Sintefex Audio

The Digital Analogue: The Sintefex Audio Replicator FX8000

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Digital Audio has suffered from the basic lack of excitement of digital processing. For this reason the totally digital signal path has failed to materialise in the real world, with frequent returns to analogue being needed both in recording and mastering. The Sintefex Audio FX8000 Digital Audio Effects Replicator brings a new technology to bear to solve this problem, making the digital domain a true home for complex audio processing.

Recording - Digital or Analogue?

There is no doubt that audio recording for speech, music or film is moving inexorably towards an all-digital process. Initially this was driven by the need to improve one of the major weak links in the chain, the analogue tape recorder. However it is now becoming obvious that the digital solution in fact brings the benefits of cheapness and speed of working, while perhaps failing to deliver the promised benefits of sound quality. In fact it is a rare recording that is all digital from microphone amplifier to final release.

Why not all digital?

An engineer attempting to make an all digital recording rapidly finds the limitations of the digital signal processing available. For example if an engineer needs to equalise a signal he or she finds that it is perfectly possible to achieve any equalisation curve, and this works flawlessly and effortlessly. However it does not impart any character to the basic sound. Sounds can remain thin and unexciting, often tempting the operator to apply excessive equalisation to “get an effect”. Another example is in dynamics compression. A track can be compressed with all the benefits of digital perfection, including for example look-ahead and off-line level maximisation, but it often still fails to cut through or stand out in a mix.

These problems fall away if the signal is passed through an analogue equaliser or compressor. Each processor has a richness of processing complexity that does more than simply equalise or compress the sound. Each process adds a character which makes a significant improvement in the basic sound of the track being processed. This usually leads to less EQ or compression being needed, preserving the quality of the original sound. Balanced against this of course is the potential for noise and distortion in returning to analogue, and the significant problem of lack of repeatability either for recreating the effect at a later date or for applying a matched effect across 2 or more channels.

How is a typical recording studio arranged to avoid the digital processing problems?

A popular digital recording arrangement is shown in figure 1, in which the recording is made on a disk based audio editing system. It is frequently required to send previously recorded items, singly or in groups, out through a digital to analogue converter and through an analogue processing device to give the sounds the necessary character.

Each path through an external analogue effect adds potential problems with noise, alignment and repeatability as already mentioned. Partly for this reason the original microphone signal is usually fed through an analogue processing stage prior to digitisation. A disadvantage of this initial processing stage of course is that some of the benefits of multitrack recording are sacrificed by excessive pre-processing of the signal. This reduces options later in the recording or mixing process - it is not easy to remove excessive compression later for example.

