DREAM DSP LIBRARY

One of the pioneers in digital audio, DREAM has been developing DSP code for over 30 years. But the company's roots go back even further to 1977, when their founder was granted his first digital audio patent. Today they offer a tiered approach to code development. At the highest level is DSP Designer, a brand new graphically based tool that enables the designer to rapidly build and test his design. It has a graphical user interface where the designer can drag and drop DSP blocks as required, making connections and setting set scales or limits. For most applications this is all that is required for the design process.

The next level down uses their established tool, SAMvs, an IDE (Integrated Development Environment) with a C compiler to generate the code used on the SAM5000 processors. There is a vast number of pre-defined functions and extensive help files to assist you when working at this level. Finally, there is the assembly layer, where you can write new functions at the lowest level.

To save you from needing to write low-level code, DREAM has a comprehensive selection of audio DSP processes and functions described below. These will cover the vast majority of pro-audio and prosumer applications.

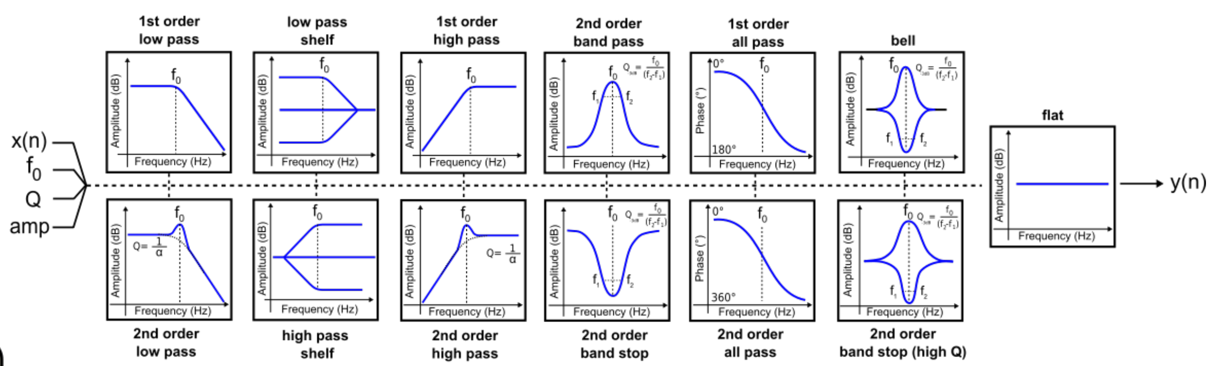
All DREAM development tools are free to use, as is their DSP library. In addition, there are extensive sound libraries for every instrument, particularly drums and pianos.

Contents

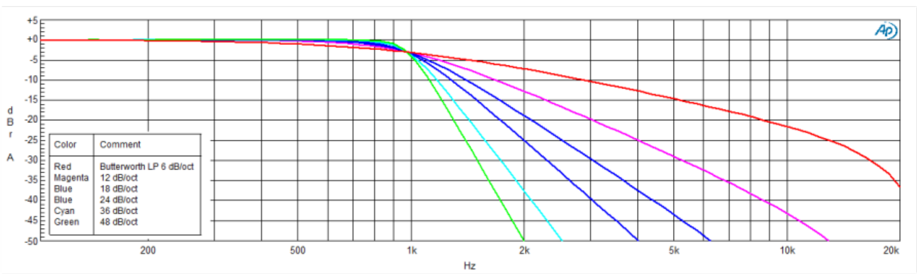
Parametric Equalisation & Speaker Crossover Filters

Parametric equalization is the process of filtering audio to enhance or tone down certain frequencies. It is fundamental to getting good sound, as it can compensate the frequency response for a room or a speaker cabinet, or to just add colour to the music. The basic filter blocks are second order, commonly referred to as biquads; these are implemented digitally using IIR topology, which results in sharper filter response for a given code size. With these building blocks DREAM's DSP Designer has 36 cross over filters to select from including Bessel, Butterworth and Linkwitz-Riley up to 8th order or 48dB/octave. The DREAM DSP calculates and stores coefficients in double precision (48 bits), minimizing distortion which can otherwise occur at low levels. Their internal MAC (Multiplier-Accumulator) uses 56 bits giving 8 guard bits so there is no clipping whilst processing audio.

