The 1-Bit Advantage – Future Proof Recording

Korg has developed and is introducing a line of mobile digital audio recorders — the first in their class to utilize 1-bit audio recording. The hand-held MR-1 is capable of high quality 1-bit/2.8 MHz recording and playback, while the tabletop MR-1000 delivers up to 1-bit/5.6 MHz, doubling industry DSD recording quality standards. An integral part of these 1-bit digital recorders is Korg's exclusive AudioGate™ software, included free with each unit. AudioGate provides the compatibility between 1-bit recording made on the MR-1 and MR-1000, and other current 1-bit and PCM audio formats.

In order to discuss the capabilities and benefits these recorders offer, we have prepared this digital audio primer. In it, we will explore and discuss current PCM digital audio techniques and some of the principles and advantages that 1-bit recording provides, both in terms of fidelity and archiving for future use.

Capturing audio in 1-bit / 5.6 MHz enables recordings to be archived in the highest quality ever available to professionals and consumers, and allows the re-purposing and distribution of archived recordings into any current PCM format types while preserving the original pristine 1-bit audio files for upcoming future formats.

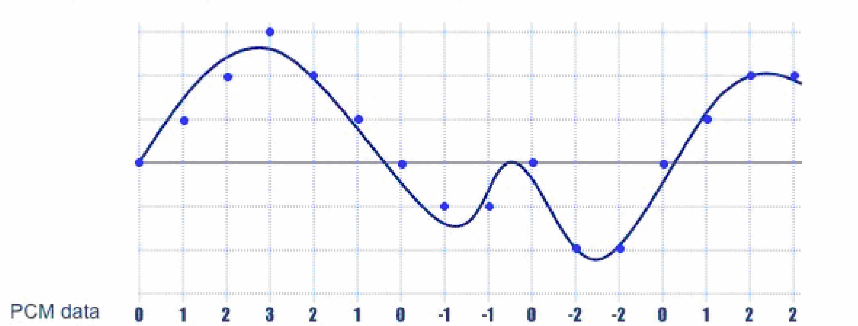
1-bit recording technology is not new to the pro audio world. It was originally developed by Dr. Yoshio Yamasaki in the late 80's at Waseda University in Japan. Dr Yamasaki patented the 1-bit process in 1992, and has written history with his research and developments with 1-bit technology. The technology has been adopted and promoted by Sony and Phillips as Direct Stream Digital recording (DSD) in their SACD configuration, and these 1-bit SACD's have been commercially available since 1999 with over 4,000 titles available.

Digital Audio – a brief history

In typical analog reproduction systems, the fidelity and dynamic range were limited by the physical medium they were stored on – tape, vinyl records, etc. In addition, each pass of a tape across the heads and capstan, or each time a stylus ran through the groove on a record, the original was degraded by this mechanical contact. Tapes and vinyl had to be stored in a specific environment, and tapes had to be kept free of any magnetic fields – such as speakers or computer monitors. They were expensive, their storage capacity was limited, and they continue to degrade with every passing year.

Enter digital audio. In the fall of 1982 (spring of 1983 in the United States), the Compact Disc introduced digital audio recordings to the consumer market. Digital audio offered a number of improvements over analog audio systems – namely the higher fidelity and increased dynamic range provided by non-mechanical reproduction; and greater storage and archiving abilities

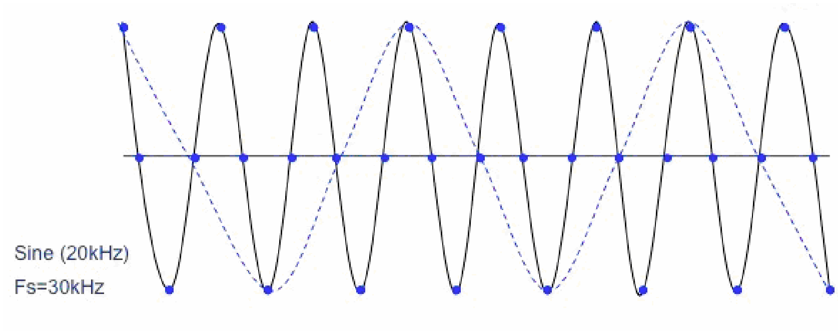
So what is digital audio? Basically, digital audio analyzes a continuous audio waveform by taking "samples" of points along this curve. When we refer to sampling – converting analog audio signals to digital data – we are generally discussing two specifications: bit-depth and sampling frequency.



