



# X-Phase User Guide

## Introduction

### About SSL X-Phase

X-Phase is an all-pass 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 all-pass filter and time alignment tool for manually adjusting the phase.
- The gain is linear across the frequency range, but a change in phase is applied such that it 'wraps' at a specific frequency and converges to 0r 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 degrees phase shift with All-pass 'phase' button.

## Installation

You can download installers for a plug-in from the [website's Download page](#), or by visiting a plug-in product page via the [Web Store](#).

All SSL plug-ins are supplied in VST, VST3, AU (macOS only) and AAX (Pro Tools) formats.

The installers provided (macOS Intel .dmg and Windows .exe) copy the plug-in binaries to the common VST, VST3, AU and AAX directories. After this, the host DAW should recognise the plug-in automatically in most cases.

Simply run the installer and you should be good to go. You can find more information about how to authorise your plug-ins below.

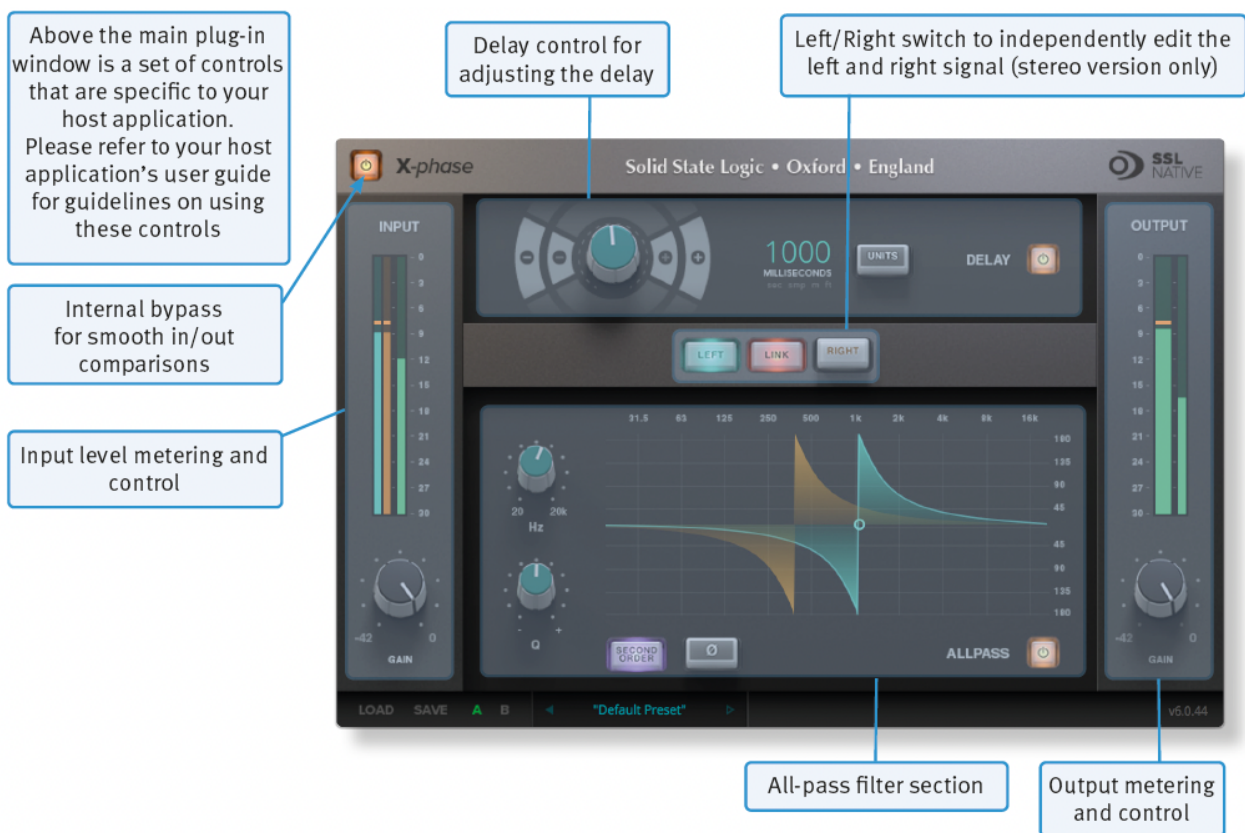
## Licensing

Visit the [online plug-ins FAQ](#) for guidance in authorising your SSL plug-in.

## Using SSL Native X-Phase

### Overview

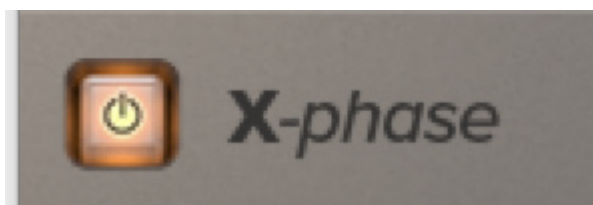
The illustration below gives an overview of some of the X-Phase features which are described in full over the following pages.



### Interface Overview

The basic interface techniques for the X-Phase are largely identical to those for the Channel Strip.

### Plug-in Bypass



The **power** 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 plug-in to be in circuit.

