XLogic Alpha Channel

Putting it to work—Tips & Tricks

www.solid-state-logic.com

Solid State Logic
SOUND || VISION

XLogic Alpha Channel

Putting it to work—Tips & Tricks

Contents

Introduction

This guide is intended to show you the various function of the Alpha Channel, and some of their more common applications.

The Alpha Channel offers far more audio features and routing possibilities than you would first imagine, and by reading this you will be made aware of these which will help you to get the most out of the channel in a creative and intuitive manner.

Monitoring

A major consideration when recording into a host based DAW (Cubase, Logic, Sonar etc) is that there is always a latency associated with a signal passing through the software.

In a DAW setup it is desirable for the musician and engineer (who may be the same person!) to monitor the signal before it reaches the computer audio interface. If you wish to monitor the signal after it has passed through the hardware and software then you will need to run the system with a very low buffer size to avoid hearing a delayed signal; this is not always possible.

Analogue recording and monitoring

The Alpha channel provides a feature which allows one output for the recording send and another for monitoring, meaning zero latency monitoring can be fully realised. This is achieved by simultaneously using the Line Output and Insert Send jacks.

Whilst recording a vocalist wishes to hear the vocals with some added reverb and some EQ. The engineer, however is not keen to record with these processes on the signal, but prefers to record the unprocessed, dry signal coming from the Alpha channel. In this situation the engineer wants to use his own A/D converters, so for now the digital connections of the Alpha Channel will be ignored.

The diagram below shows a typical set up for this situation.

Analogue recording and monitoring cont'd...

- The INSERT IN button should be down to enable the insert send.

- The SUM button should be up as there is nothing connected to the insert return in this situation.

- The POST EQ button should be up as we wish to record the non-EQ'ed signal. This means the signal appearing at the Insert Send jack will be immediately after the pre-amplifier and before the filter, EQ and Limiter.

- The filters can be used to taste to cut out incidental microphone noise and this will be passed to the line output and hence the singers monitoring.

- The EQ IN button should be depressed so the vocals can be EQ'ed to suit the singer. The limiter can also be engaged to catch the peaks in the vocalists monitor path.

A major consideration when recording into a host based DAW (Cubase, Logic, Sonar etc) is that there is always a latency associated with a signal passing through the software.

This method allows you to extract the cleanest possible signal into your chosen recorder whilst still utilising the Alpha Channels EQ,filter and limiter functions in the monitor path.

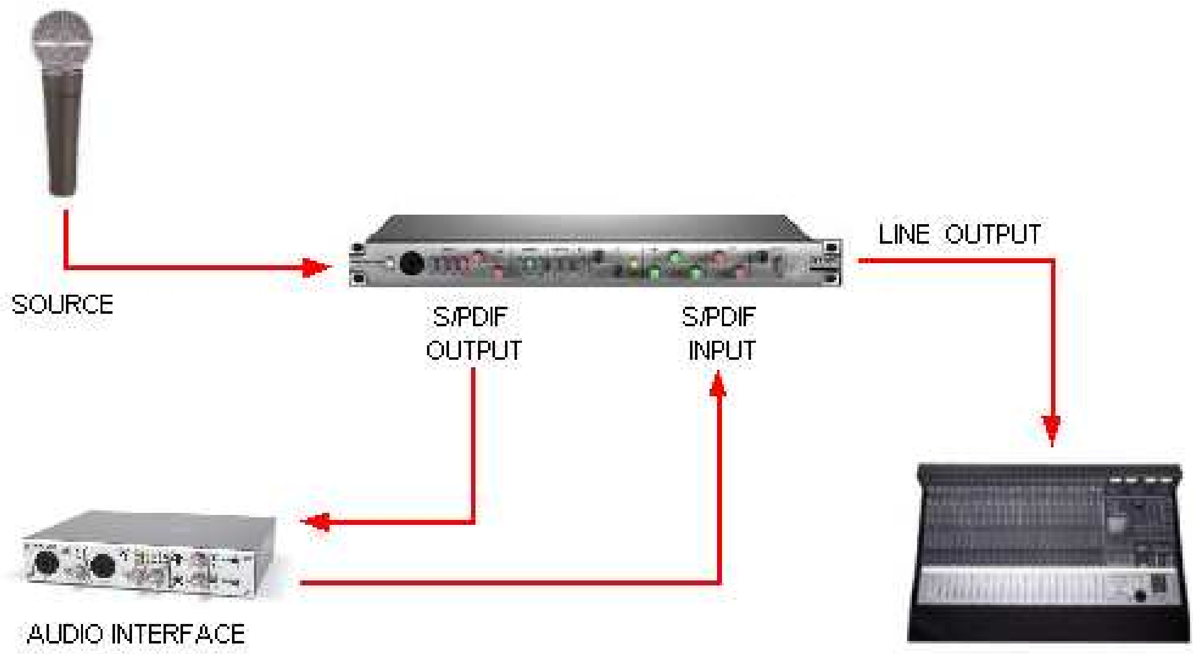
In a DAW setup it is desirable for the musician and engineer (who may be the same person!) to monitor the signal before it reaches the computer audio interface. If you wish to monitor the signal after it has passed through the hardware and software then you will need to run the system with a very low buffer size to avoid hearing a delayed signal; this is not always possible.