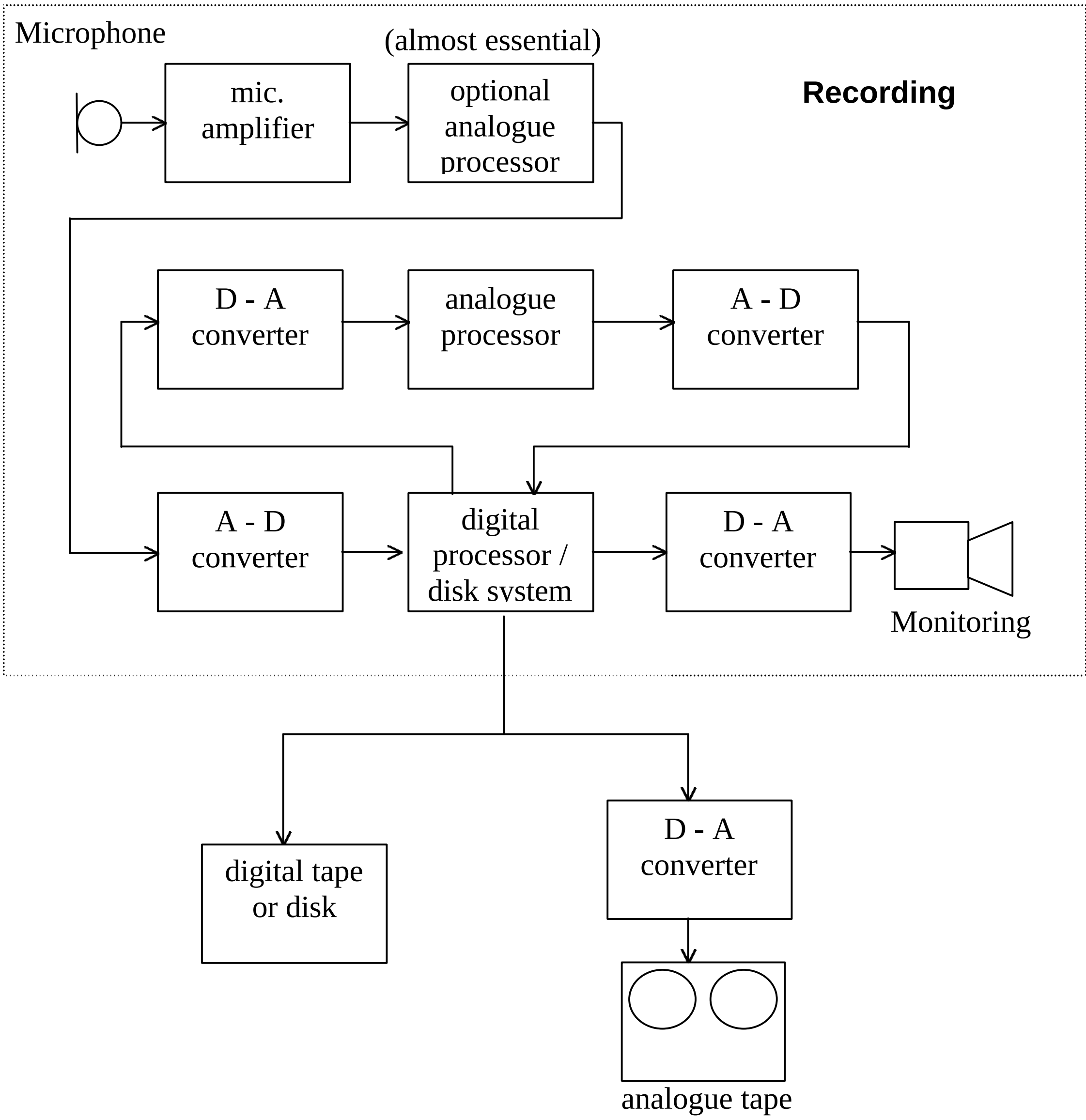

Fig 1: Digital Recording Process: Microphone to Master Tape

The final result of the recording process can either be committed to digital disk or tape for mastering. However it is not uncommon to return to analogue one further time to record the master on analogue tape. This is usually one final admission that digital cannot achieve a desirable sound by itself. The added tape distortion, compression and non-flat frequency response add further character to the sound that is frequently missing prior to this stage.

Mastering

The final stage of transferring a recording to a CD or other distribution format is the mastering process; a typical arrangement is shown in figure 2. Again, a number of analogue conversions frequently take place in the path from the master tape to the release version.

