NUGEN Audio

AB Assist — See the forest for the trees

AB Assist is a tool for quickly comparing multiple pieces of audio, for example different takes, mix revisions or plug-in options. Blind tests can be conducted in 'test' mode, with each source labelled randomly.

Comparing multiple audio sources is something that comes up in practically every project or session; deciding between different takes, mic placements, plug-in options, signal chains, or further down the line comparing mix revisions and potential masters. Should be simple, right? But in fact this can be one of the most awkward things to set up in many DAWs, navigating various combinations of “mute” and “solo” in order to hear what you need to hear.

AB Assist can seamlessly compare up to four audio sources, loudness-matched, all within a single window. The plug-in also includes a blind-testing mode, allowing you to compare options without knowing which is which, making decisions with ears rather than expectations.

- **Receive and compare up to four audio sources.**
- **Compare surround sources in any channel-count.**
- **Auto-level-match: match Short-term loudness (LUFS) of sources.**
- **Mono-check: compare mono fold-down.**
- **Smooth fades between sources.**
- **Multiple plug-in instances link automatically.**

Aligner — Perfectly aligned audio in seconds

Aligner is an automatic phase correction tool, designed to speed up the daily workflow of audio engineers. Typically, a recording engineer would use Aligner to phase-align multiple microphones used to record the same audio source.

Manual phase adjustment is boring! Perhaps you had to hurry through a recording session and didn't spend enough time on mic placement, perhaps you're mixing someone else's recording, or perhaps you're tinkering with an old recording from before your ears were as good as they are now. Whatever the situation, no-one wants to spend the first hour of a session manually aligning phase!

Aligner is designed to automate this process and speed up the daily workflow of audio engineers, leaving you more time to focus on the fun stuff.

- User-defined reference track.
- Numeric sample delay.
- Phase-invert (e.g. for top snare/bottom snare).
- Bypass per-channel and Bypass All.

NUGEN Audio

AMB — For one file, or thousands

AMB is a customizable standalone application for faster-than-real-time batch processing of multiple audio files. A license for AMB includes at least one “core” module (Loudness or Upmix) plus whichever extensions the facility requires.

Sometimes you don’t need to meticulously comb through each and every piece of audio and fine-tune settings for each of them. Sometimes you just need to batch process tens, hundreds or even thousands of files through the same chain as quickly as possible!

AMB is the product that has your back. Whether you’re normalizing the Short-term loudness of a whole game’s worth of one-shot dialog phrases, performing a basic upmix in order to play a batch of archive footage with stereo audio through a 5.1 system, or re-versioning an entire season of a TV series to play at a new Integrated loudness for a different broadcast standard, AMB processes large batches of files quickly and consistently.

- Typically at least 100x real-time processing.
- Extensions include DynApt, ProRes, MXF, Dolby E, Enterprise, and Queue/Thread expansions.
- Watch folders and automated data logging.
- Up to 16 watch folders, 16 queues and 16 threads.
- Loudness standards supported:
    - ITU-R B.S. 1770 (revisions 1-4)
    - ATSC A/85 (CALM Act)
    - EBU 128
    - ARIB TR-B32
    - OP-59
    - AGCOM 219/9/CSP
    - Portaria 354
    - DPP
    - Leq(m) (TASA and SAWA)

NUGEN Audio

DialogCheck — Every word counts

DialogCheck is a dialog intelligibility meter, designed to provide an objective measure of speech clarity. The plug-in uses speech recognition, psychoacoustic modelling and dual neural networks to determine the "listening effort" required to understand any speech within a mix.

Poor dialog intelligibility is one of the biggest sources of complaints from film and television audiences, with studies suggesting that almost half of viewers prefer to watch with subtitles. For an audio engineer who might have been working on a scene for hours or even days, hearing the same dialog hundreds of times, it's a challenge to predict how a mix will sound to someone hearing it for the first time.

