



DUENDE
NATIVE

Duende Native V5

User Guide

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Introduction

WELCOME TO THE WORLD OF SSL

Congratulations on purchasing SSL's Duende Native plug-ins. Duende Native delivers a taste of the SSL mix experience – powerful channel and dynamics processing, the legendary Stereo Bus Compressor, plus a variety of contemporary audio tools developed by SSL to provide you with extensive processing versatility.

SSL is a company that has built its considerable reputation on excellence in audio production. With the rise in popularity of DAWs, there was an increasing demand to bring SSL's values and sonic signature to the world of plug-ins. Duende was our response to that demand. Duende was the result of our unique understanding of SSL's sonic heritage coupled with years of DSP development in SSL's digital console range.

Initially to bring Duende to life, we took the DSP heart of our acclaimed C-Series consoles and transplanted it into a hardware plug-in host. As native processing has become faster, the ability to run the sophisticated Duende plug-in algorithms efficiently in the host CPU has removed the need for additional DSP, while still leaving the DAW to deliver smooth workflow. Duende Native plug-ins deliver the same acclaimed sonic performance as their hardware DSP predecessors, but also offer the convenience and reliability of a complete 'in the box' solution.

Duende is a term used in flamenco connected to magic, spirit and passion, it's a name that evokes all that we've achieved with this product – and all we hope it will inspire its users to achieve.

ABOUT THIS MANUAL

This manual starts with a complete overview of the processes involved in installing and authorising your plug-ins then going on to a detailed description of all seven of the plug-ins in the Duende Native range.

Please note that the procedures involved in using plug-ins within your host application are not covered in this manual. Please refer to your host application's user guide for more information on how to use plug-ins in general.

If you are reading this on-screen, you will find that a number of helpful hyperlinks have been included, providing easy navigation around the manual, as well as quick access to key parts of the SSL website.

PLUG-IN TUTORIALS AND SUPPORT

If you would like any advice or support about your SSL plug-ins, the SSL website has an extensive support FAQ section:

www.solidstatelogic.com/support

SSL have created a number of tutorials which provide plug-in users with clear explanations and expert advice on signal processing techniques. They include Pro Tools, Logic and Cubase/Nuendo project files which allow you to experience real-life examples of the techniques within the Duende plug-ins. These are located within the Duende Product section of the SSL website:

www.solidstatelogic.com/music/duende-native/tutorials.asp

You can access these tutorials by clicking the Tutorial icon on the right – these icons are also found throughout the manual.



ABOUT THE SSL PLUG-IN RANGE

Channel Strip

The EQ and Dynamics Channel Strip plug-in is based on the EQ and compressor sections of the XL 9000 K Series console. SSL were the first mixing console manufacturer to feature dynamics and EQ on every channel on an in-line console with the SL 4000 B Series in 1977. The plug-in includes separate high and low pass filters, an independent compressor/limiter and gate/expander, a four band parametric equalizer which is assignable to the dynamics sidechain, variable processing order as well as input and output gain adjustment and phase inversion.



Stereo Bus Compressor

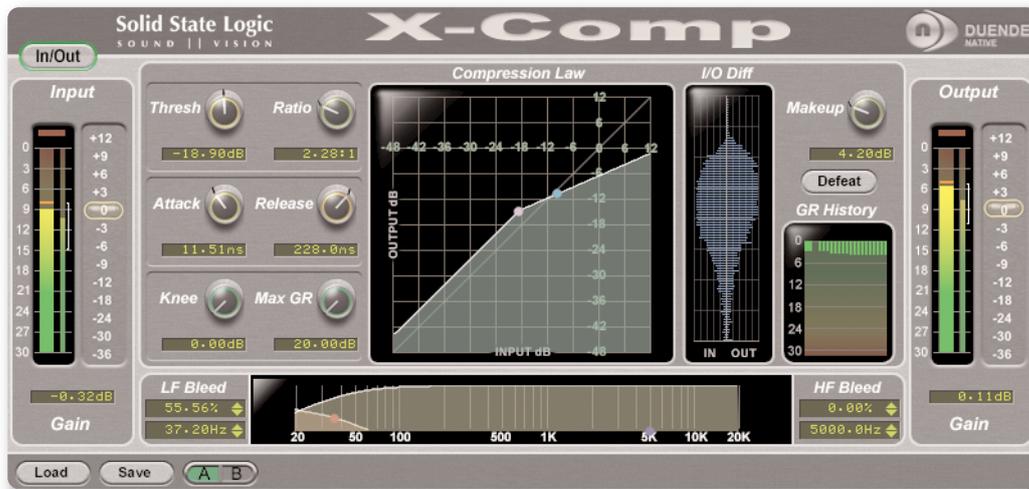
The Stereo Bus Compressor plug-in is a stereo version of the legendary centre section bus compressor found on the XL 9000 K Series console. It provides high quality stereo compression, giving you critical control over the dynamic range of audio signals.



Uses may include inserting the bus compressor over a stereo mix which has the effect of 'gluing' the mix together whilst still maintaining a big sound. The dynamics of drum overheads or whole drum kits can be controlled very effectively with the bus compressor. As it is available as either a stereo or mono plug-in the bus compressor can be used for practically any application that requires superior compression.

X-Comp

X-Comp can deliver the transparent audio finesse of a mastering grade stereo compressor or be driven to inject character and raw power to rival the SSL Listen Mic Compressor. In X-Comp we haven't just modelled a particular compressor, but have provided a set of features and controls that allow the emulation of many vintage and modern compression designs based on a well loved SSL feed forward compressor algorithm.

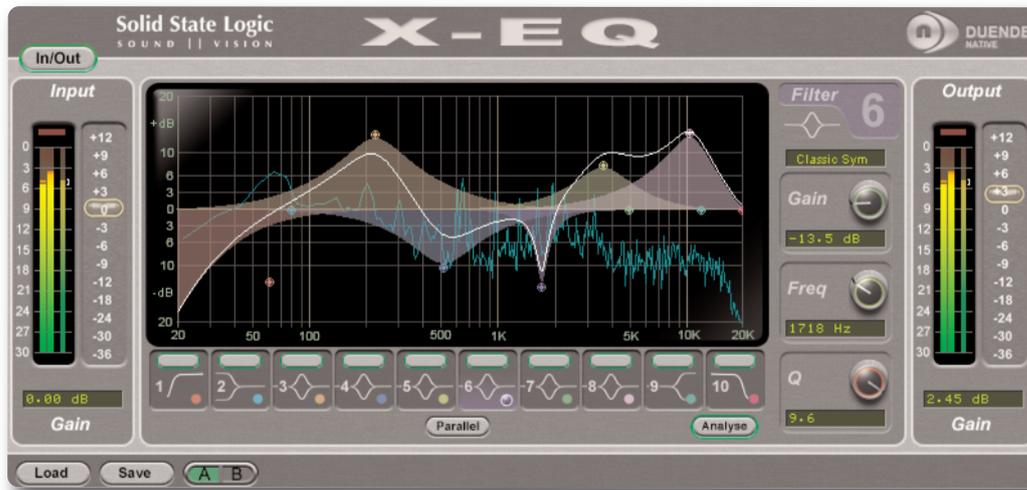


Key Features

- Dual symmetrical knee design allows detailed shaping of the compression characteristic
- Advanced side chain architecture using 1st order filters delivers user friendly Frequency Dependent Parallel Compression
- Amplitude Histogram and Gain Reduction history displays provide advanced real time pre/post signal analysis
- Max Gain Reduction control provides genuine vintage compressor characteristics
- Intuitive user interface with drag and move graphic, mouse wheel and numeric editing
- A/B facility for instant comparison of two different compression set ups
- Proprietary preset management functions providing compatibility between all DAW platforms
- Global latency free bypass
- Superb mastering grade audio quality delivered by Duende's 64-bit floating point engine
- Preset library based on settings used by some of the worlds top mixing engineers

X-EQ

X-EQ is a versatile and comprehensive mastering grade parametric EQ plug-in built from the ground up for Duende. It has an unprecedented clarity, smoothness, and excellent high-frequency response. A total of 10 EQ bands are provided, split into dedicated HP and LP filters, LF and HF shelves and 6 bell bands. A Parallel EQ mode is also provided which recreates the characteristic sound of an old parallel passive EQ circuit built with inductors and capacitors.



Key Features

- 10 band fully parametric high quality digital EQ
- A total of 16 different filter types: 5 different cut filters, 9 bell shapes, shelving filters and a parallel mode
- Extensive control options including draggable EQ graph nodes, mouse wheel adjustment and numerical data entry
- Extremely low noise and low non-linear distortion filter algorithms resulting in the residual THD+N significantly lower than 24-bit quantisation noise
- Comprehensive stereo/mono peak (with clip-hold), RMS and dynamic history metering at input and output
- Different bell filter types are all normalised to look identical at +6dB boost allowing for quick comparison between filter types
- Proprietary preset management functions providing compatibility between all major DAW platforms
- A/B functionality for easy comparison of any two settings
- Individual band bypass
- Global latency free bypass
- Real-time FFT Analyser showing the result of the EQ processing on the audio spectrum
- Superb mastering grade audio quality delivered by Duende's 64-bit floating point engine
- Preset library based on settings used by some of the worlds top mixing engineers

Vocalstrip

Vocalstrip is a one stop solution for superior vocal processing, with all the tools you need to take your vocal sound into the major league. Vocalstrip features 4 configurable processing blocks to take care of unwanted artifacts, dynamics and equalisation and extensive metering and visual feedback give you invaluable information during your work.



Key Features

- Intelligent De-esser
- Intelligent De-ploser
- 3 band EQ
- Compander featuring compression, downwards expansion and output drive
- Extensive visual feedback including a real-time FFT analyser showing the result of the EQ processing on the audio spectrum
- Complete control over process order
- Extensive control options including mouse wheel adjustment and numerical data entry
- Proprietary preset management functions providing compatibility between all DAW platforms
- A/B functionality for easy comparison of any two settings
- Global latency free bypass
- Superb mastering grade audio quality delivered by Duende's 64-bit floating point engine
- Preset library based on settings used by some of the worlds top mixing engineers

Drumstrip

The Drumstrip plug-in for Duende brings a unique blend of tools to the Duende Native platform, which provides an unprecedented amount of control over the transient and spectral elements of drum and percussion tracks. Manipulation which previously may have been time-consuming or impossible with traditional EQ and dynamics processing becomes elegant and rewarding with SSL Drumstrip.



Key Features

- Transient shaper capable of drastically changing the attack characteristics of rhythmic tracks. An audition mode makes for easy setup
- Highly controllable Gate featuring both open and close thresholds, attack, hold, release and range control
- SSL Listen Mic Compressor – as featured in the LMC-1 plug-in – with extra functionality
- Separate high and low frequency enhancers provide spectral control not achievable with traditional EQ
- Peak and RMS metering on both input and output
- Wet/dry controls on both the main output and the LMC, allow parallel processing to be easily dialled in
- Process order control over all five sections gives complete flexibility over the serial signal chain
- Latency-free bypass of all processing

X-Verb

X-Verb is a versatile and comprehensive studio grade reverberation plug-in, delivering the density, warmth, depth and stunning detail usually only associated with high class hardware reverb units. The use of innovative Diffusion Engines and innovative processing design provides superior performance that surpasses other Reverb Plug-Ins in both artistic freedom, depth and sonic quality. This plug-in may well provide all the different Reverb options you will ever need.



Key Features

- Unique and modular SSL HiD Reverb Algorithm Engine
- 4 Core Algorithms create a complete collection of processors
- Quick Controls panel to easily alter the superb Presets
- Stunning Graphical User Interface to access the detailed parameter set
- Automatable Morphing feature to dynamically morph from one preset to another

X-ValveComp

X-ValveComp is a fully-featured channel compressor with valve emulation, side-chain with filters and a wet/dry “Mix” control for parallel processing.



Key Features:

- Based on SSL’s C300 compressor
- Mode: Switches between Peak & RMS response
- SC in (with high and low pass filters)
- Valve introduces a tube compressor emulation for saturating the signal by adding harmonics
- Auto Make-Up
- Threshold, ratio, adjustable knee
- Mix control for parallel compression

X-Saturator

X-Saturator allows the user to introduce clipping and harmonic distortion to a signal to add some pleasing “analogue character”. The Saturator emulates analogue gain circuits that introduce harmonic distortion when driven and can act as a veritable time-machine of processing sounds, covering everything from smooth tube harmonics to screaming crushed techno sounds.



Key Features:

- The Saturator emulates analogue gain circuits
- The Drive controls the input level of the plug-in
- By adjusting the Harmonics control the distortion characteristics can be shifted from 50's valve-style overdrive (if the Harmonics selection control is turned fully anti-clockwise for 2nd harmonic distortion) to 70's transistor-style grit (if the Harmonics selection control is turned fully clockwise for 3rd harmonic distortion).
- Depth and Shape control the number of harmonics injected back into the signal, enriching the sound
- Boost adds 6dB of headroom to avoid internal clipping
- Dry/Wet allows parallel processing

X-Phase

X-Phase is an Allpass filter and time alignment tool for manually adjusting the phase. It can be used to time/phase align signals, for example from a stereo set of overhead mics on a drum kit.



Mono Version



Stereo Version

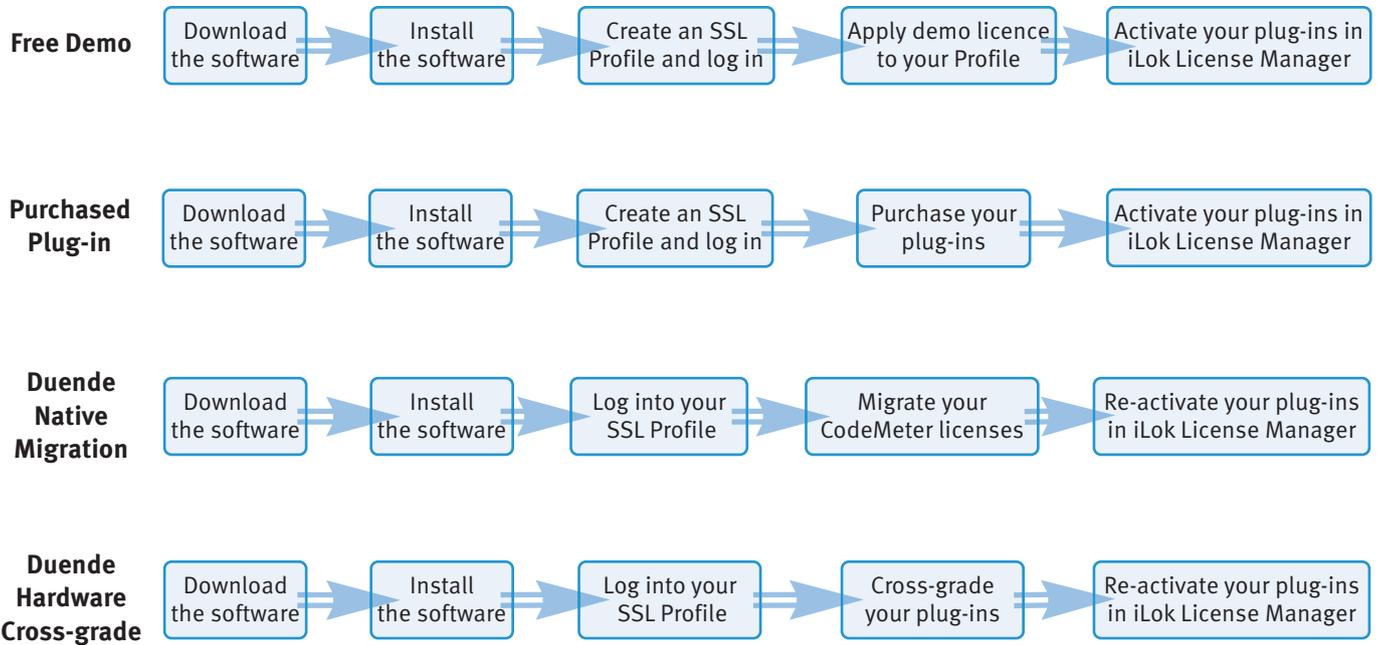
Key Features

- X-Phase is an Allpass filter and time alignment tool for manually adjusting the phase
- The gain is linear across the frequency response but a change in phase is applied such that it 'wraps' at a specific frequency and converges to 0 for all other frequencies
- The Q-Factor of the filter will adjust the rate of phase-change across the frequency spectrum
- 2nd Order inverts the filter
- Delay (in various units)
- Link L/R for stereo signals
- 180 degree phase shift with Allpass 'phase' button

1. Software Setup

OVERVIEW

In order to get your SSL software working, here are the steps you will need to go through:



Each of these steps will be covered over the following pages. There is a good chance that you are already some way through this process – feel free to jump to the appropriate section.

You can navigate to any of the sections by clicking on the appropriate blue box. If you would like more information on any stage of the setup process, please refer to the Duende Native area of the SSL website.

Cross-grade: *This option is for users who already have either a Duende unit or the Codemeter plug-ins and wish to take advantage of the free cross-grade offer. See page 21 for details.*

SYSTEM REQUIREMENTS

Before starting the process, you may want to confirm that no upgrades to your computer or its software are required to run Duende Native:

Windows

- Windows XP (SP2 or higher), Vista (32- or 64-bit), 7 (32- or 64-bit) or 8 (64-bit)
- Intel Core2 (or comparable) CPU running at 2GHz or higher
- 2GB of RAM minimum
- VST (32- or 64-bit), VST3 (32- or 64-bit), AAX Native (32- or 64-bit), RTAS (32-bit) compatible host

Duende Native does not support Pentium or Celeron CPUs

Macintosh

- OS X 10.6.8 or higher (32- or 64-bit kernel), OS X 10.7.1 or higher (32- or 64-bit kernel), OS X 10.8.5, OS X 10.9 or higher
- Intel Dual Core Mac running at 2GHz or higher
- 2GB of RAM minimum
- AU (32- or 64-bit), VST (32- or 64-bit), VST3 (32- or 64-bit), AAX Native (32- or 64-bit), RTAS (32-bit) compatible host

Duende Native does not support Power PC Macintosh.

Host Applications

Duende Native is designed to run within the following host applications:

Windows

- SSL Soundscape 6
- SSL Soundscape 7
- Ableton Live 8
- Ableton Live 9
- Avid Pro Tools 9
- Avid Pro Tools 10
- Avid Pro Tools 11
- Steinberg Cubase 6
- Steinberg Cubase 7
- Steinberg Cubase 7.5
- Steinberg Wavelab 7
- Steinberg Wavelab 8
- Steinberg Nuendo 5
- Steinberg Nuendo 6
- PreSonus Studio One v2
- Cakewalk Sonar X2
- Cakewalk Sonar X3
- Motu Digital Performer 8
- Cockos Reaper 4
- Sound Forge Pro 11
- Vegas Pro

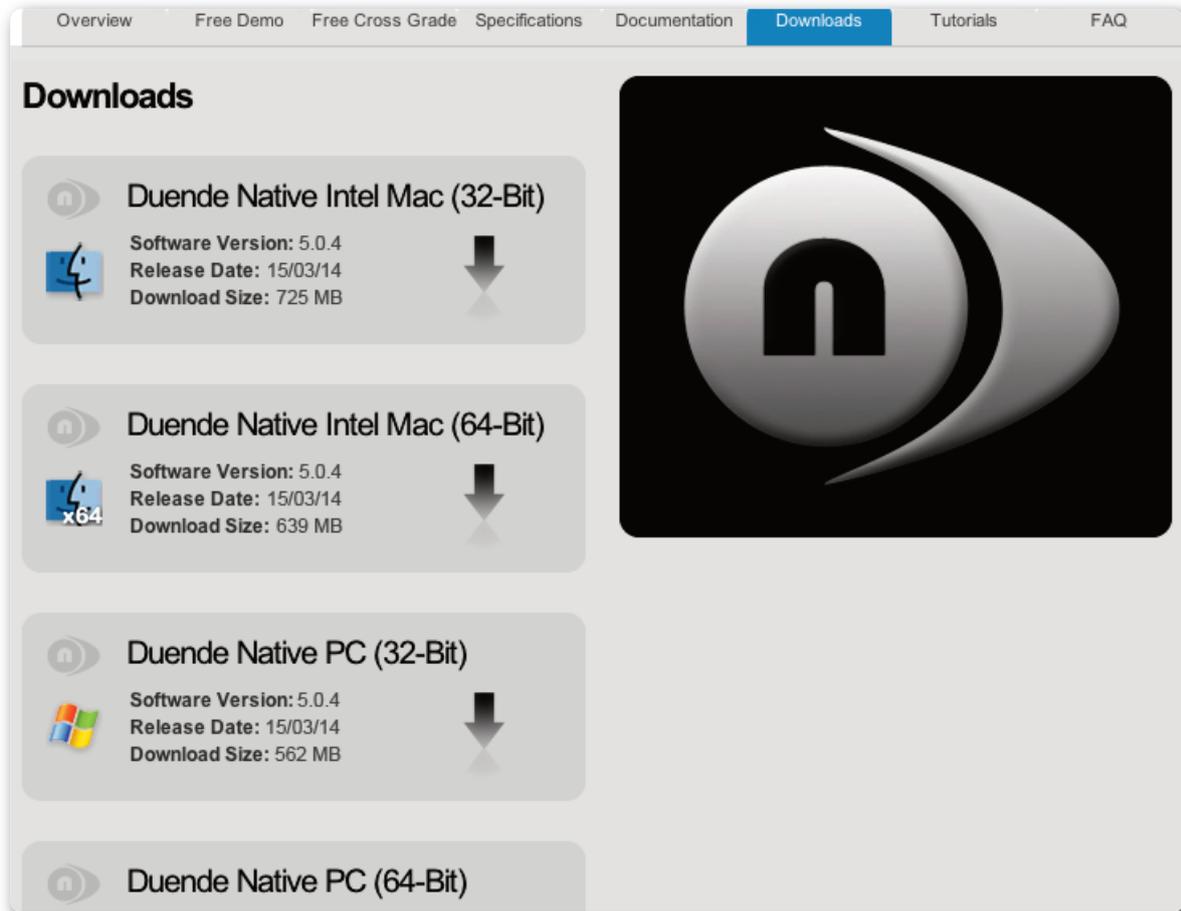
Mac

- Logic Pro 9
- Logic Pro X
- Ableton Live 8
- Ableton Live 9
- Avid Pro Tools 9
- Avid Pro Tools 10
- Avid Pro Tools 11
- Steinberg Cubase 6
- Steinberg Cubase 7
- Steinberg Cubase 7.5
- Steinberg Wavelab 7
- Steinberg Wavelab 8
- Steinberg Nuendo 5
- Steinberg Nuendo 6
- PreSonus Studio One v2
- Cakewalk Sonar X2
- Cakewalk Sonar X3
- Motu Digital Performer 8
- Cockos Reaper 4
- Sound Forge Pro 11

DOWNLOADING THE SOFTWARE

Please download the latest Duende Native software from www.solidstatelogic.com. Note that you do not need to purchase plug-ins in order to download them although you will need to register on the SSL website and activate demos onto your iLok 2 before use. To use them for more than one month you will need to purchase authorisation – described on Page 19.

