Bob Moog

sampling instruments, part 4: more listening tests for digital samplers

MY LAST COLUMN (Nov. '85) presented three quick listening tests that you can perform in order to shine an aural spotlight on the performance of a digital sampling instrument. Here are a few more that you can perform with nothing more than a simple programmable synth (preferably an analog model), a good monitor system, and your ears:

Test No. 4. Set up your test synth to produce a single sustained tone of low harmonic content (sine or triangle wave), at a frequency of about 4kHz (four octaves above Middle C). Sample a couple of seconds of the sound, in strict accordance with the instrument manufacturer's instructions. Connect the instrument's output to a high-quality monitor system's input.

Next, press the key that plays back the sound at its original pitch. Adjust the monitor gain level so that the sound is clearly audible, but not loud. If you listen carefully to the sound, you will probably hear (a) the original tone, (b) some extra brightness or harshness coming from aliasing and distortion, and (c) some unpitched, hissy noise.

Now, here's the test: Press two keys simultaneously—the key that plays the sound at its original pitch, and the key immediately below it. You will hear the two high tones, but you will probably also hear a third tone at a pitch somewhere near Middle C. This lower tone is a difference frequency, or sideband, resulting from intermodulation between the two pitches. Intermodulation causes muddiness in the sound quality, and increases with the number of notes that you play.

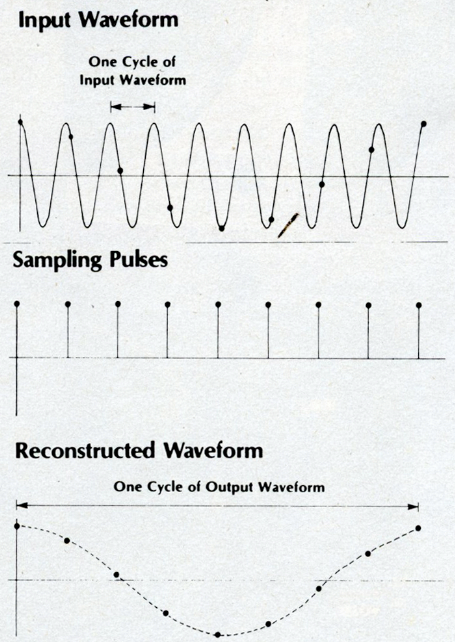
Very high-quality sampling instruments often have a completely separate digital-to-analog channel for each individual sound that is produced. Separating the tones during the conversion process minimizes or eliminates intermodulation. Instruments with a single D-to-A converter, on the other hand, are much more likely to produce intermodulation. Sometimes intermodulation occurs because of the way the instrument's software works, and sometimes it occurs due to limitations on the part of the circuitry itself. You can't tell how the intermodulation is being produced by just listening—but you can sure tell whether or not it's there.

A warning is appropriate here. Since this is not an A/B test (that is, you're not comparing the sampled sound with the original), you should be aware of the possibility that the monitor system itself is generating intermodulation. To check for this, you should set the sampling instrument's output level control fairly low, to make sure that you're not overloading the monitor system at its input. Then, if you hear intermodulation, turn up the monitor level so that the loudness increases by a moderate amount. If the intermodulation seems to get louder at a faster rate than the original tones, then your monitor system is probably introducing the intermodulation.

Test No. 5. Set up your test synth to produce a single sustained tone of about 1kHz (two octaves above Middle C), with a sine or triangle waveform. Put a very short envelope on it: attack time as short as possible, decay time of about 50 milliseconds, and zero sustain level. The sound should be as short as that of a woodblock. Again following the instrument manufacturer's instructions, sample that sound.

Now connect the sampling instrument's output to one of the monitor system's inputs, and the output of your test synth to another. Play one instrument and then the other. Listen carefully to the differences between the envelopes of the two sounds.

Fig. 1. If the input signal is in the same frequency range as the sampling rate, low-pitched alias tones will be introduced when the original waveform is reconstructed at the instrument's output.



Many sampling instruments use compansion (compression during recording, expansion during playback), either in a hardware circuit, or in a software algorithm. Compansion is an effective way to reduce the audible noise of a sampling instrument. (All standard noise reduction systems, for instance, employ some form of this technique.) Adjusting the response times of the compressor and expander, however, is a tricky design job involving lots of tradeoffs and experimentation. Asking a sampling instrument to process a very percussive waveform is a good way to test the overall response time of the instrument. An instrument that accurately reproduces a slowly varying tone (like a horn or flute sound) might do a less-than-adequate job on a sound with a sharp attack and a fast decay.

Test No. 6. All sampling instruments have some sort of lowpass filter with a sharp cutoff, intended to remove all frequency components above one-half of the sampling rate. Without this filter, wide bandwidth signals will introduce alias tones (see Fig. 1).

A good test for the sharpness of this filter is to sample a tone that has lots of high harmonics. If your test synth is analog, open its lowpass filter all the way and set the waveform to the skinniest rectangle available. (If your test synth if digital, it may not be capable of producing very high frequencies. Check the specifications to see what its bandwidth is.) At any rate, the idea is to set up a clean but very bright, thin tone whose harmonics go up to at least 15-20kHz. The fundamental pitch of the tone should be a couple of octaves above Middle C. A little slow low-frequency modulation (a wide vibrato, for instance) will help to reveal the audible aliasing.

Sample the above-described tone. A/B the sampled tone with the original. You will probably hear some frequency components in the sampled sound that are not in the original. If you have set up slow low-frequency modulation of the test tone, you will hear the extra frequency components of the sampled sound going up and down in pitch differently from the pitch of the original tone.

Now take a few more samples of the same tone. With each succeeding sample, close the test synth's lowpass filter a little bit. Listen carefully to the samples. At one setting of the filter, you will probably hear the alias tones decrease markedly. If this happens, then you know that the sampling instrument's filter is letting through frequency components that are too high for it to sample cleanly.

This concludes my list of listening tests that you can perform on any sampling instrument. All of the tests are qualitative. That is, they won't help you check the instrument against published specifications. However, they will enable you to hear distortion components, one at a time. When you sample a "natural" sound from the outside world, or sample complex electronic tones, your sampling instrument will probably be generating several types of distortion in small amounts. By understanding what the sources of the distortion are, you can often make adjustments to the harmonic content or amplitude level of the original sound in order to reduce distortion and improve the overall sound.

Bob Moog designed and built the first commercially successful synthesizer.