> ## LiquidControl™ Software
>
> **Please visit www.ffliquid.com to download the latest version of the free LiquidControl™ software application, for Mac OS X or Windows XP. The application allows you to edit The Liquid Channel™ remotely, plus load, save and archive the mic-pre and compressor replicas and programs. The LiquidControl™ user guide is also available to download in PDF format.**

Contents

Introduction

Thank you for purchasing The Liquid Channel™ brought to you by the Focusrite team – Ian, Trevor, Peter, Martin, Tom, Mick A'C, Phil, Chris G, Micky, Pauline, Melissa, Chris W, Rob J Snr, Simon J, Vernon, Giles, Rob J Jnr, Mick G, Tim, Dave, Nick, Paul, Peter and Simon.

The chaps at Focusrite are a jolly hard working bunch and take a great deal of pride in designing, building and delivering products which are considered the best audio units around; we hope your new Focusrite unit lives up to that reputation and that you enjoy many years of productive recording. If you would like to tell us about your recording experiences then email us at: sales@focusrite.com

The Focusrite Team

Important Safety Instructions

Please read all of these instructions and save them for future reference. Follow all warnings and instructions marked on the unit.

- Do not obstruct air vents in the rear panel. Do not insert objects through any apertures. Ensure adequate ventilation at all times.
- Do not use a damaged or frayed power cord.
- Unplug the unit before cleaning. Clean with a damp cloth only. Do not spill liquid on the unit.
- Unplug the unit and refer servicing to qualified service personnel under the following conditions: if the power cord or plug is damaged; if liquid has entered the unit; if the unit has been dropped or the case damaged; if the unit does not operate normally or exhibits a distinct change in performance. Adjust only those controls that are covered by the operating instructions.
- Do not defeat the safety purpose of the polarised or grounding type plug. A polarised plug has two blades with one wider than the other. A grounding type plug has two blades and a third grounding prong. The wider blade or the third prong are provided for your safety. When the plug provided does not fit into your outlet, consult an electrician for replacement of the obsolete outlet.

This unit is capable of operating over a range of mains voltages as marked on the rear panel. Ensure correct mains voltage setting and correct fuse before connecting mains supply. Do not change mains voltage settings while mains supply is connected. To avoid the risk of fire, replace the mains fuse only with the correct value fuse, as marked on the rear panel. The internal power supply unit contains no user serviceable parts. Refer all servicing to a qualified service engineer, through the appropriate Focusrite dealer.

> **WARNING:**
> **THIS UNIT MUST BE EARTHED BY THE POWER CORD. UNDER NO CIRCUMSTANCES SHOULD THE MAINS EARTH BE DISCONNECTED FROM THE MAINS LEAD.**

Power Connections

There is an IEC mains lead supplied with the unit, which should have the correct moulded plug for your country. The wiring colour code used is:

For units shipped to the USA, Canada, Taiwan and Japan:
Live Black; **Neutral** White; **Earth** Green

For units shipped to any other country:
Live Brown; **Neutral** Blue; **Earth** Green & Yellow

Liquid Channel™ Architecture

The Liquid Channel™ is fundamentally different from any other Focusrite product previously built – and indeed any other dynamics processor currently available. Although The Liquid Channel™ is extremely simple to use, please take a moment to read this section of the user guide, which gives a brief overview of the signal path and storage capabilities of the unit.

Overview

The Liquid Channel™ is the first implementation of Focusrite's groundbreaking 'Liquid Technology'. The Liquid Channel™ transports classic front-end processing into the digital age, removing the need for endless patching and adding fluidity and reliability to the studio environment. Quite simply, The Liquid Channel™ is capable of being configured to replicate the finest mic-pres and compressors in history, all inside a single 192 kHz, 2U device.

This is achieved through a unique two-part process. Firstly, the use of Dynamic Convolution techniques is applied, utilising lightning fast SHARC chips applying unique level-dependent impulse responses to every sample of audio. However, whilst this is more than sufficient to closely emulate a compressor's sonic behavior, in order to replicate subtle nuances of classic units, the pre-amplifier must operate in a suitable physical environment to mirror the way in which any classic unit interacts with a given microphone. The Liquid Channel™'s solution to this problem lies in its vast analogue front-end. The pre-amp has the ability to change its impedance and vary its signal path to either transformer or electronic, replicating the interaction characteristics of the original, whilst remaining transparent within the signal path.

The Liquid Channel™ can therefore closely replicate the sound of the classic mic-pres and compressors from history. With entirely digital front panel controls, all parameters can be saved in one of 100 program memories, meaning that entire session setups can all be recalled at the touch of a button. A brand new digital EQ is also available, providing a comprehensive and truly 'liquid' channel strip.

A USB connection on the rear panel links to the LiquidControl™ software application, enabling the archiving of both replicas and surplus program memories, as well as providing full editing and remote operation of the unit itself. The Liquid Channel™ comes complete with replicas of 40 classic mic-pres and forty classic compressors but is infinitely expandable, since the USB port also facilitates the upload of further replicas downloaded from a dedicated website – **www.ffliquid.com**.

Finally, to account for variances in amounts of second, third and fifth order harmonic distortion (usually perceived as warmth) from one pre-amp to the next, a dial is included to permit control over this value. This essential feature ensures satisfaction for even the most hardcore vintage enthusiast, bringing accurate control over the key sonic attributes.

Liquid Assets

The Liquid Channel™ ships with 40 pre-amp replicas and 40 compressor replicas as standard. The free LiquidControl™ software allows additional replicas downloaded from **www.ffliquid.com** to be uploaded to The Liquid Channel™ via USB.

Program memories

The Liquid Channel™ has 100 program memories to allow storage of preferred pre-amp/compressor/EQ combinations. These can be saved, archived and reloaded from a remote computer using LiquidControl™ via the USB connection. All channel parameters are stored, offering 'total recall' including mic gain. Using LiquidControl™, a virtually unlimited number of replicas and programs can be stored on your PC or Mac and uploaded to The Liquid Channel™ hardware as required. You can also email your favourite Liquid Channel™ replicas and program settings to other users anywhere in the world – the ultimate in session recall.

Signal Path

The Liquid Channel™ features both analogue (mic and line) and digital (AES-EBU) inputs, plus analogue (line) and digital (AES-EBU) outputs. Analogue signals connected to the mic input are routed through an analogue mic-pre stage, before passing through an A/D and entering the digital pre-amp convolution section. The unique sonic characteristics of the selected pre-amp replica are then applied to the signal. The signal then passes through the compressor and EQ sections (in that order, unless the EQ 'Pre Comp' switch has been engaged – see page 7 for details). This processed signal then routes to the digital output, and via the D/A to the analogue output.

Analogue signals connected to the line input follow the same signal path as above, but bypass the initial mic-pre stage routing directly to the A/D. The pre-amp type will be fixed at FLAT trfmr (transformer) whilst the line input is selected on the front panel. Digital signals connected to the digital input are routed directly to the pre-amp convolution section. However this can be bypassed – see 'Pre-amp and Compressor Select' on page 5 for details.

Rear Panel Connections

XLR Audio Inputs and Output

All 3-pin XLR audio connectors (MIC I/P, LINE I/P and LINE O/P – from left to right above) are wired as follows:
Pin 1: Screen/Chassis
Pin 2: Audio 0°
Pin 3: Audio 180°

AES Input and Output

The AES I/P and AES O/P XLR connectors provide The Liquid Channel™ with AES-EBU format digital input and output. See pages 11-15 for details of digital applications and connectivity.

Wordclock Input and Output

The WORDCLOCK INPUT BNC connector allows The Liquid Channel™ to be linked to an external clock source. The WORDCLOCK OUTPUT BNC connector allows The Liquid Channel™ to supply a clock source to external equipment. See page 9 for more details on synchronising The Liquid Channel™ to an external Wordclock source.

Dynamic Link Bus

The DYNAMIC LINK BUS connectors allow two or more units to be linked together for stereo and multi-channel applications, whereby one unit (the 'master') will control the settings on multiple units. In addition, if using multiple units in digital stereo pairs, the dynamic link connectors also transmit digital audio data between devices. The connectors are both RCA (phono). See page 10 for more details on linking multiple units.

USB

The USB connector allows The Liquid Channel™ to be connected to a computer running the free LiquidControl™ software, which allows remote operation, download and archive management. See the separate LiquidControl™ user guide, downloadable from **www.ffliquid.com**, for more information. For remote operation at a greater distance (up to 45m), contact Focusrite for information about purchasing a USB to CAT5 (Ethernet) converter.

Front Panel Controls

Digital controls

The dials are all tactile rotary encoders, meaning that they can be rotated infinitely and their relative value will be displayed (except for the DATA wheel) by the LEDs surrounding them. The exact numerical parameter values can be seen on the large display when active, in the centre of the front panel.

As all controls are digital, the settings can be saved and recalled with ease, and edited from the LiquidControl™ software application (please see the separate LiquidControl™ documentation, available from www.ffliquid.com for details). N.B. The hardware's onboard memory will store the last front-panel settings to have been left standing on the unit for over 10 seconds while powered up, while the unit is turned off.

Power

Applies power to the unit. Turn on The Liquid Channel™ before powering up devices to which the outputs are connected.

Mic-Pre and A/D
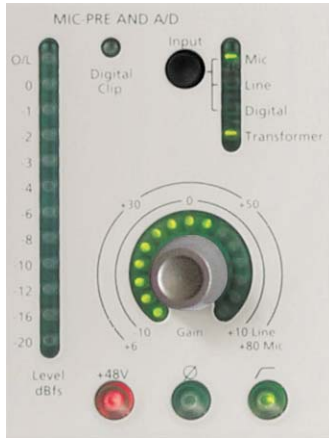
Meter

A single, vertical, peak hold LED bar graph displays the level of the signal being fed from the mic-pre to the A/D converter, controllable by the gain dial in the mic-pre section. The scale shows from –20 dBFS up to the maximum 0 dBFS (digital clipping level) with an additional LED to represent overload (O/L). Exceeding this maximum will result in the A/D clipping the signal, which may cause audible and unwanted distortion.

