Meridian MLP Encoder User Guide

MLP™
LOSSLESS

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Contents

Introduction

This chapter gives an introduction to MLP, with information about its benefits and the expected savings it achieves for different materials.

Overview of MLP

Meridian Lossless Packing (MLP) is an encoding system designed to compress high-quality digital audio data with bit-for-bit accuracy. Unlike perceptual, or lossy data reduction, MLP guarantees not to alter the final decoded signal in any way, but merely packs the audio data more efficiently into a smaller data rate.

MLP encoding provides two main benefits:

- It minimises the size of the compressed data, allowing a larger amount of data to be stored in a given capacity.
- It can reduce the maximum instantaneous peak data rate.

This second feature is important for DVD-Audio which has an upper limit of 9.6Mbps. Six channels of 96kHz 24-bit Linear PCM audio have a data rate of 13.824Mbps, which is well in excess of the capability of DVD-Audio. Also at this data rate the data capacity of the disk would limit the playing time to approximately 45 minutes. MLP encoding can reduce the worse case data rate to 9.6Mbps, in addition to extending the playing time to the industry norm of 74 minutes.

The following table gives examples of the typical compression that can be achieved with different audio sample rates and wordsizes:

Format	**Minimum**	**Typical**
48kHz, 16-bit	0%	50%
96kHz, 20-bit	40%	55%
96kHz, 24-bit	38%	52%
192kHz, 24-bit	43%	50%

The following table gives examples of playing times on DVD-Audio that can be obtained with different channel, sample rate, and wordsize combinations:

Format	**Playing time**
6 channels, 96kHz, 24-bit	86 minutes
5.1 channels, 96kHz, 24-bit	100 minutes
2 channels, 192kHz, 24-bit	2 hours
2 channels, 96kHz, 24-bit	4 hours
2 channels, 44.1kHz, 16-bit	12 hours
1 channel, 44.1kHz, 16-bit	25 hours (talking book)

How it works

Because MLP was originally designed with consumer applications in mind, one of its design aims is that any complexity must be in the encoder rather than the decoder. The design of the decoder ensures that it will remain lossless irrespective of the hardware platform or processor it is implemented on. In addition, the decoder is driven by information in the bitstream, allowing improved versions of the encoder to be developed without becoming incompatible with the installed base of decoders.

MLP encoding uses a combination of three methods to reduce the data rate:

- Lossless matrixing is used to reduce the correlation between channels.
- Lossless waveform prediction is used to reduce the inter-sample correlation.
- Entropy coding is used to reduce the data rate by efficiently encoding the most likely occurring values in the audio data.

MLP does not make any assumptions about the assignment of channels, or the correlation between channels, but takes advantage of whatever redundancy is present in the overall signal to encode the data using the smallest possible bandwidth.

In addition to these procedures for reducing the data rate, MLP uses stream buffering to reduce the variations in the transmitted data rates, and absorb transients that are hard to compress, in order to ensure a maximum instantaneous peak data rate. The buffer allows the peak data rate to be minimised for virtually all practical audio data.

If the audio data cannot be compressed within the specified peak data rate the MLP Encoder will signal an error. The producer can then use one or more options for reducing the data rate, or reducing the total space used by the recording. These include:

- Reducing the bit width of one or more channels, such as from 24-bit to 22-bit.
- Filtering one channel to LFE.
- Reducing the audio bandwidth; for example, by filtering information above some arbitrary frequency, such as 40kHz when sampling at 96kHz.

All of these options will increase the amount of compression that MLP can achieve, thus increasing the playing time or reducing the peak data rate.

Other features

Two-channel downmix

Content providers will often want to make a two-channel version of a multi-channel audio stream available on a DVD-Audio disk, for consumers who only have a two-channel playback device. One option is to create separate multi-channel and two-channel streams, and write these separately to the disk. However, this requires two separate mastering and authoring processes, and uses disk capacity.

MLP provides an elegant and simple solution to providing a two-channel downmix. The encoder includes lossless matrixing, which can encode a two-channel downmix as a linear combination of the multi-channel mix and encode this alongside the multi-channel version on the DVD-Audio disk. The advantage of this approach is that the producer can listen to the downmix at the encoding stage in the knowledge that it will be delivered bit-for-bit to the end user at the decoding stage. Another advantage is that a two-channel-only playback device does not need to decode the multi-channel stream, and need only decode the stereo.

Other features

In addition to audio the MLP stream can carry hierarchical metadata, which can include:

- Dynamic range control data.
- Ownership and copy protection fields.
- Time codes.
- Descriptive text fields.

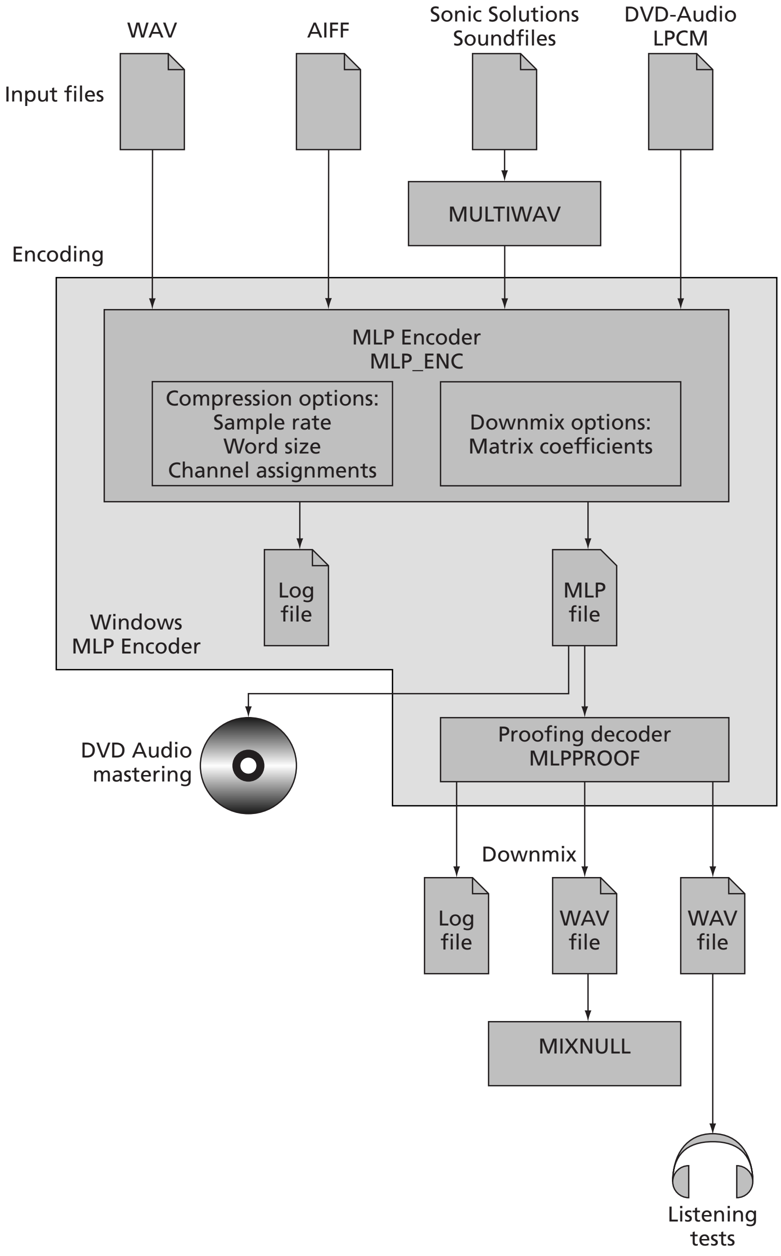
In addition MLP has powerful built-in error detection that allows rapid recovery from bit-stream errors, and prevents any erroneous noises, clicks, or bangs following data errors.

Making an MLP Stream for DVD-Audio

This chapter describes the workflow of an MLP encoding project for DVD-Audio using the Windows-based MLP Encoder, together with the MLP command-line tools described in subsequent chapters of this guide.

Typical workflow

The following diagram shows a typical workflow from the input audio files through to the output MLP file:



The different sections of this workflow are explained in greater detail in the following sections, and in the PDF files on the installation disk.

Encoding

Input files

The MLP Encoder processes audio data from one or more input file. The MLP Encoder directly supports WAV, AIFF, and LPCM (uniform-rate) audio formats, and in addition the command-line version supports raw binary. Files in Sonic Solutions soundfile format, or DVD-Audio LPCM mixed-rate format, can be converted into WAV format for use with the MLP Encoder using the command-line tools MULTIWAV and DVDWAV respectively.

MLP Encoder

The MLP Encoder is an intuitive Windows-based program that simplifies the steps involved in encoding a set of one or more audio files, and then checking the resulting MLP file.

To use the MLP Encoder the following options need to be specified:

Sample rate and wordsize

The sample rate and wordsize will be set to the corresponding values in the input files.

Channel order and assignments

DVD-Audio supports up to six channels in one of 21 alternative channel assignments.

To run the MLP Encoder you need to specify the channel assignment, and the location of each of the channels in the source files.