To download the software, go to www.solidstatelogic.com, hover over the **PRODUCTS** tab (at the top of the page) followed by the **MUSIC** entry in the drop-down which appears and click on **Duende Native** in the second drop-down which appears. Select the **Downloads** tab to be taken to the page shown below. Alternatively, just click on the pdf image below:



There are four download boxes; for 32-bit Mac, 64-bit Mac, 32-bit Windows and 64-bit Windows machines. Click on the **Download** label within the appropriate download box and the download will start automatically.

It is important that all users install the correct software version – please refer to your computer's User Guide to establish which of the following host/OS permutations applies to you:

Mac OS X 10.6 – 10.9, 32-bit Host *:	32-bit installer required
Mac OS X 10.6 – 10.9, 64-bit Host *:	64-bit installer required
32-bit Windows, 32-bit Host:	32-bit installer required
64-bit Windows, 32-bit Host:	32-bit installer required
64-bit Windows, 64-bit Host:	64-bit installer required
64-bit Windows, Separate 32-bit and 64-bit Hosts:	both installers required!

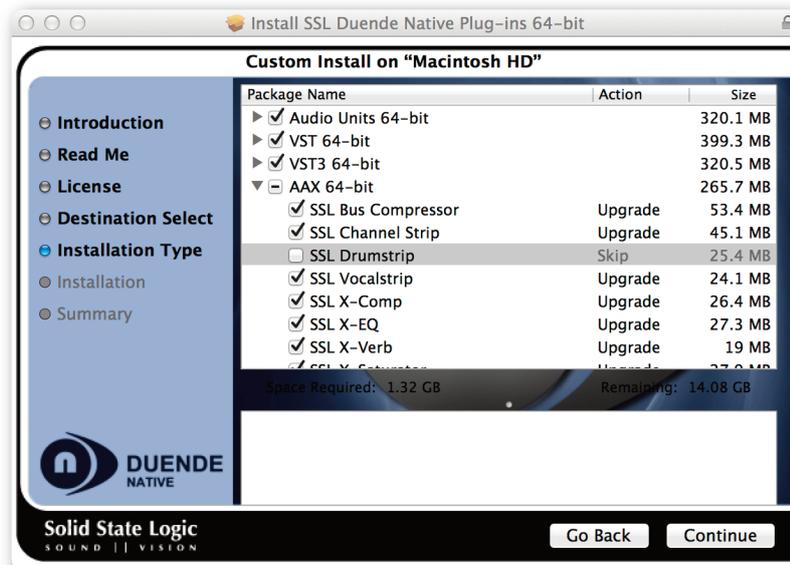
** Pro Tools 10, and below, is a 32-bit host on both Mac and Windows and therefore requires the 32-bit Mac or 32-bit Windows installer. Pro Tools 10 supports both RTAS and AAX, but when both are installed the AAX plug-ins take precedence. Pro Tools 11 is a 64-bit host and therefore Pro Tools 11 users should install the AAX plug-ins from the 64-bit Mac or 64-bit Windows installer.*

INSTALLING THE SOFTWARE (MACINTOSH)

- Choose the appropriate installer for your DAW; 32-bit or 64-bit. Double click the Duende Native disk image file that you downloaded to mount the disk image.
- Run the Duende Native Plug-ins installer and follow the on-screen instructions.



During the installation process you can choose which plug-ins and formats you don't require by unticking the appropriate boxes:



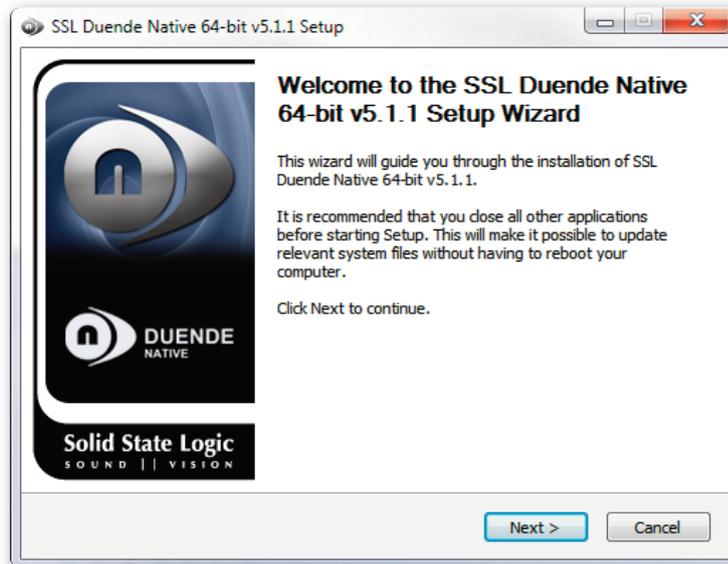
In the disk image you will also find iLok License Support.dmg. If you do not already have the iLok software installed – for example if this is the first time you have used your iLok 2 – you should double click this disk image and run the installer. If you have a newer version of the software already installed you will be notified and prompted to quit the installer.

If this is the first time you are installing the Duende Native V5 plug-ins you will need to activate the licenses onto your iLok 2. Please see the 'Activating Plug-In Licenses' or 'Claiming Demos' section of this user guide for more information.

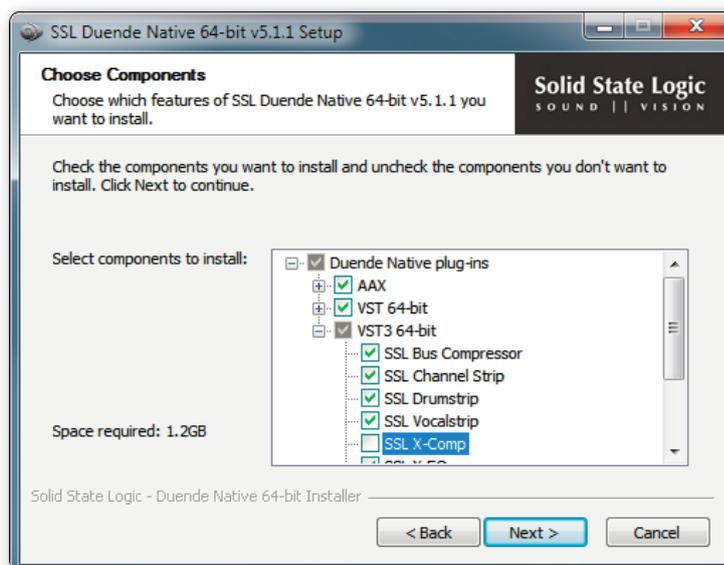
Please note that a newer version of iLok License Manager may be available here: <https://www.ilok.com/>

INSTALLING THE SOFTWARE (WINDOWS)

- Choose the appropriate installer for your DAW; 32-bit or 64-bit. Double click the Duende Native zip file that you downloaded to open it.
- Run the SSL Duende Native Setup.exe file and follow the on-screen instructions:



During the installation you can choose which plug-ins and formats you don't require by unticking the appropriate boxes:



In the zip file you will also find License Support.exe. If you do not already have the iLok software installed, for example if this is the first time you have used your iLok 2, you should double click this file to run the installer. If you have a newer version of the software already installed you will be notified and prompted to quit the installer.

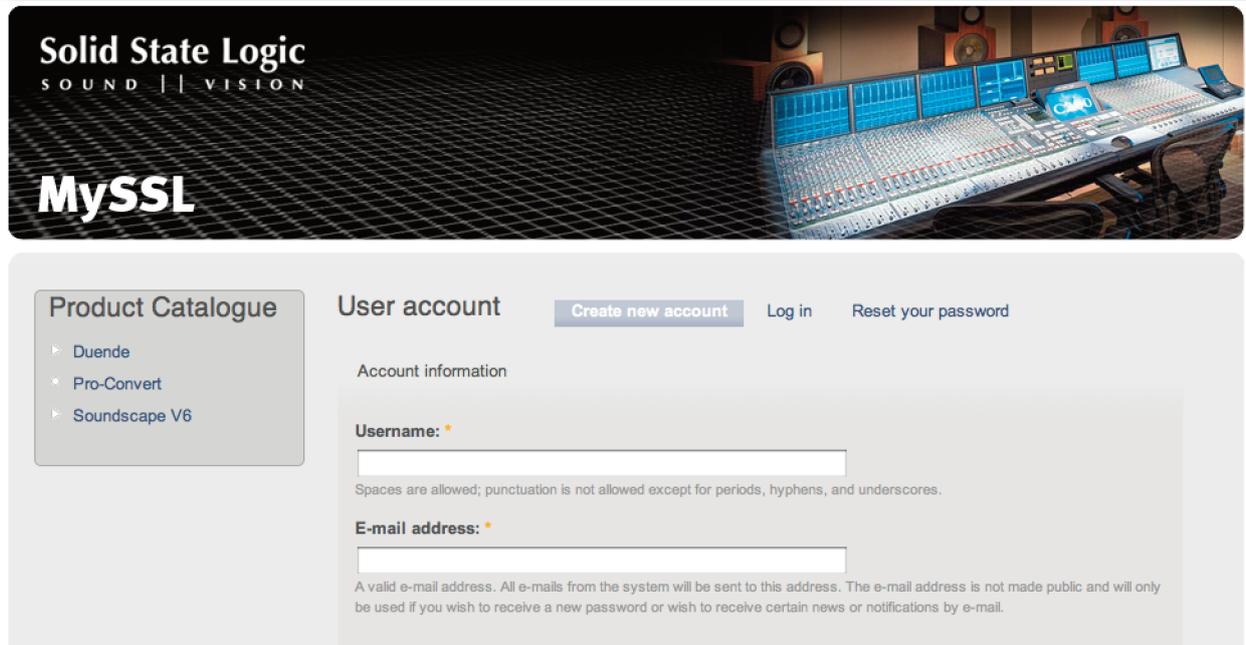
If this is the first time you are installing the Duende Native V5 plug-ins you will need to activate the licenses onto your iLok 2. Please see the 'Activating Plug-In Licenses' or 'Claiming Demos' section of this user guide for more information.

Please note that a newer version of iLok License Manager may be available here: <https://www.ilok.com/>

CREATING AN SSL PROFILE

In order to authorise your SSL software, you need to create an SSL profile:

Go to www.solidstatelogic.com and click on **REGISTER**, located in the top right-hand corner of the SSL home page. Clicking on the picture below will take you there:

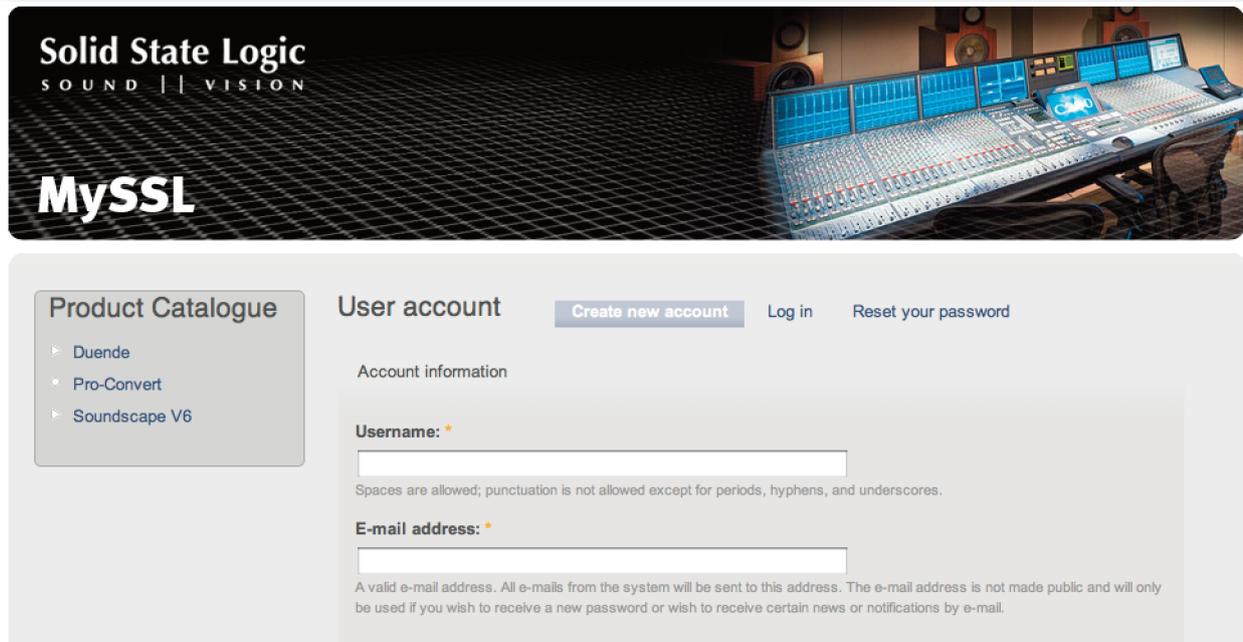


Fill in your details and click on the **Create New Account** button at the bottom of the page.

You will be sent a password by email. Once you have received this email, you are ready to continue.

LOGGING INTO THE SSL WEBSITE

To log into the SSL website, go to www.solidstatelogic.com and click on **LOG IN** in the top right-hand corner. Clicking on the picture below will take you there:



Type your username you created and the password sent to you by email, and click on the **Log In** button below the password.

BUYING PLUG-INS

Duende Native plug-ins are now licensed exclusively using iLok 2. To buy Duende Native plug-ins you will need both an SSL Account and an iLok User Account – and to activate and use them you will need an iLok 2 USB Dongle.

If you don't already have an iLok User Account you should create one here: [iLok Register](#)

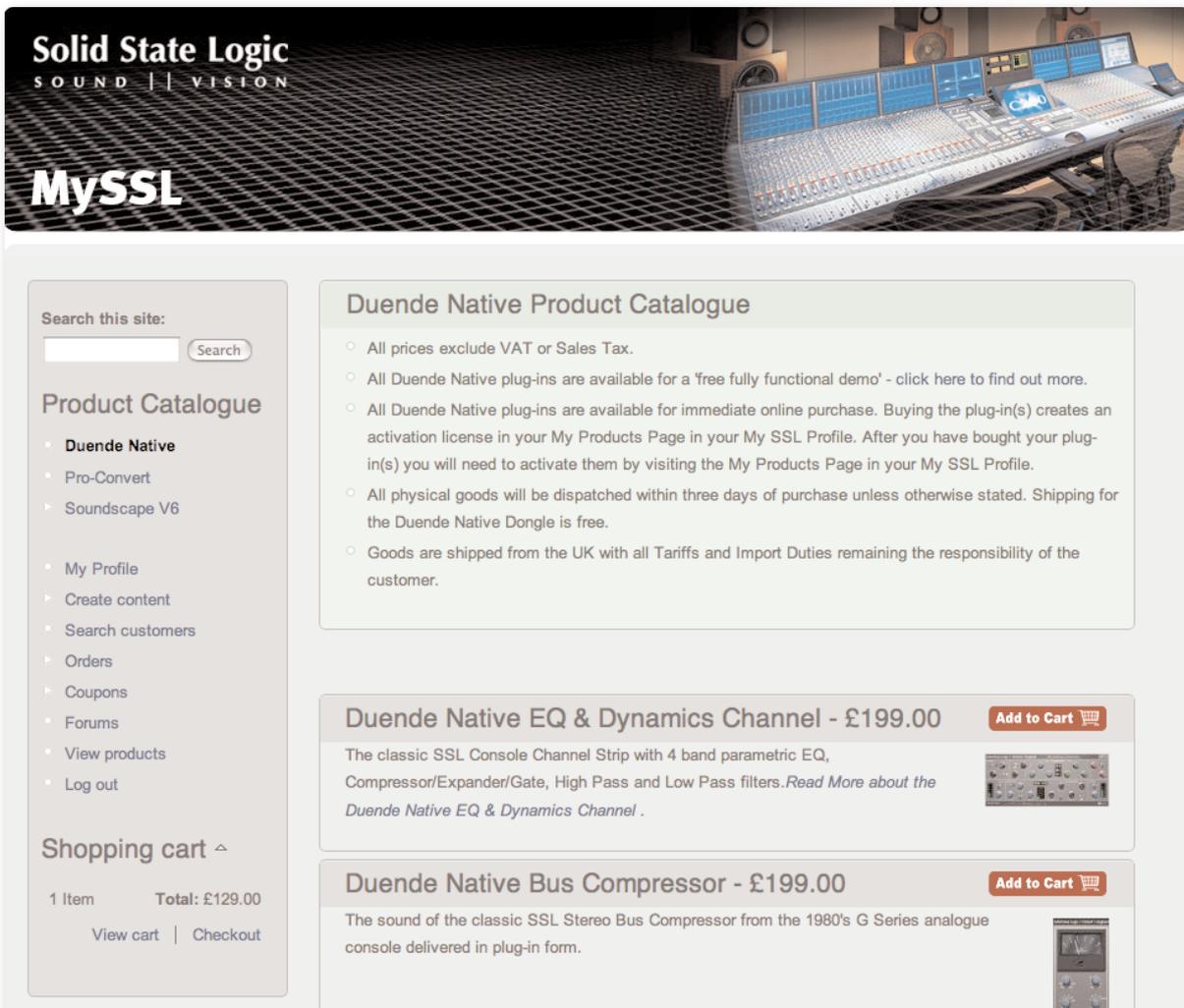
You can purchase an iLok 2 Dongle here: [SSL webstore-iLok](#)

If you already own Duende Native plug-ins using the Codemeter system or Duende Hardware then please read the 'Crossgrading From Duende Hardware' or 'Migrating Licenses' sections below for information.

Please refer to earlier in this guide for information about creating and logging into an SSL profile.

When you log into the website you are automatically taken to the **MySSL** area. To navigate to there from elsewhere in the website, click on your name in the top right-hand corner of the screen.

In the **Product Catalogue** down the left-hand side of the **MySSL** area, click on **Duende Native** to be taken to the Duende Native store:



The screenshot shows the Solid State Logic MySSL website interface. At the top, there is a banner with the Solid State Logic logo and the text 'SOUND | VISION' and 'MySSL'. Below the banner, the main content area is divided into several sections:

- Search this site:** A search bar with a 'Search' button.
- Product Catalogue:** A list of navigation links including 'Duende Native', 'Pro-Convert', 'Soundscape V6', 'My Profile', 'Create content', 'Search customers', 'Orders', 'Coupons', 'Forums', 'View products', and 'Log out'.
- Shopping cart:** A section showing '1 Item' and 'Total: £129.00', with links for 'View cart' and 'Checkout'.
- Duende Native Product Catalogue:** A list of product descriptions with 'Add to Cart' buttons. The first product is 'Duende Native EQ & Dynamics Channel - £199.00', described as 'The classic SSL Console Channel Strip with 4 band parametric EQ, Compressor/Expander/Gate, High Pass and Low Pass filters.' The second product is 'Duende Native Bus Compressor - £199.00', described as 'The sound of the classic SSL Stereo Bus Compressor from the 1980's G Series analogue console delivered in plug-in form.'

Select the plug-ins you wish to purchase by clicking on their Add to Cart buttons and complete the check-out and purchase processes. During the purchase process you will be asked to enter your iLok User ID if it is not already present in your SSL account, then validate it.

For each purchased plug-in a license will be deposited into your iLok account. You can access your iLok account by logging into it here: <https://www.ilok.com>

ACTIVATING PLUG-IN LICENSES

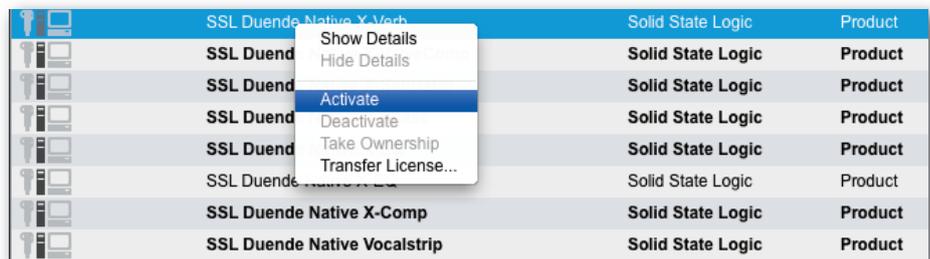
- Open 'iLok License Manager'.
 - On Mac OS it is located in your *Applications* folder. You may wish to drag this to your Dock for easy access.
 - On Windows it is accessible via the *Start Menu*.
- Log in to your iLok account which must be the same as you entered into the SSL Webshop when you bought the plug-ins.
- Once you've signed in, connect your iLok. If an unregistered iLok is connected, a popup will appear asking if you would like to add the iLok to your account; select "yes". The iLok will be registered to your account.
- Activate your SSL License(s).

Note that only iLok 2s are valid license locations

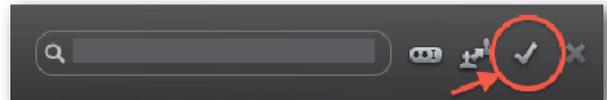
There are five ways to activate (download) a license:

1 Drag and drop: click on a license to select it, then drag the license you wish to activate onto the desired activation location and drop it.

2 Context Menu: right-click the license you wish to activate and select "Activate", select a location, and then click the "Activate" button.



3 Icon menu: select the license you wish to activate by clicking on it, then click on the check mark in the top right corner of the iLok License Manager window, select a location, and click the "Activate" button.



4 From the Detail pane: select the license you wish to activate by clicking on it, then, if you don't already have the Details pane open, click on "Show Details". Click the "Activate" link, select a location, and then click the "Activate" button.

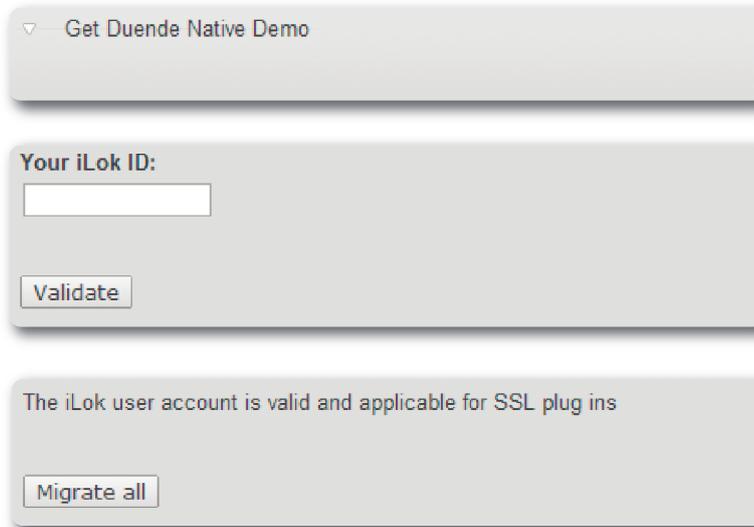
5 Application menu: select the license you wish to activate by clicking on it, go to the Licenses in the menu bar, and select "Activate".

Your licenses are now activated and the plug-ins are ready for use within your DAW.

