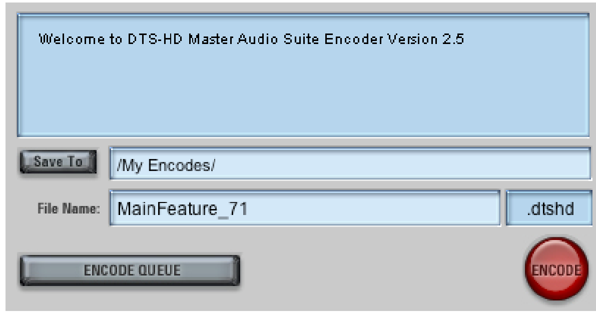
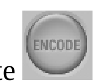
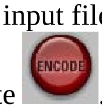
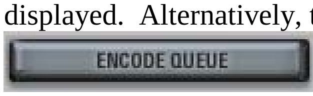


Figure 7-12 Diagnostics and Output

The  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 and DTS Music Disc. For the DVD destination format the filename extension will be ".cpt". For the DTS Music Disc destination format the filename extension will be ".wav".

The encode button will be in the inactive state  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 . When pressed, if a path or filename is not included in the  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. If the encode queue window is not active, when the  button is pressed, the encode queue window will be automatically displayed. Alternatively, the Encode Queue window may be activated by pressing the  button.

If there is an error with the encoder, a message will appear in the Encode Queue after the Job name that is listed. The Job itself will highlight Red, a red X will be placed to the left of the encode, and the right Progress bar area will say 'Error.' While the error message will display in the Queue window it is generally too long to be read here. If the user opens the log file by clicking on the info button to the right of the Failed Encode and scrolls to the bottom of the log file, the entire Error message will be displayed there.

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.txt" as its filename extension.

7.3 Bitstream

When the Audio panel is displayed, selecting the BITSTREAM tab will activate the Bitstream Panel as shown in Figure 7-13 Bitstream Panel. The Panel is divided into independent sections.

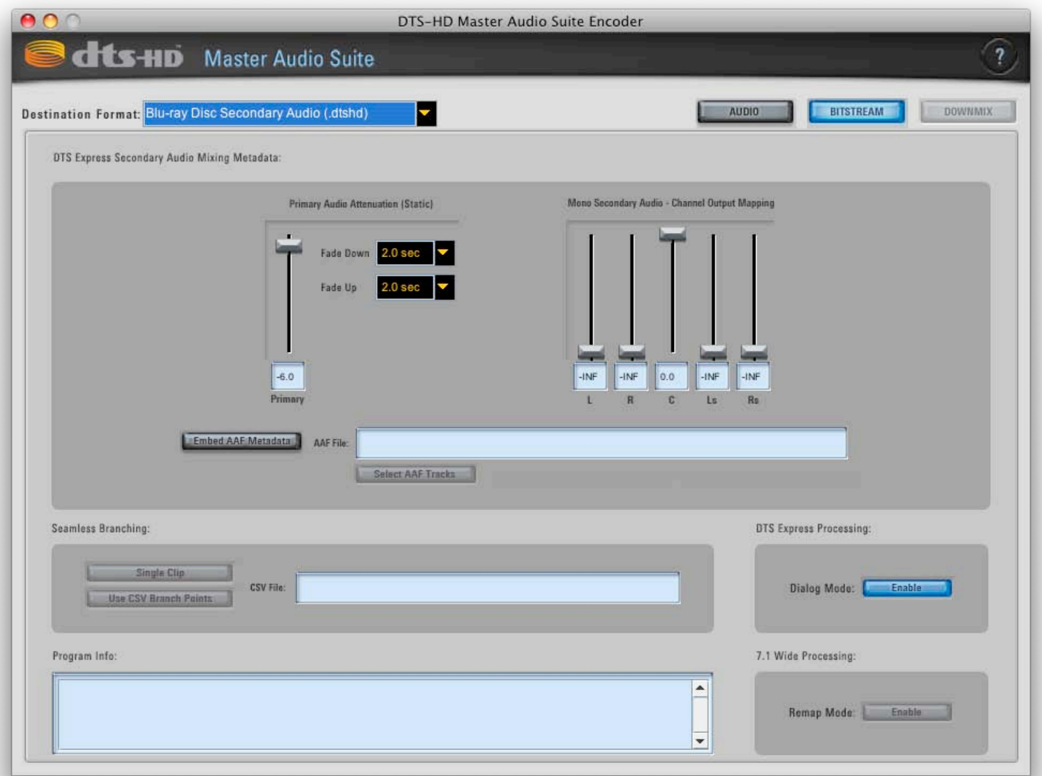


Figure 7-13 Bitstream Panel

7.3.1 Secondary Audio Mixing Metadata

The Secondary Audio Mixing Metadata section allows for manipulation of Blu-Ray Primary and Secondary Audio DTS Express Streams. All fader information is placed into the Secondary Audio Stream including the Primary Audio Attenuation fader information. The Primary Audio Stream does not contain any information about the Secondary Audio Stream. The Primary Stream will be affected by the information within the Secondary Audio Stream if the two streams are played together.

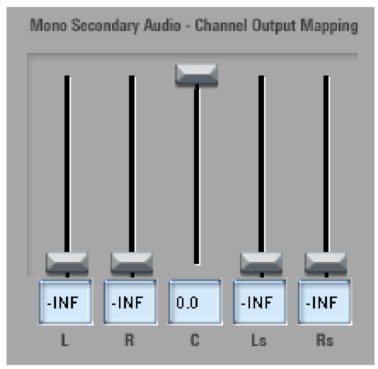
☑ **Note:** All faders are static.

7.3.2 Mono Secondary Audio – Channel Output Mapping

These 6 faders are only active for Blu-Ray Secondary Audio DTS Express Mono streams. Mono Secondary Audio - Channel Output Mapping (1.0 channel layout only) This function allows a user to place the single input audio channel into a 5.1 output configuration. (This can be used for 5.1 panning) All 6 faders receive input from the Center speaker input file and they allow the user to adjust the output gain of each individual speaker in a 5.1 setting. Each fader has a dynamic range from 0 to -60 (0 to -60 dBFS) inclusive or INF. The default setting will use only the Center channel at full input gain.

Note: These faders only affect the Secondary Audio streams playback mapping. They do not affect the playback of the Primary Audio stream in any way.

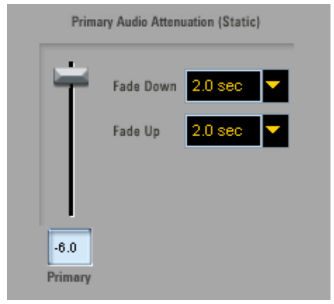
Note: At least one of the 6 faders must be greater than 'INF' to create an encode.



7.3.3 Static Primary Audio Attenuation Metadata

This fader controls the attenuation of the Primary Audio Stream Primary Audio Attenuation Metadata. The attenuation data is held in the Secondary Audio stream. When the Primary and Secondary Audio streams are played together, this attenuation data will statically attenuate the Primary Audio stream by the user specified amount. Fader coefficients range from 0 to -60 (0 to -60 dBFS) inclusive with -INF. The default is -6 dBFS.

☑ **Note:** Because the attenuation data is contained in the Secondary Audio stream, any Primary stream that is played with the Secondary Audio stream will be attenuated by the coefficient amount contained within the Secondary Audio stream. There is no direct bond between Primary and Secondary streams. A Primary stream can be paired with any Secondary Audio stream and visa versa; they are interchangeable.



7.3.4 Primary Audio Fade Down/Up for Static Attenuation

Next to the Primary Audio Attenuation Fader are two menus which are used to set Fade Down and Fade Up time spans for the Beginning and End of the Secondary Audio File. Each menu allows a Fade range from 0 Sec to 9 Sec at 0.5 Sec intervals.

☑ **Example:** If the Primary Attenuation Fader is set to -6dB and the Fade Down Menu is set to 2.0 sec then when the Secondary Audio file begins playback in it is related Primary stream it will start with Primary Attenuation at 0 dB and fade down to -6dB over the 2 Seconds specified by the user. The Secondary stream will continue playback with Primary Attenuation set to -6dB from the user specified 2 seconds forward.

☑ **Example:** If the Primary Attenuation Fader is set to -6dB and the Fade Up Menu is set to 4.0 sec then the related Primary stream which as been playing with a Primary Attenuation of -6dB at 4 Seconds prior to the end of the Secondary Audio File will begin to fade up to -0dB over the last 4 Seconds of the Secondary Audio File.

7.3.5 DTS Express Dialog Mode

DTS Express Dialog Mode: enhances the quality of dialogue-based DTS Express encodings (for Blu-ray Secondary Audio). If input material is mostly dialogue or contains excessive high frequencies, it is recommended that DTS Express Dialog Mode to the Enable state.

☑ **Note:** DTS Express Dialog Mode may not be appropriate for all dialogue-based input material.

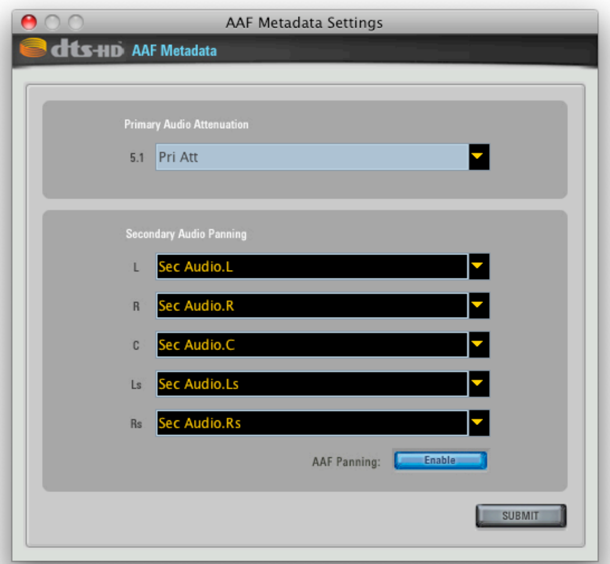
7.3.6 AAF File Import – Primary Audio and Panning Automation

When Blu-ray Secondary Audio is selected as the Destination Format the Embed AAF Metadata button on the Bitstream panel will un-gray. AAF files can then be loaded into MAS and used to control the volume automation of the Primary Audio Attenuation control. When an AAF file created by ProTools is loaded into the AAF file field to the right of the Embed AAF Metadata button or when the Embed AAF Metadata button is selected, the AAF file field will become active and a browser window will display requesting the selection of an AAF file. When an AAF file is loaded the AAF Metadata Settings Window will automatically appear. Once the desired channel layouts have been selected and Submitted, the AAF Metadata Settings window will close. The Select AAF Tracks button will bring up the AAF Metadata Settings window if further changes are needed. Please look at Section 14 - AAF File Import Supplement. for more information about the AAF Metadata Settings Window.

Please look at Section 14 - AAF File Import Supplement for detailed instructions on how this feature should be used as well as the proper way to create AAF Files.

> ☑ **Note:** All users attempting to use the AAF File feature should read and follow the instructions at Section 14 - AAF File Import Supplement.

7.3.7 AAF Metadata Settings Window



The AAF Metadata Settings Window holds AAF specific settings. With the Embed AAF Metadata button selected, the Primary Audio Attenuation (Static) fader will gray out and be completely ignored at Encode time. Instead, volume automation pulled from an Automation track contained within the AAF file will be used to dynamically control the Primary Audio Attenuation Metadata contained within the Secondary Audio stream. The 5.1 menu under the Primary Audio Attenuation heading is used to select the Automation track that will be used for Primary Audio Attenuation.

When the AAF Panning Metadata: Enable button is selected, the Mono Secondary Audio - Channel Output Mapping static faders will gray out and be completely ignored at Encode time. Instead, volume automation pulled from individual Automation tracks contained within the AAF file will be used to dynamically control the Mono Secondary Audio Channel Output Mapping contained within the Secondary Audio stream. Each channel menu under the Secondary Audio Panning heading is used to select the volume Automation track that will be used to control output volume for the given output track.

☑ **Example:** Selecting an automation track for the R channel under the Secondary Audio Panning heading will essentially take the mono audio input file (imported into the MAS Encoder) and map it out the right channel copying the volume rides that were used in the selected AAF track. The same will occur for all other channels.

Please look at Section 14 - AAF File Import Supplement for detailed instructions on how this feature should be used as well as the proper way to create AAF Files.

☑ **Note:** All users attempting to use the AAF File feature should read and follow the instructions at Section 14 - AAF File Import Supplement.

7.3.8 AAF Automation Use Summary

For clarification purposes, a visual summary of the current AAF settings are displayed directly under the AAF File field. This summary depicts the current state of Primary Audio Attenuation. Ex. Using AAF means the Resulting Encode will ignore the STATIC Primary Audio Attenuation Fader (and ramp Down/Up) and will instead use the selected track of the AAF file.

☑ Seeing Using AAF after a title means the Resulting Encode will IGNORE the related STATIC faders (and menus for Primary Attenuation) and will USE the volume automation information found in the track selected within the AAF Metadata Settings Window

☑ Seeing Static after a title means the Resulting Encode will USE the related STATIC faders (and menus for Primary Attenuation) and will IGNORE data with volume automation information that might have been previously selected within the AAF Metadata Settings Window

☑ Seeing NA after a title means the feature does not apply to the current setup and will therefore not be used

Please look at Section 14 - AAF File Import Supplement for detailed instructions on how this feature should be used as well as the proper way to create AAF Files.

☑ **Note:** All users attempting to use the AAF File feature should read and follow the instructions at Section 14 - AAF File Import Supplement.

7.3.9 Seamless Branching

The Seamless Branching section features two buttons and an input file field. The Single Clip button is used for Seamless Branching Encodes that do NOT have exit/entry points within the length of the file. The 

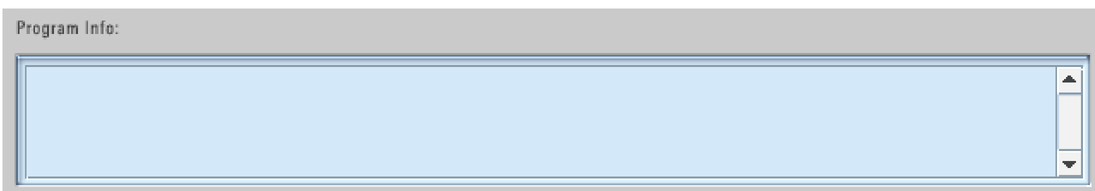
button enables the import of a CSV file which will contain a list of branch points used as exit/entry points within the encoded file.

Please look at Section 15. CSV File Import Supplement – Seamless Branching for detailed instructions on how this feature should be used as well as the proper way to create CSV Files.

☑ **Note:** All users attempting to use either Seamless Branching feature should have a clear understanding of how Seamless Branching works prior to use. If questions arise, Please read and follow the instructions at Section 15. CSV File Import Supplement – Seamless Branching

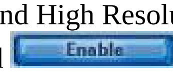
7.3.10 Program Info

The Program Information text field allows the user to type in any text data. All text entered in the field will be placed within the selected encode. All standard keyboard characters are accepted in the text field. The information in the resulting encode can be viewed using StreamPlayer or StreamTools File Info and Verify. Program Info is limited to Blu-Ray media type only. There is also a 2000 character limit to the program info. (DVD and DTS Music Disc formats do not support this feature)



7.3.11 Wide 7.1 Remapping Metadata

The 

button enables the preset Wide 7.1 speaker remapping configuration. This remapping is only allowed for Master Audio and High Resolution Streams with a channel layout of 7.1 – L, R, C, LFE, Ls, Rs, Lw, Rw. When enabled 

, this function effectively remaps the channel layout to that of a 7.1 – L, R, C, LFE, Lss, Rss, Lsr, Rsr layout. This will apply a preset speaker remapping that will change the 5.1 downmix mapping such that the Wide fronts (Lw and Rw) will be dropped to INF (not present in the 5.1 downmix) The remaining channels will downmix at a 1:1 ratio into their related channels. [see table below for details] The 2.0 downmix coefficients will remain the standard default. All custom Downmix settings will be overwritten with the preset values. The selected 7.1 – L, R, C, LFE, Ls, Rs, Lw, Rw channel mapping will become a 7.1 – L, R, C, LFE, Lss, Rss, Lsr, Rsr channel mapping for 7.1 playback.

7.1 to 5.1 preset downmix coefficients are as follows:

	L	R	C	LFE	Lw	Rw	Ls	Rs
L	0.0	INF	INF	INF	INF	INF	INF	INF
R	INF	0.0	INF	INF	INF	INF	INF	INF
C	INF	INF	0.0	INF	INF	INF	INF	INF

LFE	INF	INF	INF	0.0	INF	INF	INF	INF
Ls	INF	INF	INF	INF	INF	INF	0.0	INF
Rs	INF	INF	INF	INF	INF	INF	INF	0.0

7.4 Downmix Panel

When the Audio panel is displayed, selecting the 

tab will activate the Downmix Panel as shown in Figure 7-14, 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.x 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.x channels can only be performed when a 7.x channel layout is selected (6.x Discrete will use the built-in Legacy downmix). Downmixes to stereo can only be performed when the selected channel layout contains at least 5.x-channels.

Figure 7-14 Downmix Panel

7.4.1 Downmix to 5.1

The Downmix to 5.1 section of this panel allows the user to specify the appropriate parameters for the downmix as shown in Figure 7-15. This section will only be active if the selected channel layouts have 7.1 or 7.0 channels.