Many of the factors that negatively impact dialog intelligibility are outside an audio engineer's control. Speech might be in a language that the listener doesn't understand, or might not have been enunciated clearly, the listener may be hearing impaired, or in a poor listening environment. But there are other factors we do have control over; dialog level versus music and background noise, dialog level, the quality of audio signal, and the use of reverb and effects, to name a few. DialogCheck provides an objective value that signposts where there might be room for improvement, or whether it might be time to resort to ADR!

- **Based on Fraunhofer IDMT's LE-Meter.**
- **Supports up to 9.1.6 channels.**
- **Real-time bar meter.**
- **Timecode-locked history graph with optional session offset.**
- **Realtime momentary clarity readout.**
- **User-configurable Upper and Lower Percentile readouts.**

Halo Downmix — Decisive downmixing

Downmixing is the process of collapsing a surround audio mix to play in fewer channels, for example from 7.1.4 to 5.1 or from 5.1 to stereo. Halo Downmix offers more precision than a typical downmixer, with additional controls such as delay compensation, allowing the user to produce better downmixes as well as rebalancing existing surround mixes.

Typical "stock" downmixing doesn't always get the desired results, especially if legacy upmixing practices were used for the source material. Halo Downmix can remove delay artefacts and tame excessive artificial reverb, creating a more coherent end result.

With intuitive controls for re-balancing surround channels and direct versus ambient sound, Halo Downmix is integral for any parallel surround/stereo/immersive workflow. If you are able to spend as much time and care on the downmix as you do the full surround mix, why wouldn't you?

- **Standard plug-in supports up to 7.1 channels.**
- **Optional 3D Immersive Extension adds support for 7.1.2 and 7.1.4 channels.**
- **Downmix surround-to-stereo.**
- **Fine-tune surround-to-surround.**
- **Maintain dialog clarity.**
- **Control excess ambient sound.**
- **Remove delay artefacts.**

NUGEN Audio

Halo Upmix — Full control, intuitive ease

Upmixing is the process of creating a surround or immersive mix from a mix with fewer channels. Halo Upmix produces a phase coherent spatial upmix without adding artificial reverb, chorus or delays, preserving the character of the original audio and ensuring mix integrity for downmixes further downstream.

Halo Upmix is far more than just a “set and forget” automatic upmixer for your full mix. This plug-in’s real strength lies in the ability to layer mix elements, expanded to different extents and pushed out into the surround field in different ways.

Working on a 7.1 mix of a film originally made in stereo? You can create expansive soundscapes by pushing environmental sounds as far as they’ll go across the surround panorama, whilst keeping dialog and important action tight across the LCR channels. Mixing a pop album in Dolby Atmos? You can fill the room with synth pads whilst keeping most other elements as point sources, automating a different instance of Halo Upmix to pull them wider at key musical moments.

- **Standard plug-in supports up to 7.1 channels.**
- **Optional 3D Immersive Extension adds support for 7.1.2, 7.1.4 and 1st, 2nd and 3rd order Ambisonics.**
- **‘Exact’ control for seamless downmixes.**

Halo Vision — Seeing is believing

Halo Vision is a real-time visual analysis suite for monitoring surround and immersive audio. Customizable modules provide engineers with a clearer understanding of their audio, allowing for confident decision making, and troubleshooting problem areas which might be missed using ears alone.

A mix can contain complex information that isn't always easy to decipher, and this is especially true for surround and immersive mixes. The product of a series of interviews with many of our most trusted and respected customers, Halo Vision provides a clear visual reference, helping engineers to identify potential problems.

What makes Halo Vision unique is the ability to display a wide range of information intuitively, without the need for a crib sheet or tons of on-screen clutter. We have built upon features, ideas and inspiration from other products on the market — as well as other highly-regarded plug-ins of our own, including but not limited to Halo Upmix, Paragon and VisLM — in order to create a concise, streamlined analysis suite.