DVD-Audio channel assignments

Channel Assign	Ch 0	Ch 1	Ch 2	Ch 3	Ch 4	Ch 5	Number in Group 1	Number in Group 2
0	M						1	0
1	L	R					2	0
2	Lf	Rf	S				2	1
3	Lf	Rf	Ls	Rs			2	2
4	Lf	Rf	LFE				2	1
5	Lf	Rf	LFE	S			2	2
6	Lf	Rf	LFE	Ls	Rs		2	3
7	Lf	Rf	C				2	1
8	Lf	Rf	C	S			2	2
9	Lf	Rf	C	Ls	Rs		2	3
10	Lf	Rf	C	LFE			2	2
11	Lf	Rf	C	LFE	S		2	3
12	Lf	Rf	C	LFE	Ls	Rs	2	4
13	Lf	Rf	C	S			3	1
14	Lf	Rf	C	Ls	Rs		3	2
15	Lf	Rf	C	LFE			3	1
16	Lf	Rf	C	LFE	S		3	2
17	Lf	Rf	C	LFE	Ls	Rs	3	3
18	Lf	Rf	Ls	Rs	LFE		4	1
19	Lf	Rf	Ls	Rs	C		4	1
20	Lf	Rf	Ls	Rs	C	LFE	4	2
	Group 1				Group 2			

Key:	M	Mono	C	Centre
	L	Stereo Left	LFE	Low Frequency Effects
	R	Stereo Right	S	Surround
	Lf	Left front	Ls	Left surround
	Rf	Right front	Rs	Right surround

Channel groups

The channel assignments group the channels into two groups, referred to as Group 1 and Group 2. The significance of the group is that the sample rates and wordsize can be specified independently for each group. See *DVD-Audio channel assignments*, above.

The DVD-Audio channel assignments 8 to 12 are almost identical to the channel assignments 13 to 17, and differ only in the way the channels are grouped. For these channel assignments you therefore might want to specify whether the number of channels in Group 1 is 2 or 3 in order to uniquely identify the channel assignment. If this is not specified the encoder will place as many channels as possible into Group 1.

Restart gap

A restart gap of between 8 and 32 can be specified to control the overhead of the MLP stream. The lowest, default setting of 8 allows a restart every 7ms, and this setting gives the fastest start up in cueing and the fastest recovery from a disk error. Selecting the highest permitted value of 32 will increase the apparent disk capacity, but increase the recovery/cue time to 27ms.

Pre-audio gap

To allow players or downstream decoders to lock onto the MLP stream it is recommended that you include a one-second period of silence or irrelevant audio (eg ambient noise) at the start of each track.

Downmix

To encode a downmix from the input channels you need to specify the downmix coefficients to the encoder. Normally this will be done by listening to a suitable mix on an audio workstation, and then reading the fader settings. The command-line version of the MLP Encoder requires the matrix coefficients to be specified as decimal fractions, and the following table is provided to convert from dB values:

dB	+6	+3	0	-3	-6	-9	-12	-15	-18	-21	-24	-∞
Coefficient	2.00	1.41	1.00	0.71	0.50	0.35	0.25	0.18	0.13	0.09	0.06	0.00

The Windows MLP Encoder automatically converts from dB values.

Downmix coefficients can range from +2 (ie +6dB) to 0 (ie infinite attenuation), and from -2 to 0 for negative phase, where 1 corresponds to 0dB.

The downmix process does not prevent overload (clipping) of the stereo output signal, because that may be the artistic intent. When overload occurs, the signal is not clipped in the multi-channel MLP stream (which continues to operate losslessly) but in the player. You can detect an output clip using the Proofing Decoder, MLPPROOF.

Checking an encoded file

It is strongly recommended that all encoded files be checked carefully with the tools provided before proceeding to author them onto a DVD. The following sequence of checks is recommended:

- Check the encoder log file for any error reports.
- Decode the encoded file using MLPPROOF and check its log file for errors.
- Check downmixes by creating a decoded stereo WAV file of the downmix.
- Listen to the decoded audio files.

Logging and reporting

The MLP Encoder and Proofing Decoder each produce log files to confirm the options that have been specified, and report error messages. In the Windows MLP Encoder the log files are displayed in the **Progress** dialogue box so that any errors can be identified, as well as being saved to disk for later reference.

MLP Encoder log

If the MLP Encoder session has been successful the last line of the encoder log should display:

```
Encoding completed
```

If this line is missing do not proceed to decode. Check your input material is correctly formatted (eg view it in an editor or play it back), or re-run the encoding as an isolated process so that the error messages on the console can be seen.

The MLP Encoder log also displays the average bits/sample achieved by encoding.

Checking Proofing Decoder logs

The Proofing Decoder log will end with the line

```
PASS
```

if the stream is correct. Other useful information is also included in the log, including an alert if a downmix will clip in the player, and confirmation of the actual channel assignment, sample rates, wordsizes, and grouping.

MLP file playing time

The MLP encoding process extends the audio sample by up to 1ms of silence. The exact duration is displayed in the log for use in subsequent authoring.

Listening tests

Listening tests are important in case the source file format was incorrect.

Checking downmixes

It is strongly recommended that all downmixes are checked using the following procedure:

- Use the Proofing Decoder MLPPROOF to create a WAV file of the downmix from the MLP Stream.
- Listen to the downmix and confirm it is correct.
- Check the downmix file for errors using MIXNULL.

Using the MLP Encoder

The Meridian MLP Encoder program provides a convenient Windows interface to the MLP tools to simplify the creation of MLP streams for DVD-Audio.

This chapter explains how to install the Meridian MLP Encoder, and use it to create and check MLP streams.

Installing the Meridian MLP Encoder

Requirements

To use the Meridian MLP Encoder, and its associated tools, you need:

- A computer running Windows 95, 98, NT 4 or 2000.
- A 100MHz or faster Pentium-class processor.
- At least 16Mbytes of RAM.
- 5Mbytes of free disk space.

The graphical user interface requires a mouse or similar pointing device – it cannot be operated solely from the keyboard.

In addition, you must have sufficient free disk space for:

- The input audio you intend to encode.
- The resulting MLP file.
- Any downmix auditioning and null-checking files you wish to generate; see *Specifying the checking options*, page 28.

If encoding speed is important to you, it is strongly recommended that you fit 128Mbytes of RAM and use a disk drive or network capable of delivering at least 5Mbytes per second.

As a rough guide, the encoder should run at between 1 and 2 kilobytes of input data per second per megahertz of Pentium II or III CPU speed, depending on circumstances.

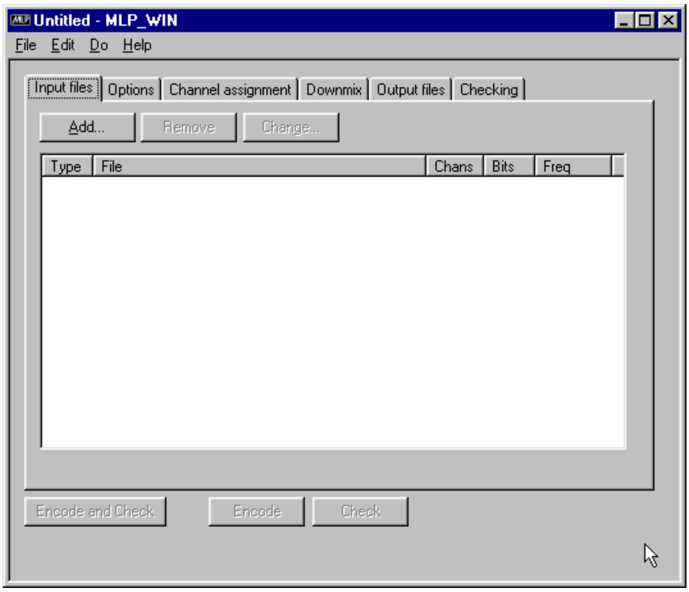
To install the Meridian MLP Encoder

- Insert the Meridian MLP Encoder CD-ROM.
- Follow the instructions on the *Installation Sheet* included in the package.

To run the Meridian MLP Encoder

- Click **Start,** point at **Programs**, **Meridian**, then click **Meridian MLP Encoder**.

The **Meridian MLP Encoder** window will be displayed:



The **MLP Encoder** window provides five tabbed pages of settings, to allow you to specify the settings for a particular encoding session.

The following table summarises the function of each page of settings:

Page	**Allows you to specify**
Input files	Between one and six input files containing the channels of audio data to be encoded.
Options	The encoding options, such as the number of channels and the data rate.
Channel assignment	How the channels in the input files will be mapped to the channels in the final encoded file.
Downmix	How the channels in the input files should be mapped to the stereo downmix.
Output files	The filenames for the encoded output file, and a file containing encoding statistics.
Checking	Options for the optional checking stage, including the name of the report file, and the name for optional decoded files.

The operation of each of the pages of settings is explained in greater detail in the following sections.

The buttons along the bottom of the window allow you to encode and/or check the files you have selected. If encoding or checking cannot be preformed the corresponding buttons are greyed out, and the reason is displayed below the buttons; for example:

Can’t encode: No input files

Creating a new project

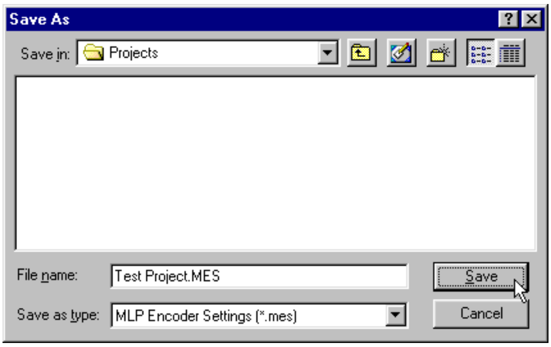
The combination of input files, output files, and settings that you create using the MLP Encoder is referred to as a project. You can save the project settings in a project file, which can then be reloaded at a later date to perform the same encoding, or for use as the starting point for a similar encoding.