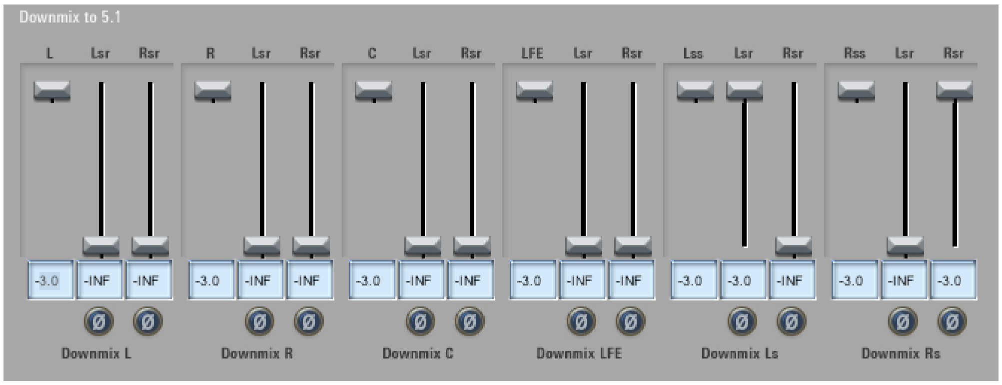


Figure 7-15 Downmix to 5.1

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 -6.0 (0 to -6.0 dBFS) inclusive. The mixing coefficients for the extra channels (i.e. Lh and Rh in Figure 7-15), known as Downmix Coefficients, range from 0 to -60.0 (0 to -60.0 dBFS) inclusive or -INF. INF implies that there is no contribution to the channel (-INF = Negative 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 -60.0, 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.x Discrete channel layout, only the 

is valid. If this button is not activated for the 6.x 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 Audio stream 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.

7.4.2 Downmix Saturation Check

The Downmix Saturation Check feature allows a user to check for Saturation (or clipping) in the 5.1 downmix. When the 

button is clicked, a File Save browser will display to allow a user to pick the output directory and file name for their saturation check results log file. Once a directory and file have been selected by the user, a new job will appear in the Encode Queue with an identical name to the file name they just chose. When run in the Queue, the Downmix Saturation Check will perform a 'fake' encode which will result in a log file containing all saturation warnings (clipping) that were encountered. No encoded file will be created.

- ☑ **Note:** A saturation or clip in the downmix will be recorded anytime 0dBFS is reached for 3 or more consecutive samples.
- ☑ **Note:** Downmix Saturation Check will be disabled when the 'Folder-Based Encode' function is active
- ☑ **Warning:** Downmix Saturation Check ONLY checks Saturation in the 5.x downmixes for 7.x (Master Audio and High Resolution) and 6.x Discrete (Master Audio) channel layouts.
- ☑ **Warning:** The Downmix Saturation Check runs a 'mock' encode based on the parameters of the Encoder at the time the 

button is clicked. If ANYTHING is changed that would affect the Downmix after the Saturation Check is performed, the previously run Saturation Check should be considered invalid. Each Final Encode log file should always be looked at to ensure the final 'Actual' encoded file does not contain Saturations in the Downmix. (All Saturations will be located towards to bottom of the related encodes log file)

7.4.3 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 7-16. This section will only be active if the selected channel layout has 5.1 channels.

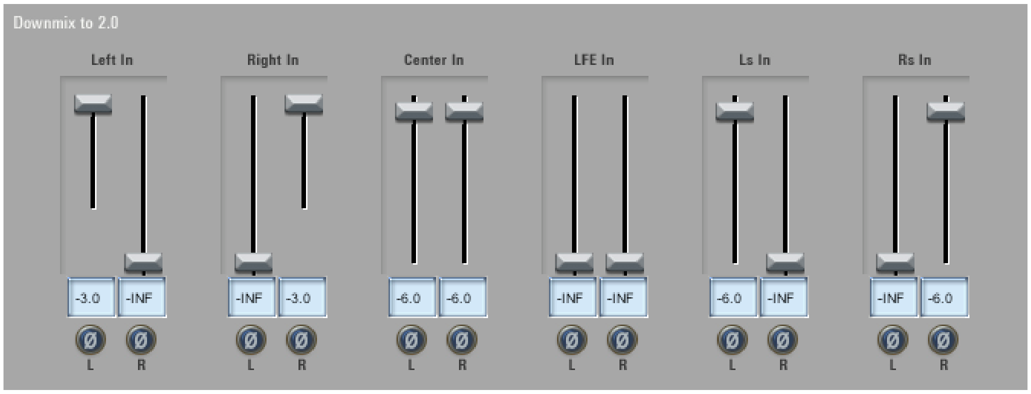


Figure 7-16 Downmix to Stereo Panel

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. Scaling coefficients (L -> L and R->R) range from 0 to -40dBFS for Lo/Ro downmixes. Scaling coefficients (L ->L and R ->R) are not used for Embed Downmixes.

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 -60.0, 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 be deactivated as this channel is not included in the selected channel layout

7.4.4 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.

Downmix Button States	Description
Reset Downmix to 5.1 Settings	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.
Use Legacy Matrix	If this button is illuminated, then the encoder will use the built-in legacy downmix coefficients at encode time.
Enable Downmix to 2.0	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.
Enable Downmix to 2.0	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.
Embed Downmix	If this button is illuminated, the stereo downmix is performed during the encoding process. Using the specified downmix coefficients, all channels are returned to their original state upon full decode. 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.
Embed Downmix	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.
Reset Downmix to 2.0 Settings	When this button is pressed, the settings for the Downmix to 2.0 section will be reset to their initial factory default states.
Save Settings	This Button Saves all Downmix parameters available on the Downmix Panel. It holds the same functionality as the Save Downmix menu option. (See 0 Save Downmix Settings)
Recall Settings	This Button Recalls a saved Downmix Settings file 'dtshddmx.' Same functionality as the Recall Downmix menu option. (See Section 0)
Saturation Check	This Button Checks for Saturation in the 5.1 downmix. (See Section 0)

Table 7-6 Downmix Buttons

7.4.5 Discussion on 6.1 Matrix Processing

DTS-HD Master Audio Encoder supports the ability to encode a sixth ‘phantom’ channel from a 5.1 channel layout. This is accomplished by selecting the 6.1 Matrix or 6.0 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 7-17.

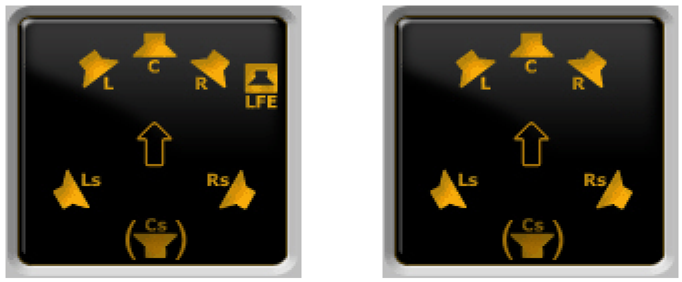


Figure 7-17 6.0 and 6.1 Matrix Phantom Speaker Layout

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 aid the user to better understand DTS 6.1 Matrix processing.

When the DTS-HD Master Audio stream type and 6.x 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 itself.

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

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

Note: Due to an issue in this version of the encoder library, when selecting a 6.x 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.

7.5 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.

A job can have one of six states: **In-Progress, Pending, Completed, Canceled, Passed, Error** or **Failed.** Figure 7-18 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.

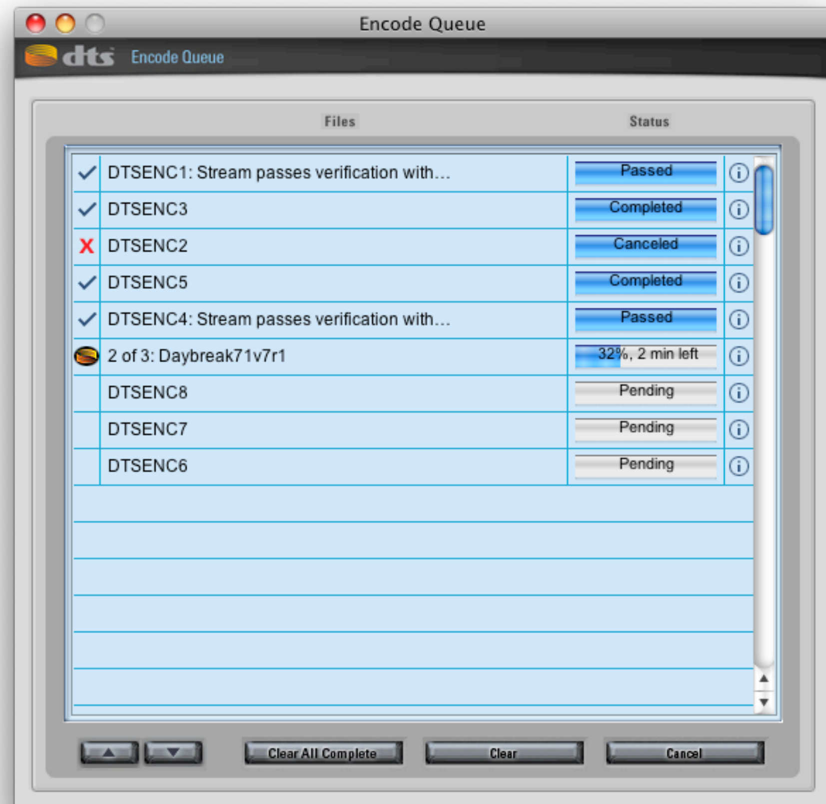
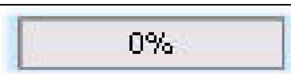


Figure 7-18 Encoder Queue

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 0 13. 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.

7.5.1 Encode Queue Control Buttons

The control buttons at the bottom of the page (see Figure 7-19) allow the user to manage jobs present in the Encode Queue.

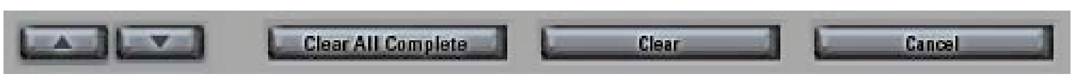


Figure 7-19 Encoder Queue Control Buttons

The  buttons at the bottom of the page (see Figure 7-19) allow the user to alter the priority of a single encode job currently in the queue that is in a '**Pending'** state. Folder-Based Encode jobs cannot be moved in the queue and single encode jobs cannot be moved passed them.

The  button, allows the user to lower the selected single encode jobs priority by moving it "down" in the queue. The  button, allows the user to raise the selected single encode 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 7-18 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  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  button. This will remove the job from the queue and will NOT remove the corresponding files stored on disc (i.e. log file, encoded file, etc.).

To clear all jobs that are not currently "**In-Progress"** or Pending click the button. Once clicked, all jobs that have Completed, Error'ed, Canceled, Passed, or Failed will be removed from the Queue window.

7.5.2 Encode Jobs

An Encode job is loaded into the Encode Queue by clicking the Encode Button located on the main MAS encoder Audio Panel. Once this Encode Button is clicked, the resulting encode job will be placed in the Encode Queue as the last job. An Encode job can found in four states: **In-Progress, Verifying, Pending, Passed, Completed, Error,** or **Canceled.** (See Figure 7-20 Encode Jobs)

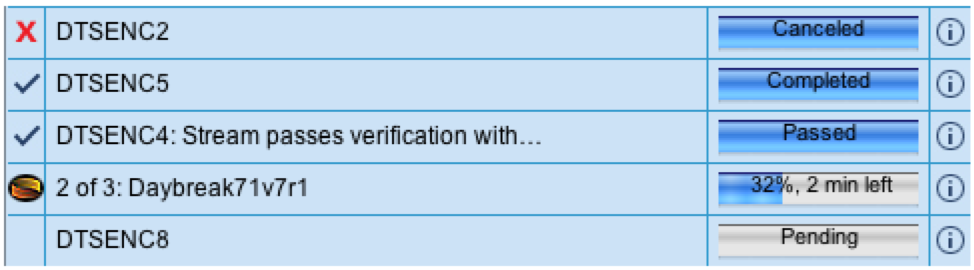


Figure 7-20 Encode Jobs

7.5.3 Verify Jobs

If the Auto Verify function is checked when the Encode Button is clicked, the resultant encode will be automatically run through the file verification process. (See Auto Verify) The Verify job will begin immediately upon encode completion and will display on same entry as the Encode job in the Encode Queue. When a Verify job begins, the output of the verify task will be appended to the verifying file name in the Encode Queue. Once the Verify job completes the 'Status' bar will display the results either **Passed,** or **Failed.** (See Figure 7-21 Verify Job**)** To manually verify a DTS file, please use the StreamTools Verify function. If a Verify job fails the related encode should be considered faulty and deleted.

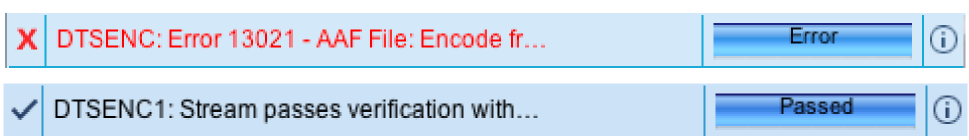


Figure 7-21 Verify Job

7.6 Log Files

Once an Encode job has completed it can be found at the destination path. A DTSHD file will contain all of the encode parameters used to create itself. This information can be found using the File Info Tool (Section 10.6) in the StreamTools program, but this information can be somewhat cryptic and difficult to decipher. (See StreamTools File Info portion of this manual for details) As an aid, the encoder automatically places a Log file that contains all information used to create the encoded file in the same Save To directory as the encode for future reference. A DTSPBR file can also be created in the same fashion by checking the Auto PBR option from the Options menu.

The log file contains all information that the Encoder used to create the resulting encode plus any other notable information that arose during the encode process. When an Encode is created, a log file is placed in the same Save To directory as the encoded file. The log file will share the same name as the encoded file with the extension (_log.txt).

- ☑ **Warning: Log files are not protected and can be edited. Thus, information contained within the LOG FILES SHOULD BE BACKED UP!**

7.6.1 Encode Settings

The Encoder Settings portion of the log file contains all of the information the Encoder used to create the resulting Encode. This includes the full path of the input material and output destination. The destination type, stream type, sample rate, bit width, constant bit rate, dialog normalization, timecode parameters, button options, downmix coefficients, and program information. This is a quick way to determine what settings were used to create the encoded file. The Encode Settings is located at the beginning of the log file and will makeup most all of the information provided in most log files.

7.6.2 Encode Results

The Encode Results portion begins at the bottom of the log file directly below the downmix information section. It contains information about events that occurred in the encoder once the encoded file began processing. There are three type of information that can display here; MD5 Verification, PBR information, and Saturations Warnings. (See the three sections below for details)

7.6.3 MD5 Validation

MD5 data verification code: This letter/number code allows a user to late validate that the encoded file is identical to the original encoded file. Data can be lost during file transfers, thus an MD5 check should be run on any encoded file that has been transferred and the new MD5 letter/number code should be cross referenced with the original letter/number code to ensure they match. If the codes do not match the transferred encode should be considered corrupt and the transfer should be done again. (some chip sets in external hard-drives will cause corrupt data transfer) The MD5 data verification check should be performed if an encoded file has been transferred from the original encode location.

7.6.4 PBR Data in Log Files

When the ✓ Auto PBR Analysis option is checked and a Master Audio stream is encoded, a Peak Bit Rate analysis is performed on the encoded file and a '.dtspbr' file is created in the destination directory.

The peak and average bit rates of the encoded file are placed in the log file. (See Section 10.8 Peak Bit Rate for more information)

7.6.5 Saturation Warnings

If the encoder detects saturation (digital clipping of 0dBFS) in the downmixed streams a warning message will be displayed in the Encode Queue. (it will not display the full message) The related Log file will contain all saturation warning messages that occurred during the encode process along with the time that each saturation occurred. Each saturation message informs the user that the encode will playback without saturation as a full decode (Example: A Master Audio file will playback the full decode Losslessly), but the lossy downmix of that same encode will have saturation at the given time on playback. (Example: the same Master Audio file played on a 5.1 Legacy system will playback with saturation at the given time)

☑ **DTS Strongly recommends reducing the 5.1 downmix coefficients level until the stream can encode in full without a Saturation Warning.**

7.6.6 Dtshd File Delivery via DVD-R

With the increased capacity of Blu-ray Discs discs, it is possible for .dtshd files to exceed 2 GB. Given this file size, there is a cross platform file limitation when moving files from Macintosh to Windows, and visa versa. Errors from transfers (from Mac to PC) can lead to truncated files.

When delivering .dtshd files via DVD-R, you must burn them using the UDF format. On Mac OSX, it is recommended you use Toast or a similar application for this burning process. On the Windows PC platform the latest version of Nero is recommended with UDF enabled.

If you encounter truncated files, the transfer should either be redone via DVD-R burnt in the UDF format or via a different method.

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 7 pertain to the DTS Surround Audio Suite Encoder with the exception that the only available destination formats are DVD and DTS Music Disc. Features that are not available in the DTS Surround Audio Suite include Downmix, and Bitstream Settings. Timecode can be used to determine encode start and stop times within the source material but the timecode will not be saved in the resulting encodes like '.dtshd' files. These limitations exist because the legacy encoder does not allow a custom stereo downmix. The DTS-HD StreamTools cannot operate on a DVD or DTS Music Disc encoded stream. ('.cpt' or '.wav')

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 via email at proaudioinfo@dts.com for details on how to upgrade to DTS-HD Master Audio Suite.

