Paul Frindle

The man responsible for the product concept and design of Sony plug-ins talks about the state of digital, what can be achieved, clouding the issue, brute processing power, and the underestimation of what we can hear.

FRINDLE STARTED HIS professional design career around 1970 — as an extension of childhood hobbies and a fascination with electronics and music — designing a multitrack audio tape machine, audio and RF test equipment and various industrial control systems. Sound and technical engineering in studios began in Paris in 1975 and he went on to work as a technical engineer and mobile recording engineer for Virgin during the late 1970s.

He joined SSL at the beginning of 1980s as a design engineer, working on parts of the E-Series, and led a design team making an analogue assignable console system. Frindle later designed the channel electronics on the G-Series and his final project at the company was the digital 0.1 system, where he designed the A-D and D-A conversion and clock/sync PLL systems.

He is one of the founder members and co-director of Oxford Digital, involved in many aspects of the desk project from the design of the conversion systems, EQ, dynamics and the overall philosophy and operation of the system and assignable panel concepts. He is now responsible for the product concept and design of Sony plug-ins.

How much Sony Oxford console is in Sony Oxford plug-ins?

The EQ and dynamics are pretty much identical in processing terms to those in the OXF-R3 console, with some minor GUI modifications to make them more compatible with screen-based operation. There was a basic version of the TransMod in the later releases of the OXF-R3, but the plug-in version has far more control for greater flexibility and control of sound effect, resulting from feedback from beta testers. The Inflator is an entirely new process that did not exist in any previous form.

The applications that are common to previous OXF-R3 processes actually achieve higher technical performance than the console versions did, mainly because we can more easily make use of double precision (48-bit) calculations. But of course EQ and Dynamics are only one part of a large console system and do not in themselves define all aspects of a console design.

What differentiates the Sony Oxford approach to plug-ins?

An enduring professional audio application of merit is not defined only by absolute measured performance. Rather it is a whole entity designed to interact with the user and the art in such a way as to inspire and encourage great artistic results — much like a musical instrument. Much of this sensitivity and design attitude, built up over the past decades of professional audio design is directly applicable to the workstation environment.

Our intention is to identify areas in which we can bring improvements to overall quality and artistic freedom, drawing from our experience in professional audio and console design. At the basis of this approach is the understanding that all users actually aspire to excellence and can tell when they have achieved it — regardless of the host platform style they are using. We believe that a professional level of facility is no longer the sole reserve of those that have the honour of working on expensive and exclusive equipment. We want to bring that quality, artistic nuance and excellence within the grasp of today's workstation users. Recent advances in the processing power and affordability of workstation systems makes this a reality.

What are the most tempting compromises that plug-in manufacturers can make?

Like in any design process, if an atmosphere of compromise exists it can take many forms and operate on multiple levels. In reality all design is in fact a compromise — nothing can ever be perfect, and you cannot muster infinite resources. However if you know exactly what you are aiming for, it should be possible to restrict unavoidable technical compromise to issues that don't affect the performance and character of what you are trying to achieve. One always has the option of simply not making something that cannot yet be realised properly or at reasonable cost using current technology.

In my long experience, the biggest misconceptions in our industry revolve around underestimation of what we can hear. The process of professional recording and mixing is an extremely complex and highly sensitive, essentially artistic, process that succeeds or fails largely on the depth of involvement between the engineer, the art and the equipment.

Under these conditions we are far more sensitive to aberrations in sonic character than people would like to believe or would seem academically reasonable.

If we add to this the whole issue of the user's subjective perceptions based on operational styles, control types, laws, ranges and interactions and general 'feel', it should be fairly evident that there is much that could interfere with the success of this whole delicate artistic process, if compromise begins to intrude.

How real an issue is the need to reduce plug-in processor 'hunger' and what are the implications?

As in any design initiative, there is always pressure to reduce cost to the user. The workstation environment is no different from any other in this respect. This can easily lead to applications that are of lower quality or less capability than they could have been — if you let it.

But it is the responsibility of the designer to resist this pressure. A good designer who has a firm grasp on the whole product concept, its application and its user's aspirations should understand why certain levels of quality are needed, even if everyone from the users to the marketing guys are screaming at him! The people who are screaming would not necessarily know what they would lose due to compromises if they never got to hear the real thing! The art of good design is understanding in great detail the nuances of what will affect the user within the nature of your design concept. In many respects it is this that differentiates a great product from an average one.

We strive very hard to reach a point where we believe that not a single processing tick is wasted in our plug-ins, but at the quality level in the market we are aiming at, every plug-in has to do exactly and precisely what we intended it to do. Nothing else would make any sense.

How dependent on the host computer or host DAW is plug-in performance?

For systems that use supplementary signal processing modules, the host processor is confined mostly to control calculation processing, GUI handling and communications, etc. Therefore generally speaking, lack of host processing power or high system usage results in excess control or display latency. Although the program itself is not necessarily affected, at the limit this can still result in a loss of immediacy and 'control' in the overall operational experience.

For systems that use only the host system for signal processing, the complexity and number of plug-ins that can run simultaneously is directly related to available processing power and internal communications bandwidths. However, no attempt is made in any Sony plug-ins to downgrade host versions to save processing loads. All perform to the same standard whatever the platform and they will either find sufficient processing to run or the system will stop. It would be entirely unfair to saddle the engineer with nebulous, subtle or unpredictable performance variations depending on the platform format and condition.

What are the technology bottlenecks and limitations that are holding plug-ins back? What can be going on inside that the user doesn't know about?