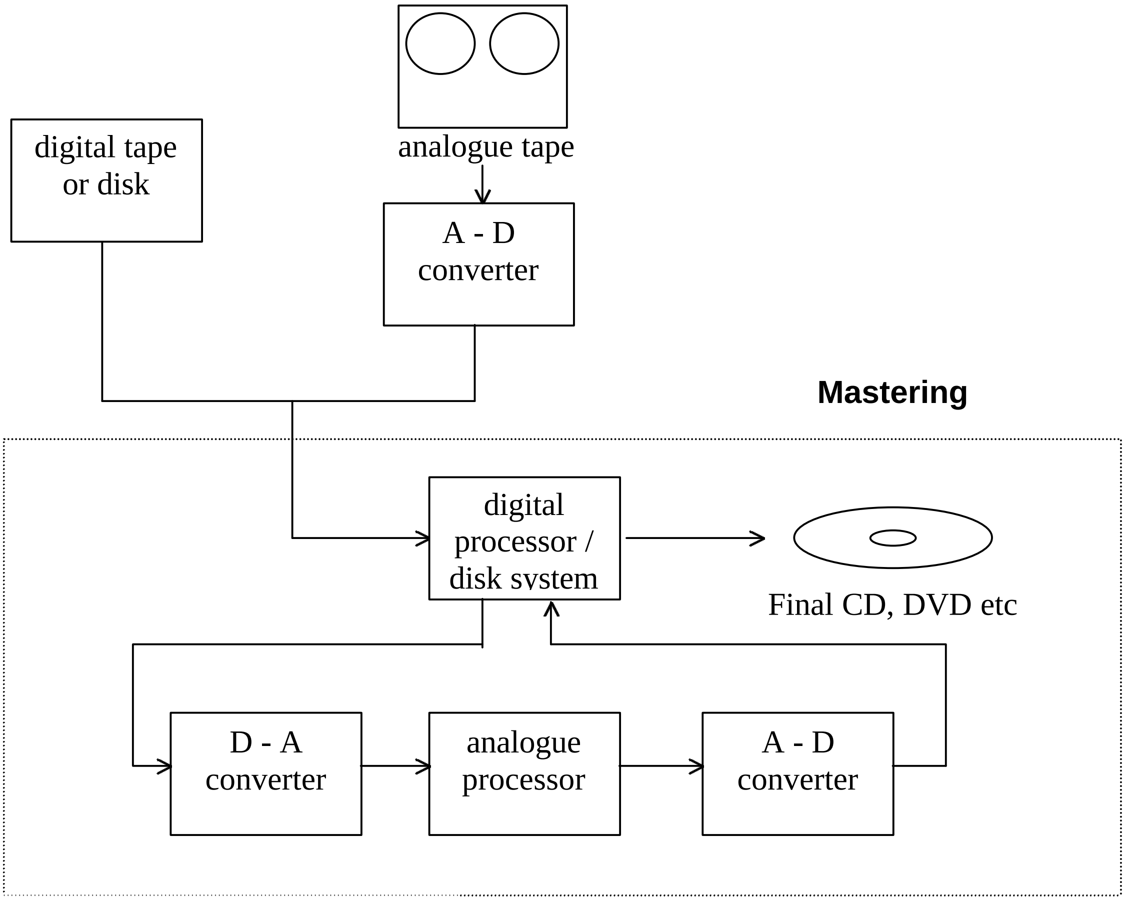

Fig 2: Mastering Process: Master Tape/Disk to CD

Sampling Analogue Processors

Reviewing the large number of conversions between analogue and digital that have taken place in a typical recording leads on to the obvious question: can the effect of the analogue processor be performed in the digital domain? This is the question that Sintefex Audio set out to answer when it began researching into the sampling of analogue processing devices over three years ago.

We quickly realised that a failing of digital processors lies in the simplicity of the algorithms. Many equalisers can be totally characterised with a few parameters. A typical digital peak equaliser requires just 3 parameters to describe, frequency, Q and gain, and rarely more than that in the digital realisation, where a couple of feed-forward and feedback terms describe the entire process.

(To counter any technical objections, I should mention that finite impulse response filters have many more terms, but they are derived from the same three basic parameters by a simple choice of well-know algorithms).

By contrast, an analogue processor has a complex effect on the signal. Choice of circuit arrangement, component specifications, the use of valves (tubes), transistors or other active devices, the use of transformers or inductors, and many choices of gain control elements in compressors, all lead to a varied and difficult to characterise processing effect. Clearly it is possible to make bad sounding analogue processors, but in general those that have survived have done so because they just sound good. It is hardly surprising that digital processing is fatally flawed in its simplicity when compared with the richness and complexity of the analogue world.

It was quickly realised that the tonal qualities of a signal path can be captured by means of measuring its impulse response, but this is limited to linear processes. By contrast, many analogue processors achieve their distinctive sound by being non-linear. To put it another way, audio engineers are familiar with the fact that if you put a signal through an analogue processor at a higher signal level it sounds different to using a lower signal level. This effectively means that the processor is distorting the signal, but in an attractive way, for example unexpected peaks can be controlled by a certain amount of modification rather than simple clipping.

To capture this variation with signal level, we decided to take a whole sequence of impulse responses of the original analogue equipment at a set of different signal levels. We thus achieved a characterisation of the analogue processor that included its variation with level. It was then possible to devise a new form of convolution processor, known a dynamic convolution, where the impulse response in use is varied continuously (or at least sample by sample) as the input signal to the simulation varies in intensity.

The result of these endeavours has been built into the Sintefex Audio FX8000 Digital Audio Effects Replicator.