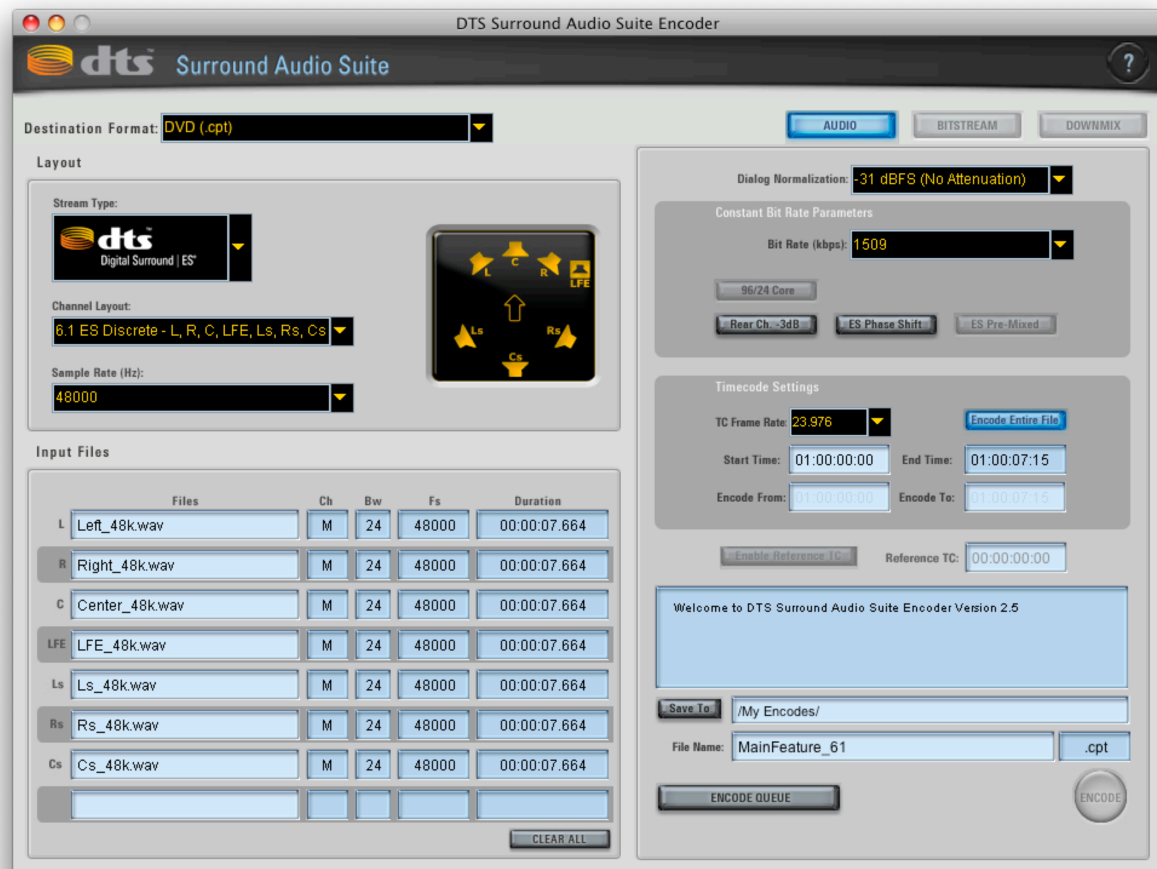


Figure 8-1 Surround Audio Panel

9. DTS-HD StreamPlayer

The DTS-HD Master Audio Suite utilizes a stand-alone software player to QC encoded streams. Please consult the StreamPlayer user manual for operation details. The StreamPlayer user manual can be viewed by clicking the small  button in the top right corner of the StreamPlayer user interface.

Figure 9-1 DTS-HD StreamPlayer

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.

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/Replace 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/Replace Tool
- Append Tool
- Trim Tool
- Split Tool
- Re-stripe Tool
- Add Silence Tool
- File Info Tool
- Verification Tool
- PBR Analysis

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

c) Quality of audio transition

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

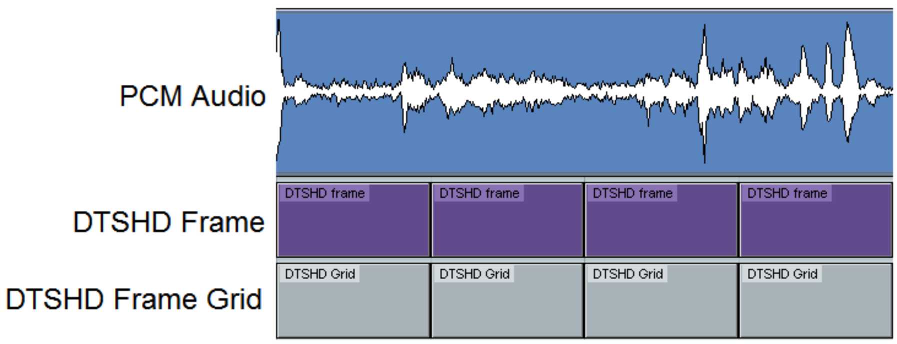


Figure 10-2 DTS Tools Legend

(not to scale)

☑ **DTS Express (LBR):** The use of DTS Express (LBR) encoded streams is only permitted with the Verify tool.

10.1 Join/Replace Tool

The Join/Replace tool allows users to join two DTS-HD encoded streams, or replace a portion of one encoded stream with another, where audio to timecode synchronization and bit exact editing are *mandatory*[1]. The Join/Replace Tool is activated by selecting the 

button, as shown in Figure 10-3 Join/Replace Tool and Main Tools Screen. The Join/Replace tool requires two DTS-HD encoded streams as its input and a destination file to save the results of the 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. Use the "Save To" button and the "File Name" fields to specify the output directory and filename of the resultant join/replace operation. Pressing the "Process" button will initiate the join/replace processing. Pressing the "Cancel" button will stop the running process.

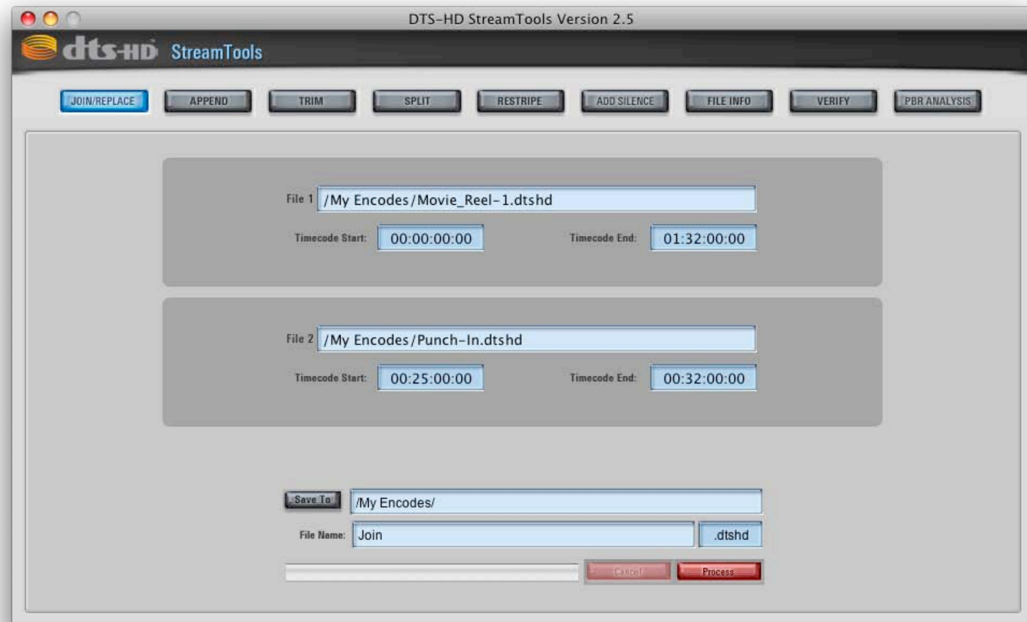


Figure 10-3 Join/Replace Tool and Main Tools Screen

The Join/Replace operation will join two DTS-HD encoded streams, or replace a portion of an encoded stream (also known as a Punch In), given the Start and End timecode values of the provided DTSHD encoded files. File 1 will be joined to File 2 in the order dictated by the timecode of the input files.

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

---

[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/Replace operation if the selected DTS-HD file has been restriped (see section 0).*

❒ Join/Replace operations must be performed with overlapping audio regions. The audio contained in the overlapping regions 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/s in order for a join operation to be permitted.

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

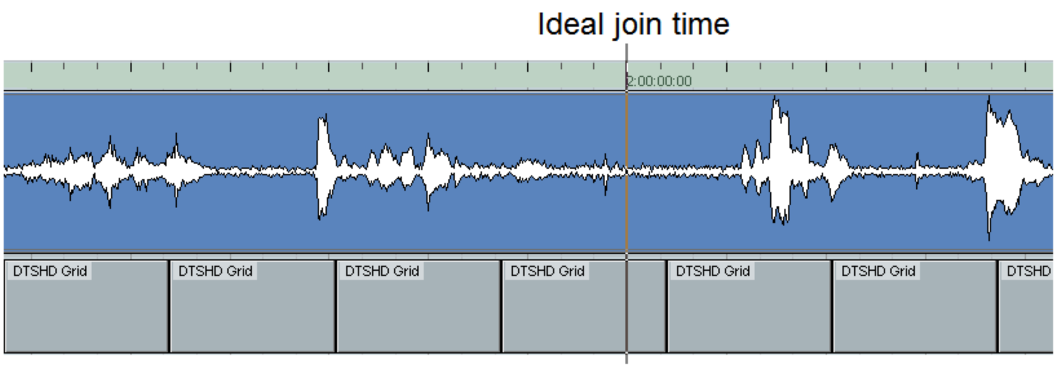


Figure 10-4 Audio Alignment to DTS Frame

To create the required overlaps, 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 duplicate audio material of at least 1 second or more (seen below).

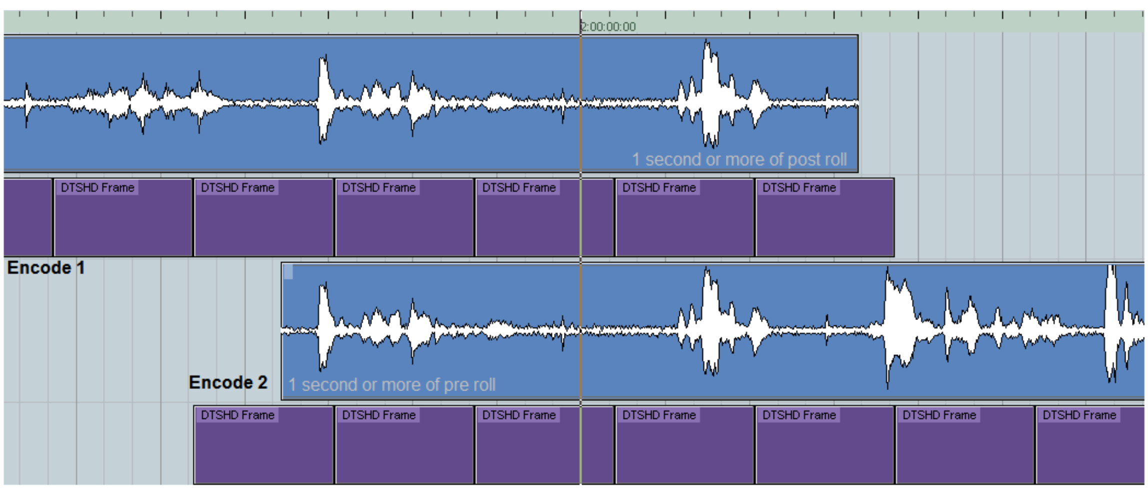


Figure 10-5 Join Overlap Diagram *(not to scale)*

Once the resulting audio segments are encoded with the same reference timecode *(see section 0 for reference timecode explanation)* the files will be joined at the user defined join time (02:00:00 as seen in Figure 10-4).

In the case of a Replace operation, the overlapping identical audio regions would be found one second from the beginning and end of the shortest encode (Encode 2 in Figure 10-5.1 below). The regions in Encode 2 (green and yellow selections) must contain bit identical audio to the same regions in Encode 1. The audio between the overlaps in Encode 2 is the replacement audio. The shorter encode will always replace the longer encoder.

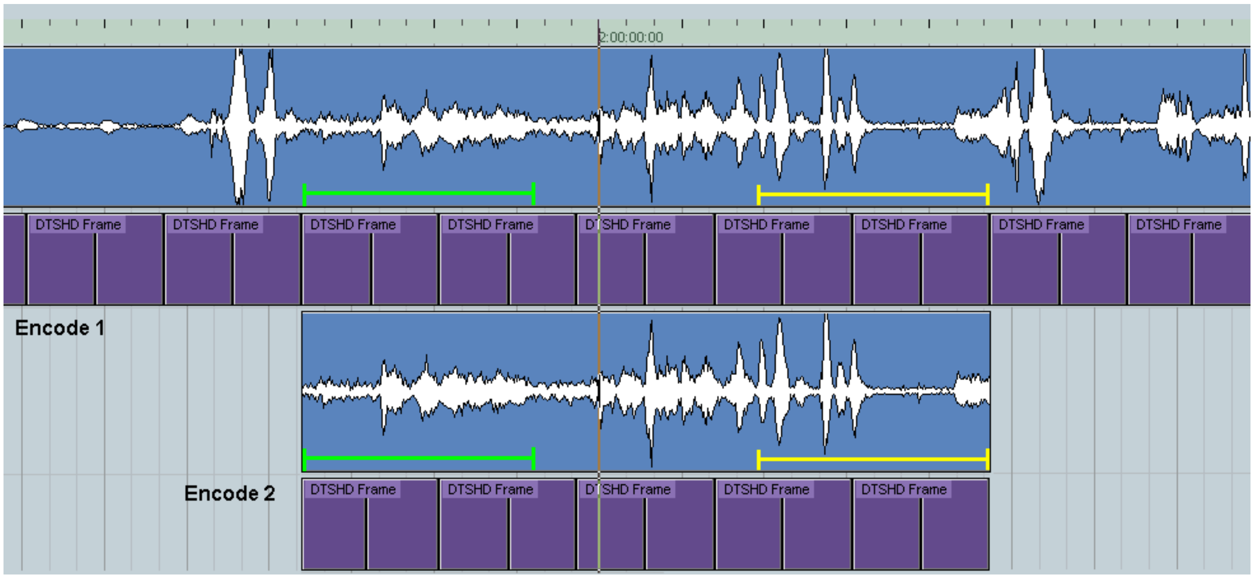


Figure 10-5.1 Replace Overlap Diagram *(not to scale)*

❒ The DTS-HD encoded streams used with the Join/Replace operation must contain a matching Reference Times (which is specified during the encoding process). 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.

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

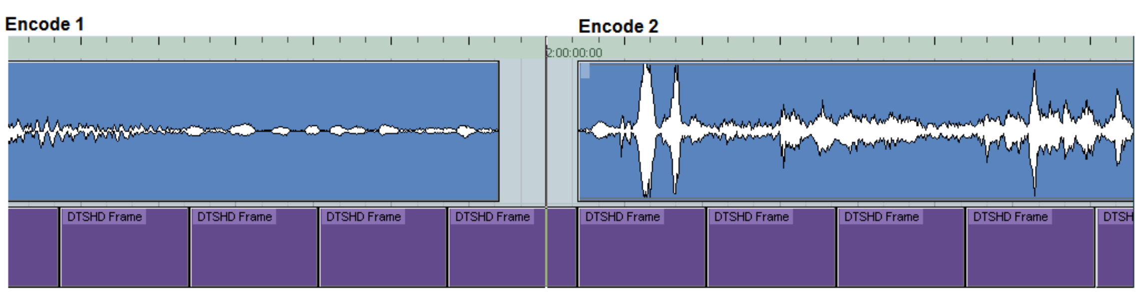


Figure 10-6 Example with Differing Reference Times

From the example in Figure 10-6, Encode 2 could not be placed at 02:00:00:00, it's intended start time due to asynchronous DTS frame alignment between the two encodings. DTS-HD StreamTools editing occurs between DTS-HD frame boundaries. Encode 2 could not be placed exactly at 2:00:00:00 and would then be out of sync, relative to it's intended start time, after being joined to Encode 1.

The scenario in Figure 10-6 is ***not*** permitted using the Join/Replace operation. Encode 2 would need to be re-encoded with a reference time identical to Encode 1 (01:00:00:00) and with the required overlapping audio regions (as described above) in order to be joined with Encode 1 using the Join/Replace tool.

If audio to timecode synchronization is not required, the Append operation can be used to join these DTS-HD encoded files as depicted in Figure 10-6, allowing encode 2 to shift in time. *(see section 0 , Append Tool)*

☑ **DTS Express (LBR):** The use of DTS Express (LBR) encoded streams is only permitted with the Verify 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:** Figure 10-7 does *not* show the overlapping audio regions required for the Join/Replace operation, thus, this operation would not be permitted by the join tool.

Figure 10-7 Example with the same Reference Times

❐ 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/replace or append operations to detect "-3dB Rear Channel Attenuation", "ES Phase Shift" and "Downmix Coefficient" discrepancies between encoded streams, 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/Replace operation if the selected DTS-HD file has been restriped (see 10.5 Restripe Tool).

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

Summary of JOIN requirements

❐ Overlapping audio regions must be present at join points. 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 /replace operation to be permitted.

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

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

❐ Audio to timecode sync cannot be maintained by the Join/Replace operation if the selected DTS-HD file has been restriped.

☑ **DTS Express (LBR):** The use of DTS Express (LBR) encoded streams is only permitted with the Verify tool.

10.2 Append Tool

The Append Tool is activated by selecting the 

button. As shown in Figure 10-8 Append Tool, the operation requires two DTS-HD encoded streams as its input and a destination file to save the results of the append operation. Double-click in each of the filename fields to load the files that are to be joined. A little as 2 and as many as 8 files can be appended together in one single process. The Timecode Start and Timecode End text fields show the start and end time of the input materials for each of selected files.

