AudioTools Server
Module Specifications

AudioTools Server

AudioTools Server is an enterprise–level software platform providing seamless, end–to–end audio processing capabilities for broadcast and post-production facilities.

Use Cases

AudioTools Server provides comprehensive solutions for use-cases such as Loudness Control, Dolby E Automation, Audio Quality Control, Metadata Quality Control, Channel Management, Upmixing, Downmixing, Audio Program Management, Watermarking, and Frame Rate Conversion among others. AudioTools Server may be configured in a simple, single-machine environment or in a multi-node, load-balanced environment.

Server Core

The core module is AudioTools Server. The Server runs as a Windows Service, deployed on a Microsoft Windows PC. Supported versions of Windows include Windows Server 2008 R2, Windows Server 2003 R2, and Windows 7. AudioTools Server performs media handling and audio processing according to the processing modules and parameters specified in the Job or Workflow XML. The Server executes Jobs and Workflows, and communicates Job and Workflow Status to any clients polling the Server.

Jobs

AudioTools Jobs are specified in XML and define units of work. A job defines one or more assets to work on, as well as one or more processing modules to apply to those assets. The Job XML specifies all processing parameters such as Loudness Control parameters or Dolby E Encoding metadata.

Workflows

AudioTools Workflows are also specified in XML and can execute any number of Jobs against an asset. Workflows enable conditional, exceptional, parallel, and iterative processing of audio content, as well as load balancing across multiple nodes. Workflows provide a simple programming syntax for controlling Jobs, parsing Job results, setting Job parameters, and controlling high-level workflows. See the AudioTools WFC module description for more details.

Client Applications

Clients submit Jobs or Workflows to the Server and monitor Job/Workflow Status via a well-documented SOAP interface. AudioTools Server ships with the AudioTools Queue Control and its integrated Job Wizard. Custom Client Applications may be developed in any SOAP-enabled language, and may be deployed on any operating system capable of making SOAP calls.

Third-party applications may communicate with AudioTools Server via a well-defined SOAP interface. Third-party applications may control all AudioTools Server operations or a subset of operations. Third-party applications may be deployed on any platform supporting SOAP calls, including but not limited to Windows, Linux, and MacOS. The optional AudioTools Web Services Interface provides easy third-party integration. .NET sample code is provided.

Watch Folders

Watch Folders are supported for all operations. Watch Folders can act on Assets dropped into the Watch Folder, or Job/Worfklow XML files dropped into the Watch Folder. The latter approach supports Watch Folder operation without requiring full asset copying to the Watch Folder.

Supported Processing Modules

- AudioTools Loudness Measurement
- AudioTools Channel Assignment Detection
- AudioTools SurCode for Dolby E Encoder
- AudioTools SurCode for Dolby Digital Plus Encoder
- AudioTools SurCode for MPEG Audio
- AudioTools Audio Quality Analysis
- AudioTools Audio Extractor for MXF

- AudioTools Loudness Adjustment
- AudioTools Channel Manager
- AudioTools SurCode for Dolby E Decoder
- AudioTools SurCode for Dolby Digital Plus Decoder
- Dolby Media Meter
- AudioTools Dolby E Wave File Analyzer
- AudioTools ReWrap for MXF

- AudioTools Audio Extractor for LXF
- AudioTools Audio Extractor for GXF
- AudioTools Audio Extractor for QuickTime
- AudioTools Audio Extractor for Transport Streams
- AudioTools Program Correlation Check
- Soundfield UPM-1 Upmix
- AudioTools Downmix
- AudioTools Watermarking
- AudioTools Workflow Control

- AudioTools ReWrap for LXF
- AudioTools ReWrap for GXF
- AudioTools ReWrap for QuickTime
- AudioTools ReWrap for Transport Streams
- AudioTools Frame Rate Toolbox
- DTS Neural UpMix
- AudioTools Metadata Verifier
- AudioTools AudioSecter
- AudioTools Web Services

◊ General Specifications:

Number of Programs	1 - 32
Channels per Program	1 - 64
Sample Rate	32 -384 kHz
Bit Depth	8, 16, 24 bit integer, 32 bit float
Bit Packing	interleaved, non-interleaved
File Types	MXF, QuickTime, LXF, GXF, transport stream, BWF, WAV, AIFF, AC-3, E-AC-3
Audio Formats	LPCM, Dolby E, Dolby Digital, Dolby Digital Plus, MPEG Audio, AAC
Transport Stream Audio Types	MPEG1A, MPEG2A, AES3_337M_MPEG1A, AES3_337M_MPEG2A, AAC_LATM, AAC_ADTS, AC3, AES3_337M_AC3, AC3_DDPlus, AES3_337M_DDPlus, AES3_302M
Sample Rate Conversion	arbitrary ratio, fixed ratio
Redithering Types	MBIT, +TPDF
Plug–in Formats	VST (pre–qualified)
Supported Web Services Protocols	SOAP, XML
3rd Party Interoperability	Aveco automation, Avid Interplay MAM, Grass Valley, IBM AREMA, Sony Media Backbone, Viz Ardome & others
3rd Party Modules	Dolby Media Meter, DTS Neural UpMix, Flux:: Sound and Picture development VST plug–ins including Alchemist, Soundfield UPM-1 Upmix, Videomenthe Plotter, other Minnetonka Audio–approved VST plug–ins

