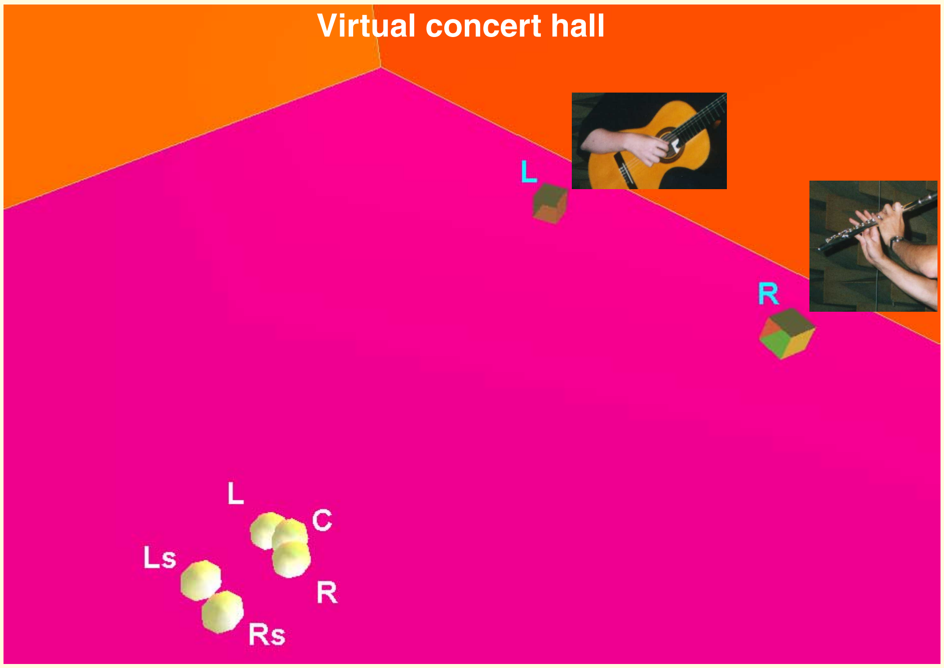
PureVerb™ MultiVolver™

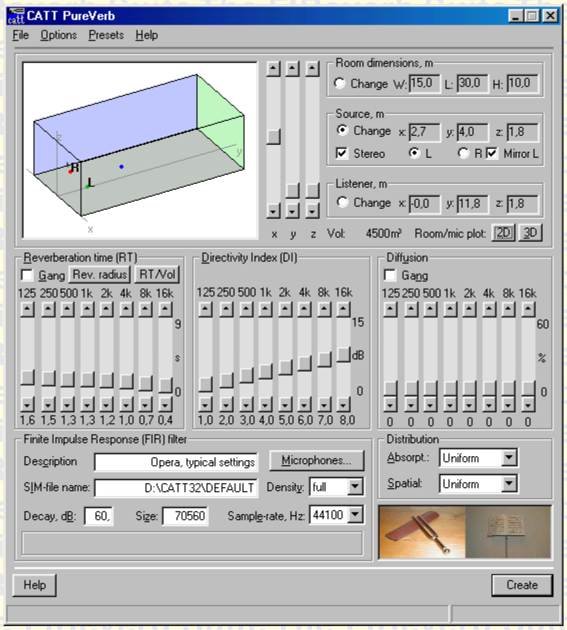
The FIReverb Suite™
3rd Edition



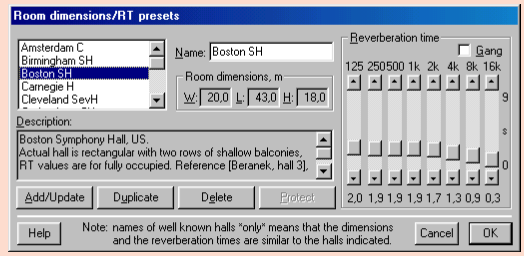
The FIReverb Suite™ consists of two applications for 32-bit *Windows*®. *PureVerb*™ creates natural-sounding, high reflection-density Finite Impulse Responses (FIRs) that imitate real room responses (in mono, stereo, AB-stereo, 5-channel or B-format at 16, 44.1, 48, 88.2 or 96 kHz samplerate and with mono or stereo sources) and *MultiVolver*™, an off-line 8x8 multi-channel convolver, that convolves created FIRs with dry music or performs an asortment of general convolution/filtering tasks such as 5-channel to binaural down-mix. Also a *Lake*[1] *Huron*™ or *CP4*™ processor can be used (*PureVerb*™ creates all files required for *Lake AniScape*™).