The Start Time and End Time text fields allow the user to specify a specific portion of each input file to be used. The timecode values in the Append function ONLY relate to the output length of the file they are tied to. The End Time of the file above can be greater than or less than the Start Time of the file below as they are not related to each other. The Start Time of the file below will always begin immediately following the End Time of the file above and the resultant concatenated, combined, file will use the contents of the below file while continuing the timecode of the first file. 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.

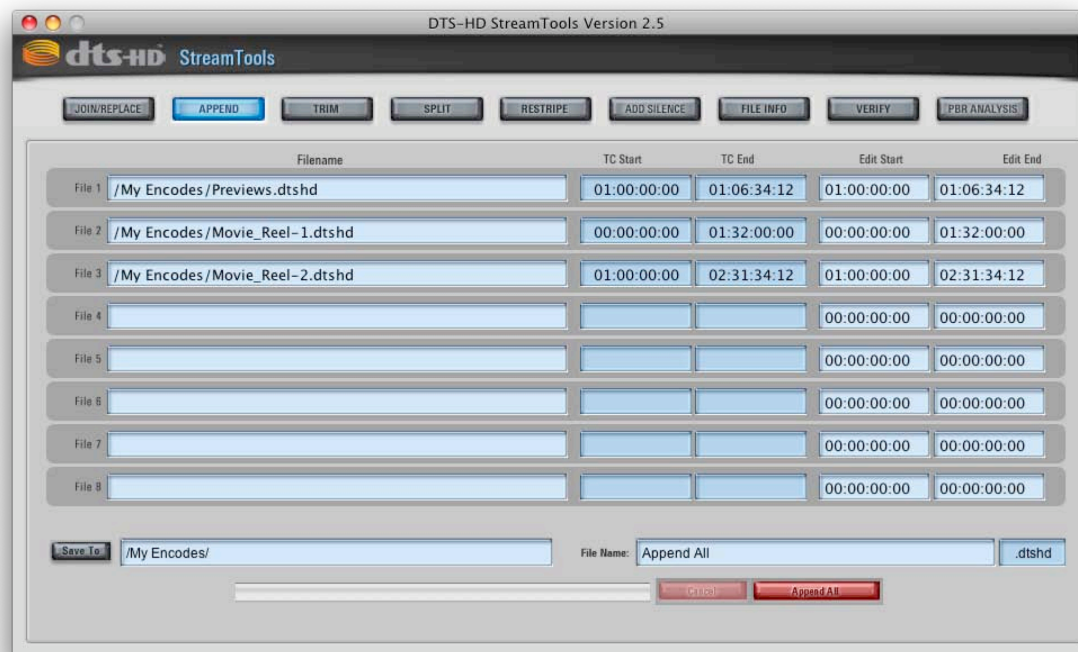


Figure 10-8 Append Tool

The Append tool allows the user to concatenate, combine, sections of a DTS-HD encoded streams with no regard for audio to timecode synchronization as is illustrated in Figure 10-9. In this basic 2 file append example, File 1 represents the initial file in the resultant appended stream. File 2 would be appended to the end of File 1 after the append operation.

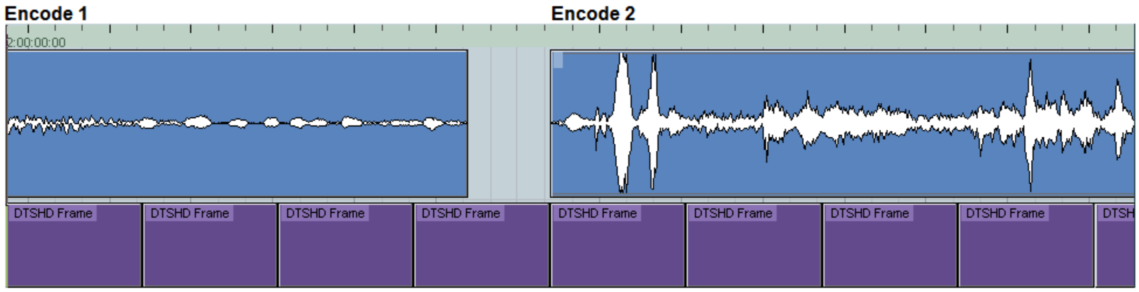


Figure 10-9 Append Tool Diagram

Audio to timecode synchronization for all encodes following the initial encode will be lost. As demonstrated in Figure 10-9, appending one Encode to another will cause Encode 2 to drift later in time, no greater than 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 near zero crossing in order to avoid audio anomalies at the edit point. If bit exact editing is required, use the Join tool to perform the operation.

☑ **Note:** For Appends using more than 2 input files, a temporary file will be held in the user directory of the user's computer. Should a space issue arise, please ensure that both the destination directory and the user directory (OS Drive) have enough free space to create the output file.

10.3 Trim Tool

The Trim Tool is activated by selecting the 

button, as shown in Figure 10-10 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 input 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.

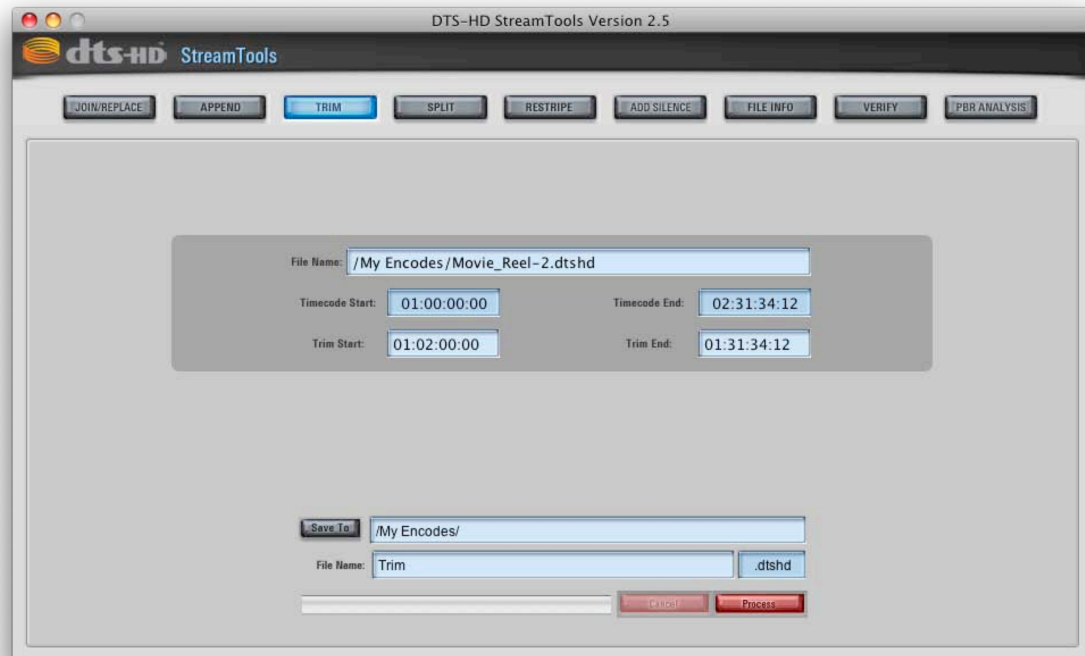


Figure 10-10 Trim Tool

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 is shown in Figure 10-11.

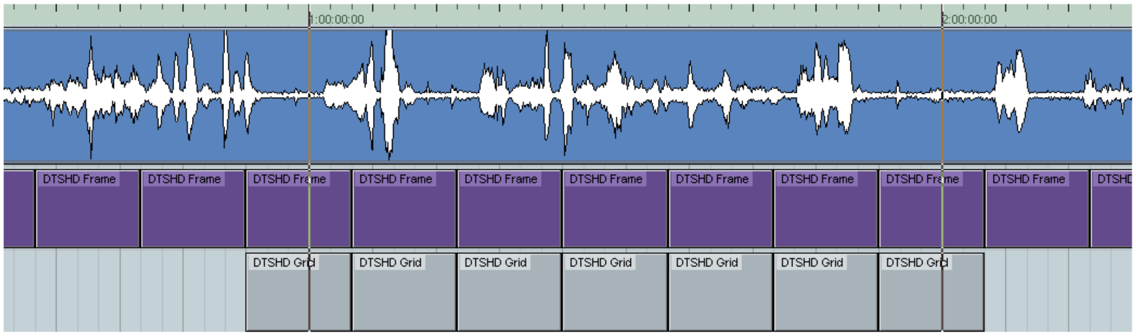
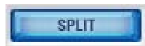


Figure 10-11 Trim Tool Diagram

The DTS frames located at the specified Trim timecodes are retained in their entirety. However, the user specified start and end times specify where audio playback will begin and where it will end. Upon decoding, only the audio between the specified Trim timecode values will be decoded. If this file were joined or appended to the end of another encoded stream, the entire first frame would be heard when decoded.

☑ **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 10-12, 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.

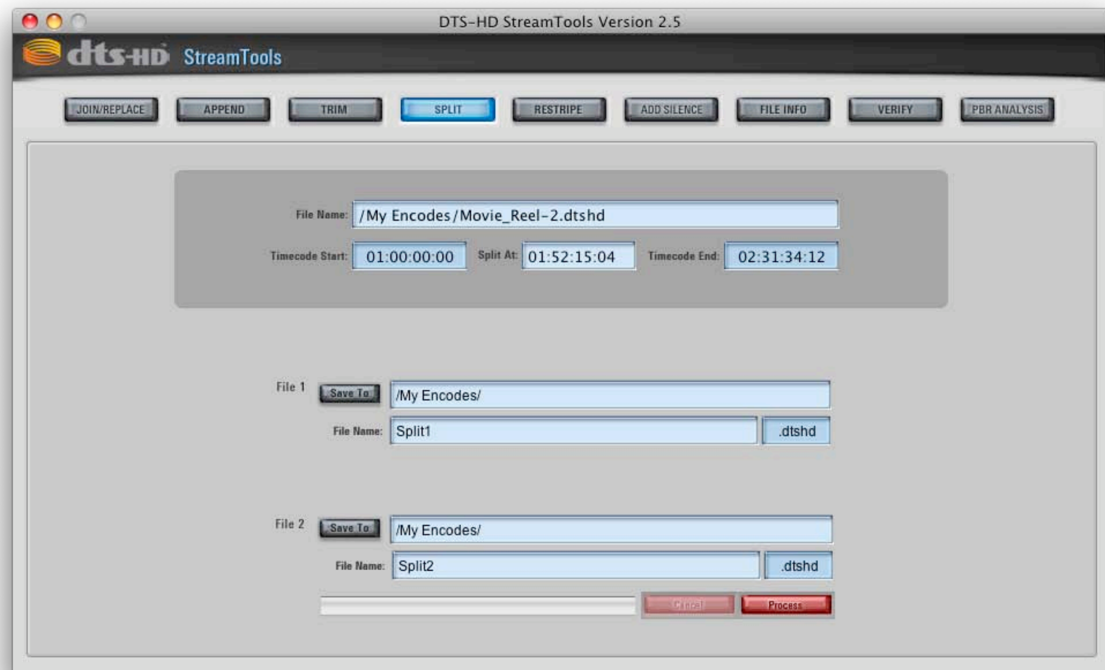


Figure 10-12 Split Tool

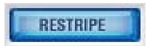
The Split tool operates by dividing an encoded stream at the DTS frame nearest to the user specified split time, resulting in the creation of two additional encodes as illustrated in Figure 10-13.

Figure 10-13 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:** 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 10-14 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. The original Start Timecode and Frame Rate will display in the read only fields. Specify the New Start Timecode and the New Frame Rate for the restripe operation. Pressing the "Process" button will initiate the restripe processing.

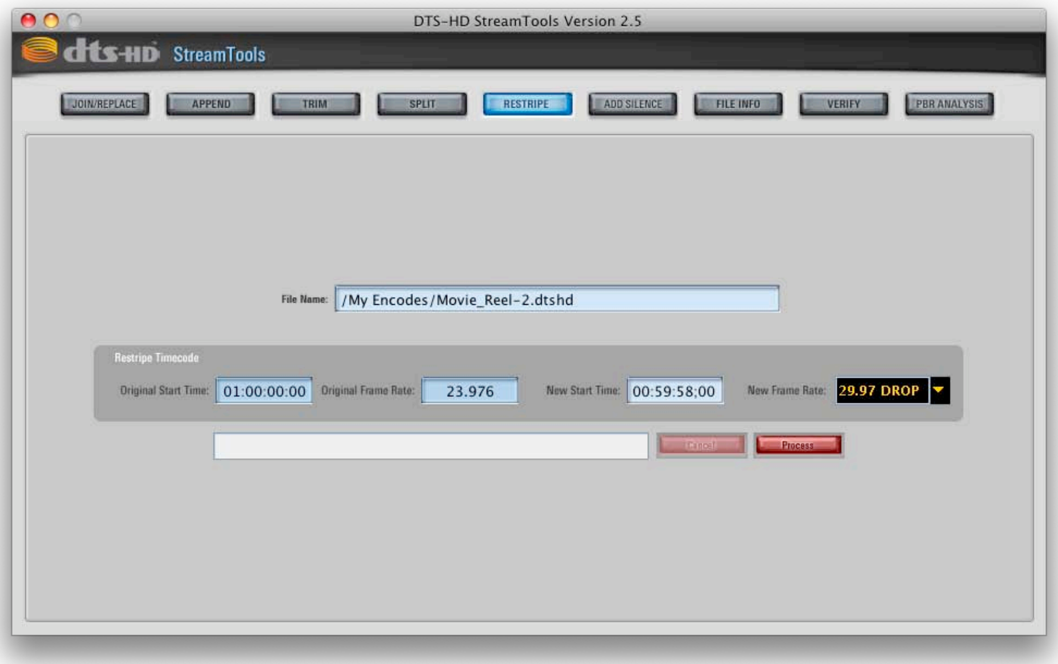


Figure 10-14 Restripe Tool

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 than 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 Add Silence Tool

The Add Silence Tool is activated by selecting the ADD SILENCE button as shown in Figure 10-16

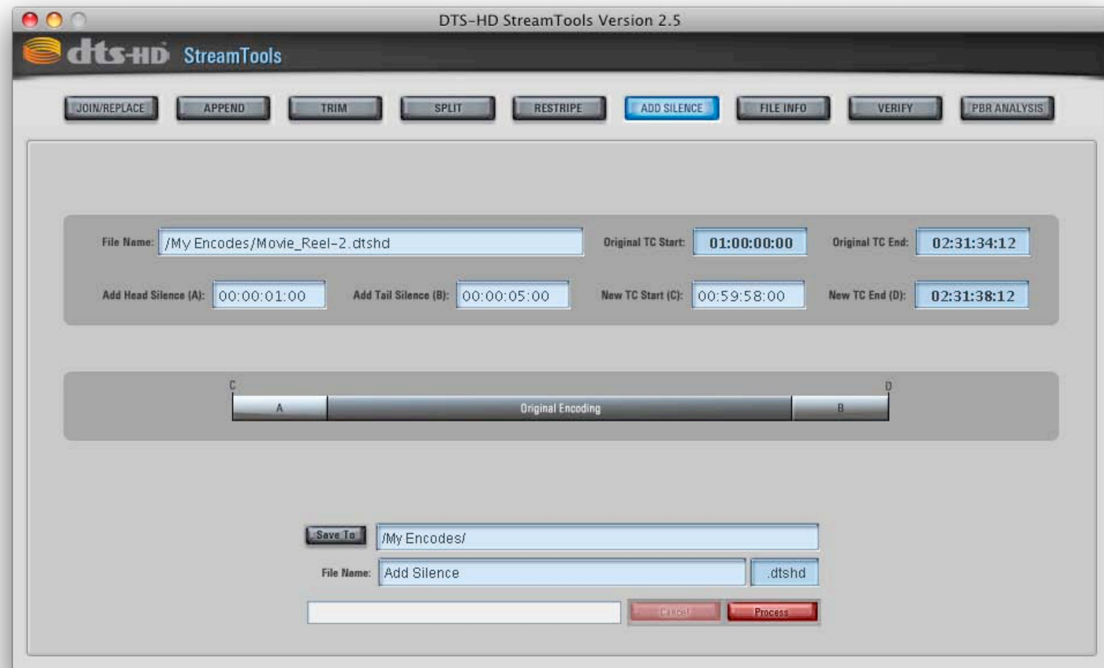


Figure 10-15 – Add Silence Tool

This Tool appends silence (-INF) audio to the Head, Tail, or Head and Tail of a Blu-Ray Primary Audio encoded file. When the encode is loaded, the Original TC Start: and Original TC End: will display the current timecode Start and End times for the file. The timecode value entered into the Add Head Silence (A): timecode field will determine the length of silent audio that will be added onto the Head/Front of the original encode. The timecode value entered into the Add Tail Silence (B): timecode field will determine the length of silent audio that will be added onto the Tail/End of the original encode. The New TC Start (C): timecode field allows a user to perform a Restripe command at the same time the silence is added. Every instance in which a user malleable timecode field is committed (ie. Enter is hit for A, B, or C timecode fields) the New TC End (D): will adjust accordingly.

> **CAUTION**: Consider audio to timecode synchronization lost if either Add Head Silence (A): is anything other than 00:00:00:00 or if the New TC Start (C): is not equal to the Original TC Start: Timecode synchronization Drift will be no greater than one DTS-HD frame in the worst case.

☑ **Note:** Changing either Add Head Silence (A): to anything other than 00:00:00:00 or changing the New TC Start (C): to something other than the Original TC Start: will result in an encoded file with an adjusted Reference Time to a valid timecode per the new frame count.

