What can Engine do for you?

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Efficiency gains made by analysing 10000 files a month.

Using Engines automation, companies can experience significant efficiencies in their workflow. The software's advanced capabilities allow for precise analysis and detection of the source structure of audio files. For example, it can accurately identify whether an 8-channel MXF file consists of a 5.1 surround sound and a stereo component. This information is invaluable for the Quality Control (QC) team in the downstream process. By automating this task, the software eliminates the need for manual inspection, saving time and reducing the risk of human error. With streamlined source structure detection, the QC team can focus on other critical aspects of their work, ensuring high-quality audio output and a smoother production process overall.

Efficiency gained by automating the analysis of 60 files

Our conditional workflow module offers significant efficiency gains for users. Traditionally, selecting the appropriate workflow for a specific source file has been a manual and time-consuming process. With our conditional workflow module, this burden is alleviated. The module is designed to intelligently analyse the characteristics and properties of the source file, accurately determining the most suitable workflow to be applied. This automation eliminates the need for operators to manually select individual workflows from a potentially extensive bank of options. As a result, valuable time is saved, and human error is minimised. By seamlessly and accurately handling workflow selection, the conditional workflow module enables operators to focus their efforts on other critical tasks, ultimately enhancing productivity and streamlining the entire audio production workflow.

Efficiencies gained in using faster than real time loudness analysis.

Utilising our file-based faster-than-real-time loudness analysis and correction provides significant efficiencies in audio processing. Traditional real-time loudness analysis and correction methods require the audio to be played back in real-time, which can be time-consuming and may slow down the overall production process. In contrast, our file-based solution performs loudness analysis and correction at an accelerated rate, significantly reducing processing time. This means that audio files can be analysed and corrected for loudness discrepancies much faster than real-time playback. As a result, production teams can save valuable time, allowing for quicker turnarounds and increased productivity. Additionally, the file-based nature of our solution enables simultaneous processing of multiple files, further enhancing efficiency and throughput. By leveraging our faster-than-real-time loudness analysis and correction, audio professionals can optimise their workflows, deliver high-quality content more efficiently, and meet tight deadlines with ease.

Philosophy & Focus

- Save money for the Customer
- Easy to use, automatable, and scalable
- Cross Platform – Windows, Mac and Linux
- Software based license supports Virtual Machines
- Audio workflows for MXF, MOV, AIFF, WAV, LXF, GFX, MP4, ATMOS
- Analyse, report, process
- Highest quality algorithms

Loudness

There is a global movement to adapt Program Loudness and True Peak correction. **Engine** allows the creation of file based workflows that deal with all the worldwide standards and can also Measure and Correct a wide range of Channel layouts (4 Stereos, Stereo + 5.1, etc…)

Engine supports up to 64 channels of audio. Optional modules are available to loudness correction of files with Dolby E encoded audio, and for LRA correction.

The Loudness Compliance algorithm is designed to provide correction with minimal change to the creative mix. This is achieved by using global gain for Program Loudness and local attenuation for True Peak, Short Term and Momentary Loudness.

LRA

LRA (Loudness Range) is a parameter defined in the EBU R128 Loudness specification. Some countries or broadcasters specify a maximum LRA. LRA reduction is often required for converting original cinematic mixes for broadcast use.

This option in the **Loudness** module, lets you specify the maximum LRA for the output file to meet the requirement. Compression is applied, changing the overall mix, but making the content more appropriate in a broadcast environment.

Channel Mapping & Mute

Engine's Channel Mapping module allows remapping, replication and muting of selected audio channels.

Engine's workflow configurator lets you quickly and simply configure complex channel mapping tasks as shown in the diagram.

This example workflow is for files with 16 audio tracks. It shows channels 1&2, and channels 3&4 being swapped.

Channels 5 & 6 are muted.

Channels 7 to 12 are rearranged from a SMPTE 5.1 order to a file 5.1 order.

Channels 13 & 14 are duplicated to channel locations 15 & 16, whilst still also remaining on channel 13 & 14.

Dolby E Encode

Dolby E encoding is in regular use by many Broadcasters. The **Dolby E Encode** module has been designed to simplify the complexity and operational difficulties associated with encoding the different program configurations and the management of metadata profiles. Additionally, the module allows positioning of the guard band and channel selection of where the encoded Dolby E is placed.

Dolby E Decode

As with the Dolby E encoding, Dolby E decoding is regularly used by Broadcasters. **Engine's Dolby E Decode** module has been certified and has additional features that allow robust decoding from files that have a few frames of PCM at the beginning of the file and or misaligned guard bands.

Dolby E Guard Band Correction

Dolby have a concept called the Guard Band position, and this refers to the delay from the start of the video frame, to the start of the Dolby E audio frame. For every different video format, Dolby have defined an 'ideal' start position for Dolby E data.

The **Dolby E Guard Band Correction** module measures the position of the Dolby E data within the frame structure and reports this. It can also be used to adjust the position to the ideal position.