AudioTools Server Web Services

The optional Web Services API provides a SOAP/XML interface to AudioTools Server. The Server provides interfaces to Job and Workflow management. The Web Services Interface adheres to the SOAP 1.1 standard and provides multiple services, including:

◊ Job Web Services API:

- Submit/RemoveJob
- GetJobList/XML

- ChangeJobPriority/State
- Set/GetConcurrentJobs

◊ Workflow Web Services API:

- SubmitWorkflow
- RemoveCompletedWorkflow
- GetWorkflowList/XML

- ChangeWorkflowState/Priority
- GetNodeList
- Set/GetNodeState

AudioTools Loudness Measurement

The optional AudioTools Loudness Measurement module provides a comprehensive loudness measurement toolkit supporting current and forthcoming international standards and recommended practices, including VLC or Voice Level Consistency. VLC looks at the distance or amplitude difference between dialog and program by measuring the difference between Voice Level and various

loudness metrics. If this value is reasonably small, and Voice Level remains consistent throughout a program, then the program can be intercut with other programs or commercials without jarring subjective discontinuity. The result of VLC measurements can inform "special case" downstream processing to correct out of bounds material.

◊ The Loudness Measurement module provides the following measurement types:

- Programme Loudness (EBU)
- Program Loudness (ITU)
- Maximum True Peak Level
- Loudness Range
- Maximum Momentary Loudness
- Maximum Short Term Loudness
- Sample Peak Level
- Dialogue Intelligence™

◊ The Loudness Measurement module provides a reference implementation of the following standards:

- ITU-R BS. 1770-1, 2 & 3
- EBU R128
- ATSC RP A/85
- Commercial Advertisement Loudness Mitigation (CALM) Act

Measurement

Measurement results are provided in two ways:

- Measurement summaries are available in the Job Status XML returned to any client. This provides overall, or "infinite", measurement results for the entire program within an asset.

- Periodic measurements along with the measurement summaries are provided in a separate log file on a momentary basis. By default, measurements are reported every second throughout an asset. The reporting period can be customized for more granular data to drive dynamic graphing displays, for example.

Measurements can be performed independently on any number of programs within a single asset via Program Configuration specifications in the Job XML. This means that, in a container file which contains 8 x 2 independent audio programs for example, each program is measured and reported independently in one pass.

Measurement results can be used by the AudioTools Workflow Control module or 3rd party applications to not only program Loudness Adjustment properly, but also conditionally process assets if they are already within specification, or properly categorize assets according to custom loudness specifications.

Accelerated TruPeak™ provides optimized True Peak Measurement. AudioTools Server features Accelerated TruPeak measurement which results in a 50-90% decrease in processing time when compared to a non–accelerated computation.

◊ Module Input:

Parameters	Default	Range
Measurement Type	-	True Peak, Max Momentary Loudness, Max Short Term Loudness, Programme Loudness, Loudness Range, Average ITU, Max Sample Peak Loudness, Dialogue Intelligence™

Additional Parameters

- Location for detailed Results XML, Gating, Program (Channel) Configuration, Program, Program Name, Program ID, Channel, Channel Offset, Channel Name, Channel Name External, Measurement Period   -   -

◊ Module Output:

- Measurement Summary within the Job XML returned via web services
- Detailed Results Log File including momentary measurements and summary

AudioTools Loudness Adjustment

The optional AudioTools Loudness Adjustment module provides a comprehensive loudness adjustment toolkit supporting current and forthcoming international standards and recommended practices. These include Movie Adaptation, which adjusts a wide dynamic range theatrical audio mix for broadcast. A related process, Audio Conforming creates the best possible audio experience for diverse modern platforms such as OTT/web, mobile/handheld and VOD/SVOD.

For Maximum Momentary and Maximum Short Term Loudness adjustment, the AudioTools Loudness Adjustment module employs TruCorrect™ smart loudness processing, which eliminates perceived loudness changes and degraded intelligibility associated with traditional methods of correcting content containing extreme dynamic range.