☑ **Note:** The Add Silence tool will ONLY work with Primary Audio files created by MAS Encoder v2.0 and after.

10.6 File Info Tool

The File Info Tool is activated by selecting the FILE INFO button, as shown in Figure 10-16.

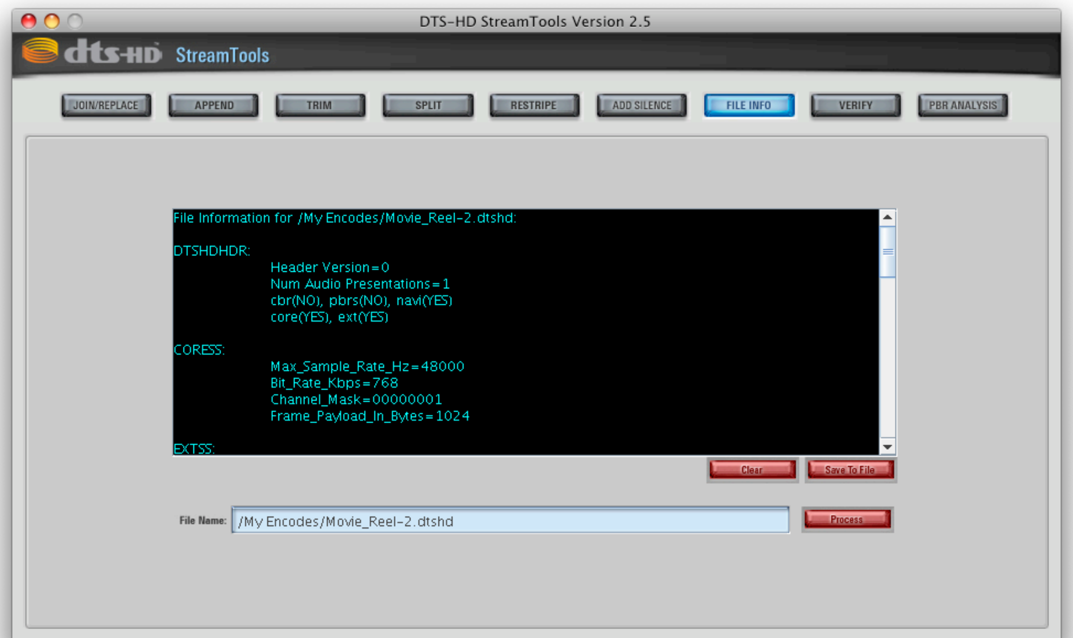


Figure 10-16 File Info Tool

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. 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 Encodes\Movie.dtshd:

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

CORESS:
        Max_Sample_Rate_Hz=48000
        Bit_Rate_Kbps=1509
        Channel_Mask=0000000F
        Frame_Payload_In_Bytes=2012

EXTSS:
        Ext_Ss_Avg_Bit_Rate_Kbps=274
        Ext_Ss_Peak_Bit_Rate_Kbps=273
        Pbr_Smooth_Buff_Size_Kb=128
        bcc(CORE), ll(YES), lbr(NO)

Source Samples    : 92160000
Sample Rate       : 48000Hz
Samples Per Frame : 512
Codec Delay       : 1024

LOSSY:
        FSize=2012
        Amode=9
        LFF=2
        SFreq=13
        PCMR=16 bit
        DialNorm=-31 dBFS LeqA

LOSSLESS:
        Version=1
        NumChSets=1

Frames Content    : 0 + 180000 frames + 0
Frames Min Reqd   : 180002
Frames Hdr        : 180002
Frames Navi       : 180002

Timecode RATE     : 30
Timecode START    : 01:00:00:00
Timecode END      : 01:32:00:00
Timecode REF      : 00:00:00:00

----------
```

10.7 Verify Tool

The Verification Tool is activated by selecting the  button, as shown in Figure 10-17.

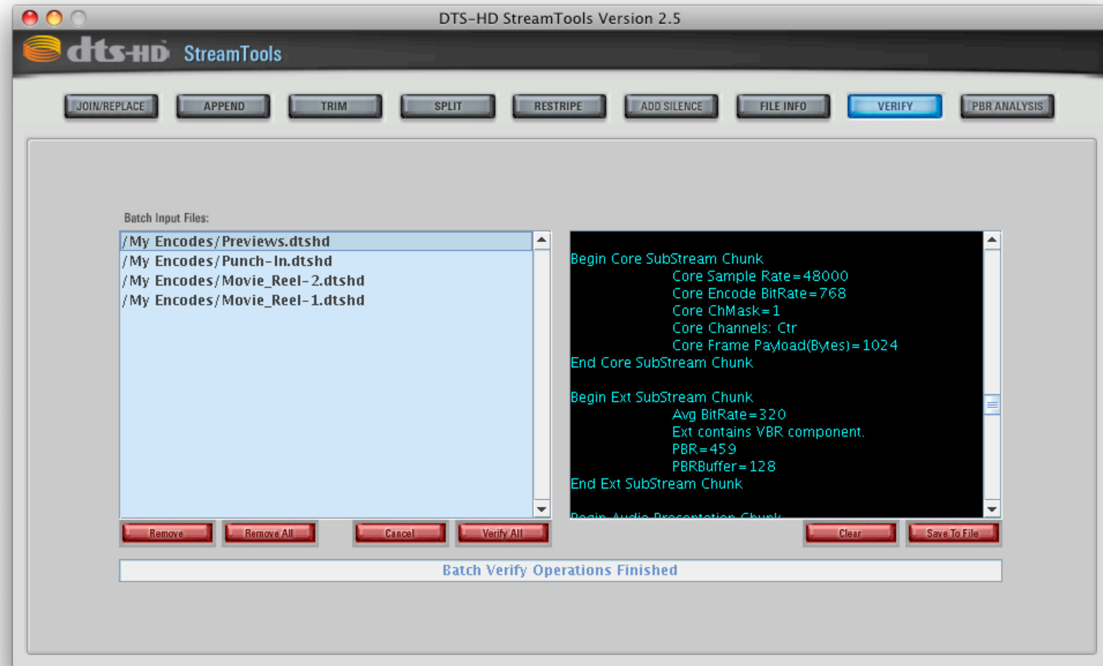


Figure 10-17 Verification Tool

The left blue field is the file input field. Double clicking this blue file field will open a browser allowing the user to select a '.dtshd' '.cpt' or '.wav' file for validation. A user can also Drag and Drop valid DTS file(s) into the file field. A single selected file can be removed by clicking the 'Remove' button. A user can clear all loaded files by clicking the 'Remov2 e All' button. Once all the files are loaded click the 'Verify All' button to start the verification process. Files will be verified from top to bottom of the input files list. To cancel the process at any time, click the 'Cancel' button.

The Verify process 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 displayed in the black window on the right side and the readout 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 'Save To File' button will open a Save Browser that will save the results of the verify job (the contents of the black window) to a '.txt' file specified in the browser. The readout 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. To clear the contents of the black output window simply click the 'Clear' button located directly beneath the window.

The status bar above the file name selection area provides feedback on the number of frames that have been processed.

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 Encodes\Movie.dtshd

****** DTSHD Verification Tool Version 325.26 May  5 2007 04:42:58 ******
DTSHD Header Chunk
        DTSHD Header Verison 0
        TimeCode reference clock=1.0/48000.0
        TimeCode Frame Rate=30
        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=15
        Core Channels: Ctr L R LS RS LFE
        Core Frame Payload(Bytes)=2012
End Core SubStream Chunk

Begin Ext SubStream Chunk
        Avg BitRate=274
        Ext contains VBR component.
        PBR=273
        PBRBuffer=128
End Ext SubStream Chunk

Begin Audio Presentation Chunk
        Audio Presentation Index =0
        Max SampleRate=48000
        Total Number of Frames=180002
        Total Number of samples per frame=512
        Total Number of samples in original(@ Max SampleRate)=92160000
        Channel Mask=15
        Channels: Ctr L R LS RS LFE
        Codec delay in samples=1024
        LSB Trim percentage=0
End Audio Presentation Chunk

Begin Stream Data Chunk
Encoded Stream Size=471090176 bytes
End Stream Data Chunk

Begin Navigation Table Chunk
        Num of Entries=180002
```

```
            Frame Interval=1
            Bytes Per Entry=4
            Skipping printing of Navigation entries.
End Navigation Table Chunk

Begin Time code Chunk
            Time code clock =48000
            TimeCode Frame Rate=30
            Start TimeCode=01:00:00:00
            End TimeCode=01:32:00:01
            Reference TimeCode=00:00:00:00
End Time code Chunk

Begin Build Ver Chunk
BUILD DATE = Mar 29 2007
BUILD TIME = 04:42:09
VERSION = 325
REVISION = de

End Build Ver Chunk

Begin File Info Chunk
#
END FILEINFO
#
End File Info Chunk

Find sync word: 7ffe8001
Find sync extension: 3f

****** End of input bit stream ******

****** End of input bit stream ******
180002 out of 180002 frames verified successfully
Stream passes verification with no errors.

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

10.8 Peak Bit Rate Analysis

The Peak Bit Rate (PBR) Analysis Tool is activated by selecting the 

button, as shown in Figure 10-17.

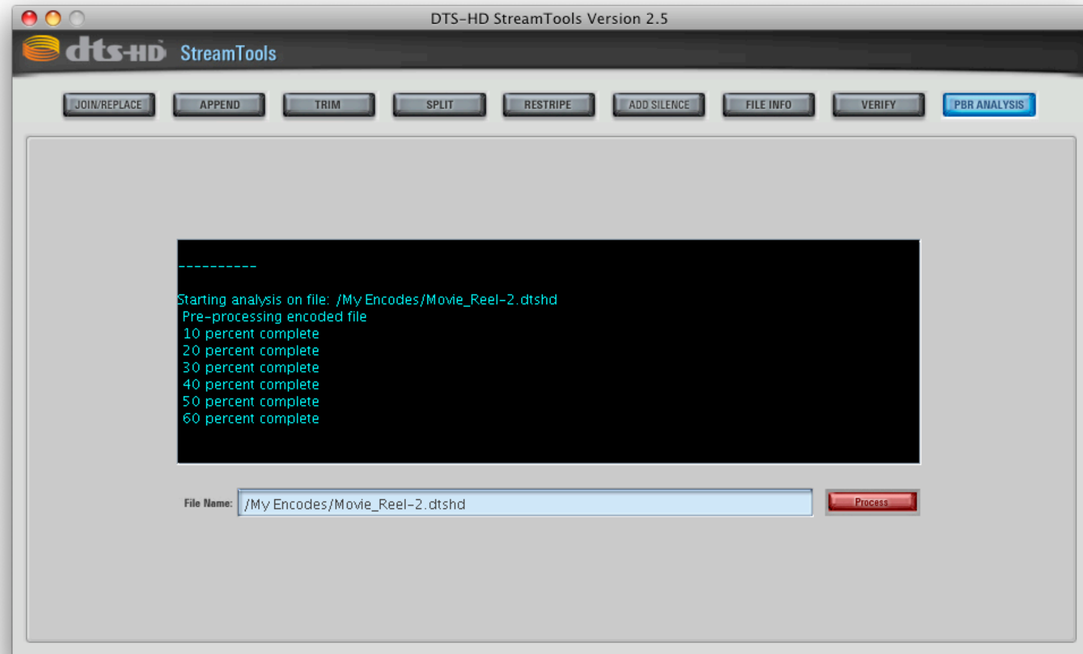


Figure 10-18 Peak Bit Rate

The Peak Bit Rate Analysis tool analyses variable bit rate encodings (DTS-HD Master Audio encoded streams) graphically plotting the selected encoding’s bit rate over time, as if the encoding had been “smoothed” for authoring using a Peak Bit Rate scheduling utility. The smoothing process redistributes data throughout the encoded stream for a more constant flow of data during disc played back. Smoothing is performed during the authoring process of a disc. The PBR Analysis tool is an informational tool only; the selected encoding will not be modified. However, a .dtspbr file will be created, at the location of the selected encoding, containing the analysis data.

Upon "processing" an encode, the status window is populated with the performance related data, seen below, as the .dtspbr file is created on disk.

Note, the Peak and Average Bit-rate are stated within the status window text.

```
----------

Starting analysis on file: C:\DTSENC.dtshd
 Pre-processing encoded file
 10 percent complete
 20 percent complete
 30 percent complete
 40 percent complete
 50 percent complete
 60 percent complete
 70 percent complete
 80 percent complete
 90 percent complete
 Pre-processing complete
 Avg  Bitrate CBR core + VBR Lossless ext (kbps):  4099
 Peak Bitrate CBR core + VBR Lossless ext (kbps):  4320
 PBR max from second 0 to second 1
 PBR max from second 20 to second 21
 PBR max from second 59 to second 60
 PBR max from second 80 to second 81
Ending analysis on file: C:\DTSENC.dtshd
Creating graph on file: C:\DTSENC.dtspbr
```

PBR Analysis Status Window Output

Upon analysis completion, the PBR Analysis graph window will be presented as shown in Figure 10-18.

You may open a .dtspbr file that has already been created, simply click the Tools File menu (top left), click "Open PBR Analysis Graph", then select the desired .dtspbr file.

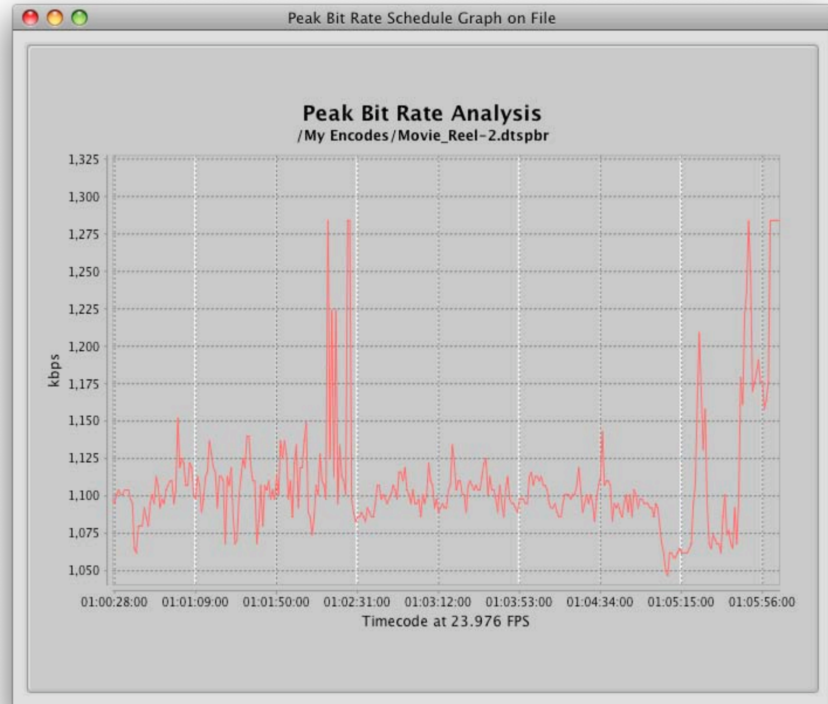


Figure 10-19 Peak Bit Rate Graph

Right click on the graph window for more options such as properties, save as, print, zoom, and auto range.

Along with the ability to zoom via the right click menu, the graph window also allows drag and drop zoom functionality. Click and drag down and to the right over the desired selection to zoom in. Repeat for a closer view. Click and drag from right to left for a full zoom out.

11. Encoder Error Codes

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

General Error xxxx6 – Parameter Invalid
General Error xxxx7 – Parameter object Invalid
General Error xxx11 – Data type invalid
General Error xxx12 – Data type unsupported
General Error xxx13 – Bus port unsupported
General Error xxx14 – Bus invalid
General Error xxx15 – Bus unassigned
General Error xxx16 – Bus channel count insufficient
General Error xxx17 – Metadata Invalid

Error 1 – Job Queue: Job canceled
Error 2 – Job Queue: Insufficient input channels provided
Error 3 – Job Queue: Audio Bus allocation
Error 4 – Job Queue: Queue command invalid
Error 5 – Job Queue: Queue allocation
Error 6 – Job Queue: Encode transaction allocation
Error 7 – Job Queue: Module ID invalid
Error 8 – Job Queue: Module allocation
Error 9 – Job Queue: Encode duration is too short to implement the specified static primary audio attenuation
Error 10 – Job Queue: Unable to delete specified log file
Error 11 – Job Queue: Unable to open PBR output file
Error 12 – Job Queue: PBR initialization unable to allocate timecode object
Error 13 – Job Queue: PBR unknown library error
Error 14 – Job Queue: PBR unable to delete PBR output file
Error 15 – Job Queue: Input file bit depth does not match config file
Error 16 - Job Queue: Input file sample rate does not match config file

Error 2023 – Audio Bus: Channel count Invalid
Error 2024 – Audio Bus: Frame size invalid
Error 2025 – Audio Bus: Buffer type invalid
Error 2026 – Audio Bus: Read Buffer unavailable
Error 2027 – Audio Bus: Write Buffer unavailable

Error 3018 – Audio File Input: Sample start/end times invalid
Error 3019 – Audio File Input: No Files Specified
Error 3023 – Audio File Input: File type unsupported
Error 3025 – Audio File Input: Files have different bit depths
Error 3026 – Audio File Input: Files have different sample rates
Error 3030 – Audio File Input: Unable to open file for reading
Error 3031 – Audio File Input: Unable to open file for writing
Error 3032 – Audio File Input: File bit depth unsupported
Error 3033 – Audio File Input: File sample rate unsupported
Error 3035 – Audio File Input: File get position
Error 3036 – Audio File Input: File set position
Error 3037 – Audio File Input: Cannot read header ID
Error 3038 – Audio File Input: Cannot read header Format
Error 3039 – Audio File Input: Cannot read header data
Error 3040 – Audio File Input: File Seek
Error 3041 – Audio File Input: File Seek out of range

Error 3042 – Audio File Input: Unable to read file
Error 3043 – Audio File Input: Unable to write file

Error 8018 – Encoder: Initialization – invalid configuration file
Error 8019 – Encoder: Un-initialize
Error 8020 – Encoder: Get mode
Error 8021 – Encoder: Encode frame
Error 8022 – Encoder: Get Header
Error 8023 – Encoder: Get appendix
Error 8024 – Encoder: Get MD5
Error 8025 – Encoder: Unable to create encoder object
Error 8026 – Encoder: Unable to open configuration file
Error 8027 – Encoder: Configuration input stream allocation
Error 8028 – Encoder: Unable to parse configuration file
Error 8029 – Encoder: Configuration stream invalid
Error 8030 – Encoder: Unable to obtain input buffer
Error 8031 – Encoder: Unable to obtain encoded frame
Error 8032 – Encoder: Audio data allocation
Error 8033 – Encoder: Unable to allocate MD5 buffer
Error 8034 – Encoder: Unable to convert filename to UTF-16
Error 8035 – Encoder: Encoded file failed MD5 validation"

Error 12018 - Encoder File Output: Mode invalid
Error 12019 - Encoder File Output: Unable to open file for writing
Error 12020 - Encoder File Output: File seek
Error 12021 - Encoder File Output: Unable to write to file - disk may be full
Error 12022 - Encoder File Output: Unable to delete encode output file

Error 13018 - AAF File: Parser allocation
Error 13019 - AAF File: Unable to parse specified file
Error 13020 - AAF File: Encode frame rate does not match AAF frame rate of 23.976
Error 13021 - AAF File: Encode frame rate does not match AAF frame rate of 24
Error 13022 - AAF File: Encode frame rate does not match AAF frame rate of 25
Error 13023 - AAF File: Encode frame rate does not match AAF frame rate of 29.97 Drop
Error 13024 - AAF File: Encode frame rate does not match AAF frame rate of 29.97
Error 13025 - AAF File: Encode frame rate does not match AAF frame rate of 30 Drop
Error 13026 - AAF File: Encode frame rate does not match AAF frame rate of 30
Error 13027 - AAF File: Encode frame rate does not match AAF frame rate of 50
Error 13028 - AAF File: Encode frame rate does not match AAF frame rate of 59.94 Drop
Error 13029 - AAF File: Encode frame rate does not match AAF frame rate of 59.94
Error 13030 - AAF File: Encode frame rate does not match AAF frame rate of 60 Drop
Error 13031 - AAF File: Encode frame rate does not match AAF frame rate of 60
Error 13032 - AAF File: Encode frame rate does not match AAF frame rate (unknown)
Error 13033 - AAF File: More than one primary scaling mode was defined
Error 13034 - AAF File: Insufficient number of primary scaling indices were specified
Error 13035 - AAF File: Invalid primary scaling slot index specified
Error 13036 - AAF File: Insufficient number of panning indices were specified
Error 13037 - AAF File: Invalid panning slot index specified
Error 13038 - AAF File: Unable to allocate UTF-16 file buffer
Error 13039 - AAF File: Unable to convert AAF file to UTF-16

Error 14018 - Branch Point: Unable to open file for reading
Error 14019 - Branch Point: Invalid SMPTE timecode or sample rate

Error 14020 - Branch Point: EOF
Error 14021 - Branch Point: Error reading CSV file
Error 14022 - Branch Point: No valid entry within encode time frame found
Error 14023 - Branch Point: SMPTE timecode object allocation
Error 14024 - Branch Point: Timecode string allocation
Error 14025 - Branch Point: Branch point invalid
Error 14026 - Branch Point: Branch point out of time sequence

Error 15018 - Downmix Saturation File: Unable to open file for writing
Error 15019 - Downmix Saturation File: Unable to write to file - disk may be full
Error 15020 - Downmix Saturation File: Unable to delete output file

12. DTS Tools Error Codes

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

12.1 File Error Codes

"Format not supported"
The format of the .dtshd file is not supported by the given operation. For example, Sub Audio (CA) encoded streams are not permitted for use with the Join operation. DTS Express encoded stream editing is not permitted for sue with tools operations.

"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 could not be opened"
File does not exist or cannot be created.

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

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

"File extension is not present"
File extension required.

"File extension is not .dtshd"
Only .dtshd files can be edited with tools.

"File is zero size"
File has no data or is corrupt.

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

"File could not be opened"
File may be in use by another program.

"Bad error number"
Unknown error.

"Operation canceled"
The attempted operation was not permitted, was invalid, or canceled by user.

12.2 Timecode Error Codes

”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).

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

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

"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.

"The timecode format is unrecognized"
The manner in which the timecode information was formatted is unrecognized.

"Not a valid time"
Timecode does not conform to SMPTE standards.

"No timecode information present in file"
Previous versions of DTS Express encoded .dtshd files did not contain timecode data. Editing encoded stream lacking timecode is not possible.

"File did not have a 2 frame delay"
File lacks expected codec delay and cannot be edited.

12.3 Join / Append Error Codes

"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 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.

"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.

12.4 Encoder Log File Output Example

Due to the use of tabs and other white spaces contained within the log file, Table 13-1 depicts the editors that should be used for best on-screen viewing on the log file.

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

Table 13-1 Recommended Word Processors for Viewing Data

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.

