FEATURES & TECHNICAL DETAILS

HLConvolver 2.16.0 for macOS

Multichannel FIR | Live PCM meters | Native DSD & WavPack

HIGH-PERFORMANCE FIR CONVOLUTION | UNIVERSAL 2 FOR APPLE SILICON & INTEL

HLConvolver is a high-performance FIR convolution application and plug-in suite for stereo and multichannel audio. Part of the Hang Loose DSP Software Suite, it supports both Apple Silicon and Intel Macs.

One convolver - local or networked audio

CoreAudio → CoreAudio	CoreAudio → UPnP
UPnP → CoreAudio	UPnP → UPnP

Route audio from a Mac player via a virtual CoreAudio device such as BlackHole, or receive audio from a UPnP source. Send the convolved result to local audio hardware or a compatible network renderer. BlackHole is not included.

Software formats and interfaces

macOS: Standalone, Audio Unit (AUv2) and VST3 2.16.0, with native arm64 and x86_64 code. **Separate Suite components:** AAX 1.2.6 and HLHost 2.1.6.

Native UI, local browser-based Web UI (http://127.0.0.1:8081 for Standalone), optional headless operation and macOS Background Service. Avoid competing access to audio resources.

Highlights in 2.16.0

Live monitoring: Web input and processed output PCM meters with per-channel RMS, sample peak and held peak. See the 12-channel display on page 2.

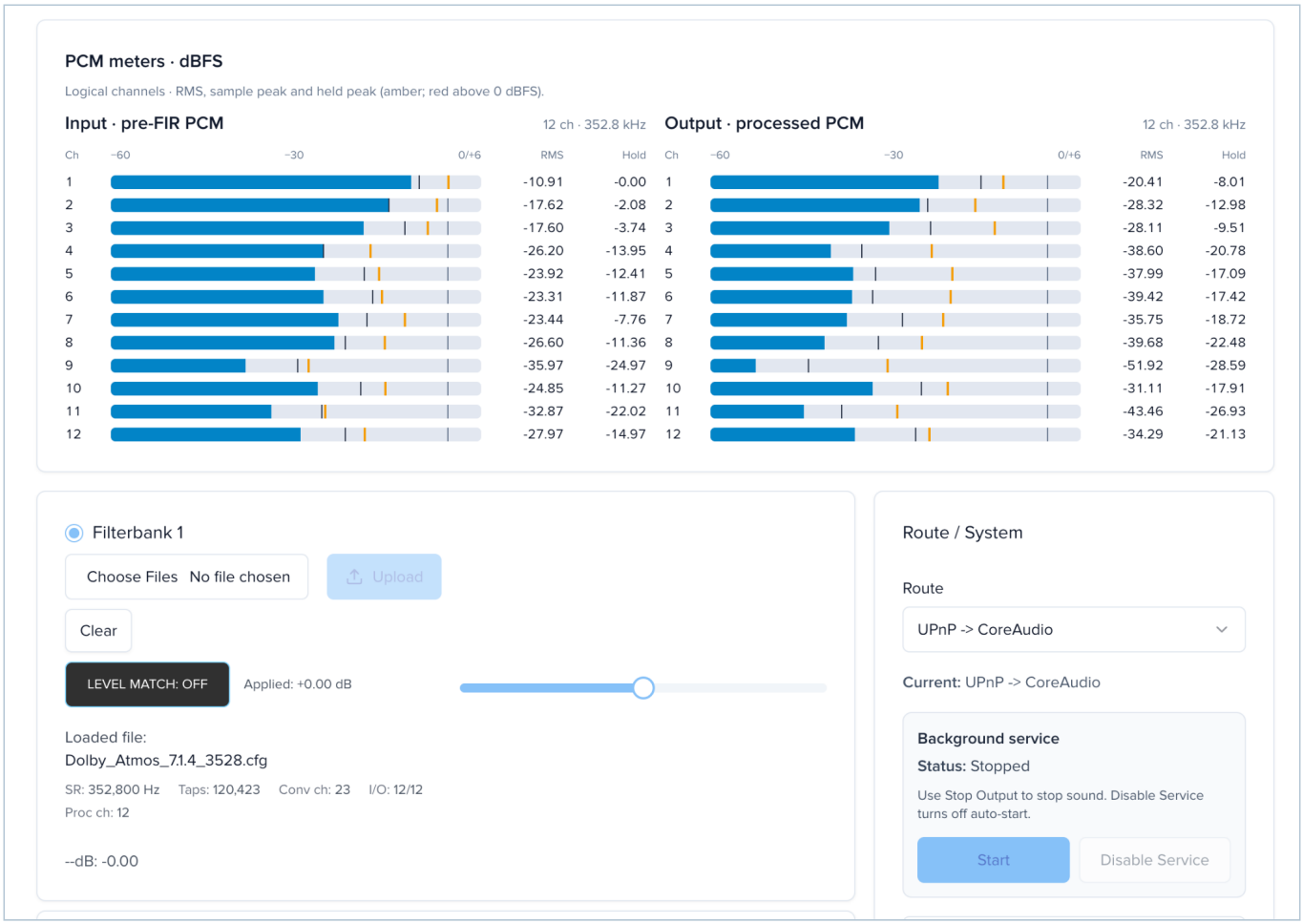
Level Match: Restored independent filterbank controls in Native and Web UIs, with explicit ON/OFF and applied gain readouts. Manual gain and Master Trim remain separate.