The project file is a text file containing a list of the command-line commands, together with their parameters, that perform the encoding and checking process. It can therefore be useful as a template to create a batch file, to perform batch encoding and checking of a series of related audio files.

To create a new project

- Choose **Save** from the **File** menu.

The following dialogue box prompts you to name the project file:



Save in: Projects

File name: Test Project.MES

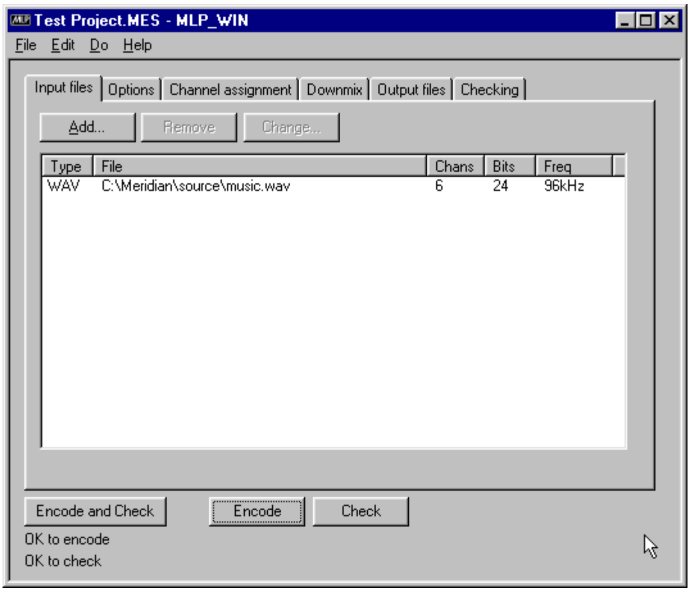
Save as type: MLP Encoder Settings (*.mes)

- Enter a suitable name for the project and click the **Save** button.

The project name is displayed in the **MLP Encoder** window title bar.

Specifying the input files

The **Input files** page allows you to specify between one and six input files containing the audio channels for encoding. The files can be mono or multichannel, and the encoder can use part of the file.



Type	File	Chans	Bits	Freq
WAV	C:\Meridian\source\music.wav	6	24	96kHz

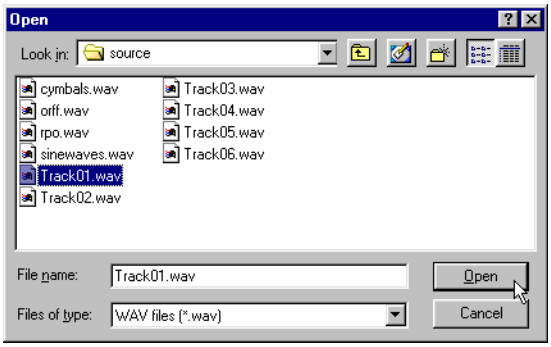
OK to encode
OK to check

The MLP Encoder supports WAV files, with a `.wav` extension, AIFF files, with a `.aif` or `.aiff` extension or fixed-rate DVD-A LPCM files, with a `.pcm` or `.bin` extension. If you have input files in Sonic Solutions Soundfiles format or DVD-A LPCM mixed-rate format you should first convert these to WAV format using the command-line utilities DVDWAV and MULTIWAV respectively. For more information refer to the online documents *Using DVDWAV* and *Using MULTIWAV*.

To specify the input files

- Click the **Add...** button.
- Choose the type of file from the **Files of type** drop-down menu.

- Select the file and click the **Open** button to add it to the list of input files:



Open

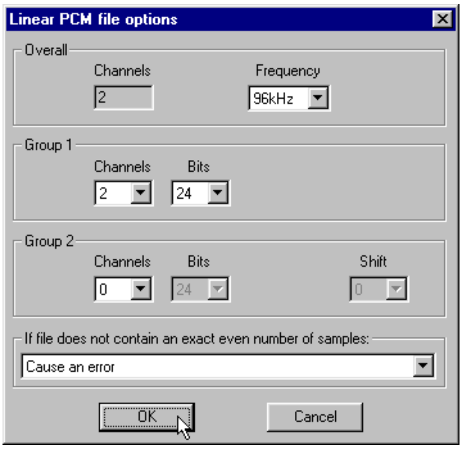
Look in: source

cymbals.wav
orff.wav
rpo.wav
sinewaves.wav
Track01.wav
Track02.wav
Track03.wav
Track04.wav
Track05.wav
Track06.wav

File name: Track01.wav

Files of type: WAV files (*.wav)

If the file is DVD-A LPCM format the following dialogue box is displayed to allow you to specify how the data should be interpreted:



Linear PCM file options

Overall
Channels: 2
Frequency: 96kHz

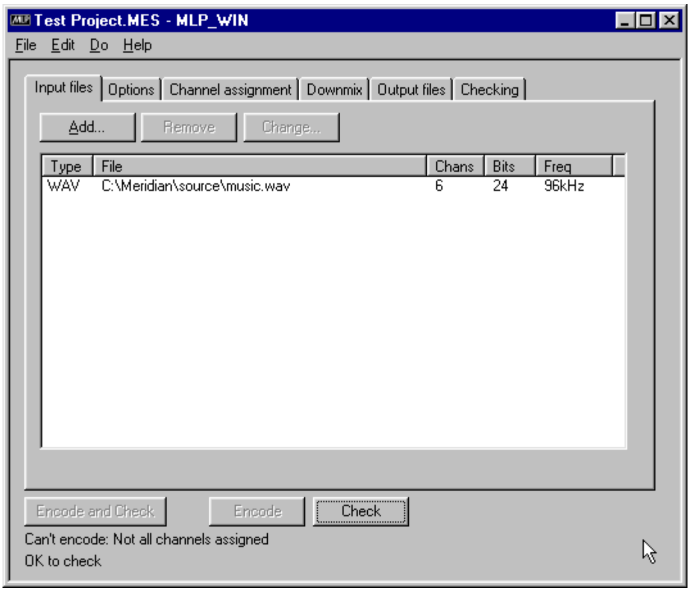
Group 1
Channels: 2
Bits: 24

Group 2
Channels: 0
Bits: 24
Shift: 0

If file does not contain an exact even number of samples: Cause an error

- Specify the data rate using the **Frequency** drop-down menu.
- Specify the number of Group 1 and Group 2 channels using the **Channels** drop-down menus, and the number of bits for each group.
- If there are Group 2 channels use the **Shift** drop-down menu to specify the bit shift.
- Use the drop-down menu to specify whether incomplete data should cause an error, be padded with zeros, or truncated.
- Repeat for any other files you want to add.

The files are displayed in the **Input files** page:



Type	File	Chans	Bits	Freq
WAV	C:\Meridian\source\music.wav	6	24	96kHz

Can't encode: Not all channels assigned
OK to check

To remove a file

- Select the file you want to remove by clicking in the **Type** column.
- Click the **Remove** button.

To replace a file

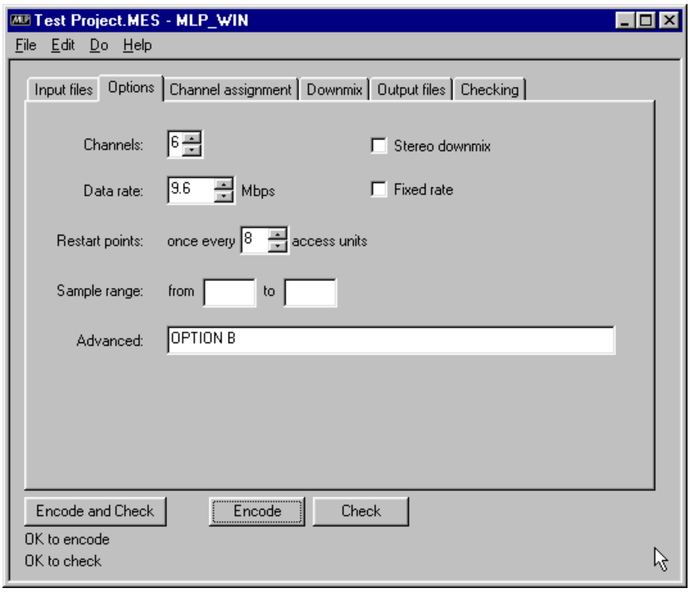
- Select the file you want to replace by clicking in the **Type** column.
- Click the **Change...** button.

The **Open** dialogue box shows a list of available files.

- Select the file and click the **Open** button to add it in place of the selected file.

Specifying the encoding options

The **Options** page allows you to specify options affecting the encoding process. Initially the options are set to default values, but you can override these if you want to have specific control over the encoded file.



Channels: 6

□ Stereo downmix

Data rate: 9.6 Mbps

□ Fixed rate

Restart points: once every 8 access units

Sample range: from to

Advanced: OPTION B

OK to encode
OK to check

Channels

Specifies the number of channels to encode.

Stereo downmix

When three or more channels are being encoded this option allows you to include a stereo downmix in the encoded file. The **Downmix** page allows you to specify the parameters of the downmix.