For further information on using your iLok please see here: www.ilok.com/ilm.html

CLAIMING DEMOS

To claim a 30 day demo for all plug-ins, press **Get Duende Native Demo** under **MY PRODUCTS** in your SSL account. If your SSL account is not already associated with your iLok account you should enter your iLok User ID here and validate the iLok account by pressing the **Validate** button. Once validated, the button will change to **Get Demo**. Press this to send licenses for all Duende Native V5 plug-ins to your iLok account.



▼ Get Duende Native Demo

Your iLok ID:

Validate

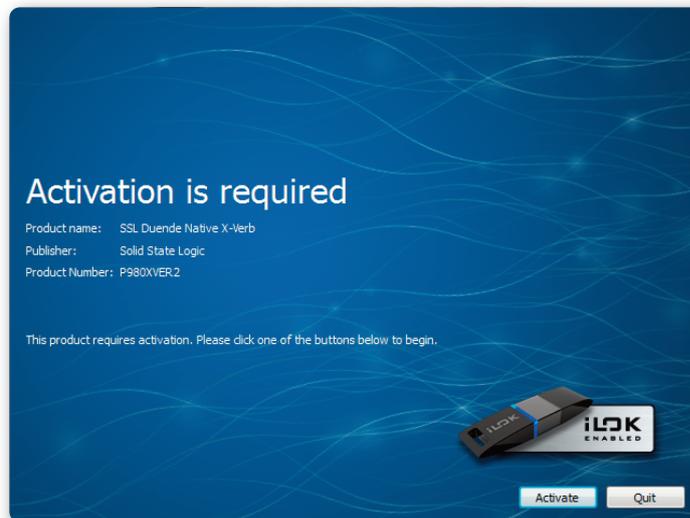
The iLok user account is valid and applicable for SSL plug ins

Migrate all

The licenses will now be available for activation in your iLok account. Please see the 'Activating Plug-in Licenses' section of this user guide for more information.

PLUG-IN EXPIRATION

When your plug-in demo period comes to an end the next time you load a project or instantiate a new instance of a plug-in you will see a window similar to this:

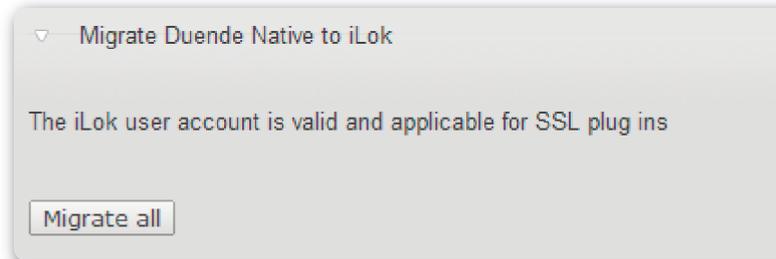


When this happens you will need to purchase the plug-ins to continue using them. This is done through the **MySSL** area of the SSL website, as described on the previous and following pages.

If you load a previously saved project without the appropriate iLok present or with expired demos and press 'Quit' multiple times when you see the above window, the project may load without the plug-ins. In this scenario, be careful not to save the project in its current state as you will lose any plug-in settings! If you need to continue working on the project in this state it is recommended to save it under a new name.

MIGRATING LICENSES

If you already own Duende Native plug-ins using either the Codemeter Virtual Dongle or USB Dongle you can migrate them to iLok 2. In your **SSL MY PRODUCTS** page, press the arrow to the left of the **Migrate Duende Native to iLok** text. If your SSL account is not already associated with your iLok account you should enter your iLok User ID here and validate the iLok account by pressing the **Validate** button. Once validated, the button will change to **Migrate all**. Press this to carry out the migration.



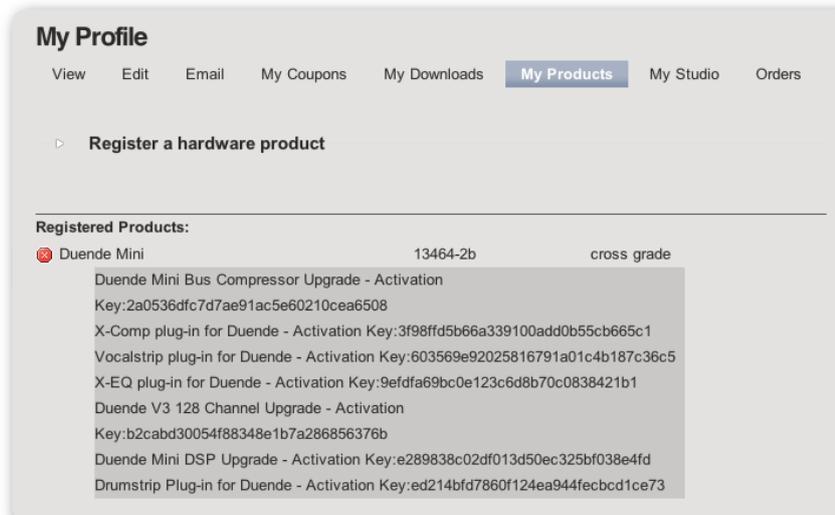
All Codemeter licenses will be migrated at once. It is not possible to migrate only some of your licenses.

The migration may take a few minutes, but the system will inform you of the estimated time. After this time the licenses will be available for activation in your iLok account. Please see the 'Activating Plug-In Licenses' section of this user guide for more information. The Codemeter licenses will disappear from your **MY PRODUCTS** page and be replaced by the equivalent iLok licenses for reference.

If for some reason the migration fails please contact SSL Support here: [SSL Support: Migration](#)

CROSS GRADING FROM DUENDE HARDWARE

If you are a registered owner of Duende Classic, PCI-e or Mini you can cross grade your plug-ins to Duende Native v5. Under **MY PRODUCTS** in your SSL Profile press **Cross Grade** next to the Duende Hardware plug-ins.



If your SSL account is not already associated with your iLok account you should enter your iLok User ID here and validate the iLok account by pressing the **Validate** button. Once validated, the button will change to **Crossgrade all**. Press this to carry out the crossgrade.

All plug-in licenses will be crossgraded at once. It is not possible to crossgrade only some of your licenses.

The crossgrade may take a few minutes, but the system will inform you of the estimated time. After this time the licenses will be available for activation in your iLok account. Please see the 'Activating Plug-In Licenses' section of this user guide for more information. The licenses will be shown in your **MY PRODUCTS** page for reference.

If for some reason the migration fails please contact SSL Support here: [SSL Support: Crossgrade](#)

See this link for Cross Grade Terms and Conditions: [SSL Crossgrade T&C](#)

Your plug-ins should now be installed and activated, and should appear in the plug-in list within your host application. You can run Duende plug-ins simultaneously with your other plug-ins in any combination. Below are the locations of Dende plug-ins in common audio applications:

Logic:

Audio Units > SSL

Cubase/Nuendo:

VST2 – Solid State Logic

VST3 – The plug-ins appear in their respective categories:

Channel:	EQ
Bus Compressor:	Dynamics
Drumstrip:	EQ
Vocalstrip:	EQ
X-EQ:	EQ
X-Comp:	Dynamics
X-Verb:	Reverb
X-Saturator:	Distortion
X-ValveComp:	Dynamics
X-Phase:	Other

Pro Tools:

The plug-ins appear in their respective categories:

Channel:	Dynamics, EQ
Bus Compressor:	Dynamics
Drumstrip:	Dynamics, EQ, Harmonic
Vocalstrip:	Dynamics, EQ
X-EQ:	EQ
X-Comp:	Dynamics
X-Verb:	Reverb
X-Saturator:	Harmonic
X-ValveComp:	Dynamics
X-Phase:	Other

duende /*do-end-ay*/
noun
a quality of passion and inspiration

2. EQ & Dynamics Channel Strip

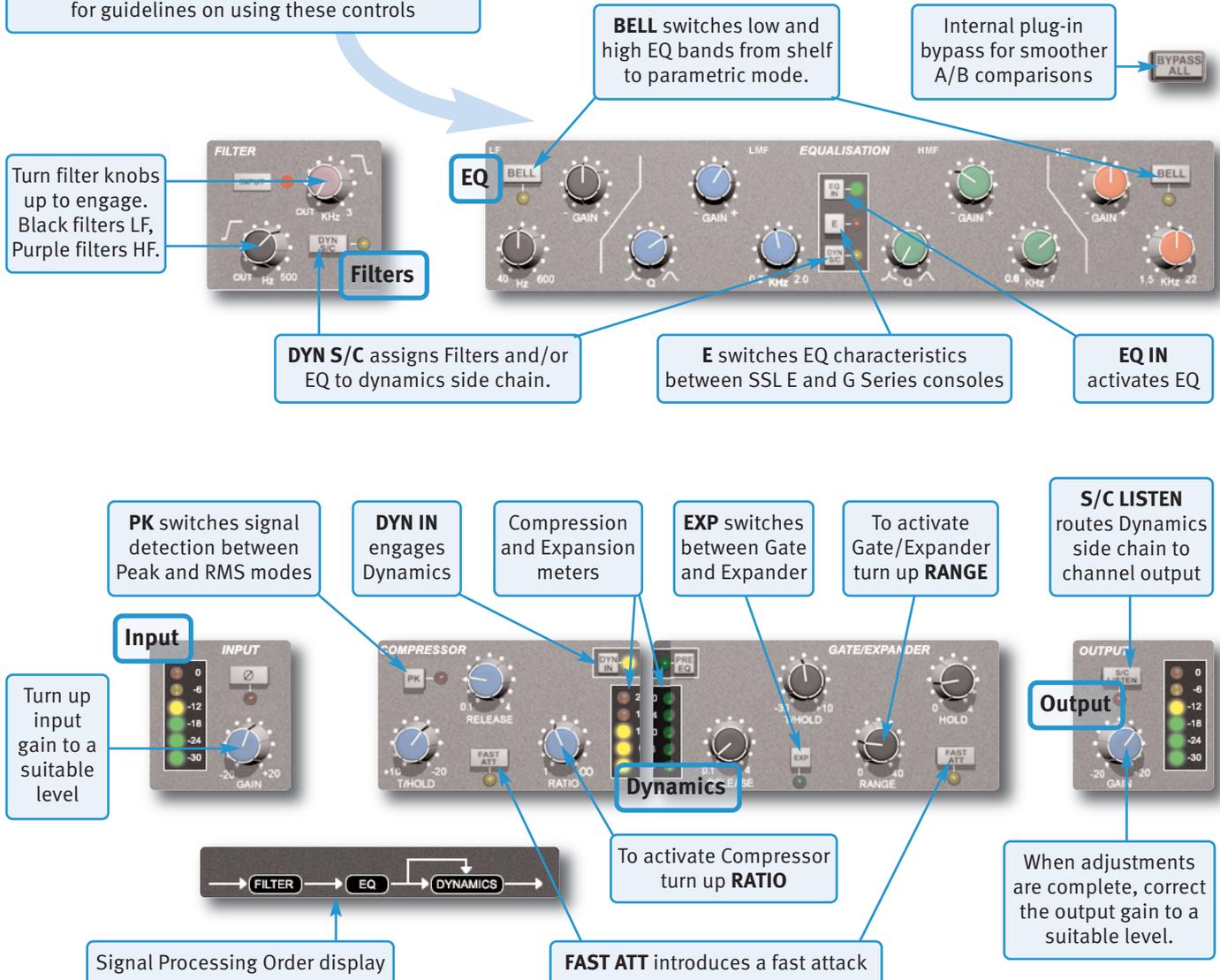
INTRODUCTION

The EQ & Dynamics Channel Strip plug-in provides a complete SSL channel strip, modeled on the XL 9000 K Series console, and includes high and low pass filters, a four band equalizer, compressor/limiter and gate/expander. The channel strip can run in mono or stereo.

Before examining the plug-in in detail, the diagram below provides an operational overview. If you are unfamiliar with any of the operations described, you may also want to view SSL's on-line tutorials (See Page 5 for details). Click on any section of the plug-in to be taken to its description:



Above the main plug-in window is a set of controls which are specific to your host application. Please refer to your host application's user guide for guidelines on using these controls



INTERFACE TECHNIQUES

The following techniques and automation information for controlling the EQ & Channel Strip Plug-in are common to all of the Duende Native plug-ins.

- To press a switch in a plug-in, simply click on it. With all plug-ins, an on-screen indication adjacent to or surrounding the switch will indicate when the switch is active.
- To turn a knob, click on it, and drag it up and to the right (as shown right). If your mouse has a scroll wheel, you can also turn knobs by hovering over them and turning the scroll wheel.
- To move a knob slowly for fine adjustments, hold the following on your keyboard whilst turning/scrolling:



To reset a knob to its default value, click on the knob whilst holding the following on your keyboard:

Fine Adjust for Host Type	Mac	Windows
AU	Shift	n/a
VST or VST3	Shift	Shift
AAX	Shift	Shift
RTAS	⌘ ('Command')	Ctrl

Reset for Host Type	Mac	Windows
AU	⌘ ('Command')	n/a
VST or VST3	⌘ ('Command')	Ctrl
AAX	⌘ ('Command')	Ctrl
RTAS	Alt	Alt

- To view the value for any knob, hover over the knob cap with the mouse. The current value will appear immediately above the knob (shown right).



Automation

Every plug-in parameter can be automated in host applications which support automation. The method for recording and editing automation varies from host to host. For specific instructions on using automation within the host, consult the host application documentation.

INPUT SECTION

Turn the **GAIN** knob to control the level of the incoming audio signal. The post-gain signal level is shown to its left. As a rough guide, the '-6' yellow indicator should occasionally come on but the red 'o' indicator should remain off.

Press **Ø** to invert the phase of the input signal.



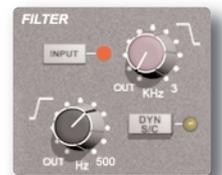
FILTER SECTION

There are two filters in the Filter section:

- The black knob controls an 18dB/Octave high pass filter (20Hz to 500Hz).
- The purple knob controls a 12dB/Octave low pass filter (3kHz to 22kHz).

Filters are inactive when turned fully left (**OUT**). Turn them clockwise to move the filter frequency in from its extremity.

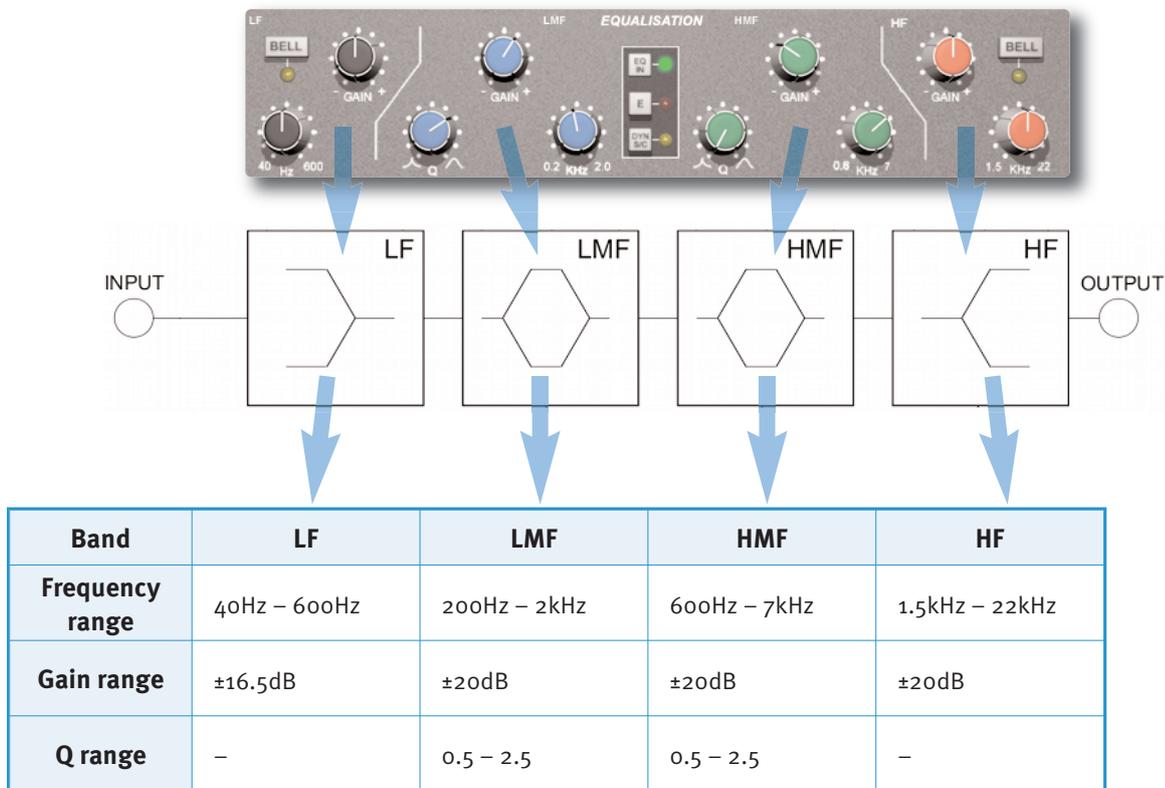
To place the Filters straight after the input control, press **INPUT** (see previous page), and to switch the Filters into the Dynamics sidechain, press **DYN SC** (see Page 27 for more info). Note that these switches cancel each other out.



EQUALISER SECTION

To use the EQ, switch it into circuit by pressing the **EQ IN** switch.

The EQ section has four bands, each with its own knob colour. All bands have gain and frequency control. The low (LF) and high (HF) bands are shelved by default (as shown below) but can be switched to a bell shape (parametric) by pressing the **BELL** button. The low-mid (LMF) and high-mid (HMF) bands also have Q control:



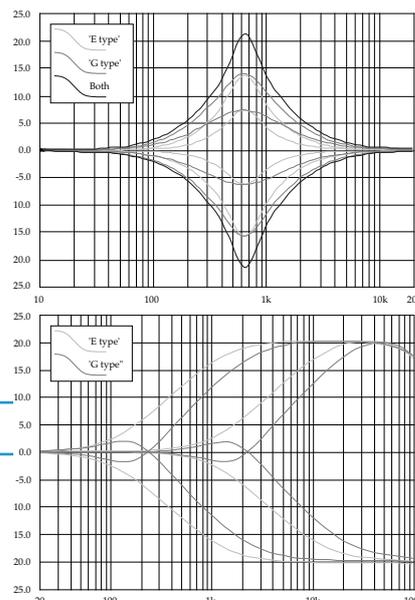
Press the **E** to switch the EQ emulation from G Series to E Series consoles. The diagrams to the right display the difference between them:

G Series: The bell curve has a more rounded shape at low gains, and the shelf curve overshoots zero slightly at the base of the curve. G Series EQ is more subtle and is generally more suited to instruments and vocals.

E Series: The bell curve is slightly more pointed, and there is no overshoot on the shelf curve. E Series EQ is more aggressive and is therefore better for removing problem frequencies. It is generally more suited to drums.

At full boost or cut both curves are identical.

To switch the EQ into the Dynamics sidechain, press **DYN SC** (see Page 27 for more info).



DYNAMICS SECTION

The Dynamics section comprises a Compressor/Limiter and an Expander/Gate. Both sections work independently but can be operational at the same time, providing sophisticated control of signal levels. The diagrams down the right-hand side of the page give a simplified overview what each control represents – If you are unfamiliar with any of the operations described, you may want to view SSL's on-line tutorials (See Page 5 for details).



Press **DYN IN** to switch the Dynamics into circuit. To place the Dynamics before the EQ in the signal chain, press the **PRE EQ** switch (See Page 27 for more info on processing order).

Compressor/Limiter



To activate the Compressor/Limiter, turn the **RATIO** knob so that it's ratio is no longer 1:1.

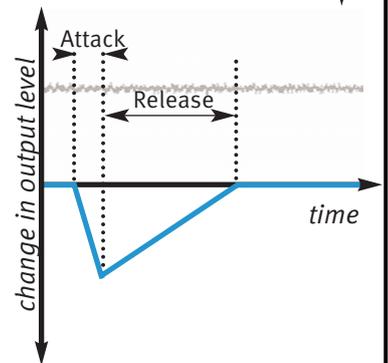
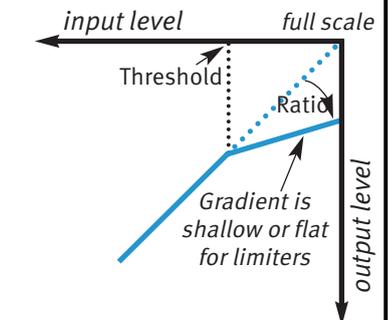
To turn the compressor into a ∞ :1 limiter, turn the knob fully to the right.

There is no gain makeup control as the **T/HOLD** (threshold) knob controls both the level at which gain reduction is introduced and the gain make-up, keeping the output level steady regardless of the compression.

RELEASE controls how quickly the level returns to normal after the input level has dropped below the threshold (measured in seconds). The attack time is adjusted automatically to match the audio. To choose a constantly fast attack time, press the **FAST ATT** switch.

Press the **PK** button to switch from RMS to peak signal detection. In normal RMS mode the compressor reacts to the average signal level, and has a soft knee characteristic. When switched to peak mode, it responds to peak signal level and introduces a hard knee characteristic, resulting in more dramatic compression.

The level of compression being introduced is shown in the left-hand of the two meters in the centre of the Dynamics section.



Expander/Gate



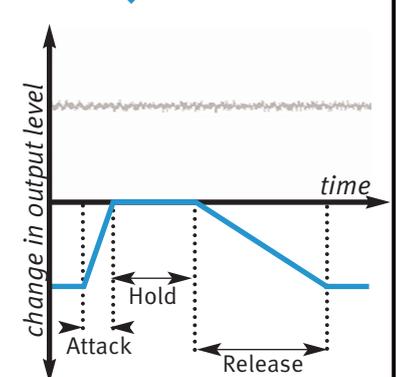
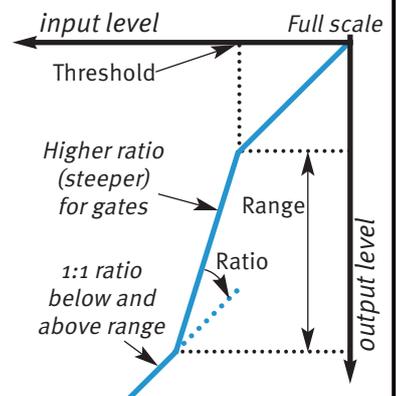
To activate the Expander/Gate, turn the **RANGE** knob so that it's range is no longer zero. The green indicators in the right-hand of the two meters in the centre of the Dynamics section show the amount of gain reduction being introduced.

By default, the Expander/Gate section functions as a gate. To switch to the expander, press the **EXP** switch.

The **THRESHOLD** function uses different levels to 'open' the gate to audio and to 'close' it again – the level at which the expander opens is higher than the level at which it closes again. In other words, when the expander is opened, it stays open until the signal level crosses the quieter 'close' threshold. This is known as hysteresis and is very useful as it allows instruments to decay more naturally. The word 'Threshold' normally refers to the 'open' threshold.

