Oversampling in the Emulator III

by Riley Smith

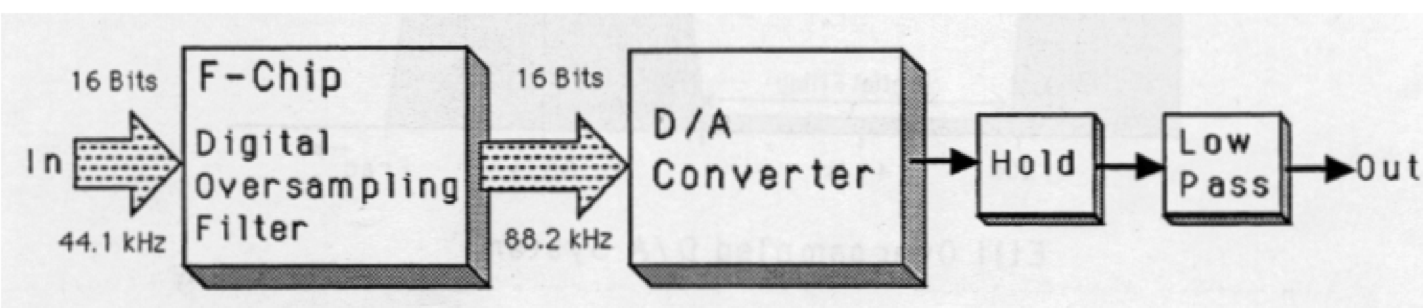
Digital Audio has become quite familiar since the advent of the Compact Disc. Compact Disc players are being billed as the “ultimate in sound reproduction” with “unmeasurable noise and distortion” and that the lowest priced CD player sounds as good as the most expensive. Now that 16-bit sampling instruments have finally become affordable we hear comparisons being made between CD players and Sampling Instruments. “CD quality sampling” and the ubiquitous “96 db S/N” read the promotions of the new 16-bit machines.

Is all 16-bit digital audio created equal? If the differences are not audible, why spend more money than you have to? There may be more to digital audio equipment then the spec sheets reveal.

I had an interesting and educational experience about a year ago when I decided to purchase my first CD player. I had shopped around, compared the spec sheets of the various brands, and settled on a model that seemed to have all the bells and whistles that I wanted. One of the first things I did was to buy the CD version of some of my favorite records so that I could hear all the new detail and sonic clarity. Well, the CDs did sound cleaner and I sure didn’t miss those clicks and pops, but there seemed to be something else missing. The music from the CD player seemed to be coming from behind a thick curtain, while the music from the record seemed to be alive in the room. What was wrong with this audio picture? I asked an audiophile friend of mine and was told that the problem was the output filtering on my new CD player and that I should try another brand. He recommended I look for one that used a technique called “oversampling” which effectively doubled or quadrupled the output sample rate and required less output filtering. I took the CD player back to my dealer to compare it with the same model and with other brands. No, the CD player wasn’t defective but when I A/B’d it with another brand (one that used over-sampling), the difference was like night and day. I returned home with the oversampling unit and have been happy ever since. Since my ears could easily tell the difference between the two units and since the noise and distortion specs were virtually identical, I concluded that there must be types of distortion or coloration that do not appear in standard spec sheets.

Sampling instruments are not CD players although they do have similarities. One of the differences between a CD player and a sampling musical instrument is that on a CD player the sample rate stays at a constant 44.1 kHz while on a sampling instrument, the sample rate changes as a function of the pitch of the sound.[1] This at once creates problems in that it is much easier to design a good fixed frequency reconstruction (lowpass) filter than one that can track a variable sample rate. The reconstruction filter must have a slope which is steep enough to remove the “staircase voltage” harmonics created by the D/A process without seriously cutting into the audio band and at the same time be “quiet” enough to be considered for use in a “no compromise” 16-bit system. We have found that when a four-pole reconstruction filter is used (as is common on 12-bit machines), most users will use less filtering than needed to remove the “staircase voltage” or “clock” noise in an attempt to preserve the high frequency content of the sound. This results in a harsh, grainy sound as well as excess background noise.

The Emulator III uses custom designed oversampling filters and custom low-noise analog tracking filters to deal with the problem of output filtering. The oversampling filter chip actually doubles the sample rate by examining a number of samples on either side of the current sample and inserting a new sample at the appropriate level. This means that a sound sampled at 44.1 kHz played back at its original pitch will actually be output at 88.2 kHz and if transposed up one octave will appear at the output at a 176.4 kHz rate!



1 'It should be noted that some sampling instruments shift pitch using a constant sample rate, but since most of these methods cause serious distortion we will not consider them for use in a high quality system.

The use of oversampling has several important advantages in a sampling instrument.

First, because the quantization noise is distributed over an area which is twice as great, the residual noise within the audio bandwidth is only one half of the original figure. This adds 3 dB to the signal-to-noise specification.

