Sonnox Fraunhofer Codec Toolbox
User Guide
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1 Introduction

The Sonnox Fraunhofer Codec Toolbox is designed for real-time auditioning of audio signals by the Fraunhofer codecs, in addition to offline decoding, encoding and metadata editing of the resulting files.

Mixing engineers can produce compensated mixes optimised towards specific target codecs, thereby ensuring maximum fidelity. Similarly, mastering engineers may audition material in the final format, and produce compensated, optimised masters for final encoding and distribution.

The Codec Toolbox Plug-in provides an interface to choose from a range of codec types and settings. When inserted on a DAW main output, the Toolbox plug-in allows real-time comparison between the input signal and the encoded/decoded signal. The plug-in includes meters to indicate input level and the presence of overloads caused by the encoding process. These will require correction to ensure an optimised bit-stream level for the target codec. The NMR METER (Noise-to-Mask Ratio) will indicate the frequency range(s) where encoding artefacts and codec-induced noise might be audible.

Having created your mix and bounced to WAV or AIFF, the Codec Toolbox Manager is where offline encoding takes place, and where metadata can be added or edited. It is also possible to decode previously compressed files to WAV or AIFF formats.
2 Supported Codecs and Formats

Major codecs supported:

- MP3
- AAC-LC
- HE-AAC
- HE-AAC v2
- MPEG Surround
- Apple AAC (iTunes Plus; Mastered for iTunes) (*Mac only*)

Formats supported:

- Mono: MP3, AAC-LC, HE-AAC & HD-AAC
- Stereo: MP3, AAC-LC, HE-AAC, HE-AACv2 & HD-AAC,
- 5.1: AAC-LC, HE-AAC & HD-AAC, MPEG Surround
  - L R C LFE Ls Rs (*SMPTE/ITU*)
3 Summary of Codec Features and Applications

**MP3**
General purpose audio codec, compatible with most players.
*Typical bit rate: 128 kbps (stereo)*

**AAC-LC (AAC Low Complexity)**
High performance audio codec for excellent audio quality at low bit rates. Used for ISDB television (Japan).
*Typical bit rate: 128 kbps (stereo)*

**HE-AAC (High Efficiency AAC)**
High performance audio codec for good quality at bit rates of 32–48 kbps per channel. Used for XM Radio, mobile music downloads and Digital Radio Mondiale.
*Typical bit rate: 64 kbps (stereo)*

**HE-AAC v2**
High performance audio codec for good quality at bit rates of 16–24 kbps per channel. Used for 3GPP music download, Digital Radio DAB+, and internet radio streaming to mobile devices (e.g. iPhone).
*Typical bit rate: 48 kbps (stereo)*

**HD-AAC (High Definition AAC)**
Lossless audio codec with an optional lossy core. Used for music distribution and archival.
*Typical bit rate: roughly half the bit rate of a 16-bit uncompressed WAV file.*

**Apple AAC**
High performance audio codec for excellent audio quality at low bit rates. Used for Apple iPod, iTunes etc.
*Typical bit rate: 256 kbps VBR (stereo)*
4 Terminology

**Bit rate**

Number of bits transmitted or processed per unit of time. This is expressed here in kilobits per second, or kbps.

**CBR**

Constant Bit Rate – bit rate remains the same for the duration of the audio signal

**VBR**

Variable Bit Rate – bit rate changes over time, based on the complexity of the signal. In this case, the VBR value expressed is an average over the length of the file.

**Encoder**

In this context, an encoder converts an uncompressed LPCM audio stream into a smaller, compressed format through the use of lossy compression (HD-AAC excluded).

**Decoder**

In this context, a decoder converts a compressed audio stream back into an uncompressed LPCM audio stream.

**Metadata**

Categorical information about a file, such as the song Title, Artist and Album

**Tag**

One specific piece of metadata, such as the song Title
5 Codec Details

Apple AAC iTunes Plus (Mastered for iTunes) (Mac only)

The addition of the Apple AAC codec allows real-time audition of files produced under the Mastered for iTunes initiative that are destined for the iTunes store.

This codec is available for Mac computers only. It uses the same Apple codecs and re-samplers that are used for the current iTunes Catalogue. It very specifically re-samples to 44.1 kHz if necessary, and encodes as AAC-LC, 256 kbps, variable bit rate and maximum quality, which are the settings used for the iTunes catalogue. For online audition, the signal is then resampled back to the host DAW sample rate if necessary.

This option is available only for stereo configuration. There are no options to change the iTunes codec settings because this would not be representative of the iTunes Plus standard. The Apple codecs and re-samplers are components of the Mac OSX operating system, so can be upgraded during an OSX update. On Windows computers, the closest approximation to the iTunes Plus codec is the Fraunhofer AAC-LC codec set to VBR at 256kbps.

MPEG Surround (Fraunhofer)

MPEG Surround is a feature-rich open standard compression technique for multi-channel audio signals. Operating on top of any core audio codec – including AAC, HE-AAC and MPEG Layer 2 – the system provides a comprehensive feature set, including highest surround and stereo audio quality, in addition to multi-channel support at stereo bit rates.

HD-AAC (Fraunhofer)

The HD-AAC codec from Fraunhofer has a very clever feature; the single compressed lossless file includes a lossy core channel. It therefore acts as a lossless archival format, a lossless distribution format for the masters, and a final playback format for both lossless and lossy decoders — and all of this in a single file.

In the case of HD-AAC playback, if the decoder in your player has full HD-AAC capability, you will be able to listen to a perfect replication of the original WAV file. However, the same HD-AAC file will still play through a decoder that doesn’t have HD-AAC capability, and instead you will be listening to the embedded lossy AAC channel.
The primary purpose of the Codec Toolbox Plug-In is to provide a way of monitoring your host session material using the highly regarded Fraunhofer encoders, without the need for offline processing.

By auditioning and comparing codecs in real-time, the plug-in enables you to make executive decisions about the most appropriate codecs to use.

1. The **INPUT SIGNAL METERS** range from –40 to 0 dBFS, and are calibrated to provide higher resolution in the range from –12 to 0 dBFS in order to be most useful for monitoring final mix/master bus levels.
   - The plug-in includes input sample peak metering and 16-bit clipping indication.
   - Since the input signal is dithered and truncated to 16 bits, red clip lights indicate the occurrence of hard clipping at the plug-in input.
   - Metering options are provided for clip light Hold Time and to Force Channel Configuration (see section 3.2). The options window is displayed by a **right-click** inside the Input Meter area.
2. Click inside the CODEC SELECTOR to reveal the codecs, modes and bit rates available for the current sample rate and plug-in channel configuration. This allows fast selection of different codec/setting combinations.