◊ The Loudness Adjustment module provides a reference implementation of the following standards:

- ITU-R BS. 1770-1, 2 & 3
- EBU R128
- ARIB TR-B32

- ATSC RP A/85
- Commercial Advertisement Loudness Mitigation (CALM) Act
- OP-59

Loudness Adjustment, together with Loudness Measurement, modules can be chained together in Jobs or Workflows to provide any combination of adjustment types. For example, a typical Loudness Adjustment Job consists of Loudness Measurement, Programme Loudness, True Peak Limiting, and a final Loudness Measurement in a single Job.

Adjustment can be performed independently on any number of programs within a single asset via Program Configuration specifications in the Job XML. This means that in a container file which contains 8 x 2 independent audio programs, for example, each program is adjusted independently in one pass.

Dialog Intelligence™ provides the option to perform Programme Loudness Adjustments based on the overall Programme Loudness levels or the Dialog levels throughout a programme. Control of such functionality requires the AudioTools Server Workflow Control module or a third party application implementing logic supporting this feature.

Accelerated TruPeak™ provides optimized True Peak Adjustment, with a 50-90% decrease in processing time relative to standard adjustment methods.

◊ Module Input:

- Parameters specifying Target Loudness for each of the Loudness Adjustment types; For Programme Loudness and Loudness Range Adjustments, the Targets are calculated based on Measurements from a previous processing step. For True Peak, Maximum Momentary, and Maximum Short Term, the Targets can simply be specified without a preceding Measurement step.
- Parameters controlling any limiting functions such as Attack, Release, Sustain, and LookAhead
- Program Configuration specifying the number of programs within an asset and the number of channels within each program
- Optional Channel Offset parameters enabling adjustment of only specific channels within a multi-channel asset

Loudness control is an especially significant use case that employs Minnetonka Audio's proprietary A.I.M. digital signal processing. While AudioTools Loudness Normalization processes are designed to fully preserve the existing audio and only apply a gain change combined with optional peak limiting, there are other use cases that require changing the dynamic range and other more complex parameters of the audio content. Examples include "movie adaptation," adapting a theatrical audio mix for broadcast, and creating the best possible audio experience for modern platforms such as OTT/web, mobile/handheld and VOD/SVOD.

Movie Adaptation as accomplished by the AudioTools A.I.M. algorithm:

- Reduces dynamics and/or Loudness Range
- Preserves transients & artistic intent of original while preventing subjectively compressed sound quality
- Maintains dialog intelligibility
- Provides a compliant, automated process for all program genres & all target requirements

AudioTools A.I.M. processing, in conjunction with the AudioTools WFC workflow control module, employs several measurement techniques, including Dialogue Intelligence™–based detection and measurement, in a fully standards–compatible workflow, preserving dialog intelligibility and providing ITU-R BS.1770/R128 target normalization.

◊ Module Parameters Per Adjustment Type:

Parameter
Programme Loudness Normalization (ITU & EBU)
- Normalization Gain (dB)
True Peak Limiting
- Target True Peak (LU), Attack, Sustain, Release (mS), Maximum Sample Peak (dB)
Loudness Range Adjustment
- Compression Ratio. Pivot Level (LU), Maximum Gain (dB), Attack, Look–ahead (mS), Release (mS)

Maximum Momentary Limiting
- Target Momentary Loudness (LUFS), Attack (mS), Release (mS), Look Ahead (mS)
Maximum Short Term Limiting
- Target Short Term Loudness (LUFS), Attack (mS), Release (mS), Look Ahead (mS)
◊ Module Output:

- Adjusted Audio
- XML log file

AudioTools SurCode for Dolby E Encoder

The optional SurCode for Dolby E Encoder module provides comprehensive Dolby E encoding and supports all Dolby E program configurations, parameters, and metadata. Dolby E parameters and metadata can be specified completely via Job XML, or the Encoder can be instructed to retrieve Dolby E metadata from an upstream Dolby E Decoder, if present. Since certain metadata may change between the Decode and Encode operations, some Encoding parameters can be overridden to reflect changes during processing. These parameters include Dolby E Bit Depth, Dialog Normalization, and Description Text. See the AudioTools Server documentation for a complete description of Dolby E parameters and metadata.

When used in conjunction with SurCode for Dolby E Decoder, Workflow Control, and other processing and analysis modules, AudioTools Server provides a comprehensive, automated system for processing of any Dolby E content. This includes multiple Dolby E streams as well as mixed PCM/Dolby E streams within container formats.