In the example above, the blue dots represent the sample points taken of this audio signal. The vertical lines represent the sampling frequency, or rate. This is defined in hertz, representing how many "readings' are taken each second. The horizontal lines represent the bit-depth, or resolution. The first thing you will notice is that the dots do not accurately represent the original signal. This is because the resolution in this example is quite low.

The CD that we began this conversation with was recorded using a bit-depth of 16, and the sampling frequency is 44.1 kHz (commonly called 16-bit / 44.1 kHz). Before we go any further, let's look at why those numbers were chosen. The human ear responds to audio signals ranging roughly from 20 Hz (twenty cycles per second) to 20 kHz (twenty-thousand cycles per second). Research has proven that in order for sampling to be done accurately the sampling rate must be slightly more than double the frequency of the highest note being sampled. This is often called the Nyquist theorem.

In the graphic below, the sampling rate is only 30 kHz, while the frequency of the audio signal is 20 kHz. The dotted line shows the signal created from the sampling error. These errors are referred to as “aliasing”noise.



As we mentioned, the human hearing range tops out at about 20 kHz, and the Nyquist theorem shows that we need to sample at slightly faster than double that. This is how CDs ended up using 44.1 kHz as their sampling frequency. It is the lowest rate that assures accurate sampling through the entire hearing range.

Sampling bit depths are expressed in powers of two – eight, sixteen, twenty-four, etc. Deeper bit rates provide more detailed definition in the audio, increasing the accuracy and cleanliness of the lower level signals before they get lost in the “noise floor”. This means that the dynamic range of the system is increased by pushing down the range – loud, or full-level signal are easier to represent. It is the quieter signals, and the decaying into silence where the inaccuracies of most systems become apparent. Increasing the bit depth from 8 to 16 produced dramatic results, increasing the theoretical dynamic range from 48 dB to 96 dB.

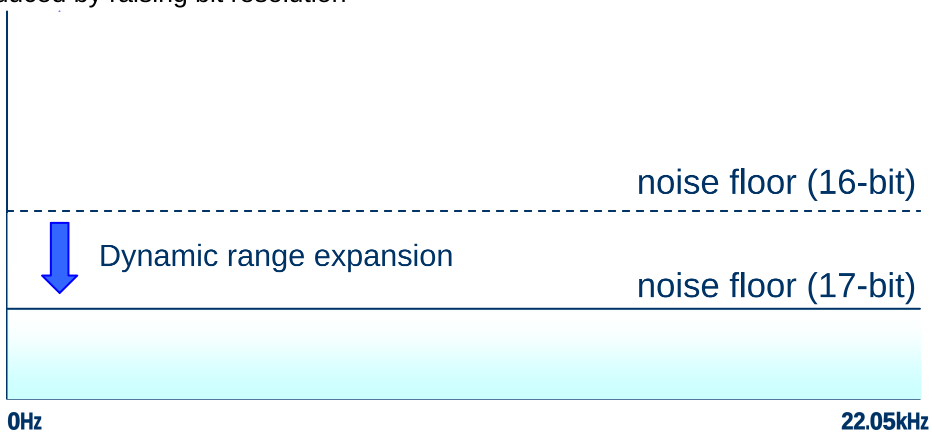
16-bit resolution represented a significant increase in dynamic range over analog recording mediums, which were typically 50 – 60 dB. Eight and twelve bit systems did not offer this advantage, so sixteen was adopted for the CD format. Keep in mind that at the time, memory was still at a premium and the idea was to fit as much data as possible on a CD while keeping it affordable as a consumer entertainment medium.