- The default codec settings are AAC-LC, 256 kbps, CBR (Constant Bit Rate).
- Selected codec settings are immediately reflected in the auditioned audio signal.
- Double-click a codec, mode or bit rate, or move the mouse cursor outside of the selector to close the window.
- Selecting VBR (Variable Bit Rate) will re-generate the bit rate list. The available bit rates may not be the same as for CBR bit rates. In this case, the Toolbox Plug-In will provide the closest possible match to the previously selected CBR bit rate.

3. The NMR METER (Noise-to-Mask Ratio) provides an indication of the frequency areas where the difference between the codec output and the original input might be audible.

4. The bit-stream OVERS meter monitors the peak sample level of the output of the decoder. This provides an indication of the potential for signal overload when the encoded signal is decoded in an end-user’s playback device.

5. The LAUNCH MANAGER button allows you to launch the Manager directly from within the Plug-In.

6. The INPUT/CODEC button is used to smoothly switch the audition path of the plug-in between the selected codec output and the input signal (dithered and truncated to 16-bits). The signal being monitored is indicated by INPUT (blue) and CODEC (green).

   - By default, this button is set to the CODEC position.

7. The STATUS PANEL displays additional information about the codec. Some Codecs specify a particular sample rate, either for some or all bit rates they support. When one of these configurations is selected, the STATUS PANEL will specify whether up-sampling or down-sampling is being carried out by the plug-in.

Changing the session/project sample rate can change the available bit rates for the selected codec. In this case, the closest-possible available bit rate will be selected.

Changing the channel configuration of the plug-in can change the codecs that are available to select. If the previously selected codec is unavailable for the new channel configuration, the codec settings will revert to default (AAC-LC, CBR, 256 kbps).
6.1.1 NMR Meter

The NMR METER (Noise to Mask Ratio) provides an indication of the frequency areas where the difference between the codec output and the original input might be audible.

All lossy codecs will produce a very slightly different output from their input. The very nature of a perceptual coder is that this difference should be inaudible (i.e. masked by the output signal). You can choose to trade off more data compression (and smaller files) against increased audibility of artefacts and codec-induced noise. Theory states that this codec-induced noise should be inaudible when the NMR indicator is green.

Under some circumstances (codec, frequency and input signal dependent) one or more of the NMR LEDs will turn orange, indicating the frequency range where encoding artefacts and codec-induced noise might be audible. The listening environment, training and sensitivity of the listener’s ears are also variable factors that must be taken into account.

It is possible to make very quick comparisons of a selection of different bit rates while auditioning music and monitoring the NMR METER for such potential artefacts.

The NMR calculation is less accurate for parametric codecs (those that use enhancements such as Parametric Stereo or Spectral Band Replication). HE-AAC and HE-AAC v2 use parametric enhancements to achieve very high compression ratios. The NMR METER is still enabled for these codecs, because it can still give an indication of the frequency areas that might require examination.

Note that the NMR METER is not present if the selected codec is lossless (i.e. mp3-HD or HD-AAC). The NMR meter is not valid or displayed for frequencies above 16 kHz, and is not supported at a sample rate of 32 kHz.
6.1.2 Bit-stream OVERS Meter

Any form of filtering can, under certain circumstances, increase the peak level of the signal. If your input is hot, this can potentially produce overloads when the signal is decoded. This effect can be very easily demonstrated with this plug-in.

The bit-stream OVERS meter can be used to monitor this potential for overload. The post-decoder level is continually monitored, and the meter indicates levels over 0 dBFS. The bit-stream OVERS clip LED will indicate that a clip event has occurred. **Right-click** inside the OVERS meter or clip LED to select the told time for the clip light and meter peak indicator.

The mix level should be reduced by the amount required to avoid any overloads being indicated. This may be done manually by lowering the overall mix level, or through use of a Limiter, for example the Sonnox Oxford Limiter.

It is important to note that the indicated bit-stream OVERS metering and clip LED are provided as a guide only. If a segment of audio is being looped or cycled, these maximum values are not exactly repeatable from one loop to the next. Furthermore, in some host applications, repeatedly rewinding to the beginning of the timeline/audio file will also not result in repeatable maximum values. The discrepancies in both cases should be minor, and not all audio or codecs will suffer this.

There is a highly technical reason for this, stemming from the difficulty of synchronizing the start point of the buffer that is used to input data to the codecs.
6.1.3 Channel Configuration

In most DAW hosts, the Codec Toolbox plug-in can automatically detect the number of channels whether it is inserted on a mono, stereo or 5.1 surround track. Some hosts, however, have no method of passing channel configuration information to the plug-in, which subsequently defaults to a surround configuration.

To overcome this limitation, the Toolbox plug-in provides an independent method to force the channel configuration of the plug-in to match that of the host. The options window is displayed by a right-click inside the Input Meter area. These options are only available when the plug-in is instantiated on a six-channel track, or on mono and stereo tracks in host DAWs where they are required.

6.1.4 Dither

The encoders in the plug-in will accept an input signal with greater than 16-bit precision, but in such cases the signal will be dithered and truncated to 16-bits prior to the encoding process.

The plug-in will not introduce dither at the 16-bit level if the input to the plug-in has been already been dithered and truncated to 16-bits.

6.1.5 Data Compression

All of the available codecs compress an audio data bit-stream. Different codecs compress using different algorithms, and will compress to very different degrees.

HE-AAC is optimized for the best quality at very low bit rates, and will produce very small files (potentially around one hundredth of the input file size). Lossless encoders, in contrast, are not so effective at compressing data, and will produce files that are around half the size of the input. The familiar mp3 codecs typically produce compression factors of around 12.

Remember that using perceptual coding is always a trade-off between more accurate reproduction of the original signal and a higher compression ratio and smaller files.
6.1.6 Internal Buffer Sizes

For most codecs, the plug-in uses internal buffer sizes of 1024 samples for up to 48kHz operation and 2048 samples for 48kHz. We recommend that the buffer sizes of the sound card, and thus of the host application, are set to match these figures. Buffer sizes for the HE-AAC and HE-AAC v2 codecs are twice these figures (so 2048 samples for up to 48kHz operation).

The plug-in runs most efficiently and smoothly when supplied with sample buffers of 1024 samples or multiples thereof. Smaller buffer sizes than 1024 samples can cause CPU spiking, which different hosts may have a different tolerance to.

6.2 Creating your Final Mix

When working on finalising your mix or master processing, insert the Toolbox plug-in as the last insert on your master output.

The Toolbox plug-in should be placed after your final limiter and dithering processor.