HOLD controls the delay before the signal level starts reducing again, and **RELEASE** controls how quickly the level then reduces. Note that the **RELEASE** interacts with the **RANGE**, which determines the depth of gain reduction.

The attack time (the time taken for the Expander/Gate to 'recover' once the signal level is above the 'deactivate' threshold) is normally set to 1.5ms per 40dB. Press the **FAST ATT** switch to introduce a faster attack time of 100µs per 40dB. This is useful when gating signals with a steep rising edge, such as drums.



Dynamics Values

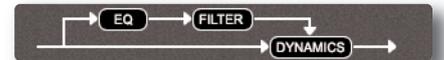
The value ranges associated with the dynamics controls are as follows:

	Ratio	Threshold / Range	Release time	Attack time
Compressor	1:1 to ∞:1	-20dB to +10dB	0.1s to 4s	3ms to 30ms (automatic) 3ms in Fast Attack
Expander	Gate: 40:1 Expander: 2:1	30dB to +10dB Range: 0dB to 40dB	hold 0s to 4s release 0.1s to 4s	1.5ms / 40dB 100µs / 40dB in Fast Attack

Side Chain Processing Order

The EQ and filter sections can be assigned to the Dynamics sidechains, allowing for advanced processes like de-essing. This is done using the **DYN S/C** switches in the respective sections.

The Process Order display at the base of the plug-in window indicates the sidechain assignments, as shown to the right. Both EQ and filter sections can be assigned to the sidechain together, in which case the EQ precedes the filter.



To listen to the signal feeding the side chain, press the **S/C LISTEN** button in the Output section to route the side chain signal to the channel output.

*Remember to cancel the **S/C LISTEN** button once you have finished auditioning the side chain!*

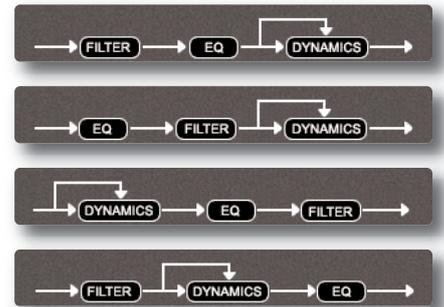
CHANNEL PROCESSING ORDER

Signal processing order is shown in the graphic at the bottom of the plug-in window. The default plug-in order is **Filters > EQ > Dynamics**.

To place the after the EQ section, deactivate the **INPUT** switch in the Filters section. The **INPUT** indicator should go off.

To place the Dynamics before the EQ, press the **PRE EQ** switch in the Dynamics section.

When the **INPUT** and **PRE EQ** switch are active simultaneously, the processing order becomes **Filters > Dynamics > EQ**.



OUTPUT SECTION



The Output section allows you to ensure that the signal retains a good level after all the signal processing. The signal level is shown to the right of the knob. As a rough guide, the '-6' yellow indicator should occasionally come on but the red '0' indicator should remain off.

S/C LISTEN routes the sidechain directly to the output, so you can monitor the sidechain signal.

Blank. Reserved for text that never was.

3. Stereo Bus Compressor

INTRODUCTION

The Stereo Bus Compressor plug-in is a stereo version of the legendary centre section stereo bus compressor found on the XL 9000 K Series console. It provides high quality stereo compression, giving you critical control over the dynamic range of audio signals.

The compressor can be run in mono or stereo modes and can be used for practically any application that requires superior compression. For example, place it over a stereo mix to 'glue' the mix together whilst still maintaining a big sound, or use it on drum overheads or whole drum kits for very effective control of drum dynamics.

INTERFACE TECHNIQUES

The interface techniques for the Stereo Bus Compressor are identical to those for the EQ & Dynamics Channel Strip (see Page 20).



Control Parameters

Below is a description of the Stereo Bus Compressor's parameters, as shown in the diagrams to the right. If you are unfamiliar with how to use any of the parameters described, please refer to SSL's on-line tutorials (See Page 5 for details).

Compression Meter
Shows gain reduction in dB

THRESHOLD
Controls level at which gain reduction is introduced.
Continuously variable:
-20dB to +20dB.

ATTACK
Controls response time when Threshold is crossed.
Choose between:
0.1, 0.3, 1, 3, 10 and 30ms.

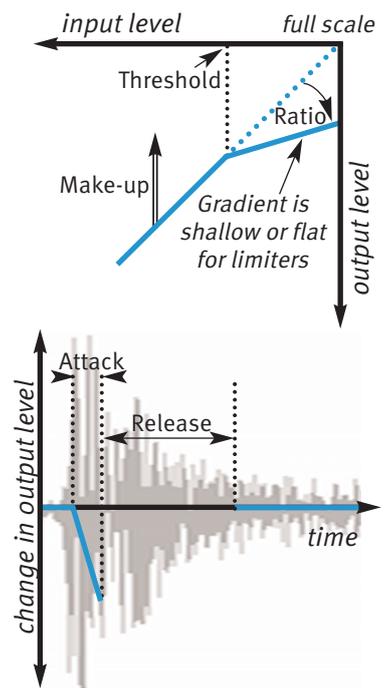
RATIO
Controls the degree of compression.
Choose between:
2:1, 4:1 and 20:1.



MAKE-UP
Controls level compensation to offset compressor action.
Continuously variable:
-5dB to +15dB.

RELEASE
Controls how quickly level returns to normal.
Choose between
0.1, 0.3, 0.6, or
1.2 seconds, or Auto.
Auto: release time is dependant upon duration of signal peak.

COMP IN
Switches the compressor in and out of the signal path.



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4. X-Comp

INTRODUCTION

The illustration below gives an overview of some of the X-Comp features which are described in full over the following pages. Click on the blue boxes to be taken to fuller descriptions of their contents.

The screenshot shows the X-Comp plugin interface with several callout boxes:

- Interactive compressor display with draggable nodes:** Points to the Compression Law graph.
- IO Diff display displays the signal's dynamic range:** Points to the IO Diff waveform display.
- GR History displays gain reduction over last second:** Points to the GR History graph.
- Defeat button for auditioning bleed signal:** Points to the Defeat button in the Makeup section.
- Above the main plug-in window is a set of controls which are specific to your host application. Please refer to your host application's user guide for guidelines on using these controls:** Points to the In/Out gain meters.
- Internal bypass for smooth in/out comparisons:** Points to the A/B switch.
- Input level metering and control:** Points to the Input Gain meter.
- Switch between main and alternative plug-in settings:** Points to the A/B switch.
- Bleed section allows LF and HF bands to bypass the compressor. Display includes draggable nodes:** Points to the LF and HF Bleed frequency sliders.
- Output level metering and control:** Points to the Output Gain meter.

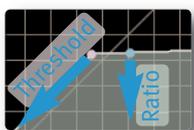
INTERFACE TECHNIQUES

The basic interface techniques for the X-Comp are largely identical to those for the EQ & Dynamics Channel Strip (see Page 20). In addition to these basic techniques, the following are also available:

To control **Input** and **Output** levels, click and drag upwards (see right).



To enter a precise value for any parameter, double-click on its value display, enter a value on your computer keyboard, and press the *Return* key.



Threshold and **Ratio** values can also be controlled directly within the Compression Law display. Move the pink node along its line to control the Threshold, and move the blue node up and down to control the ratio.

The bleed bands can be adjusted in a similar way, as described on Page 33.



Automation

Automation support for X-Comp is the same as for the EQ & Dynamics Channel Strip (see Page 20).

CONTROL PARAMETERS

Plug-in Bypass

The **In/Out** switch located above the **Input** section provides an internal plug-in bypass. This allows for smoother In/Out comparisons by avoiding the latency issues associated with the host application's Bypass function.



The button must be 'lit' for the compressor to be in circuit (as shown right).

Presets

  The **Load** and **Save** buttons at the base of the screen provide quick access to the host application's Import and Save Settings functions. For specific instructions on using presets within the host, consult the host application documentation. Presets use a standard file format which allows them to be moved between a variety of platforms and host applications.

A file of ready-made X-Comp presets are included in the plug-in installation. These provide starting points for a variety of instruments and effects. To load SSL presets, click on the **Load** button and navigate as follows:

Macintosh:	/Applications/Solid State Logic/Presets/X-Comp
Windows 32-bit:	c:\program files\solid state logic\presets\X-Comp
Windows 64-bit:	c:\program files (x86)\solid state logic\presets\X-Comp

A-B comparisons

The **A B** button at the base of the screen allows you to load two completely independent set ups and compare them quickly. When the plug-in is opened, setting A is selected by default, as shown right. Pressing the **A B** button will switch between setting A and setting B.



You may wish to use this function to compare two very similar settings or to keep a 'maybe' setting in reserve if your current one does not work out.



Common Compressor Parameters

The diagrams down the right-hand side of the page give a simplified overview what each control represents, as shown in the **Compression Law** display in the centre of the plug-in window. If you are unclear about how the **Thresh** (threshold), **Ratio**, **Attack**, **Release** and **Make up** work, please refer to SSL's on-line tutorials (See Page 5 for details).

Knee

The **Knee** controls how focussed the threshold level is:

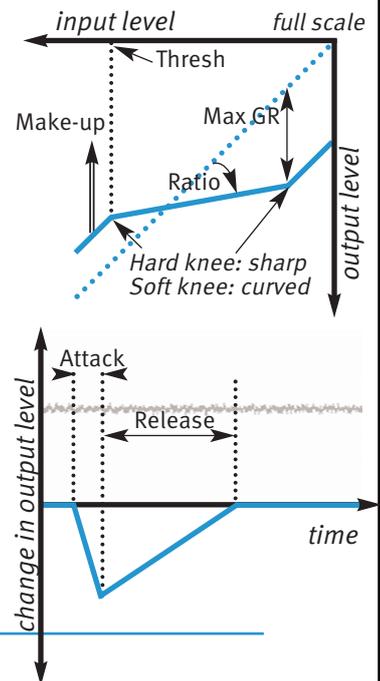
- With a hard knee (knob at minimum) the compressor's parameters all come into force at precisely the point at which the threshold is crossed. In the **Compression Law** diagram, this is shown by a sharp change in gradient at the threshold.
- With a soft knee, the ratio is introduced gradually, starting below the threshold and reaching its full value above the threshold. In the **Compression Law** graph, this is shown by a curve in the gradient around the threshold.

The **Knee** control affects both the main threshold point and the **Max GR** point (see below) in equal measure.

Operational tip. A hard knee allows for greater precision, but can sound more obvious. A soft knee generally provides a more transparent result.

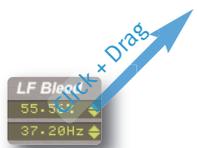
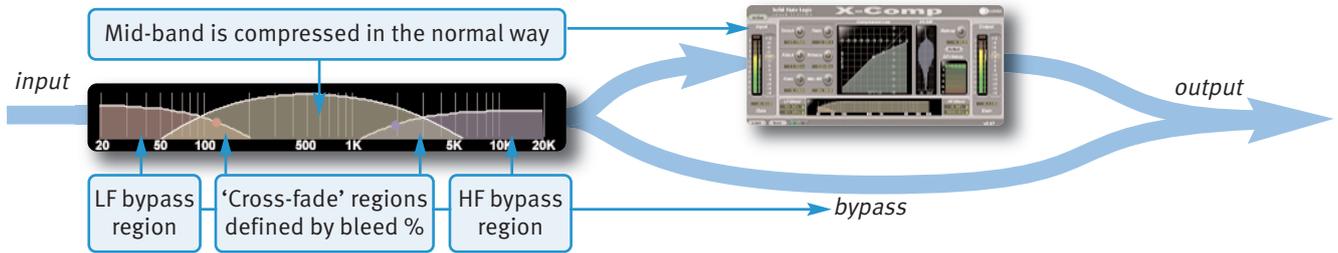
Max GR

The **Max GR** (Maximum Gain Reduction) allows you to set a limit on how much gain reduction can be introduced, replicating the performance of older optical compressors (as shown in the upper diagram above). With a Max GR of 20dB, for example, any signals that would normally be reduced by more than 20dB will be attenuated by 20dB. By only compressing the middle of the dynamic range you can exert some general dynamic control whilst still preserving the affect of a signal's peaks.



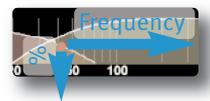
Bleed

The Bleed controls, located at the bottom of the X-Comp plug-in window, allow you to remove the compressor's effect from high and low frequencies. Frequencies outside the mid-band bypass the compressor and are recombined with the compressed signal at the output. Note that the bypassed signal is not affected by the **Make up** gain.



The frequencies and slope of the bleed-band filters are indicated either side of the graph; LF to the left and HF to the right. The slope is measured in percentage.

The bleed bands can be adjusted by clicking on the values in the displays and dragging up and right in the usual way. Alternatively, they can be adjusted directly using the nodes in the display – brown for low frequencies and blue for high frequencies. Move the nodes horizontally to control frequency and vertically to control percentage.



Auditioning Bleed Bands

To listen to the bleed bands on their own, press the **Defeat** button, located below the **Make up** gain. This mutes the compressed signal, so only the bypass portion of the signal is heard.



COMPRESSOR VALUES

The table below lists the ranges of all of X-Comp's parameters:

Parameter	Min	Max
Input gain	-36dB	+12dB
Threshold	-48dB	+12dB
Ratio	1:1	50:1
Knee	0dB	40dB
Max GR	20dB	60dB
Attack time	0.5ms	100ms
Release time	1ms	2000ms
Make up gain	-6dB	+36dB
LF Bleed	30Hz	300Hz
HF Bleed	2kHz	12kHz
Bleed slopes	0%	100%
Output gain	-36dB	+12dB

Input and Output Sections

The input and output sections at either side of the plug-in window provide input and output gain control, along with displays of the following information:

Clip Light

Indicates that the signal is clipping. It clears after 1 second, or click on it to clear manually.

Peak Meter

Allows you to observe the exact level at any point in time. There are two meters in stereo and one in mono.



Range Meter

The thin white bracket indicates the signal's recent dynamic range. Longer brackets indicate a larger dynamic range. If the bracket edges are moving quickly, this implies an 'energetic' signal.

RMS Meter

The thinner bar to the right shows the RMS (average) level over time.

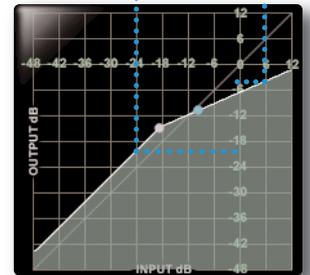
Compression Law Graph

The display in the centre of the plug-in window compares the input and output signals given the current compressor settings. The thicker white line indicates the relationship between input and output across the range of levels, while the faint line indicates a 1:1 relationship. In the example to the right, an input level of -24dB results in an output level of -20dB (as this is below the threshold, this indicates a Make-up gain of 4dB), while an input level of $+6\text{dB}$ results in an output level of -4dB .

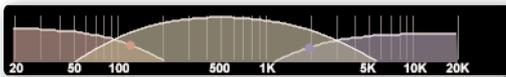
Threshold and Ratio can be defined by moving the nodes within the display: Move the pink node along its line to control the Threshold, and move the blue node up and down to control the ratio.

In: -24dB
Out: -20dB

In: 6dB
Out: -4dB



Bleed Graph



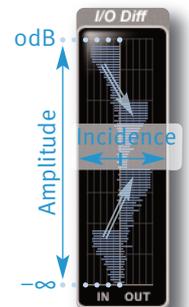
The bleed graph indicates the frequencies which are bypassing the compressor, as described on the previous page.

I/O Diff Graph

The I/O Diff meter to the right of the **Compression Law** graph shows how often each level occurs within the input and output signals. The input is shown on the left and the output on the right.

The vertical scale is amplitude, with 0dB at the top and $-\infty$ at the bottom. The length of each line protruding from the centre represents the number of incidents of that amplitude over a period of seconds.

In the graphic to the right, the input signal displays a mixture of very loud and very quiet, indicating that the signal has a large dynamic range which is changing very quickly. The output signal indicates that this dynamic range has been made smaller, with the loud bars being reduced in level and the quiet bars being increased.



GR History

The **GR History** (Gain Reduction History) meter shows the current gain reduction and how it has fluctuating.

The thicker line towards the left of the display shows how much gain reduction is currently being used, while the thinner lines to its right show how this has been changing over the past second.



Colour of knobs



You may also notice that the colour of the knob surrounds change to indicate their level – green for low, yellow for mid, red for high.

5. X-EQ

INTRODUCTION

X-EQ is a 10-band highly configurable EQ plug-in. The six central bands have bell curves, the second and ninth bands are shelf bands, while the first and last bands are filters. The illustration below introduces some of the X-EQ plug-in features which are described in full over the following pages. Click on the blue boxes to be taken to fuller descriptions of their contents. If you are unclear on how to use an EQ, then it may be worth starting with the SSL on-line tutorials (See Page 5 for details).

[? click here for on-line tutorials](#)

Above the main plug-in window is a set of controls which are specific to your host application. Please refer to your host application's user guide for guidelines on using these controls

Internal bypass for smooth in/out comparisons

Interactive EQ display with draggable nodes

Input level metering and control

Switch between main and alternative plug-in settings

Switches between Serial and Parallel signal-flow

Activates bands and assigns them to Edit area

Displays graph of frequency content

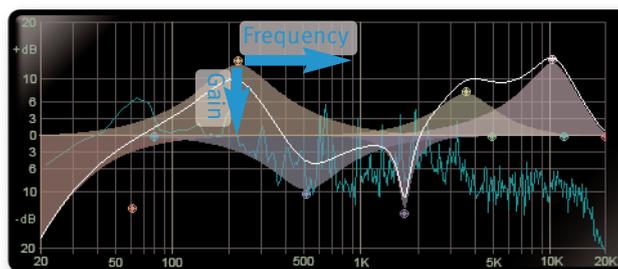
Edit area for the band selected below the main graph

Output level metering and control

INTERFACE OVERVIEW

EQ Graph Display

The EQ graph provides visual feedback for a number of things:



The white line across the graph shows the frequency response of the current EQ settings.

The nodes represent the settings of each individual band, and can be dragged to change the settings:

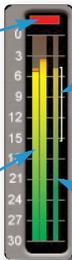
- For the eight central EQ bands, move the nodes left and right to control Frequency, or up and down to control Gain.
- For the filter bands (Bands 1 and 10), move the nodes left and right to control Frequency or up / down to control Slope.

Each band is colour coded, with the graph node colour matching the coloured dots in their band box below the graph. Each coloured shade area represents the impact of the node with that colour.

The real-time refluency content of the processed signal can be displayed by clicking the **Analyse** button below the display

Input and Output Sections

The input and output sections at either side of the plug-in window provide input and output gain control, along with displays of the following information:



Clip Light
Indicates that the signal is clipping. It clears after 1 second, or click on it to clear manually.

Range Meter
The thin white bracket indicates the signal's recent dynamic range.

Peak Meter
Allows you to observe the exact level at any point in time. There are two meters in stereo and one in mono.

RMS Meter
The thinner bar to the right shows the RMS (average) level over time.

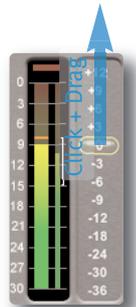
Interface Techniques

The basic interface techniques for the X-EQ are largely identical to those for the EQ & Dynamics Channel Strip (see Page 20). In addition to these basic techniques, the following are also available:

To control **Input** and **Output** levels, click and drag up- or downwards (see right).



To enter a precise value for any parameter, double-click on its value display, enter a value on your computer keyboard, and press the Return key.



You may notice that the colour of the knob surrounds change to indicate their level: (green for low, yellow for mid, red for high).



Band 'on' switch



To switch each band on, click on the oblong switch at the top of each band box.

The edit controls to the right of the EQ graph control one of the 10 EQ bands. The band being controlled is selected by clicking on the appropriate band box, located in a row underneath the graph, or by clicking on the band's node in the EQ graph.

The band assigned to the edit controls is indicated by the band box being 'lit' with its band colour, and by the band number and colour appearing at the top of the edit controls, as shown right for band 6.



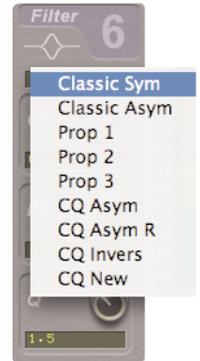
Automation

Every plug-in parameter can be automated in host applications which support automation. The method for recording and editing automation varies from host to host. For specific instructions on using automation within the host, consult the host application documentation.

EQ CONFIGURATION

When one of the band boxes below the EQ graph is selected, its band is assigned to the band editing controls to the right of the graph.

- **Freq** (frequency) control is provided for each band.
- **Gain** and **Q** control is provided for all bands apart from the filter bands (Band 1 and 10).
- The filter bands have slope control in place of the Q control.
- In addition, you can select from a number of filter shape types by clicking on the text box above the gain control and selecting a shape from the drop-down list which appears. These shapes are described on Page 39 and following. Note that there are no shape types available for the shelving bands (Bands 2 and 9).



Plug-in Bypass

 The **In/Out** switch located above the **Input** section provides an internal plug-in bypass. This allows for smoother In/Out comparisons by avoiding the latency issues associated with the host application's Bypass function.

The button must be 'lit' for the compressor to be in circuit (as shown left).

Presets

The **Load** and **Save** buttons at the base of the screen provide quick access to the host application's Import and Save Settings functions. For specific instructions on using presets within the host, consult the host application documentation. Presets use a standard file format which allows them to be moved between a variety of platforms and host applications.



A file of ready-made X-EQ presets are included in the plug-in installation. These provide starting points for a variety of instruments and effects. To load SSL presets, click on the **Load** button and navigate as follows:

Macintosh:	/Applications/Solid State Logic/Presets/X-Eq
Windows 32-bit:	c:\program files\solid state logic\presets\X-Eq
Windows 64-bit:	c:\program files (x86)\solid state logic\presets\X-Eq

A-B comparisons

 The **A B** button at the base of the screen allows you to load two completely independent set ups and compare them quickly. When the plug-in is opened, setting A is selected by default, as shown right. Pressing the **A B** button will switch between setting A and setting B.

You may wish to use this function to compare two very similar settings or to keep a 'maybe' setting in reserve if your current one does not work out.

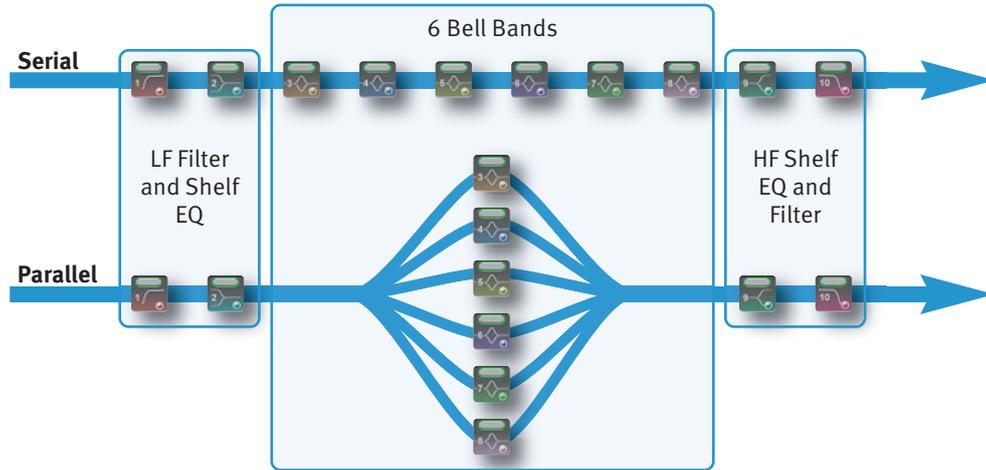
Analyse

If the **Analyse** button is pressed, a display of the input signal's frequency response is shown within the EQ graph.