Data rate

Specifies the data rate for the encoded file; defaults to the maximum allowed on DVD-Audio, 9.6 Mbps, but can be set to lower values for other applications.

Fixed rate

Normally MLP streams are variable rate. This option holds the data rate at the value specified by **Data rate**, and should only be selected for special applications, such as building CD streams.

Restart points

A value between 8 and 32 controlling the overhead of the MLP stream. The lowest default setting of 8 allows a restart every 7ms, which gives the fastest startup in cueing and the fastest recovery from a disk error. Selecting the highest permitted value of 32 will increase the apparent disk capacity, but increase recovery/cue time to 27ms. This setting has an important effect on disk capacity at low sample rates (44 or 48kHz), and can be used to assist in encoding very noisy material at higher rates.

Sample range

Allows you to specify a sample range to encode a subset of the data. If no sample range is specified the entire data is encoded.

Advanced

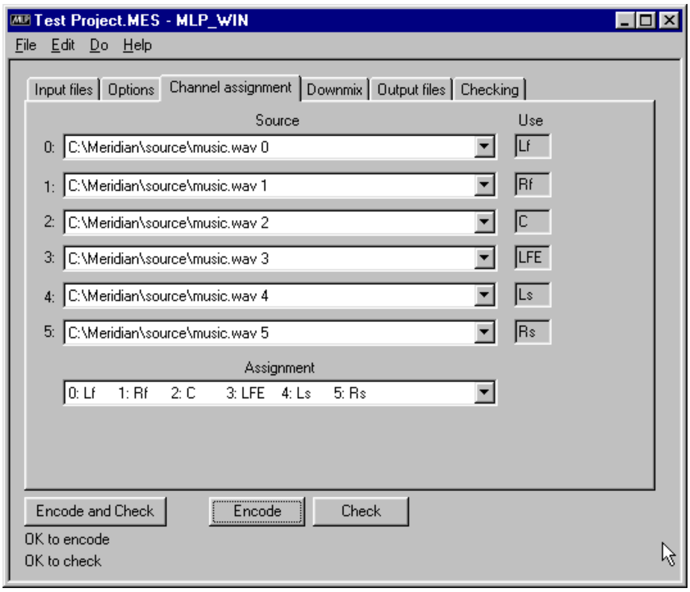
Allows you to specify one or more encoder commands, separated by semicolons, in the same format as for control files; see *Control files and interactive mode*, page 37.

The following advanced commands are useful:

Command	**Description**
OPTION B	Should be specified when including a stereo downmix and the input files have 22 bits or more per channel.
OPTION H	Specifies that the material was HDCD encoded.

Specifying the channel assignments

The **Channel assignment** page allows you to specify the mapping between the channels from the input files you specified, and the channels in the final encoded MLP file.



	Source	Use
0:	C:\Meridian\source\music.wav 0	Lf
1:	C:\Meridian\source\music.wav 1	Rf
2:	C:\Meridian\source\music.wav 2	C
3:	C:\Meridian\source\music.wav 3	LFE
4:	C:\Meridian\source\music.wav 4	Ls
5:	C:\Meridian\source\music.wav 5	Rs

Assignment: 0: Lf 1: Rf 2: C 3: LFE 4: Ls 5: Rs

OK to encode
OK to check

The **Source** menus contain an entry for each channel in each of the input files you have specified on the **Input files** page, and allow you to assign one to each of the output channels.

The **Assignment** menu allows you to select one of the DVD-Audio channel assignments for the number of channels you have specified on the **Options** page. See *DVD-Audio channel assignments*, page 8.

The order of channels in channel assignments 8 to 12 is the same as the order of channel assignments 13 to 17 respectively. These differ in the number of channels in each group. For these the MLP Encoder assumes channel assignments 13 to 17, with three channels in Group 1.

To specify a channel assignment

For each channel to be encoded:

- Specify the source file and channel combination from the appropriate **Source** drop-down menu.

The **Source** drop-down menus contain an entry for every channel in each of the source files.

- Specify the channel assignments from the **Assignment** menu.

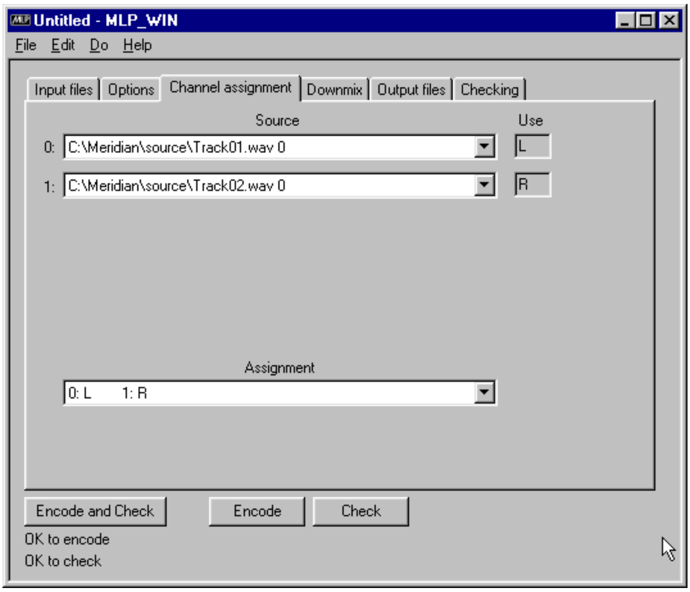
The assignment for each channel is shown in the **Use** column.

The assignments are as follows:

Meaning	Description
L, Lf	Left front
R, Rf	Right front
C	Centre (or mono)
LFE	Low Frequency Effects
Ls	Left surround
Rs	Right surround
S	Surround

Examples

The following illustration shows an assignment corresponding to DVD-Audio channel assignment 1, with each channel provided in a separate audio file:

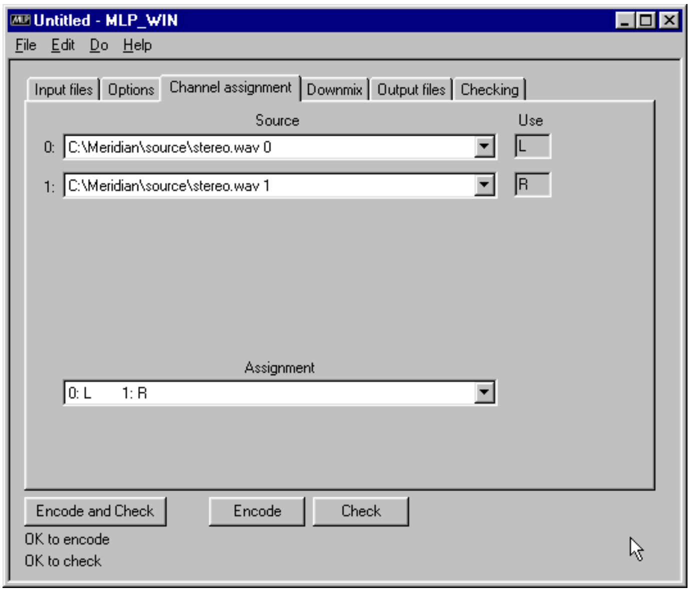


Source | Use
0: C:\Meridian\source\Track01.wav 0 | L
1: C:\Meridian\source\Track02.wav 0 | R

Assignment: 0: L 1: R

OK to encode
OK to check

An alternative example is where both channels are provided in a single input file. In this case you would specify the same source file name for each of the channels to be encoded:



0: C:\Meridian\source\stereo.wav 0 L

1: C:\Meridian\source\stereo.wav 1 R

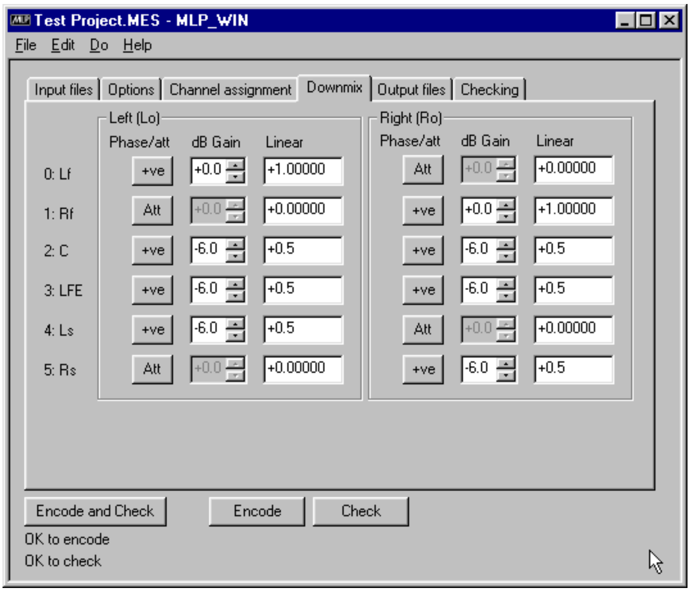
Assignment: 0: L 1: R

OK to encode

OK to check

Specifying the downmix

If you have checked the **Stereo downmix** option on the **Options** page, the **Downmix** page allows you to specify the contribution of each channel in creating the downmix.