```
****************************************
MAS Version Number = 2.50.op


AUDIO INPUT SETTINGS
------------------------------------------------------
Media Type              = Blu-ray Disc (.dtshd)
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
Using 96/24 Core        = false

INPUT FILES
------------------------------------------------------
Left                        = /MainFeature_L.wav
Right                       = /MainFeature_R.wav
Center                      = /MainFeature_C.wav
Low Frequency Effects       = /MainFeature_LFE_Test.wav
Left Side Surround          = /MainFeature_Lss_Test.wav
Right Side Surround         = /MainFeature_Rss_Test.wav
Left Surround Rear          = /MainFeature_Lsr_Test.wav
Right Surround Rear         = /MainFeature_Rsr_Test.wav

BITSTREAM SETTINGS
------------------------------------------------------
Program Info                                       =
Enable Remapping                                   = false

TIME CODE SETTINGS
------------------------------------------------------
Frame Rate              = 24
Encode Entire File      = true
Start Time              = 01:00:00:00
End   Time              = 01:00:10:00
```

```
Use Reference         = false

OUTPUT LOCATION
------------------------------------------------------
Directory             = /Encodes/
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 Not Enabled

----------------------------------------------------


MD5: 2d7c3b85d8885e9c26bbf10d88f779a2

Avg  Bitrate CBR core + VBR Lossless ext (kbps): 4750
Peak Bitrate CBR core + VBR Lossless ext (kbps): 6675