This allows problem frequencies to be pinpointed visually and filtered out by moving the nodes within the graph, such as in the example below. As the analyser is placed post-EQ, you can see what effect the processing has on the frequency spectrum.

Parallel

The parallel button changes the order in which the six central bell-shaped bands are processed. In normal (serial) mode, they are processed one after the other. In parallel mode, once the signal has passed through the LF filter and shelf bands, it is split off and routed through each bell band simultaneously, before being combined again for the HF shelf and filter bands:



The parallel EQ is modelled on the old 'Parallel Passive' EQ units, and the different style of band interaction means that the EQ performs quite differently. See the EQ History section on Page 41 for more information.

EQ PARAMETER VALUES

Band	Gain	Frequency	Q
HP Filter	n/a	20Hz – 1020Hz	0 – 48dB/oct in 6dB steps
Low Shelf	±20dB	20Hz – 1020Hz	0.3 – 10.3
Midband 1	±20dB	20Hz – 20kHz	0.3 – 10.3
Midband 2	±20dB	20Hz – 20kHz	0.3 – 10.3
Midband 3	±20dB	20Hz – 20kHz	0.3 – 10.3
Midband 4	±20dB	20Hz – 20kHz	0.3 – 10.3
Midband 5	±20dB	20Hz – 20kHz	0.3 – 10.3
Midband 6	±20dB	20Hz – 20kHz	0.3 – 10.3
High Shelf	±20dB	500Hz – 20kHz	0.3 – 10.3
LP Filter	n/a	500Hz – 20kHz	0 – 48dB/oct in 6dB steps

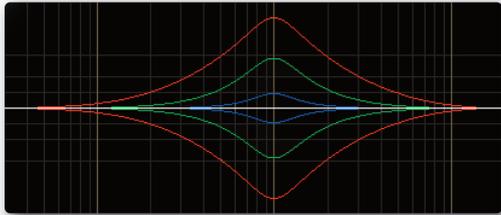
EQ AND FILTER SHAPES

Bell Shapes

Q Definitions:

- P – 3dB** Classical definition that means Q is measured 3dB below peak for boost or 3dB above peak for cut.
- o – 3dB** Definition used for bells in some US products, the bandwidth for Q calculation is measured 3dB below 0dB line for cut or 3dB above for boost. Q in both definitions correct only for +6dB boost/cut.
- P/2** New ‘musical’ definition based on the bandwidth measurement in the middle of a bell filter, between peak and 0dB line.

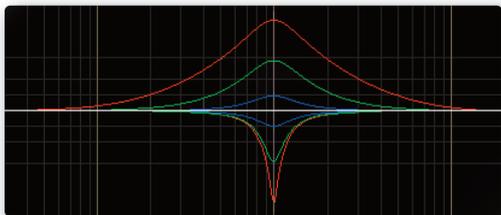
Normalisation 12 equalisers are normalised to have exactly the same bell shape for +6dB boost.



Classic Symmetrical (Classic Sym)

The most popular parametric EQ shape used in various mixing consoles and outboard gear. Almost constant Q characteristic.

P – 3dB

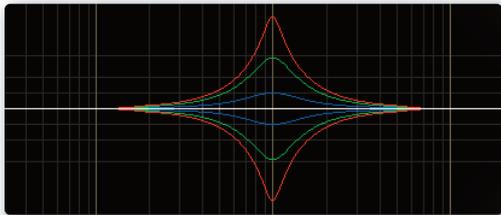


Classic Asymmetrical (Classic Asym)

Often features in older equalisers. Boost as above but much narrower cut characteristic.

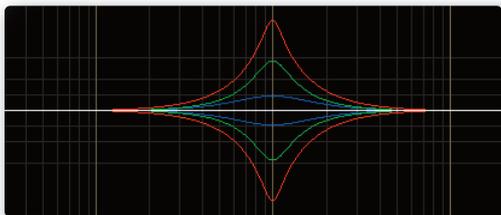
P – 3dB boost

o – 3dB (x2) cut



Proportional 1 (Prop 1)

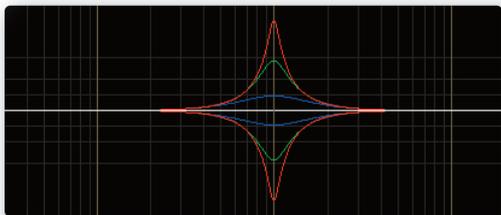
Proportional equalisers are recognised as being more ‘musical’ than Constant Q. The bells are wider below +6dB and narrower above +6dB (or –6dB for cut).



Proportional 2 (Prop 2)

Like Proportional 1 but with larger changes below and above +6dB peak (or –6dB for cut).

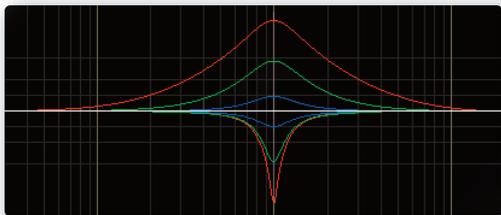
P – 3dB



Proportional 3 (Prop 3)

Like Proportional 1 but with extra widened bells between 0 and +3dB (or –3dB for cut).

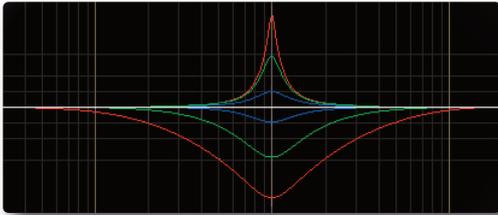
P – 3dB



Constant Q Asymmetrical (CQ Asym)

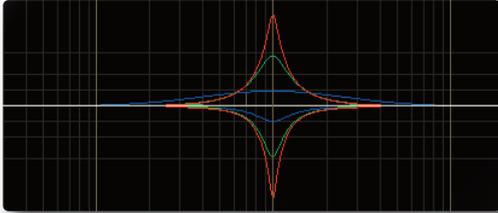
Used in some US products and mixers. The boost characteristics are defined according to o – 3dB formula. Extra widened between –3dB and 0dB.

P – 3dB boost, o – 3dB cut



Constant Q Asymmetrical Reverse (CQ Asym R)

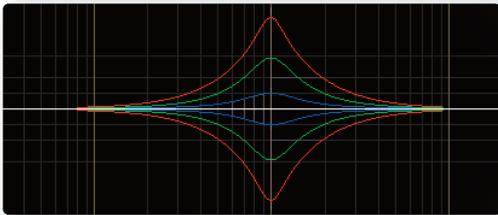
Exactly like above but with mirrored boost and cut characteristics.



Constant Q Invert (CQ Invert)

Both boost and cut characteristics are both defined according to 0 – 3dB formula.

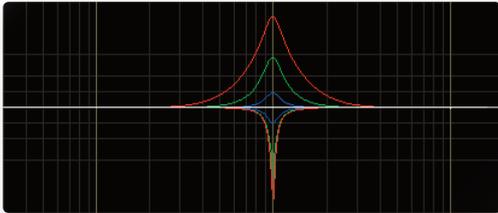
0 – 3dB



Constant Q New (CQ New)

Defined Q is always perfectly maintained independent of bell gain (also below 6dB). In terms of P – 3dB definition of this equaliser can be classified as proportional.

P/2



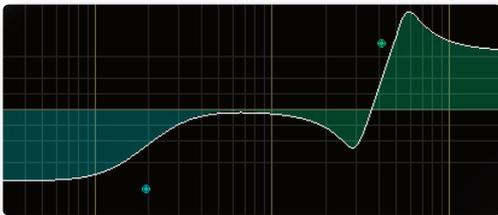
Parallel

Note this is accessed via the Parallel switch, not the drop-down menu.

Recreates the passive LC parallel equaliser with all its advantages (sound) and disadvantages (band interaction, asymmetry). As found in graphic equalisers.

3dB boost, 0 – 3dB (x2) cut

Shelving Bands

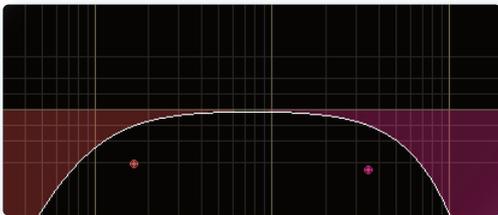


Low and High shelves

Q value is used to control overshoot characteristic. On the left is the Low shelf filter with a low Q value, on the right the High shelf filter exhibits overshoot with a large Q value.

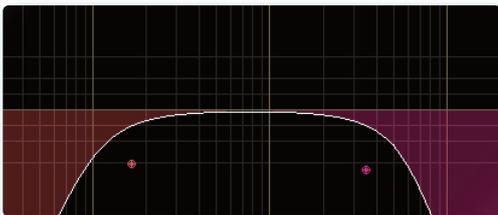
Note that there are no alternative shapes for shelving bands

Filter Shapes



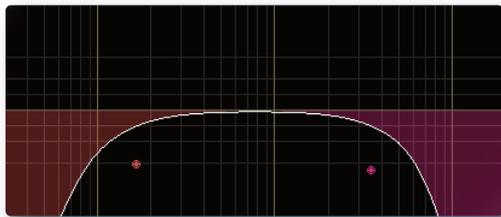
Critical

‘Critical Damped’ filters simulate a chain of passive analogue RC (for highcut) and CR (for low-cut) stages fixing a behaviour similar to a series of RC elements in vintage analogue equipment.



Bessel

Linear phase behaviour leads to no overshoot or ringing resulting from a sudden transition between signal levels. The drawback is a sluggish roll-off rate.



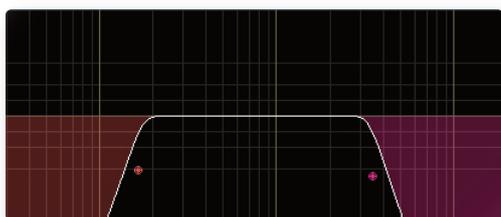
Gaussian

No ringing or overshoot in the time domain, but slow roll-off in the frequency domain.



Butterworth

Characterised by having a maximally flat magnitude response, ie. no amplitude ripple in the passband.



Chebyshev

Characterised by having an equiripple magnitude response, meaning the magnitude increases and decreases regularly from DC to the cutoff frequency.

EQ HISTORY

An Audio Engineer's Best Friend

The equaliser is the oldest and the most popular sound processing tool. From the earliest days, its main function has been to correct or enhance sound by boosting or cutting certain frequency ranges. Engineers have developed countless equalisers for over 50 years and some of them became legendary and were considered bench marks. The most popular type of EQ in recording and post-production studios is the parametric equaliser or PEQ. It offers maximal flexibility due to direct access to all relevant filter parameters. Properly used the PEQ is a very powerful tool and the best friend of every sound engineer in the battle for perfect sound. If misused, it can be the greatest enemy of any recording.

The Best of the Analogue and Digital Worlds

X-EQ is a creative equalising tool combining the best of both the analogue and digital worlds. We recreated the most legendary analogue equalisers and added a few experimental characteristics only possible in the digital domain. By using proprietary filter algorithms, we have achieved a huge dynamic range as well as extremely low noise and distortion level, and thus unparalleled sound purity; impossible with any analogue circuitry.

Analogue Parametric EQ Modelling

In today's era of digital audio workstations, hundreds of software parametric equalisers are available. Many of them are intended to be "THE best sounding equaliser ever". The truth is that only few of them are recognised and adored by the experts. You may ask why one equaliser sounds great while another does not. This question is almost as old as the equaliser itself and still is not completely answered. On the one hand there are some obvious rules which must be followed when designing a good sounding PEQ, on the other hand some never really proven esoteric claims driven by marketing departments or self-nominated audio evangelists. Especially treasured analogue equalisers are considered by some people as being absolutely unique and unmatched by any 'dirty' digital equaliser. The truth is that with a properly designed, fully parametric analytic EQ every amplitude and phase characteristic of any other equaliser setup can be recreated. Of course, the contribution of distortions to the specific sound of a particular analogue equaliser caused by the respective electronics has to be considered. If the distortions are 'good', they may make certain applications sound better. Usually however, 'bad' non linear distortions and other deficiencies like limited dynamic range are surely not responsible for a 'magic' sound. Therefore our policy in digital PEQ design has always been to make the equaliser filters as precise and clean as possible. For controlled generation of distortions we recommend using enhancers or any other specialised processors with proper built-in anti-aliasing technology.

Non Linear-phase EQ

Phase shift in equipment is probably one of the most misunderstood topics in audio. Phase shifts often get blamed for anything that is not classed as typical distortion. The reality is that all analogue EQs produce a deviation from linear phase which is specifically related to the shape of the EQ curve, and it is precisely these phase changes which produce the required gain reductions. This is a very natural process which can be found in the audio characteristics of rooms, materials and elsewhere.

Though it is not possible for us to detect phase in isolation, the phase shift of an EQ can often be the cause of the ‘magical’ sound of that device once the EQ signal is added back into the mix. For example, the Duende channel EQ derived from the SSL analogue consoles – plenty of phase shift and plenty of magic!

In contrast, a linear-phase EQ delays all frequencies by the same amount. This can only be achieved in the digital domain by using special techniques. While this process is very effective in certain situations (eg. ‘surgically’ removing troublesome frequencies), it should not be considered as a superior tool for the majority of EQ tasks.

Linear-phase equalisers have become popular over the past few years due to the proliferation of powerful digital processors. We decided not to go down that route with X-EQ and we think you’ll agree with that decision. Maybe linear-phase EQ is something we will explore in the future but... who knows? An equaliser not only changes the relative frequency balance of a signal but also its phase. If you EQ a sound until it sounds right or natural, then the chances are that you are EQing to make it sound in phase with the rest of the mix. We need say no more on this subject.

Parallel Passive EQ

Parallel EQ exhibits quite different sonic properties to the familiar serial parametric EQ. We are generally used to hearing the effect of one EQ band superimposed on another, as opposed to the band interaction inherent to a parallel EQ. Because the bands are placed in a parallel configuration, phase cancellations and re-enforcements happen which is not always obvious when first encountered.

Passive EQ is something that is found in old equaliser units and is generally known for its transparent and natural sound, but has some problems associated with it. However, in the digital domain these shortcomings do not have such an influence.

A passive EQ does not have any gain elements, but can still have controls to seemingly boost frequencies as well as cut. What actually happens is that the entire signal is cut by an amount, but the frequencies which are apparently ‘boosted’ are simply not cut as much. Therefore the unit must attenuate either the input, the output, or both to allow enough headroom.

Unfortunately in the analogue domain, a 20dB reduction in signal level produces a 20dB increase in the noise floor. Luckily, in the digital domain with a 64-bit floating point DSP, these issues do not remain.

In X-EQ – when the ‘parallel’ button is engaged – you are presented with a parallel passive EQ model which the original designers of these devices could only have dreamed of. The noise floor can be disregarded due to the huge resolution available in Duende plug-ins.

You may find yourself entering this mode more and more as you become familiar with the sonic signature. Larger gain changes are possible without colouration, and boost starts to become something that is useable to a significant degree in a digital EQ!

Parallel EQ does however exhibit asymmetry in its boost and cut characteristics. But this is not such a bad thing as most engineers would agree that boost is best done with low (wide) Q values and cut with a higher (narrower) Q.

6. Vocalstrip

INTRODUCTION

Vocalstrip is a one stop solution for superior vocal processing, providing tailor-made tools for fixing and polishing your vocal sound. The illustration below introduces the main Vocalstrip plug-in features which are described in full over the following pages. Click on the blue boxes to be taken to fuller descriptions of their contents. If you are unclear on how to use the tools provided by Vocalstrip it may be worth starting with the SSL on-line tutorials (see Page 5 for details).



Above the main plug-in window is a set of controls which are specific to your host application. Please refer to your host application's user guide for guidelines on using these controls

De-esser module, for removing excess sibilance

De-plorer module, for removing low frequency wind noise

Graphic Display, shows EQ and dynamics information

Internal bypass for smooth in/out comparisons

Input level metering and control

EQ module

Switch between main and alternative plug-in settings

Processing order controls

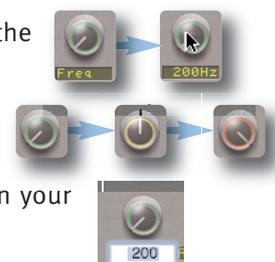
Expansion and Compression module

Output level metering and control

INTERFACE OVERVIEW

The basic interface techniques for the Vocalstrip are largely identical to those for the EQ & Dynamics Channel Strip (see Page 20). In addition to these basic techniques, the following are also available:

To view the value associated with a knob, hover over it with the mouse – the text box underneath the knob will switch from displaying its parameter to displaying its value.



You may notice that the colour of the knob surrounds change to indicate their level – green for low, yellow for mid, red for high.

To enter a precise value for any parameter, double-click on its value display, enter a value on your computer keyboard, and press the Return key.

Plug-in Bypass

The **In/Out** switch located above the **Input** section provides an internal plug-in bypass. This allows for smoother In/Out comparisons by avoiding the latency issues associated with the host application's Bypass function.

The button must be 'lit' for the compressor to be in circuit (as shown left).

Presets

The **Load** and **Save** buttons at the base of the screen provide quick access to the host application's and Save Settings functions. For specific instructions on using presets within the host, consult the host application documentation. Presets use a standard file format which allows them to be moved between a variety of platforms and host applications.



A file of ready-made Vocalstrip presets are included in the plug-in installation. To load SSL presets, click on the Load button and navigate as follows:

Macintosh: /Applications/Solid State Logic/Presets/Vocalstrip
Windows 32-bit: c:\program files\solid state logic\presets\Vocalstrip
Windows 64-bit: c:\program files (x86)\solid state logic\presets\Vocalstrip

A-B Comparisons



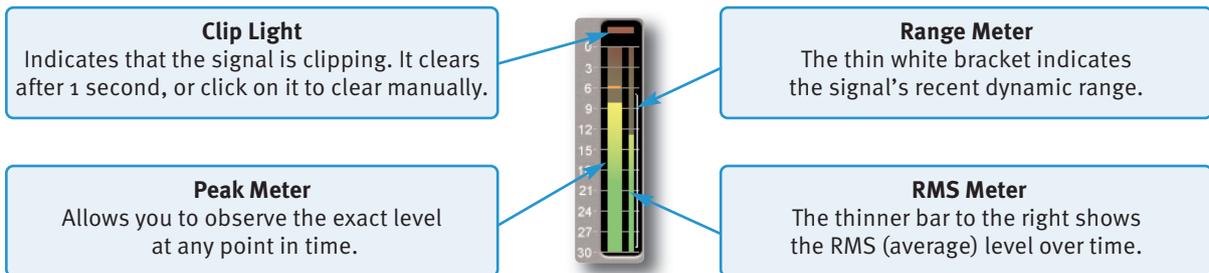
The **A B** button at the base of the screen allows you to load two completely independent set ups and compare them quickly. When the plug-in is opened, setting A is selected by default, as shown right. Pressing the **A B** button will switch between setting A and setting B.

Automation

Automation support for Vocalstrip is the same as for the EQ & Dynamics Channel Strip (see Page 20).

INPUT AND OUTPUT SECTIONS

The input and output sections at either side of the plug-in window provide input and output gain control, along with displays of the following information:



Clip Light

Indicates that the signal is clipping. It clears after 1 second, or click on it to clear manually.

Range Meter

The thin white bracket indicates the signal's recent dynamic range.

Peak Meter

Allows you to observe the exact level at any point in time.

RMS Meter

The thinner bar to the right shows the RMS (average) level over time.

VOCALSTRIP MODULES

De-esser

Sibilance is a resonance often found in vocal recordings which can result in 'S' consonants becoming too pronounced. The Vocalstrip De-esser can detect and remove sibilance.

Switch the De-esser on by clicking on the power button in its top left-hand corner. The blue indicator between the controls 'glows' to show that it is working.

- Move the **Amount** knob to control how much sibilance is removed.
- Move the **Thresh** (threshold) control to set the detection level, so that the De-esser can distinguish between 'S' and 'non-S' sounds.

To listen only to the sibilance that is being removed, press the **Aud** button. This can be particularly helpful in ensuring that the threshold is correctly set.



De-ploser

Plosives are bursts of low frequency energy in a signal which can cause compressors to sound unpleasant. They are most often caused by the breath of a singer standing too close to the microphone. The Vocalstrip De-ploser detects these artefacts and lets you manage their level. The De-ploser's controls are very similar to those of the De-esser:



Switch the De-ploser on by clicking on the power button in its top left-hand corner. The blue indicator between the controls 'glows' to show that it is working.

- Move the **Amount** knob to control how much plosive is removed.
- Move the **Thresh** (threshold) control to set the detection level, so that the De-ploser can distinguish between plosives and normal low frequency sounds.

To listen only to what is being removed, press the **Aud** button. This can be particularly helpful in ensuring that the threshold is correctly set.

Equaliser

Switch the EQ on by clicking on the power button in its top left-hand corner. Three popular EQ types are provided to allow you to cut low frequencies, find and attenuate resonances and shape the top end of the signal. Switch each band on by clicking on the oblong switch above the band's curve graphic.



The low cut filter operates in the range of 30Hz to 300Hz, with a slight boost around its cut-off frequency.



The notch filter operates in the range 200Hz to 10kHz and offers 12dB of boost and 36dB of attenuation with a high Q value.

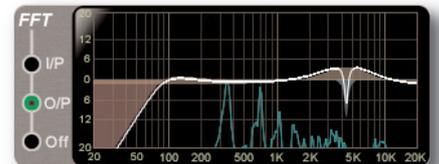


The high band EQ offers 12dB of boost/attenuation over the range of 1kHz to 20kHz with a low Q value.

EQ Display

When any controls within the EQ section are moved, the display in the top right-hand quarter of the window becomes an EQ graph:

- The white line across the graph shows the frequency response of the current EQ settings.
- The brown shaded area represent the impact of each of the three EQ bands.
- A display of the signal's frequency response is shown in green. The signal displayed can be switched between the plug-in input and output and can be switched off, all using the **FFT** buttons to the left of the graph.



! EQ Tip: Finding and Attenuating a Resonance

A vocal recording may contain a resonant frequency caused by a number of factors, including bad mic placement, room modes, or even unpleasant resonances caused by the singer's skull!