	Left (Lo)			Right (Ro)		
	Phase/att	dB Gain	Linear	Phase/att	dB Gain	Linear
0: Lf	+ve	+0.0	+1.00000	Att	+0.0	+0.00000
1: Rf	Att	+0.0	+0.00000	+ve	+0.0	+1.00000
2: C	+ve	-6.0	+0.5	+ve	-6.0	+0.5
3: LFE	+ve	-6.0	+0.5	+ve	-6.0	+0.5
4: Ls	+ve	-6.0	+0.5	Att	+0.0	+0.00000
5: Rs	Att	+0.0	+0.00000	+ve	-6.0	+0.5

OK to encode
OK to check

You can specify each downmix coefficient either as a dB value and a phase, or as a linear coefficient, and the program will automatically convert between the two. For more information see *Downmix*, page 9.

To specify the downmix coefficients

For each channel:

Either:

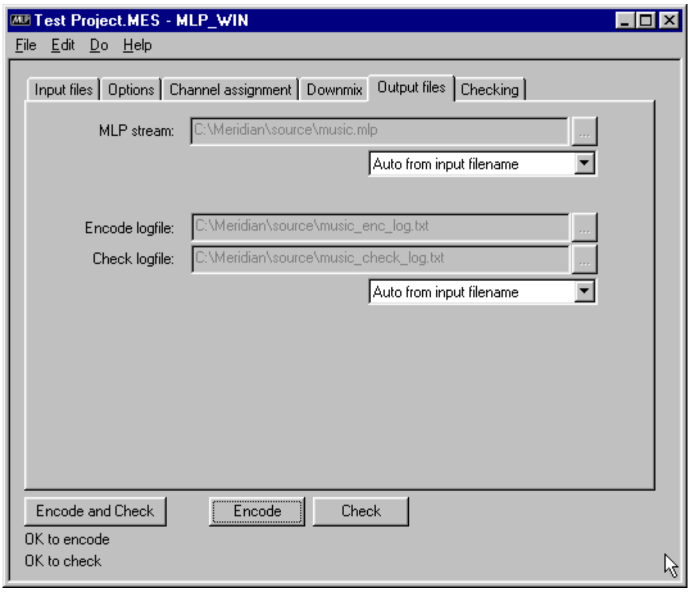
- Click the **Phase/att** button to cycle between **Att** (attenuated), **+ve** (positive phase), or **-ve** (negative phase) and enter a dB value from +6dB to -24dB.

Or:

- Enter a linear coefficient from +2 to 0 for positive phase or from -2 to 0 for negative phase.

Specifying the output files

The **Output files** page allows you to specify the names of the files to contain the encoded MLP stream, and the encode and check log files.



MLP stream: C:\Meridian\source\music.mlp

Auto from input filename

Encode logfile: C:\Meridian\source\music_enc_log.txt

Check logfile: C:\Meridian\source\music_check_log.txt

Auto from input filename

OK to encode

OK to check

The filename defaults to the name of the first input file, with a `.mlp` extension.

To specify the output filenames manually

- Choose **Manual** from the appropriate drop-down menu.
- Edit the pathname, or click the **...** button and specify the filename in the **Save As** dialogue box.

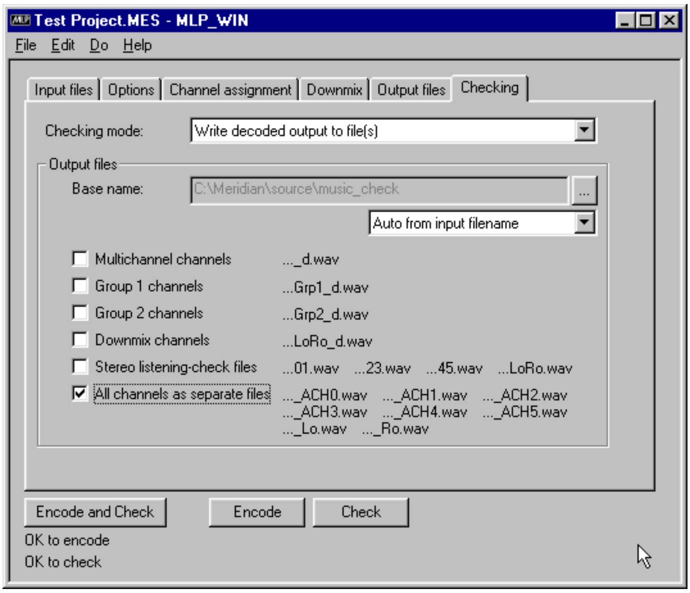
To specify the output filenames automatically

- Choose **Auto from input filename** or **Auto from settings filename** from the appropriate drop-down menu to name the output files from the input or settings file respectively.

Output file	**Name**
MLP stream	`input.mlp`
Encode logfile	`input_enc_log.txt`
Check logfile	`input_check_log.txt`

Specifying the checking options

The **Checking** page allows you to specify the options affecting the checking stage, to validate the encoded output file.



Checking mode: Write decoded output to file(s)

Output files

Base name: C:\Meridian\source\music_check

Auto from input filename

- □ Multichannel channels ...._d.wav
- □ Group 1 channels ...Grp1_d.wav
- □ Group 2 channels ...Grp2_d.wav
- □ Downmix channels ...LoRo_d.wav
- □ Stereo listening-check files ...01.wav ...23.wav ...45.wav ...LoRo.wav
- ☑ All channels as separate files ..._ACH0.wav ..._ACH1.wav ..._ACH2.wav ..._ACH3.wav ..._ACH4.wav ..._ACH5.wav ..._Lo.wav ..._Ro.wav

OK to encode
OK to check

To specify the checking options

- Choose the appropriate option from the **Checking mode** drop-down menu.

The following options are provided:

Option	Description
DVD-Audio compliance check only	Performs a compliance check to verify that the output file conforms with the DVD-Audio specification.
Write decoded output to file(s)	Performs the compliance check and converts the encoded file back into unencoded WAV format files, so that these can be listened to or analysed to ensure that the encoding process has been performed correctly.

To create decoded output files

- Choose **Write decoded output to file(s)** from the **Checking mode** drop-down menu.
- Choose **Manual** from the **Base name** drop-down menu to specify the base name to be used for the decoded files manually, or one of the **Auto** options to derive the base name from the input, settings, or MLP stream filenames.

- Select the output files you want to create.

The following options are available:

Option	Description
Multichannel channels	A single file containing all the channels in the encoded file.
Group 1 channels	A file containing the Group 1 channels in the encoded file.*
Group 2 channels	A file containing the Group 2 channels in the encoded file.*
Downmix channels	A two-channel file obtained by decoding only the downmixed channels in the encoded file using the specified downmix parameters.
Stereo listening-check files	A set of four two-channel files containing the channels in pairs, and a downmix The decoded outputs are also crudely downsampled (every second sample at 88.2 and 96kHz, and every fourth sample at 176.4 and 192kHz). The option is useful for being able to do basic listening checks using sound cards which only work up to 48kHz.
All channels as separate files	A set of eight files containing each of the channels, and the two downmixed channels, as separate files.

* For an explanation of Group 1 and Group 2 channels see *DVD-Audio channel assignments*, page 8.

Any combination of options can be selected to create multiple sets of files for different listening checks. For detailed information about each output option see *Appendix B – Command-line Proofing Decoder*, page 51.

Note that WAV files have a size limit of 4 Gbytes. If you are checking a large MLP file you may need to check only the downmix, or all channels as separate files.

Encoding the project

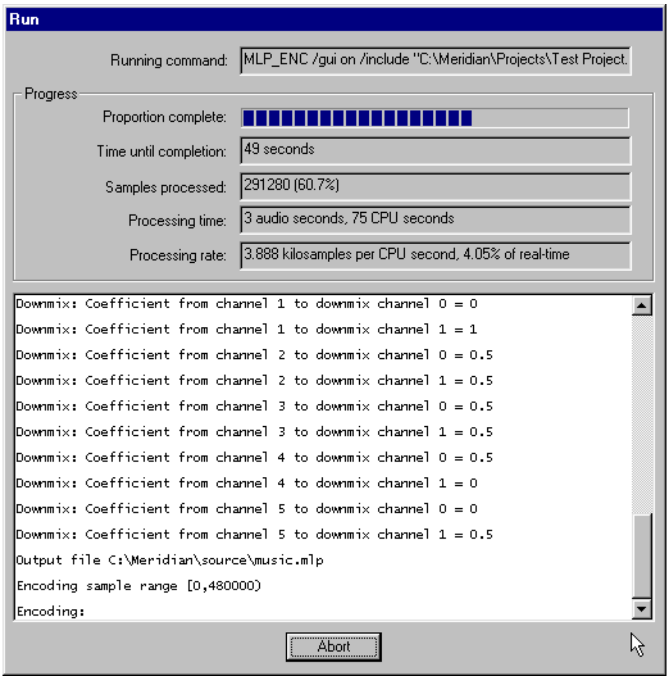
After specifying the options on each of the pages of options it is good practice to save the project before proceeding with the encoding and checking process.

- Choose **Save** from the **File** menu.

To encode the project

- Click the **Encode** button at the bottom of the **MLP Encoder** window.

The following dialogue box is displayed to show the progress of the encoding process, together with estimates of the time until completion, the processing time, and the processing rate.



Running command: MLP_ENC /gui on /include "C:\Meridian\Projects\Test Project.