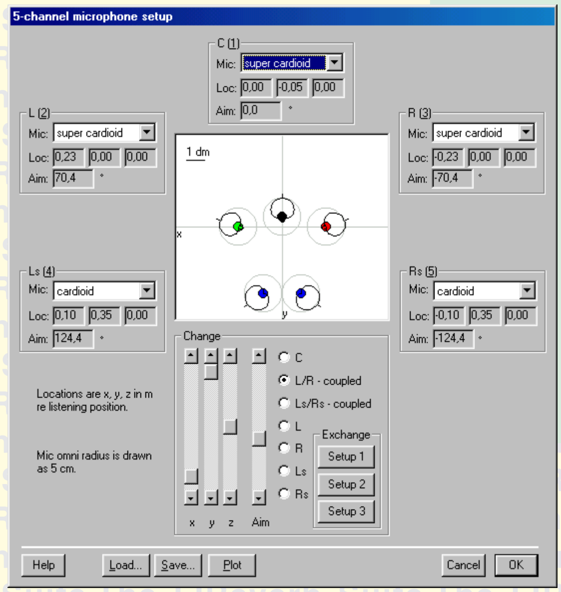
PureVerb™

Natural-sounding FIR Reverberation for Studio and VR

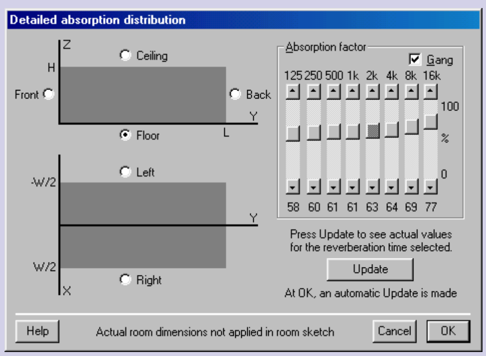


Generic Halls selection, an aid to create realistic reverberation settings by using the reverberation times, volume and overall shape of actual concert halls.

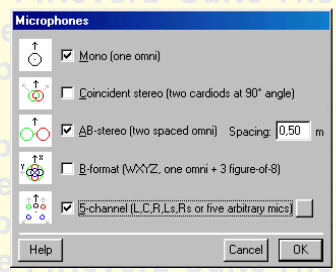


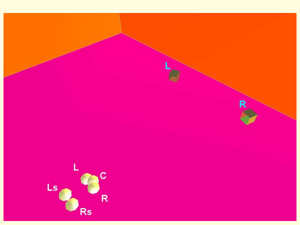


5-channel microphone setup. Microphones can be individually placed and aimed and be of types omni, cardioid, hyper-cardioid, super-cardioid, figure-of-8 or a custom single-sided figure-of-8 (positive side).



Absorption distribution "detailed", other distributions are "uniform" and "audience". The reflection incidence distribution can be set to "uniform" or "lateral".

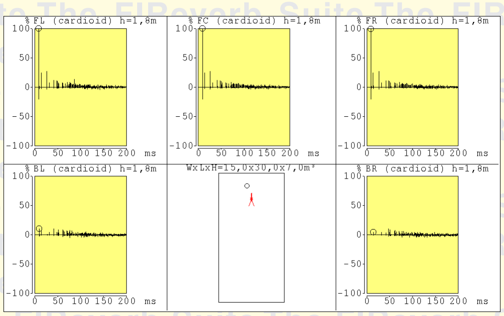




3D-view of a 5-channel mic set-up (shown as spheres) and two sources placed in the virtual concert hall. All combinations of source to listener/mic reflections are calculated (2x5=10 FIR filters for a stereo to 5-channel upmix)

Why not just use filters measured from real concert halls?

Of course, using measured filters should be the optimum but it is also associated with inflexibility since after the filters are measured they cannot be freely changed to suit a particular room or recording situation. With *PureVerb* there are many more degrees of freedom and responses can be tailored to specific needs. In addition, measuring filters from real rooms may introduce noise and properties of the measuring loudspeaker (directivity, frequency response, distortion) are difficult to remove. A further complication is that real room responses often have to be measured high above the seats to give a clean direct sound, in contrast the *PureVerb* direct sound is always perfectly flat and clean (the direct sound can also be excluded from the filter and the unprocessed sound bypassed and mixed in). Some editing software and hardware that can handle FIR filters and long convolution allows the filters to be modified, e.g. dry-out, wet-out, changed initial delay, that typically will result in unrealistic filters since in real rooms decay rate, reflection density, volume and initial delays are tightly coupled. *PureVerb* instead makes it possible to create new filters that have realistic reflection densities and relationships based on either *Generic Halls* presets or suggested values taken from optimum reverberation time to volume ratios. The reverberation radius of the room is used to suggest mic/listener distance.