In a real time environment, Dolby E audio encoding usually causes a one frame delay. The Guard Band correction module can be configured to move the Dolby E audio either forwards or backwards by up to two frames in case there is an issue caused by frame offsets.

Immersive Mix

Broadcasters who multicast (SD and HD) or have archive material with a stereo mix have a frequent requirement to provide a high-quality Upmix from stereo sources and vice versa. The Upmix can generate any surround format (up to 9.1.6).

The immersive mix can also be used to easily convert any immersive format file to stereo, and the stereo file.

The Downmix algorithm offers the standard Lo/Ro and Lt/Rt options as well as adaptive EQ and independent direct and ambient downmix levels for the surround channels. The Downmix algorithm can be used as a reference to verify the quality of your stereo downmix in both Lo/Ro and Lt/Rt or as a high-quality adaptive downmixer to create a separate stereo transmission feed.

Examine

The Examine module within **Engine** provides easy metadata extraction and reporting. Files with any number of audio channels may be processed and the metadata is reported in PDF or XML formats. An example of the onscreen display is shown below. The full Dolby E AC3 metadata can be exported to XML.

Audio Description

The **Audio Descriptor module** takes Program Audio, Mono Audio descriptor Audio and uses the Control track to creates a new combined Audio Mix.

(Note "File Manipulation" module required for multiple input files).

(Note other I/O options also possible)

File Manipulation - Channel Replace used in Language addition example

The support of multiple languages and Audio Descriptor means that there is often a requirement to insert audio from a WAV file into an MXF or QuickTime MOV file. This process, conventionally, takes a long time. The 'Channel Replace' function, part of the "File Manipulation" module allows you to avoid having to use an expensive edit suite by using a automated file based workflow that is quicker and more efficient.

Estimated time for 1-hour programme

Conventional workflow – 3 hours + edit suite + specialist editor

Using **Engine** – one hour on computer

File Manipulation - Channel Extract – Example workflow

This function, available with the "File Manipulation" module, can be used to automatically extract the M&E or any other track(s) out of a media file, without using an edit suite.

Estimated time for 1-hour programme

Conventional workflow – 3 hours + edit suite + specialist editor

Using **Engine** – one hour on computer

File Manipulation - Add Channels

This function, part of the "File Manipulation" module, allows you to add audio channels to an audio file. For example, you could start with a two channel MXF file and convert it to an eight channel MXF file. Additional channels may be required, for example if you want to upmix a stereo file, and keep the new 5.1 within the same MXF file.

File manipulation – Remove Channels

Using the **Remove Channels** function, part of the "File Manipulation" module, can be used to reduce the number of channels in an audio file. Reducing the number of channels in a file is often a requirement after downmixing, or Dolby E Encoding.

File Wrapping

In a multilingual environment, there is often a single video essence file and multiple language versions of audio files stored as individual WAV files. This module allows the selection of the video file and the desired audio files to be wrapped into an MXF file. Choice of files for wrapping can be specified using a RESTful API.

File Metadata

This module includes the display of language and channel layout metadata from source files, and lets you specify new metadata.

- Display/insert AS11 Language metadata (MXF)
- Display/insert AS11 Channel position metadata (MXF)
- Display/insert DMS-1 Language metadata (MXF)
- Display/insert Channel position metadata (QT/MOV)
- Display/insert timecode (all appropriate formats)

Pitch/Duration Adjustment

When converting a file from USA to European standards, or the other way, it is increasingly common to restamp the frame rate thus preventing ugly conversion artefacts. This leaves the audio with the wrong duration. The Pitch/Duration module can adjust the duration, whilst leaving the pitch unchanged from the source material.

This module uses a very high-quality algorithm and after two successive conversions, the result is usually indistinguishable from the source.

Mono to Stereo

This module is used to convert mono audio content into stereo. A broadcaster may have a requirement to play out archived footage which has mono audio content. In order to be able to play this out on modern-day television, broadcasters should at least have a stereo file which has differing levels on both the Left and Right channels, so the file will not be rejected before playout. Our module offers the chance to convert a mono file to stereo, as well as offering spatial enhancement so the VU meter will accept the file.

Binaural Mix

The binaural mix module can take any immersive format (up to 9.1.6) or stereo file and generate an enhanced spatial stereo mix of this content, for monitoring purposes.

Conditional Workflows

Using a combination of the examine module, channel layout detection and silent channel detection - Engine is able to identify key characteristics of the file as it is loaded in, and these characteristics define which workflow the file will be processed by. This process is able to be automated through the use of watch folders or a MAM system.

The characteristics Engine is able to identify and then use to select an appropriate workflow are: Channel layout, encoded audio, Dolby E program configuration, duration, video resolution, audio bit depth, file type, frame rate.

Dolby Digital/Plus Encode

Dolby Digital/DD+ encoding is in regular use by many Broadcasters. Just like the Dolby E Encoding module; the DD/DD+ Encode module has been designed to simplify the complexity and operational difficulties associated with encoding the different program configurations and the management of metadata profiles. Additionally, the module allows the user to add extra streams to be encoded if and, correct the loudness to the relevant ‘Dialnorm’ level.