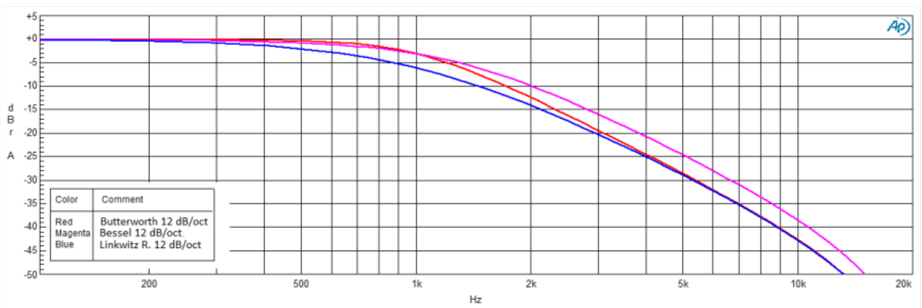
Standard filters



Measurement 1: multi scope Butterworth filters (fc=1kHz)



Measurement 2: Low pass 12 dB/oct Butterworth, Bessel and Linkwitz Riley filters (fc=1kHz)

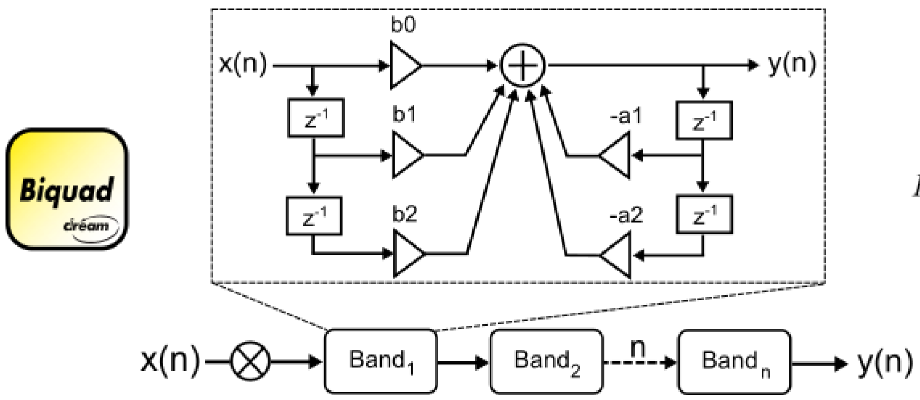


Parametric Equalisation & Speaker Crossover Filters (cont)

Control and Monitoring

You can select from 1 to 31 bands of eq, and for each band the parameters that can be controlled are:

- Select filter type and slope. 6–48dB per octave, 36 crossover filter types.
- Q or bandwidth
- Centre frequency
- Gain
- Filter on/off
- Invert phase



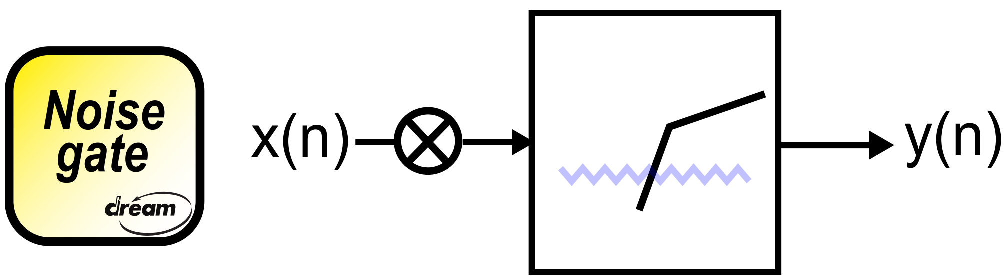
2nd order IIR Z transform:

$$H(z) = \frac{b_0 + b_1 * z^{-1} + b_2 * z^{-2}}{1 + a_1 * z^{-1} + a_2 * z^{-2}}$$

Noise Gate

The noise gate is a dynamics processor designed to reduce the level of background noise during quiet periods by reducing the signal below a threshold. Control and Monitoring

The standard controls are threshold level, gain, attack time, release time and boost level. In addition, there is compressor, on/off and invert signal.