Proportion complete:

Time until completion: 49 seconds

Samples processed: 291280 (60.7%)

Processing time: 3 audio seconds, 75 CPU seconds

Processing rate: 3.888 kilosamples per CPU second, 4.05% of real-time

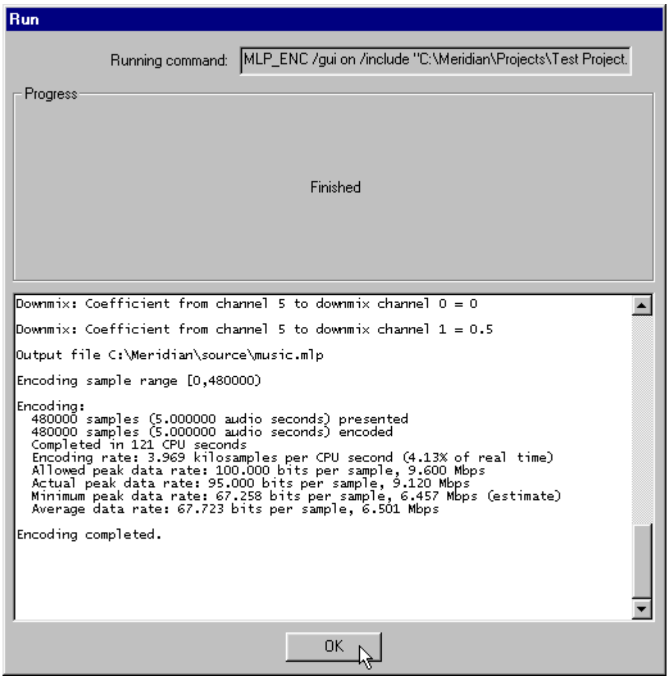
```
Downmix: Coefficient from channel 1 to downmix channel 0 = 0
Downmix: Coefficient from channel 1 to downmix channel 1 = 1
Downmix: Coefficient from channel 2 to downmix channel 0 = 0.5
Downmix: Coefficient from channel 2 to downmix channel 1 = 0.5
Downmix: Coefficient from channel 3 to downmix channel 0 = 0.5
Downmix: Coefficient from channel 3 to downmix channel 1 = 0.5
Downmix: Coefficient from channel 4 to downmix channel 0 = 0.5
Downmix: Coefficient from channel 4 to downmix channel 1 = 0
Downmix: Coefficient from channel 5 to downmix channel 0 = 0
Downmix: Coefficient from channel 5 to downmix channel 1 = 0.5
Output file C:\Meridian\source\music.mlp
Encoding sample range [0,480000)
Encoding:
```

The lower half of the dialogue box displays the encoder status. For more information about interpreting the status refer to *Appendix A – Command-line MLP Encoder*, page 35.

To abort the encoding process before completion:

- Click the **Abort** button.

Once the encoding is complete the **Abort** button changes to an **OK** button, and the **Run** window displays statistics for the encoding process:



Run
Running command:
MLP_ENC /gui on /include "C:\Meridian\Projects\Test Project.
Progress
Finished
OK

```
Downmix: Coefficient from channel 5 to downmix channel 0 = 0

Downmix: Coefficient from channel 5 to downmix channel 1 = 0.5

Output file C:\Meridian\source\music.mlp

Encoding sample range [0,480000)

Encoding:
  480000 samples (5.000000 audio seconds) presented
  480000 samples (5.000000 audio seconds) encoded
  Completed in 121 CPU seconds
  Encoding rate: 3.969 kilosamples per CPU second (4.13% of real time)
  Allowed peak data rate: 100.000 bits per sample, 9.600 Mbps
  Actual peak data rate: 95.000 bits per sample, 9.120 Mbps
  Minimum peak data rate: 67.258 bits per sample, 6.457 Mbps (estimate)
  Average data rate: 67.723 bits per sample, 6.501 Mbps

Encoding completed.
```

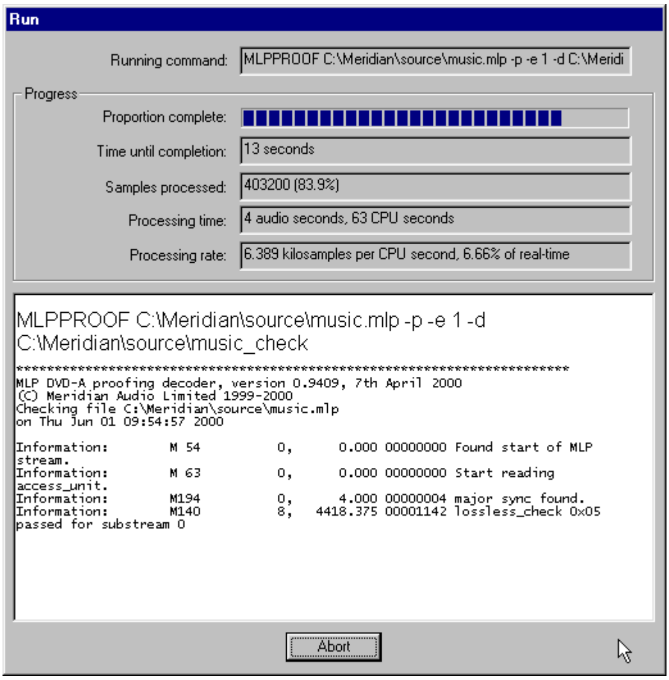
- Click the **OK** button to return to the **MLP Encoder** window.

Checking the encoded file

To check the encoded output file, according to the options specified on the **Checking** options page:

- Click the **Check** button at the bottom of the **MLP Encoder** window.

The following dialogue box displays the progress of the checking stage:



Running command: MLPPROOF C:\Meridian\source\music.mlp -p -e 1 -d C:\Meridi

Time until completion: 13 seconds

Samples processed: 403200 (83.9%)

Processing time: 4 audio seconds, 63 CPU seconds

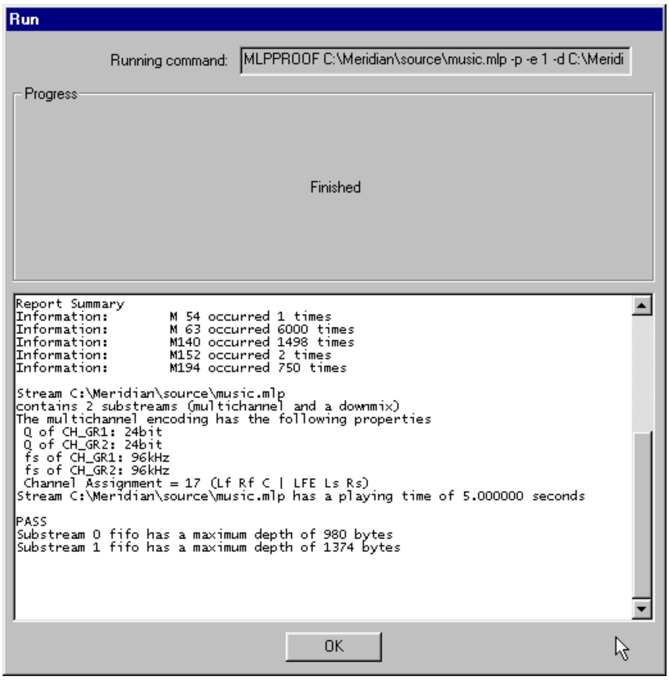
Processing rate: 6.389 kilosamples per CPU second, 6.66% of real-time

MLPPROOF C:\Meridian\source\music.mlp -p -e 1 -d
C:\Meridian\source\music_check

```
************************************************************************
MLP DVD-A proofing decoder, version 0.9409, 7th April 2000
(C) Meridian Audio Limited 1999-2000
Checking file C:\Meridian\source\music.mlp
on Thu Jun 01 09:54:57 2000

Information:        M 54          0,       0.000 00000000 Found start of MLP
stream.
Information:        M 63          0,       0.000 00000000 Start reading
access_unit.
Information:        M194          0,       4.000 00000004 major sync found.
Information:        M140          8,    4418.375 00001142 lossless_check 0x05
passed for substream 0
```

When the check has finished the dialogue box shows a report summary, together with the actual playing time:



Running command: MLPPROOF C:\Meridian\source\music.mlp -p -e 1 -d C:\Meridi

```
Report Summary
Information:        M 54 occurred 1 times
Information:        M 63 occurred 6000 times
Information:        M140 occurred 1498 times
Information:        M152 occurred 2 times
Information:        M194 occurred 750 times

Stream C:\Meridian\source\music.mlp
contains 2 substreams (multichannel and a downmix)
The multichannel encoding has the following properties
 Q of CH_GR1: 24bit
 Q of CH_GR2: 24bit
 fs of CH_GR1: 96kHz
 fs of CH_GR2: 96kHz
 Channel Assignment = 17 (Lf Rf C | LFE Ls Rs)
Stream C:\Meridian\source\music.mlp has a playing time of 5.000000 seconds

PASS
Substream 0 fifo has a maximum depth of 980 bytes
Substream 1 fifo has a maximum depth of 1374 bytes
```

Copies of the logs are saved in the encode logfile and check logfile specified on the **Output files** page.

Assuming no errors are generated you now have a MLP format file suitable for use with DVD-Audio.

Appendix A – Command-line MLP Encoder

This appendix describes the command-line MLP Encoder, MLP_ENC, which is designed to convert an input audio file to an encoded MLP file, with a wide range of options over the encoding process.