Advances and Issues in Multi-Bit Science

Almost as soon as the CD was introduced, new sample formats began to proliferate, offering deeper bit depths and faster sampling times. One could assume that higher sampling rates coupled with increased bit depths would improve the quality of sampling/recording. And they do in many ways. The increase from 16 to 24 bits delivers a practical dynamic range of approx. 110 dB, which while significant, was a markedly smaller increase when compared with the move up to 16-bits. So while each increase in bit depth produces a real expansion of dynamic range, the improvements are getting smaller each time.

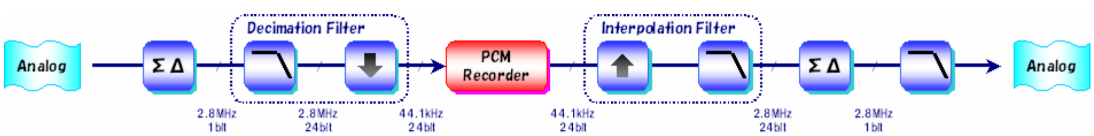
Quantization Noise

- Distributed equally in the whole band (white noise)
- Reduced by raising bit resolution



There is no doubt that current 24-bit / 192 kHz audio sounds very good. But there are still areas that can be improved, and approached from a different perspective with other unique benefits.

The Multi-Bit PCM Encoding Process



- A/D & D/A conversion is done in low-bit (e.g.1-bit) at high sample rates, then one in X samples is kept (the decimation process)
- Decimation occurs during recording (i.e. A/D conversion) then interpolation and sigma-delta modulation during playback (D/A conversion)

Surprisingly, most current 24-bit converters actually use 1-bit conversion at the front end already. After capturing a high-speed 1-bit stream the converter uses what is called a Decimation Filter to change the 1-bit data into the desired multi-bit format. A simple explanation of the decimation filter is that it is a form of sample rate converter or divider which parses the 1-bit stream into the needed number of samples for the multi-bit format. Meaning it throws away sample information (in an intelligent fashion, of course) that is not able to be used.

It must also contain a filter at half of the sample rate (Nyquist theory) to eliminate aliases - i.e. at 44.1 kHz a filter of 22.05 kHz is employed. Since the design of the filter *is* affecting the audio, considerations of phase, linearity, transient response and ripple are all subjective to the math being done, which in turn is the responsibility of the person who wrote the code for the decimation filter. This also may be compared to how the same microphone sounds different in two different mic preamps.

In the D/A process the PCM data stored is manipulated again to re-conform the data to audio (actually it's still technically voltage till it gets to a speaker). During this process, more calculations have to happen to "reassemble" the data to an audio stream, including more estimation processes try to reconstruct the audio as close to how it was when it was originally captured. These include interpolation to reconstruct a more detailed approximation of the original analog signal, and sigma-delta modulation to control the inevitable noise and errors which occur when estimating the info.

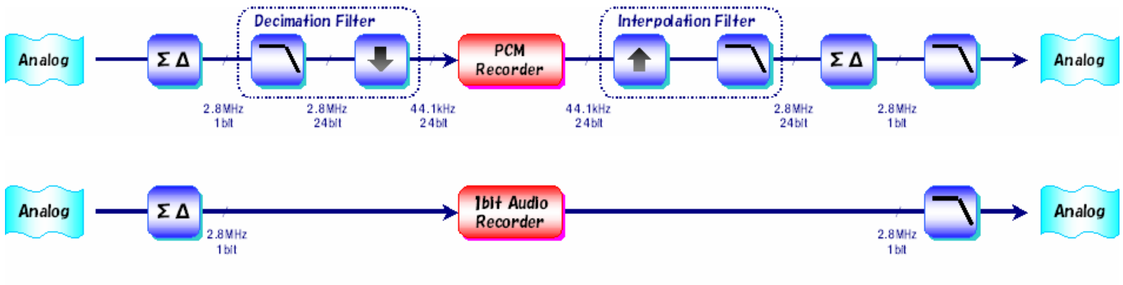
A common practice is to use oversampling for playback to double the sample rate of a source by "estimating" where the data would be if it had recorded at say two times the rate. Although it can be effective, it is still an estimation and is nowhere near as good as capturing and retaining all the information in the first place.

