Audio Processing - Processors

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Automixer - Stereo

Overview

The Automixer - Stereo is a Processor for combining two stereo program sources with additional mono voice signals. It provides input conditioning, fail-over handling, equalization, level control, voice-over processing, mix-minus generation, clean mono generation, output conditioning, limiting, and metering in one processing chain.

The Processor is intended for stereo production workflows where program audio and voice signals need to be mixed, controlled, and delivered as program, mix-minus, and clean mono outputs.

Typical Applications

Typical applications include:

- Stereo broadcast program mixing
- Voice-over integration into stereo program audio
- Mix-minus generation for production and contribution workflows
- Clean mono output generation
- Loudness-controlled stereo output processing
- Program fail-over workflows

Included Modules

The Stereo version includes the following processing and measurement stages:

- Input Meter
- Input Conditioner
- Fail Over
- Equalizer
- Level Magic
- Voice Over
- Mix-minus
- Matrix Mixer
- True Peak Limiter
- Output Conditioner
- Output Meter
- Loudness Measurement

The signal flow includes separate input conditioning for stereo program and mono voice signals, followed by fail-over selection, equalization, level control, voice-over mixing, mix-minus generation,

clean mono generation, output limiting, output conditioning, and measurement.

Channel Formats

This article describes the Stereo version.

Automixer - 5.1

Overview

The Automixer - 5.1 is a Processor for combining two 5.1 program sources with additional mono voice signals. It provides input conditioning, automatic upmixing, equalization, level control, voice-over processing, mix-minus generation, downmixing, output conditioning, limiting, and metering in one processing chain.

The Processor is intended for surround program production workflows where program audio and voice signals need to be mixed, controlled, and delivered in multiple output formats.

Typical Applications

Typical applications include:

- 5.1 broadcast program mixing
- Voice-over integration into surround program audio
- Mix-minus generation for production and contribution workflows
- Clean mono output generation
- Program downmix generation
- Loudness-controlled surround output processing

Included Modules

The 5.1 version includes the following processing and measurement stages:

- Input Meter
- Input Conditioner
- Auto Upmix
- Equalizer
- Level Magic
- Voice Over
- Mix-minus
- Downmix
- Matrix Mixer
- True Peak Limiter
- Output Conditioner
- Output Meter
- Loudness Measurement

The signal flow includes separate input conditioning for program and mono voice signals, followed by upmixing, equalization, level control, voice-over mixing, mix-minus generation, downmixing, output limiting, output conditioning, and measurement.

Channel Formats

This article describes the Surround (5.1) version.

Delay

Overview

The Delay is a Processor for applying delay to 8 or 16 audio channels. It is used to align audio signals in time or to compensate for timing differences between processing, routing, video, or transmission paths.

The Processor provides a dedicated delay stage for multichannel audio workflows. It can be loaded as an 8 or 16 channel version. All channels can be used individually or for multi-channel applications.

Typical Applications

Typical applications include:

- Audio-to-video synchronization
- Timing compensation between signal paths
- Delay alignment in multichannel workflows
- System integration and routing compensation
- Satellite link compensation

Included Modules

The 8- and 16-channel versions includes the following processing stage:

- Delay

The signal flow is:

Input → Delay → Output

Channel Formats

This article describes the 8- and 16-channel versi

Dolby® E Decoder and Encoder

Dolby® E Decoder and Encoder

AIXpressor can optionally be equipped with a Dolby® E decoder and encoder. Please see Adding a Processor for instructions on installing a new processor.

Dolby E Overview

Dolby E was introduced in 1999 as a professional audio coding system that allows the distribution of multichannel digital audio within two-channel infrastructures. Up to eight channels of broadcast-quality audio, plus metadata, can be distributed via a single AES3 pair.

Dolby E Decoder

The decoder will decode all standard Dolby E programs.