Command format

The MLP encoder is controlled by a series of commands, each of which may be followed by one or more parameters. The commands can be entered on the command line, read from a control file, or entered interactively.

Specifying commands

The commands and their parameters can be specified in either upper or lower case, and the order of parameters is not important. In the following descriptions of each command the parameters are shown in italics, and optional parameters are shown in square brackets.

Many of the command names can be abbreviated, as shown in the following table:

Command	Abbreviation
allchannels	ac
assign	a
bitspersample	bps
channel	c
encode	enc
fixed-as-variable-rate	favr
fixed-rate	fr
include	i
interact	int
label	lbl
option	opt
output	o
preencodewav	pewav
progressreport	pr
restartgap	rg
static-downmix	sd
static-downmix-coefficient	sdc
static-downmix-identity	sdi
static-downmix-matrix	sdm
static-downmix-vector	sdv
upsample	up
variable-rate	vr

Entering command on the command line

Commands can be entered on the command line in the following format:

`MLP_ENC /`*`commandname parameters`*` /`*`commandname parameters`* etc.

Control files and interactive mode

Commands can also be supplied in a control file, using the `include` command, or typed interactively, following an `interact` command.

Commands can be put on separate lines, or several commands can be put on one line, separated by semicolons. Comments can be included in the file by prefixing them with a # sign, and blank lines are ignored. For example:

```
commandname parameters
commandname parameters; commandname parameters
# Process next file
```

The MES settings files saved by the **MLP Encoder** can be viewed in a text editor as an example of an encoder include file.

Command categories

The following sections give a detailed description of each of the MLP Encoder commands. The commands are organised into the following categories:

Category	Description
Input	Specify the input files.
Options	Define option affecting the encoding process.
Channel assignment	Specify how the channels in the input file are to be processed.
Downmixing	Define information controlling the downmixing of channels in the encoded file.
Output	Specify the output file for the encoded file, and for encoding statistics.
Encoding	Control the encoding process.

For examples of typical sets of commands see the examples in the following sections.

Input

wav *file*

Specifies a WAV-format input file. This file may have between 1 and 6 channels, but it must be in PCM format, and must have more than 8 bits per sample. By default, single-channel WAV files are assumed to be mono, and two-channel WAV files are assumed to contain the left and right channels in that order.

Examples

```
/wav tracks.wav
```

aiff *file*

Specifies an AIFF-format input file. AIFF is completely supported, but AIFF-C is not permitted.

The following channel assignments are assumed. Note that channels in AIFF are normally indexed starting at 1, but are indexed starting at 0 within `MLP_ENC`:

Channels	0	1	2	3	4	5
1	M					
2	L	R				
3	L	R	C			
4	*	*	*	*		
5	*	*	*	*	*	
6	L	†	C	†	R	S

* AIFF permits several incompatible alternatives, so no assumptions are made.

† Channel uses which AIFF implies, but which cannot be represented in DVD-Audio-compatible MLP channel meanings.

Examples

```
/aiff tracks.aif
```

lpcm *file attribute=value* ...

Specifies an LPCM (Linear PCM) file for input, as used for non-packed audio on DVD-Audio disks and described in section 7.2.4.1 of Part 4 of the DVD specification for read-only disks. Because the format is completely unknown the data must be explicitly specified by the attribute values:

Attribute	**Name**	**Description**
fs=*sample-rate*	Sample rate	Samples per second.
g1=*channels*	Group 1 channels	Number of channels in group 1.
g2=*channels*	Group 2 channels	Number of channels in group 2.
ch=*channels*	Channels	Total number of channels; equal to g1+g2.
ww=*word-width*	Word width	Word width of the samples, in bits.
w1=*word-width*	Group 1 word width	Word width of samples in group 1.
w2=*word-width*	Group 2 word width	Word width of samples in group 2.
bs=*bitshift*	Bit shift	Bit-shift to apply to group 2 (defaults to 0)
rd=*rounding*	Rounding	Behaviour if given an incomplete number of sample pairs.

Only two of **ch**, **g1** and **g2** need to be specified.

Either **ww** may be used to specify the same word width for all channels, or **w1** and **w2** may specify separate widths for the two groups. If group 2 contains no channels, **w2** need not be specified. LPCM files with differing sample rates in groups 1 and 2 are not currently supported.

The input file must contain an exact number of sample pairs. Normally, if this is not the case, an error is raised (rd=Error). Alternatively, the input file can be truncated after the last complete sample pair (rd=Down), or padded with zero bytes to the next complete sample pair (rd=Up).

raw *file attribute=value*

Specifies a raw file for input. Because the format is completely unknown, the data must be explicitly specified by the attribute values:

Attribute	**Name**	**Description**
fs=*sample-rate*	Sample rate	Samples per second.
ch=*channels*	Channels	Number of channels.
ww=*word-width*	Word width	Word width of the samples, in bits.
ws=*word-stride*	Word stride	Offset in bytes between consecutive samples (defaults to ww rounded up and expressed in bytes).

Continued

Attribute	Name	Description
ss=*sample-stride*	Sample stride	Offset in bytes between consecutive sample points for a channel (defaults to ws x ch).
bo=*order*	Byte order	Order in the input data in which the high, medium, and low order bytes of each sample appear. This is expressed as the letters h, m and l in the appropriate order. For word widths of 8 or less, the only choice is 'h'. (Defaults to hml, hm or h as appropriate to ww.)
sb=*n*	Start byte	Offset into the file at which reading should begin (defaults to 0).
eb=*n*	End byte	Offset into the file at which reading should stop (defaults to the end of the file).

Note that, although the raw file reader can cope with files of any size, the start byte and end byte options can only refer to addresses within the first 2Gbytes.

Examples

```
/raw samples.bin fs=48000 ch=2 ww=24
```

label *label*

Specifies the label to attach to the next file that is declared. This command is only required if more than one input file is used, so that they can be distinguished in channel assignment commands.

Examples

```
/lbl A /wav filea.wav /lbl B /wav fileb.wav
```

Options

include *file*

Causes a file to be included as additional commands to the encoder.

Examples

The following example runs the MLP Encoder using commands from `session.ctl`:

```
MLP_ENC /include session.ctl
```

interact

Enters interactive mode. Commands are read from the console and treated as if contained in a file that was included at the point where the interact command occurs.

Type `end` ↵ to finish entering interactive commands.

variable-rate, fixed-rate, fixed-as-variable-rate

Designates the form the output stream should take (default is `variable-rate`).

bitspersample *n*

Specifies the peak bits per multichannel sample the encoder may use. This value may be non-integer. If the stream cannot be encoded within this budget, an error occurs. The following defaults apply:

Sample rate	Bits per sample
44.1 kHz	217.625
48kHz	200
88.2kHz	108.8125
96kHz	100
176kHz	54.375
192kHz	50

These are the maximum bits per sample for the given sample rate.

The bits per sample is equal to the data rate divided by the sample rate.

Example

```
/bps 100
```

restartgap *n*

Specifies how often an access unit must contain a restart. The default is 8, and the parameter must always lie in the range 8 to 32.

Example
Maximum restart gap to maximise available capacity:

```
rg 32
```

iec61937

Equivalent to:

```
/fixed-rate /bitspersample 32
```

except that major syncs are also modified as required for IEC61937 compliance.

range *start end*

Specifies the sample interval to be encoded. By default, this is taken as the intersection of the sample ranges available on the input streams.

Encoding will begin with sample number `start` and finish with the sample immediately before `end`.

Example
To encode exactly the first three minutes at 48kHz:

```
/range 0 8640000
```

where 8640000 = 48000 x 3 x 60.

progressreport *on/off*

Normally the encoder provides a real-time indication of the progress it is making on an encode. If the output is being fed to a file, however, this is unhelpful, and may be turned off using this command.

Unlike other commands, the setting of this parameter is not restored to the default after an encode. Thus it may be specified once on the command line, and applied to all encodes done in a given run.

gui *on/off*

This is a special option for use when the encoder is being driven by a graphical user interface. Its only function is to make progress reports available in a format that can be distinguished from other output, and interpreted then presented graphically.

mes *option*

Will be present in `.mes` command files created by the Windows-based MLP Encoder, and is ignored by the command-line version.

option h/b

Provides the following options:

Command	**Description**
OPTION B	Should be specified when including a stereo downmix and the input files have 22 bits or more per channel.
OPTION H	Specifies that the material was HDCD encoded.

Channel assignment

channels *n*

Specifies the total number of channels to encode. Channels must always be specified, unless implicit in an `allchannels` command.

downmixes *n*

Specifies the number of downmixed channels. Downmixes must be either 0 or 2, and defaults to 0.

group1 *n*

Specifies the number of channels to place in Group 1. The number of Group 1 channels must not exceed the total number of channels, and defaults to the highest number possible.

allchannels [*label*]

This is a shorthand channel assignment, for simple cases. It specifies that the number of channels to encode is exactly the number in the specified input file, and there is a one-to-one mapping between the channels in that input file and encoded channels. If no label is provided, the unlabelled input file is used.

channel *n* [*label*:]*c*

Specifies that encoder channel `n` should have channel `c` from the specified input file associated with it. If the label is omitted, channel `c` from the unlabelled input file is used.