◊ Module Input:

- Source File
- Dolby E Encoding parameters and metadata
- Supports all Dolby E Program Configurations and Encoding parameters
- PrePend Guard Band option
- Optional flag instructing the encoder to retrieve metadata from an upstream Decoder
- Optional override parameters: All
- Optional list of input channels instructing the Encoder which channels to input from a multichannel stream
- Optional list of output channels instructing the Encoder into which channels to place the Dolby E output in a multi-channel stream

◊ Module Output:

- Dolby E Encoded Audio

AudioTools SurCode for Dolby E Decoder

The optional SurCode for Dolby E Decoder module decodes Dolby E to PCM Program Audio, Dolby E Metadata, and Program Metadata.Decoded audio and metadata can be accessed by downstream processors within a single Job or Workflow. This allows the Program Configuration, for example, to be used to program a Downmix operation or split programs to multiple PCM files.

The Decoder also provides Dolby E Detection for determining whether a stream contains Dolby E. This information can be used by Workflow Control or a third party application for conditionally controlling downstream processing.

When used in conjunction with SurCode for Dolby E Encoder, Workflow Control, and other processing and analysis modules, AudioTools Server provides a comprehensive, automated system for processing of any Dolby E content. This includes multiple Dolby E streams, as well as mixed Dolby E/PCM streams, within container formats.

◊ Module Input:

- Optional list of input channels instructing the Decoder which channels to pull input from in a multi-channel stream
- Supports all Dolby E Program Configurations and Metadata

◊ Module Output:

- Decoded PCM Audio

- Dolby E Metadata
- Program Metadata

AudioTools SurCode for Dolby Digital Plus Encoder

The optional AudioTools SurCode for Dolby Digital Plus Encoder module provides comprehensive Dolby Digital and Dolby Digital Plus encoding for up to 5.1 streams. Dolby Digital and Dolby Digital Plus parameters and metadata can be specified completely via Job XML, or the Encoder can be instructed to retrieve Dolby Digital Plus metadata from an upstream Dolby Digital Plus Decoder OR Dolby E Decoder if present. Since certain metadata may change between the Decode and Encode operations, all Encoding parameters can be overridden to reflect changes during processing, including Dialog Normalization. See AudioTools Server documentation for a complete description of Dolby Digital and Dolby Digital Plus parameters and metadata.

When used in conjunction with SurCode for Dolby Digital Plus or Dolby E Decoder, Workflow Control, and other processing and analysis modules, AudioTools Server provides a comprehensive, automated system for processing of any Dolby Digital or Dolby Digital Plus content.

◊ Module Input:
- Dolby Digital or Dolby Digital Plus Encoding parameters and metadata
- Optional flag instructing the encoder to retrieve metadata from upstream Dolby Digital Plus or Dolby E Decoder
- Optional override parameters: All
- Optional list of input channels instructing the Encoder which channels to pull input from in a multi-channel stream.
- Optional list of output channels instructing the Encoder which channels to place the Dolby Digital Plus output in a multi-channel stream

◊ Module Output:
- Dolby Digital or Dolby Digital Plus Encoded Audio

AudioTools SurCode for Dolby Digital Plus Decoder

The optional AudioTools SurCode for Dolby Digital Plus Decoder module decodes Dolby Digital and Dolby Digital Plus to PCM Program Audio and Program Metadata. Decoded audio and metadata can be accessed by downstream processors within a single Job or Workflow. The Decoder also provides Dolby Digital Plus Detection for determining whether a stream contains Dolby Digital or Dolby Digital Plus. This information can be used by Workflow Control or a third party application for conditionally controlling downstream processing. When used in conjunction with SurCode for Dolby Digital Plus Encoder, Workflow Control, and other processing and analysis modules, AudioTools Server provides a comprehensive, automated system for processing of any Dolby Digital or Dolby Digital Plus content.

◊ Module Input:
- Decoding parameters include Compression Mode, Dynamic Range Scaling, optional LFE output, and Output Mode
- Output Mode provides a mechanism for automatic downmixing of decoded audio
- Optional list of input channels instructing the Decoder which channels to pull input from in a multi-channel stream

◊ Module Output:
- Decoded PCM Audio
- Program Metadata

AudioTools SurCode for MPEG Audio

The optional AudioTools SurCode for MPEG Audio module encodes and decodes MPEG-2 Audio Layer II (mp2) and MPEG-2 Audio Layer III (mp3). Decoded audio and metadata can be accessed by downstream processors within a single Job or Workflow. The Decoder also provides MPEG Audio Detection for determining whether a stream contains mp2 or mp3–encoded material. This information can be used by Workflow Control or a third party application for conditionally controlling downstream processing. When used in conjunction with AudioTools WFC and other processing and analysis modules, AudioTools Server provides a comprehensive, automated system for processing of any MPEG Audio content.