With increasing pressure on engineers to mix entirely 'in the box' the bottleneck is brute processing power. One of the essential factors in conventional dedicated console workstyles is the facility of having all your applications, processing and potential routing running simultaneously so you can grab an EQ or compress a track by simply reaching for a control. If people are to continue down this route with their workstations, the requirement for processing is likely to be open ended.

There are several ways in which plug-in designers might try to reduce the processing costs of their applications, for example sub-sampling, coefficient or signal accuracy restrictions, or the removal of less important processes. But all are likely to degrade performance in one or several ways.

A plug-in's absolute performance is dictated by the user's choice of convertors and the 'structure' of the system used to run it on. What steps can a user take to maximise performance?

This is a very big subject indeed. There are many popular misconceptions about digital systems, many of which, despite coexisting in the popular perception, are actually completely opposed in concept. Industry buzzwords like 'resolution' and the like, despite seeming obvious are in fact little understood (because it's more complex than that) and cloud the issues permanently. For instance, people used to claim that because digits are numbers the results must be perfect and beyond reproach. Many people are now actually arguing about whether a digital processor can add two numbers together!

In reality and more seriously, there are a great many factors that affect the resulting quality from a whole system, many of which involve design decisions and shortcuts taken in the various parts of the interconnected installations and the way they perform together. Some of the most hotly debated issues are defined by the host application's control and signal structures and not by 'the nature of digital technology'. Some revolve around ill-conceived modulation levels and industry wide misconceptions about subjects like 'resolution' and sampling rates. We have issues at all levels from hardware to software, from jitter in cables all the way up to the presence or absence of appropriate dithering — it's a real zoo — I have written whole AES papers that cover only a fraction of it!

My advice in the current environment using current applications is:

1 Find a really good convertor system that has impeccable signal performance (when connected in a realistic installation) and very crucially has very good internal PLL based clock recovery.
2 Always use the best possible quality of mic preamp during recording sessions. The analogue bits are still the biggest cause of quality loss!
3 Record at lower modulation levels (i.e. -3dBFS peak) to avoid signal clipping that does not always show up on the sample level type metering that comes with your workstation application. Avoid using the red 'over' light as your operating level indicator.
4 Attenuate all signal returns into the mixer during the mixing session to avoid internal overloads that won't necessarily show up on your meters (i.e. perhaps -10dBFS peak). And make up the level at the final output of the mixer.
5 Always re-dither at the final mixer output to the intended word length of the target media.
6 Forget the words 'resolution' and 'digital', set your faders where they naturally want to be and just interact with it.

What impact do the higher bit and sampling rates being offered on DAWs have on plug-ins?

Generally speaking doubling the sampling rate halves your available processing power and doubles the cost of the plug-in processing. In some cases an increase in coefficient accuracy is required to maintain comparable performance at higher sampling rates. Where there are host platform process partitioning restrictions or memory limitations, it may be that some plug-ins

cannot be made to work at all at higher sampling rates and there may be pressure to sub-sample the internal processing of the plug-in to enable it to work.

Under some conditions, under-performing plug-ins may produce better (or different) results at higher sampling rates. Or others may produce worse performance if they have different intrinsic deficiencies.

All the Sony plug-ins produce identical performance for signals up to 20kHz regardless of the sampling rate.

How close to a hardware equivalent can a plug-in be made to be?

If you are talking about digital hardware applications, there is no philosophical reason why a plug-in should not be identical to a hardware digital box. However, limitations in processing within workstations may currently prevent this being realised for some applications.

For analogue hardware it is far more difficult since some of the character of these units may reside in the residual effects due to imperfect circuitry or intentional exploitation of complex behaviour in certain components. But ultimately anything can be emulated if you have sufficient inside knowledge, time and available processing. If it produces the same signal under all conditions it will sound the same — there is no magic — it's all signal.

Once all the 'emulations' of hardware have been served, plug-in developers will ultimately have their imagination challenged to come up with more original plug-in ideas. What sort of direction are these new ideas likely to take?

That's an interesting statement because it illustrates a fairly widespread notion that all digital processes are somehow emulations of existing analogue ones. I view this very differently in that none of the Sony plug-ins are actually emulations of anything particular (with the exception of the GML EQ option). Rather they are realisations of applications that I had always wanted to design but could never achieve using analogue circuitry. The performance of these applications is many times what would be theoretically possible using analogue circuits — which sadly are constrained by the laws of physics.

A great many totally new applications have been designed because digital processing allowed it (the TransMod is one such example). Whole new artistic styles and fashion idioms have already resulted from some of these applications. In my opinion this is very exciting and the field is still fairly wide open — limited only by imagination and appropriateness. I would predict that there will be many more ground-breaking realisations as the cost of processing decreases over time. I would also hope that more people will use increasing processing capability to improve the quality of their existing application functions.

I have several other plans based on existing functions I consider to be insufficient and open to improvement. I also have plans to try several other ideas out that couldn't previously be achieved in analogue technology. We have a couple of new applications in development now that should be released later this year.

Can plug-in quality be improved on, can digital be improved?

Yes – in principle it is always possible to improve things. I have never made anything ever where I could not have conceived of something even better. Certainly the advent of digital processing has significantly breached the barriers that contained us during the analogue design years. And for familiar and comparable applications like EQ we are achieving performance beyond anything I could ever imagine 25 years ago.

When it comes to out and out signal integrity, maybe it's getting difficult to imagine how further improvement could ever be significant and noticed. But there are always new functions, features and nuances lurking in the back of our minds and they are often inspired by listening to the changing artistic trends all around us and perceiving the new artistic opportunities they suggest to us. Perhaps we can never be truly satisfied, when driven by art and imagination.