Figure 3 shows the way Replicator can connected to an analogue processor to sample its processing effect. With Replicator it is possible to sample signal paths and replicate them digitally with both accuracy of frequency and phase response and simulation of variation with level of applied signal. A favourite signal path can be captured and stored for indefinite reuse in the future,

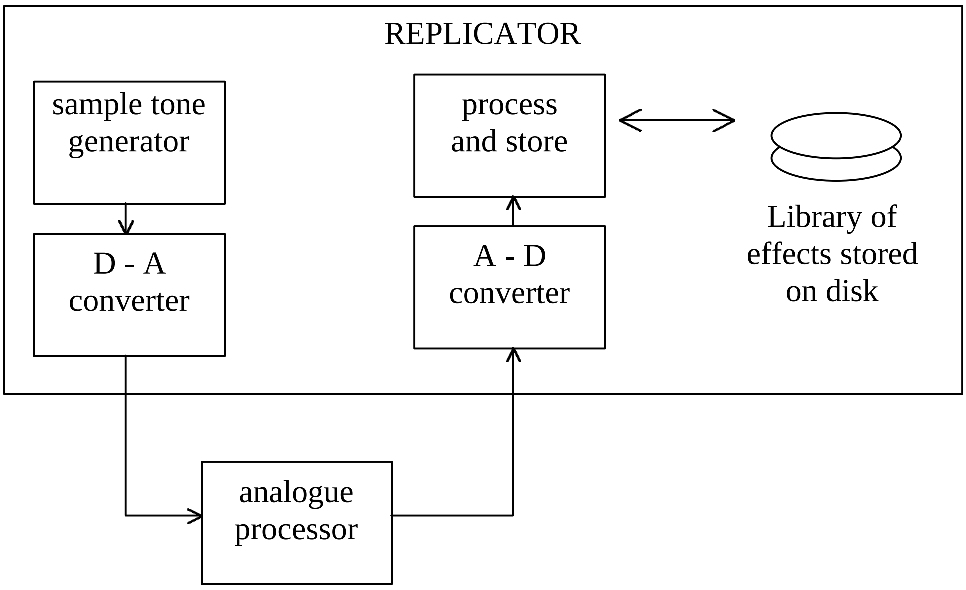

Fig 3: Sampling the Analogue Processor

With Replicator it is possible to apply precisely the same effect across multiple channels for perfect stereo matching and indeed the same process can be applied across all 6 channels of a 5.1 surround sound. Try matching 6 channels of analogue compression or equalisation! In fact Replicator is available with up to 8 channels in a single unit for applications up to 7.1 surround or for multiple independent effects.

Figure 4 shows how the recording set-up can now be organised to use Replicator for providing the complex signal processing traditionally provided by analogue equipment.

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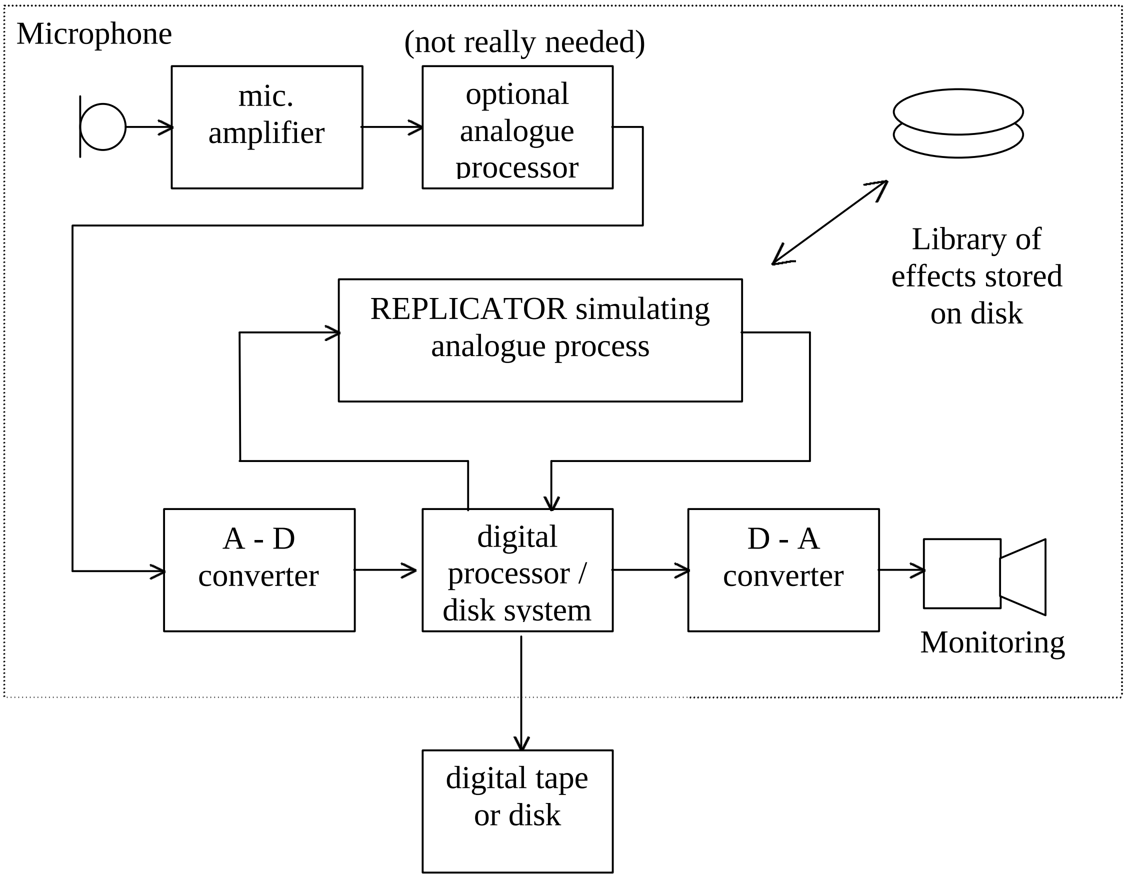