◊ Module Input:

- Encoding input is PCM Audio

- Decoding input is MPEG Audio

◊ Module Output:

- Decoded PCM Audio

- Encoded MPEG Audio

- Program Metadata

◊ Partial Module Parameters:

MP2/MP3:

- Layer:

  II & III– supported (Layer I not supported)

- Bit Rates:

  Fixed – 20 - 320, Variable – Good, Better, Best

- Additional

  Quality, Stereo Encoding, Deemphasis, Copyright, Original Bitstream, Protection, Private

AAC:

- Audio Object Type:

  LC, LTP, HEAAC, ER_LC, ER_LTP, ER_SCAL, PS, MP2_LC, MP2_SBR, SBR_DS, ER_SCAL_SBR, ER_SCAL_SBR_PS, MPS

AudioTools SurCode for AAC Audio

The optional AudioTools SurCode for AAC Audio module encodes and decodes Advanced Audio Codec (AAC) and many AAC variants such as Low Complexity AAC (AAC-LC), Low Delay AAC (AAC-LD), and High Efficiency AAC (HE-AAC). Decoded audio and metadata can be accessed by downstream processors within a single Job or Workflow. The Decoder also provides AAC Audio Detection for determining whether or not a stream contains AAC–encoded material. This information can be used by Workflow Control or a third party application for conditionally controlling downstream processing. When used in conjunction with AudioTools WFC and other processing and analysis modules, AudioTools Server provides a comprehensive, automated system for processing of any AAC Audio content.

◊ Module Input:

- Encoding input is PCM Audio

- Decoding input is AAC Audio

◊ Module Output:

- Decoded PCM Audio

- Encoded AAC Audio

- Program Metadata

  All HE-AAC bit rates & modes are supported, including Quality, Channel Modes, Transport Type, Signaling Modes and Metadata Modes.

AudioTools Dolby E WavFileAnalyzer

The optional AudioTools Dolby E WavFileAnalyzer module performs QC analysis and correction of Dolby E files. The results of the processing are formatted as an XML element within the job file, which can be retrieved from the system. Those results include a summary section as well as details classified as Errors, Warnings, and Information.

The module may be optionally configured to correct Dolby E guard band and preamble alignment throughout the Dolby E stream as specified by SMPTE 337M. Source files for non–SMPTE 337M analyses can also be linear PCM, in either WAV or Broadcast WAV format, without Dolby E essence.

You may specify *analyze* or *align*. If you are analyzing, no output file is generated. If you specify alignment, the source asset is analyzed and aligned, and a new output file is generated. If the Operation element is missing, the operation defaults to analyze.

- ◊ QC & Correction Functions:
    - Chunk Analysis for BWF & RF64 files
        - File Size
        - Word Length
        - Sample Rate
        - Track Type
    - SMPTE 337M Analysis includes Bit Depth, Metadata Key, Program Configuration, Frame Rate, Video Format, Word Spacing, No. of SMPTE 337M Words Found, Error Listing, No. of Errors, Warnings, Listing, No. of Warnings
    - Peak Detection
    - Preamble Spacing includes Video Alignment, Alignment Correction, File Duration
- ◊ Module Input:
    - Dolby E Data

Parameters	Default	Range
Operation	analyze	analyze, align

- ◊ Module Output:

	Default	Range
Log File (XML)	-	-
Optionally corrected Dolby E Data	-	-

AudioTools Metadata Verifier

The optional AudioTools Metadata Verifier module decodes and checks the consistency of Dolby E metadata.

- ◊ Decoding of metadata is performed for each frame of the Dolby E stream and for each program within the stream. Metadata is checked for consistency across all frames.
- ◊ The results of the processing are formatted as an XML element within the job file. The results include the Dolby E metadata and program metadata of the first frame of the stream and identify any changes to the metadata that occur in subsequent frames of the stream.
- ◊ Results describe which metadata element has changed and the value before and after the change.
- ◊ A Metadata Verifier Log file can be specified which would contain the complete set of Dolby E metadata for every metadata change within the file.
- ◊ Module Input:
    - Dolby E Data
- ◊ Module Output:
    - Analysis Summary within the Job XML returned via web services
    - Detailed Analysis Results including Per Frame Metatadata Changes In an external Log File

Upmix — SoundField & DTS Neural

AudioTools Server utilizes qualified third party plug–in modules to provide upmix functionality. Upmix is useful for converting any stereo content to surround format, particularly back catalog films, television shows, documentaries, deleted scenes, and featurettes. Though each module produces unique results based on proprietary algorithms, both modules are designed to ensure that all resulting material is fold down–compatible, essential to preserving the intent and quality of the original stereo mix.