Encode completed successfully.
```

13. AAF File Import Supplement – Dynamic Secondary Audio

1. Required Software / Overview

Required Software

1. Pro Tools HD 7.0 or higher with DigiTranslator 2.0 or higher.

Overview

With the integration of Secondary Audio for Blu-ray Disc, DTS-HD streams can be encoded with dynamic volume automation for real-time mixing between Secondary and Primary Audio. Mono Secondary Audio can also be encoded to incorporate dynamic panning between channels.

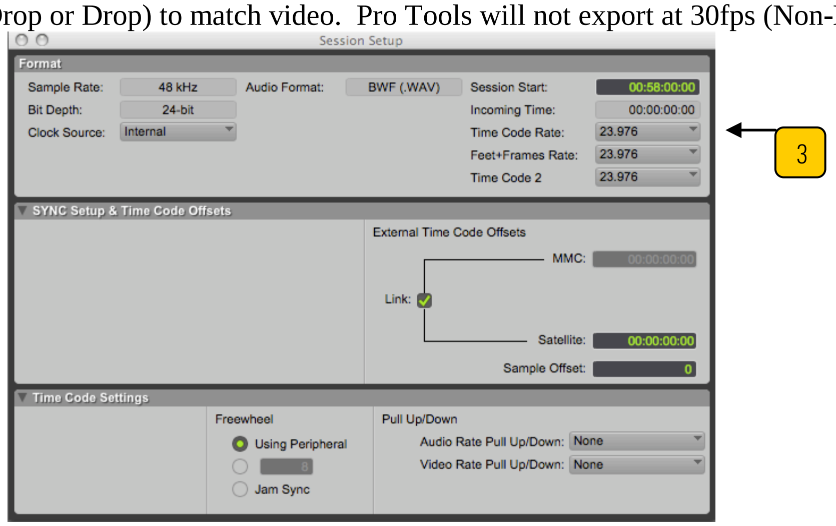
The purpose of this document is to explain how to create such automation within Pro Tools, export the volume changes and panning data as an AAF file, import this file into the DTS-HD Master Audio Suite, and encode your Secondary Audio tracks.

NOTE: This document is created with the understanding that the user is already familiar with the basic functions of Pro Tools. Although overall methodology is very similar, the illustrations that follow reflect Pro Tools workflow. You may need to consult your DAW manual for assistance.

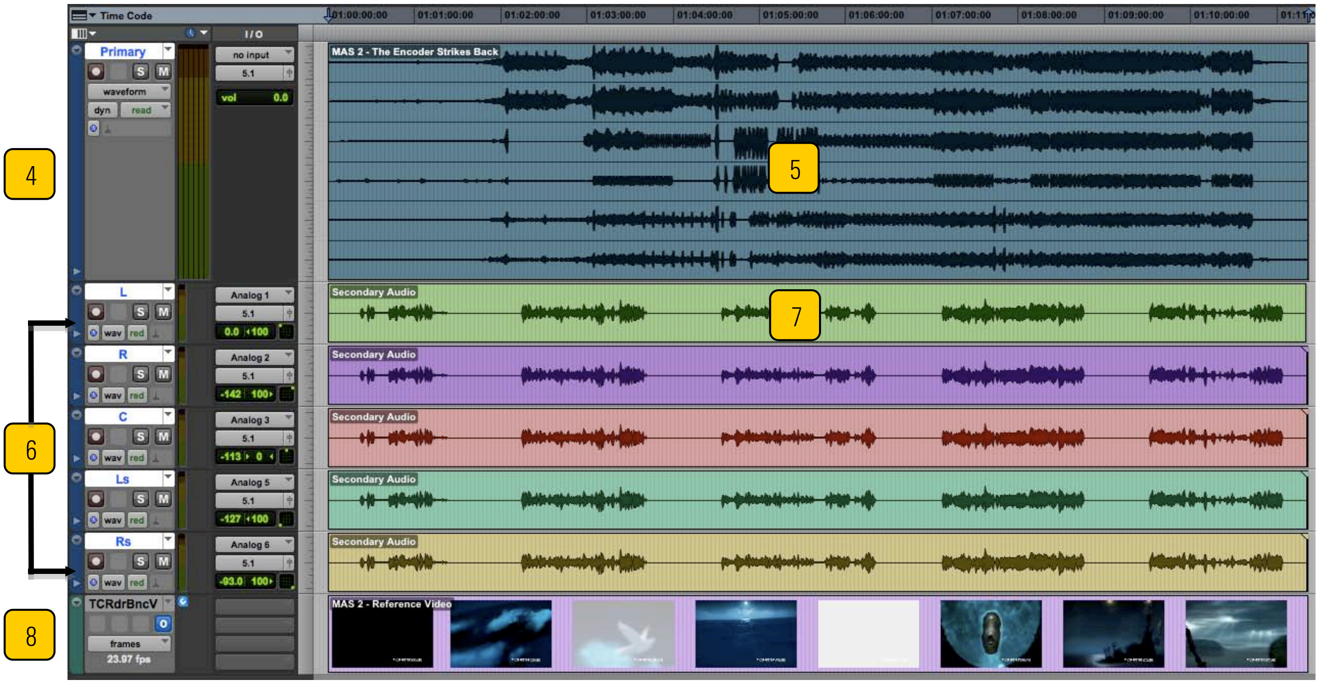
---

2. Session Setup

1. Create a new 48k session, 24 or 16 bit supported. (48k only, no 96k) Note: Secondary Audio is limited to 48 kHz.
2. Use Timecode as your Main Counter format.
3. Check Session Setup and choose the appropriate Time Code Rate of 23.976, 24, 25, or 29.97fps (Non-Drop or Drop) to match video. Pro Tools will not export at 30fps (Non-Drop or Drop).



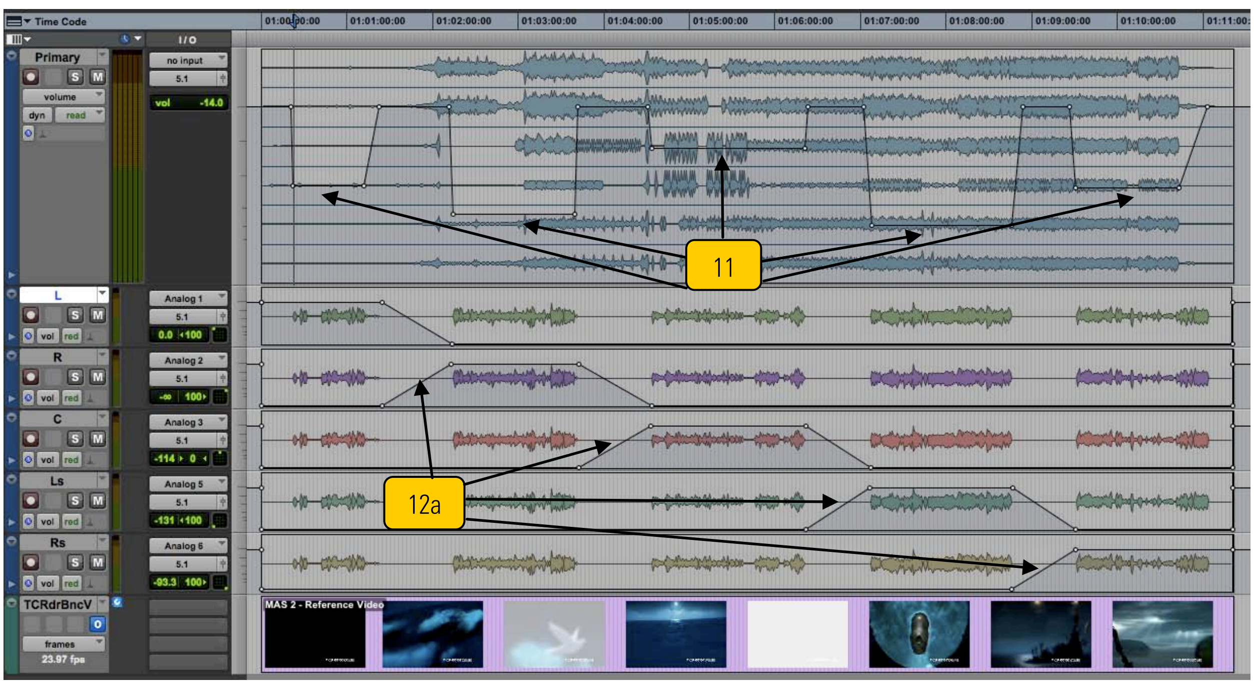
4. Create a single multi-channel audio track for your Primary PCM audio.
5. Load your Primary PCM audio into your multi-channel audio track.
    a. **NOTE: Your Primary Audio must be one solid region.**
    b. **NOTE: The Primary Audio should be spotted to match the eventual start and end time of your MAS encode**. (EX: If the first reel starts at 01:00:00:00, then the Primary Audio track should start at 01:00:00:00 in the timeline.)
6. Create an audio track for the Secondary Audio soundtrack (5.1, 2.0, 1.0)
7. Spot Secondary Audio PCM file(s). (Their start and end times should match the final in and out times of the Secondary Audio on the final authored disc.) Note that Secondary Audio can exist as individual regions or one complete region that is equal to or less than the Primary Audio duration as shown below. *(Refer to Section 5, Preparing Secondary Audio for Authoring Systems)*
    a. If you are making a mono Secondary Audio soundtrack which will utilize dynamic panning, create five mono audio tracks and load your mono Secondary Audio PCM file into each of those tracks, as shown below.
    b. Name each of the tracks L, C, R, Ls, and Rs, and bus the outputs accordingly.
8. Import Video to track



9. Verify that the timecode frame rate of the session is the same as the frame rate of the video file in Session Setup.
10. Verify that proper sync of Primary, Secondary, and video tracks is correct.

3. Creating Automation

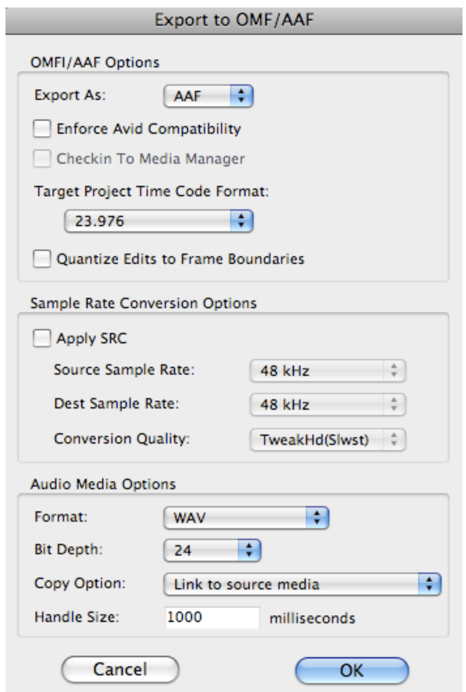
11. On the Primary track, write your volume automation. This is going to reflect how the Primary Audio is attenuated when the Secondary Audio playback is enabled. *(If you will be using Dialog Normalization on your DTS Encodes, please refer to the Appendix, Section B, Dial Norm Note)* When creating your automation points on the Primary track, note the following:
    a. Automation should begin and end with a 0dBFS point/node within the Secondary Audio region boundaries (Drawn on the Primary Track)
    b. Automation data will be thinned to a resolution of approximately 2 SMPTE frames.
    c. If a volume fade greater than 10dB is necessary, DTS recommends the fade to gradually take place over the minimum duration of a second.
    d. ALL volume automation points/nodes must reside on the single Primary Audio region track and must occur within the time span of the Secondary Audio region boundaries.
    e. Any automation value above 0dB will reduced to 0dB upon Encode
    f. The supported range of volume automation on the Primary Track is from 0 to -60dB, INF (negative infinity). Volumes below -60dB will be treated as INF upon encode
12. Unless you are using dynamic panning, the Secondary Audio levels should be at 0dB with no automation. If Secondary Audio level adjustment is necessary, it must be made in the PCM domain.
    a. If you are using dynamic panning, write your Secondary Audio volume automation to reflect where and at what level the mono Secondary Audio will be heard. The same points as stated in Section 11 “a” through “f” apply.



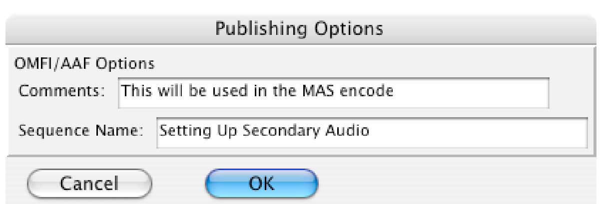
13. Play back both Primary and Secondary audio to verify and QC mixing levels.
14. To begin the AAF export process, click on the name of the Primary Audio track to select it.

a. If you are not using dynamic panning, this should be the only track selected. This will include all volume automation within the Primary Audio region.
b. If you are using dynamic panning, select the Primary Audio track AND each Secondary Audio track. This will include all volume automation within the Primary Audio region, as well as all volume data across Secondary Audio channels.

15. For Pro Tools, go to File>Export>Selected Tracks as New AAF/OMF...The AAF Options should be as follows:
    a. Export as AAF
    b. Uncheck Enforce Avid Compatibility
    c. Ensure Target Project Time Code Format matches your session TC
    d. Uncheck Quantize Edits to Frame Boundaries
    e. Sample Rate Conversion should not be enabled
    f. Audio Format must be set to .wav, Audio Bit Depth should match session
    g. Select the Link to Source Media option. (Any other selections could cause issues.)
    h. Click OK



    i. Publishing Options are at your discretion
    j. Click OK



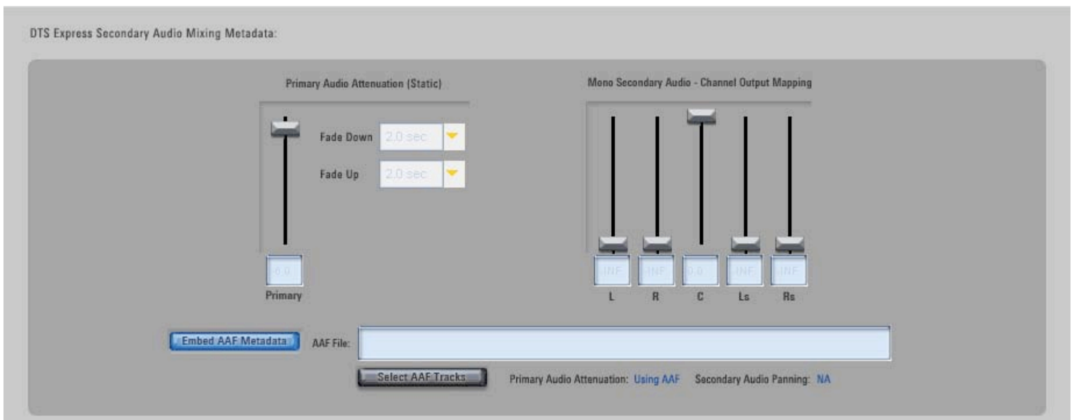
    k. The Save Dialog box will open. Choose the AAF File Name and Destination.
    l. Click Save.

m. Save and Close your session .

---

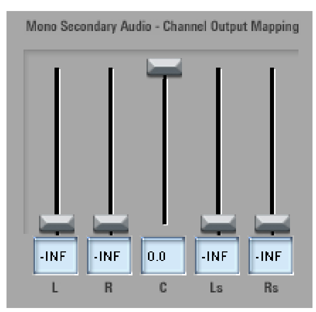
4. Importing AAF into the DTS-HD Master Audio Suite

16. Open the DTS-HD Master Audio Suite encoder.
17. Choose your Destination Format to be Blu-ray Secondary Audio
18. Select your channel layout and load your Secondary Audio wave file(s) *(Refer to Section 5. Preparing Secondary Audio for Authoring Systems)*
19. Choose the desired bit-rate
20. Match your TC Frame Rate and Start/End Times to those established in Pro Tools
21. Go to the Bitstream Panel



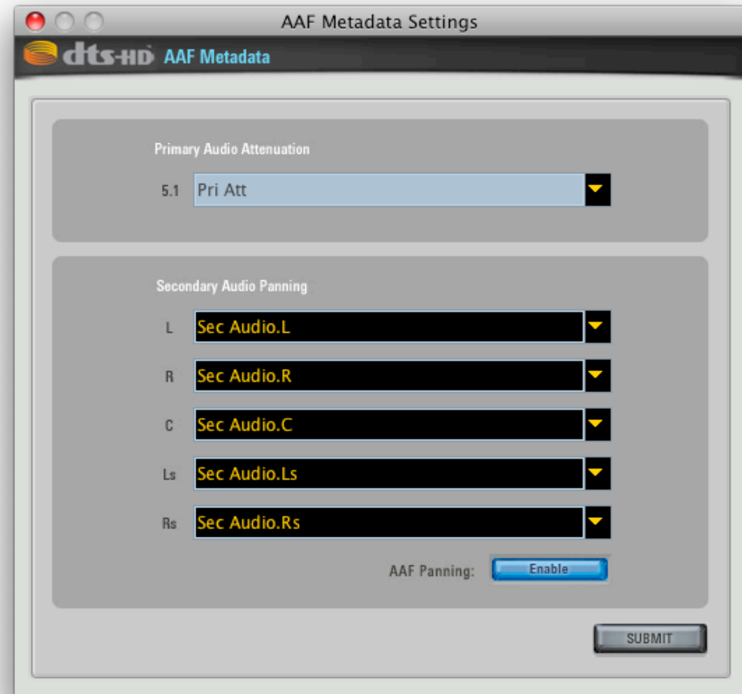
22. Enable the Embed AAF Metadata button 

and select the AAF file you want to use or drag-and-drop your AAF file to load the volume automation for the encode. *
23. If you are using a Mono encode WITHOUT Dynamic Panning, set the static channel output mapping:



24. If you are using a Mono encode WITH Dynamic Panning, enable the Select AAF Tracks button.

a. This will open a new window where you will assign each track in the AAF file to its corresponding track for the Secondary Audio encode via the channel layout drop-down menu.



25. Once you have made all of your selections, click  to exit this window.

26. Go back to the Audio Panel and choose the Save To destination
27. Name the encode appropriately and click the Encode button
28. You are done.

***If you need to encode multiple Secondary Audio streams, there is no need to reload the AAF file.**

5. Preparing Secondary Audio for Authoring Systems

There are two ways to deliver Secondary Audio streams to Authoring. Please confirm with the documentation of the authoring system and/or authoring house that will be handling the integration of Secondary Audio encodes to choose the proper way to proceed.

Out-of-Mux authoring specifies that Secondary Audio be in separate regions, corresponding only to when Secondary Audio is being mixed with Primary Audio. This saves space when bit budgeting by only utilizing bits when audio is needed and avoids encoding silence. However, you must follow the rule about points of automation, noted in Appendix section A, part B. Separate Secondary Audio regions result in more Secondary Audio encodings that need to be performed as each region will be encoded separately, synced by timecode start and end times within the Primary Audio presentation. Out-of-mux authoring is most commonly used for content that is delivered via BD-Live or media that is attached to a BD player.

In-Mux authoring, the most common usage, specifies that the Secondary Audio track be one complete region when being imported into the authoring system. The best practice in this scenario is to consolidate the Secondary Audio regions into one complete region that matches the duration of the Primary Audio track, prior to writing automation in Pro Tools. If the final volume automation has already been written on the Primary Audio track, do not worry. Consolidate your Secondary Audio track and export the region. Load the consolidated Secondary Audio track into the MAS with the same start time as the Primary Audio track. Load the AAF file as stated above and encode.

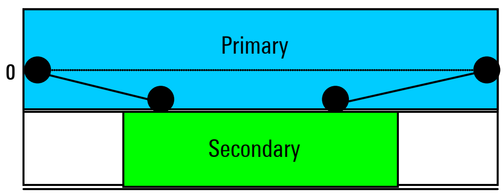
APPENDIX:

a) Automation Rules (Due to the Blu-ray Specifications)

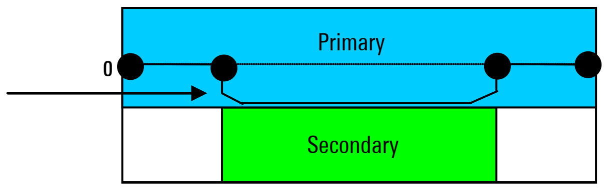
a. ALL automation points/nodes must be contained within the single Primary Audio Region. The AAF file uses the timecode of the Primary Audio region as the reference for the embedded volume automation changes during Secondary Audio playback. If automation lies outside of the Primary Audio file, the AAF does not retain it.

b. If the Secondary Audio files are placed as independent regions, the automation should begin and end with a 0dBFS point/node at the boundaries of each Secondary Audio region (Drawn on the Primary Track). This is because the AAF format does not recognize points outside the region boundaries and this can cause abrupt volume changes. Here is an example:

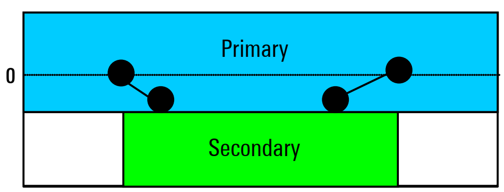
You believe you are creating a smooth fade…



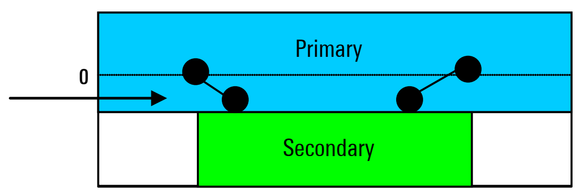
But this is how it is being interrupted.



If you create 0db points at the beginning and end…



Your fades will be recognized properly.



An easy way to achieve 0db points at the start and end of the referenced region:

1) Make sure the A to Z button is on to enable key commands:
2) Make sure that your Primary Track is set to show volume automation on its track.
3) Highlight your entire Secondary Audio region (For this example, it should be on the track below your Primary Audio track.)
4) Hit the P button, which will move your selection area up to the Primary Track.
5) Select the normal trim tool (F6).
6) Click in the middle of the region, which will make the tool face downwards.
7) Click, pull the volume slightly down, and release. This will automatically create points at the exact beginning and end points, relative to the Secondary Audio region below.
8) Fine-tune your points of automation as needed.

b) Advisements

a. Any automation value above 0dB will reduced to 0dB upon Encode
b. The supported range of volume automation on the Primary Track is from 0 to -60dB, INF (negative infinity). Volumes below -60dB will be treated as INF upon encode
b. Automation data will be thinned to a resolution of approximately 2 SMPTE frames.
c. If a volume fade greater than 10dB is necessary, it is recommended the fade to gradually take place over the minimum duration of a second.

DIAL NORM NOTE: If you are using a dial norm level other than -31db for your encoded Primary Track, your Primary Track will be attenuated by the corresponding amount of db when it is mixed in a Blu-ray player (Profile 1.1). For example, if a dial norm setting of 27 is applied, the resulting volume of the Primary Track will actually be 4db lower than what you hear in your Pro Tools session. This will change your overall mix when Secondary Audio is introduced in your final content and not give you an accurate example during the stage of writing your automation. <u>It is recommended to leave your dial norm setting at -31db.</u>

To simulate/test the effect of dial norm on your Primary Track in Pro Tools without affecting your automation:

1) **In the Inserts section of your Primary Track, choose Other>Trim to bring up the Trim plug-in.**
2) **Choose the amount of attenuation that will be applied to the track when your chosen dial norm setting is used (EX: Dial norm of 27 = -4db attenuation)**

This will allow you to monitor how dial norm will affect Primary Audio tracks when mixed in a Blu-ray Player (Profile 1.1) in addition to the volume automation of Primary Audio tracks.

If you must use a different dial norm than -31db for your Primary Track, and you would like your Secondary Audio to have the same attenuation applied, there is currently one solution. Dialog normalization on Secondary Audio is not supported in the MAS Encoder. So, attenuate the final Secondary Audio in your Pro Tools session by the amount equal to the amount of dialog normalization to be applied to the Primary Track, before bouncing/exporting your Secondary Audio track(s) to be encoded in the MAS Encoder. When encoding your Secondary Audio, ignore the dial norm parameter.

c) AAF Error Codes

Double-check your start time and timecode frame rate before encoding in the DTS-HD Master Audio Suite with your embedded AAF file or the following errors might occur:

Error 13018 - AAF File: Parser allocation
Error 13019 - AAF File: Unable to parse specified file
Error 13020 - AAF File: Encode frame rate does not match AAF frame rate of 23.976
Error 13021 - AAF File: Encode frame rate does not match AAF frame rate of 24
Error 13022 - AAF File: Encode frame rate does not match AAF frame rate of 25
Error 13023 - AAF File: Encode frame rate does not match AAF frame rate of 29.97 Drop
Error 13024 - AAF File: Encode frame rate does not match AAF frame rate of 29.97
Error 13025 - AAF File: Encode frame rate does not match AAF frame rate of 30 Drop
Error 13026 - AAF File: Encode frame rate does not match AAF frame rate of 30
Error 13027 - AAF File: Encode frame rate does not match AAF frame rate of 50
Error 13028 - AAF File: Encode frame rate does not match AAF frame rate of 59.94 Drop
Error 13029 - AAF File: Encode frame rate does not match AAF frame rate of 59.94

Error 13030 - AAF File: Encode frame rate does not match AAF frame rate of 60 Drop
Error 13031 - AAF File: Encode frame rate does not match AAF frame rate of 60
Error 13032 - AAF File: Encode frame rate does not match AAF frame rate (unknown)
Error 13033 - AAF File: More than one primary scaling mode was defined
Error 13034 - AAF File: Insufficient number of primary scaling indices were specified
Error 13035 - AAF File: Invalid primary scaling slot index specified
Error 13036 - AAF File: Insufficient number of panning indices were specified
Error 13037 - AAF File: Invalid panning slot index specified
Error 13038 - AAF File: Unable to allocate UTF-16 file buffer
Error 13039 - AAF File: Unable to convert AAF file to UTF-16

14. CSV File Import Supplement – Seamless Branching

1. Required Software / Overview

Required Software
1. Microsoft Excel – any version (Macintosh/Windows) capable of exporting CSV files

<u>Overview</u>
Seamless Branching is a space-saving mechanism supported by Blu-ray Disc players, commonly used when multiple versions of a film are included on a single disc. Instead of storing the entire film on the disc several times, branch points are established within authoring to jump between extended and abridged versions of scenes. For more information about Seamless Branching, please contact your authoring system provider.

In order to mitigate audible anomalies when switching between dissimilar (non-contiguous) PCM waveforms, the DTS decoder within players and AVRs will automatically insert a quick crossfade at each branch point. To further avoid audible anomalies, DTS suggests that a specific encoding filter be temporarily disabled at these points. Please note that disabling this filter will not affect the audio quality or lossless capabilities (DTS-HD Master Audio) of the encoder.

When Seamless Branching has been enabled on the BITSTREAM tab, the DTS-HD MAS Encoder will accept a CSV file containing branch point timecodes at which the encoder will temporarily disable the above-mentioned filter. Stated simply, CSV (comma-separated values) is a file format that has columns separated by commas and rows separated by newlines. Since the CSV file accepted by MAS only utilizes one column, it will actually not contain commas, only newlines for each timecode value.

2. *Single Clip* or *Use CSV Branch Points*?

If the Seamless Branching feature will be utilized on a Blu-ray Disc, all encodings created for that disc should use a Seamless Branching mode: *Single Clip* or *Use CSV Branch Points*.

<u>Single Clip</u>
As described above, the encoder must receive all branch points prior to encoding so that it can disable an encoding filter appropriately. If an encode contains no specific branch points (e.g. an extended or inserted scene), set the destination format to *Blu-ray Disc Primary Audio* on the **AUDIO** tab and enable *Single Clip* within the Seamless Branching section of the **BITSTREAM** tab. <u>The *Single Clip* mode does not require a CSV file</u>.

Use CSV Branch Points

Unlike *Single Clip* mode, *Use CSV Branch Points* mode uses a CSV file containing the same branch points that will be defined in authoring. This mode is often used to encode the complete versions of the film. Follow the steps in the next section to create a CSV file.

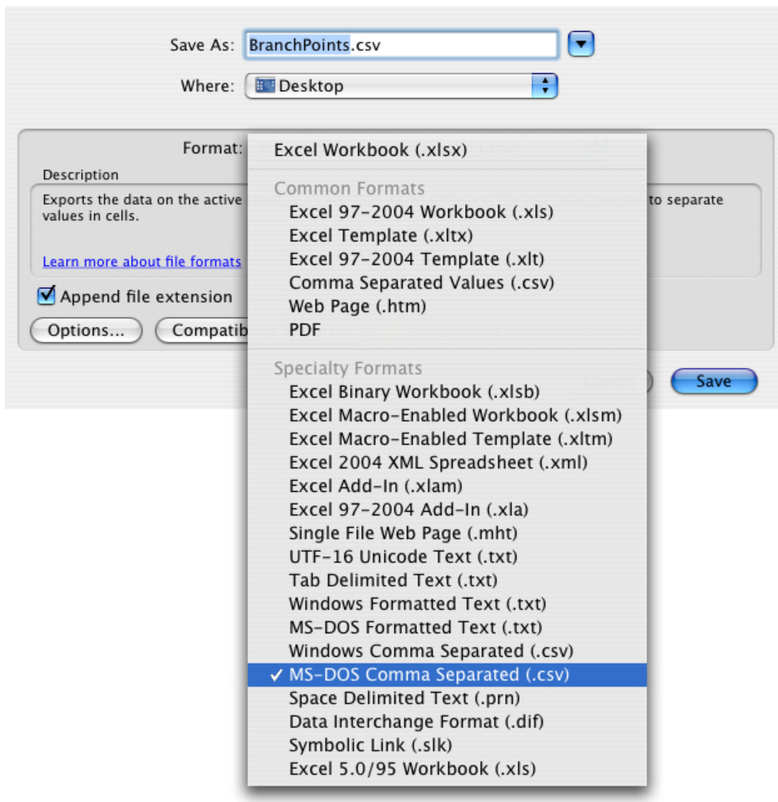
3. Creating a CSV file

   1. Create a new workbook within Microsoft Excel.
   2. Beginning with cell A1, populate column A with all the branch points of the encoding in chronological order.* If the branch points are contained within another document, this step can be easily completed via copy and paste.

	A	B	C	D	E
1	01:03:15:17				
2	01:05:17:12				
3	01:06:13:14				
4	01:07:19:17				
5	01:14:12:19				
6	01:18:14:10				
7	01:34:16:11				
8	01:56:18:13				
9	01:58:19:18				
10	02:08:02:19				
11	02:17:10:14				
12	02:21:11:22				
13	02:23:13:02				
14	02:33:18:20				
15	02:39:06:08				
16					
17					
18					

***Per the SMPTE timecode specification, drop frame is represented with a semi-colon (i.e. XX:XX:XX;XX). If the encoding utilizes drop frame timecode, ensure that the CSV file timecode values contain semi-colons accordingly.**

3. Save as "MS-DOS Comma Separated (.csv)" (File > Save As). **Note: The ".csv" extension must be included in the filename so that the DTS-HD Master Audio Suite can recognize the file.**



4. Importing CSV into the DTS-HD Master Audio Suite

4. Open the DTS-HD Master Audio Suite encoder.
5. Choose the Destination Format *Blu-ray Primary Audio.*
6. Select the channel layout and load wave file(s).
7. Choose a bit rate.
8. Define TC Frame Rate and Start/End Times that coincide with the branch points established within the CSV file.
9. Go to the **BITSTREAM** tab.
10. Enable the *Use CSV Branch Points* button. 

11. Double-click the CSV File field or drag-and-drop the CSV file to load the branch points for the encode.*
12. Go back to the **AUDIO** tab and choose a *Save To* destination.
13. Name the encoding and click the *ENCODE* button.

***Multiple encodes (e.g. foreign or alternate languages) can be performed with a single CSV file; there is no need to reload the CSV file. If branch points within the CSV file are outside the bounds of the encode Start/End Times, those branch points will not be applied to the encode. Please verify applied branch points within the encode log file.**

5. Verifying branch points applied during encoding

14. Open the encode log file stored in the same *Save To* directory as the encode.
15. Locate the *BRANCH POINTS* section and verify that the applied branch points are accurate.