Fig 4: Digital Recording Process with Replicator: Microphone to Master Tape

What can be sampled?

Of course, Replicator had to be refined somewhat from the simple signal path snapshot described above. It is possible to handle dynamic control processors not only by measuring the signal path but also by measuring the compression curves achieved by analogue compressors. The so-called "soft knee" offered by some digital compressors is usually only a programmer's guess as to what might sound nice, but it is now possible to measure the knee of analogue compressors and store these curves exactly. It is also possible for Replicator to interpolate between sampled curves which gives even more possibilities than the original machine.

In addition, we were able to extend the sampling and replication to a "multi-sample" approach. In this we take an equaliser and sample a large number of combinations of the controls' settings. Replicator can then interpolate smoothly between curves, and in this way it is possible to Replicate the entire range of controls of an original equaliser in real time.

It should be mentioned that any signal path that can be characterised by its set of impulse responses can be sampled and replicated. This includes analogue tape and vinyl disk, loudspeaker and microphone combinations and other electro-mechanical devices. Distorting devices like telephones, public address systems and sound reproducing equipment (TVs, radios etc) may be sampled allowing for a library of real sound processing effects. This is useful in film post production where a standard set of effects can be provided with a film for dubbing into other languages in the knowledge that the same audio effects can be applied to the dubbed version as to the original.

Further Possibilities

Of course, once a sample of a signal path is taken it is possible to modify it in ways not possible with the original device. For example it can be pitch shifted to move any resonances in the original sample into tune with a new musical piece, or to achieve a different effect. Any distortion in the original sample can be reduced or eliminated if this is desired.

The Answer is: Analogue Complexity in the Digital Domain.

With Replicator we have discovered that the answer to effective digital audio processing is to sample and replicate the complexity of response of analogue processors, to allow the character of signal processing devices to be applied without needing to continually switch forward and backward from analogue to digital as has been done until now.

This complexity just can't be "programmed in" by clever DSP programmers. A digital processor with algorithmically generated complexity is unlikely to capture the desirable sounds of tried and tested analogue. By contrast the Sintefex Audio Replicator side-steps this by using actual samples of real devices, and building a library of such effects that can be stored indefinitely for future use, as well as allowing your favourite compressor or equaliser signal path to be applied perfectly matched, in stereo or multi-channel formats.

Mike Kemp, Biographical Data

Mike Kemp studied mathematics and computer science at Emmanuel College Cambridge, UK. He founded his first recording business before graduating in 1974, and after a further brief period of research at the university's computer laboratory he left with an MA degree to concentrate on recording. He innovated microprocessor controlled equipment for his studio including a multitrack recorder and a 56 channel desk and also engineered and produced many recordings at his Spaceward Studios in Cambridge.

In the 1980's he branched out into television computer graphics equipment design before returning to audio in 1991 where to further his interest in computer-based audio editing he co-founded Studio Audio and Video Ltd and conceived of the SADiE audio editor.

In 1994 he moved to Portugal where he founded Sintefex Audio to develop his ideas for a digital audio effects processor capable of simulating analogue effects.

He has presented several technical papers to the Audio Engineering Society including aspects of the Replicator design and is the named inventor on a number of patents and patent applications in audio and computer graphics.

Sintefex Audio is dedicated to innovating products for the professional audio market.

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