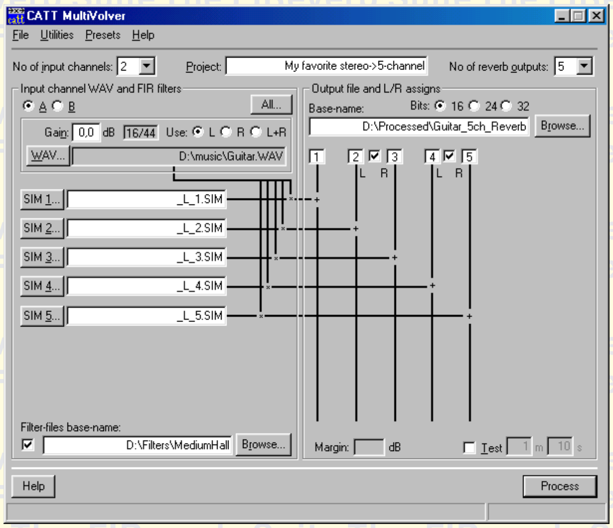


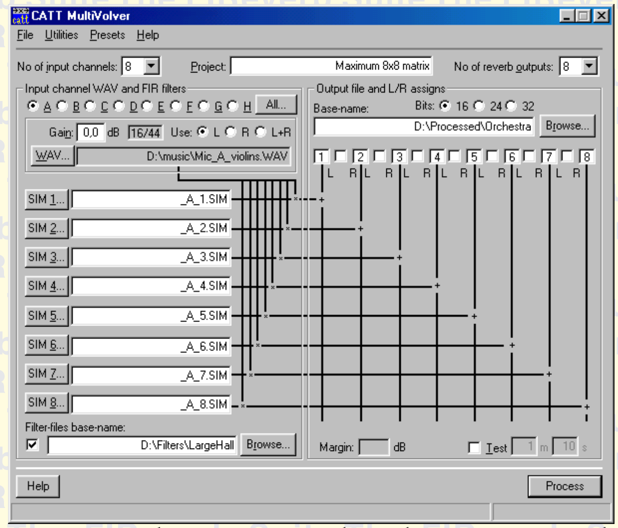
Sample 5-channel responses for the left source (first 200 ms shown).

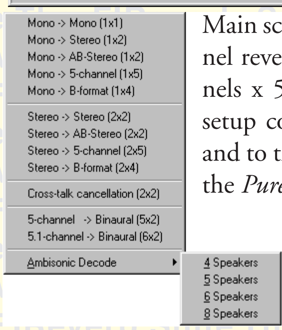
Preset options

MultiVolver™

An 8 x 8 channel off-line reverb/mix convolver



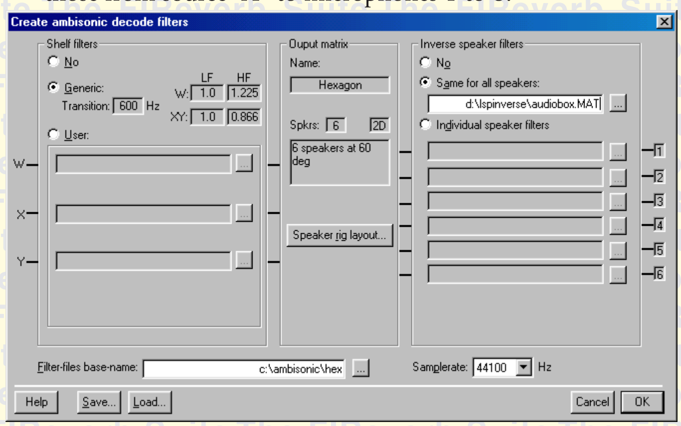




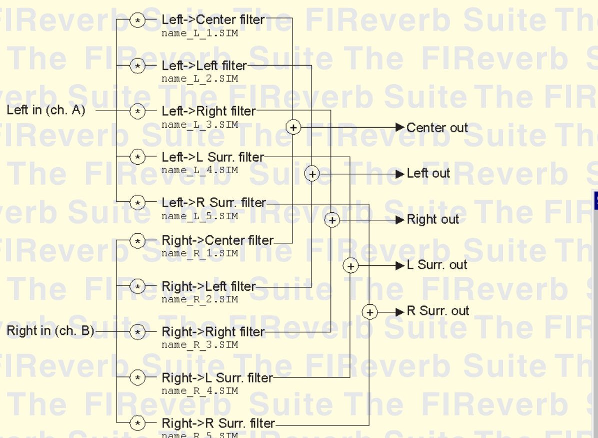
Typical setups/presets

Main screen showing a stereo to 5-channel reverb/up-mix setup (2 input channels x 5 output reverb channels). This setup corresponds to the figures below and to the virtual recording 3D-view on the *PureVerb* page.