Use the NMR METER to help assess the areas of your mix that may be affected by the effects of lossy compression. You may adjust areas of your mix, or adjust your Codec, Mode and Bit Rate until you find a suitable balance between audio quality and amount of data compression. (Remember, lower bit rates equate to greater data compression).

Play your mix or master from start to finish.

It is important to pay attention to the bit-stream OVERS meter. If this meter registers greater than 0 dBFS (red), it indicates that your mix or master has the potential to clip the DAC of the end-user’s playback device, leading to undesirable distortion.

The sound of hard or soft clipping can be desirable for certain programme material. However in such cases, it is much wiser to impart this desired sound by using a suitable clipping processor prior to your final dither processing. This will ensure sonic consistency across various playback devices.

To accommodate for bit-stream OVERS, simply reduce signal level prior to your final dither processing, by enough to prevent the Toolbox plug-in OVERS meter from registering greater than 0 dBFS.
When you have decided on suitable codec settings, and optimised your mix/master levels accordingly:

1. **Bypass or disable** the Codec Toolbox plug-in
2. Bounce to uncompressed WAV or AIFF and save the resulting file to your chosen folder.
3. Open the Codec Toolbox Manager. This can either be done by clicking the Manager icon at the top right of the plug-in, or by opening the application directly from disk.
7 The Codec Toolbox Manager

The Codec Toolbox Manager is an application, bundled with the Toolbox Plug-In, primarily intended for encoding files using the latest codec technology from Fraunhofer IIS. In addition, the Manager provides the ability to decode files, and features a metadata editor for adding metadata to your encoded files.

This section provides an overview of the user interface, and discusses each section of the application in detail. Technical specifications are given at the end of this chapter.

The graphical user interface of the Manager is logically separated into five main sections:

1. The **Folder Browser** is used to navigate folders on your file system containing supported media. Selecting a folder containing files supported by the manager will load them into the File List. The selected folder is denoted by an orange highlight.

2. To the right of the Folder Browser is the **File List**. It provides a detailed view of all supported files in the folder currently selected by the Folder Browser.

3. Below the File List is the **Encode/Decode Section**. This is where settings pertaining to encoding and decoding can be adjusted, as well as output file name and location options.

4. The **Audition Section** is situated at the bottom of the window. It contains controls for playing back audio files selected for audition in the File List.
5. Occupying the top right hand area of the screen is the **Metadata Editor**. If a supported M4A or MP3 file is selected, any metadata it contains can be examined, edited, and written back to the file. If a WAV or AIFF file is selected, the section is used to edit the metadata that will be written when this file is encoded.

7.1 **Folder Browser**

The leftmost section of the Codec Toolbox Manager presents controls for browsing directories.

From here, individual folders can be selected so that their contents can be loaded into the File List. This section describes the different components that make up the Folder Browser section, including details about some common operations you may need to perform.

To help minimise the size of the Manager, the Folder Browser can be hidden and revealed at any time using the hide/reveal button. This button is located on the top right hand side of the section.

7.1.1 **Setting up a Root Folder**

The Root of the Folder Browser is the folder whose path is displayed at the top of the section. All folders contained in the current Root Folder will appear in the Folder Browser. The Root Folder can be selected in one of five ways:

- Select one of the provided default Root Folders from the drop-down list.
- Browse to a new Root Folder via the folder browse button.
- **Right-click** a folder in the Folder Browser and choose ‘Make Root’.
- Drag and drop a folder onto the section. This folder will become the Root.
- Drag and drop a supported audio file onto the section. This file’s parent folder will become the Root.
7.1.2 Finding Audio Files on your System

The Folder Browser includes a feature designed to help locate audio files on your system. Thus, using the built-in Folder Browser may be more efficient than browsing to directories using the operating-system pop-up browser (accessible via the Browse button at the top of the section). This feature works as follows:

- Folders that contain audio files in any of their sub-folders are displayed with a full-colour blue folder icon.
- Folders that contain audio files in their top-level are displayed in full colour along with the number of audio files that they contain.
- Folders that do not contain audio files in any of their sub-folders are displayed with an empty folder icon.

It is important to note that this feature can be CPU intensive. If the root of the Folder Browser contains a significant number of sub-folders, or is located on a slow external storage device, the background scanning of all folders in the Folder Browser’s tree can be correspondingly slow.

If this becomes a hindrance on your system, the background scan can be disabled in the Settings tab, by deselecting the option Folder Browser: Background scan for audio files". With this feature disabled, all folders will be displayed with a full-colour folder icon, and the number of files in each folder will not be displayed.
7.1.3 Supported File Types

The Manager supports the following file types:

- M4A AAC (including AAC-LC, HE-AAC and MPEG Surround)
- MP3
- WAVE (Uncompressed)
- AIFF (Uncompressed)

Unsupported file types will not be recognised as audio files by the background scan process, nor will they be displayed in the File List (see next section).

Click on a folder in the Browser to display the supported audio files it contains in the File List section.

7.2 File List

This section displays all compatible audio files that are in the top-level of the folder currently selected in the Folder Browser. Incompatible file types are not displayed. The section provides the following information:

- File Name
- File Type
- File Size (in MB)
- Presence of Metadata (for MP3 and M4A files). If a file contains metadata supported by the Manager, an Arrow Button is displayed in the rightmost column of the list.
  - Hover the mouse cursor over one of these buttons to preview the file’s metadata.
  - Single-click to import the file’s metadata into the Metadata Editor.

To select a file to audition, single-click in the playback column to the left of the file name. The file will begin to play immediately, and clicking in the same column again will pause playback.

To select a file for processing, single-click on the file. This is denoted by an orange highlight.
7.2.1 Selecting Multiple Files – Batch Processing

It is possible to select multiple files in the File List, for encoding/decoding to the same output format. To select multiple files:

- Command+click (Mac) or control+click (Windows) to add a file to the current selection.
- Shift+click to select a range of files.
- Press command+A (Mac) or control+A (Windows) to select all.

It is important to remember that the available bit rate list for each codec changes depending on the source sample rate and channel configuration.

Because of this, and the modality of the Codec Selector and decoder output format selector, there are necessary restrictions on the types of files that can be selected simultaneously.

- Encoding and Decoding cannot coexist in a single batch operation
- When selecting uncompressed WAV or AIFF files for encoding, all files in the selection must have:
  - the same sample rate
  - the same channel configuration

In order to enforce these restrictions, when keyboard focus is set to the File List, and a multi-select modifier (command (Mac), control (Windows), or shift) is held, incompatible files in the list are greyed out and cannot be selected.

