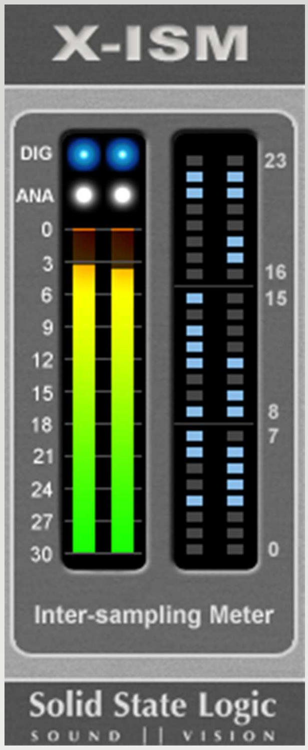
X-ISM

Background Information

Introduction



Thank you for your interest in Solid State Logic X-ISM plug-in for PC / Mac.

At Solid State Logic, we are committed to the pursuit of sound quality excellence. As part of our continuing research into digital audio, an area of constant debate is the effect of inter-sample peaks on the listener's experience. The availability of tools to analyse this phenomenon and predict occurrence for DAW users is limited and this is the reason for developing and releasing X-ISM as a VST/AU plug-in.

There is currently a trend in the in the audio industry known euphemistically as 'The loudness war' in which the engineering process is driven to produce a product which is as loud as is technically possible within the constraints of the medium. Debate rages as to whether this produces problems that degrade and distort the audio. One reason for this distortion is often blamed on the presence of 'inter-sample peaks', where signals that are usually missed by the Fs sampling of DAW meters, may actually exceed 0dBfs in reconstruction from digital to analogue domain in some Digital to Analogue Converters (DACs).

The Solid State Logic X-ISM is a state of the art VST/AU plug-in which can interpret inter-sample peaks, allowing engineers to make informed judgements about the resultant sound quality of the mix.

Why is this important?

Reliable metering and monitoring is a key to consistent results. Recording studios of all sizes choose the best equipment in order to produce the best results possible, and this invariably includes audio interfaces. So it is very likely that the circuitry used in professional audio interfaces will be more sophisticated than those used in consumer devices. It is therefore entirely possible for an engineer to produce a mix which would sound great in the studio but may sound different on some domestic systems.

There have been several tests performed on domestic CD players with signals known to produce inter-sample peaks, with interesting results. The tests indicate that some players are quite capable of handling the peaks without audible results, but some models produce audible clipping.

So if the recording engineer has no idea whether the mixes he/she produces will sound distorted on some domestic playback systems how can the problem be fixed? Probably the best solution would be for the studio to have a variety of types of players to play mixes on, along with an engineer with very good hearing!

A practical alternative is to use a meter which simulates the oversampling DAC filtering processes used most commonly, and can therefore indicate the presence of ›0dBfs inter-sample peaks, even if the peaks can't be heard in the control room. Preferably, this meter would take the form of a VST plug-in that can be inserted at the end of the DAW mix. In this way the pure digital signal can be monitored and results predicted.

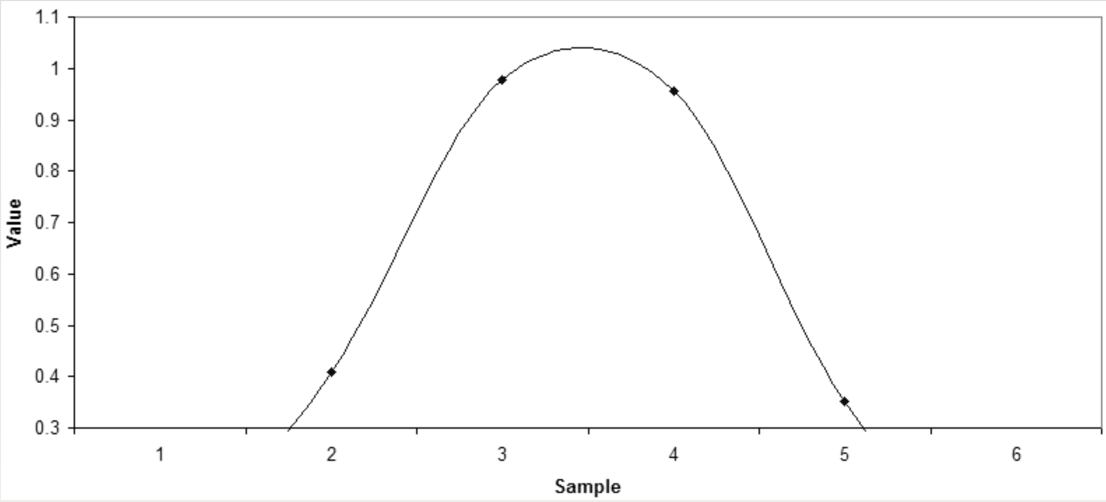
Enter X-ISM...

Inter-Sample Peaks

When a digitally-recorded sound is played-back it must be converted into an analogue representation at some point. This is performed by the DAC (digital to analogue converter). Commonly used over-sampling DACs apply a 'reconstruction' filter to convert the periodic sample levels back into a discrete signal. In simple terms they 'fill in the gaps' using interpolation so that digital artefacts are reduced to a minimum.

Regardless of the reconstruction filter's design, in certain circumstances it is possible for a reconstruction filter to produce signals greater than odBfs even if the samples used to create these signals are all within the odBfs limit.

Take the following example:



Here is a portion of a high frequency sine wave that is represented in the digital domain by 4 consecutive samples. The line that joins these points is the product of an interpolation algorithm designed to provide a smooth, discrete wave between the points.