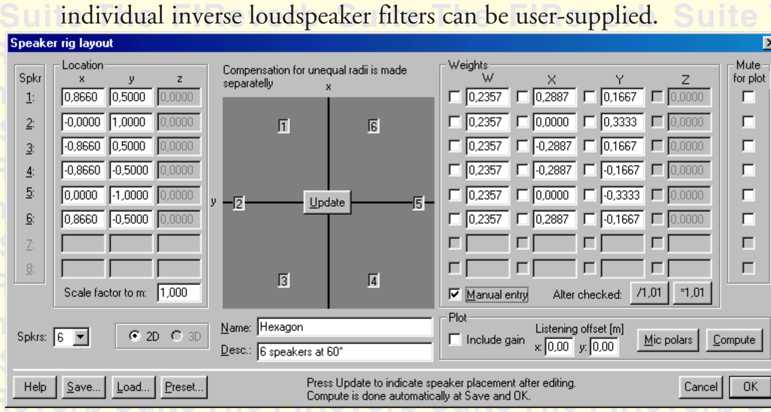
Main screen showing 8 input channels x 8 output reverb channels corresponding e.g. to a reverb/re-mix of 8 dry (or close recorded) instruments to e.g. 7.1. Selected filters are those from source "A" to microphones 1 to 8.



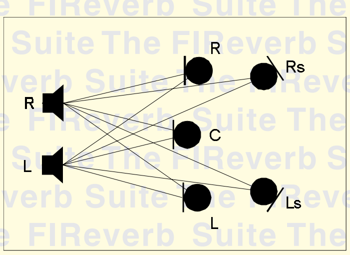
Ambisonic decode FIR filter creation dialog. Shelf and global or individual inverse loudspeaker filters can be user-supplied.



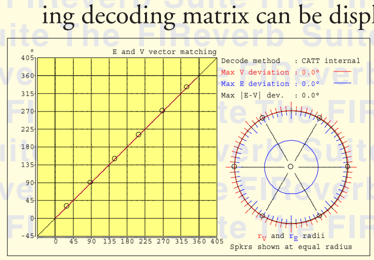
Signal flow for stereo to 5-channel reverb/up-mix.

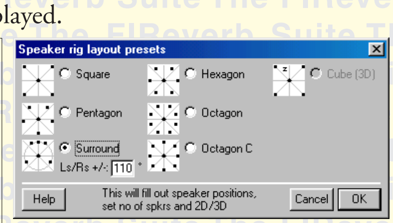


Speaker rig layout dialog where also the properties of the resulting decoding matrix can be displayed.



Schematic picture of all combinations for stereo to 5-channel reverb/up-mix.





Presets for common rig layouts

How is the FIR reverberation created?

A basic room shape is first defined as a rectangular 3D outline. Within this shape is placed a mono or a stereo source plus a listener (or virtual microphone) position. The desired reverberation time is given in octave-bands 125 Hz - 16 kHz and a number of other parameters are selected such as diffusion, absorption distribution and reflection incidence distribution. From this information, a reflection generation process begins, based on how sound is reflected in a real room.

Why is this different from 'normal' reverberation units/plugins?

The difference between pure FIR reverberation, as created by *PureVerb*, and the various combinations of FIR and IIR (Infinite Impulse Response) reverberation algorithms employed by most reverberation units/plugins is that a pure FIR filter has a much higher potential to imitate the reverberation of a real room such as a concert hall. The very high, and with time increasing, reflection density in natural reverberation tails can be created avoiding the granularity apparent in many reverberation units/plugins. The quality of the tail is most apparent with impulsive sounds. If properly done, a synthesized FIR can be equivalent to one measured directly in a room (but noise-free and with a clean direct sound). Since in *PureVerb* reflections are literally created "one by one", and do not get lost in a feedback loop, the spatial location of each individual reflection can be controlled in detail allowing for spatial response properties virtually impossible with other techniques. *PureVerb* is used e.g. by *Deutsche Grammophon* to enhance too dry recordings.