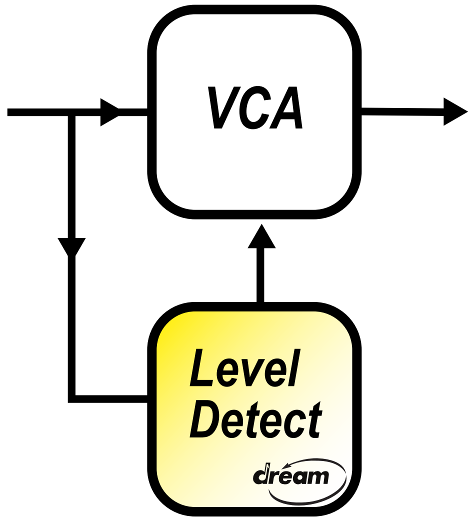
Compressor/Limiter and Level Detector

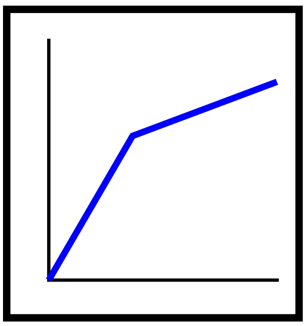
DREAM's compressor limiter is one of the fundamental dynamics processing algorithms used to enhance audio quality. For example, it may be used to boost the level to ensure the signal is clearly heard above the noise floor, or reduce the level to prevent clipping in an amplifier. The level detector is an averaging detector, with the rms option coming soon.

Control and Monitoring

The standard controls are:

- Threshold level
- Compression ratio
- Attack and release time
- Boost level
- Compressor on/off
- Invert signal phase
- Display of compression level





Delay

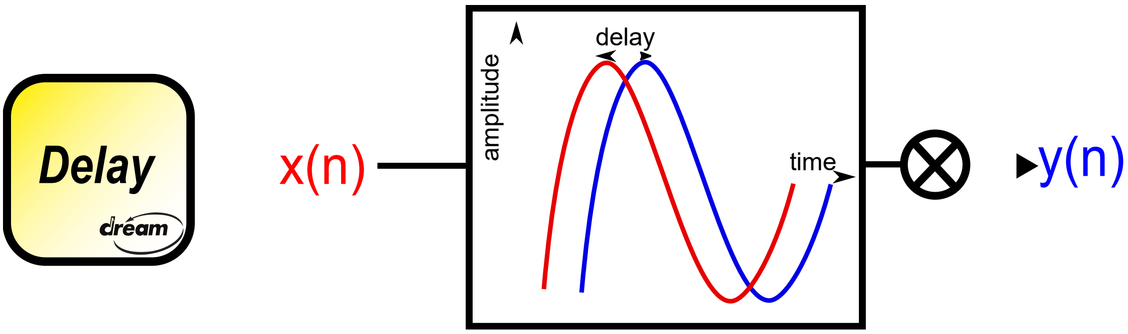
Delay is used for both time alignment as well as effects for guitar amplifiers and voice processing. The use in effects is described under vocal reverbs. Relatively long delays are used to time align multiple speakers; for example in cinema, stadiums and station platforms. Short delays are used in active crossovers when several drivers are in the same cabinet and the mounting positions of the tweeter and base driver need to be compensated for.

The delay can be in increments of one audio sample, and the DREAM DSP can access internal memory or external RAM for when long delays are needed.

Control and Monitoring

The standard controls are:

- Delay time
- On/off
- Gain
- Invert phase



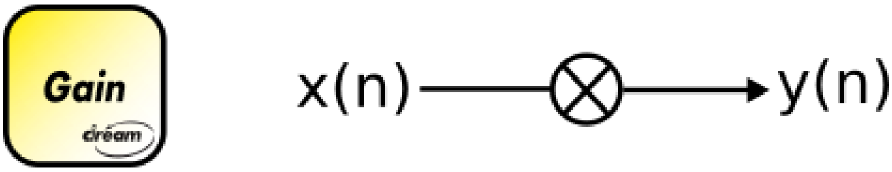
Gain, Mixing, Adding and Subtracting

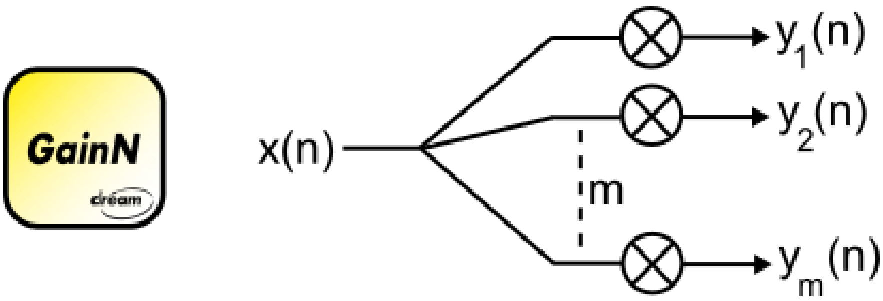
The basic mixer functions for between 2 and 12 channels per DSP block.