## Automation

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Automation support for X-Phase is the same as for the Channel Strip.

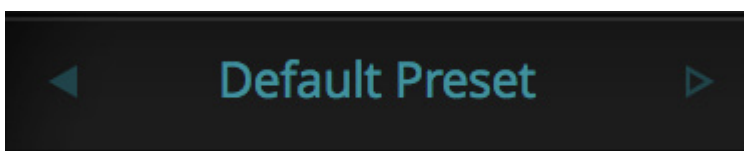
## Presets

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Factory presets are included in the plug-in installation, installed in the following locations:

**Mac:** Library/Application Support/Solid State Logic/SSLNative/Presets/X-Phase

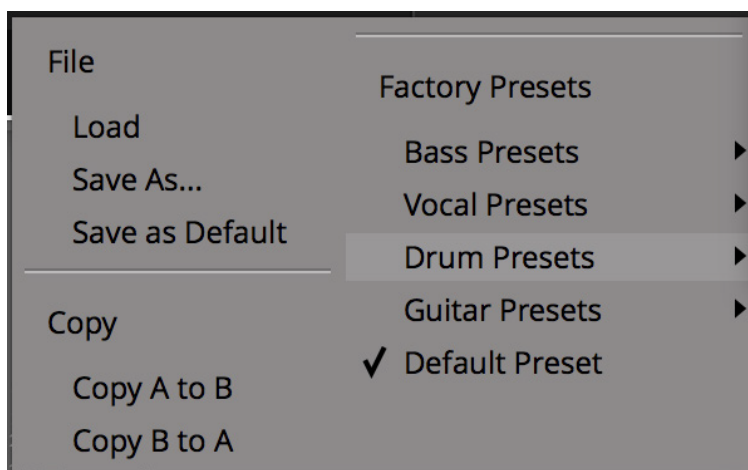
**Windows 64-bit:** C:\ProgramData\Solid State Logic\SSL Native\Presets\X-Phase



Switching between presets can be achieved by clicking the left/right arrows in the preset management section of the plug-in GUI, and by clicking on the preset name which will open the preset management display.

## Preset Management Display

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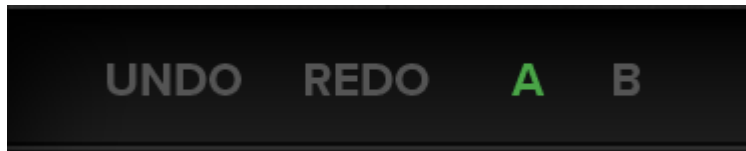


There are a number of options in the Preset Management Display:

- **Load** allows loading of presets not stored in the locations described above.
- **Save As...** allows for storage of user presets.
- **Save as Default** assigns the current plug-in settings to the Default Preset.
- **Copy A to B** and **Copy B to A** assigns the plug-in settings of one comparison setting to the other.

## A-B Comparisons

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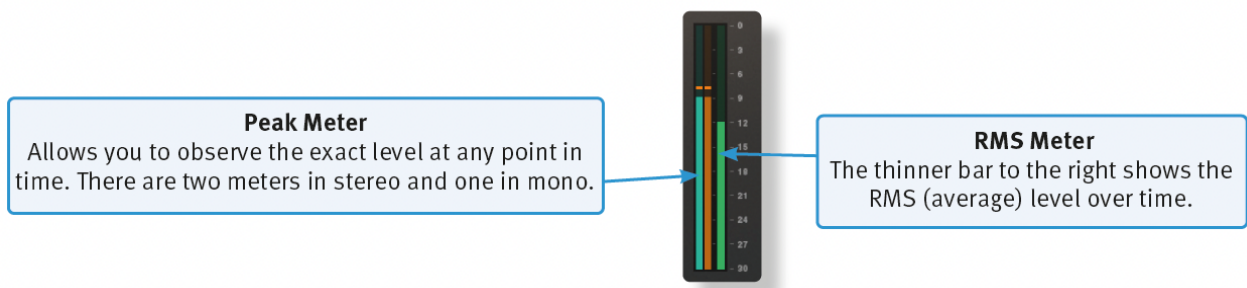
The **A B** buttons at the base of the screen allows you to load two independent settings and compare them quickly. When the plug-in is opened, setting **A** is selected by default. Clicking the **A** or **B** button will switch between setting **A** and setting **B**.

**UNDO** and **REDO** functions allow undo and redo of changes made to the plug-in parameters.

## Input and Output Sections

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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:



### Peak Meter

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

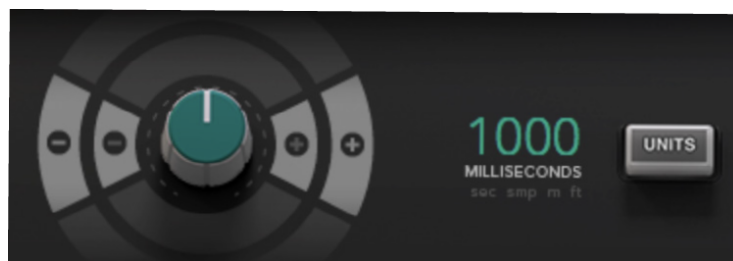
### RMS Meter

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

When clipping occurs, the meter will turn red. It will remain red until the meter is reset by clicking on the meter.