Digital recording with analogue monitoring

If you prefer to use the digital output of the Alpha Channel to interface with your DAW but also need the analogue monitoring capabilities..

Whilst recording a vocalist wishes to hear the vocals with some added reverb and some EQ. The engineer, however is not keen to record with these processes on the signal, but prefers to record the unprocessed, dry signal coming from the Alpha channel. In this situation the engineer wants to use his own A/D converters, so for now the digital connections of the Alpha Channel will be ignored.

The diagram below shows a typical set up for this situation.



The left hand side of the S/PDIF output is identical to the signal appearing at the analogue output.

Digital clocking considerations in the studio

Please note that in order to use the S/PDIF output reliably, all digital devices in your studio should be synchronised from the same clock source. The Alpha Channel will presume it is the master sync source unless a S/PDIF signal is connected to it's S/PDIF input. If presented with a S/PDIF signal in this manner then the Alpha Channel will automatically slave to this clock source.

- If nothing is connected to the S/PDIF input then the Alpha Channel as the master sync source will output a clock of 44.1kHz. In this case all other devices should be set to sync from their digital inputs and hence the Alpha Channel.

If the Alpha Channel is presented with a sync reference of a higher sample rate than this then it will not output a valid S/PDIF signal or clock. If you need to work at these higher sample rates then you would need to use the analogue outputs of the Alpha Channel only.

Using the S/PDIF output to record two signals

The Alpha Channel utilises the stereo S/PDIF output in a very useful way. The left hand side of the S/PDIF output will carry the same signal (after A/D conversion) as appears at the Analogue output. This means that if you select the left hand side of the S/PDIF input of the audio interface in your DAW then you will record exactly what you are hearing.
However, if you select the right input of your audio interface, the signal that comes from the Alpha Channel is taken from the pre-insert, post filters point. This means it will have no EQ, insert or limiter effects. It is also 12db below the left hand side because that allows enough headroom for the signal to avoid clipping the converter.

By creating two mono audio tracks and selecting the left S/PDIF input for one and the right input for the other you can record both these signals simultaneously.

The decision over which track to use can then be made later on in the recording process.

By taking the right hand side of the S/DPIF signal you are provided with a slightly cleaner, unprocessed version of the source signal.

This signal is 12db below the left hand side signal, but when recording at 24bit this is an acceptable recording level with enough headroom for peaks.

Recording an Acoustic Source

Please refer to the first few sections of this tutorial to show the various monitoring configurations offered by the alpha channel.

We will now assume that the full features of the channel are required to record an acoustic source such as vocals or acoustic guitar digitally into a DAW.
Plug the microphone into the front panel input preferably with Alpha Channel off

Engage the 48V button to provide phantom power to the mic if needed

Move the gain control to the appropriate position; the output meter should show about -6db on the loudest parts of the signal. Engage the PAD button if you require more headroom which also doubles as an input clip which turns red when clipping is occurring. This also indicates when the VHD is producing a noticable harmonic effect within the pre-amp, so may be desired in some cases. Please see the section on using VHD for more information on this.

The configuration of the INSERT section will depend on if you wish to utilise the send/return loop of the Alpha Channel with a compressor for example. This is covered elsewhere in the tutorial.

A good starting point for the filter settings would be only the 80Hz button in. If you feel this misses some useful information very low down in the signal then use only the 40Hz or no filters in the case of a bass instrument.