Digital Clip LED

A single separate red LED is also included to signify when a digital clip is occurring.

Input Select

The input Select switch, pressed repeatedly, will step through the mic, line and digital inputs, with an LED beside each, which illuminates when that input is active. A further LED is provided to show when the transformer has automatically been switched in circuit; this will change according to the mic-pre replica selected.

Gain dial

The dial below the input select switch adjusts the level fed to the A/D. The levels for mic and line are shown by the outside (mic) and inside (line) arc around the dial, with the exact numeric value displayed next to the pre-amp replica name on the front panel LCD (indicated by 'Pre-amp Gain dB').

+48V

Pressing the +48V switch provides +48V phantom power, suitable for condenser microphones, to the rear panel XLR microphone connector. This switch does not affect the other inputs.

If you are unsure whether your microphone requires phantom power, refer to its handbook, as it is possible to damage some microphones (most notably ribbon microphones) by providing phantom power.

Ø

Pressing Ø inverts the phase of the selected input, primarily used to correct phase problems when using multiple microphones on a single source.

⌐

This switches an analogue, pre-A/D high-pass filter into the signal path, in order to roll off low frequencies. This is switchable (in the Setup menu) between a 75 Hz and 120 Hz roll-off. See page 9 for more information about the Setup menu. The roll-off is 12 dB per octave, 6 dB down at 75 or 120 Hz.

Session Saver

Next to the Mic-Pre and A/D section and above the Clock Select switch is the Session Saver switch. Press this switch once to arm the Session Saver. Once this circuit is armed (indicated by the corresponding LED) a gain reduction will occur if the signal starts to clip. This is a way of protecting your session from continual overload if the level is consistently 'hot'.

The Session Saver's action is different from a limiter's action, as the signal isn't compressed in any way, just turned down by an appropriate amount to avoid overloading. The feature monitors the signal at the input and output. If the input is overloading, the mic-pre gain is reduced. If the output is clipped but the input is clean, the level of the makeup gain is lowered. When the section has reduced the gain, the Activated LED will illuminate. Once the level is decreased, it remains activated until the switch is pressed to disarm the Session Saver. Pressing once more will then re-arm the Session Saver.

Clock Select and Synchronisation
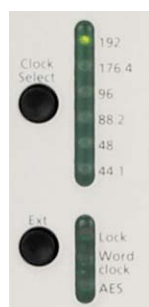
Clock Select

This switch changes the clock to the sample rate value indicated by the corresponding LED: 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz, 176.4 kHz or 192 kHz. If using an external digital input, the sample rate must be set to match that of the digital signal fed to the input.

Ext

This switch allows The Liquid Channel™ to synchronise to an external source, either the Wordclock or AES input on the rear panel, each shown by an LED. When a lock is achieved, the corresponding LED will illuminate. If the LED flashes the unit is not locked and jitter will be evident.

N.B. If either Wordclock or AES are selected and there is no cable connected to the relevant input, The Liquid Channel™ will not achieve lock.

When processing in 176.4 or 192 kHz, The Liquid Channel can accept and regenerate either 88.2/176.4 kHz or 96/192kHz respectively, depending on the clock source and destination requirements. See the Setup menu section on page 9 for more details.

Main LCD and Controls

Pre-amp Gain DB

This area of the display shows the gain (as set using the Gain control in the Mic-Pre and A/D section) in dB.

Program Name

This area of the display shows the name of the currently loaded program memory. Naming and saving of programs are separate commands; it is advisable to name the program first, then save it.

Number

This area of the display shows the number of the currently loaded program memory, from 00-100. Selecting '00' allows the user to return to the factory default program memory, which always loads the 'FLAT trfmr' mic-pre and 'FLAT comp' compressor.

Unlike program memories 01-100, program memory 00 cannot be overwritten.

Pre-amp and Compressor Select

When beginning a session for the first time, it is recommended to load default program 00 (see above), as this is an ideal starting point.

Pressing the Pre-amp Select switch causes its LED to flash. Rotating the DATA wheel then allows a microphone pre-amp replica (shown in the top left hand corner of the display window) to be selected by scrolling through a bank of pre-amp replicas. When the desired pre-amp is found, press the DATA wheel once to load the required pre-amp replica.

N.B. To use The Liquid Channel™ as a standalone compressor with an analogue input, select the Line input, which will automatically select the transparent

'FLAT trfmr' (transformer) input stage. To use The Liquid Channel™ as a standalone compressor with a digital input, you can select the 'FLAT dig'l' pre-amp on the front panel.

Pressing the Compressor Select switch causes its LED to flash. Rotating the DATA wheel then allows a compressor replica (shown in the top right hand corner of the display window) to be selected by scrolling through the bank of compressor replicas. When the desired compressor is found, press the DATA wheel once to load the required compressor replica.

N.B. Loading a new mic-pre or compressor will not affect any values dialled in on the front panel. For example, if the user changes a compressor's threshold value from −7 dB to −9 dB and then loads a new compressor replica as above, that new compressor replica's threshold value will initially be −9 dB. (If 'As Original Model' is selected in the Setup menu (see page 9) the value will reset to the nearest available value.) To save all edited values when you have finished editing so that they may be recalled later, use the Save switch (see page 8).

Harmonics
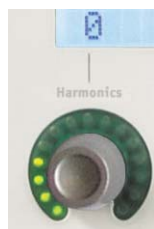
The dial directly below the pre-amp name allows desirable harmonic distortion, perceived as 'warmth', to be applied. The exact amount of 2nd-, 3rd-and 5th-order harmonics depends both on the type of pre-amp chosen and the amount of mic-pre gain. This is because the amount of each harmonic with respect to each other is different for pre-amps with valves compared to those with transformers; and also because the balance of harmonics changes with level. This means that a higher gain at the mic-pre stage will result in a greater, and hence more noticeable, effect as the harmonics setting is increased. Special 'hot' replicas, which can be downloaded from www.ffliquid.com, provide greater levels of harmonics application, for those wanting to achieve a seriously overdriven sound.

The harmonics control allows replication of the subtle variations between analogue devices, (sometimes two units of the same type can have different amounts of harmonic distortion), and allows the user to create an 'overdriven' sound without having to overdrive the pre-amp. Careful use of the Harmonics control will

thus allow the user to 'tune in' The Liquid Channel™ replicas to their own individual vintage unit.

The amount of harmonic distortion is indicated by the LEDs surrounding the dial and a value from 0-15 shown above it. '+ODD' appears next to the value when third and fifth order harmonics are present.

Compressor Settings

N.B. If the Compressor Controls option in the Setup menu has been set to 'As Original Model', then depending on the compressor replica selected, not all of the following settings may be available. With 'As Original Model' selected, only those controls that were present on the original unit are available for editing, all other dials display 'FIXED' and are inactive. Similarly, if the dials' values are reversed on the original or they have some special operational features, then these are also replicated on the front panel. See page 9 for more information on the Setup menu.

Comp In

The compressor is activated by pressing the Comp In switch, located in the bottom right corner of the middle section.

Threshold

The Threshold control sets the level at which compression begins. The lower the threshold, the more the signal is compressed. Setting a higher threshold allows quieter passages in the music or speech to remain unaffected; only passages that exceed the threshold will be compressed. The level of threshold varies depending on the chosen compressor, and the exact value is shown in dB directly above the dial on the display window.

Ratio

The Ratio control determines the amount of compression applied to the signal with increasing input, and is the ratio of change in input level compared to change in output level. Higher ratio settings will produce more noticeable compression, so for the least obtrusive result, the ratio should be set at the minimum necessary for the application. For example, using low threshold and low ratio will produce a less subjectively noticeable effect than a high threshold and high ratio, even though the total amount of compression may be the same. The range of ratios selectable varies depending on the chosen compressor replica, and the exact value is shown directly above on the display window.

Attack

The Attack control determines how quickly compression is applied once the level of the source signal has risen above the threshold. When turned anti-clockwise the response is very fast, which tends to make the compressor react to the peak levels of the signal. This is sometimes desirable, but short transients can cause unwanted 'pumping' of steadier low-level signals. A slower attack will cause the compressor to ignore short transients and respond more to the average loudness of the signal; however this may seem to increase the relative volume of the transients. The range of attack times selectable varies depending on the chosen compressor, and the exact value is shown in milliseconds (mS) directly above on the display window.

Release

The Release control determines how quickly compression is removed once the level of the source signal has fallen below the threshold. When in the anti-clockwise position, the compression releases very quickly, which may be appropriate on rapidly varying signals to avoid compressing the beats that follow, but can result in excessive distortion on more sustained material. Clockwise rotation increases the release time, giving a smoother effect, but may also result in transients causing audible 'pumping'. The Release time varies depending on the chosen compressor, and the exact value is shown in milliseconds (mS) directly above on the display window.

Makeup

Compression results in an overall reduction in level. The Makeup control allows you to increase the gain of the compressed signal. The range of makeup values selectable varies depending on the chosen compressor, and the exact value is shown in dB directly above on the display window.

Stereo Link

This switch enables Liquid Channel™ units to be linked, so that the 'master' can control multiple 'slave' units. To set whether the unit is the 'master' or 'slave', the Setup menu must be accessed. See page 9 for more information on the Setup menu, and page 10 for more information on linking multiple units.

Gain Reduction bar graph

The vertical LED meter indicates the action (Gain Reduction) of the compressor, in expanding increments down to –15 dB.