How does this relate to auralization?

For auralization based on models of actual rooms, such as made in *CATT-Acoustic*[2], detailed information on room geometry and wall materials is required and the frequency range is often limited to the octave-bands 125 Hz to 4 (or 8) kHz, simply because of lack of available material data for the higher octaves (*CATT-Acoustic*[2] extrapolates the higher octaves for auralization). This approach is based on prediction since it ***analyzes*** a room and the resulting response will sound according to the design of the room. The approach used in *PureVerb*, however, is the reverse since based only on a few parameters, like the gross dimensions and some octave-band dependent quantities (reverberation time, individual "wall" absorption (optional), "wall" diffusion · source directivity index), a well-sounding response is ***synthesized***. The frequency control available is for the eight octave-bands 125 kHz to 16 kHz and rather than tweaking a number of parameters to make an IIR reverberation sound natural, *PureVerb* room response creation is based on how a real room behaves and, unless very unrealistic input parameters are selected, will always create an impulse response that sounds like a room (be it a small hall or a church). For special effects, less than full reflection density can be selected and the reverberation tail will have echoes. These echoes will, however, still sound like if they come from a real room rather than like a synthetic reverberation.

Virtual Microphones

Responses can be created using a variety of microphones: Mono (using omni or cardioid patterns) · Coincident stereo (using cardioid patterns) · AB stereo (using omni-directionals) · 5-channel formats (using various types of patterns and placements/aims ; virtually any microphone combination possible ; sets of microphone configurations can be saved and reused) · B-format (WXYZ) for Ambisonic and VR

Responses created

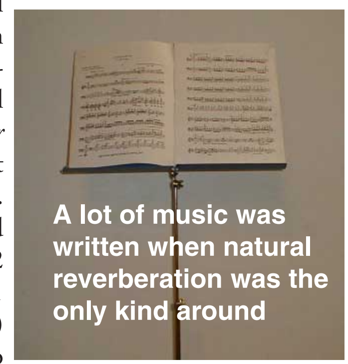
Responses created are full frequency-range at 44.1 or 48 kHz samplerates and are flat below the lower limit of the 125 Hz octave-band. For *Lake*[1] *AniScape* a 16 kHz mode is included but in that case full frequency-range direct sound and first-order reflections are created by *AniScape* in real time for moving sources and receivers so that actually only the tail is run at the lower samplerate. If 88.2 or 96 kHz samplerates are requested, the responses are created using 44.1 or 48 kHz internally and sample-rate conversion is made when creating the final FIRs (in 32-bit *Lake*[1] SIM-file format). It makes little sense to do the actual reverberation at the highest samplerates due to the fast increase of wall and air absorption as a function of frequency. The direct path can be selected to be excluded from the filter and by bypassing and mixing while applying the filter for full benefit of e.g. a 96 kHz samplerate.

Handling

Settings selected are saved in a settings-file and all created filter files will have names based on the settings-file name so that the origins of a filter set can be traced. A plain text version of these settings are also saved as documentation. Batch processing of a list of settings-files can be performed.

Convolution

Since the filters created are FIRs a convolver is required for the processing. The two most natural choices of convolvers are the *MultiVolver* that can perform multi-channel off-line convolutions, a *Lake*[1] *Huron* or a *Lake CP4*. With a well equipped *Huron* convolution processor, high-quality real time stereo to 5-channel upmix can be made from the ten FIR filters created by a stereo source to 5-channel setting in *PureVerb*. The *MultiVolver* can perform off-line convolution at around 8x speed on a 500 MHz PIII. Translated to processing times and assuming 8 minutes of music and 2 second FIRs (88,200 taps at 44.1 kHz) applying mono reverb (1x1) takes 1 minute and stereo to stereo reverb (2x2) takes 4 minutes. With *PureVerb* or *CATT-Acoustic* filters the *MultiVolver* automatically takes into account filter gains and in one step runs the number of convolutions required and mixes the outputs giving a margin value to full scale.



Licensing

The *FIReverb Suite (PureVerb + MultiVolver)* can be licensed separatelly (a 30-days trial license is available) but is also included with the full version of *CATT-Acoustic*[2].

References

[1]*Lake Technology Ltd.*, Sydney, Australia, http://www.lakedsp.com, [2]*CATT-Acoustic*, see separate info-sheet and the www page below where the *FIReverb Suite* and a fully working *CATT-Acoustic* demo can be downloaded.

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