```
****************************************
MAS Version Number = 2.5

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
Using 96/24 Core       = false

INPUT FILES
------------------------------------------------------
Left                       = /Users/dtsdts/Desktop/AUDIO_L.wav
Right                      = /Users/dtsdts/Desktop/AUDIO_R.wav
Center                     = /Users/dtsdts/Desktop/AUDIO_C.wav
Low Frequency Effects      = /Users/dtsdts/Desktop/AUDIO_LFE.wav
Left Side Surround         = /Users/dtsdts/Desktop/AUDIO_Lss.wav
Right Side Surround        = /Users/dtsdts/Desktop/AUDIO_Rss.wav
Left Surround Rear         = /Users/dtsdts/Desktop/AUDIO_Lsr.wav
Right Surround Rear        = /Users/dtsdts/Desktop/AUDIO_Rsr.wav

BITSTREAM SETTINGS
------------------------------------------------------
Program Info                                   =
Using BD Secondary Audio MetaData              = false
Using Express Dialog Mode                      = false
Enable Remapping                               = false

TIME CODE SETTINGS
------------------------------------------------------
Frame Rate             = 23.976
Encode Entire File     = true
Start Time             = 00:59:58:00
End   Time             = 02:50:37:17
Use Reference          = false

OUTPUT LOCATION
------------------------------------------------------
Directory              = /Users/dtsdts/Desktop/
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 not Enabled

----------------------------------------------------

BRANCH POINTS
----------------------------------------------------
01:03:15:17
01:05:17:12
01:06:13:14
01:07:19:17
01:14:12:19
01:18:14:10
01:34:16:11
01:56:18:13
01:58:19:18
02:08:02:19
02:17:10:14
02:21:11:22
02:23:13:02
02:33:18:20
02:39:06:08
```

15. Folder Based Encoding Supplement

1. Overview

Folder-based encoding is designed to streamline workflow when performing a large number of encodes with identical file attributes.
Examples of projects where this can be useful include the following:

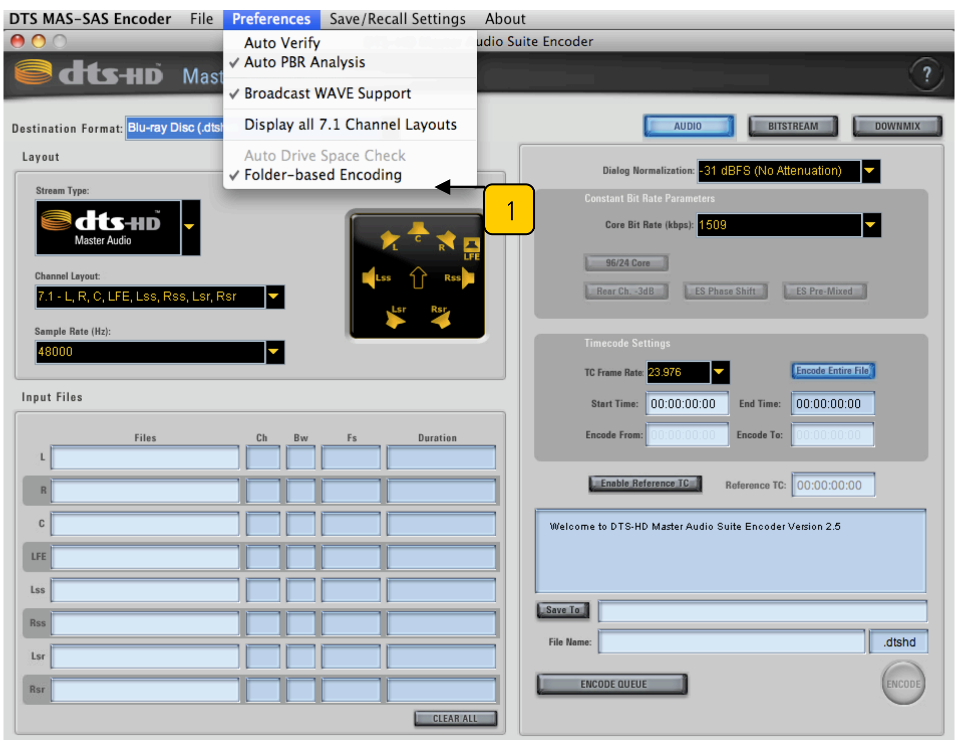
- Multiple language dubs of the same feature
- Multiple episodes of a TV series, (e.g. 5 disc set with 4 episodes each = 20 encodes)
- Downloadable content (trailers, featurettes and other BD live content)
- Game clips

This document explains the operation of this feature.

---

2. Process

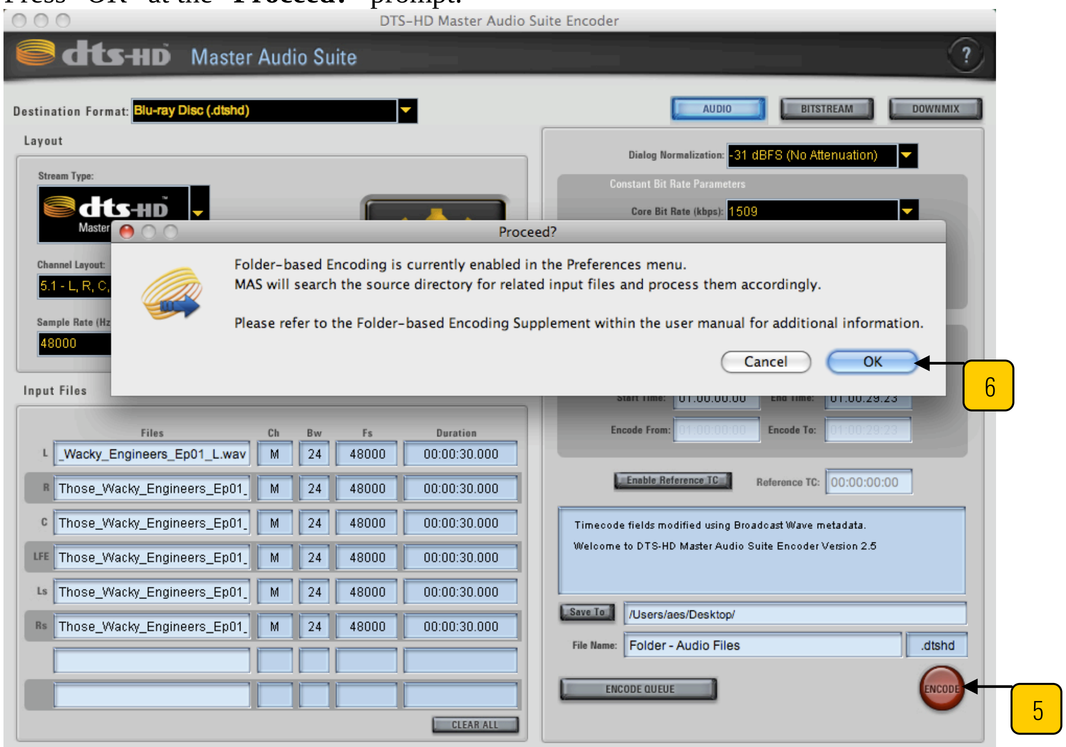
1. Enable **Folder-based Encoding** mode in the Encoder **Preferences** menu.



2. Select appropriate *Destination Format, Stream Type, Channel Layout, and Sample Rate*. These settings will apply to all encodes in the batch.
3. Navigate to the folder containing the input files to be processed, and select a file for one of the encodes. This can be done in one of two ways:

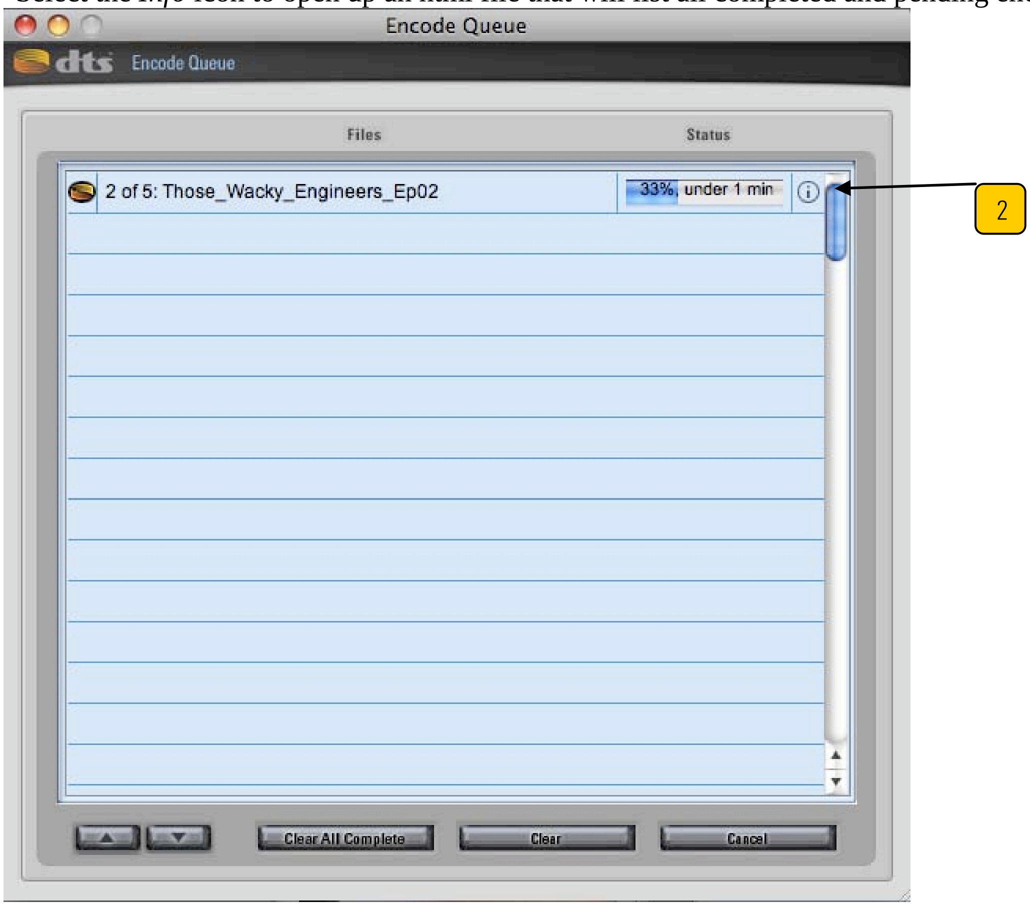
a. Double-click inside the first Input Files window. This will open a browser window. Navigate to the folder that will be the source for the group of encodes, and select an input file.
b. Drag an audio file from the batch folder into an Input File window.

4. Select appropriate *Dialog Normalization, Constant Bit Rate Parameter, Timecode Settings, and Save To* location.
    a. NOTE: Settings will apply to ALL encodes in the batch. Broadcast Wave timecode metadata from the first input files will apply to all encodes.
5. Press *ENCODE*.
6. Press "*OK*" at the "**Proceed?**" prompt.

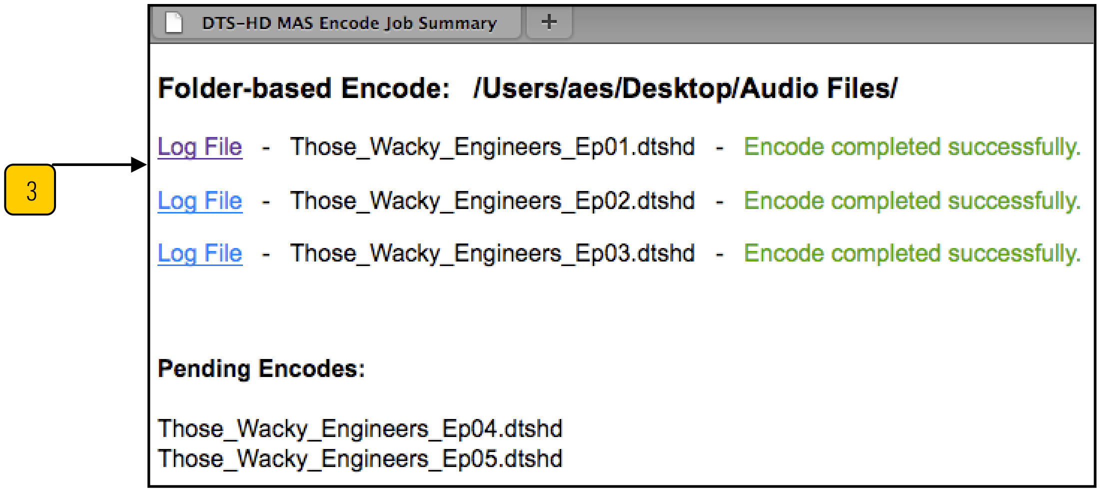


3. Output

1. The **Encode Queue** window will appear. This indicates the number of encodes that will be processed, as well as the status of each individual encode.
2. Select the *Info* icon to open up an html file that will list all completed and pending encodes.



3. Click the "*Log File*" hyperlink if you wish to open the log file for a specific encode.



4. Additional Notes

- Output encode file names will match their corresponding input file names, minus the channel ID descriptor.
- Channel ID's must match the supported DTS channel ID naming conventions. (See Table 7-4-2 Supported Channel IDs and Definitions)
- Audio files for all channels in a selected channel layout must be present for an encode to process.
    - e.g. A 5.1 encode must have all six (L, R, C, LFE, Ls, and Rs) channel files present in the source file folder.
- If a group of files contains more audio files than a channel layout calls for, the additional files will be ignored.
    - e.g. If there are 7.1 files, but the folder-based encoding process calls for 5.1 encodes, then the two additional files in the 7.1 grouping will be ignored.
    - All unused files will be noted in the Job Summary.
- Stereo source files are not supported. All folder-based encode source files must be mono (one channel).
- Auto Drive Space Checking will be disabled when Folder-Based Encoding is enabled.