3-Band EQ
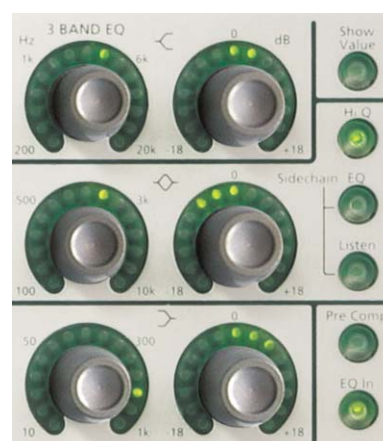
The Liquid Channel™ features a new digital ('modelled' rather than 'convolved') EQ, loosely based on the classic Focusrite sound of the original ISA 110. This is designed to allow small amounts of corrective shaping. All of the EQ settings can be saved, along with those of the mic-pre and compressor, in one of the Program slots (see the DATA SELECT section below).

EQ In

Located in the bottom right corner of the section, this switch activates the EQ section and is illuminated when the EQ is active.

Pre Comp

Pressing this switch (illuminated when active) allows the EQ to be moved to a position prior to the compressor in the signal path.

Show Value

Pressing this switch (illuminated when active) causes the exact values of each EQ dial to be shown in the display window, for more visual accurate parameter modification.

High-frequency shelving

The two upper dials control the frequency (left dial) and gain (right dial) of the high-frequency shelf. The frequency varies from 200 Hz to 20 kHz, and the gain between –18 dB and +18 dB.

Low-frequency shelving

The two lower dials control the frequency (left dial) and gain (right dial) of the high-frequency shelf. The frequency varies from 10 Hz to 1 kHz, and the gain between –18 dB and +18 dB.

Parametric mid-band

The two dials in the centre control the frequency (left dial) and gain (right dial) of the parametric mid-band. The frequency varies from 100 Hz to 10 kHz, and the gain between –18 dB and +18 dB.

Hi Q

This switch (illuminated when active) changes the Q to a higher value, causing the parametric controls to affect a narrower band of frequencies. With Hi Q switched out the Q value is 0.8. With Hi Q switched in the value is 2.5.

Sidechain EQ

This switch (illuminated when active) routes the parametric mid-band EQ section to the sidechain of the compressor for frequency-conscious compression. Use this to configure the compressor to compress some parts of the frequency spectrum more than others, for example when de-essing.

N.B. The high- and low-frequency shelving bands will still be active in the audio path, and will still be affected by the Pre Comp switch (see above) if engaged.

Sidechain Listen

This switch (illuminated when active) allows the sidechain to be monitored independently, allowing easy setup of frequency-conscious compression. Release the switch when setup is complete.

Comp & EQ Settings

Bypass

This switch (illuminated when active) allows the user to hard bypass the harmonics, compressor and EQ sections.

Compare

This switch allows the user to temporarily revert back to the saved program memory settings, allowing an A/B comparison between the saved and adjusted session setups. Press the switch once to revert to the saved program memory settings, and a second time to return to the currently edited settings.

Revert

Having used the Compare switch to qualify new settings, should the original (saved) program

parameters be preferred, then this switch can be pressed to revert to the previously saved settings. **N.B. Any edits made to the program settings will then be permanently lost.** Wordclock and phantom power settings are not saved in the unit's Program memories.

Data Select

This section is responsible for managing Program data, saving, naming and recalling sessions.

DATA

Rotating this dial allows the pre-amp or compressor replicas to be chosen (when the Pre-amp Select or Compressor Select switches are activated) and allows various parameters to be modified or selected in the DATA SELECT section. **To choose a parameter, once found, press the DATA wheel in once**.

Save

Pressing this switch allows the current session setup (all front panel controls) to be saved to an allocated program memory slot. Once the Save switch has been pressed and is flashing, the DATA wheel will scroll through the 100 program memories. When the relevant space is found (this may be an already named slot or an empty one) pressing the DATA wheel will save the settings. Press the Save switch again to cancel (without pressing the DATA wheel). When a parameter is changed on a saved program, a star will appear next to the Program number, showing the user that the Program has been edited and may need to be saved.

N.B. If you overwrite another program memory, that memory will be lost.

If you run out of spare program memories, the LiquidControl™ application allows you to store an unlimited number of programs on your Mac or PC and upload them to The Liquid Channel™ as required. See the LiquidControl™ documentation for details.

Recall

This switch allows a program memory to be loaded, recalling all settings for a particular session. Once the Recall switch has been pressed and is flashing, the DATA wheel will scroll through the 100 program memories. When a relevant program memory is found, pressing the DATA wheel will load the program memory. Press the Recall switch again immediately to cancel (without pressing the DATA wheel). Wordclock and phantom power settings are not saved in the unit's Program memories.

Name

This switch enables program memories to be named. Once the Name switch has been pressed and is flashing, the letters can be adjusted. Rotating the DATA wheel changes the current letter/symbol and pressing it moves on to the next character. **When the program memory name is completed, the program will need to be saved (unless already done). See the Save section above.**

Clear

When in Name mode, this switch erases all characters, to allow a new name to be entered from scratch. (Clear does not affect any other program settings.)

Setup

This menu allows various Liquid Channel™ settings to be adjusted. The DATA wheel scrolls through options when rotated, and will save the setting and move on to the next screen when pressed. Press the Setup switch once to exit. The Setup menu options are:

LF Filter
Use the DATA wheel to select a cut-off frequency of 75 or 120 Hz for the LF filter (see Mic-Pre and A/D section on page 4 for details of the LF filter).

Compressor Controls
This option allows the user to specify whether or not the compressor control ranges are restricted to those of the original units being replicated. Use the DATA wheel to select 'As Original Model' or 'FREE'. Selecting 'FREE' allows you to create new settings that were not possible on some original units.

Wordclock Input
If using the unit's digital input, running at 176.4 or 192 kHz and the Ext sync switch is set to wordclock input, then the unit can be set to lock to 88.2/176.4 kHz or 96/192 kHz respectively, depending on the clock source.

Wordclock Output
If using the unit's digital input and running at 176.4 or 192 kHz, the unit's wordclock output can be set to regenerate 88.2/176.4 kHz or 96/192 kHz respectively, depending on the clock destination.

Digital Link Bus (Gain Linkage)
If using more than one Liquid Channel™, this setting specifies how many machines are to be linked. Up to eight Liquid Channels may be chained via the DIGITAL LINK BUS connectors. See page 10 for more information about linking multiple units. **If using only one unit, ensure this is set to 'OFF'.**

Gain Linkage
If Digital Link Bus has been set to a value greater than 'OFF' (see above), this setting specifies whether The Liquid Channel™ is to act as the master or slave. With more than two units connected, each slave machine must be identified with a unique number, from 2 to 8. The available options are machine #1 (master) and machine #2 – #8 (slave). N.B. This option is only available when Compressor Gain Linkage is set to a number greater than 'OFF'.

Slave Controls to Master
If The Liquid Channel™ unit has been set up to slave to another master Liquid Channel™ unit, a 'Slave Controls to Master' menu will appear. Using the DATA wheel, the user may select 'Yes' or 'No'. Selecting 'Yes' will cause the slave Liquid Channel™ unit's controls to follow any changes made to the master Liquid Channel's controls. (The slave unit's controls become inactive.) Mic-pre and compressor replicas called up on the master unit will also be called up on the slave unit(s.) This is the standard setup for most linked applications.

Selecting 'No' will allow the user to continue to edit the slave's controls independently, even though the master and slave units' gain reductions are linked. Hence selecting 'No' allows the user to independently set up e.g. mic-pre gain levels, compressor ratios etc. independently on a slaved unit.

AES Linking
If the Digital Link Bus is not set to 'OFF', then an AES Linking option will appear in the Setup menu. This allows multiple Liquid Channels™ to be used as grouped AES pairs, by sending both units' signals down a single wire (as in diagram 5 on page 12).

There are three possible settings:
- Both units are receiving analogue signals and the master is sending AES – set master to 'Link AES – right: **from slave**' – and set slave to 'Link to master AES – right: **output**'.
- Both units are receiving analogue signals and the slave is sending AES – set master to 'Link AES – right: **to slave**' – and set slave to 'Link to master AES – right: **input**'.
- The master is receiving AES and sending AES – set master to 'Link AES – right: **to and from slave**' – and set slave to 'Link to master AES – right: **in and out**'.

USB ID

This setting allows the USB ID to be set between 1 and 8. When using the LiquidControl™ software application with multiple Liquid Channel™ units, each unit must be assigned a unique USB ID to allow it to be selected in the LiquidControl™ application. Turn the DATA wheel to change the USB ID value and press the DATA wheel once to enter the selected value. See the downloadable LiquidControl™ documentation for more information.
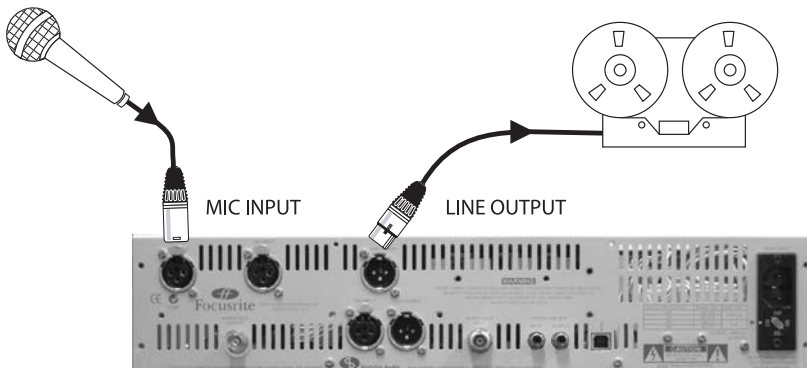
Linking Multiple Units

Using the DIGITAL LINK BUS connectors, up to eight Liquid Channel™ units may be linked, so that one 'master' unit controls the compressor, mic-pre gain and other parameters of all connected 'slave' units. This means you can use multiple Liquid Channel™ units for perfectly matched stereo or multi-channel surround mixing applications.