Drag an audio file onto the File List to set the file’s parent folder as the root of the Folder Browser, select the file for processing. Please note that this will only work for audio file types that are supported by the Manager, such as WAVE, AIFF, MP3 and AAC encoded M4A.

Right-click a file in the list and select ‘Reveal in Finder’ (Mac) or ‘Reveal in Explorer’ (Windows) to open the operating system file browser at that file’s location.
This section occupies the lower-middle area of the Toolbox Manager window. It provides information about the selected file, Encoder settings or Decoder output format settings, output file name editing, and output file path selection.

1. Name of the selected file – If the file is not in the currently selected folder, click the Input File Name to reselect its parent folder in the Folder Browser.

2. File Info
   - Duration
   - Number of audio channels
   - Sample rate
   - Bit depth or Bit rate (depending on selected file type)
   - Input file size
   - Output file size. This is dependent on the current Codec settings.

3. Encode/Decode Selection. If a WAV or AIFF file is selected, this panel will display the Codec Settings Selector. This selector is identical to the one found in the Toolbox plug-in. If a supported M4A or MP3 file is selected, this panel will display a WAV/AIFF toggle to choose the decoded file’s output format.

4. Output File Name – Click to edit the name of the output destination file.

5. Output File Path – By selecting the ‘save to input folder’ option, the location of the source file will be used. By unselecting the ‘save to input folder’ option, a further output path panel is displayed. Click the browser icon on the right to choose your output location.
6. **Encode/Decode Button.** If a WAV or AIFF file is selected, this button will be in ENCODE mode. If a supported M4A or MP3 file is selected, the button will be in DECODE mode. While an encode/decode operation is in progress, the button displays an internal progress bar; clicking the button will CANCEL the current encode/decode operation.

### 7.3.1 Encoding

The **CLIP SAFE** button enables a level-correction feature. This works by decoding the newly compressed file, calculating the maximum sample value in the PCM stream and, if necessary, applying the precise amount of gain trim required when the source file is encoded a second time. This guarantees that the final encoded file, once decoded for playback, will not contain any illegal sample values (or overloads). However, if your audio files have been correctly prepared using the Toolbox Plug-in, using Clip Safe should not be necessary.

When you are ready, press the ENCODE button to encode the file and write it to the target location. A status bar will indicate progress during the encoding process. During encoding, it is possible to press the encode button again if you wish to cancel the process. The current status of the process is also indicated in the STATUS window in the bottom right corner.

All edited tags in the Metadata Editor will be written to the encoded output file.

Some computers may indicate a ‘spinning progress cursor’ during the first few seconds of the encoding process. This is quite normal and does not indicate that the application is not responding.

### 7.3.2 Dither

As in the Toolbox Plug-in, the encoders in the Manager will accept an input signal with greater than 16-bit precision, but in such cases the signal will be dithered and truncated to 16-bits prior to the encoding process.

The manager will not introduce dither at the 16-bit level if the input signal has been already been truncated to 16-bits.
7.3.3 Decoding

The Manager can also decode compressed files for import into a host sequencer (or for general purpose decoding). To decode supported M4A or MP3 files, simply select them in the File List, choose WAV or AIFF, and press the DECODE button.

The Output file path panel also allows you to select where you would like the decoded files to be saved to. By selecting the ‘save to input folder’ option, the source folder will be used. By unselecting the ‘save to input folder’ option, a further output folder field is displayed. Click the browser icon on the right to choose your output folder. Click the output filename to rename the destination file.

7.4 Auditioning Audio Files in the Manager

It is possible to listen to any supported audio file using the Toolbox Manager. Click in the audition column to the left of the file name to load a file into the internal playback section. The file will begin to play immediately. Press the spacebar to pause or resume playback. Use the SETTINGS tab to select your audio output device.

This section provides familiar controls – play/pause, volume and play position seeking. Clicking the speaker icon toggles playback mute. The loop button has three modes:

- No looping
- Loop the currently selected file
- Loop all files in the list in sequence. At the end of playback, the next file in the list is selected, and playback begins. Playback will stop at the end of the last file in the list.

The audio output device used by the Manager can be selected from the SETTINGS tab. If a device is connected to the system while the Manager is open, press the rescan button, and the device will become available via the drop-down list.

Note – it is recommended to configure the plug-in’s host DAW to release the audio driver in the background.
7.5  The Metadata Editor

This section describes the features and operation of the Manager’s Metadata Editor, which occupies the right hand side of the user interface.

7.5.1  Overview

Many audio file formats, such as MP3 and M4A, can optionally contain information about a file. This information is commonly known as METADATA. Typical metadata tags in an audio file could include the Title, Artist, Track Number, and Cover Art. The Title tag of a file is distinct from its file name.

Adding metadata to your audio files makes them easier to identify, and allows other applications to categorise and sort them based on your needs. It is also a sign of quality, offering the information that you would normally find on an album CD case or vinyl sleeve.

The Toolbox Manager provides the ability to add the most commonly used tags to your files during the encoding process, and edit tags in existing MP3 and supported M4A files.
When an MP3 or M4A file is selected in the File List (single-click), any supported metadata it contains is imported into the Metadata Editor. Once a tag has been edited, its row is displayed with a light grey highlight. Metadata in files that are not currently selected can be previewed in context, by hovering the mouse cursor over the Metadata Import button in the File List.

**Undo/Redo**
Undo or Redo a metadata edit. Importing metadata from other files, and from presets, sets an undo point so can also be undone/redone.

**Clear**
Remove all the existing metadata in the table.

**Revert**
Undo all changes and restore the section to the metadata held within the selected file.

**Write to File**
Write the contents of the Metadata Editor to the selected file.
Pressing the ‘Write to File’ button alters the selected file, and cannot be undone.
7.5.2 Supported Tag Formats

The Manager provides the ability to add and modify metadata in files with an ‘mp3’ extension, and some files with an ‘m4a’ extension. Adding and modifying metadata in m4a files compressed with the Apple Lossless Audio Codec (ALAC) is not supported. The Status panel will display an error message if one of these files is selected. The adding and editing of metadata in ‘wav’ and ‘aiff’ formats is not supported.

The metadata contained within mp3 and m4a files is distinct from the audio data, and is stored in a separate format of its own. The Manager supports the handling of 2 different metadata formats:

ID3v2

This format is typically for use in mp3 files. The ID3v2 standard is now in its 4th revision, but many players, including Windows 7 utilities, only support up to ID3v2.3. The Manager can read and write metadata in all versions: 2.2, 2.3 and 2.4.

iTunes Metadata

This is the standard developed and popularised by Apple, and found in m4a files.