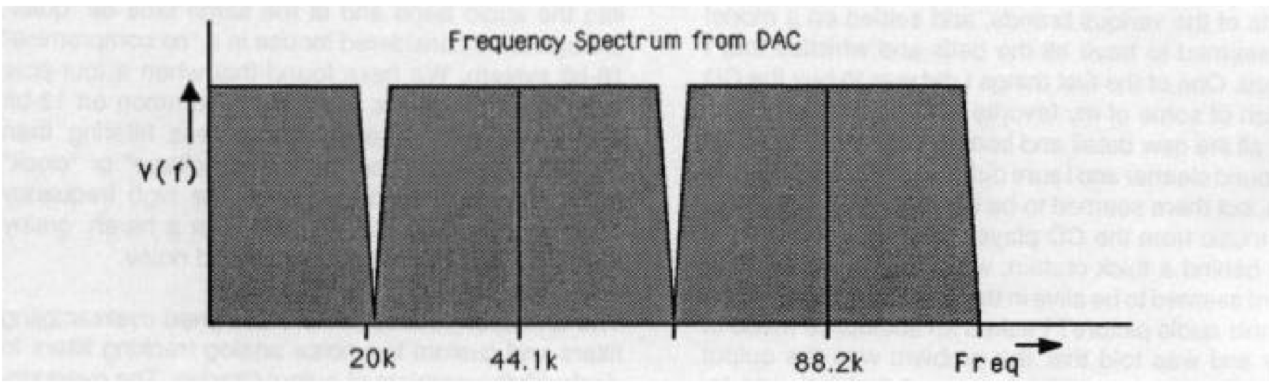
Second, when using sampling instruments, the pitch of a sound is often shifted down an octave or more to create an unusual sound effect. Normally when this is done, the sample rate begins to enter the audio spectrum and it becomes audible. By use of oversampling the sample rate does not begin to enter the audio range until it has been shifted down two octaves.

Third, since the sample rate is now an octave higher, so are the “staircase voltage” harmonics. They can now be filtered out with a much simpler filter (i.e. one with much less phase distortion and ripple). The oversampling filter can thus be thought of as a zero phase-shift digital jitter. Phase distortion is accountable for a lifelessness or lack of presence for which digital audio is often criticized.

The analog filter used in the Emulator III is a custom designed, 7-pole, voltage controlled filter with a signal-to-noise ratio of greater than 96 dB. Three poles are used for reconstruction and the remaining four poles are used for traditional synthesizer type signal processing. The reconstruction filtering is automatic and eliminates the audible artifacts of the A/D, D/A process.

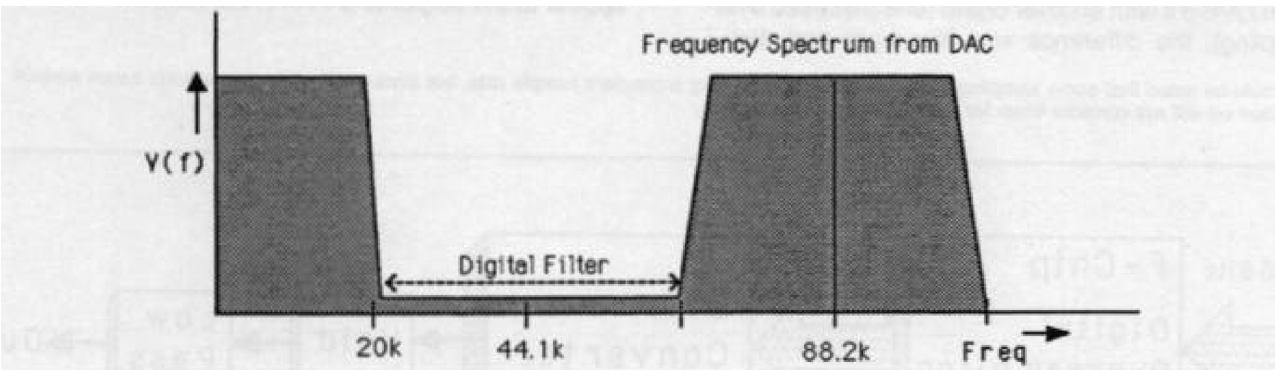
The combination of a phase-linear digital filter with a high performance D/A converter and simple phase-linear analog output fitter results in a nearly ideal sampling system for digital audio, the Emulator III.

The frequency plot below shows the frequency spectrum generated from a normal D/A converter system. Note the harmonic frequencies above the audio range that must be filtered out to avoid production of audible intermodulation products.



Normal D/A Converter System

The frequency plot below shows the frequency spectrum generated from the Emulator III D/A converter system.
The first harmonics appear far above the audio range at 66 kHz, which requires much less filtering to remove.



EIII Oversampled D/A System