Presentation: Clearer zero-gain values and improved physical-output selection controls.

Actual media-format, device, host and controller compatibility depends on the end-to-end path. See technical scope on pages 3–4.

Live PCM monitoring and Level Match

The Web UI shows input PCM before FIR processing alongside processed output PCM. The same view also presents each filterbank's Level Match state and applied gain. Metering follows available processor measurements rather than inferring physical audio-device output.



2.16.0 Web UI: 12-channel PCM input and output at 352.8 kHz, with Filterbank 1 Level Match controls shown below.

Reading the meters

Blue bars: RMS level in dBFS, with visual smoothing. **Thin markers:** measured sample peaks. **Amber markers:** held sample peaks; red indicates a hold peak above 0 dBFS. Numeric RMS and hold-peak values accompany each channel. When no current measurement is available, the panel indicates this rather than displaying old readings.

Independent Level Match controls

Each filterbank has a saved Level Match ON/OFF state, with applied gain displayed in dB. The native and Web interfaces control the same parameter. Level Match is separate from manual filterbank gain and overall Master Trim.

Audio formats, routing and transport

Standalone UPnP input and output

PCM and decoded formats	DSD / DoP formats
WAV, AIFF/AIFC, FLAC	Sony DSF; uncompressed DSDIFF/DFF
Lossless integer PCM; supported float WavPack	DoP-over-WAV with verified markers

Native DSD input supports DSD64, DSD128, DSD256 and DSD512 in the 44.1-kHz family. DSD is decoded to 352.8 kHz planar floating-point PCM before speaker routing and FIR processing. Ordinary PCM WAV is not classified as DoP without coherent markers. DST-compressed DFF, DSD1024, lossy/hybrid WavPack and WavPack DSD are outside this input scope.

UPnP output supports selectable 16-, 24- or 32-bit PCM, with separate configurable DSD/DoP output on supported UPnP-output routes. Physical CoreAudio output and host plug-in audio use PCM. Input support does not imply native DSD passthrough in every topology.

Multichannel speaker-position routing

Trusted source positions can be normalized into a fixed 7.1.4 system bus. This is routing, not upmixing: unused positions remain silent. Auto uses authoritative metadata; an explicit music-layout assumption is available for documented layouts when metadata is insufficient.

1 L	2 R	3 C	4 LFE
5 Ls	6 Rs	7 Lrs	8 Rrs
9 Ltf	10 Rtf	11 Ltr	12 Rtr

For example, 5.1 with rear surrounds maps to positions 1–4, 7 and 8; authoritative 5.0 Sony DSF maps to 1–3, 7 and 8. A 7.1.4 source with matching order maps directly. The Web UI shows the active source layout and effective map. Validated physical CoreAudio masks can select channels other than the first N.

Automatic sample-rate following and UPnP control

Standalone follows valid CoreAudio source-rate changes subject to device capabilities and FIR readiness. Local output coordinates physical DAC reopen, output-clock readiness, ASRC and buffering; network output does not require an unrelated local DAC to reopen.

Compatible renderer discovery, versioned service types, SetURI/Play and supported queued-track controls are available. UPnP input advertises supported FLAC, AIFF, DSF and uncompressed DFF. Seeking is supported for compatible PCM streams and gapless playback for compatible same-format queued PCM on UPnP-to-CoreAudio. Renderer buffering, network conditions and format transitions can affect audible starts and track boundaries; seamless playback is not guaranteed across every route or endpoint.

FIR processing, diagnostics and deployment

Long FIRs, multiple filterbanks and rate-specific configurations are supported. Bundled r8brain can resample filters when a matching-rate FIR is unavailable, including direct stereo WAV FIRs. Filterbanks have independent transition-band, stopband and phase settings. FIR latency is reported in samples and milliseconds; plug-ins report applicable latency to the host.

Zero: Uniform partitioning; zero-latency convolution setting. **4096 / 8192 / 16384:** Non-uniform convolution; trades initial latency for lower sustained processing load.

Historical measured examples (macOS 2.12, M1 Mac mini): Stereo 65,536-tap FIR at 48 kHz, resampled to about 963,000 taps/channel at 705.6 kHz: average DSP load 65.0% (Zero) vs 29.9% (16384), about 54% lower. For 12-in/12-out at 352.8 kHz with 23 active paths and over 11 million resampled coefficients, Zero reached 132% and did not sustain clean playback; 8192 and 16384 achieved clean playback at 93.6% and 77.5%. These are historical measurements, not 2.16.0 benchmarks or universal limits.

Diagnostics and disk safeguards

Managed diagnostic logs use 8 MiB segments, two rotations per active log and a 500 MiB rolling budget for recognized managed logs. The Support ZIP includes a manifest and included/omitted counts. Default log directory: ~/Library/Logs/HLConvolver.

The budget does not cover every custom, historical or unrelated file. Earlier oversized logs may require safe manual cleanup with all audio processes stopped. See the companion README. Share Support ZIPs privately only; they may contain local paths and network details.

Installing or updating

Quit HLConvolver and plug-in hosts; stop the Background Service. Open the signed and notarized Suite DMG when supplied and run HLDSPSuite.pkg. Confirm the installed version, route, audio devices, channel assignments and FIR settings before playback. Use a safe volume, particularly with active crossovers or separate driver amplifiers.

Trial, license and documentation

A 14-day trial and perpetual six-activation license with free updates are available from Accurate Sound. No subscription. Confirm current licensing and downloads at the official website.

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