These two tag formats are widely supported, and can be parsed by most commercially available music player applications.

Note that m4a files can contain metadata in the ID3 format, but mp3 files cannot contain metadata in the iTunes format.
### 7.5.3 Supported Tags

The list of available tags varies between MP3 and AAC files due to the different metadata specifications of each format. Tags that are not supported for the selected file type will be disabled, and appear in italics.

<table>
<thead>
<tr>
<th>TAG</th>
<th>MP3 Files (ID3)</th>
<th>AAC Files (Apple iTunes)</th>
<th>Comments</th>
</tr>
</thead>
<tbody>
<tr>
<td>Title</td>
<td>✓</td>
<td>✓</td>
<td></td>
</tr>
<tr>
<td>Artist</td>
<td>✓</td>
<td>✓</td>
<td></td>
</tr>
<tr>
<td>Album Artist</td>
<td>✓</td>
<td>✓</td>
<td></td>
</tr>
<tr>
<td>Album</td>
<td>✓</td>
<td>✓</td>
<td></td>
</tr>
<tr>
<td>Composer</td>
<td>✓</td>
<td>✓</td>
<td></td>
</tr>
<tr>
<td>Genre</td>
<td>✓</td>
<td>✓</td>
<td>Supports standard and non-standard genres</td>
</tr>
<tr>
<td>Comments</td>
<td>✓</td>
<td>✓</td>
<td>Multi-line text</td>
</tr>
<tr>
<td>Grouping</td>
<td>✓</td>
<td>✓</td>
<td></td>
</tr>
<tr>
<td>Lyrics</td>
<td>✓</td>
<td>✓</td>
<td>Multi-line text</td>
</tr>
<tr>
<td>Webpage</td>
<td>✓</td>
<td>not supported</td>
<td></td>
</tr>
<tr>
<td>Publisher</td>
<td>✓</td>
<td>not supported</td>
<td></td>
</tr>
<tr>
<td>Original Artist</td>
<td>✓</td>
<td>not supported</td>
<td></td>
</tr>
<tr>
<td>Remixer</td>
<td>✓</td>
<td>not supported</td>
<td></td>
</tr>
<tr>
<td>Compilation</td>
<td>✓</td>
<td>✓</td>
<td>Toggle</td>
</tr>
<tr>
<td>BPM</td>
<td>✓</td>
<td>✓</td>
<td>Numeric, 3 digits</td>
</tr>
<tr>
<td>Track Number</td>
<td>✓</td>
<td>✓</td>
<td>Numeric, in the format x/y (e.g. 1/12)</td>
</tr>
<tr>
<td>ISRC</td>
<td>✓</td>
<td>not supported</td>
<td>International Standard Recording Code</td>
</tr>
<tr>
<td>Gapless</td>
<td>not supported</td>
<td>✓</td>
<td>Toggle</td>
</tr>
<tr>
<td>Date</td>
<td>✓</td>
<td>✓</td>
<td>Numeric, 4 digits (year)</td>
</tr>
<tr>
<td>Cover Art</td>
<td>✓</td>
<td>✓</td>
<td>Image (.png, .jpg, .jpeg)</td>
</tr>
</tbody>
</table>
7.5.4 Editing Text Tags

In order to edit a text tag, click on the tag row in the Metadata Editor. This opens an inline text field; type your changes into this field.

There are two kinds of text tag:

- Single line
- Multi-line (Lyrics and Comments)

To accept changes for a single line tag, press the **ENTER** key, or press **TAB** or **shift+TAB** to accept and move to the next or previous tag.

In multi-line tags, pressing **ENTER** will insert a new line into the tag text field.

To accept changes for a multi-line tag, press **command+ENTER** (Mac) or **control+ENTER** (Windows). Alternately, press **TAB** or **shift+TAB** to accept and move to the next or previous tag.

The **ENTER** key has one more behaviour in the Metadata Editor. If there are no tags currently being edited, press **ENTER** to move back to and edit the most recently touched tag. This can be useful if you have just accepted an incorrect edit. Rather than clicking on the tag again, or tabbing back/forwards, simply press the **ENTER** key.

While editing a text tag, press **ESCAPE** to discard your edits.

Some fields have format restrictions and will only accept a certain type and/or number of characters. See the Supported Tags table for details.

7.5.5 Adding Cover Art

To add Cover Art to your encoded files, first select a file by **single-clicking** it in the File List. There are two ways to load an image file into the Metadata Editor:

1. Click on the ‘Image’ tag field in the lower right corner of the Metadata Editor, and browse to an image file.
2. Drag an image file onto the Manager.

To remove an image file from the Metadata Editor, hover your mouse cursor over the ‘Image’ tag, and click the red ‘X’ button that appears.

If this image file exists in the Cover Art tag of the selected audio file, press the ‘Write to File’ button to overwrite the previous Cover Art tag with empty contents.
7.5.6 Importing Tags from Another File

Any audio files that contain metadata are listed in the File List with a small arrow button in the right-most column.

Hover your mouse cursor over a button to PREVIEW the metadata from that file in the Metadata Editor. Click a button to IMPORT the metadata from that file into the Metadata Editor.

Only non-empty tags will be imported, so existing edited tags cannot be overwritten with an empty tag from another file.

Importing metadata from another file in this way is compatible with the Undo/Redo mechanism, and the Revert feature.
7.5.7 Saving and Loading Metadata Presets

Metadata tags can be saved into presets for later use. To display the metadata presets manager, click the PRESETS tab in the lower right hand area of the Manager. In this presets manager, a list of saved presets is displayed.

Hover the mouse cursor over a preset name to preview the saved tags in the Metadata Editor. Click on a preset to load the saved tags into the Metadata Editor as changes for the selected file.

Since the presets are imported as changes to the existing content in the Metadata Editor, if a preset does not contain anything different, nothing will be previewed or imported.

To undo metadata tag changes after importing a preset, press the Undo button, or use the keyboard shortcut command+Z (Mac) or control+Z (Windows).
To save a new metadata preset, press **command+S** (Mac) or **control+S** (Windows), or press the PLUS button that appears on mouse over in the upper-left corner of the Preset Tab. This will open a text field where the preset name can be typed. Press **Enter** to save the preset, or **Escape** to dismiss the text field, and cancel the saving process.

The list of presets is ordered alphabetically, and is immediately re-ordered when a new preset is saved. Empty presets cannot be saved, so the plus button is not displayed when the Metadata Editor is empty.