The EQ section could be used to remove any resonances found in the instrument and the room or to generally shape the sound to taste. The LF band could be used to find any muddyness below 600Hz with the BELL button depressed.
The MF band could be used to either boost or cut any frequencies in the all important 300Hz to 5kHz range.
Because the HF band goes all the way up to 22kHz, this could be used to add some 'air' to the signal at or slightly below this frequency.

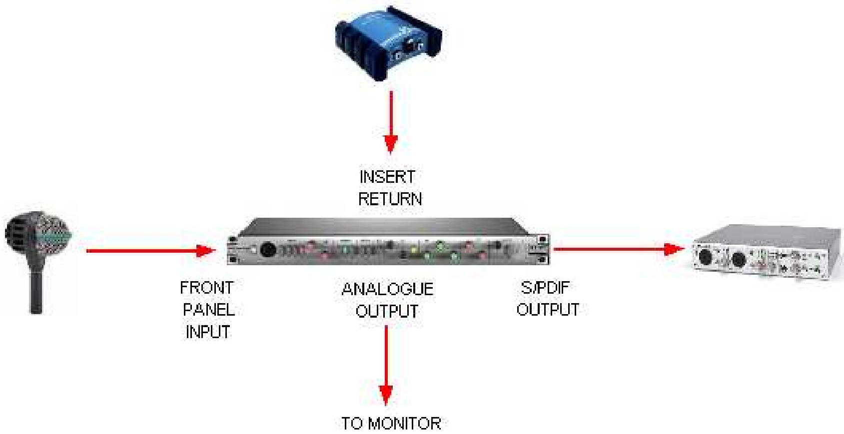
The Lite Limit can be engaged to prevent the A/D converters from clipping. Use the OUTPUT knob to drive the limiter to the required degree.

The signal(s) can be recorded from the S/PDIF output and monitored from the analogue output.
Please ensure that all clocking arrangements are correct as mentioned above.

Recording Amplified Instruments

When recording an amped instrument (electric guitar, bass guitar, keyboards etc) it can be a good idea to take both a mic'ed signal from the cabinet and a DI signal directly from the instrument or amplifier.

The Alpha Channel is capable of receiving these two signals and summing them in various ways to provide a mono output source.



Lets deal with the mic'ed signal first:

- The microphone is plugged into the front panel input, possibly with the PAD button engaged to accommodate the loud signal that is often produced by an amplifier.

- The gain control should be used to bring the input up to a decent level. It may be worth driving the pre-amp at this stage in order to bring in some of the VHD colouration. Please the section on VHD in this tutorial for more information on this.

- If recording bass guitar it may be worth leaving the filter section out of circuit as you will want to capture the very low frequencies of the instrument. If recording electric guitar or keyboard then a little experimentation may be necessary in using the filters to remove any unwanted noise yet retain the all the musical aspects of the signal.

Adding the DI signal:

- The direct signal either from a DI box or from the amplifier itself is input to the INSERT RETURN jack. Pressing the SUM button will add the mic and DI signal together.

- The PHASE button can be used to alter the relative phase between the two signals. By 'flipping' the phase you will notice a drastic change in tone and you may find this negates the need to EQ the signal. It may also be worth experimenting with the distance of the mic from the speaker grill at this point. This again will alter the relative phase of the two signals and consequently it's tone.

- The two signals can now be balanced by adjusting either the pre-amp input gain or the output of the amplifier if available.

EQ'ing the signal(s)

You have the choice to EQ either the mic signal or both the DI and Mic.

- If the POST EQ button is depressed then the DI'ed signal will not be affected by the EQ.

- The EQ could well be used to add a little bottom end to the signal if recording bass guitar. The fully parametric mid- band can be used to pinpoint and pull out any offending frequencies. The HF band would be ideal in a situation where a significant amount of noise is present in the signal which can sometimes be the case with guitar and bass amps.

Lite Limit

Finally it may be worth experimenting with the Lite Limit function. This appears right at the end of the signal chain, and it's affect will be present at both the left S/PDIF signal and the analogue output.

It's intention is to provide a dynamic safety net before the A/D converters to prevent overload. Be careful to check that the limiter is not infringing on the dynamic range of the signal too much. Due to its transparent nature, the Lite Limiter can be hit quite hard before any detrimental effects are heard.

The amount of limiting action is determined by the level of signal going into it. This means there is no single control to govern the action, but it will be a combination of pre-amp gain, insert section, EQ and output gain.