Control and Monitoring

The standard controls are:

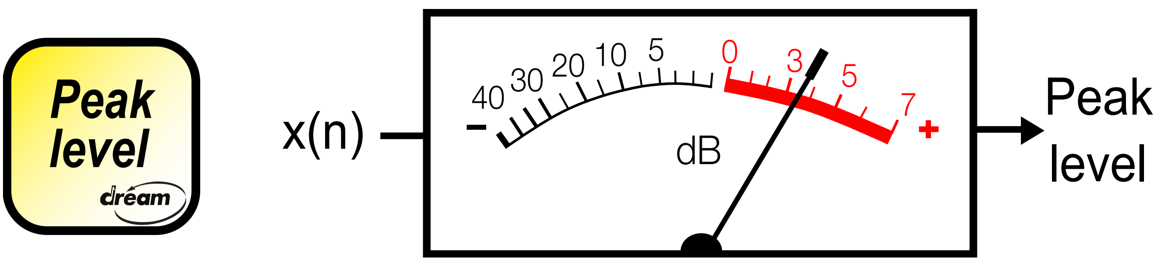
- Gain
- Invert phase





Peak Level

A function that returns the peak level to be used in peak level monitoring and bar graph displays.



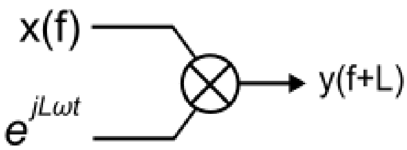
Frequency Shifter

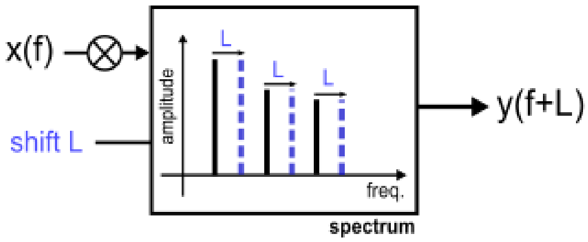
The frequency shift algorithm does just that, changing the frequency by up to ±16Hz, although you set the initial scale for the controls from ±5Hz to ±16Hz. It is primarily used for feed-back cancelling with simple to use controls and uses less DSP power so can be combined with notch filters and other DSP functions. Can also be used as a guitar effect.

Control and Monitoring

There are parameters to control the following:

- Shift amount
- Gain
- High pass filter
- Invert phase
- On/off





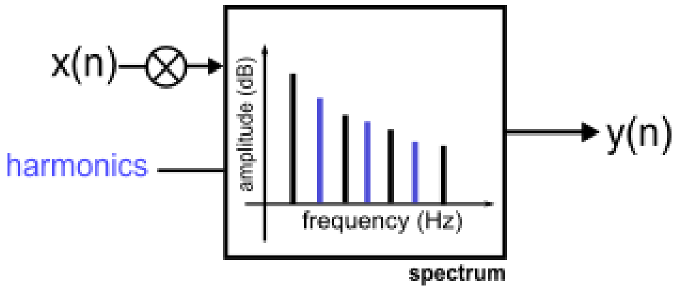
Exciter

The Exciter is often used with microphones. Its aim is to give a perceived increase in dynamic range with a mix of boost and compression of certain frequency bands and adding harmonic content.

Control and Monitoring

There are parameters for:

- Gain
- Low frequency compression threshold
- High frequency compression threshold
- Intensity
- Low contour
- On/off



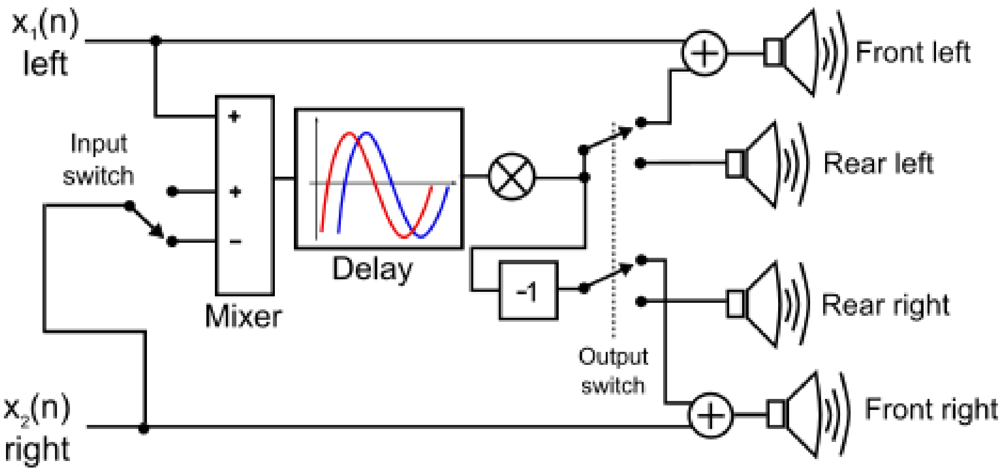
Spatializer

The spatializer, or 3D audio as it is sometimes referred to, mixes and delays a basic stereo stream to provide outputs to four surround sound speakers to create a more immersive experience.