To delete or remove a preset, hover your mouse cursor over the preset name, and press the red X button that appears at the right-hand side of the row. A pop-up window will ask for confirmation before deleting the preset.
7.5.8 The Metadata Lock Mechanism

It is important to remember that the changes performed in the Metadata Editor are not written to the selected file until the ‘Write to File’ button is pressed.

If a new file is selected for processing when the metadata of the previously selected file has been edited (but not written-to-file), those changes will be lost. A pop-up window will appear in this case, to warn the user and allow the operation to be cancelled.

Each field in the Metadata Editor can be protected to ensure that it isn't modified during editing. Clicking at the beginning of that field’s row, in the padlock column, will toggle a small padlock on that row. Click and drag in the padlock column to lock or unlock multiple tags.

Locked rows will retain the current tag content when clearing, reverting, or importing metadata from other files or presets. Additionally, when changing the selected-for-processing file, that row will keep its state. The cover art can also be locked, by clicking on the padlock icon in the lower right corner.

When a non-encoded file is selected for processing (double-clicked), metadata info can also be added in the Metadata Editor. The main difference with encoded files is that the ‘Write to File’ button is disabled, since this metadata cannot be written to an uncompressed PCM file. Instead, the metadata in the section will be written to the new encoded file.


8 Keyboard Shortcuts

<table>
<thead>
<tr>
<th>Global Context</th>
<th>Mac</th>
<th>Windows</th>
</tr>
</thead>
<tbody>
<tr>
<td>Process File(s) (encode or decode)</td>
<td>Command+P</td>
<td>Control+P</td>
</tr>
<tr>
<td>Write metadata to selected file</td>
<td>Command+W</td>
<td>Control+w</td>
</tr>
<tr>
<td>Undo metadata edit</td>
<td>Command+Z</td>
<td>Control+Z</td>
</tr>
<tr>
<td>Redo metadata edit</td>
<td>Shift+cmd+Z</td>
<td>Shift+ctrl+Z</td>
</tr>
<tr>
<td>Save metadata preset</td>
<td>Command+S</td>
<td>Control+S</td>
</tr>
<tr>
<td>Toggle playback</td>
<td>Spacebar</td>
<td>Spacebar</td>
</tr>
</tbody>
</table>

The grey highlight that appears around the Folder Browser, File List or Metadata Editor indicates the section that is currently the keyboard focus target for the remaining key commands. Focus can be changed by pressing TAB or SHIFT+TAB, and also follows the most recently clicked section.
9 Workflow Example

The first thing to do after launching the Codec Toolbox Manager, is load your uncompressed WAV/AIFF audio files. This can be done in one of three ways:

1. Browse to your folder using the built-in Folder Browser
2. Drag a folder from your operating system’s file browser onto the Manager
3. Drag one of your audio files onto the Manager

Now, you should see your audio files listed in the Manager’s File List. **Single-click a file to select it for processing. Once the file is selected, the Codec settings panel will update to only provide access to the codec settings which are compatible with the sample rate and channel configuration of the file.

- Using the Metadata Editor, set up all of the tags that you wish to be written to the output compressed file.
- Click the Encode button. The output file will be written to the location indicated in the Output file path field, and all non-empty metadata tags will be written to the resulting file.

However, you may wish to encode all of your files at once, and write the common metadata tags at the same time. To do this:

1. Select the files that you wish to encode using command/control+click or shift+click.
2. In the Metadata Editor, fill in the tags that will be common to all selected files (for example, Artist, Album, Genre, Cover Art).
3. Choose your desired Codec Settings
4. Point to the desired output folder
5. Press **ENCODE**

When the Manager has finished encoding each of the selected files and writing the common metadata tags, navigate to your chosen output folder in the Folder Browser.

1. Select the first file in the File List, and edit the metadata tags specific to that file.
2. Press command+W (Mac) or control+W (Windows) to write your edits to the file.
3. Press command+] (Mac) or control+] (Windows) to select the next file in the File List.
4. Continue until the metadata for all files is complete.
10 Specifications

10.1 Metadata Text Encoding

Text encoding formats are a way of describing the internal byte representation of individual characters. The ID3 and iTunes metadata formats strictly define which text encodings are supported, so that any text tags (e.g. Title, Artist) can be read and written by different applications. This section describes how the Manager handles metadata text encoding for the different metadata formats supported by the Manager.

There are 3 different types of text encoding defined in the ID3 and iTunes metadata standards:

**Latin–1**
This is an extension of ASCII. It is very space efficient, storing only 1 byte per character; however it has very limited character support.

**UTF–16**
Was a standard introduced in 1990 to address universal character support. It is far more comprehensive than latin–1, but is also significantly less space efficient.

**UTF–8**
This is a later revision of UTF–16. It supports the same number of characters as UTF–16, but has the benefit of being far more space efficient. UTF–8 is generally accepted as the de facto text encoding standard, and is what the Manager uses to store metadata tags internally.

The Manager always writes iTunes metadata text tags using UTF–8 encoding. The handling of ID3 text encoding, however, is more subtle, as it supports Latin–1, UTF–16 and UTF–8 text encoding formats. It is important to note that UTF–8 support is only available in ID3v2.4.

To ensure maximum space efficiency, all text tags less than or equal to ID3v2.3 are written with Latin–1 encoding. If the characters in the tag are not in the Latin–1 subset, the tag is written as UTF–16. Files with ID3v2.4 text tags will always have the text encoding fixed to UTF–8.
10.2 Sample Rate and Bit Depth

The Codec Toolbox plug-in and Manager support sample rates of 32, 44.1 and 48kHz.

The input to the encoders will be dithered and truncated to 16 bits if necessary. If the signal has no activity below the 16th bit, the input to the encoders will not be dithered.

10.3 Codec Sample Rate vs. Bit Rate

10.3.1 Sample Rate: 32 kHz

**MP3 – Mono**

<table>
<thead>
<tr>
<th>CBR Bit rate (kbps)</th>
<th>Codec Sample rate (kHz)</th>
<th>VBR Bit rate (kbps)</th>
<th>Codec Sample rate (kHz)</th>
</tr>
</thead>
<tbody>
<tr>
<td>40</td>
<td>44.1</td>
<td></td>
<td></td>
</tr>
<tr>
<td>48</td>
<td>44.1</td>
<td></td>
<td></td>
</tr>
<tr>
<td>56</td>
<td>32</td>
<td></td>
<td></td>
</tr>
<tr>
<td>64</td>
<td>44.1</td>
<td></td>
<td></td>
</tr>
<tr>
<td>80</td>
<td>44.1</td>
<td>84</td>
<td>44.1</td>
</tr>
<tr>
<td>96</td>
<td>44.1</td>
<td>88</td>
<td>44.1</td>
</tr>
<tr>
<td>112</td>
<td>44.1</td>
<td>112</td>
<td>44.1</td>
</tr>
<tr>
<td>128</td>
<td>44.1</td>
<td>128</td>
<td>44.1</td>
</tr>
<tr>
<td>160</td>
<td>44.1</td>
<td></td>
<td></td>
</tr>
<tr>
<td>192</td>
<td>44.1</td>
<td></td>
<td></td>
</tr>
<tr>
<td>224</td>
<td>44.1</td>
<td></td>
<td></td>
</tr>
<tr>
<td>256</td>
<td>44.1</td>
<td></td>
<td></td>
</tr>
<tr>
<td>320</td>
<td>44.1</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
MP3 – Stereo