- Views:
    - Correlation Matrix
    - Correlation Web
    - Frequency Haze
    - Location Haze
    - Spectrum
    - True Peak
    - Timecode
- Supports up to 7.1.4 channels.

NUGEN Audio

ISL — Transparent limiting in surround

Most limiters use sample peak limiting, but many situations in both music production and audio post production warrant True Peak limiting, which preempts potential intersample peaks. ISL is a True Peak limiter which can be used to prevent ITU-R B.S. 1770 violations, streaming codec distortion, and more, supporting up to 7.1.4 channels.

In today's landscape there's a good chance that listeners won't hear your actual mix, at least not exactly as you hear it! Whether that's due to streaming codecs or simply the limitations of a physical format like vinyl, one of the hardest things to predict is intersample peaking. Typical sample peak limiters prevent the digital waveform from peaking in your DAW session, but additional processing further downstream can lead to peaks between the samples. Sounds scary, right? But using a True Peak limiter like ISL means you can rest easy.

True Peak limiting is a requirement for practically all broadcast compliance standards worldwide. And with surround support for up to 7.1.4 channels, ISL is a must-have for audio post production workflows.

- **ITU-R B.S. 1770-4 and Apple afclip compliant.**
- **Auto-release dynamically adapts for low frequency content.**
- **Unique steering and ducking meters.**

ISLst — Transparent limiting in stereo

Most limiters use sample peak limiting, but many situations in both music production and audio post production warrant True Peak limiting, which preempts potential intersample peaks. ISLst is a True Peak limiter which can be used to prevent ITU-R B.S. 1770 violations, streaming codec distortion, and more, supporting mono and stereo audio.

In today's landscape there's a good chance that listeners won't hear your actual mix, at least not exactly as you hear it! Whether that's due to streaming codecs or simply the limitations of a physical format like vinyl, one of the hardest things to predict is intersample peaking. Typical sample peak limiters prevent the digital waveform from peaking in your DAW session, but additional processing further downstream can lead to peaks between the samples. Sounds scary, right? But using a True Peak limiter like ISL means you can rest easy.

ISLst is a simplified version of ISL that only supports mono and stereo audio, perfect for users who don't work in surround and immersive formats.

- **ITU-R B.S. 1770-4 and Apple afclip compliant.**
- **Auto-release dynamically adapts for low frequency content.**
- **Unique steering and ducking meters.**

Jotter — Streamline your collaboration

Jotter is a utility plug-in for making timestamped notes and annotations within a DAW, easily exported and shared. Although the plug-in is a paid-for product, the standalone application is free; collaborators can download the free version of Jotter and add notes, which can then be imported directly into the plug-in.

In a world of increasingly fractured workflows, we often aren't in the same room or even the same city as our colleagues and collaborators. So how best to share comments and feedback in a timely and straightforward fashion? Jotter creates timestamped notes, saved as part of the DAW session, or easily imported and exported between sessions. And for members of the team who might not ever work within a DAW, the standalone application allows timestamped annotations using only a WAV file.

No more tracking feedback across physical notepads, emails, text messages, shared spreadsheets and countless other places. No more ambiguity over exactly which moment a comment refers to (is the "bridge" a pre-chorus or a middle eight?).

- Exports as CSV.
- Full waveform and macro waveform views.
- "To-do" checkboxes.
- Custom colour allocation.

NUGEN Audio

LM-Correct — Immediate quick-fix loudness compliance

LM-Correct is a quick-fix tool for loudness compliance, available as a standalone application or via AudioSuite. Faster than real time, LM-Correct adjusts audio to hit user-defined targets for Integrated loudness, Max True Peak, Max Short-term loudness and Max Momentary loudness. The optional DynApt extension adds the ability to make dynamic changes, including a Max LRA target.

For a mix that feels “finished” but sits just outside spec, or a project mixed to one loudness standard that now requires an additional alternative, LM-Correct is an immediate, hassle-free route to compliant audio. Avoid repeated iterative mix adjustments — simply enter the target values and let LM-Correct handle the rest.