Control and Monitoring

There are controls for:

- Delay time
- Level
- Invert phase
- On/off



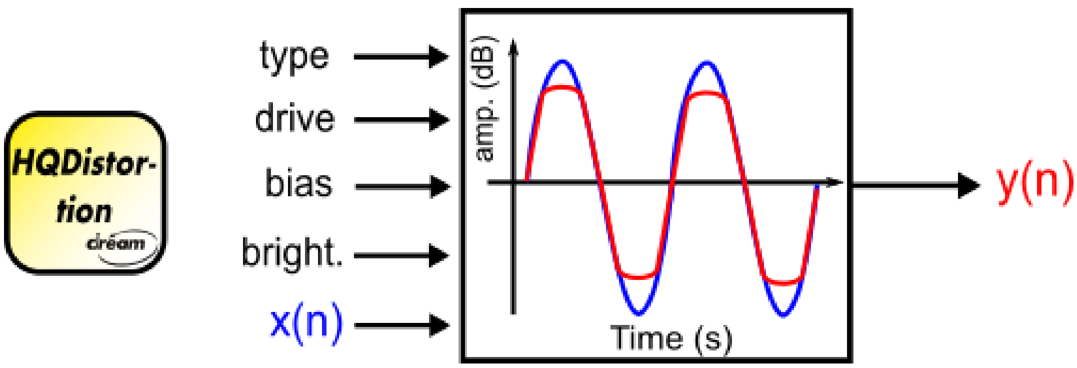
HQ Distortion

A digital effect that simulates a tube-based guitar amplifier being overdriven or clipped into distortion. Many of the leading guitar amplifier brands have their own sound as a result of the tube distortion characteristics. DREAM's HQ Distortion algorithm has all the parameters needed to create unique sounding amplifiers or mimic famous old ones.

Control and Monitoring

You can choose from a menu of possible controls including:

- Distortion types: gentle overdrive, overdrive, light distortion, medium distortion, distortion, fuzz, smooth tube, tube rectifier, asymmetrical gain
- Select number of tube or distortion stages from 1 to 3
- Bias
- Drive for each stage
- Brightness for each stage
- Volume



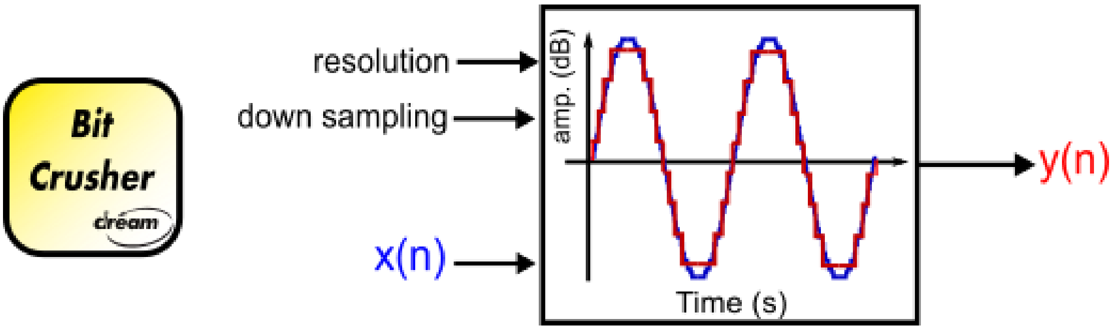
Bit Crusher

Used by DJs for effects, this algorithm adds quantization noise, making the sound digitally distorted. Used sparingly it can make the sound warmer.