Important - Unlike legacy Dolby E implementations in which the second program of a 4x2 configuration appeared on outputs 7/8, **AIXpressor always shows the audio outputs in ascending order regardless of which program configuration is being decoded.**

Input Page

Click on the **Dolby E Decoder button (A),** then on the **In button (B)** to display a running total of the **number of frames received (C).**



Figure 1 - Dolby E decoder input page

Decoder Page

Click on the **Dec button (A)** to display a **list of all parameters (B),** including (but not limited to) the following:

- **Library Version -** The version of the installed Dolby library.
- **Input Signal -** Green indicates a Dolby E signal; yellow indicates PCM audio; red indicates mute.
- **CRC Errors -** Displays the number of CRC errors in the incoming Dolby E stream. The counter can be reset with the **Execute button** at the bottom of the metadata display.
- **Frame Rate -** Indicates the frame rate of the incoming Dolby E signal.
- **Frame Offset -** Shows the offset between the incoming Dolby E signal and the system frame reference.
- **Bit Depth -** Indicates the bit depth of the incoming Dolby E signal.
- **Program Configuration -** Displays the Program Configuration metadata parameter, which determines how the audio channels are grouped within the Dolby E bitstream.
- **Program Description -** Shows the Program Description metadata, typically the name of the program or a description of the program source.
- **Channel Mode -** The Channel Mode (also known as the *Audio coding mode*) indicates the active channels within the encoded bitstream. In the decoder, this ensures the proper routing of the audio channels to downstream devices.
- **LFE Channel -** Indicates whether or not there is an LFE Channel present in the incoming bitstream.
- **Bitstream Mode -** Displays the audio service contained within the incoming Dolby E bitstream.
- **Dialog Normalization -** Also referred to as Dialog Level or *dialnorm*, this indicates the average level of dialog in the presentation and determines the proper level shift in the decoder to ensure proper normalization within and between program sources.

Note - The Decode page displays considerable additional information A detailed explanation of all metadata parameters is available in the Dolby Metadata Guide.



Figure 2 - Dolby E decoder parameter page

Meter Page

Click on the **Mtr button (A)** to view the decoder meter page. The **Peak Level meters (D)** show sample-accurate peaks for all eight incoming audio channels with a hold function. **Peak Accuracy (B)** is currently fixed at sample-accurate resolution. The **Peak Hold control (C)** allows for Automatic or Manual control of the hold feature. Clicking the **Reset Measurement Execute button (E)** resets the peak hold indicators.



Figure 3 - Dolbe E decoder meter page

Output Page

Click on the **Out button (A)** to show the **number of clients (B)** currently connected to the unit and a running total of the **number of frames emitted (C)** from the decoder.



Figure 4 - Dolby decoder output page

Important - Any preset saved and stored in the Dolby E decoder will also be available in the Dolby E encoder, providing an easy way to use the same settings and metadata values. Note that you may receive an error message if the metadata in one of the received programs does not match the program type - for instance, if LFE is activated for a 2/0 Channel Mode of a 5.1+2 Program Configuration. The encoder will automatically correct this, and you may dismiss the message.

Dolby E Encoder

Input Page

Click on the **Dolby E Encoder button (A),** then on the **In button (B)** to display a running total of the **number of frames received (C).**



Figure 5 - Dolby encoder input page

Meter Page

Click on the **Mtr button (A)** to view the encoder meter page. The **Peak Level meters (D)** show sample-accurate peaks for all eight encoded audio channels with a hold function. **Peak Accuracy (B)** is currently fixed at sample-accurate resolution. The **Peak Hold control (C)** allows for Automatic or Manual control of the hold feature. Clicking the **Reset Measurement Execute button (E)** resets the peak hold indicators.



Figure 6 - Dolby encdoer meter page

Encoder Page

Click on the **Enc button (A)** to display a **list of all parameters (B),** including (but not limited to) the following:

- **Library Version -** The version of the installed Dolby library.
- **Encoder Status -** Green indicates the encoder is active; red indicates it has stopped.
- **Video Sync Frame Rate Status -** Green indicates proper sync; red indicates a mismatch.
- **Video Sync Frame Rate -** The actual measured reference frame rate extracted from the sync signal.
- **DSP 1/2 - DSP 7/8 -** A label of the encoder inputs based on the signal type to ensure proper encoding for the chosen Program Configuration.
- **Frame Rate -** The Frame Rate of the Dolby E-encoded bitstream.
- **Video Sync Shift Offset -** Sets the guard band relative to the video switching point (see SMPTE RP 168) to ensure proper Dolby E alignment. Shift steps are measured in audio samples using a 48kHz sample rate.
- **Program Configuration -** Sets the Program Configuration metadata parameter, which determines how the audio channels are grouped within the Dolby E bitstream.
- **Program Description -** Sets the Program Description metadata, typically the name of the program or a description of the program source.
- **Channel Mode -** The Channel Mode (also known as the *Audio coding mode*) sets the active channels within the encoded bitstream.
- **LFE Channel -** Informs the downstream decoder as to whether or not an LFE channel is present.
- **Bitstream Mode -** Sets the audio service contained within the incoming Dolby E bitstream.
- **Dialog Normalization -** Also referred to as Dialog Level or *dialnorm*, this sets the average level of dialog in the presentation and determines the proper level shift in the

downstream decoder to ensure proper normalization within and between program sources.

- **Line Mode Profile -** Sets the dynamic range metadata profile for line outputs of the downstream decoder.
- **RF Mode Profile -** Sets the dynamic range metadata profile for the RF outputs of the downstream decoder.
- **DC Filter -** Determines whether or not a DC-blocking 3Hz highpass filter is applied prior to encoding to remove DC offsets in the program audio.
- **Lowpass Filter -** Determines whether or not a lowpass filter is applied to the main input prior to encoding to remove high frequencies that would not be encoded, thereby preventing aliasing upon decoding.
- **LFE Filter -** Determines whether or not a 120Hz eighth-order lowpass filter is applied to the LFE channel prior to encoding, thereby removing frequencies that would cause aliasing when decoded.

Note - A detailed explanation of all metadata parameters is available in the Dolby Metadata Guide.



Figure 7 - Dolby encoder parameter page

Output Page

Click on the **Out button (A)** to show the **number of clients (B)** currently connected to the unit and a running total of the **number of frames emitted (C)** from the encoder.



Figure 8 - Dolby encoder output page

Note - Dolby, Dolby Audio, and the double-D symbol are trademarks of Dolby Laboratories Licensing Corporation. The Dolby E encoder and decoder in AIXpressor are manufactured under license from Dolby Laboratories.

Dolby® E Decoder with S-ADM output

Overview

The Dolby® E Decoder + S-ADM is a Processor for decoding Dolby E content to PCM and making the decoded audio available for workflows that include S-ADM metadata handling.

The Processor is intended for workflows where Dolby E input signals need to be decoded and prepared for downstream immersive or metadata-aware processing. A version without the S-ADM metadata output can be loaded.

Typical Applications

Typical applications include:

- Dolby E decoding
- Conversion of encoded contribution signals into decoded audio
- Preparation of decoded audio for S-ADM workflows
- Broadcast and production ingest workflows

Included Modules

The Processor includes the following processing and measurement stages:

- Dolby E Decoder
- Output Meter

The signal flow is:

Input → Dolby E Decoder → Output Meter → Output

Dolby® ED2 Decoder with S-ADM output

Overview

The Dolby® ED2 Decoder is a Processor for decoding Dolby ED2 input signals into audio that can be used by downstream processing, monitoring, or routing stages.

The Processor is intended for broadcast and immersive audio workflows where Dolby ED2 content must be decoded before further processing.

Typical Applications

Typical applications include:

- Dolby ED2 decoding
- Immersive audio ingest workflows
- Broadcast contribution signal decoding
- Preparation of decoded audio for monitoring or further processing

Included Modules

The Processor includes the following processing and measurement stages:

- Dolby E Decoder
- Output Meter

The signal flow is:

Input → Dolby E Decoder → Output Meter → Output

Dolby® ED2 Encoder

Overview

The Dolby® ED2 Encoder is a Processor for encoding multichannel or immersive audio into Dolby ED2 format.

The Processor is intended for broadcast and contribution workflows where processed audio must be encoded as Dolby ED2 for delivery or downstream transmission.