## 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.



## All-pass In

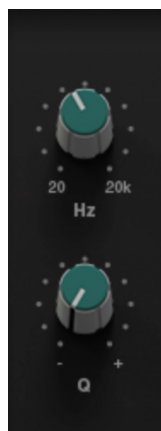
Activates the All-pass Filter.

## All-pass Phase

Inverts the phase of the All-pass Filter.

## Frequency

Sets the Frequency of the filter.



## Second Order

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

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

## Phased by Phase?

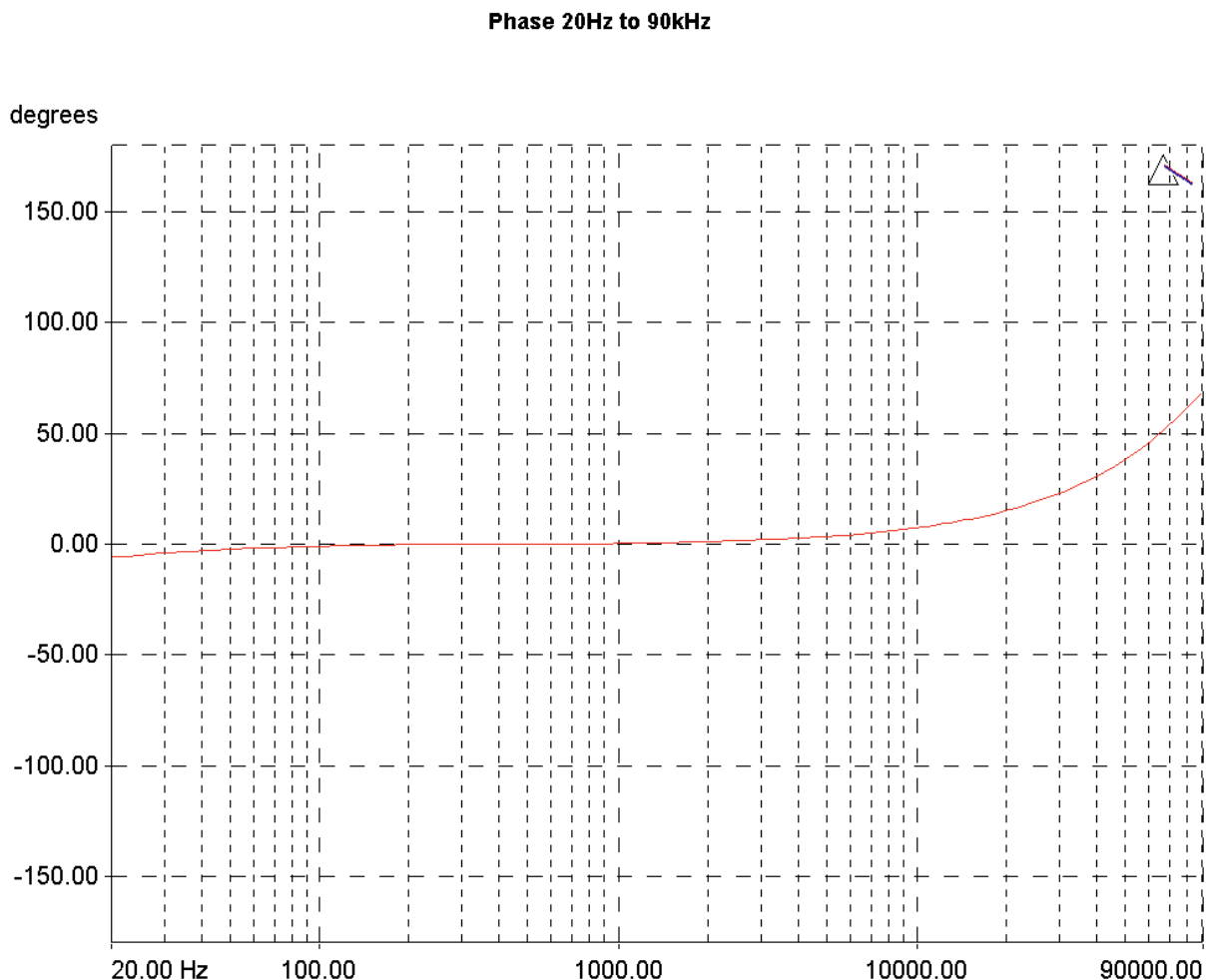
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Phase. It's a funny subject, one that many of us think we know a bit about but unfortunately don't often fully understand how to handle all of it. The idea that two microphones, set at different distances from a guitar, will have the sound arrive at 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 two 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 two 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 remains 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 with 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 +/- 70° 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 All-pass Filter Idea**

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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 inter-symbol interference (ISI) in the demodulation of digital information from an analogue carrier signal.

X-Phase can solve these problems; for example, by aligning the overhead microphones of a drum-kit 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 two-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 artefacts 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 for 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 three microphones for a piano they are all separated by different physical distances and are all picking up similar frequencies. But when you try and mix those three microphones together you will have a 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 overall 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.