The DynApt extension is perfect for more complex specifications, using dynamic loudness adjustments to correct for Integrated loudness, Short-term/Momentary loudness and LRA (loudness range). In order to remain transparent, DynApt recognises and maintains deliberate dynamic transitions, including scene-changes.

- Up to 7.1.4 channels.
- Loudness standards supported:
  - ITU-R B.S. 1770 (revisions 1-4)
  - ATSC A/85 (CALM Act)
  - EBU 128
  - ARIB TR-B32
  - OP-59
  - AGCOM 219/9/CSP
  - Portaria 354
  - Leq(m) (TASA)

NUGEN Audio

MasterCheck — One master, every platform, verified

MasterCheck is an optimisation plug-in for cross-platform mastering, designed to ensure that masters sound consistent across all platforms — physical, digital and streaming. Audition how a track will be affected by loudness normalisation and audio codecs, integrated with industry standard loudness and True Peak measurement.

How does your music sound when it reaches the listener? A track might sound great in the studio but will behave differently depending on the listening environment. You already optimise masters for domestic hi-fi systems, car stereos, headphones and earbuds. In the streaming era, shouldn't you optimise for each streaming platform too?

To some extent, loudness and cross-platform optimisation is primarily a concern for the mastering engineer, not the mixer, artist or producer. But if a mix is too hot or over-compressed then there is nowhere further to go! So it makes sense to reference Integrated loudness, PLR, PSR and True Peak even at the mixing stage. Plus, MasterCheck can be a great demonstration tool for those clients who might still believe that "louder is better" no matter what.

- **Using Codec mode and Offset to match, audition data compression and loudness normalisation for:**
  - **Spotify**
  - **Apple Music**
  - **Tidal**
  - **YouTube**
  - **Soundcloud**
  - **Bandcamp**
  - **Qobuz**
  - **Pandora**
- **Supports up to 7.1.4 channels.**

Monofilter — Essential bass management

Monofilter is a bass management plug-in that takes a stereo audio signal and sums the low frequencies to mono. This helps to optimise low-frequency content for louder playback, enables safer mastering, and creates a "tighter" and more defined-sounding mix.

When cutting vinyl for mastering, wide bass frequencies require deeper grooves than normal which can lead to mechanical difficulties during playback. In some extreme cases this may cause the needle to jump. If a track is intended for vinyl release then summing low frequencies to mono is the safest solution.

For music which is likely to be played out in public, many club PA systems are either set up entirely in mono or set up with the subs in mono, but you cannot necessarily rely on this! If the whole system is in stereo, wide bass frequencies can create unwanted "wobbling" effects as someone moves through the space. Again, this can be resolved by summing the bass to mono.

- Stereo width envelope.
- HPF cut-off.
- Individual M and S trim controls.
- Stereo spectrum analysis.
- Phase correlation control.
- Zero Latency mode.
- Linear Phase mode.
- HQ mode.

Monofilter Elements — Bass management made simple

Monofilter Elements is a bass management plug-in that takes a stereo audio signal and sums the low frequencies to mono. This helps to optimise low-frequency content for louder playback, enables safer mastering, and creates a "tighter" and more defined-sounding mix. This plug-in is a simplified version of Monofilter, with fewer, simpler controls.

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For music which is likely to be played out in public, many club PA systems are either set up entirely in mono or set up with the subs in mono, but you cannot necessarily rely on this! If the whole system is in stereo, wide bass frequencies can create unwanted "wobbling" effects as someone moves through the space. Again, this can be resolved by summing the bass to mono.

- Single 'Monofilter' control.
- Bass trim control.
- HPF cut-off.

Paragon — Boundless reverb in surround

Paragon is a reverb plug-in designed for surround and immersive mixes, supporting up to 7.1.4 channels, making it suitable for mixing in Dolby Atmos. It uses state-of-the-art re-synthesis technology to combine true convolution reverb with the flexibility of an algorithmic reverb.