Control and Monitoring

There are inputs for:

- Resolution from 0 to 12
- Brightness
- High frequency damping
- Downsampling
- On/off



Vocal Reverb

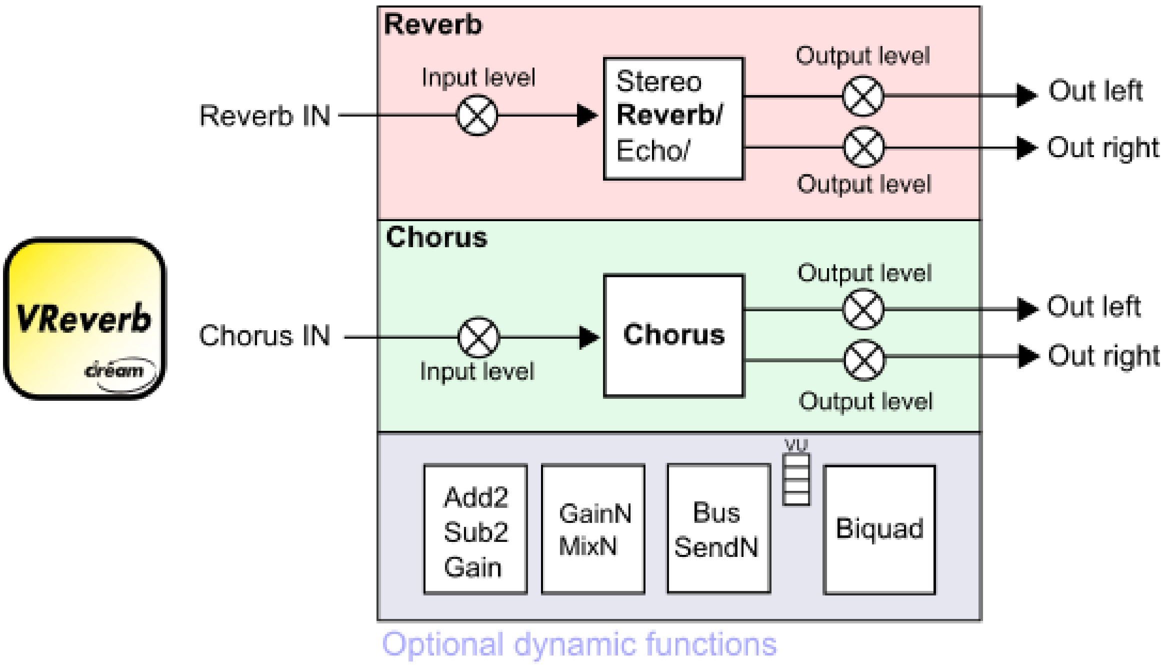
The vocal reverb is used extensively with piano and guitar to add harmonies resulting in a more natural sound. When used with voice it will give the impression that the singer is not in a closed mic or studio setting where there is no natural reverberation, but in a concert hall or room.

Many of these algorithms have taken many years to develop and refine, and they are available without a license fee.

Control and Monitoring

There are controls for:

- Select type (room, hall, plate, echo - mono and stereo)
- Size (small, medium, large)
- Input and output levels
- Low and high frequency cut
- Tone gain and frequency
- Reverb time (for hall, room and plate)
- Pre-delay
- Early reflection (for hall, room and plate)
- High frequency damping
- Diffusion (for hall, room and plate)
- Modulation
- Modulation speed and switch for evolving
- Echo feedback (for echo only)
- Echo time (for echo only)



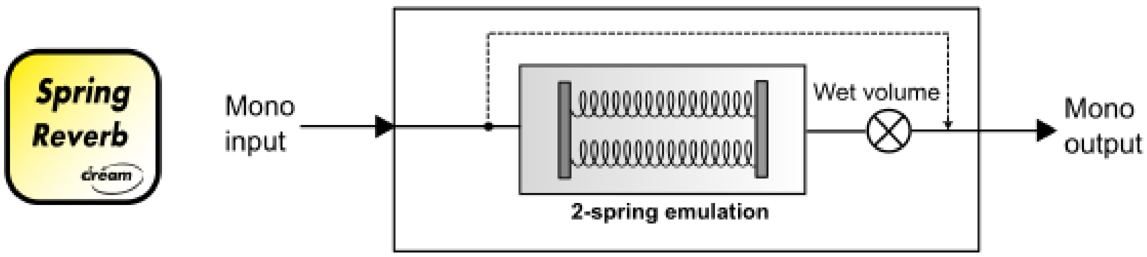
Spring Reverb

A digital simulation of a real spring reverb used in guitar amplifiers. Has the benefit of not suffering the spring crash when someone kicks the amp.