SoundField UPM-1 for AudioTools Server option
Parameters
- Inputs, Output, Front Direct Sound (dB), Front/Rear Ambient Sound (dB), Width, Center Divergence, Rear Delay (ms.), Output Program Configuration

DTS Neural Surround™ UpMix for AudioTools Server option
Parameters
- Inputs, Outputs, Depth, Width, 80 Hz LFE LPF, Peak limiter [with bypass] (dBFS)

- ◊ Module Output:
    - Processed Audio

Module Specifications

AudioTools Downmix

The optional AudioTools Downmix module accepts up to 7.1 multi-channel PCM audio and performs a downmix to a stereo program. The Downmix module can be controlled by Dolby E metadata derived from the AudioTools SurCode for Dolby E Decoder module.

◊ Accepts up to 7.1 and provides control over input trims and the use of an optional limiter

◊ The Downmix module accepts parameters that specify the center and surround downmix levels, and selection of Lo/Ro or Lt/Rt downmix coefficients.

◊ The Downmix module can use Dolby metadata for Downmix from a previous Dolby decoding process, applying those values.

◊ The Downmix module includes a Low Pass Filter (LPF), with user adjustable frequency and gain. The LPF can be disabled.

◊ The Downmix module takes 6 channels as input, and produces either 2 channels out or 8 channels out, with the original 5.1 + downmixed channels on 7 & 8.

◊ Module Input:
   Parameters
   • Source File, Channel Configuration, LoRo Center Gain (dbFS), LoRo Surround Gain (dbFS), LtRt Center Gain (dbFS), LtRt Surround Gain (dbFS), Limiter On, Limiter Threshold (ms.), Limiter Attack (ms.), Limiter Sustain (ms.), Limiter Release (ms.), LFE On, LFE LPF On, LFE LPF Cutoff (Hz), AddDownmixChannelsToSource, Results Directory

◊ Module Output:
   • Processed Audio

AudioTools Channel Manager

The optional AudioTools Channel Manager module provides altering, swapping and replacing of audio programs or audio channels in MXF, QuickTime, Dolby E or multi–channel PCM files. Note that AudioTools for Extractor/ReWrap for MXF, AudioTools Extractor/ReWrap for QuickTime or SurCode for Dolby E Encoder/Decoder are required for processing MXF, QuickTime or Dolby E content.

◊ The channel replacement process accepts a Dolby E WAV or PCM file as input, one or more PCM audio files for replacement (WAV or AIF), and the channel numbers to replace and program description text for the replacement channels.

◊ The Dolby E file is decoded to PCM and then re-encoded, replacing the specified channels with the PCM audio from the input files. The PCM audio may be supplied as mono or stereo files.

◊ All metadata from the Dolby E input file, including program mode, encode start time and AC3 metadata will be copied from the Dolby E input file to the Dolby E output file. The output file will include a broadcast wave chunk (BEXT) which specifies the encode start time.

◊ Program description text, if specified for replacement, will be used to replace the existing text for the replacement channels.

◊ The Channel Manager swap function can swap and re-order any audio channel or audio program configuration within MXF or QuickTime container formats, and in Dolby E or multi-channel PCM audio files.

◊ The results of the processing are formatted as an XML log file, which can be retrieved from the system.

◊ Module Input:
   • PCM Audio
   Parameter
   • Channel, Channel (Input) ID, Channel Operation, Channel (Copy) ID

◊ Module Output:
   • Processed Audio

AudioTools Audio Quality Analysis

The optional AudioTools Quality Analysis module uses a highly flexible algorithm to analyze incoming PCM audio content and detect anomalies such as data corruption and program gaps. Given a specified energy threshold in a specified frequency range, the

module reports the results for a given period of time. Using those params, the module can be configured to detect a wide variety of anomalies. Data corruption tends to produce large amplitude bursts at high frequency, whereas program gaps are the absence of energy in critical bands.

This module generates an output log that indicates the occurrences of anomalies found and the time code locations where they occurred. By selectively setting the frequency range and threshold, the absence or presence of energy at particular frequencies can be detected. Prolonged absence of moderate energy in the 0.4 to 4 kHz range usually indicates a gap in the program, while inordinate short term energy at 17 to 20 kHz usually indicates data corruption. Subsonic noise, time code bleed, perf noise and mains buzz and hum are also easily flagged.

For each analysis or reporting period, a value in dB is returned each time the threshold is exceeded. As this is purely numeric data, the AudioTools Workflow Control option is highly recommend so that subsequent processes and modules can take advantage of the data produced and act on those results.

> *Parameter*
> - Low Cutoff (Hz), High Cutoff (Hz), Analysis Period (sec.), Analysis Threshold (dBFS)

AudioTools Channel Assignment Detection

The optional AudioTools Channel Assignment Detection module determines channel locations within multi-channel programs by evaluating relative loudness levels, channel correlations, and frequency range on a representative section of the audio.