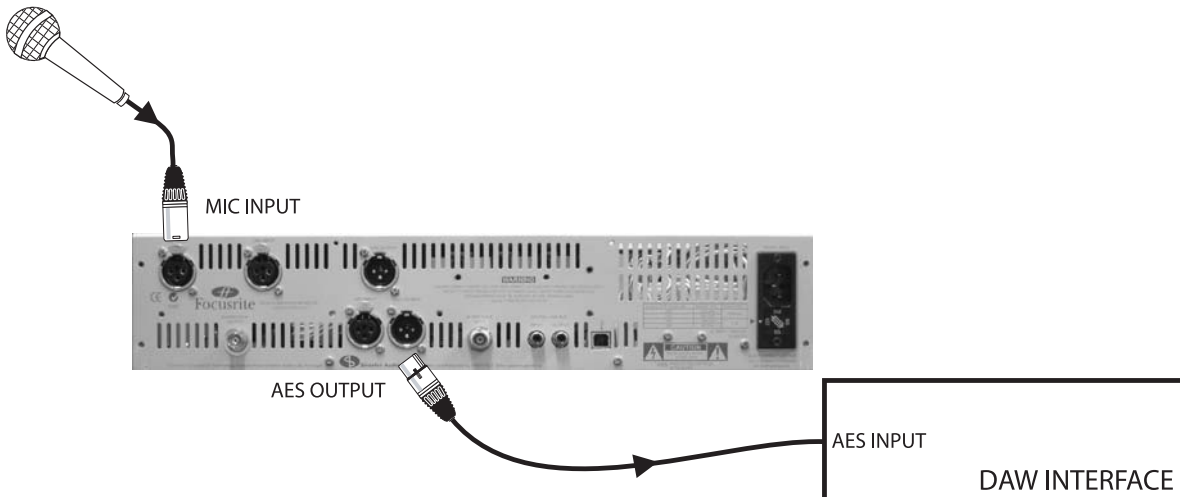The procedure for linking multiple units is as follows:

1. Connect an RCA cable from the DIGITAL LINK BUS output of the unit you intend to use as the 'master' to the DIGITAL LINK BUS input of the first 'slave' unit. When more than one slave is to be used, connect an RCA cable from the output of the first slave to the input of the second slave and repeat as required until all slave units are 'chained' together.

2. Connect the DIGITAL LINK BUS output of the final 'slave' unit in the chain to the DIGITAL LINK BUS input of the 'master' unit. See diagram 5 on page 12.

3. Set the 'Digital Link Bus' option in the Setup menu to display the number of units in use.

4. Set the 'Gain Linkage' option in the Setup menu (see page 9) to 'machine #1 (master)' on the master unit, and to 'machine #2 (slave)', 'machine #3 (slave)' etc. on each linked slave unit.

5. If all units are to be controlled by the master, then the slave units must all have their 'Slave Controls to Master' option in the Setup menu set to 'Yes'.

6. Ensure that the pre-amp and compressor replicas required by the program are loaded. If necessary use the LiquidControl™ application to upload the required Program and replicas to each unit. See the separate downloadable LiquidControl™ documentation for more information.

7. If you wish to use two Liquid Channels™ with digital input/output at sample rates of 44.1, 48 or 96 kHz, a stereo AES-EBU signal can be fed via a single AES connection (despite the fact that each Liquid Channel™ is a mono device). To route stereo digital audio to a pair of Liquid Channels™, first connect a suitable cable to the AES digital input of the first Liquid Channel™. Connect a pair of RCA (phono) to RCA (phono) cables between the DIGITAL LINK BUS output of the first Liquid Channel™ and the DIGITAL LINK BUS input of the second Liquid Channel™, and between the DIGITAL LINK BUS output of the second Liquid Channel™ and the DIGITAL LINK BUS input of the first Liquid Channel™. Then connect an AES cable from the AES digital output of the first Liquid Channel™ to route the stereo AES signal to the signal's destination. See diagram 9 on page 15 (Digital in/out stereo for digital insert, 44.1-96 kHz) for further detail. For multiple applications involving more than two Liquid Channels™ repeat this procedure for each pair of Liquid Channels™. N.B. If wanting to use more than 2 units as AES pairs, the units cannot all be dynamically linked. See diagram 6 on page 13. The AES Linking option in the Setup menu on each unit must also be set appropriately – see page 9 for details.

8. If using the LiquidControl™ application on an attached computer, assign each unit a unique USB ID using the 'USB ID' option in the Setup menu (see page 9). N.B. To avoid confusion, we recommend using a USB ID that matches the machine number set in 'Gain Linkage', i.e. 'machine #2 (slave)' should have a USB ID of 2.