Broadly speaking there are two types of reverb plug-in: convolution and algorithmic. Convolution reverb uses recordings of real spaces, known as Impulse Responses, whereas algorithmic reverb is generated by an algorithm. Convolution reverb typically sounds more realistic, but is limited by the sound of the space where an Impulse Response was recorded. Algorithmic reverb tends to sound less realistic, but has more scope for control over the reverb parameters. If a convolution reverb plug-in is able to adjust parameters like room size and decay time, this usually requires timestretching, which can cause audible artefacts.

Paragon uses technology developed at the University of York to analyse, decompose and re-synthesise Impulse Responses to create new authentic spaces, without timestretching artefacts. The end result is detailed, natural-sounding reverb with unprecedented tweakability to cater to the needs of an immersive mix.

- True convolution reverb.
- Frequency-dependent decay rate.
- Individually configurable crosstalk per channel.
- HPF and LPF per channel.
- Spectral analysis.
- Switchable LFE.
- Compact mode.

NUGEN Audio

Paragon ST — Boundless reverb in stereo

Paragon is a reverb plug-in designed for stereo mixes. It uses state-of-the-art re-synthesis technology to combine true convolution reverb with the flexibility of an algorithmic reverb.

Broadly speaking there are two types of reverb plug-in: convolution and algorithmic. Convolution reverb uses recordings of real spaces, known as Impulse Responses, whereas algorithmic reverb is generated by an algorithm. Convolution reverb typically sounds more realistic, but is limited by the sound of the space where an Impulse Response was recorded. Algorithmic reverb tends to sound less realistic, but has more scope for control over the reverb parameters. If a convolution reverb plug-in is able to adjust parameters like room size and decay time, this usually requires timestretching, which can cause audible artefacts.

Paragon uses technology developed at the University of York to analyse, decompose and re-synthesise Impulse Responses to create new authentic spaces, without timestretching artefacts. The end result is detailed, natural-sounding reverb with unprecedented tweakability to cater to the needs of your mix.

- True convolution reverb.
- Frequency-dependent decay rate.
- Configurable crosstalk.
- HPF and LPF.
- Spectral analysis.
- Compact mode.

SEQ-S — Powerful sonic sculpting in surround

SEQ-S is a surround EQ supporting up to 7.1.4 channels with "match" and "morph" functionality. The plug-in uses a direct-draw interface with three independent curves that can be assigned to specific channels.

With three assignable EQ curves and precise linear phase, SEQ-S is a powerful solution for surgical sonic sculpting, and its secret weapon is the "match" function. Say you're recording ADR and struggling to match the frequency response of the original take — set the original as your reference track and analyse to create a perfectly matched EQ curve. Say your voiceover is fighting against a music bed — set the voiceover as your reference then invert the curve and apply it to the bed in order to carve out perfect space.

SEQ-S is also capable of curve-morphing; transitions between two EQ curves that can be either manually automated or locked to the project tempo. Try the "morph" function to transform the sonic environment, either for controlled scene transitions or for extreme mindbending sound design.

- Match EQ.
- Curve reset.
- EQ cut, copy and paste between instances.
- Curve invert.
- EQ banding:
    - Thirds
    - Sixths
    - Chromatic
    - Bark
    - Mel
    - Bark Thirds

SEQ-ST — Powerful sonic sculpting in stereo

SEQ-ST is a mono and stereo EQ with “match” and “morph” functionality. The plug-in uses a direct-draw interface with three independent curves that can be assigned to specific channels, including mid/side.

With three assignable EQ curves and precise linear phase, SEQ-ST is a powerful solution for surgical sonic sculpting, and its secret weapon is the “match” function. Say you’re struggling to match a guitar tone from a previous session — set the original as your reference track and analyse to create a perfectly matched EQ curve. Say your vocal is fighting to be heard — set the vocal as your reference then invert the curve and apply it to the instrument bus in order to carve out perfect space. Match EQ can even be used on the master bus to match the frequency profile of a mix reference track!