Based on relative loudness levels, channel correlation and frequency range, the module identifies likely channel pairs, front/rear channels, center and LFE. This module is typically used to process a portion of an audio stream, until the desired level of confidence in channel assignment is reached.

The LFE cutoff defines a LPF resonant frequency used to detect the LFE. The Mid Cutoff Low and High define the passband used to evaluate the content for midrange energy.

> *Parameter*
> - LFE Cutoff (Hz), Mid Cutoff Low (Hz), Mid Cutoff High (Hz)

AudioTools Program Correlation Check

The optional AudioTools Program Correlation module performs a correlation check between two audio programs and is used to determine whether both represent the same content. The number of channels in each program may range from 1 to 8. A common workflow uses the Program Correlation Check module to evaluate eight channel, 3/2 + 2/0 PCM decoded from the SurCode for Dolby E Decoder module to correlate the 5.1 and stereo programs and determine if the stereo program is derived from the surround program.

◊ An initial alignment step is performed to account for small differences in the start of each program. The alignment tolerance is a parameter that can be controlled.

◊ The log output of the module indicates the level of correlation found during the program comparison for each second of audio, an overall correlation value, and the initial alignment time.
> *Parameter*
> Correlation Period , Noise Floor (dBFS), Mid Cutoff Low (Hz), Mid Cutoff High (Hz)

AudioTools Frame Rate Toolbox

The AudioTools Frame Rate Toolbox option is a signal processing module that provides multichannel, phase–coherent pitch shift, time stretch and sample rate conversion for frame rate and format adjustment and correction. This module is useful for standards conversion, duration modification and other audio time base and time domain–correction tasks.

> *Pitchshift & Time Stretch Parameter*
> Semitones, Cents, Quality, Solo Mode, Transient Sensitivity, Noise Generation, Window Size, Phase Coherence, Formant Shoft Active, Formant Shoft Semitones , Formant Correction Strength, Formant Correction Width
> Sample Rate Convertor
> Target (bps), Quality

AudioTools Audio Extractor for MXF

The optional AudioTools MXF Audio Extractor module provides the ability to access the audio and metadata stored in MXF files. Audio tracks within MXF files are extracted efficiently, without unwrapping video tracks. This module also supports functions for

reading MXF metadata. MXF OP1 and OP1a profiles are currently supported, and more profiles will be added based on project requirements.

The MXF Audio Extractor module can optionally store an MXF Channel map, which can be used for other jobs. This eliminates having to re–read the file to generate a map each time the file essence needs to be accessed.

The module can also extract audio from continually growing MXF files during an Ingest operation, for read-only processes such as Loudness Measurement or audio analysis.

AudioTools ReWrap™ for MXF

The optional AudioTools MXF ReWrap module re–wraps audio data back into existing MXF files. Audio is re–wrapped "in place," meaning the same audio format is used and samples are simply replaced. MXF OP1 and OP1a profiles are currently supported, and more profiles will be added, based on project requirements.

AudioTools Audio Extractor for LXF

The optional AudioTools LXF Audio Extractor module provides the ability to access the audio and metadata stored in LXF files. Audio tracks within LXF files are extracted efficiently, without unwrapping video tracks.

AudioTools ReWrap™ for LXF

The optional AudioTools LXF ReWrap module outputs audio data into existing LXF files. New audio data replaces existing data. Existing video within the LXF file is not modified.

AudioTools Audio Extractor for GXF

The optional AudioTools GXF Audio Extractor module provides the ability to access the audio and metadata stored in GXF files. Audio tracks within GXF files are extracted efficiently, without unwrapping video tracks.

AudioTools ReWrap™ for GXF

The optional AudioTools GXF ReWrap module outputs audio data into existing GXF files. New audio data replaces existing data. Existing video within the GXF file is not modified.

AudioTools Audio Extractor for QuickTime™

The optional AudioTools Audio Extractor for QuickTime module extracts PCM audio streams from QuickTime files of all video formats. The module provides the ability to access the audio and metadata stored in QuickTime files. Audio tracks within Quick-Time files are extracted efficiently, without unwrapping video tracks. This module also supports functions for reading QuickTime metadata.

AudioTools ReWrap™ for QuickTime™

The optional AudioTools ReWrap for QuickTime™ re–wraps audio data back into existing QuickTime files or audio–only Quick-Time files can be created from scratch. Audio can re–wrapped "in place," meaning the same audio format is used and samples are simply replaced. Or, a new Track/Channel configuration can be generated and the QuickTime file is effectively recreated using the untouched video essence.