Connections and Applications

1. All-analogue signal path

MIC INPUT   LINE OUTPUT

2. Digital out, Liquid Channel™ is clock master

MIC INPUT

AES OUTPUT

AES INPUT

DAW INTERFACE

3. Digital out, Liquid Channel™ slaved to Wordclock

MIC INPUT

AES OUTPUT   WORDCLOCK INPUT

WORDCLOCK OUT

AES INPUT

DAW INTERFACE

4. Digital out, Liquid Channel™ slaved to AES clock

MIC INPUT

AES INPUT   AES OUTPUT

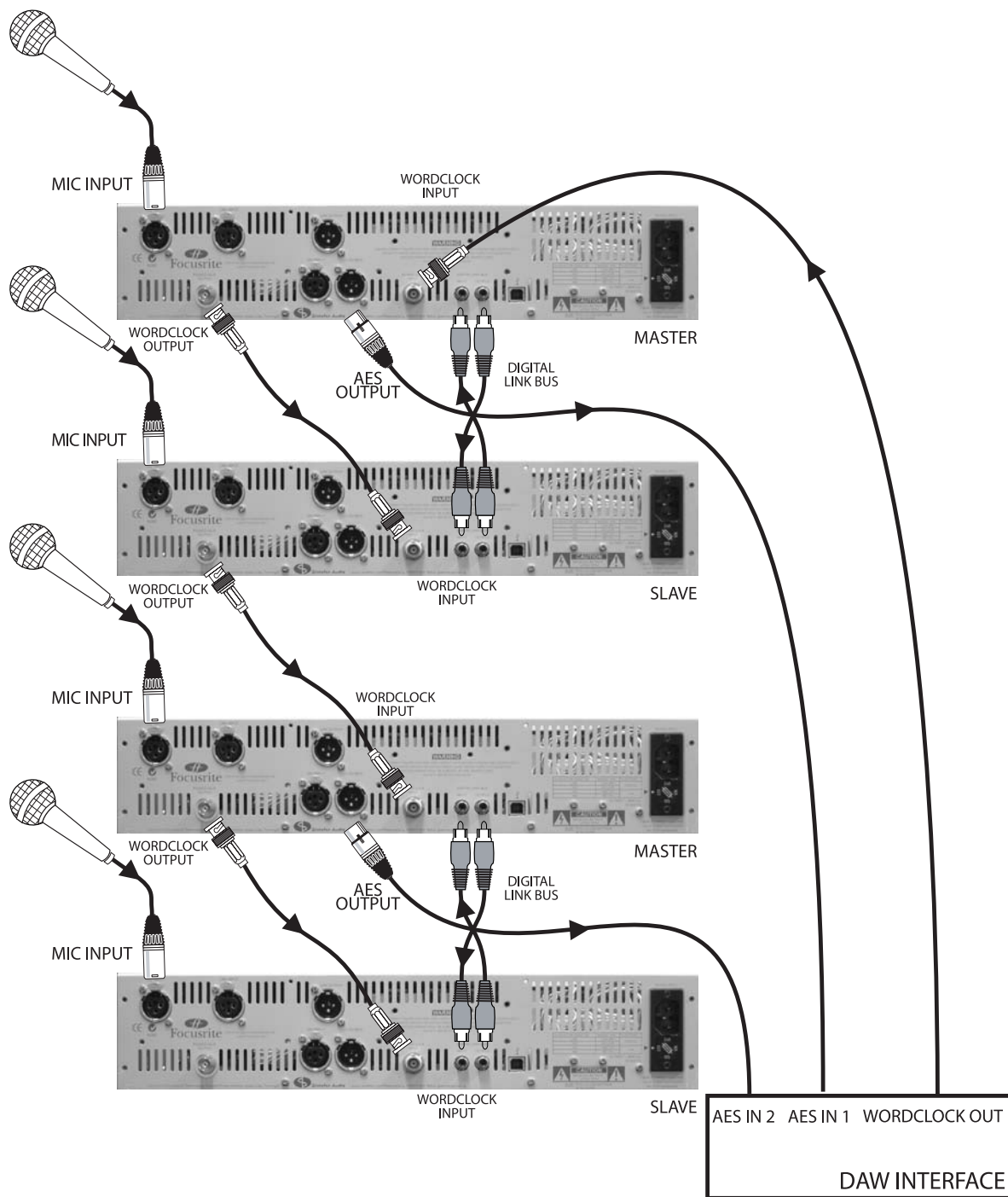AES INPUT

AES OUTPUT   DAW INTERFACE
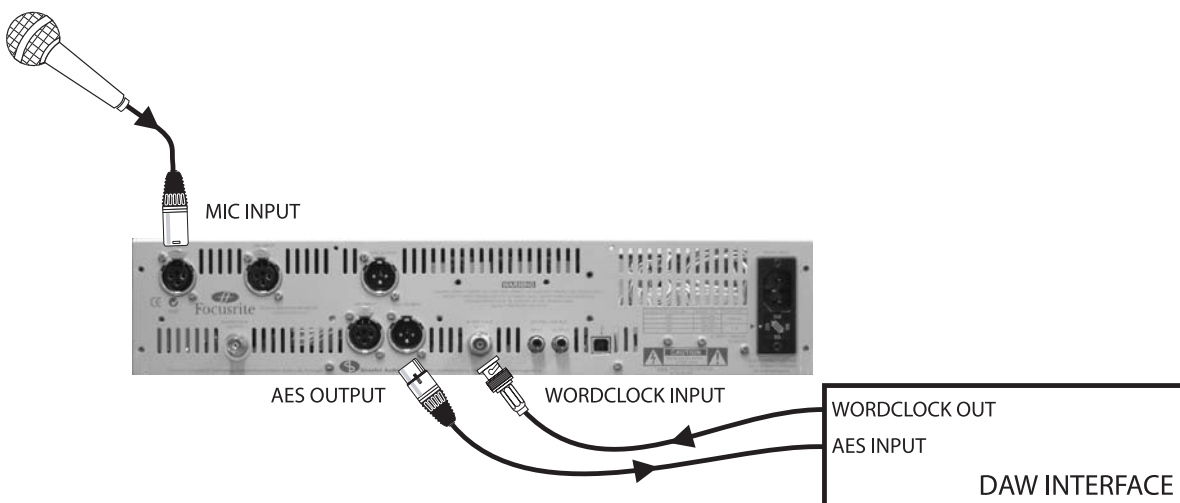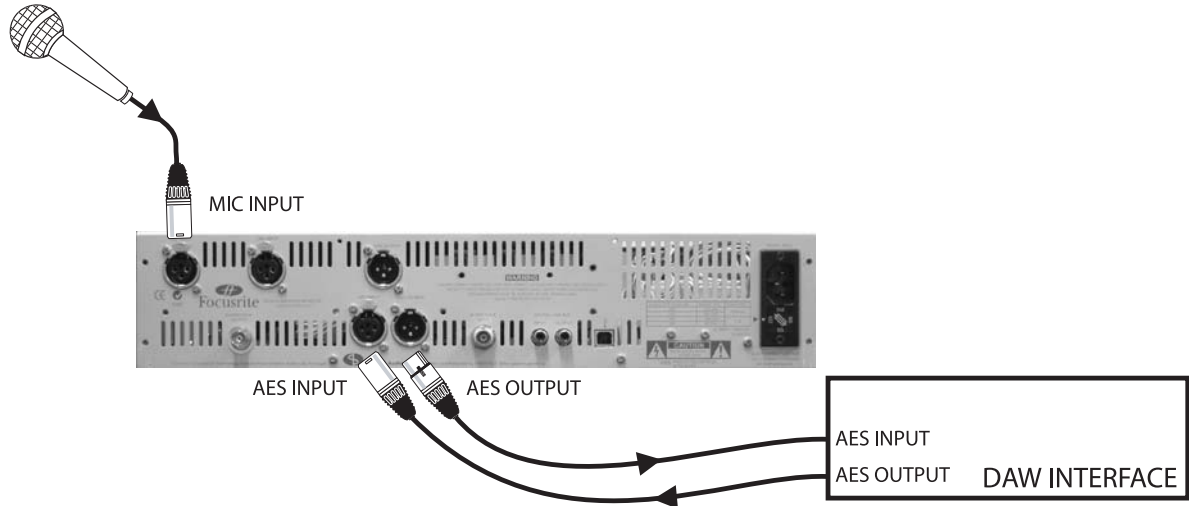
(Liquid Channel™ receives clock from AES connection.)

5. Stereo tracking, 44.1-96 kHz, Liquid Channel™ slaved to Wordclock

MIC INPUT

WORDCLOCK
INPUT

SLAVE

DIGITAL
LINK BUS

MIC INPUT

MASTER

WORDCLOCK
OUTPUT

WORDCLOCK
INPUT

AES
OUTPUT

WORDCLOCK OUT

AES INPUT

DAW INTERFACE
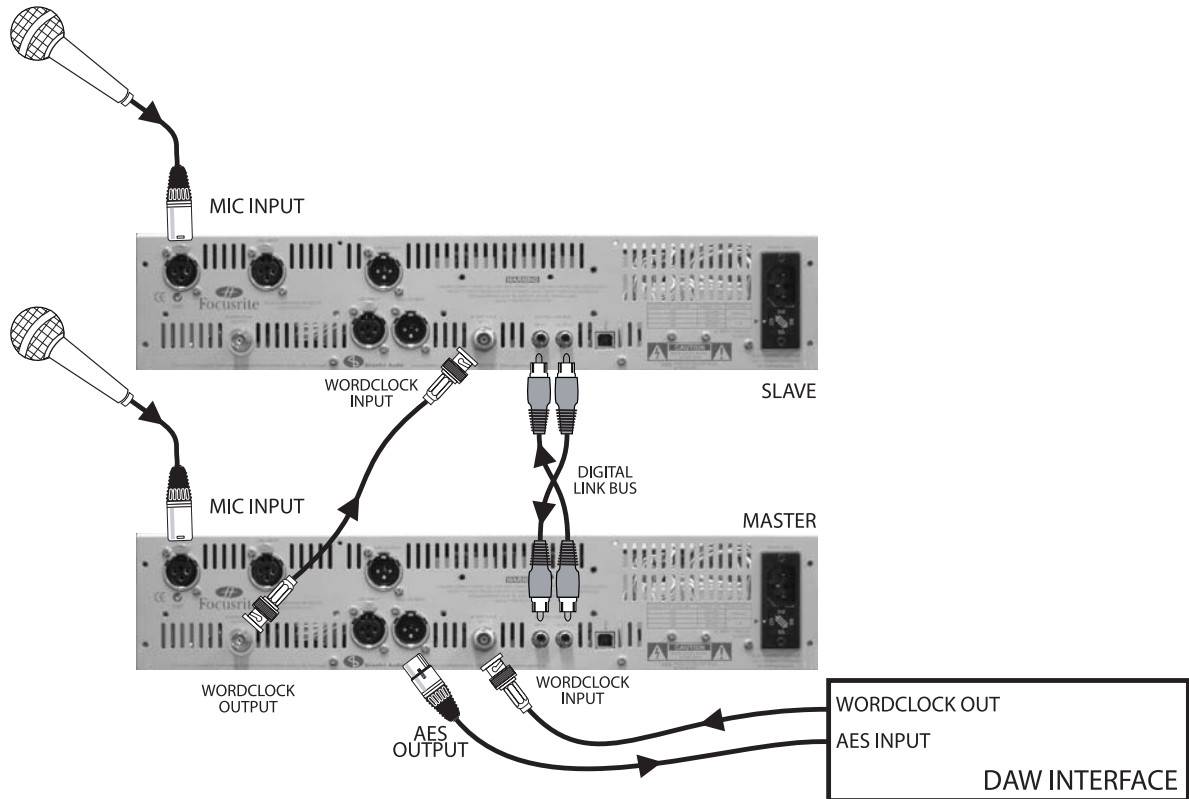
(See 'Linking Multiple Units' on page 10 for more information.)

6. Four-channel tracking, 44.1-96 kHz, Liquid Channel™ slaved to Wordclock

MIC INPUT

WORDCLOCK
INPUT

WORDCLOCK
OUTPUT

AES
OUTPUT

DIGITAL
LINK BUS

MASTER

MIC INPUT

WORDCLOCK
OUTPUT

WORDCLOCK
INPUT

SLAVE

MIC INPUT

WORDCLOCK
INPUT

MASTER

WORDCLOCK
OUTPUT

AES
OUTPUT

DIGITAL
LINK BUS

MIC INPUT

WORDCLOCK
INPUT

SLAVE

AES IN 2   AES IN 1   WORDCLOCK OUT

DAW INTERFACE

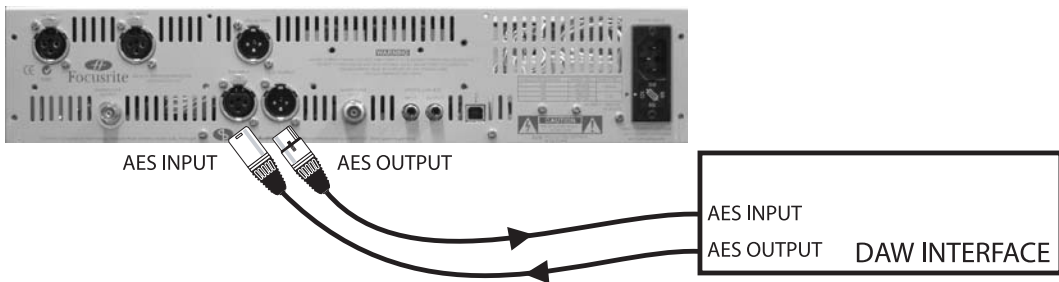(See 'Linking Multiple Units' on page 10 for more information.)