Typical Applications

Typical applications include:

- Dolby ED2 encoding
- Immersive audio contribution workflows
- Broadcast delivery workflows
- Encoding of multichannel audio for downstream distribution

Included Modules

The Processor includes the following processing and measurement stages:

- Input Meter
- Dolby ED2 Encoder

The signal flow is:

Input → Input Meter → Dolby ED2 Encoder → Output

Downmix Matrix

Overview

The Downmix Matrix is a Processor for creating a downmix from a multichannel input signal. It uses a matrix-based processing stage to derive a lower-channel output format from a higher-channel input format.

The Processor is intended for workflows where immersive or surround content must be made available in a reduced channel format.

Typical Applications

Typical applications include:

- Creating downmixes from multichannel sources
- Format conversion for monitoring
- Generating compatibility outputs
- Routing multichannel content to lower-channel destinations

Included Modules

The Processor includes the following processing stage:

- Downmix Matrix

The signal flow is:

Input → Downmix Matrix → Output

Channel Formats

The backup shows a 7.1.4 input-related configuration and an 8-channel output-related configuration.

[PLACEHOLDER: confirm exact input and output channel format naming]

Block Diagram

[PLACEHOLDER: insert block diagram for Downmix Matrix]

Notes

[PLACEHOLDER: block diagram required]

[PLACEHOLDER: confirm target downmix format]

Fail Over

Overview

The Fail Over - Stereo (+Conditioner) is a Processor for stereo source fail-over with input and output conditioning. It selects or switches between stereo signal paths and provides conditioning before and after the fail-over stage.

The Processor is intended for workflows that require a conditioned backup or fallback source path for stereo audio.

Typical Applications

Typical applications include:

- Stereo program fail-over
- Backup source switching
- Conditioned fallback signal paths
- Operational redundancy in broadcast workflows

Included Modules

The Stereo version includes the following processing stages:

- Input Conditioner
- Fail Over
- Output Conditioner

The signal flow is:

Input → Input Conditioner → Fail Over → Output Conditioner → Output

Channel Formats

This article describes the Stereo version.

The backup also shows an 8-channel layout associated with the Processor configuration.

Block Diagram

[PLACEHOLDER: insert block diagram for Fail Over - Stereo (+Conditioner)]

Notes

[PLACEHOLDER: block diagram required]

[PLACEHOLDER: confirm whether the 8-channel layout is user-facing or internal]

FM Conditioner

Overview

The FM Conditioner Channel Strip - Stereo is a Processor for stereo program processing with a special FM conditioning stage for preparing the audio signal for FM broadcast. It combines input measurement, input conditioning, fail-over handling, phase rotation, de-essing, spectral processing, equalization, dynamics processing, loudness control, FM conditioning, output conditioning, and measurement.

The Processor is intended for stereo broadcast workflows that require a complete program processing chain including FM-oriented signal conditioning. The Processor is for pre-conditioning only and does not replace the functionality of a stereo coder.

> Please see the FM Conditioner Module description for detailed setup instructions.

Typical Applications

Typical applications include:

- Stereo FM program processing
- Spectral program shaping
- Program fail-over workflows
- De-essing and phase optimization
- Loudness-controlled stereo output processing
- Generation of program and monitor outputs

Included Modules

The Stereo version includes the following processing and measurement stages:

- Loudness Measurement
- Input Conditioner
- Fail Over
- Phase Rotator
- De-Esser
- Spectral Signature
- Equalizer
- Expander
- Compressor
- Level Magic

- FM Conditioner
- Output Conditioner

The signal flow includes input measurement and conditioning for two stereo sources, fail-over selection, phase rotation, de-essing, spectral signature processing, equalization, expansion, compression, loudness control, FM conditioning, output conditioning, and output measurement.

Channel Formats

This article describes the Stereo version.

Monitoring Controller - Immersive

Overview

The Immersive Monitoring Controller is a Processor for immersive production workflows. It combines input metering, input conditioning, source switching, monitoring control, monitor mix processing, downmixing, bass management, equalization, sample playback, output conditioning, and output metering.

The Processor is intended for controlled monitoring of 7.1.4 (lower channel configurations can be loaded) audio in production, quality-control, and immersive audio environments.