AudioTools Audio Extractor for Transport Streams

The optional AudioTools Audio Extractor for Transport Streams module extracts the relevant audio streams from transport streams containing H.264 and MPEG-2 video formats. The module can extract PCM, AC-3, E-AC-3, AAC and MPEG audio types. The module can extract arbitrary numbers of audio streams in an arbitrary number of video programs. Existing video within the transport stream will not be modified. Note that Program Streams are not supported.

AudioTools ReWrap™ for Transport Streams

The optional AudioTools ReWrap for Transport Streams module inserts audio into transport stream files, including H.264 and MPEG-2 video programs. Existing video within the transport stream file will not be modified. The module will only insert the same audio codec type into a stream that previously carried that codec type. Combined use of both the AudioTools Audio Extractor for Transport Streams and the AudioTools ReWrap for Transport Streams modules requires either external automation system or the AudioTools Workflow Control module. Note that Program Streams are not supported.

AudioTools AudioSecter

The optional AudioTools AudioSecter module provides time/region–based insertion, deletion and/or muting. Predefined region definitions are driven via Workflow XML.

The AudioTools Quality Analysis module can also be employed, in conjunction with the AudioTools AudioSecter module, to define regions based on frequency and/or amplitude. Detected regions can be removed or muted. Arbitrary region detection is driven by specifying an amplitude threshold and/or a frequency range. Null or silent audio can also be inserted at the start or end of a detected region. In frequency and/or amplitude mode, the AudioTools AudioSecter module requires the AudioTools Quality Analysis and Workflow Control modules.

AudioTools Watermarking

The optional AudioTools Watermarking module adds Civolution NexTracker Audience Measurement watermarking for audio essence and integration with Civolution's SyncNow Automatic Content Recognition service. SyncNow provides server-side content identification and triggering for real-time recognition of broadcast content, and AudioTools Server provides secure integration with SyncNow for watermark detection and validation.

The Civolution NexTracker Watermarking module works within the AudioTools Server environment to seamlessly watermark content within larger workflows involving many AudioTools Server processes, or as a standalone watermarking solution. The Civolution NexTracker Watermarking module for AudioTools Server can be applied at any point in the AudioTools Server Workflow, the most common point being post-processing as the audio is re-wrapped to the final container.

Civolution Watermark Metadata is defined via AudioTools Server Workflow XML. Any number of metadata specifications are supported via multiple workflow profiles. This enables simple handling of multiple downstream watermarking requirements in a completely automated environment.

AudioTools WFC — WorkFlow Control

The optional AudioTools WFC workflow control module enables AudioTools Server to act as a high level business logic automation engine. Workflows can drive any number of AudioTools Jobs in a logical flow of processes and tie together the logic between Jobs. Workflows apply Conditional, Exceptional, Parallel, and Iterative logic to achieve highly complex business processes in an automated environment.

Workflows use a well-defined XML syntax to define Steps, Parameters, Logic Test Cases, Success and Error Conditions, and Utility Functions. Workflow Steps execute Jobs, retrieve Job Results, program Job and Processing Module parameters, test Parameters, modify Workflow State, and call Utility Functions.

Workflow Parameters hold values. Values can be Literal, Reference or Calculation values. Examples if a Literal values would be a number representing Target Loudness or a file path. Reference values refer to another parameter's value or a Processing Results value in Job XML. Calculations values result when a calculation is performed on two parameters to produce a new Parameter Value.

Workflow Tests determine the relationship between two Parameters and define behavior if a certain condition is met. Tests produce a True/False answer based on whether Parameters are "Equal," "Not Equal," "Less Than," "Less Than or Equal," "Greater Than" or "Greater Than or Equal." Success Conditions define Workflow behavior if a Step Succeeds. Error Conditions define Workflow behavior if a Step Fails.

Functions provide access to enhanced system features or special case behavior. Available Functions include Move File, Delete File, Rename File, Consolidate Job Results, and Execute Process.

In addition to process control, the AudioTools WFC module provides load balancing of multiple AudioTools Server Nodes. A Node containing the AudioTools WFC license communicates with other nodes on the network to negotiate resource availability and ensure Nodes are optimized based on performance and license availability. Workflow control failover is also part of the AudioTools WFC module. For enterprise–class environments, this enables processing redundancy which, in the event of a failure, ensure processing continuity and uninterrupted productivity via non–stop operation. Note that one AudioTools WFC license can control workflows on multiple AudioTools Server machines with or without load balancing. For failover, an additional license of AudioTools WFC is required for the failover host.

Customer–Defined & 3rd Party Modules

AudioTools Server has been designed to also host customer–defined modules. Other manufacturers are welcome to use the AudioTools Server platform as a vehicle for making their specific functionality available in an enterprise level, file–based environment.

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