<table>
<thead>
<tr>
<th>CBR Bit rate (kbps)</th>
<th>Codec Sample rate (kHz)</th>
<th>VBR Bit rate (kbps)</th>
<th>Codec Sample rate (kHz)</th>
</tr>
</thead>
<tbody>
<tr>
<td>80</td>
<td>44.1</td>
<td></td>
<td></td>
</tr>
<tr>
<td>112</td>
<td>32</td>
<td></td>
<td></td>
</tr>
<tr>
<td>128</td>
<td>44.1</td>
<td>130</td>
<td>44.1</td>
</tr>
<tr>
<td>160</td>
<td>44.1</td>
<td>150</td>
<td>44.1</td>
</tr>
<tr>
<td>192</td>
<td>44.1</td>
<td>190</td>
<td>44.1</td>
</tr>
<tr>
<td>224</td>
<td>44.1</td>
<td>220</td>
<td>44.1</td>
</tr>
<tr>
<td>256</td>
<td>44.1</td>
<td></td>
<td></td>
</tr>
<tr>
<td>320</td>
<td>44.1</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

AAC LC – Mono
At all bit rates, the codec operates at 32 kHz – there is no resampling.

AAC LC – Stereo
At all bit rates, the codec operates at 32 kHz – there is no resampling.

AAC LC – 5.1
At all bit rates, the codec operates at 32 kHz – there is no resampling.

HE AAC – Mono
At all bit rates, the codec operates at 32 kHz – there is no resampling.

HE AAC – Stereo
At all bit rates, the codec operates at 32 kHz – there is no resampling.

HE AAC – 5.1
At all bit rates, the codec operates at 32 kHz – there is no resampling.
HE AAC V2 – Stereo

At all bit rates, the codec operates at 32 kHz – there is no resampling.

iTunes+  (Stereo only)

<table>
<thead>
<tr>
<th>Bit rate (kbps)</th>
<th>Codec Sample rate (kHz)</th>
<th>Bit rate (kbps)</th>
<th>Codec Sample rate (kHz)</th>
</tr>
</thead>
<tbody>
<tr>
<td>256</td>
<td></td>
<td>256</td>
<td>44.1</td>
</tr>
</tbody>
</table>

10.3.2   Sample Rate: 44.1 kHz

MP3 – Mono

<table>
<thead>
<tr>
<th>Bit rate (kbps)</th>
<th>Codec Sample rate (kHz)</th>
<th>Bit rate (kbps)</th>
<th>Codec Sample rate (kHz)</th>
</tr>
</thead>
<tbody>
<tr>
<td>56</td>
<td>32</td>
<td></td>
<td></td>
</tr>
<tr>
<td>64</td>
<td>44.1</td>
<td>78</td>
<td>44.1</td>
</tr>
<tr>
<td>80</td>
<td>44.1</td>
<td>84</td>
<td>44.1</td>
</tr>
<tr>
<td>96</td>
<td>44.1</td>
<td>88</td>
<td>44.1</td>
</tr>
<tr>
<td>112</td>
<td>44.1</td>
<td>112</td>
<td>44.1</td>
</tr>
<tr>
<td>128</td>
<td>44.1</td>
<td>128</td>
<td>44.1</td>
</tr>
<tr>
<td>160</td>
<td>44.1</td>
<td></td>
<td></td>
</tr>
<tr>
<td>192</td>
<td>44.1</td>
<td></td>
<td></td>
</tr>
<tr>
<td>224</td>
<td>44.1</td>
<td></td>
<td></td>
</tr>
<tr>
<td>256</td>
<td>44.1</td>
<td></td>
<td></td>
</tr>
<tr>
<td>320</td>
<td>44.1</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
MP3 – Stereo

<table>
<thead>
<tr>
<th>CBR Bit rate (kbps)</th>
<th>Codec Sample rate (kHz)</th>
<th>VBR Bit rate (kbps)</th>
<th>Codec Sample rate (kHz)</th>
</tr>
</thead>
<tbody>
<tr>
<td>96</td>
<td>32</td>
<td>96</td>
<td>44.1</td>
</tr>
<tr>
<td>112</td>
<td>44.1</td>
<td>130</td>
<td>44.1</td>
</tr>
<tr>
<td>128</td>
<td>44.1</td>
<td>150</td>
<td>44.1</td>
</tr>
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<td>160</td>
<td>44.1</td>
<td>190</td>
<td>44.1</td>
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<tr>
<td>192</td>
<td>44.1</td>
<td>220</td>
<td>44.1</td>
</tr>
<tr>
<td>224</td>
<td>44.1</td>
<td></td>
<td></td>
</tr>
<tr>
<td>256</td>
<td>44.1</td>
<td></td>
<td></td>
</tr>
<tr>
<td>320</td>
<td>44.1</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

AAC LC – Mono
At all bit rates, the codec operates at 44.1 kHz – there is no resampling.

AAC LC – Stereo
At all bit rates, the codec operates at 44.1 kHz – there is no resampling.

AAC LC – 5.1
At all bit rates, the codec operates at 44.1 kHz – there is no resampling.

HE AAC – Mono
At all bit rates, the codec operates at 44.1 kHz – there is no resampling.

HE AAC – Stereo
At all bit rates, the codec operates at 44.1 kHz – there is no resampling.

HE AAC – 5.1
At all bit rates, the codec operates at 44.1 kHz – there is no resampling.

HE AAC V2 – Stereo
At all bit rates, the codec operates at 44.1 kHz – there is no resampling.
HD AAC – Mono

Audition not available. Plug-in output = input.