Examples

The following example encodes four channels from `myfilea.wav` and two from `myfileb.wav` (which may have different bit-depths and/or sample rates) to make `myout.mlp`, with the specified channel meanings.

```
MLP_ENC /include mycommands.ctl
```

where `mycommands.ctl` contains:

```
label A; wav myfilea.wav
label B: wav myfileb.wav
channels 6
channel 0 A:0; channel 1 A:1; channel 2 A:2
channel 3 A:3; channel 4 B:0; channel 5 B:1
assignall L R C LFE Ls Rs
output myout.mlp
```

upsample *n*

Upsamples the specified channel. Normally, upsampling is deduced automatically from the combination of sample rates presented to the encoder. This option is only useful in the rare circumstance that all channels should be upsampled, which is only normally of use in testing the upsampling filter, but can be used to upsample content.

assign *n use*

Declares the speaker feed name which should be associated with encoder channel *n*. By default, the use inferred for the corresponding input channel, if any, is assumed. The possible uses are as follows:

Use	Description
Lf or L	Left front
Rf or R	Right front
C or M	Centre or mono
LFE	Low Frequency Effects
S	Surround
Ls	Left surround
Rs	Right surround

assignall *use0 use1*...

Equivalent to:

```
assign 0 use0; assign 1 use1;...
```

The number of uses given must correspond with the number of channels declared.

Downmixing

static-downmix

Specifies that a fixed matrix will be used to provide the downmixed channels. The following commands provide three ways to specify that matrix. Generally $cf_{c,m}$ is the coefficient for mixing encoded channel `c` into mixdown channel `m`.

static-downmix-coefficient *m c* $cf_{c,m}$

Specifies the coefficients one at a time.

static-downmix-vector *m* $cf_{0,m}$ $cf_{1,m}$ ...

Specifies all the coefficient for making a given downmix channel. Must provide exactly C coefficients, where C is the number of channels.

static-downmix-matrix *m* $cf_{0,0}$ $cf_{1,0}$... $cf_{0,1}$ $cf_{1,1}$ ...

Specifies the entire matrix in one go. Must provide exactly C x M coefficients, where C is the number of channels and M is the number of downmix channels.

static-downmix-identity

Provides an identity matrix, which is 1.0 on the diagonal and 0.0 elsewhere, ie mixdown channel N is exactly channel N in all cases.

Output

output *file*

Specifies the file into which encoded output will be placed.

Examples

```
/output myfile.mlp
```

Encoding

encode

Performs an encode, then resets parameters ready for another encode.

The inferences made are:

- If no channel assignment has been performed, and there is exactly one input file, perform an `allchannels` on that file.
- If no upsample commands have been used, perform an upsample on all channels which are at half the main sample rate.
- If the range has not been specified, choose it to be the intersection of the ranges stipulated on all the input channels.
- If no output file has been specified, and there is exactly one input file, take the input filename, strip off the file extension and add `.mlp` for use as the output filename.

Once encoding is complete various statistics are displayed.

The `encode` command may be omitted, and will be assumed as the last command.

You can include it to perform several encodings in a single run of MLP_ENC.

Examples
The following example encodes `myfile1.wav` as `myfile1.mlp`, `myfile2.wav` as `myfile2.mlp` and `myfile3.wav` as `myfile3.mlp`, using default parameters:

```
MLP_ENC /i mycommands.ctl
```

where `mycommands.ctl` contains:

```
wav myfile1.wav; encode
wav myfile2.wav; encode
wav myfile3.wav; encode
```

preencodewav *file*

This command takes the place of `encode`. It infers unspecified parameters (apart from the output filename) and tidies up after itself in the same way. However, instead of encoding, it produces a WAV file containing the data that would have been encoded. This data is represented at 24 bits, regardless of input precision, contains only the interval intended to be encoded, and has been upsampled as appropriate. The output is useful for diagnostic purposes.

abort

Stops the MLP Encoder without reading more commands or performing an encode. Mainly used in interactive mode.

Appendix B – Command-line Proofing Decoder

This appendix describes the MLP Proofing Decoder, MLPPROOF, which can be used to check whether an MLP stream is compliant with the MLP specification for DVD-Audio.

Command format

The Proofing Decoder should be run as a console application under Win32 operating systems.

```
MLPPROOF filename options
```

For example, to proof an MLP stream `test.mlp` the command is:

```
MLPPROOF test.mlp
```

and an example of the console output produced is shown below:

```
****************************************************************************************
MLP DVD-A proofing decoder, version 0.9409, 7th April 2000
(C) Meridian Audio Limited 1999-2000
Checking file source\music.mlp
on Thu Jun 01 10:38:26 2000


Information:      M 54     0,    0.000 00000000          Found start of MLP     stream.
Information:      M 63     0,    0.000 00000000          Start reading access_unit.
Information:      M194     0,    4.000 00000004          major sync found.
Information:      M140     8,    4418.375 00001142       lossless_check 0x05 passed
                                                         for substream 0
Information:      M152     5999, 4062996.000 003dff14 Termination word found at end of
                                                         substream_segment for substream 0.

Stream is MLP compliant from byte 0 onwards.

Report Summary
Information:      M 54 occurred 1 times
Information:      M 63 occurred 6000 times
Information:      M140 occurred 1498 times
Information:      M152 occurred 2 times
Information:      M194 occurred 750 times

Stream source\music.mlp
contains 2 substreams (multichannel and a downmix)
The multichannel encoding has the following properties
Q of CH_GR1: 24bit
Q of CH_GR2: 24bit
fs of CH_GR1: 96kHz
fs of CH_GR2: 96kHz
Channel Assignment = 17 (Lf Rf C | LFE Ls Rs)
Stream source\music.mlp has a playing time of 5.000000 seconds

PASS
Substream 0 fifo has a maximum depth of 980 bytes
Substream 1 fifo has a maximum depth of 1374 bytes
```

Creating test files

MLPPROOF can also be used to make WAV files which can be used for listening or for checking using MIXNULL.

Command categories

The following sections give a detailed description of the Proofing Decoder command-line options. The options are organised into the following categories:

Category	Description
Options	Specify the proofing options.
Output	Specify output WAV files, for listening or checking.

Note that although MLPPROOF can check MLP files of any length, WAV files have an upper size limit of 4Gbytes.

Options

-s *start_byte*

This is simply to allow you to start checking the stream at some point *start_byte* bytes in from the beginning. The default value is 0 which checks from the beginning of the file. Can only be specified within the first 2Gbytes of the file.

-l *length*

This option allows you to specify the length, in bytes, which is to be checked. This (in conjunction with the -s option) allows smaller sections of the stream to be checked. This can be useful when you want detailed outputs for a small section of the stream to help with reporting any problems encountered in an encoder. By default checking will proceed to the end of the file. Can only be specified within the first 2Gbytes of the file.

-e *errorcount*

By default *errorcount* is 1. If more diagnostic information is needed then use a larger `errorcount` number.

-u

This option will enable extra checking particularly relevant in an authoring application.

In particular the stream will be failed if:

- There is data before the start of the MLP stream.
- The termination words which should be at the end of a complete MLP stream are not present.

Output

The output options allow you to generate one or more decoded WAV files from the input MLP stream, for listening or checking. Note that WAV files have an upper size limit of 4Gbytes.

All these options take an optional *root* parameter, which is prefixed to the filenames; if omitted, the default prefix `audio` is used.

-1 [*root*]

Writes one wave file *root*`Grp1_d.wav` containing the Group 1 channels.

-2 [*root*]

Writes one wave file *root*`Grp2_d.wav` containing the Group 2 channels. The output file has the original sampling rate of that group, e.g. if the encoding is from a mixed rate source where Group 1 is at 96kHz and Group 2 is at 48kHz, the WAV file will contain the original 48kHz material.

-m [*root*]

Writes one wave file *root*`_d.wav` containing the all Group 1 and Group 2 channels.

-d [*root*]

Writes one wave file *root*`LoRo_d.wav` containing the downmix channels.

-w [*root*]

Writes four wave files:

Filename	**Description**
*root*_d.wav	Wave file with all the channels
*root*Grp1_d.wav	Wave file with the Group 1 channels
*root*Grp2_d.wav	Wave file with the Group 2 channels
*root*LoRo_d.wav	Wave file with the two-channel downmix

root`Grp2_d.wav` is written for the original sampling rate of that group, e.g. if the encoding is from a mixed rate source where Group 1 is at 96kHz and Group 2 is at 48kHz, the file will contain the original 48kHz material. However the file containing all the channels, *root*`_d.wav`, will have all the material at 96kHz – the Group 2 channels have the interpolated samples as well as the original samples.

`-w` is equivalent to specifying `-1 -2 -d -m`.

-t [*root*]

Writes four wave files each containing pairs of channels.

```
rootLoRo.wav
root01.wav
root23.wav
root45.wav
```

The decoded outputs are also crudely downsampled (every second sample at 88.2 and 96kHz, and every fourth sample at 176.4 and 192kHz). The option is useful for being able to do basic listening checks using sound cards which only work up to 48kHz.

-o [*root*]

Writes a separate wave file for each channel. The files produced are called:

```
root_ACH0.wav
root_ACH1.wav
root_ACH2.wav
root_ACH3.wav
root_ACH4.wav
root_ACH5.wav
root_Lo.wav
root_Ro.wav
```

Where the ACH0 to ACH5 are the output channels according to the DVD Audio channel assignments; see *DVD-Audio channel assignment*, page 8.

Each output is written with the bit-depth and sample rate for the corresponding channel of the input MLP stream.

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