7. Stereo tracking 176.4 kHz/192 kHz, Liquid Channel™ slaved to AES clock
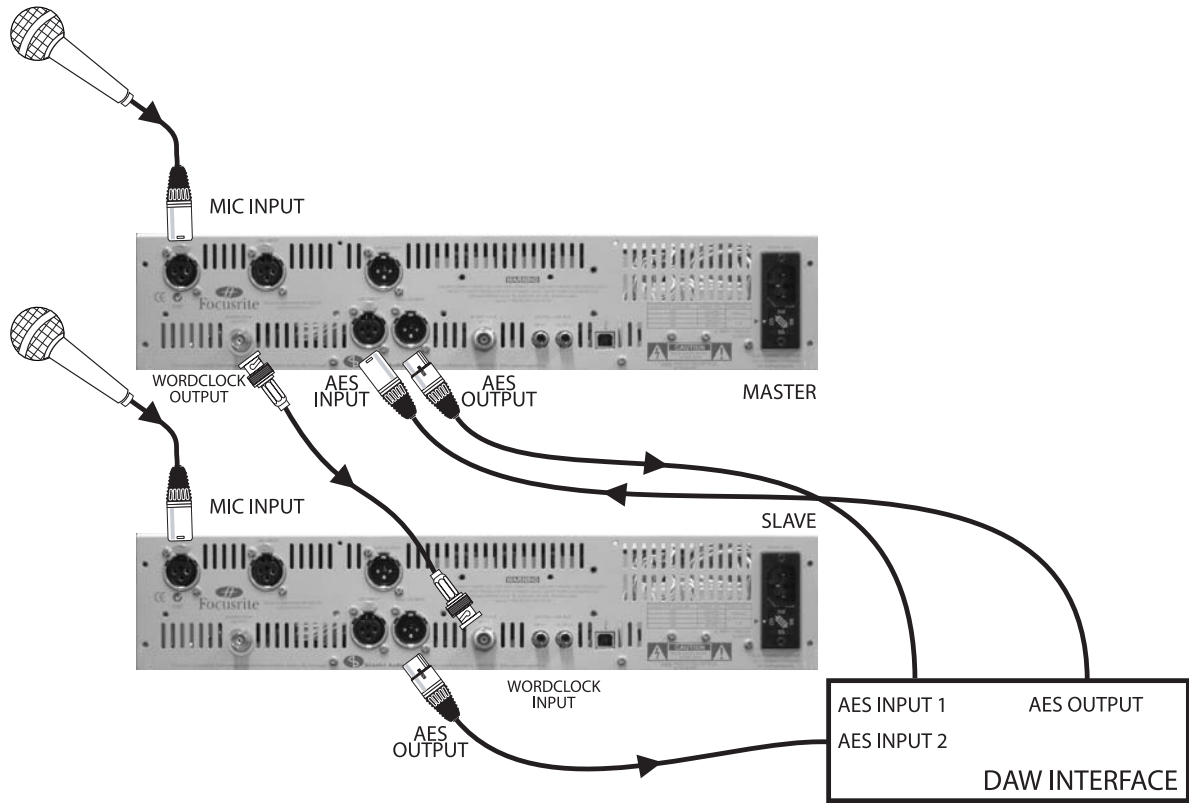
(For 176.4 kHz/192 kHz operation set DAW to send/receive in 'dual wire' mode. See 'Linking Multiple Units' on page 10 for more information.)

8. Digital in/out mono for digital insert, 44.1-192 kHz

(For 176.4 kHz/192 kHz operation set DAW to send/receive in 'dual wire' mode.)

9. Digital in/out stereo for digital insert, 44.1-96 kHz

(See 'Linking Multiple Units' on page 10 for more information.)

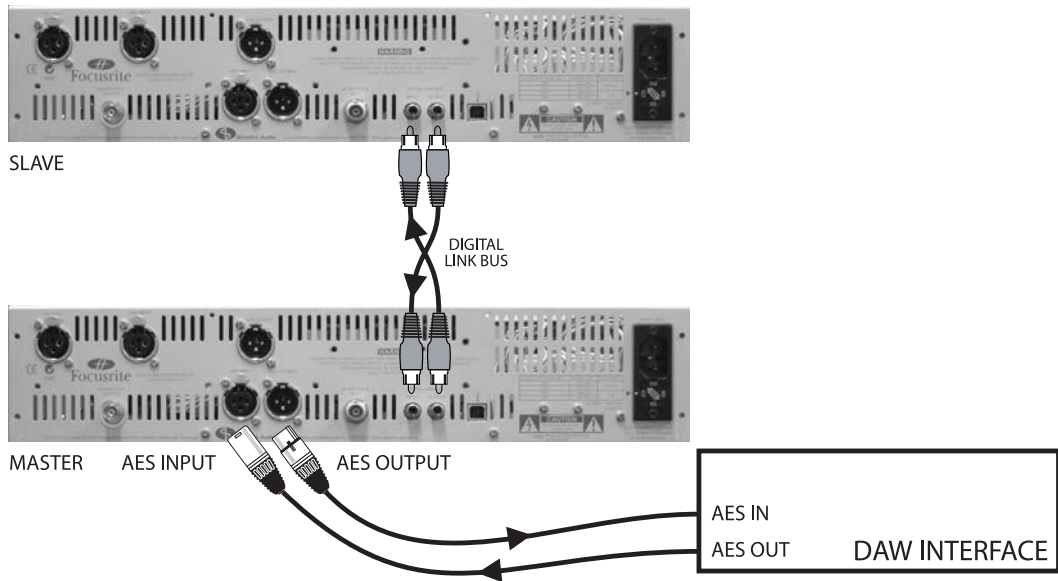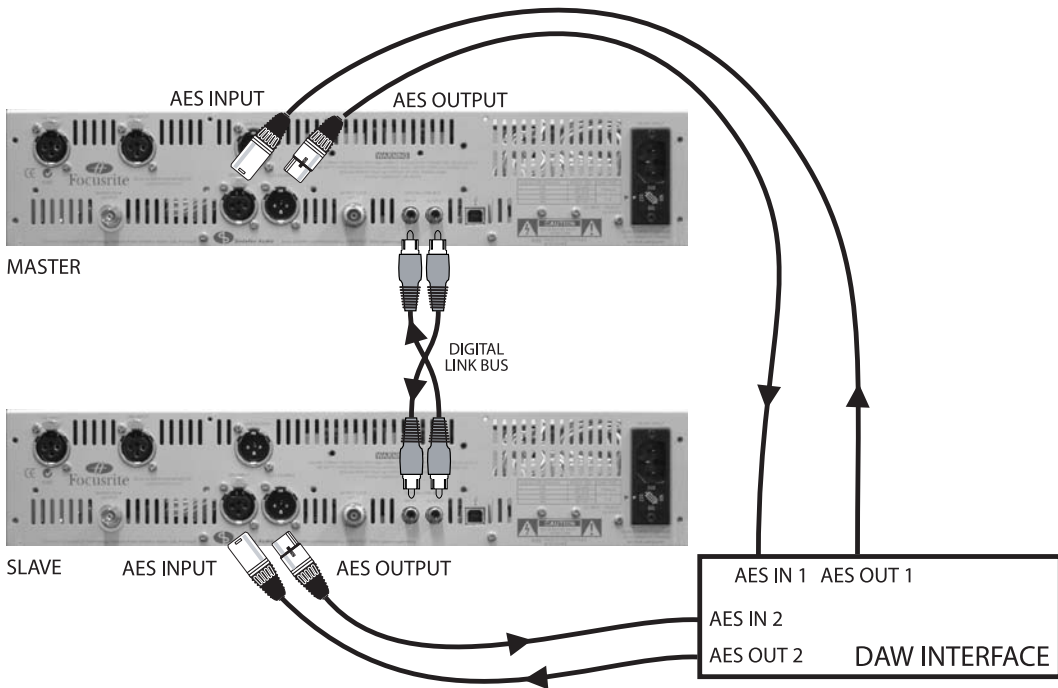## 10. Digital in/out stereo for digital insert, 176.4 kHz/192 kHz

(For 176.4 kHz/192 kHz operation set DAW to send/receive in 'dual wire' mode. See 'Linking Multiple Units' on page 10 for more information.)

Frequently Asked Questions

Q: What kind of technology is used by Focusrite in The Liquid Channel™?
A: The technology involves Dynamic Convolution techniques and a new liquid pre-amp, which together replicate vintage pre-amps and compressors.

Q: What is Dynamic Convolution?
A: The Convolution process has been defined as: 'The term given to the mathematical technique for determining a system output, given an input signal and a system impulse response.' What that means is if you know what is coming in to your system, and you can control your system's impulse response, you can define the system's output. In other words, you can replicate a compressor's sound, (or even a mic-pre's sound if you add a suitable analogue mic-pre hardware circuit.) Put another way, Focusrite have found a way to emulate the way in which any classic compressor or mic-pre ever made affects sound.

Q: How is this different from modelling and other 'simulations' we've seen and heard before?
A: Modelling looks at the way a device works and then relies on the generation of code to try to emulate the typical way in which a device would respond, usually in a certain limited set of situations. Convolution, on the other hand, records data about the way a device behaves and then replicates that. To use a simple analogy, it's like the difference between sampling and synthesis; if you want a REAL violin sound triggered from your keyboard you sample a violin, if you want a modelled sound which recalls the real sound of a violin, synthesis will generate a similar violin-like waveform.

Q: Why have modelling devices never succeeded in nailing the way a compressor or mic-pre responds?
A: The problem with a compressor is that it is a dynamic processor. That is, it is reacting to changes in input signal, and varies its response according to those changes. The problem with a mic-pre is similar- it is constantly interacting with whichever microphone is feeding signal to the pre, and it is the combination of pre and mic that characterises the sound. Dynamic convolution plus Liquid hardware enable these phenomenally complex interactive relationships to be replicated.

Q: What does the impulse response/ convolution process involve?
AA while back, Focusrite set about driving a huge set of impulse responses into the best collection of vintage and modern compressors and EQs ever assembled. The impulse response device they used for this process is called, with good reason, "The Replicator." This mysterious black box outputs an impulse (a very narrow (time-wise) voltage spike of amplitude which contains an infinite number of frequencies.) The impulse spike is sent to the device you wish to replicate. By measuring the output of the device itself, Focusrite's R and D team were then able to calculate what the device has done to the spike, hence calculate the aspects of the device that relate to frequency- and time-related parameters; frequency response, distortion.

One impulse of course, tells you how the processor will react to a particular input level, so it's necessary to sample the data of the device you wish to replicate firstly with a spike that drives the box into distortion, then with a fractionally lower amplitude spike, then lower again, all the way down to tiny spikes down into the noise floor. Once you have all this data recorded you can repeat any change in input level i.e. replicate the response to all input source types. Then all you have to do is apply convolution for every parameter setting combination, and you have the genetic blueprint of the device ready to go.

Q: So The Liquid Channel™ can emulate the sound of any compressor and any mic-pre ever made?
A: Yes, and more. Once you have The Liquid Channel™ you can mix and match the sound of your dream pres and compressors into program memories to set up and recall any combination you want.