These can be detected and reduced in the following way:

- Switch off the low cut filter and high band EQ
- Set the notch filter gain to between about +10dB
- Sweep the frequency slowly up range, listening out for ringing
- Once you have found the offending frequency, bring the gain down to a significant negative value

Another technique is to use the green frequency response line in the EQ graph and locate resonances visually.

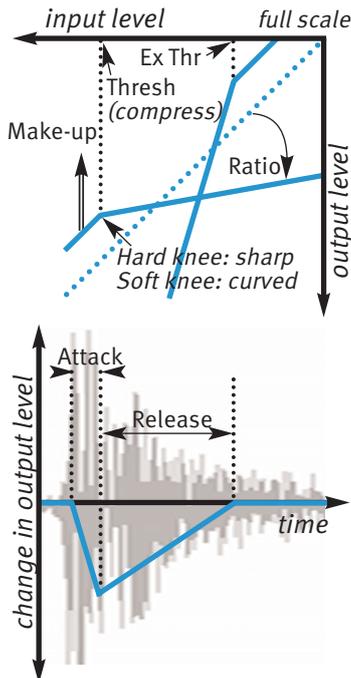
Compander

The Vocalstrip Compander is a hybrid Compressor and Expander. Switch it on by clicking on the power button in its top left-hand corner.

Expander

First the signal enters a fixed ratio downwards expander, designed to reduce room ambience, spill or breath noise that is often brought up by the subsequent compression.

To set the level at which the level reduction is introduced, turn the **Ex Thr** (Expander Threshold), located on the Compander's right-hand side. Turn the threshold to minimum to turn the expander off. The amount of level reduction introduced is shown in the green bar to the left of the **Ex Thr** knob.



Compressor

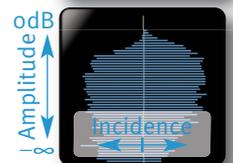
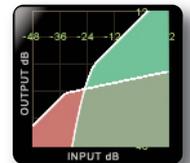
The compressor offers both hard and soft knee modes alongside fully variable **Ratio**, **Thresh** (threshold), **Attack**, **Releas** (release) and **Makeup** controls, along with a choice of **Hard** or **Soft Knee**.

The output stage which is sourced after the make-up gain features an optional **Drive** feature, which introduces harmonic characteristics to the signal. Its intensity is increase with the **Makeup** gain. If you use a lot of make-up gain to drive the circuit, the level can be reduced again using the Output level control. The amount of level reduction introduced is shown in the red bar to the right of the compressor controls.

Parameter	Min	Max
Ex Thr	-96dB	0dB
Ex Ratio	Fixed: 1.5:1	
Comp Thr	-48dB	0dB
Comp Ratio	1:1	50:1
Attack	0.1ms	100ms
Release	0.5ms	2s
Makeup	0dB	48dB

Compander Display

When any controls within the Compander section are moved, the display in the top right-hand quarter of the window shows two different graphs:



The left-hand graph is a gain-law display, showing the relationship between input and output levels, as illustrated in the graph above-left.

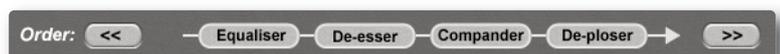
The right-hand graph is an IO difference display, showing how often each level occurs within the input and output signals. The input is shown on the left and the output on the right.

The vertical scale is amplitude, with 0dB at the top and $-\infty$ at the bottom. The length of each line protruding from the centre represents the number of incidents of that amplitude over a period of seconds.

The FFT controls are for the EQ display, not the Compander display.

Processing Order

The processing order is controlled at the base of the plug-in window.



To move a module within the order, highlight it so that it is surrounded in yellow, and press either the left **<<** arrow or the right **>>** arrow.

7. Drumstrip

INTRODUCTION

Drumstrip is a one stop solution for superior drum processing, providing tailor-made tools for fixing and polishing your drum sounds. The illustration below introduces its features which are described in full over the following pages. Click on the blue boxes to be taken to fuller descriptions of their contents. A good place to start with Drumstrip may be the SSL on-line tutorials (See Page 5 for details).



INTERFACE OVERVIEW

The basic interface techniques for the Drumstrip are largely identical to those for the EQ & Dynamics Channel Strip (see Page 20). In addition to these basic techniques, the following are also available:

To view the value associated with a knob, hover over it with the mouse – the text box underneath the knob will switch from displaying its parameter to displaying its value.



You may notice that the colour of the knob surrounds change to indicate their level – green for low, yellow for mid, red for high.



To enter a precise value for any parameter, double-click on its value display, enter a value on your computer keyboard, and press the Return key.



Plug-in Bypass

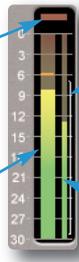
The **In/Out** switch located above the **Input** section provides an internal plug-in bypass. This allows for smoother In/Out comparisons by avoiding the latency issues associated with the host application's Bypass function. The button must be 'lit' for the compressor to be in circuit (as shown left).

Automation

Automation support for Drumstrip is the same as for the EQ & Dynamics Channel Strip (see Page 20).

INPUT AND OUTPUT SECTIONS

The input and output sections at either side of the plug-in window provide input and output gain control, along with displays of the following information:



Clip Light
Indicates that the signal is clipping. It clears after 1 second, or click on it to clear manually.

Peak Meter
Allows you to observe the exact level at any point in time.

Range Meter
The thin white bracket indicates the signal's recent dynamic range.

RMS Meter
The thinner bar to the right shows the RMS (average) level over time.

In addition, the Output area provides a **Mix** control, for adjusting the mix between the processed ('Wet') signal and unprocessed ('Dry') signal.

DRUMSTRIP MODULES

Gate

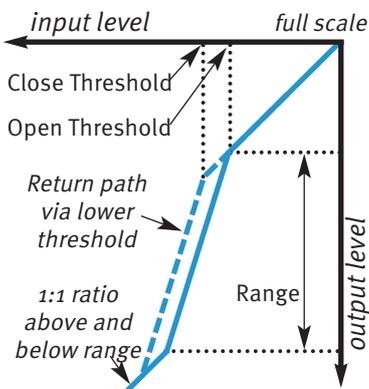
The gate is suitable for many applications, including:

- Shortening drum hits to obtain a 'tighter' sound
- Controlling ambience on live drums tracks
- Manipulating attack and decay characteristics

Switch the Gate on by clicking on the power button in its top left-hand corner.

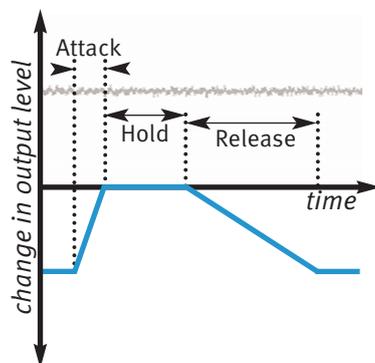


The Gate provides controls for the **Attack**, **Releas** (release) and **Hold** times, as well as **Open** and **Close** thresholds and **Range** levels, as illustrated in the diagrams below left. If you are unclear about these parameters, please refer to SSL's on-line tutorials, introduced on Page 1.



Open and Close Thresholds

The levels for 'opening' the gate to audio and 'closing' it again are set separately. Generally, the 'open' level is set higher than the 'close' level. This is known as hysteresis and is very useful as it allows instruments to decay more naturally. If the close threshold is higher than the open threshold, the close threshold is ignored.



Range

The range is the depth of attenuation applied to the signal when the gate is closed, as indicated by the white line in the right hand column. For a true gating action the range should be set to -8odB , which is effectively silence. By reducing the range, the gate takes on some of the characteristics of a downward expander where the signal is lowered in level set by the range amount, rather than being completely silenced. This can be useful in cleaning up a drum track containing reverb, where silencing the reverb would sound too artificial but attenuating it by a few dB would push it down to an acceptable level.

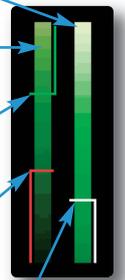
Attenuation level

Input level

Green line: 'Open' threshold

Red line: 'Close' threshold

White line: Range value

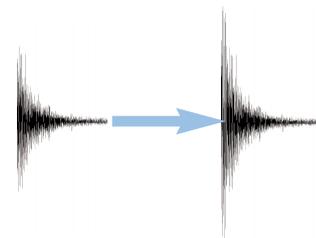


Parameter	Min	Max
Open Thr	0dB	-3odB
Close Thr	0dB	-3odB
Range	0dB	-8odB
Attack	0ms	0.1ms
Hold	0s	4s
Release	0s	1s

Transient Shaper



The Transient Shaper allows you to add attack to the start of a drum hit by increasing the amplitude of the attack portion of the signal whilst leaving the decay unchanged. The right hand waveform is a processed version of the one on the left. It has been passed through the transient shaper where the amplitude of the attack portion has been increased.

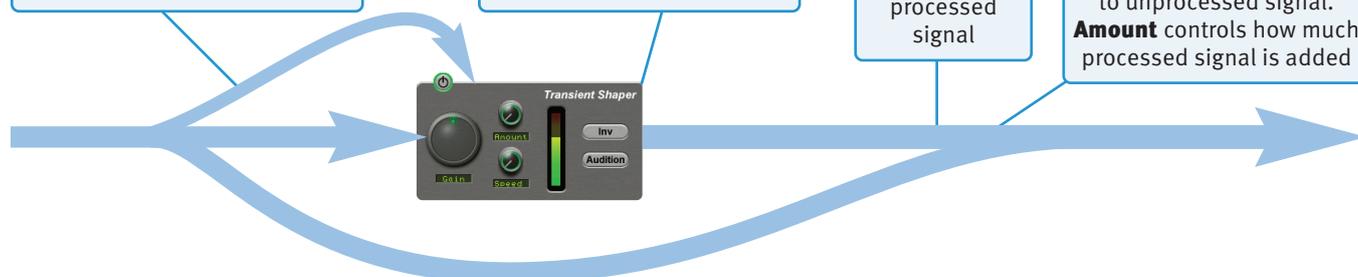


The Shaper uses a control signal to detect transients. **Gain** controls the level of this signal.

You can adjust the **Speed** of the signal's return to normal. You can also **Invert** the signal

Press **Audition** to listen to processed signal

Processed signal is added to unprocessed signal. **Amount** controls how much processed signal is added

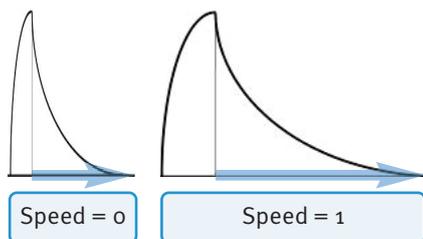


Switch the Shaper on by clicking on the 'power' button in its top left-hand corner. The central meter gives visual feedback on how much attack is being added using the **Gain** and **Amount** controls. If this rises to the top red section then reduce the effect.

Gain controls the detection level of the controller signal, and should be set so that only the transients you want to shape are detected. If this is set too low then the Shaper will do nothing; if it is set too high then the Shaper will detect too many transients, resulting in an exaggerated process, and the attack appearing too long. The default setting of 0dB should be a good starting point.

***Gain** doesn't directly affect the output signal's gain.*

Amount controls the amount of the processed signal added to the unprocessed signal. This process can increase the peak level of a signal significantly so watch the output meter carefully.



Speed controls the length of time the added attack takes to fall back down to the normal signal level once it has reached the top of the attack phase. Turn the knob clockwise for a slower speed, and longer transient.

The **Inv** switch inverts the processed signal so that it is subtracted from the unprocessed signal. This has the effect of softening the attack, resulting in more body in the drum sound.

The **Audition** switch allows you to listen to the processed signal, to assist in the set up process.

*When the **Inv** and **Audition** buttons are both pressed, the signal will not be inverted.*

HF and LF Enhancers

The HF and LF enhancers respectively enrich the high and low frequencies of the input signal. Whereas a standard EQ simply raises the level of certain frequencies, the Enhancer adds a combination of 2nd and 3rd harmonics to those frequencies, producing a more pleasing effect.



Switch each Enhancer on by clicking on the power button in its top left-hand corner. No effect is heard until an Enhancer's **Drive** and **Amount** are turned up.

HF Cutoff sets the frequency above which the HF Enhancer generates harmonics. It ranges from 2kHz up to 20kHz – To add air or sparkle to a signal, push this frequency towards the higher end of the range. To give more presence to a signal, use the lower end of the range. Note that the effect is barely audible in the 15kHz to 20kHz range.



LF T.Over (turnover) sets the frequency below which the LF Enhancer generates harmonics. It ranges from 20Hz up to 250Hz. The LF Enhancer is great for adding depth and weight to kick drums, snare or toms.

Each Enhancer has its own **Drive** and **Amount** controls:

- **Drive** (or overdrive) controls the density and amount of harmonic content, from 0 to 100%.
- **Amount** is the amount of Enhanced signal that is mixed into the unprocessed signal, from 0 to 100%.

Listen Mic Compressor

The Listen Mic Compressor was first found in the classic SL 4000 E Series console. The Drumstrip edition includes a narrowband EQ by-pass and a wet/dry Mix control.



Comp controls the amount of compression, from 0 to 100%.

MakeUp controls the level compensation for the gain reduction and **Mix** controls the balance of the compressed ('Wet') to uncompressed ('Dry') signal. Note that **MakeUp** only acts on the 'wet' part of the signal.

To simulate the original narrow-band listen mic characteristic, activate the **EQ In** button – to use the compressor on the full frequency range, leave **EQ In** deactivated.

The Listen Mic Compressor features very quick fixed time constants. This means it is easily capable of producing distortion on low frequency material.

Processing Order

The five processing blocks in Drumstrip can be configured in any order, as defined by the Process Order blocks at the base of the plug-in window.



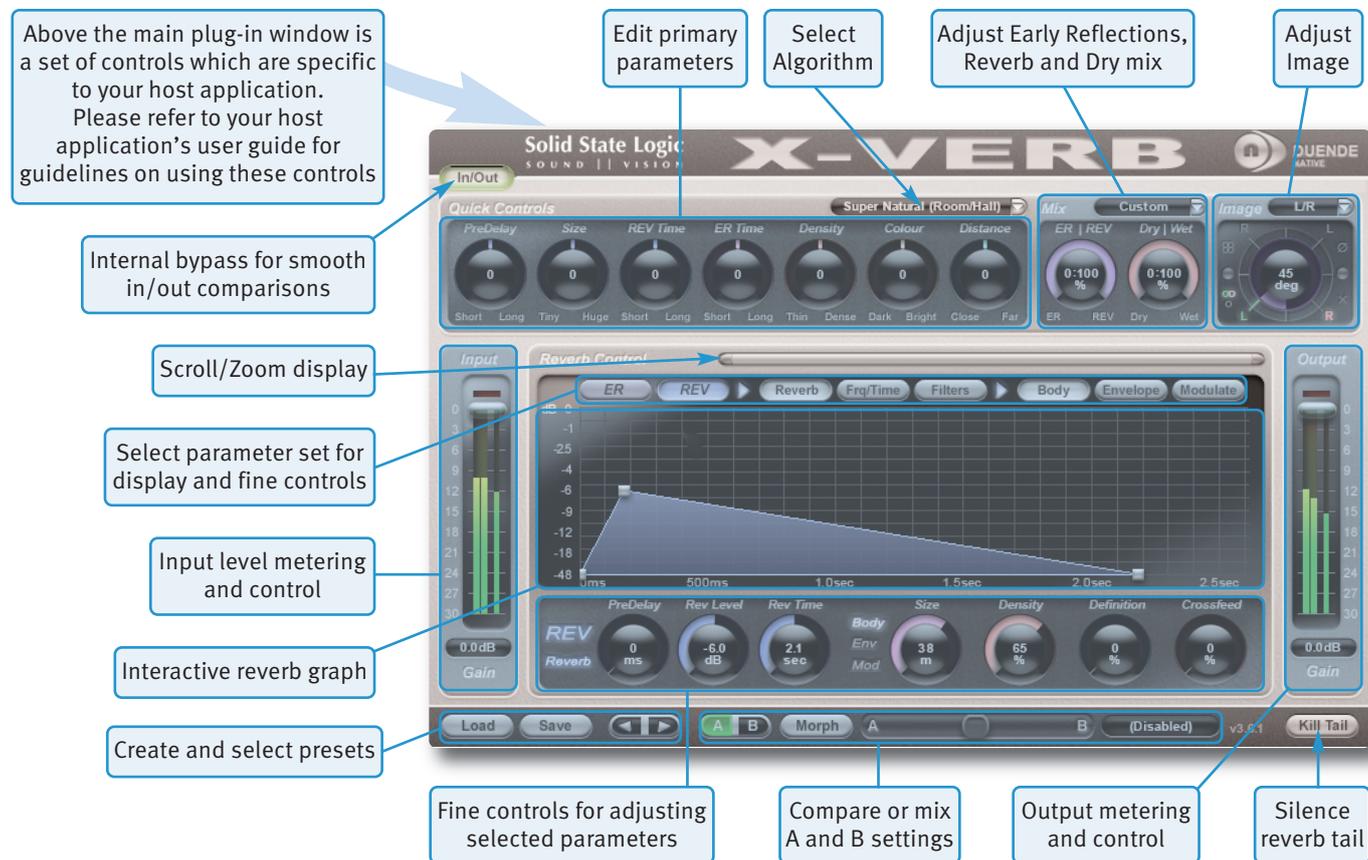
To move a module within the order, highlight it so that it is surrounded in yellow, and press either the left << arrow or the right >> arrow.

Typically, the gate is first in the chain so that it is able to act on the full dynamic range of the signal before the signal is processed by the Listen Mic Compressor.

8. X-Verb

INTRODUCTION

X-Verb is a versatile and comprehensive studio grade reverberation plug-in which delivers the density, warmth, depth and stunning detail usually only found in top-end hardware reverb processors. The primary parameters provide conventional reverb manipulation, while nearly 100 detail parameters allow an exceptional level of fine-tuning. Click on the blue boxes to be taken to fuller descriptions of their contents.



INTERFACE OVERVIEW

In/Out To press a switch in a plug-in, simply click on it. The active state is sometimes indicated by a green surround (shown left), and sometimes indicated by a shadow at the top rather than the bottom, (shown right).

Knobs To turn a knob, click on it, and drag it up or down. If your mouse has a scroll wheel, you can also turn knobs by hovering over them and turning the scroll wheel. The value for each parameter is displayed in the centre of the knob.

To move a knob slowly for fine adjustments, hold the following on your keyboard whilst turning/scrolling:

Fine Adjust for Host Type	Mac	Windows
AU	Shift	n/a
VST or VST3	Shift	Shift
AAX	Shift	Shift
RTAS	⌘ ('Command')	Ctrl

To reset a knob to its default value, click on the knob whilst holding the following on your keyboard:

Reset for Host Type	Mac	Windows
AU	⌘ ('Command')	n/a
VST or VST3	⌘ ('Command')	Ctrl
AAX	⌘ ('Command')	Ctrl
RTAS	Alt	Alt

Plug-in Bypass



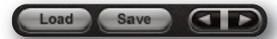
The **In/Out** switch located above the **Input** section provides an internal plug-in bypass. This allows for smoother In/Out comparisons by avoiding the latency issues associated with the host application's Bypass function. The button must be 'lit' for the compressor to be in circuit (as shown left).

Automation

Automation support for Drumstrip is the same as for the EQ & Dynamics Channel Strip (see Page 20).

Presets

The **Load** and **Save** buttons at the base of the screen provide quick access to the host application's Import and Save Settings functions. The arrow buttons to the right of the **Load** and **Save** buttons can be used to cycle through all of the stored presets within the same folder.



For specific instructions on using presets within the host, consult the host application documentation. Presets use a standard file format which allows them to be moved between a variety of platforms and host applications.

A file of ready-made X-verb presets are included in the plug-in installation. These provide starting points for a variety of instruments and effects. To load SSL presets, click on the Load button and navigate as follows:

Macintosh:	/Applications/Solid State Logic/Presets/x-verb
Windows 32-bit:	c:\program files\solid state logic\presets\Vocalstrip
Windows 64-bit:	c:\program files (x86)\solid state logic\presets\Vocalstrip

A-B Comparisons and Morphing

The **A B** button at the base of the screen allows you to load two completely independent set ups and compare them quickly. When the plug-in is opened, setting A is selected by default, as shown right. Pressing the **A B** button will switch between setting A and setting B.



It is also possible to mix between the settings: Press the **Morph** button to the right of the **A B** button and use the slider to its right to morph between A and B. The display to the right of the slider displays '**(Disabled)**' when Morph is off, and displays the A to B ratio when Morph is on.

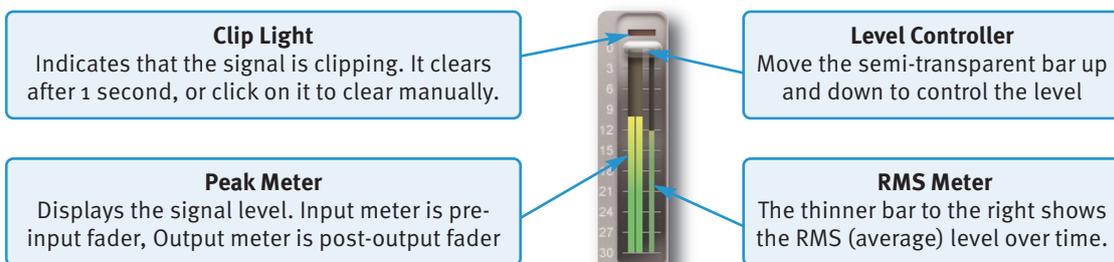
Every parameter is morphed with the exception of the Algorithm and the Size, which are taken from whichever setup is indicated as active in the **A B** button.

The Morph function is automatically deactivated when a parameter is moved.

Note also that automating Morphs will result in unpleasant artefacts if any time constants are different between setups A and B

Input and Output Levels

The input and output sections at either side of the plug-in window provide input and output gain control, along with level metering and clip lights:



The Graph Display

The graph in the centre of the plug-in window displays the impact of the settings currently selected in the controls above and below it. The row of buttons immediately above the graph (with **ER** [Early Reflections] and **REV** [Reverb] at their left-hand end) determine what information is displayed, and also which controls appear beneath the graph.

The bar to the right of the **Reverb Control** label can be used to scroll and zoom the graph's time axis: click and drag the handles on either end of the scroll bar to zoom in and out, or click and drag the main scrollbar to scroll left or right. The graph also auto-zooms to ensure that all of the important information remains visible.

The graph is interactive: move the grey nodes to adjust their associated parameters.

These buttons define contents of display

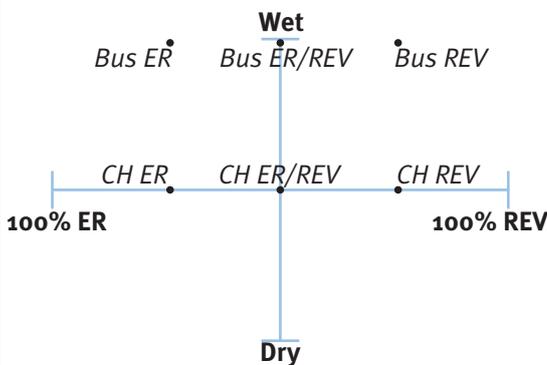


The parameters displayed in the graph are described on Page 54 onwards.

