

# Sonnox Fraunhofer Pro-Codec

We know that compressed formats are with us for the foreseeable yet optimising for distribution has remained a laborious and convoluted process. **BILL LACEY** says that all that has changed with the arrival of this unique and ingenious plug-in.



At first glance this new plug-in might raise an eyebrow. Just who is Fraunhofer, and why should we care? And is it really worth getting excited about an mp3 encoder? I've been in the business longer than the mp3 has been around, and to be honest I didn't know who Fraunhofer was. I've always been hyper-focused on working with the highest quality audio possible, and admittedly shunned the mp3 format as not being an acceptable listening format for my clients. The world has changed, I am older and hopefully wiser, and the mp3 is here to stay. The Compact Disc is on life support, our lives revolve around our iPhones and iPads, and The Beatles are now on iTunes (compressed, of course).

Fraunhofer IIS began work on improving audio

transmission over ISDN lines in the late 1980s and in 1991 completed development on the MPEG-1 Layer 3 codec. In 1995 researchers at Fraunhofer chose mp3 as the file name extension for this codec, and the rest is history. While the audio industry fought the high def wars (DVD-Audio vs Super Audio CD), the mp3 quietly entered the market and evolved into the delivery format of choice in the new portable music player and download world in which we live. Fraunhofer has continued development of new audio codecs and its joint effort with Sonnox brings to us a plug-in that for the first time gives content creators and engineers control over the compression characteristics of audio — that is sure to change the way we work from now on.

Mp3 and its AAC siblings are defined as 'perceptual'

audio codecs. They take advantage of the ear's inability to process all frequencies at differing amplitudes simultaneously. When the membranes of the inner ear receive multiple sound waves the frequencies of greater amplitude overwhelm the neighbouring ear cells making it difficult to process adjacent frequencies. This translates into one sound masking another. When we speak of data compression we are talking about throwing information away to make the file size smaller. Perceptual audio codecs accomplish this task by deciding which information the ear will be unable to hear and remove it. While reducing file size, the process also introduces errors in the form of 'coding noise', which is typically masked. The inability of the user to identify and address this noise and make creative decisions regarding the degree of data reduction has frustrated the professional audio community for years. This situation has finally been resolved.

The Sonnox Fraunhofer Pro-Codec is the first plug-in to offer real-time auditioning of the mp3/AAC encoding process. Traditionally this process involved outputting a mix via an mp3 encoder, listening to the result, making changes to the limited number of options, outputting again, listening, etc.... Now users have the ability to hear the encoded result and compare against the original signal in real-time, identifying exactly what's being removed visually and aurally, monitoring codec clipping, A/Bing different codecs, outputting multiple codes simultaneously, and more. The layout is such that almost anyone can get great results immediately, and the more technically inclined can dig a little deeper and tweak a little further. And if that's not enough, the layout is one of the most creative I've seen.

The plug-in (UK£295) ships in RTAS, Audio Unit and VST formats, and is supported on Mac Intel 10.5 or later and Windows XP or Windows 7. Audio formats include mono, stereo and 5.1 surround. Compatible workstations include Pro Tools, Logic, Cubase, Nuendo, Sequoia and Wavelab.

Typically you would insert the plug-in as the very last element in the signal chain, such as the Master Fader. Any required dithering should take place before reaching the input to the plug-in. The left hand side displays an LED meter showing the input signal, allowing you to monitor any input clipping before hitting the codec. The main window of the plug-in features a helpful FFT display, with five

slots below to select the desired codecs to audition along with a monitor select button and LED to indicate codec clipping. Any combination of codecs can be output simultaneously. This is a massive time saver. There are also options for offline encoding (which is preferred in some scenarios) as well as offline decode back to WAV or AIFF. Three tabs to the right of this section offer further control and feedback. The first is labelled Trim, which allows the user to monitor the bitstream levels and reduce the input to the codec in the event of clipping. This is important even if the input signal does not clip because the encoding process itself may increase the peak level of the signal and induce clipping. The second tab is labelled Comp, and displays the bitstream compression factor. This is for visual feedback only, and primarily confirms what you are hearing. When using lossless and variable bit rate (VBR) encoders the display provides an approximation and is coloured yellow. Finally, an A/B tab offers extensive options to compare different codecs. There is also a clever system allowing for testing of the user's listening skills, complete with a grading of the results. It's a great way to test your hearing and determine if an early retirement is in your future or not.

The FFT is unique in that in addition to being able to display the left/right and sum of the input signal (shown in yellow) users can also see the difference between the input and the output signals (shown in red), referred to as the Diff signal. Additionally, the display offers an NMR display, which is defined as the 'noise to mask ratio'. This provides a visual indication of whether or not the codec induced noise will be audible or not. A green line indicates that any codec noise will be completely masked and not audible.

Areas in red indicate artefacts that may be audible, allowing the user to determine the significance of the effect and adjust as necessary.

The tab to the right of the FFT display provides all the necessary controls to toggle the codec on and off — free of glitches — along with a master Record button. You may switch between linear and logarithmic scales for the FFT, as well as a Lissajous display to monitor phase. The highlight of this section is a button marked Diff, which allows you to listen to the audio that is being removed by the codec. This feature alone may be worth the price of admission.

As easy as this plug-in is to use, there are a few situations that are worth looking out for. Attention must be paid to the sample rate of the current session in your DAW, as the plug-in derives the sample rate of the encoded file from this. This applies as well to channel configuration. It is also important to present the correct bit depth and



dithered signal to the lossless codecs.

Other features worth mentioning are a flexible export naming scheme, mouse controls for zooming the FFT display, and a bit meter when using lossless codecs that gives you visual feedback as to whether you are dithering and truncating correctly. In addition to the expected mp3 and AAC codecs, Fraunhofer provides HE-AAC and HE-AAC V2 codecs (frequently used for streaming internet radio) and two flavours of lossless encoding — Mp3-HD and HD-AAC. Some special conditions exist that may prevent you from auditioning in real-time, and there are practical reasons for this. These show up when using a variable bit rate (VBR) at its highest settings with the mp3, HE-AAC and HE-AAC V2 codecs. In these cases the codec previews with a lower quality. The Mp3-HD doesn't offer

any real-time audition, so you're just listening to input all the time. In each case the user is alerted to the situation and is provided with the next best option. The AAC LC (low complexity) and HD-AAC lossless codecs have no real-time restrictions at all and preview in all circumstances.

The HD codecs have a special 'hidden' feature. In addition to the compressed lossless file, it contains what is referred to as a 'lossy core' channel. This allows it to function as a lossless archival format, containing the lossless and lossy file. What this means is that if your

decoder doesn't support HD-AAC, it will at least be able to play the lossy channel embedded in the file. When preparing these files it is very easy to set up an A/B comparison in the encoder to compare the lossless to lossy versions.

During testing I found the plug-in extremely easy to use. Creating great sounding mp3 files has never been easier. Record enable the individual codecs you desire, hit the master record button, and play your DAW. The encoded files are written in real-time to a preselected destination. Exact data compression ratios are displayed for all record enabled codecs.

Mastering engineers are going to love the ability to determine the best settings for the job at hand. Ultimately, the goal is to strike the right balance between audio fidelity and file size. Nothing else

allows you to accomplish this task as well as the Pro-Codec. It's a great idea, implemented well, and deserves the highest recommendation. ■

#### PROS

Ease of use; superb aural and visual feedback; outputs multiple codecs simultaneously; Diff signal monitoring worth the price; great sounding lossless codecs.

#### CONS

Maybe a bit pricey for the average user but a no-brainer for pros.

#### Contact

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