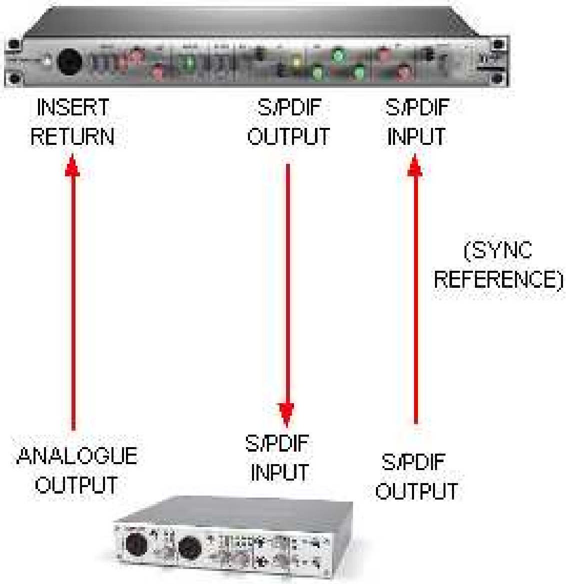
On most digital systems you tend to avoid the red 'clip' LED because this usually indicates that the converters are clipping – usually not a very nice sound! On the Alpha Channel however, hitting the clip point indicates that the Lite Limit is working and digital clipping is being avoided. In this situation the output meter will hit the top and the Lite Limit button will glow red.

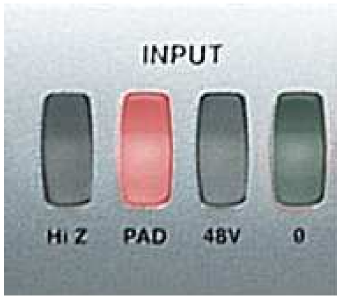
Processing a pre-recorded signal through the Alpha Channel

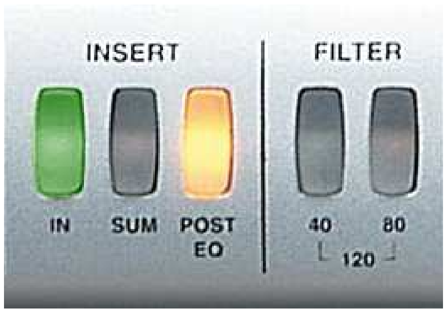
On mix down you may wish to use the Alpha Channels various functions to give signals the classic SSL analogue sound.

Because the Alpha Channel features a total 3 outputs and 2 inputs, there are a variety of ways to connect the unit in your studio for this purpose.

The example below shows the cleanest way of doing this, avoiding the colouration offered by the VHD circuit yet allowing the EQ and limiter functions to be used.







- The INSERT IN button should be depressed so the line level analogue signal from the audio interface appears after the pre-amp.

- The SUM button should be up because we are not interested in adding the front panel input to the insert return. If the SUM is depressed in this scenario then you may hear some noise being added to the signal if the input gain is wound up – this could indeed be used as a feature!

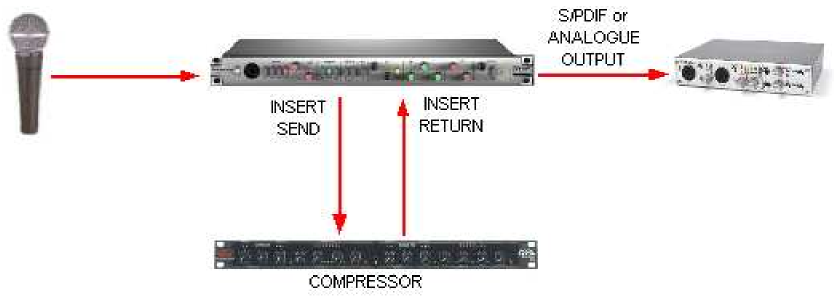
- The POST EQ button should be down so the signal is processed by the EQ section.

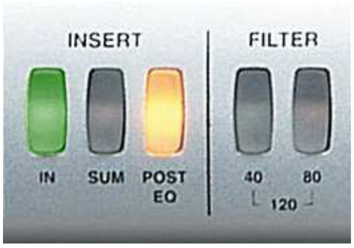
- If you wish to only use the limiter and A/D functions of the Alpha Channel then switch the POST EQ button up. This means that the signal present at the insert return (the input in this case) is placed after the EQ section but before the limiter and A/D.

- You can of course use the front panel input instead of the insert return, which would enable the VHD characteristics of the pre-amp to be utilised in the processing chain along with the filters. In this case you should depress the PAD button as we are dealing with line level signals.