Control and Monitoring

There are controls for:

- Brightness
- Colour type (1,2 or 3)
- Colour
- Reverberation time
- Spring length (long, short or medium)
- Spring width (small, medium or large)
- Diffusion
- Damping (high or low)
- Level
- On/off



Chorus, Vibrato, Flanger and Phaser

Effects used in guitar pedals and mixers to colour the music. We recommend that your development engineer works together with your musician to refine your settings, which can be done in real time with an evaluation board. Then simplified controls can be defined for your users.

Control and Monitoring

Controls for:

- Delay time
- Depth
- Feedback
- Level
- Rate
- Spread
- On/off

Tremolo

A guitar effect used on tracks such as The Rolling Stones' 'Gimme Shelter' and The Smiths' 'How soon is now?'

Control and Monitoring

Controls for:

- Depth
- Rate
- Shape
- Type (mono/stereo)
- On/off

Wah

An essential guitar effect, made famous by the likes of Jimi Hendrix and Eric Clapton, which is intended to mimic talking. Guitar pedals often have controls with creative names like vowel, humanizer, texture etc.

Control and Monitoring

Controls for:

- Amount of resonance
- Filter type (low pass/band pass)
- Rate
- Shape
- Sensitivity
- Waveform
- Mode (up/down/sharp and LFO)

Pitch Detector & Vocal Shifter

This is a sophisticated suit of algorithms currently only available using SAMvs and not DSP designer.

Used in live and recorded sound to autotune a singer's voice to make it pitch perfect, or to add harmonies and special pitch effects to instruments and voice.

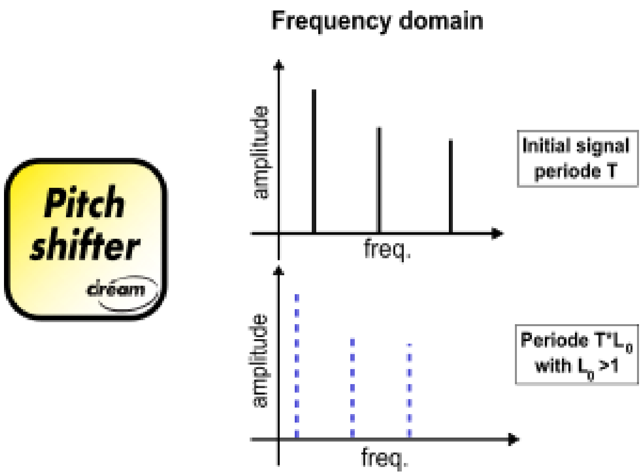
For more information there is application note 5704MICFX.

Control and Monitoring

Extensive controls and sub-functions are available providing all the flexibility you need to create your own auto tuner or effects processor.

For the pitch detector these include: minimum pitch, maximum pitch, silence threshold, enable compressor, sample rate, pitch smoothing, dry volume and dry pan.

For the vocal shifter they include: target pitch, formant, enable and set portamento time, release, smoothing, shift amount, vibrato – style, depth, delay and rate, humanize modes, levels and more.



Feedback Cancel

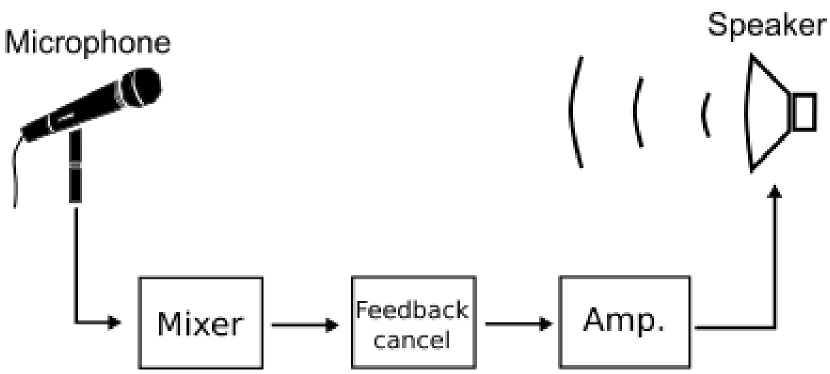
A complete quality feedback canceller that uses FFT analysis plus dynamic and static filters to identify feedback frequencies and eliminate feedback when it occurs. It uses up to 12 good notch filters, enough for any application.

Control and Monitoring

Very simple user controls considering the complexity of the coding that has functions for: check for feedback, detection speed, feedback frequency, set filter notch frequencies, set filter Q, filter status, bypass filters, reset filters.