HD AAC – Stereo

<table>
<thead>
<tr>
<th>CBR Bit rate (kbps)</th>
<th>Codec Sample rate (kHz)</th>
<th>VBR Bit rate (kbps)</th>
<th>Codec Sample rate (kHz)</th>
</tr>
</thead>
<tbody>
<tr>
<td>96</td>
<td>32 *</td>
<td></td>
<td></td>
</tr>
<tr>
<td>112</td>
<td>44.1 *</td>
<td></td>
<td></td>
</tr>
<tr>
<td>128</td>
<td>44.1 *</td>
<td></td>
<td></td>
</tr>
<tr>
<td>160</td>
<td>44.1 *</td>
<td></td>
<td></td>
</tr>
<tr>
<td>192</td>
<td>44.1 *</td>
<td></td>
<td></td>
</tr>
<tr>
<td>224</td>
<td>44.1 *</td>
<td></td>
<td></td>
</tr>
<tr>
<td>256</td>
<td>48 *</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

* Audition not available. Plug-in output = input

HD AAC – 5.1

Audition not available. Plug-in output = input.

AAC LC MPS – 5.1

At all bit rates, the codec operates at 44.1 kHz – there is no resampling.

HE AAC MPS – 5.1

At all bit rates, the codec operates at 44.1 kHz – there is no resampling.

iTunes+ (Stereo only)

The codec operates at 44.1 kHz – there is no resampling.
### 10.3.3 Sample Rate: 48 kHz

#### MP3 – Mono

<table>
<thead>
<tr>
<th>CBR Bit rate (kbps)</th>
<th>Codec Sample rate (kHz)</th>
<th>VBR Bit rate (kbps)</th>
<th>Codec Sample rate (kHz)</th>
</tr>
</thead>
<tbody>
<tr>
<td>56</td>
<td>32</td>
<td></td>
<td></td>
</tr>
<tr>
<td>64</td>
<td>32</td>
<td></td>
<td></td>
</tr>
<tr>
<td>80</td>
<td>32</td>
<td>84</td>
<td>32</td>
</tr>
<tr>
<td>96</td>
<td>32</td>
<td>88</td>
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</tr>
<tr>
<td>112</td>
<td>32</td>
<td>112</td>
<td>32</td>
</tr>
<tr>
<td>128</td>
<td>32</td>
<td>128</td>
<td>32</td>
</tr>
<tr>
<td>160</td>
<td>32</td>
<td></td>
<td></td>
</tr>
<tr>
<td>192</td>
<td>32</td>
<td></td>
<td></td>
</tr>
<tr>
<td>224</td>
<td>32</td>
<td></td>
<td></td>
</tr>
<tr>
<td>256</td>
<td>32</td>
<td></td>
<td></td>
</tr>
<tr>
<td>320</td>
<td>32</td>
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</tr>
</tbody>
</table>

#### MP3 – Stereo

<table>
<thead>
<tr>
<th>CBR Bit rate (kbps)</th>
<th>Codec Sample rate (kHz)</th>
<th>VBR Bit rate (kbps)</th>
<th>Codec Sample rate (kHz)</th>
</tr>
</thead>
<tbody>
<tr>
<td>112</td>
<td>44.1</td>
<td></td>
<td></td>
</tr>
<tr>
<td>128</td>
<td>44.1</td>
<td>130</td>
<td>44.1</td>
</tr>
<tr>
<td>160</td>
<td>44.1</td>
<td>150</td>
<td>44.1</td>
</tr>
<tr>
<td>192</td>
<td>44.1</td>
<td>190</td>
<td>44.1</td>
</tr>
<tr>
<td>224</td>
<td>44.1</td>
<td>220</td>
<td>44.1</td>
</tr>
<tr>
<td>256</td>
<td>44.1</td>
<td></td>
<td></td>
</tr>
<tr>
<td>320</td>
<td>48</td>
<td></td>
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</tr>
</tbody>
</table>
AAC LC – Mono
At all bit rates, the codec operates at 48 kHz – there is no resampling.

AAC LC – Stereo
At all bit rates, the codec operates at 48 kHz – there is no resampling.

AAC LC – 5.1
At all bit rates, the codec operates at 48 kHz – there is no resampling.

HE AAC – Mono
At all bit rates, the codec operates at 48 kHz – there is no resampling.

HE AAC – Stereo
At all bit rates, the codec operates at 48 kHz – there is no resampling.

HE AAC – 5.1
At all bit rates, the codec operates at 48 kHz – there is no resampling.

HE AAC V2 – Stereo
At all bit rates, the codec operates at 48 kHz – there is no resampling.

HD AAC – Mono
Audition not available. Plug-in output = input

HD AAC – Stereo
Audition not available. Plug-in output = input

HD AAC – 5.1
Audition not available. Plug-in output = input

AAC LC MPS – 5.1
At all bit rates, the codec operates at 48 kHz – there is no resampling.

HE AAC MPS – 5.1
At all bit rates, the codec operates at 48 kHz – there is no resampling.
iTunes+ (Stereo only)

<table>
<thead>
<tr>
<th>CBR Bit rate (kbps)</th>
<th>Codec Sample rate (kHz)</th>
<th>VBR Bit rate (kbps)</th>
<th>Codec Sample rate (kHz)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>256</td>
<td>44.1</td>
</tr>
</tbody>
</table>
11 Copyright and Acknowledgements

The Sonnox Fraunhofer Codec Toolbox plug-in and Manager were developed by the design team at Sonnox in collaboration with Fraunhofer IIS.

The Audio Coding software contained in the Sonnox Fraunhofer Codec Toolbox plug-in and Manager was developed and provided by Fraunhofer IIS of Erlangen, Germany.

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- There are currently no royalties collected for AAC commercial content distribution.

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13  **Supported Platforms**

- Avid Pro Tools (RTAS, AAX 64-bit)
- VST hosts (32 and 64-bit)
- AU hosts (32 and 64-bit)
- Mac Intel OSX 10.6 or higher
- Windows XP, 7 and 8 (32 and 64-bit)

14  **System Requirements**

For latest System requirements, please see [www.sonnox.com](http://www.sonnox.com)

**Pro Tools**

- Windows XP/7/8 or Apple Mac OSX 10.6 or higher
- Approved Digidesign/Avid CPU and hardware configuration
- Pro Tools 8 (Native or HD), or higher
- Free iLok account
- Appropriate product licence
- *iLok2 or machine authorisation*

**VST Native**

- Windows XP/7/8 or Apple Mac OSX 10.6 or higher
- VST compliant host application (e.g. Nuendo, Cubase, Ableton Live, Studio One, etc.)
- Free iLok account
- Appropriate product licence
- *iLok2 or machine authorisation*

**Audio Units**

- Approved Apple CPU and OSX 10.6 or higher
- Audio Unit Host application (e.g. Logic, Digital Performer)
- Free iLok account
- Appropriate product licence
- *iLok2 or machine authorisation*