Using a Compressor in the Insert send/return loop

It may be desirable at times to use a compressor to control the dynamic range of a signal which is being recorded. The insert send and return loop on the Alpha Channel provides a flexible way if interfacing with equipment such as this.





- The signal is presented to the front panel input and amplified accordingly.

- Pressing the INSERT IN button will not only send the pre-amplified signal to the compressor but will return it back to the Alpha Channel for further processing.

- The POST EQ button will determine whether the signal is first compressed and then EQed, or the other way round. Pressing this will place the send/return loop after the EQ. Note that the behaviour of the compressor can be changed significantly depending on whether it is pre or post EQ.

If you feel that the incoming signal is quite even across the frequency spectrum and does not contain any significant humps or troughs, it should be fine to send the unequalised signal to the compressor. It may then be worth lifting the high end of the EQ section up slightly to compensate for the high frequency degradation that compressors can sometimes exhibit.

Where you feel the EQ is needed to first even out the signal, send the signal to the compressor after the EQ – i.e. The POST EQ button down.

After the EQ, filter and insert loop the signal is sent to the Alpha Channels outputs (analogue and digital).

Note: By engaging the Lite Limit button the INSERT SEND level is dropped by 12db. This is in order to give the unit enough headroom during limiting action.
This will however affect the compressor settings quite significantly. If you have set up the compressor and subsequently engage the Lite Limit function then the compressor may well need to be adjusted again. In these cases, moving the compressor threshold down by 12db and its make-up gain up by 12db should provide the same behaviour as before the Lite Limit was activated.

Using VHD

VHD – Variable Harmonic Drive is a unique feature of the Alpha Channel pre-amp. It allows a signal (mic or line) to be over driven from subtle harmonic colouration through to full on distortion characteristics.

The VHD circuit is inherent to the pre-amp, but usually only comes into noticeable effect when the knob is in the last quarter of it's travel. This allows the pre-amp to be used in the normal manner (without distortion!) for most of it's range and the distortion characteristics to be brought into play at higher gain settings.

The level of the incoming signal will have an affect on the VHD circuit, as will the PAD button which offers 20db of gain reduction for loud signals.

The second knob in the input section, labelled 0-11 changes the character of the distortion from 3rd harmonic to 2rd harmonic. This will only have an affect at high gain settings.
2nd harmonic distortion is associated with the harder, brighter sound achieved from solid state devices, where as 3rd harmonic is more akin to the warmer sound produced by overdriven valves.

You are informed by the Alpha Channel of the distortion state where VHD will become noticeable. The PAD LED will glow red when this is happening, and this is generally best avoided for a clean signal.

The VHD is highly dependant on the input signal. It becomes more obvious when used with a signal that contains mainly low frequencies, i.e. Bass guitar. If the input signal is predominantly high frequency, then the change in effect will be less noticeable as the harmonics generated by the overdrive may be above the bandwidth of normal hearing.

Be aware of the situation that occurs if the pre-amp is over driven. This will have an affect on the gain of all parts of the unit which follow in the signal chain. The Alpha Channel has of course been designed to allow these high gain settings without overloading other parts of the circuit, but you may find that the output gain will need to be taken down to interface correctly with other equipment. You could engage the Lite Limit here to stop the A/D converters overloading if using the digital output.

Parallel Processing

This is where a processed signal and it's unprocessed counterpart are summed together to provide a single source. It often allows you to apply extreme processing but still retain the dynamic and character of the original.

The Alpha Channel provides an easy method of performing this function:

If the SUM is depressed whilst using a compressor in the send/return loop then the unprocessed signal will be added to the compressed. The output gain of the compressor can then be used to balance the two.

Two Alpha Channels as a stereo pair

Any two or more Alpha Channels have the ability to be linked together. The two phono connections above the S/PDIF sockets labelled 'LINK' provide access to the side-chain of the Lite Limit circuit.

By simply connecting a phono cable between any 2 alpha channels using the link sockets, the Lite Limit on each unit will work in sympathy with each other because their side-chains are linked.
As they are a parallel of each other it doesn't matter which of these are used and multiple units can have their side-chained linked in this manner.

This is essential for correct stereo operation of the two limiters, because each needs to be aware of what the other is doing to retain a stable stereo image.

To help get the most from your audio production adventures, SSL created Mixbuss, be sure to take a look. http://www.mixbuss.com