Q: How on earth can one machine deal with the sheer weight of calculations required to produce ALL those responses for every group of parameter settings of ALL those mic-pres AND compressors?
A: Good question. It took the world's fastest audio-implemented SHARC chip technology to be able to crunch the mind-boggling numbers. That, and a huge number of patient hours replicating the sound of the classic units from audio history.

Q: So everything is pre-programmed?
A: No, all the convolution programming has already been done for you, but of course the impulse responses have to process the audio in real time inside The Liquid Channel™.

Q: Does The Liquid Channel™ allow me to replicate my own choices of pre and compressor?
A: No, that's pretty specialised stuff and best left to the Focusrite R and D team, but the unit ships with 40 mic-pres and 40 compressors ready to go in the box. You can make up your own combinations and store them in program memories, with or without EQ.

Q: Are there 'user memories' in addition to these?
A: Yes, the mic-pre and compressor 'building blocks' can be combined into 100 user-programmable program memories. All parameters including mic-pre gain, EQ and compressor settings etc. are also stored within the program memory.

Q: What if I want the sound of a specific mic-pre or compressor, which isn't one of the chosen devices?
A: Focusrite plan to make the sound of further pres and compressors available as downloads from a new dedicated Liquid Channel™ website – **www.ffliquid.com**. The free LiquidControl™ application software then allows you to load the sound of the pres and compressors that you crave into The Liquid Channel™ via the USB port on the rear panel. You can also save program memories and/or mic-pre and compressor replicas externally to your PC or Mac, and even edit the unit remotely via USB! All parameters will be editable on-screen remotely, even mic-pre gain settings for example, and can be transferred from session to session in e.g. a Pro Tools folder.

Q: What if I should overwrite my classic compressor by mistake?
A: No problem; you can always reload a back-up set from your PC or Mac, or download the original factory settings from **www.ffliquid.com**.

Q: How can convolution replicate the interaction of mic-pre and microphone?
A: On its own it can't. You need a separate analogue circuit to be present and to work with the convolution engine.

Q: So why do mic-pres present such a problem?
A: Mic-pres have always had to connect to the source microphone, but it's an interactive system that isn't 100% efficient. Mic amps have been designed since the 1920s to suit a wide variety of different types of mics – passive carbon dynamics, then coil-based designs, then valve amplifiers, large diaphragms, phantom powered condensers etc. Hence, different vintages and types of mic amp will vary dramatically in terms of the way that their input has been designed. For example, the range of electronic/transformer front ends that have been used over the years exhibit a wide-ranging set of impedances, and this is why an analogue front end needs to be included. If a specific mic is not being loaded by the analogue circuit just as it was by an original vintage device, then the sound from that microphone will be different.

Q: So there's no real mic-pre standard?
A: Exactly. Take a transformer for instance. It has two coils of wire, the first coil generates a magnetic field, and this then passes into the second coil – which in itself is not a fixed transfer mechanism, there's a lot of variation in transformers. What impedance appears at the input of the pre is also a key factor – when you connect a mic it has an output impedance of its own. The two sides (mic and pre) react, and frequency-related level can vary wildly as a result. Capacitances also interact as both mic and transformer have capacitances that vary, so HF roll off may occur for example, or you may get an HF peak (the famous Focusrite 'airiness' typified by the ISA range for example). Older mics designed for broadcast applications often roll off at e.g. 12 kHz, since before 1970 few people cared about HF matters. (Designers used to just roll off at 12 kHz to filter out problems above this threshold.)

Q: So how do you design one mic-pre circuit that can reproduce all the variables within this wide range?
A: The only way to accommodate the full range of different designs is to allow huge flexibility in the resistance and capacitance parameters in a custom transformer designed specifically for that flexibility. Hence, The Liquid Channel™ physically changes analogue circuitry as well as using dynamic convolution technology to create mic-pre replicas.

Q: What about electronic or tube mic pres that do not include a transformer?
A: The Liquid Channel's transformer is auto-switched out when an electronic transformerless mic-pre is chosen by the user (this is indicated on the front panel). Focusrite has built in the variations required to reproduce the vagaries of a range of electronic mic-pres. The capacitance and resistance are then varied in the circuit, and Dynamic Convolution technology is used to replicate the full range of electronic pres. Tube replication is also covered 100% - this is taken care of by the Dynamic Convolution process. Whatever artefacts were present in a classic vintage tube piece are also present in The Liquid Channel.

Q: So this is really a hybrid technology that allows total control over the key aspects of the sound of analogue pres and compressors?
A: Yes, the sound of every opto, and every VCA compressor, every transformer-balanced, electronically balanced (including tube pres) can be emulated, because each device's every response has been precisely mapped.

Q: Why is it necessary to have additional circuitry for the mic-pre and not the compressor? Surely, if the convolution DSP is as thorough as you say, there should be no need for further processing.
A: As mentioned above, the interaction between the individual microphone and pre-amp is a key factor in the sound of the pre as a whole. (The ISA 428 and 430 MK II have switchable impedance values that the user can implement to specifically tailor the character and response of the device for this very reason.) By

including a 'Liquid' pre-amp circuit containing a flexible signal path (transformer or electronic) and variable impedance value, The Liquid Channel can mimic that of the classic mic-pre to ensure that the interaction with the microphone is close to the original. This issue isn't something that affects a compressor but the DSP processing required is nonetheless immense. The user's ability to affect the threshold and ratio of the compressor means that there are additional responses needed for the side-chain to account for the numerous variations in character (types of 'knee', presence of 'over-compression' etc.)

Q: Is the transformer the traditional Focusrite Lundahl™ transformer? Or another famous brand like a Jensen™?
A: No, it's a brand new custom precision-wound Focusrite 'Liquid' transformer, designed and built in the UK by Focusrite's R and D department to be extremely flexible: transparent or coloured as required.

Q: If I connect different mics to the plethora of mic-pres that The Liquid Channel™ offers, will each of the mics sound different?
A: Of course. The results will be the same as if you were connecting your collection of favourite mics to the various pres. Of course, if you're just modelling in software this is simply impossible - how can a particular mic interact in its distinctive way in real time with a particular mic-pre if the mic-pre is in fact absent?

Q: What about interaction between the mic-pres and compressors?
A: The Liquid Channel™'s dynamic convolution DSP is separate for both the pre-amp and compressor. That is to say, the device acts exactly as the separate units would, with the same signal leaving the mic-pre and entering the compressor. So, interaction is identical to the hardware equivalent without the extra cost/size and weight/plugging in required, but with all the reliability of a first class digital audio device.

Q: What about EQ?
A: Focusrite's R and D department in England decided that a truly Liquid Channel™ strip should also include a flexible EQ. So they created a brand new digital EQ, modelled on the curves of the fabled ISA 110. This EQ is a single Focusrite British EQ design that is the perfect complement to the range of mic-pres and compressors available.

Q: Can I put the EQ in front of the compressor?
A: Yes you can. You can also drive the compressor from the parametric mid-range section to use The Liquid Channel™ as a de-esser – more liquid than liquid! The EQ is editable, programmable directly from the front panel, and includes high and low shelving and

parametric mid-range across a huge range of frequencies.

Q: I understand the unit is a recording channel, i.e. mono. But what if I want to record in stereo – can I chain two units together?
A: Yes, all you need is a standard RCA (phono) cable to transmit the data between units. The pres, compressors and EQ will then all function as a perfect stereo pair, even if you are operating from the LiquidControl™ software application. With two units linked together, The Liquid Channel™ also becomes ideal for stereo mix-down and mastering applications.

Q: Is there any way to run a super-short signal path from the mic-pre to the output?
A: Yes. By not selecting Comp or EQ in, the signal will pass through the analogue front-end, A/D and mic-pre section of the DSP, then straight through to the AES digital output (or through the D/A to the analogue output).

Q: What if I record a great vocal performance only to find out later that I drove the pre too hard and caused clipping?
A: The Liquid Channel™ includes a feature called Session Saver, which will automatically prevent this if you enable it. The Liquid Channel™ notices that digital overs are in danger of occurring, and if it sees significant danger it turns down the analogue pre-amp gain – a 1 dB reduction for any level above 0 dBFS.

Q: How flexible is my record path?
A: Very. You can record in the following ways:

Analogue to digital: (mic connects to mic-pre via the balanced XLR connector, through the A/D converter, through the pre and compressor convolution processors, and then exits via the AES D/A which is included as standard.) The D/A can be used to monitor post-DSP as a super-low latency feed to bypass DAW delays if you wish.

Digital to digital: The digital input can be re-routed into the front end, feeding into the mic-pre and/or compressor convolution engines as desired.

Digital to analogue: as Digital to digital above, but the balanced XLR analogue outputs are used.

Analogue to analogue: as Analogue to digital above, but the balanced XLR analogue outputs are used.

Q: What are the A/D and D/A specs?
A: The format is the world professional standard AES-EBU, and all sample rates from 44.1 kHz – 192 kHz are supported, in and out, as standard. Hence you can digitally input a pre-recorded instrument or line level signal into the pre or direct to the compressor.

Q: What A/D encoders are featured?
A: The latest 192 kHz/120 dB spec AKM™ 5394s.

Q: I notice there's an extra parameter called 'Harmonics'. What does it do?
A: Warm is good, everybody tells us so. Of course The Liquid Channel™ will emulate the classic analogue warmth that is present in vintage pieces, along with all the other sonic artefacts. But imagine if your favourite mic-pre was for some reason warmer than most other pres of that type? (Since many vintage pieces were hand-built, there is often some component tolerance variation for example.) This parameter means it's actually possible to add in extra warmth to 'tune in' The Liquid Channel™ to your 'special' unit…

Q: What if I want to compress pre-A/D to get that "driven hard" sound?
A: There's no need; you can now optimize the level using the gain encoder, then add in the "high gain" warmth via the Harmonics pot as described above. Certain classic pres that benefit from being driven hard will also be replicated 'driven hard' at source.