Typical Applications

Typical applications include:

- 7.1.4 immersive audio monitoring
- Source switching between monitoring inputs
- Monitor mix control
- Downmix monitoring
- Bass management
- Room or monitor equalization
- Output conditioning and metering

Included Modules

The 7.1.4 version includes the following processing and measurement stages:

- Input Meter
- Input Conditioner
- Buffer Multiplexer
- MAP Monitoring Controller
- MAP Mix
- Downmix Multichannel
- Bass Management
- Equalizer
- Sampler
- Output Conditioner
- Output Meter

The signal flow includes input measurement and conditioning for two 7.1.4 sources, source switching, monitoring control, monitor mix processing, downmixing, bass management, equalization, sample playback, output conditioning, and output metering.

Channel Formats

This article describes the 7.1.4 version. Lower channel configurations can be loaded utilizing the same license.

Mix Matrix

Overview

The Mix Matrix - 16 Ch is a Processor for matrix-based mixing and routing of multichannel audio. It provides a flexible mix matrix stage for creating output signals from multiple input channels.

The Processor is intended for workflows that require channel combining, distribution, or custom signal matrixing.

Typical Applications

Typical applications include:

- Multichannel mix matrixing
- Custom channel routing
- Signal distribution
- Combining or deriving output channels from multiple inputs

Included Modules

The 16-channel version includes the following processing stage:

- Mix Matrix

The signal flow is:

Input → Mix Matrix → Output

Channel Formats

This article describes the 16-channel version.

The backup also shows an 8-channel layout associated with the Processor configuration.

Block Diagram

[PLACEHOLDER: insert block diagram for Mix Matrix - 16 Ch]

Notes

[PLACEHOLDER: block diagram required]

[PLACEHOLDER: confirm whether the 8-channel layout is user-facing or internal]

MPEG-H 3D Audio Authoring

Overview

The MPEG-H 3D Audio Authoring Processor is used to prepare audio and metadata for MPEG-H 3D Audio workflows. It provides an authoring stage with input and output metering for the audio path.

The Processor is intended for workflows where MPEG-H 3D Audio content is created or prepared before rendering, monitoring, or delivery.

Typical Applications

Typical applications include:

- MPEG-H 3D Audio authoring
- Preparation of immersive audio content
- Metadata-aware production workflows
- Input and output monitoring of authored audio

Included Modules

The Processor includes the following processing and measurement stages:

- Input Meter
- MPEG-H 3D Audio Authoring
- Output Meter

The signal flow is:

Input → Input Meter → MPEG-H 3D Audio Authoring → Output Meter → Output

MPEG-H 3D Audio Renderer

Overview

The MPEG-H 3D Audio Renderer is a Processor for rendering MPEG-H 3D Audio content for monitoring and output measurement. It combines MPEG-H rendering with a linked monitor controller and loudness measurement.

The Processor is intended for immersive audio workflows where MPEG-H content must be rendered and monitored in a controlled output format.

Typical Applications

Typical applications include:

- MPEG-H 3D Audio rendering
- Immersive audio monitoring
- Renderer output loudness measurement
- Production and quality-control monitoring workflows

Included Modules

The Processor includes the following processing and measurement stages:

- Input Meter
- MPEG-H 3D Audio Renderer
- Monitor Controller Linked
- Loudness Measurement

The signal flow includes MPEG-H rendering, linked monitor control, and output loudness measurement.

Program Processing Channel Strip – Immersive

Overview

The Program Processing Channel Strip - 7.1.4 (+Upmix) is a Processor for immersive program processing with automatic upmixing. It combines input measurement, input conditioning, upmixing, equalization, expansion, compression, loudness control, output conditioning, and loudness measurement.

The Processor is intended for 7.1.4 program workflows where source material needs to be processed, upmixed, controlled, and delivered as an immersive output.

Typical Applications

Typical applications include:

- 7.1.4 immersive program processing
- Upmix-assisted immersive workflows
- Equalization and dynamics control
- Loudness-controlled immersive output processing
- Broadcast and production workflows using 7.1.4 audio

Included Modules

The 7.1.4 version includes the following processing and measurement stages:

- Loudness Measurement
- Input Conditioner
- Auto Upmix
- Equalizer
- Expander
- Compressor
- Level Magic
- Output Conditioner

The signal flow includes input measurement and conditioning, automatic upmixing, equalization, expansion, compression, loudness control, output conditioning, and output measurement.