As can be seen, all 4 samples have a value of less than 1.0 (odBfs in this case) but the smooth line joining the points clearly contains values which are greater than 1.0. It is this mechanism which can cause analogue signals to theoretically exceed odBfs even though the digital representation of the signal is within limits. This is an 'Inter-sample Peak'.

It can be seen that it is possible for all of the digital samples to be below full scale but they could still produce a signal which has sections which are ›odBfs.

The current trend to produce louder and louder mixes ultimately can result in commercial recordings containing more inter-sample peaks than those of say 20 years ago. In reality, a highly compressed piece of audio that has been normalized to digital full-scale is almost guaranteed to have some samples in similar positions to those in the example above, and therefore the audio at an oversampling DAC is almost guaranteed to have inter-sample peaks of ›odBfs!

Is this a problem?

As already mentioned, the reconstruction filter is typically part of common Digital to Analogue converters (DAC) and the quality of this process is a factor in the overall quality of the DAC. Low quality DACs may not have the digital resolution to handle values larger than full-scale, and subsequent analogue stages may not have the headroom to handle signals larger than the theoretical maximum. Another possibility is that the DAC may be quite capable of handling ›odBfs but the surrounding design may not have taken this eventuality into consideration.

How does X-ISM work?

Firstly it must be understood that most DAW peak meters (including plug-ins) simply look for samples which are at digital maximum. They will indicate a peak condition when there have been enough of these 0dBfs samples in a row (usually 3 together). Although this can give an indication that the engineer has run out of digital headroom (and that remedial action is necessary), it really gives no indication that the resultant analogue waveform may contain >0dBfs peaks due to the inter-sample peaks.

It is possible that a mix could have many values at digital maximum but no inter-sample peaks, similarly a mix could have no values at digital maximum but lots of inter-sample peaks!

Commonly, current mass market DAC designs do the vast majority of their reconstruction process by employing a digital Finite Impulse Response (FIR) filter design tailored to provide a brick wall low-pass response. This approach overcomes the less perfect slope response issues of pure-analogue filter designs. These FIR filters work as follows; firstly, the audio is up-sampled to a multiple of the original sample rate, then the FIR is used to create the smooth curve in-between the original sample points. Of course different DACs have various ways of doing this and indeed, the finer details of the process are important design elements of the DAC design.

X-ISM uses significant processing to provide a combination of up-sampling and filtering that mimics the operation of an oversampling DAC's reconstruction process. The result is a meter that shows inter-sample errors and provides a useful tool that most DAW metering misses.

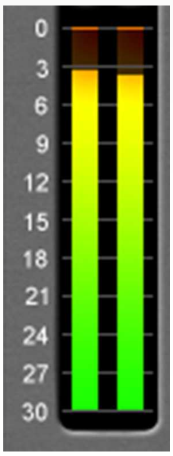
A tour of X-ISM

Digital clip indicators

A traditional stereo clip indicator that shows a digital value at full scale (0dBfs). This meter will illuminate after one such value to indicate that the digital ceiling has been reached.

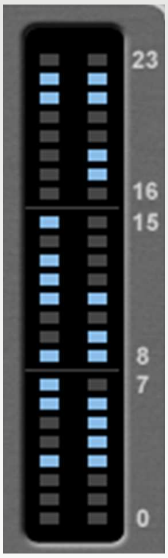
Analogue (inter-sample) clip indicators

A stereo clip indicator driven by the up-sampled, reconstructed waveform. It's function is to show the presence of inter-sample peaks.



Traditional peak meter

Monitors the digital waveform, sampling successive values to calculate peak readings.



Peak bit meter

The purpose of the bit meter is to give an indication of how many bits are being used with respect to the total available. It is also a useful display of any dithering process being used.

In this version of X-ISM, the meter is fixed to 24 bits but this means it is still quite usable on 16 bit signals too.

This meter works by taking audio values (represented within the DAW as floating point numbers) and converting them to a fixed 24-bit format, the 24-bit format was chosen as most engineers use either 16 or 24 bit depth whilst recording. This allows the engineer to 'see' the audio as the converter would.

Some bit meters are used by hardware designers and test engineers to detect hardware faults such as bits which stick on or off, but this is not the purpose of this meter.

Traditional bit meters will grab and display the current sample value as fast as it can (bearing in mind us humans can't see things which change faster than about 50 times per second). As a result, even a steady sine wave will produce a meaningless flashing light display which is more entertaining than informative for studio engineers.

The bit meter in X-ISM is driven from the same peak detector as the traditional peak meter (above) and therefore shows the largest value present and therefore how many bits are left available (headroom). This means that is a signal is too large for the converter the bit display will show all bits ON.

Compatibility

The plug-in is available in VST, AU and RTAS* formats making it compatible with virtually all audio software currently available on both PC and Mac. The product has been tested in the following applications:

PC	Mac
• Cubase	• Cubase
• Nuendo	• Logic
• Fruity loops / FL Studio	• Digital Performer
• SONAR	• Ableton Live
• Reaper	• Pro Tools*
• Ableton Live	
• Pro Tools*	

*RTAS is supported through the VST-RTAS wrapper from FXpansion, available from http://www.fxpansion.com

System Requirements

Due to the nature of the processes inside X-ISM, it is quite a CPU hungry plug-in. We therefore recommend the following:

PC	Mac
Operating System: Windows XP	Operating System: Mac OSX 10.4.8
CPU: P4 2.4GHz / AMD 2400xp+	CPU: G4 1.2Ghz
System Memory (RAM): 512MB RAM	System Memory (RAM): 512MB RAM
Hard Disk: 50MB free space	Hard Disk: 50MB free space

Useful Background Reading

http://www.digido.com/bob-katz/level-practices-part-1.html