Mix Levels

The **Mix** area towards the top right-hand corner of the display controls the mix between Early Reflections (**ER**) and reverb (**REV**), as well as the mix between the un-processed (**Dry**) and processed (**Wet**) signals.

A number of presets are available – click on the box to the right of the Mix label and select one from the drop-down menu.



The **Bus** and **CH** prefixes indicate the **Dry – Wet** mix:

Bus presets have a 100% Wet mix, for when X-Verb is being used in an auxiliary bus where it has an independent level control.

CH presets default to a 50-50 mix. Use **CH** when X-Verb is being used within a track, and you need to mix a smaller amount of the reverb signal with the original.

In the **ER/REV** presets, both **ER** and **REV** signals are at full. In the **ER** presets, the level of the **REV** signal is halved, and in the **REV** presets, the level of the **ER** signal is halved.



Image Control

X-Verb comes with an innovative Stereo Image interface which performs Stereo width, Phase, Left/Right channel swapping, and Surround Processor compatible L-R – R-L operations. The control is located in the top right-hand corner of the window.

The image can be rotated by 360° in either direction. A number of image presets are included – click on the box to the right of the Image label and select one from the drop-down list which appears (shown left).

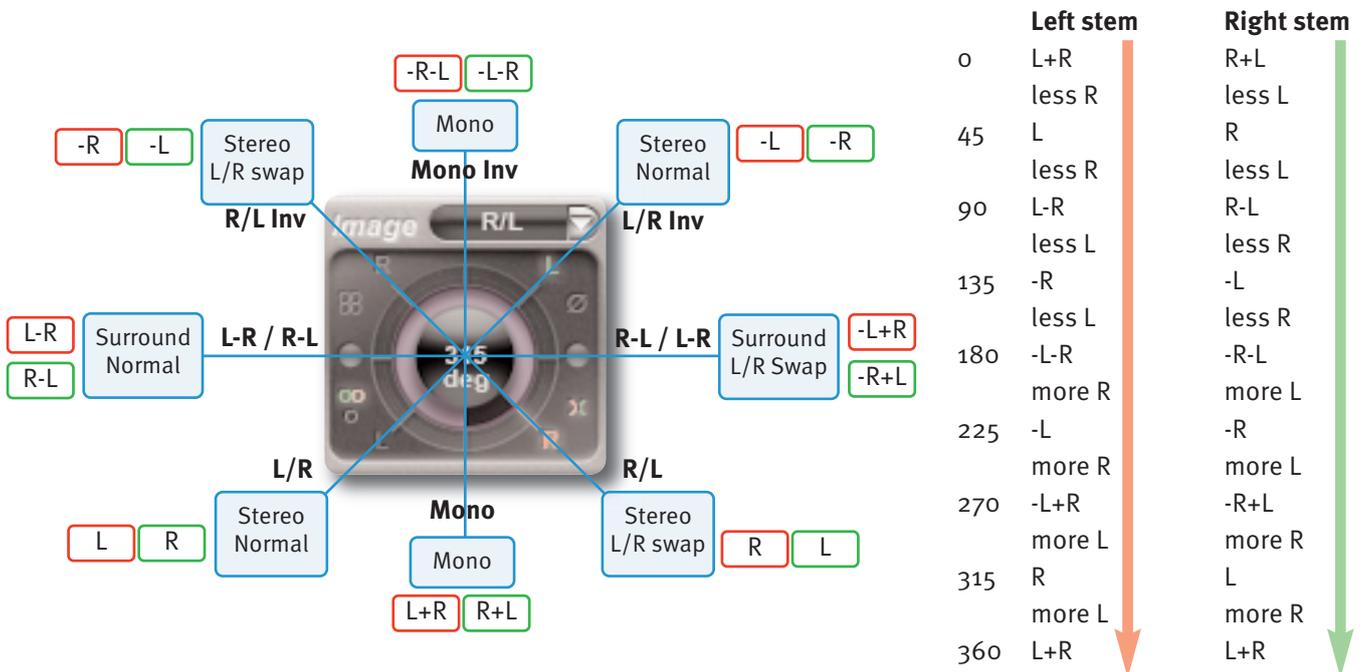


Note that the angle of the Image controller is not the position of the signal within the spectrum. In essence, the display works as follows:

As the Image controller moves left, the level of the Left signal is reduced in the Right component, and vice versa. As the image moves up, the level of the Left signal in the Left component is reduced, as is the level of the Right signal in the Right component.



The diagram below illustrates the ‘angle’ associated with each preset, and the subsequent stereo/surround imaging. The blue boxes indicate the stereo image, the red and green boxes indicate the actual signals on the left and right channels respectively, and the image preset name is shown in bold. The diagram to the right indicates what is happening as you turn the image controller clockwise.



The symbols surrounding the image control indicate the following:

- ⊗ Surround image
- ⊙ Stereo image
- Mono image

Note that often two of these symbols will be ‘lit’, as the image moves between them.

- ✕ L/R swap
- L/R L and R symbols represent the phase of the left and right signals in the left component: The lower ones indicate when that component of the signal is in phase, and the upper ones indicate when it out of phase. In other words, the signals are in phase with each other when they are in the same row, and out of phase when they are in the same column.
- ∅ Signals are more out-of-phase than in-phase

X-VERB CONFIGURATION

Kill Tail Button

Whenever you are auditioning changes in your configuration, and don't want to wait for the reverb tail to die away completely, click on the Kill Tail button in the bottom right-hand corner of the window.



1. Select an Algorithm or Preset.

It is often a good idea to begin setting up your reverb by loading a preset as a starting point as described on the previous page. When starting from scratch, the algorithm on which you are going to build the reverb needs to be selected first: click on the algorithm box, located top-centre in the plug-in window, and select an algorithm from the drop-down list which appears.



When algorithms are changed, other current settings remain as they were, making it easy to compare similar settings with different algorithms, though some parameters may change their behaviour or range. Bear in mind that changing the algorithm may lead to a short audio interruption. Here is an overview of each algorithm:

ALGORITHM	SUGGESTED USES	DESCRIPTION
Super Natural	Rooms and Halls	Produces real sounding rooms and halls of any size, with a rich diffusion and natural sounding reverb envelope.
Super Natural HiD		Produces higher density diffusion and refraction, for even more natural and smooth sounding spaces.
Ultra Plate	Plates, Larger Rooms	A rich, silky, warm reverb, providing a more traditional sound with mainly static and predictable refraction and diffusion. Ideal for making instruments and vocals stand out in a mix, and for sound designers and film composers who need huge reverbs.
Ultra Plate HiD		Adds more density, shine and depth, somewhere between digital reverberation and a real room.

2. ADJUST THE PRIMARY PARAMETERS

The seven controls across the top left of the plug-in window provide access to the primary reverb parameters. Each of these parameters represent an interaction between a number of more detailed parameters which can also be adjusted (see next page).



The following table explains what each primary parameter does, and what detail parameters are linked to it:

PARAMETER	DESCRIPTION	LINKED PARAMETERS
PreDelay (Short-Long)	Controls delay before reverb is heard, emulating the time it takes for the sound to bounce back from the closest walls.	ER Master Delay REV PreDel
Size (Tiny-Huge)	Emulates the size of the room, altering the quantity and speed of the reflections.	Size ER MasterTime ER Tap Rebound
REV Time (Short-Long)	The time it takes for the reverb to die away.	REV Time REV Shape REV PreDel
ER Time (Short-Long)	The speed of the early reflections. These are the reflections which bounce back early enough to be heard individually. Reflections quickly become too dense to be separated, at which point they become 'reverb'.	ER Master Time ER Master Delay
Density (Thin-Dense)	The level of focus and definition in the reflections. The more dense, the harder it is to discern individual reflections.	Density Definition Attack
Colour (Dark-Bright)	Alters the harmonic colour of the reflections, to emulate the materials the sound might be bouncing off.	9 linked parameters, including: ER Tap Filters, REV Shelve Filters and Freq Times
Distance (Close-Far)	Emulates the distance between the sound and the listener.	ER Time, Delay and Level Masters; REV PreDel, Master Level, Definition, Density, Shape, Attack and Mode

3. Adjust the Detail Parameters

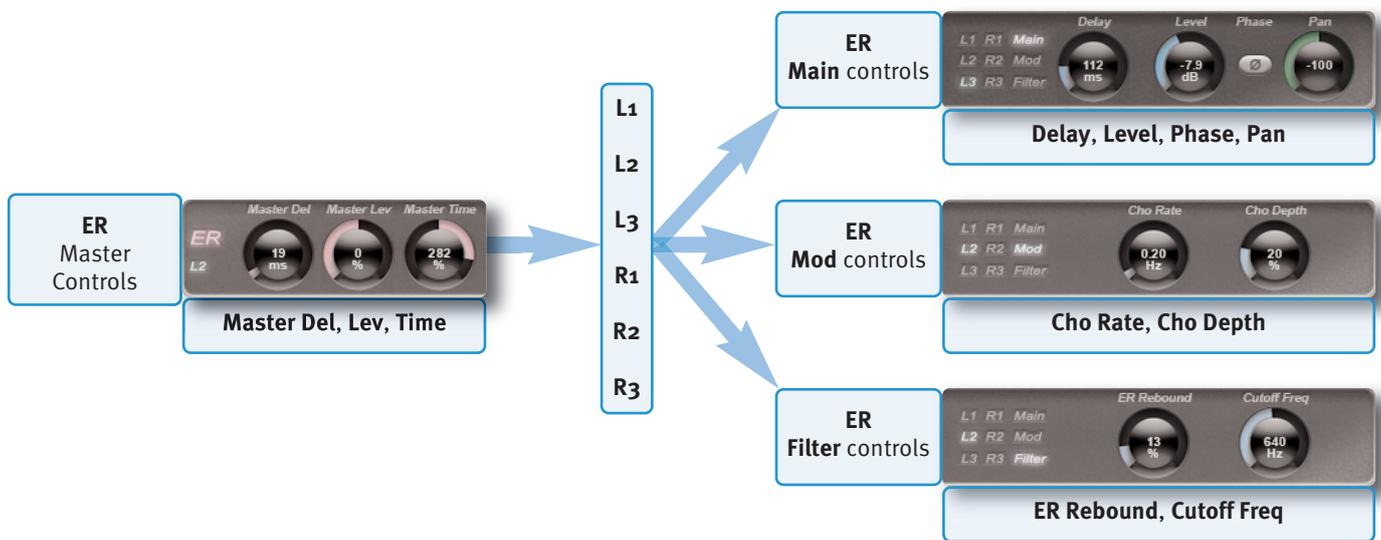
Detail parameters are adjusted in the **Reverb Control** area of the window, located between the **Input** and **Output** areas. The **ER** and **REV** buttons in the top left-hand corner of the display define whether the ER (Early Reflections) or REV (Reverb) parameters, graphs and sub-menus are displayed.

ER

If **ER** is selected, the ER Master Del (delay), Master Lev (level) and Master Time knobs appear below the graph and to the left. In addition, six buttons appear to the right of the **ER** button, labelled **L1**, **L2**, **L3**, **R1**, **R2** and **R3**. These define which of the six Left and Right early reflection ‘taps’ are being edited using the three further buttons which further to the right: **Main**, **Mod** and **Filter**. By selecting one of these three buttons, the controls for their parameters appear below the graph and to the right.

The area between the **ER** controls and the **Main**, **Mod** or **Filter** controls indicate which ER ‘tap’ is being edited, and which set of parameters are displayed to the right – the appropriate symbols light up.

The following diagram illustrates which controls are available when each button is selected. Each parameter is defined in the table on Page 60. The table on Page 56 indicates which of these parameters are affected by the Primary parameters.



When ER parameters are active, the graph displays the ER Box in pink. This represents the loudest and longest ER taps. Each tap (L1, L2, L3 and R1, R2, R3) has a handle which can be clicked/dragged horizontally to adjust the Delay time, or vertically to adjust the tap’s Level. The stem beneath each node is coloured red for left, or green for right.

The handle half way along the top of the ER Block adjusts the Master Level (increasing/decreasing the individual taps levels relatively), the handle half way down the right side of the ER Block adjusts the Master Time (increasing/decreasing the individual taps delays relatively).



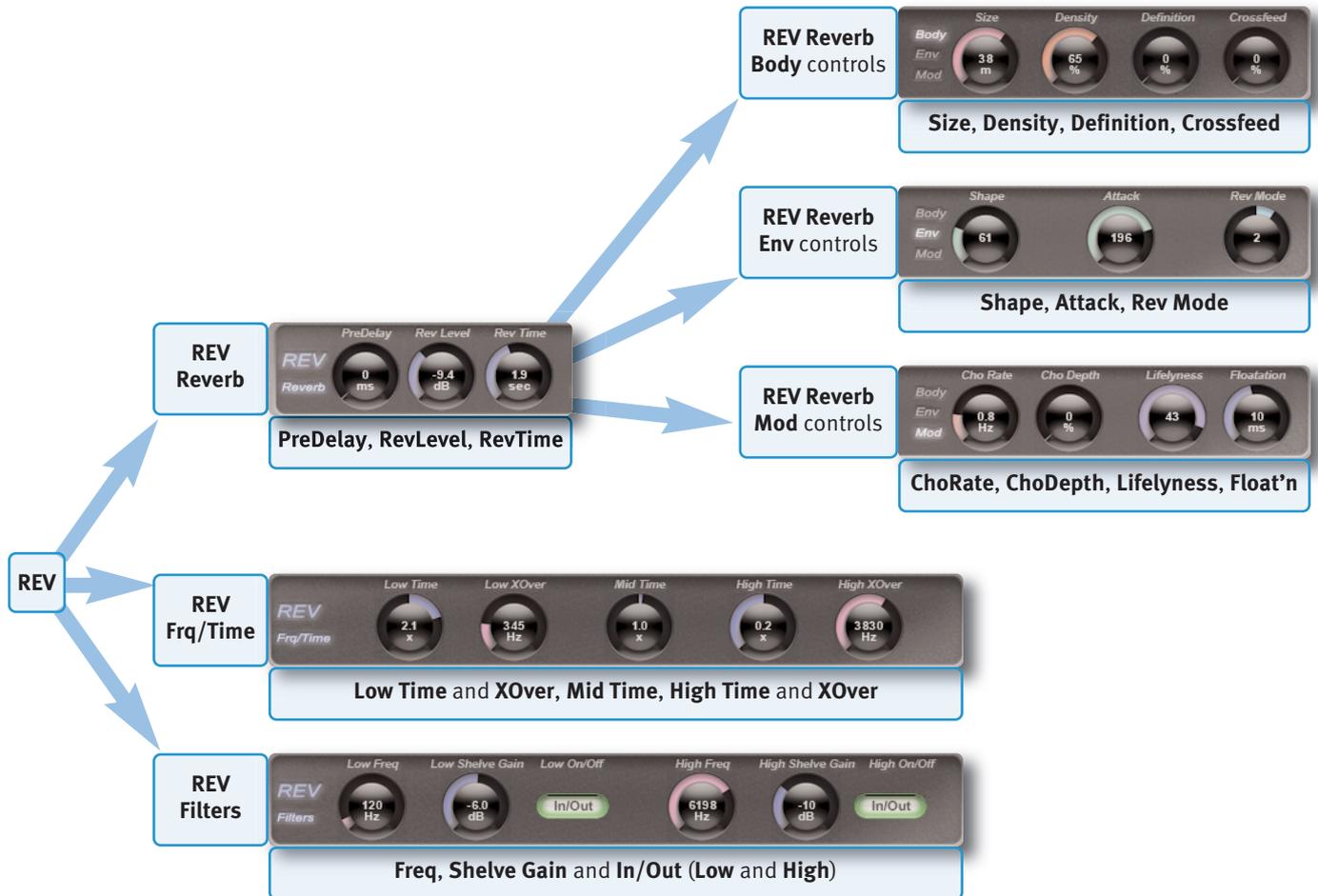
While the ER Box is displayed, the REV triangle is always visible in the background. Clicking on the REV triangle will switch the **Reverb Control** area to **REV** and the sub page to **Reverb**.

REV

If **REV** is selected, three buttons appear to the right of the **REV** button, labelled **Reverb**, **Frq/Time** and **Filters**. When each of these is selected, their parameters appear below the graph.

When the **Reverb** button is selected, a further sub-menu appears to its right, labelled **Body**, **Envelope** and **Modulate**. By selecting one of these three buttons, the controls for their parameters appear below the graph and to the right (the three left-hand parameters – **PreDelay**, **Rev Level** and **Rev Time** – are displayed whenever **Reverb** is active). The area between the left-hand master **Reverb** controls and the **Body**, **Envelope** or **Modulate** controls indicate which set of parameters are displayed to the right – the appropriate symbols light up.

The following diagram illustrates which controls are available when each button is selected. Each parameter is defined in the table on Page 60. The table on Page 56 indicates which of these parameters are affected by the Primary parameters.



When **REV** is selected, the graph display is dependent on which sub-menu is active.

When **Reverb** is selected, the REV Triangle is displayed. The left handle on the bottom adjusts the reverb predelay (horizontal), the handle on the top adjusts the reverb level (vertical) and the right bottom handle changes the reverb time.

The ER block is visible in the background and clicking on it will switch the **Reverb Control** area to **ER**.



When **Freq/Time** or **Filters** is selected, the graph changes to display the Frequency/Time blocks and High/Low Shelving filter curves. Both graphs are overlaid with the active graph being in focus (on top).

Inside the **Freq/Time** graph, the left handle allows the low reverb time factor (vertical) and low crossover frequency (horizontal) to be altered. The mid handle adjusts the mid frequency time factor (vertical) and the right handle changes the high reverb time factor (vertical) and high crossover frequency (horizontal).



In the **Filters** sub page a graph is only displayed when there is an active filter. Each filter has a handle to adjust the frequency (horizontal) and cut slope (vertical).



DETAIL PARAMETER RANGES AND DESCRIPTIONS

ER Menu

Menu		Parameter	Range	Description
ER Master Controls		Master Delay	0 - 1500ms	Adds a Pre-Delay to the whole 'ER Block' so all individual Taps move to the right.
		Master Level	0 - 400%	Adjusts the level of all Taps, counter-clockwise to decrease, clockwise to increase all tap levels.
		Master Time	0 - 400%	Adjusts the delay of all Taps, counter-clockwise to decrease, clockwise to increase all delay times.
Sub menus for each Tap	Main	Delay	0 - 2500ms	Tap Delay
		Level	$-\infty$ to 0dB	Tap Level
		Phase	On or Off	Inverts phase of tap
		Pan	-100 (L) to +100 (R)	Places tap in stereo spectrum
	Modulate	Cho Rate	0 - 4Hz	Adds a Chorus to the Tap to thicken and blur the reflection. At 0.00Hz the Chorus is switched off.
		Cho Depth	0 - 100%	Adjusts the modulation depth of the Chorus. At 0% the Chorus is switched off.
	Filters	ER Rebound	0 - 100%	Adds a modulated Feedback Loop to the delay line of a Tap to create complexity inside the Reflection.
		Cutoff Freq	25Hz - 24kHz	Adjusts the Cut off Frequency of a smooth Low Pass Filter to dampen the reflection and the rebounds.

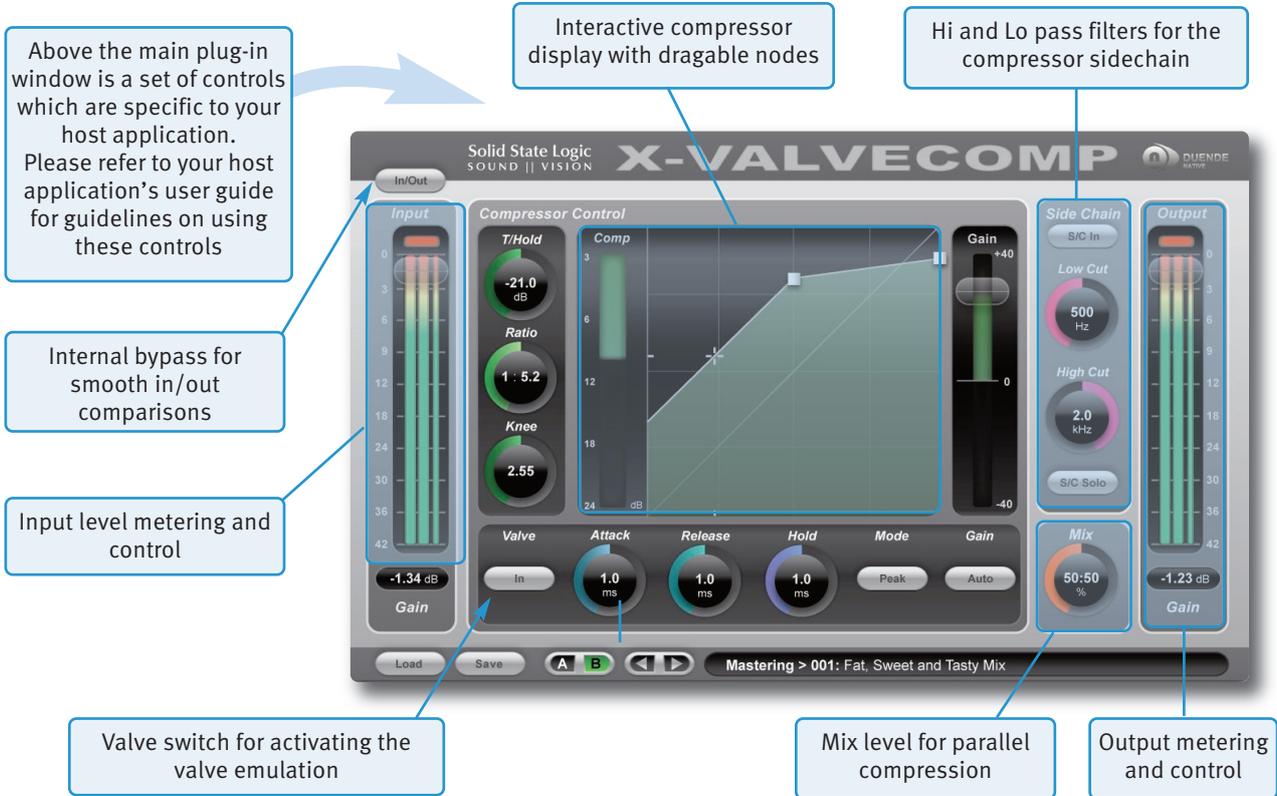
REV Menu

Menu		Parameter	Range	Description
Reverb Master Controls		PreDelay	0 - 1500ms	Adds a Pre-Delay to the reverb tail, moving the Triangle right.
		Rev Level	$-\infty$ to 0dB	Adjusts the level of the reverb tail
		Rev Time	<i>Varies</i>	Determines the time for the reverb level to fall below 60dB (RT 60). Dependant on the Size, Shape, Attack and Algorithm.
Reverb sub menus	Body	Size	0 - 60m	Adjusts room size by altering the reverb delay structure, the diffusion engine and the complexity approximation. IMPORTANT: Changing the Size can result in audible artefacts.
		Density	0 - 100%	Adjusts the density in the diffusion and refraction engine and complexity approximation. Higher values smoothen the reverb tail and individual 'refractions' or ringing echoes disappear.
		Definition	0 - 100%	Adds defined and randomly modulated reflections to the beginning of the reverb tail. Can add clarity but may reduce density. In Ultra Plate, Definition also modulates the Density in relation to Input Level to create a more natural decay.
		Crossfeed	0 - 100%	X-Verb has separate diffusion engines for left and right. Crossfeed introduces the left input to the right channel and vice versa, reducing the stereo image but adding the impression of distance.
	Envelope	Shape	0 - 255	Alters the Shape of the Reverb Triangle. At higher values the reverb build up phase (Attack) becomes longer, while the tail fade out phase becomes shorter.
		Attack	0 - 255	Expands the time of the Attack. At higher values the reverb build up time can take up to 700 ms. Note that when Shape is set to 0, Attack is inactive.
		Rev Mode	-9 to +9	Allows less natural-sounding reverb. At 0, reverb sounds most natural – positive values compress tail. LF energy may lead to artifacts – lower Input level or set RevMode to 0 to eliminate.
	Modulate	Cho Rate	0-4Hz	Adds Chorus to the tail to thicken and smoothen the reverb.
		Cho Depth	0-100%	Adjusts the modulation depth of the Chorus.
		Lifeliness	0-48	Adds natural sounding random time modulation to refractions in Diffusion engine. For Ultra Plate, adds modulation to Chorus
		Floatation	0-22ms	Determines the maximum range of 'Lifeliness' time fluctuation. Not available in the Ultra Plate Algorithms.
	Freq/Time Controls		Time (Low, Mid, High)	0.1-4
XOver (Low, High)			25Hz - 24kHz	Adjusts the crossover frequency between Low and Mid bands, and between Mid and High bands.
Filters Controls		Freq	25Hz - 24kHz	Sets filter frequency for Low and High filters
		Shelve Gain	$-\infty$ to 0dB	Sets gain reduction for Low and High filters
		In/Out	In/Out	Switches each filter in and out of circuit.