Channel Formats

This article describes the 7.1.4 version with upmixing. Lower configurations (5.1.4) with and without upmixing can be loaded.

Program Processing Channel Strip – Stereo

Overview

The Program + Spectral Processing Channel Strip - Stereo is a Processor for stereo program processing with spectral signature processing. It combines input measurement, input conditioning, fail-over handling, spectral processing, equalization, expansion, compression, loudness control, output conditioning, and loudness measurement.

The Processor is intended for stereo program workflows where tonal shaping, dynamics control, source fail-over, and loudness control are required in one processing chain.

Typical Applications

Typical applications include:

- Stereo broadcast program processing
- Spectral program shaping
- Program fail-over workflows
- Dynamic range control
- Loudness-controlled stereo output processing

Included Modules

The Stereo version includes the following processing and measurement stages:

- Loudness Measurement
- Input Conditioner
- Fail Over
- Spectral Signature
- Equalizer
- Expander
- Compressor
- Level Magic
- Output Conditioner

The signal flow includes input measurement and conditioning for two stereo sources, fail-over selection, spectral signature processing, equalization, expansion, compression, loudness control, output conditioning, and output measurement.

Channel Formats

This article describes the Stereo version.

Block Diagram

[PLACEHOLDER: insert block diagram for Program + Spectral Processing Channel Strip - Stereo]

Notes

[PLACEHOLDER: block diagram required]

Program Processing Channel Strip – Surround

Overview

The Program + Spectral Processing Channel Strip - 5.1 is a Processor for 5.1 program processing with spectral signature processing. It combines input measurement, input conditioning, automatic upmixing, spectral processing, equalization, expansion, compression, loudness control, output conditioning, and loudness measurement.

The Processor is intended for surround program workflows where tonal shaping, dynamics control, and loudness control are required in one processing chain.

Typical Applications

Typical applications include:

- 5.1 broadcast program processing
- Spectral program shaping
- Upmix-assisted surround workflows
- Dynamic range control
- Loudness-controlled 5.1 output processing

Included Modules

The 5.1 version includes the following processing and measurement stages:

- Loudness Measurement
- Input Conditioner
- Input Meter
- Auto Upmix
- Spectral Signature
- Equalizer
- Expander
- Compressor
- Level Magic
- Output Conditioner

The signal flow includes input measurement and conditioning for two 5.1 sources and one mono source, automatic upmixing, spectral signature processing, equalization, expansion, compression,

loudness control, output conditioning, and output measurement.

Channel Formats

This article describes the Surround (5.1) version.

S-ADM Authoring

Overview

The S-ADM Authoring Processor is used to prepare audio and metadata for S-ADM workflows. It provides an S-ADM authoring stage with input and output metering for the audio path.

The Processor is intended for workflows where S-ADM content is created, prepared, or passed into downstream rendering, monitoring, or delivery processes.

Typical Applications

Typical applications include:

- S-ADM authoring
- Preparation of metadata-aware audio content
- Immersive or object-based audio workflows
- Input and output monitoring of authored audio

Included Modules

The Processor includes the following processing and measurement stages:

- Input Meter
- S-ADM Authoring
- Output Meter

The signal flow is:

Input → Input Meter → S-ADM Authoring → Output Meter → Output

S-ADM Renderer

Overview

The S-ADM Dolby Renderer is a Processor for rendering S-ADM content for monitoring and output measurement. It combines S-ADM rendering with a linked monitor controller and loudness measurement.

The Processor is intended for immersive audio workflows where S-ADM content must be rendered and monitored in a controlled output format.

Typical Applications

Typical applications include:

- S-ADM rendering
- Immersive audio monitoring
- Renderer output loudness measurement
- Production and quality-control monitoring workflows

Included Modules

The Processor includes the following processing and measurement stages:

- Input Meter
- S-ADM Renderer
- Monitor Controller Linked
- Loudness Measurement

The signal flow includes S-ADM rendering, linked monitor control, and output loudness measurement.