SEQ-ST is also capable of curve-morphing; transitions between two EQ curves that can be either manually automated or locked to the project tempo. Try the “morph” function to transform the sonic environment, either for controlled scene transitions or for extreme mindbending sound design.

- Match EQ.
- Curve reset.
- EQ cut, copy and paste between instances.
- Curve invert.
- EQ banding:
    - Thirds
    - Sixths
    - Chromatic
    - Bark
    - Mel
    - Bark Thirds

NUGEN Audio

SigMod — Power up your plug-ins

SigMod is a modular plug-in for building custom signal architecture. 13 single-process modules allow routing options that many DAWs do not offer natively.

Combine modules in any order to open up unprecedented routing possibilities. Use the Tap function to set up an aux or FX send midway through the signal chain, not just pre- or post-. Add missing mid/side functionality to any stereo or multi-mono plug-in with the Mid/side module. Host VST and AU plug-ins in a DAW that doesn't support them natively using Insert. Save your ears and speakers from sudden volume jumps with Protect.

SigMod also caters to more basic functionality. Depending on your DAW a simple mono-check, left/right flip or phase invert might not be easily accessible. SigMod has modules for each of these, all in one place.

- Mid/side
- Protect
- Tap
- Crossover
- Insert
- Mute/solo
- Split
- Trim
- Switch
- DC offset
- Mono
- Phase
- Delay

Stereoizer — Expand your stereo width

Stereoizer is a stereo width enhancement plug-in that uses psychoacoustic principles to go beyond traditional linear width. This enables the user to create a pseudo stereo signal from a mono source, without introducing phase issues.

It can be tough to create a dynamic, exciting mix when many elements of the track were recorded in mono, especially if the song has a sparse arrangement. There's a limit to the number of times you can pan a guitar left and its accompanying reverb send to the right! Stereoizer uses three independent algorithms — IID (Interaural Intensity Difference), ITD (Interaural Time Difference) and Linear Width — for maximum flexibility. IID and ITD are based on how the human ear perceives width, and can create a stereo signal even from a mono source, whereas Linear Width behaves more like a typical stereo width plug-in. Together, these three modes can create everything from subtle symmetry to huge stereo panorama.

The frequency-specific controls in Stereoizer allow the user to focus width expansion on specific frequency bands, sculpting the mix and adding shine and polish without resorting to unnecessary extreme EQ. Crucially, Stereoizer maintains mono compatibility and phase coherence throughout, with a 'Collapse output to mono' audition mode for added peace of mind.

- **IID and ITD algorithm on/off.**
- **Upper and lower frequency bounds.**
- **Resolution, focus, acuity, fill and invert controls.**
- **Dynamic modulation with tempo lock.**

Stereoizer Elements — Stereo width made simple

Stereoizer Elements is a stereo width enhancement plug-in that uses psychoacoustic principles to go beyond traditional linear width. This plug-in is a simplified version of Stereoizer, with fewer, simpler controls.

It can be tough to create a dynamic, exciting mix when many elements of the track were recorded in mono, especially if the song has a sparse arrangement. There's a limit to the number of times you can pan a guitar left and its accompanying reverb send to the right! Stereoizer Elements combines IID (Interaural Intensity Difference), ITD (Interaural Time Difference) and Linear Width, all linked with one control. IID and ITD are based on how the human ear perceives width, and can create a stereo signal even from a mono source.

Crucially, Stereoizer Elements maintains mono compatibility and phase coherence, with a Mono audition mode for added peace of mind.

- Single 'Stereoizer' control.
- Stereo balance control.
- Gain trim +/-20dB.

NUGEN Audio

Stereoplacer — Next-generation pan control

Stereoplacer is a frequency-specific panning tool. It functions like an EQ, but instead of making a frequency band louder or quieter, it pans that frequency band left or right without affecting the rest of the stereo image.