9. X-ValveComp

INTRODUCTION

The illustration below gives an overview of some of the X-ValveComp features which are described in full over the following pages. Click on the blue boxes to be taken to fuller descriptions of their contents.



INTERFACE OVERVIEW

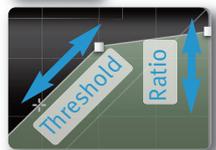
The basic interface techniques for the X-ValveComp are largely identical to those for the EQ & Dynamics Channel Strip (see Page 20). In addition to these basic techniques, the following are also available:



To control **Input** and **Output** levels, click and drag up- or downwards (see left).

To enter a precise value for any parameter, double-click on its value display, enter a value on your computer keyboard, and press the Return key.

Threshold and **Ratio** values can also be controlled directly within the Compression Law display. Move the left hand node along its diagonal line to control the Threshold, and move the right hand node up and down to control the ratio.



Automation

Automation support for X-ValveComp is the same as for the EQ & Dynamics Channel Strip (see Page 20).

Plug-in Bypass



The **In/Out** switch located above the Input section provides an internal plug-in bypass. This allows for smoother In/Out comparisons by avoiding any latency issues associated with the host application's Bypass function. The button must be 'lit' for the compressor to be in circuit (as shown).

Presets

The **Load** and **Save** buttons at the base of the screen provide quick access to the host application's Import and Save Settings functions. For specific instructions on using presets within the host, consult the host application documentation. Presets use a standard file format which allows them to be moved between a variety of platforms and host applications.



A file of ready-made X-ValveComp presets are included in the plug-in installation. These provide starting points for a variety of instruments and effects. To load SSL presets, click on the Load button and navigate as follows:

Macintosh: /Applications/Solid State Logic/Presets/X-ValveComp
Windows 32-bit: c:\program files\solid state logic\presets\X-ValveComp
Windows 64-bit: c:\program files (x86)\solid state logic\presets\X-ValveComp

A-B comparisons

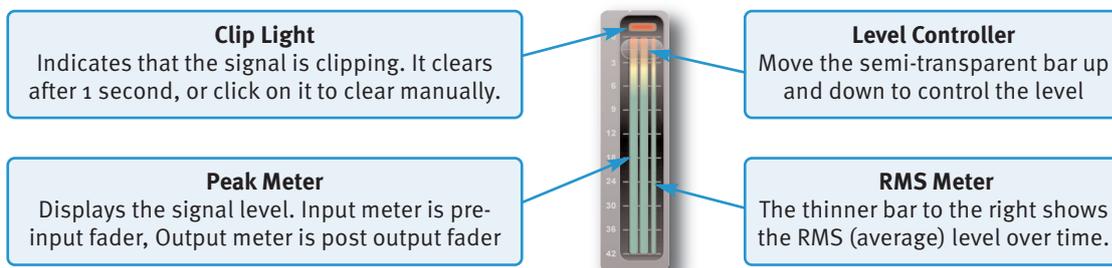


The **A B** button at the base of the screen allows you to load two completely independent set ups and compare them quickly. When the plug-in is opened setting A is selected by default. Pressing the A B button will switch between setting A and setting B.

You may wish to use this function to compare two very similar settings or to keep a 'maybe' setting in reserve if your current one does not work out.

Input and Output Levels

The input and output sections at either side of the plug-in window provide input and output gain control, along with level metering and clip lights:



Knee

The Knee controls how focussed the threshold level is:

- With a hard knee (knob at minimum) the compressor's parameters all come into force at precisely the point at which the threshold is crossed. In the Compression Law diagram, this is shown by a sharp change in gradient at the threshold.
- With a soft knee, the ratio is introduced gradually, starting below the threshold and reaching its full value above the threshold. In the Compression Law graph, this is shown by a curve in the gradient around the threshold.
- The Knee control also affects the main threshold point.



Valve

Valve introduces a tube compressor emulation for saturating the signal by adding harmonics.



Sidechain

Hi- and Lo-pass filters are available for the Compressor side-chain (see right).

To listen to the Compressor or side-chain, touch the **S/C Solo** button.

Mix



Dry/Wet option for parallel compression.

Auto Gain



You can allow the plugin to set the make-up gain automatically by touching the **Auto** button (towards the bottom of the screen).



10. X-Saturator

INTRODUCTION

The illustration below gives an overview of some of the X-Saturator features which are described in full over the following pages. Click on the blue boxes to be taken to fuller descriptions of their contents.

Above the main plug-in window is a set of controls which are specific to your host application. Please refer to your host application's user guide for guidelines on using these controls

6dB boost for adding headroom above the saturation point

Dry/Wet switch for parallel processing

Internal bypass for smooth in/out comparisons

Input level metering and control

Drive control for adjusting the amount of overtones added to the signal

Harmonics control for adding or subtracting 2nd order harmonics

Depth and Shape determine the level of the harmonics injected back to the signal

Output metering and control

INTERFACE OVERVIEW

The basic interface techniques for the X-Saturator are largely identical to those for the EQ & Dynamics Channel Strip (see Page 20). In addition to these basic techniques, the following are also available:

To control Input and Output levels, click and drag up- or downwards (see left).

To enter a precise value for any parameter, double-click on its value display, enter a value on your computer keyboard, and press the Return key.

Automation

Automation support for X-Saturator is the same as for the EQ & Dynamics Channel Strip (see Page 20).

Plug-in Bypass

The **In/Out** switch located above the Input section provides an internal plug-in bypass. This allows for smoother In/Out comparisons by avoiding any latency issues associated with the host application's Bypass function. The button must be 'lit' for the compressor to be in circuit (as shown).

Presets

The **Load** and **Save** buttons at the base of the screen provide quick access to the host application's Import and Save Settings functions. For specific instructions on using presets within the host, consult the host application documentation. Presets use a standard file format which allows them to be moved between a variety of platforms and host applications.



A file of ready-made X-Saturator presets are included in the plug-in installation. These provide starting points for a variety of instruments and effects. To load SSL presets, click on the Load button and navigate as follows:

Macintosh: /Applications/Solid State Logic/Presets/X-Saturator
Windows 32-bit: c:\program files\solid state logic\presets\X-Saturator
Windows 64-bit: c:\program files (x86)\solid state logic\presets\X-Saturator

A-B comparisons

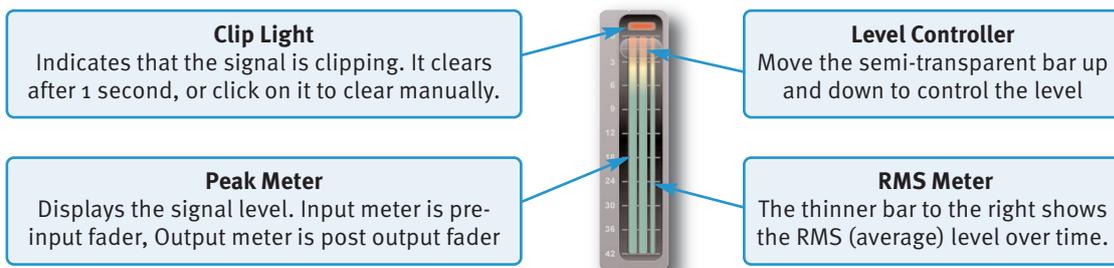


The **A B** button at the base of the screen allows you to load two completely independent set ups and compare them quickly. When the plug-in is opened setting A is selected by default. Pressing the **A B** button will switch between setting A and setting B.

You may wish to use this function to compare two very similar settings or to keep a 'maybe' setting in reserve if your current one does not work out.

Input and Output Levels

The input and output sections at either side of the plug-in window provide input and output gain control, along with level metering and clip lights:



Boost +6dB



Will add 6dB of Headroom above your saturation point.

Drive



Dialing in the drive will determine the level of saturation that will be added.

Harmonics



By adjusting the Harmonics control the distortion characteristics can be shifted from '50s valve-style overdrive (if the control is turned fully anti-clockwise for 2nd harmonic distortion) to '70s transistor-style grit (if the control is turned fully clockwise for 3rd harmonic distortion).

Depth / Shape

Depth and **Shape** control the number of harmonics injected back into the signal, enriching the sound.

Depth sets the amplitude of the added harmonics, so at 0% no harmonics at all are injected to the signal. **Shape** will give you the opportunity to add more overtones in the high frequency spectrum while -50 will be a little more subtle and smooth and +50 will sound more aggressive since the volume of 2nd and 3rd order harmonics is increased, too.



11. X-Phase

INTRODUCTION

The illustration below gives an overview of some of the X-Phase features which are described in full over the following pages. Click on the blue boxes to be taken to fuller descriptions of their contents.

The screenshot shows the X-Phase plugin interface with several callout boxes:

- Top Left:** A box pointing to the 'In/Out' switch and input meters, stating: "Above the main plug-in window is a set of controls which are specific to your host application. Please refer to your host application's user guide for guidelines on using these controls".
- Top Center:** A box pointing to the 'L R' switch, stating: "Left/Right switch to independently edit the left and right signal (stereo version only)".
- Middle Left:** A box pointing to the input gain control, stating: "Internal bypass for smooth in/out comparisons".
- Bottom Left:** A box pointing to the input level meter, stating: "Input level metering and control".
- Bottom Center:** A box pointing to the delay control knob, stating: "Delay control for adjusting the delay".
- Bottom Right:** A box pointing to the allpass filter section, stating: "Allpass filter section".
- Far Right:** A box pointing to the output gain control, stating: "Output metering and control".

INTERFACE OVERVIEW

The basic interface techniques for the X-Phase are largely identical to those for the EQ & Dynamics Channel Strip (see Page 20). In addition to these basic techniques, the following are also available:

To control Input and Output levels, click and drag up- or downwards (see left).

To enter a precise value for any parameter, double-click on its value display, enter a value on your computer keyboard, and press the Return key.

Automation

Automation support for X-Phase is the same as for the EQ & Dynamics Channel Strip (see Page 20).

Plug-in Bypass

The **In/Out** switch located above the Input section provides an internal plug-in bypass. This allows for smoother In/Out comparisons by avoiding any latency issues associated with the host application's Bypass function. The button must be 'lit' for the compressor to be in circuit (as shown).

Presets

The **Load** and **Save** buttons at the base of the screen provide quick access to the host application's Import and Save Settings functions. For specific instructions on using presets within the host, consult the host application documentation. Presets use a standard file format which allows them to be moved between a variety of platforms and host applications.



A file of ready-made X-Phase presets are included in the plug-in installation. These provide starting points for a variety of instruments and effects. To load SSL presets, click on the Load button and navigate as follows:

Macintosh: /Applications/Solid State Logic/Presets/X-Phase
Windows 32-bit: c:\program files\solid state logic\presets\X-Phase
Windows 64-bit: c:\program files (x86)\solid state logic\presets\X-Phase

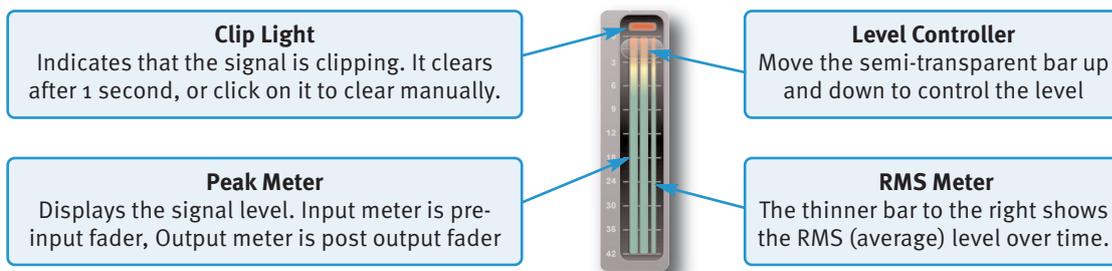
A-B comparisons

The **A B** button at the base of the screen allows you to load two completely independent set ups and compare them quickly. When the plug-in is opened, setting A is selected by default, as shown right. Pressing the **A B** button will switch between setting A and setting B.

You may wish to use this function to compare two very similar settings or to keep a 'maybe' setting in reserve if your current one does not work out.

Input and Output Levels

The input and output sections at either side of the plug-in window provide input and output gain control, along with level metering and clip lights:



Delay

Activate or deactivate a delay for the whole signal with the **Delay In** switch. Use the dial to set the delay in seconds, milliseconds and samples, meters or feet. The single +/- switches will give you smaller increments than the double +/- switches.



Allpass In

Allpass In Activates the Allpass Filter.

Allpass Phase

Allpass Inverts the phase of the Allpass Filter.

Frequency

Sets the Frequency of the filter.

Filter

Switches the filter between first and second order.

Q Factor

Sets the Q of the Filter, thus it will adjust the rate of phase change across the frequency spectrum.



Stereo Version

L R Stereo versions have an additional **L-R-Switch** to edit stereo signals.

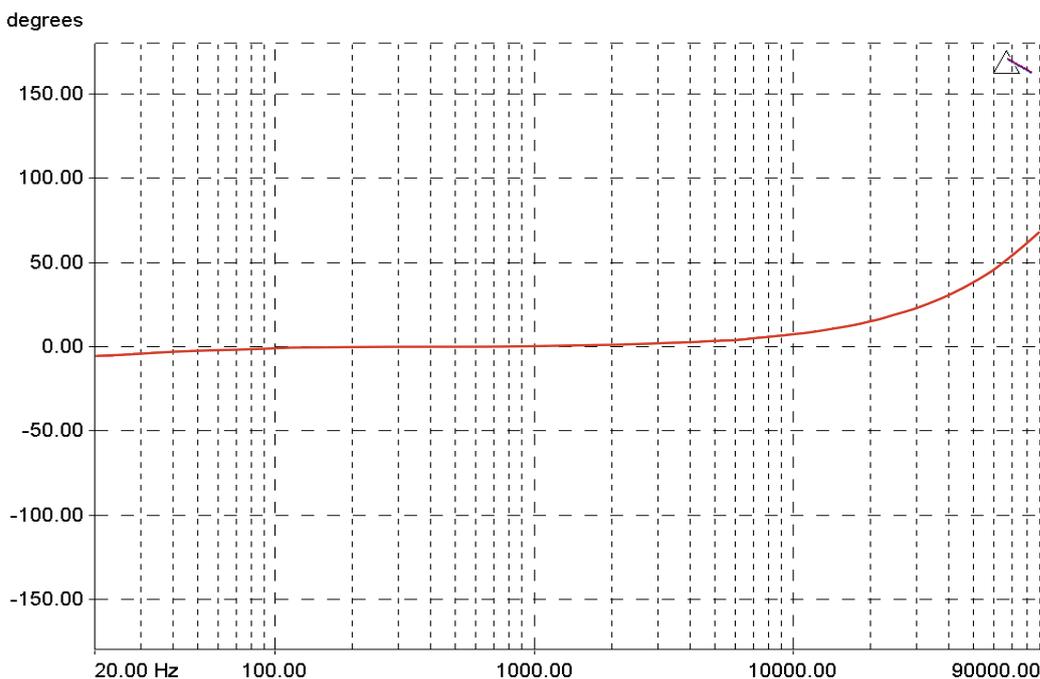
PHASED BY PHASE?

Phase. It's a funny subject, one that many of us think we know a bit about but unfortunately don't often fully understand how to handle all of it. The idea that 2 microphones, set at different distances from a guitar, will have the sound arrive at them at different times – thus making them “out of phase” with each other – is fairly simple to grasp. To fix this we could delay one of the signals by the same time it takes the sound to travel between the 2 microphones so that the signal arrives at the Mix bus at the same point in time. We're quite used to the idea of “flipping the phase” of a signal using a mixing console or our DAW to try and get 2 signals aligned therefore sounding better. What this does is to change the phase of the signal by 180° , or to reverse the polarity of it essentially “turning your waveform upside down”. We're also used to aligning audio files so that peaks of layered sounds all happen at exactly the same point in time, therefore accounting for the timing differences between microphones due to their different distances from the source.

However, how many of us have ever thought that a signal could be out of phase with itself? Or, more correctly, how many of us ever thought that different frequencies within a signal could be out of phase with each other?

You'd be surprised how much this can affect your audio...

The below chart shows the phase response chart of the X-Desk unit:



This line shows the phase relationship between different frequencies in the signal sent through the X-Desk. SSL products are known for their “width” and “depth” in the mix, and much of this is down to the phase relationships between different parts of the signal. You can see that there is a fairly large “flat” section of phase response between 80Hz and 2kHz. When audio passes through the X-Phase the phase of these frequencies remain largely unaffected by passing through the circuit. However, looking to the right hand side of the chart you can see that a frequency of 20kHz is 20° out of phase to a frequency of 1kHz. This will affect how the frequencies in that signal interact by reinforcing and cancelling each other depending on their phase which can lead to comb filtering.

You'd be amazed at some of the phase differences that happen when you run a signal through a device, with some particularly expensive devices having a $\pm 70^\circ$ degree phase difference between the low and the high frequencies, meaning that different parts of the audio signal could be 140° out of phase with each other.

This is what X-Phase helps to solve. Not only getting your signals in phase with each other, but also the frequencies within your signal to be in phase with themselves.

Here's the science...

THE ALLPASS FILTER IDEA

An all-pass filter is a signal processing filter that passes all frequencies equally in gain, but changes the phase relationship between various frequencies. All frequency components of a signal are delayed when passed through a device such as an amplifier, a loudspeaker, or when propagating through space or a medium such as air. This signal delay will be different for the various frequencies unless the device has the property of being linear phase. (Linear phase and minimum phase are often confused – they are quite different.)

The delay variation means that signals consisting of multiple frequency components will suffer distortion because these components are not delayed by the same amount of time at the output of the device. This changes the shape of the signal in addition to any constant delay or scale change. A sufficiently large delay variation can cause problems such as poor fidelity in audio or intersymbol interference (ISI) in the demodulation of digital information from an analog carrier signal.

X-Phase can solve these problems, for example, when aligning the overhead microphones of a drumkit to the direct signals while taking care of the fact that different frequencies need different delays.

Many people will be familiar with All Pass Filters as part of a loudspeaker processor and this is a very good way to visualise them. Think about a 2-way loudspeaker with a bass and high frequency driver. The acoustic centres of the two drivers are slightly different because of their physical placement. Even with the best crossover there are always frequencies that both drivers are reproducing at the same time. When the sound from each driver combines you will get ‘comb filtering’, or lobbing, as different wavelengths combine at different times. This will cause some frequencies to be completely cancelled, partly cancelled or combined to increase the level of others. These artifacts cannot be corrected using EQ filters, they can be used to partly mask the problem, but they won’t solve it.

So how can we fix this problem? The answer is to use an All Pass Filter. By setting the correct frequency and steepness we can correct the phase differences between the drivers, allowing them to sum together more accurately at the crossover point. The All Pass Filter acts like a Tilt Control, shifting the phase of some frequencies relative to others.

So how does this relate to the creative mixing process, rather than correcting loudspeakers?

If we now take the principle of the loudspeaker alignment example and apply it in the same way to instrument inputs, it will become clearer where the All Pass Filter solves problems in a way no other audio processing can.

Imagine recording Bass, a Kick Drum, a Snare, a Piano, or any instrument that will involve more than one microphone for recording. Most engineers will use a combination of Polarity switching and/or delay to try and align the two sources so that they combine more smoothly. But let's take the Snare top and bottom mics as an example.

In the same way that the high and low frequency drivers of the loudspeaker have some frequencies that overlap so do the snare top and bottom mics.

Most engineers will try switching the polarity, thinking that negative pressure on the snare top mic will be reproduced as positive pressure on the bottom. On a very simple level this is true, but the phase will not be consistent across the whole frequency range, so comb filtering will occur. Switching Polarity will just accentuate some frequencies and filter out others, it will not allow the frequencies that both microphones are picking up to combine correctly in time.

Bass DI and microphone channels suffer from a similar problem, but this time engineers tend to reach for the delay, thinking that the mic signal is behind the DI in time. This is true to some degree, but the bass speaker also has its own frequency dependent phase response, which a delay can't fix because it affects all frequencies. Both these examples use a tool to try and fix a problem, but neither delay or polarity are the right tool. They are like a sledgehammer.

What we need is a more accurate and delicate tool and this is where the All Pass Filter comes in. This is more like a precision tool, like a scalpel. Let's take a different example, an acoustic piano. If you use 3 microphones for a piano they are all separated by different physical distances and that are all picking up similar frequencies. But when you try and mix those 3 microphones together you will have similar comb filtering effect to the loudspeaker once again. This will affect your tone and your gain before feedback. Once again, delay, polarity or EQ can't fix this problem, they can only mask it to some degree.

Using an All Pass Filter we can adjust the phase response of each mic input so that they combine together more coherently. This gives us a more natural over-all tone from the piano and much higher gain before feedback as both the phase and frequency response is smoother. The All Pass Filter is a tool like no other in a sound engineer's tool kit.

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