Voice Over Mixer - STEREO

Overview

The Voice Over Mixer - Stereo is a Processor for combining a stereo main program signal with up to six mono voice signals. It provides input conditioning, voice-over processing, mix-minus generation, matrix mixing, level control, true peak limiting, output conditioning, and measurement.

The Processor is intended for stereo voice-over workflows where a main stereo program and up to six mono voice sources need to be combined and delivered in multiple output formats.

Typical Applications

Typical applications include:

- Stereo program voice-over mixing
- Up to six voice production workflows
- Mix-minus generation
- Clean mono output generation
- Loudness-controlled stereo output processing

Included Modules

The Processor includes the following processing and measurement stages:

- Input Meter
- Input Conditioner
- Voice Over
- Mix-minus
- Matrix Mixer
- Level Magic
- True Peak Limiter
- Output Conditioner
- Output Meter
- Loudness Measurement

The signal flow includes input conditioning for the main stereo program and up to six mono voice sources, voice-over mixing, mix-minus generation, clean mono generation, level control, limiting, output conditioning, and measurement.

Channel Formats

This article describes a stereo main program version with up to six mono voice inputs.

The Processor uses stereo, mono, and clean mono output-related paths.

Block Diagram

[PLACEHOLDER: insert block diagram for Voice Over Mixer - Main Stereo + 2x Mono]

Notes

[PLACEHOLDER: block diagram required]

Voice Over Mixer - SURROUND

Overview

The Voice Over Mixer - Surround is a Processor for combining a 5.1 main program signal with up to six mono voice signals. It provides input conditioning, voice-over processing, mix-minus generation, downmixing, matrix mixing, level control, true peak limiting, output conditioning, and measurement.

The Processor is intended for surround voice-over workflows where a main 5.1 program and two mono voice sources need to be combined and delivered in multiple output formats.

Typical Applications

Typical applications include:

- 5.1 program voice-over mixing
- Up to six voice production workflows
- Mix-minus generation
- Program downmix generation
- Clean mono output generation
- Loudness-controlled surround output processing

Included Modules

The Processor includes the following processing and measurement stages:

- Input Meter
- Input Conditioner
- Voice Over
- Mix-minus
- Downmix
- Matrix Mixer
- Level Magic
- True Peak Limiter
- Output Conditioner
- Output Meter
- Loudness Measurement

The signal flow includes input conditioning for the main 5.1 program and two mono voice sources, voice-over mixing, mix-minus generation, downmixing, clean mono generation, level control, limiting, output conditioning, and measurement.

Channel Formats

This article describes a 5.1 main program version with up to six mono voice inputs.

The Processor uses 5.1, mono, stereo, and clean mono output-related paths.

Block Diagram

[PLACEHOLDER: insert block diagram for Voice Over Mixer - Main 5.1 + 2x Mono]

Notes

[PLACEHOLDER: block diagram required]

Voice Processing Channel Strip

Overview

The Voice Processing Channel Strip is a Processor for processing a single voice signal. It combines input conditioning, voice leveling, phase rotation, de-essing, equalization, dynamics processing, loudness control, output conditioning, and metering in one signal chain.

The Processor is intended for professional voice processing where a consistent, controlled, and broadcast-ready voice signal is required.

Typical Applications

Typical applications include:

- Voice-over and narration processing
- Broadcast voice processing
- Podcast and studio voice processing
- Loudness control for voice recordings
- Dynamic range control for speech signals
- Audio book production

Included Modules

The mono and stereo versions includes the following processing and measurement stages:

- Input Meter
- Input Conditioner
- Voice Leveler
- Phase Rotator
- De-Esser
- Equalizer
- Advanced Dynamics
- Level Magic
- Output Conditioner
- Output Meter

The signal flow is:

Input → Input Meter → Input Conditioner → Voice Leveler → Phase Rotator → De-Esser → Equalizer → Advanced Dynamics → Level Magic → Output Conditioner → Output Meter → Output

Channel Formats

This article describes the mono and stereo version.

The Voice Processing Channel Strip is also available as a stereo version. Dual-mono operation is not supported.