The DSP Designer user controls are:

- Lock filter
- Bypass
- Mode (fast, default, slow)
- Extensive controls are available using samVS



Live-mic Dual-echo

The live-mic dual-echo is a special effect that combines a double delay with dynamics processing, and is used in live performance and Karaoke.

Control and Monitoring

Parameters include:

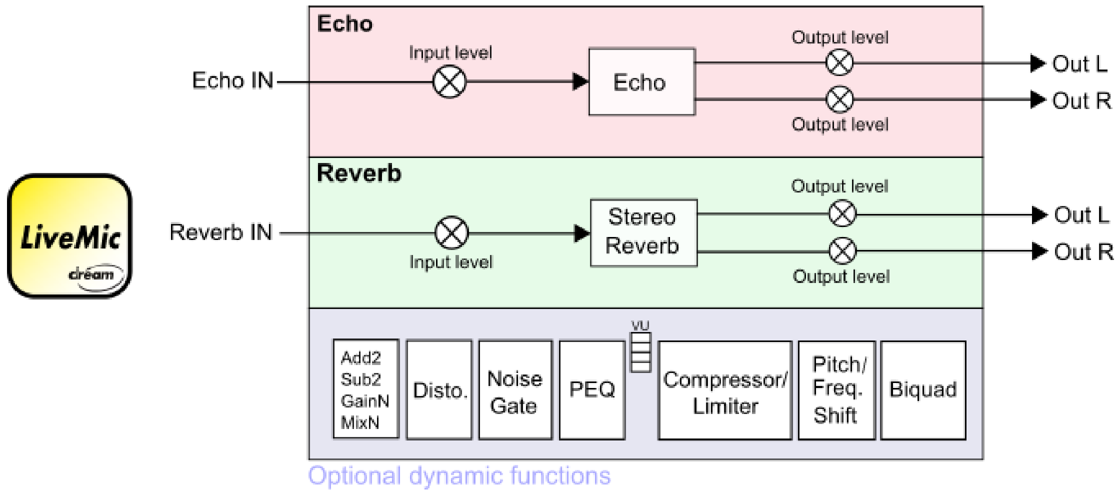
- Echo output phase
- Echo input level (and in phase selector switch)
- Level
- Low pass Filter
- Main pre delay

... then for each channel:

- Pre delay
- Echo low frequency damping
- Echo high frequency damping
- Echo time
- Echo feedback

... and finally:

- Echo output Mix (1 to 4)



The LiveMic module integrates premium quality reverberation effect + echo with miscellaneous dynamic processes.

Asynchronous Sample Rate Conversion

The inputs to the DREAM DSP can be in slave mode running at various frequencies. This function converts the sampling rate of the incoming I2S audio data stream (up to 8 channels) to synchronise it with the sample rate used in the SAM5000 processor. Dynamic range is greater than 120dB and THD less than 0.002%. %. >Asynchronous sample rate conversion can save on hardware system costs.

Control and Monitoring

Parameters include:

- Selecting the input frequency (if know or auto if not)
- Bypass

SPDIF

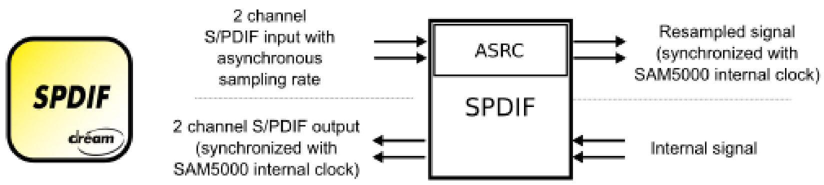
This is really a hardware feature but this function provides management of header and can be used with the Asynchronous sample rate conversion function.

Two channels of SPDIF input and output are available, and there is also a non-standard TDM mode which can be used to provide 8 channels in and out.

Control and Monitoring

There are functions for:

- Setting status bits or flags
- Channel bits and sample rates



Two channel SPDIF In/Out module with high quality Asynchronous Sample Rate Conversion (ASRC) on input to convert incoming SPDIF signal to SAM5000 internal rate.

Sample Rate Conversion engine performance: Dynamic range > 120dB and THD+N (1kHz) < 0.002%.

When using SPDIF in+out without ASRC (DSP running at the input signal sample rate), the module can operate up to 96kHz. When using SPDIF input with ASRC, the module can operate up to 192kHz, with the following limitations on the input signal sample rate:

SAM internal sample rate	SPDIF input signal maximum sample rate
44.1kHz/48kHz	192kHz
88.2kHz/96kHz	96kHz
192kHz	48kHz