Stereoplacer is most commonly used to restore the stereo image from a mono recording. This could be anything from archive material that was only ever captured in mono to new recordings limited by the number of inputs in a given session. Pan lower notes to the left and higher notes to the right in order to create a realistic stereo image from a mono piano recording, for example. Stereoplacer is also indispensable when building a track from older samples, taming drum loops with wacky panning by nudging fundamental frequencies back towards the centre.

Just like an EQ, Stereoplacer uses bell curves and high and low shelves to sculpt the audio, with up to 10 bands. And the switchable harmonic control captures linked harmonics and overtones for more natural sounding panning.

- **Restore archive recordings.**
- **Fix poor microphone placement.**
- **Pan individual sounds from sample loops.**

NUGEN Audio

Stereoplacer Elements — Next-gen pan control, streamlined

Stereoplacer Elements is a frequency-specific panning tool. This plug-in is a simplified version of Stereoplacer, with fewer, simpler controls.

Stereoplacer Elements is most commonly used to restore the stereo image from a mono recording. This could be anything from archive material that was only ever captured in mono to new recordings limited by the number of inputs in a given session. Pan lower notes to the left and higher notes to the right in order to create a realistic stereo image from a mono piano recording, for example. Stereoplacer Elements is also indispensable when building a track from older samples, taming drum loops with wacky panning by nudging fundamental frequencies back towards the centre.

Just like an EQ, Stereoplacer Elements uses bell curves and high and low shelves to sculpt the audio, with a maximum of two bands per plug-in instance.

- **Restore archive recordings.**
- **Fix poor microphone placement.**
- **Pan individual sounds from sample loops.**

NUGEN Audio

VisLM — Time to take loudness seriously

VisLM is a timecode-locked loudness meter with a history function. Suitable for measuring ITU-R BS. 1770, EBU R128, ATSC A/85, Leq(m) and Leq(a), Netflix Best Practice and more.

VisLM ReMEM remembers up to 24 hours of loudness information, locked to the timecode received directly from your DAW. Loudness overdub means that as soon as any mix tweak is played through the meter, VisLM updates Integrated measurements accordingly, removing the need for end-to-end loudness measurement. VisLM can also measure faster-than-real-time via AudioSuite, and data can be transferred from AudioSuite to a real-time plug-in instance.

The level meter displays True Peak values as defined in ITU-R BS.1770, accurately pre-empting and measuring inter-sample peaks in order to avoid potential codec distortion. VisLM's history view automatically logs True Peak violations as flags, which are navigable alongside custom user-defined flags.

- **Up to 7.1.4 channels.**
- **Loudness standards supported:**
    - **ITU-R B.S. 1770 (revisions 1-4)**
    - **ATSC A/85 (CALM Act)**
    - **EBU 128**
    - **Netflix**
    - **ARIB TR-B32**
    - **OP-59**
    - **AGCOM 219/9/CSP**
    - **Portaria 354**
    - **DPP**
    - **Leq(m) (TASA and SAWA)**
    - **Leq(a)**

NUGEN Audio

Visualizer — See what your ears can't

Visualizer is a real-time visual analysis suite for monitoring stereo audio. Customizable modules provide engineers with a clearer understanding of their audio, allowing for confident decision making, and troubleshooting problem areas which might be missed using ears alone.

A mix can contain complex information that isn't always easy to decipher; Visualizer provides an indispensable set of visual audio analysis tools. All views are highly customisable, each with an individual settings panel.

Visualizer can be used to set input levels, check phase correlation, and identify unwanted noise and problem frequencies.

- Views:
    - Level Meter
    - Spectrum Analyser
    - Stereo Spectrum Analyser
    - Spectrogram
    - Stereo Spectrogram
    - Vectorscope
    - Correlation by Frequency
    - Correlation Meter