Q: What is the latency of the unit?
A: Worst case scenario (analogue in, analogue out, all sections in circuit, 44.1 kHz) better than 4 milliseconds. At 96 kHz, analogue in, analogue out, 1.6 milliseconds. In other words, extremely small. The latency is the same for all replicas, so can easily be corrected in e.g. Pro Tools™.

Q: Does the fact that responses are measured in increments mean that the ear hears differences between the replica and the original?
A: Not at all. Firstly, the response measurement accounts for non-linear behaviour of the vintage units by sending impulse trains – literally single impulses at decreasing levels – from peak excitation to approximate noise floor level, separated by time divisions to allow for system reset. This means that the dynamic nature of the devices being replicated is taken on board unlike with most other devices that just assume linearity for ease of design.

Secondly, these 'response filters' are then applied proportionally to every sample of audio. Hence, at 96 kHz, the signal is being processed with 96,000 different dynamic responses each second! It's true that this method isn't 100% continuous, but the density of data in such a thorough system is such that any tiny errors present are not in any way audible.

Q: What about any extra knobs on the vintage models other than those present on The Liquid Channel™? How can The Liquid Channel™ replicate without them?
A: The only controllable parameters available on classic mic-pres/compressors are those featured on The Liquid Channel™ – threshold, ratio, attack, release, gain. (If anything, some models have fewer controls, with some vintage compressors just offering gain controls for example. Focusrite do also plan to have more than one replica for certain devices that may require them.) In addition, The Liquid Channel™ extends the user's control further by allowing an extra dial to change the percentage of the mic-pre's second, third and fifth order harmonic distortion, and hence the overall warmth, of the mic-pre. This extra control allows the user to account for any variation between individual vintage pres of a particular type. The Liquid Channel™ also features a Setup menu to configure the unit's parameters as per the original unit ('As Original'), or to allow The Liquid Channel™'s parameters to be fully edited ('Free'), even if this was not possible on the original unit.

Q: What components change on the analogue PCB when switching pres?
A: The transformer and the matrix of relays, which switch resistors/capacitors.

Q: How is the transformer manipulated? Are there actual variations between the primaries and secondaries?
A: Transformer variation is actually partly taken care of by the resistor/caps variation. The transformer is huge and is big enough not to load down the circuit – hence the transformer (1 to 1 type) is utterly transparent but can be configured not to be if the original device to be replicated requires it to be coloured in the analogue domain. Primary/secondary variation specifically is largely taken care of by convolution.

Q: Do the replicas sound worse at 44.1 kHz than at 192 kHz?
A: No. Sample rate does not affect replication quality; we are always 32-bit floating, and e.g. A/D quality is far more affected by lower/higher sample rates. N.B. We do not sample rate convert because we don't need to as we have all different sample rates already stored on the SHARC chip. We have low speed (44.1 kHz) and high speed replica data High is actually only at 96 kHz – nothing higher is required, since 96 kHz is high enough when applied to replication, and any further improvements are way out of even the potentially significant psycho-acoustic realm. This however is not true in the world of A/Ds for example where 192 kHz vs. 96 kHz is an audible difference.

Q: Why is there no tube? If the transformer is required in the analogue circuit how come a tube is not also required?
A: The dynamic convolution process handles the tube characteristics. However, there are additional benefits to using transformers over and above their warmth: better CMRR and the transformer's direct impact on the connected mic for example. This latter point is why we need a transformer in circuit to replicate the mic-pre - the interactivity with the mic is key for a transformer in a way in which it is not key for a tube.

Q: Are there any audible (distortion) differences between applying a "Hot" replica versus using a regular level replica and then adding 2nd/3rd/5th order distortion via the dedicated Harmonics encoder afterwards?

A: Yes. At full gain on an old mic-pre you may have 60 dB of gain at 1 kHz but only 40 dB at 10kHz. Distortion is affected in a similar way. Third order harmonic distortion is reduced at low gain, so this may be present when using a Hot replica, but not present if you just add second order distortion after the fact. This is one reason why we include Hot settings. There are also differences re: different loadings on the transformer occurring when driving high gain in at the front end, hence mic interaction changes occur. For example, HF roll-off etc may change.

Q: Why do we only use one set of impulses when creating the replicas? Surely we need to replicate all combinations of threshold, ratio parameters etc?

A: Dynamic convolution, using a single set of impulses, emulates the sound of the signal path at all frequencies/levels. However, the FF Liquid process is actually more complicated than this. After replicating using convolution, we then measure the compressor curve at different ratio/threshold positions. Then we measure the attack and release characteristics, as well as the RMS vs. Peak detection of the side-chain signal in order to see if it discriminating more towards peak or RMS. (If you put e.g. a drum kit thru a peak-detecting compressor such as a Focusrite) the compressor side-chain will follow the curve of the signal that's coming in and compress in a manner that follows that curve. An RMS compressor will compress the signal against the average level and ignore the peaks. Hence a drum kit is smoothly compressed by an FF piece, but an RMS unit will result in attacky/toppy compression with many transients that have more energy/are less smoothly compressed.

We then glue all this info together to get a compressor that acts so that when the signal enters it gets rectified, goes into Peak/RMS depending on what the original vintage unit requires, then enters the curve circuit to recreate the original side-chain, then applies that to the convolution data: This cannot be done at the impulse stage since the impulse maths will then look at the amplifier rather than the compressor.

Q: Can you route the digital input into the mic-pres?

A: Yes, you can route the digital input to any of the mic-pres, or only via the transformer, or directly to the compressor in the digital domain.

Specifications

Converter Performance

Sample Rates
> 44.1, 48, 88.2, 96, 176.4 and 192 kHz

Bit Depth
> 24-bit

A/D

Signal to Noise Ratio
> 120dB measured with 20 Hz/22 kHz bandpass A-weighted filter

Frequency Response
> ±0.05 dB between 20 Hz – 20 kHz

Maximum input level
> +22 dBu

THD+N
> 0.00035% (-109 dB)

D/A

Dynamic Range
> 116 dB measured with 20 Hz/22 kHz bandpass A-weighted filter

Frequency Response
> ±0.05 dB between 20 Hz – 20 kHz

Maximum output level
> +22 dBu

THD+N
> 0.0007% (-103 dB)

Jitter

Internal Clock
> <20 pico-seconds

AES Output
> <200 pico-seconds

External Clock:
> <1 nano-seconds

Analogue and digital paths

Mic Pre

Gain Range
> +6 dB to +80 dB, switched in 1 dB steps

Frequency Response
> Variable, set by pre-amp replica chosen

THD+N at analogue out
> 0.001% measured with a +4 dBu 1 kHz input signal with 20 Hz/22 kHz bandpass filter

THD+N at AES out
> 0.0005% measured with a +4 dBu 1 kHz input signal with 20 Hz/22 kHz bandpass filter

Mic Noise
> EIN = -126 dB measured at 80 dB of gain with 150 Ω source impedance and 20 Hz/22 kHz bandpass filter

Noise Analogue Out
> -92 dBu measured at +6 dB gain with 20 Hz/22 kHz bandpass A-weighted filter

Noise AES Digital Out
> -119 dBFS measured at +6 dB gain with 20 Hz/22 kHz bandpass A-weighted filter

Maximum Input Level
> +16 dBu

Input Impedance
> Variable, set by pre-amp replica chosen

CMRR
> Transformer: 123 dB @ 60 dB of gain
> Electronic: 102 dB @ 60 dB of gain

Line Input

Gain Range
> +10 dB to –10 dB, switched in 1 dB steps

Frequency Response
> 0 dB ±0.1 dB between 20 Hz and 20 kHz

THD+N at analogue out
> 0.001% measured with a +18 dBu 1 kHz input signal with 20 Hz/22 kHz bandpass filter

THD+N at AES out
> 0.0004% measured with a +18 dBu 1 kHz input signal with 20 Hz/22 kHz bandpass filter

Noise Analogue Out
> -92 dBu measured at 0 dB gain with 20 Hz/22 kHz bandpass A-weighted filter

Noise AES Digital Out
> -120 dBFS measured at 0 dB gain with 20 Hz/22 kHz bandpass A-weighted filter

Maximum Input Level
> +22 dBu

High Pass Filter

Roll-off frequency
> Switchable between 75 Hz and 120 Hz, frequency measured at –6 dB down point. 12 dB per octave roll-off

Harmonics

Distortion Range:
> 0 to 15 where maximum (15) = 10% of 2nd-, 20% of 3rd- and 10% of 5th-order at 0dBfs (level-dependant distortion)

Compressor

In 'As Original' mode the parameter ranges will be the same as on the original unit being replicated. In 'Free' mode the parameter ranges are as follows:

Threshold Range
> -40 dB to 20 dB switched in 1 dB steps

Ratio Range
> 1:1 to Limit

Attack Range
> 0.1 mS to 2.5 S

Release Range
> 0.1 mS to 2.5 S

Makeup Gain
> -20 dB to +20 dB switched in 0.5 dB steps

EQ

High Shelf

Frequency Range
> 200 Hz to 20 kHz

Gain
> +/-18 dB

Mid Band

Frequency Range
100 Hz to 10 kHz
Gain
+/-18 dB
Q
Variable between 0.8 and 2.5

Low Shelf

Frequency Range
10 Hz to 1 kHz
Gain
+/-18 dB

Rear Panel Connections

Mic Input
XLR female
Line Input
XLR female
Analogue Output
XLR male
AES Digital Input
XLR female
AES Digital Output
XLR male
Wordclock Input
BNC, 75Ω input impedance
Wordclock Output
BNC, 75Ω input impedance
Digital Link Bus Input
RCA connector
Digital Link Bus Output
RCA connector

Weight
8.6 kg

Dimensions
484 mm (W) x 85 mm (H) x 270 mm (D)
2U rackmount

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