So you can see that in this design the data has changed, or been manipulated a minimum of two times from the original - at encoding and then again at decoding.

The 1-bit advantage

In a 1-bit system, the audio is recorded at super-high sample rates, commonly 2.8224 MHz, and now with the MR-1000, 5.6448 MHz. At this high rate a 1-bit system is able to reproduce frequencies from DC up to 100 kHz, which exceeds all other digital systems and even magnetic tape, which can reproduce up to 50 kHz. At these high rates there is no longer a need for steep filters, which removes a possibly coloring element in the encoding chain.

1-Bit Encoding Compared to PCM



- Records the original 1bit signal directly
- D/A conversion can be as simple as running the pulse train through an analog low-pass filter

Even better, by remaining in the 1-bit format that the converter already used, there is no need for the decimation filter process during recording, and no need for the interpolation and oversampling filter processes during playback. So what comes in goes out, with no extra math in the process, thus eliminating any need for the data/audio to change. Remember, decimation filters and their design have a major influence on the sound with PCM recording, and 1-bit recording eliminates the need for them.

Less Is More

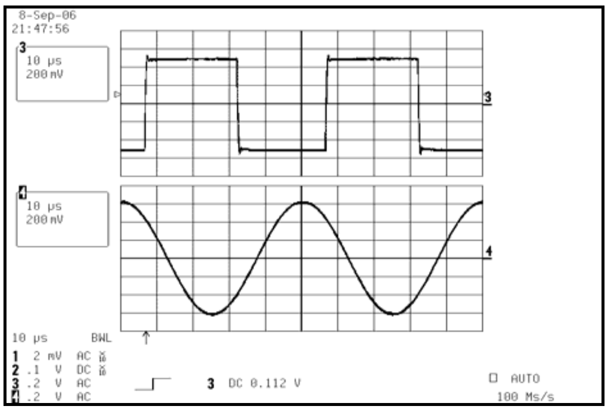
It's easy to follow the benefits of higher, and super-high sampling rates. But what might at first confuse the reader is the benefit of moving from high bit rates down to 1-bit. Surely the increased resolution of higher bit rates must be more accurate, right?

The basic concept here is that once the sampling rate is increased to such high levels, each step doesn't need to be defined with such detail. With such frequent readings of the current state of the audio waveform, each step need only be defined is the simplest of terms – has the signal increased since the last step, decreased, or remained the same. 1-bit offers only two values, a 1 or a 0. Either up from the previous sample or down. And at these super-sampling speeds a steady state can be represented by alternating 1's and 0's. The chance for error in such a system is much less than in multi-bit approaches. Consider this:

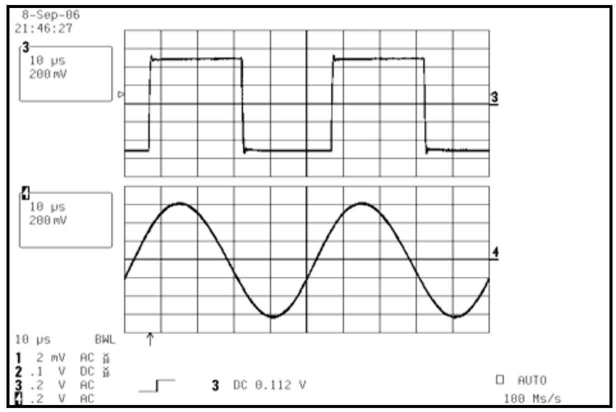
In a 1-bit system the possible values for each measurement are simple, either it's a 1 or a 0 - there's little chance to get a wrong value. In a 24-bit system there are 16,777,216 possible values. So which system is more likely to be accurate for each reading?

As practical proof of this, pictured below are the results of tests that were performed using the Korg MR-1000 showing the real advantage of 1-bit / 5.6MHz recording. We recorded an analog 20 kHz square wave signal at various sample rates and captured its analog outputs.

The top square wave is the original input signal, the bottom is the output. Each picture shows the analog input signal and the output at the given bit resolution and sampling rate. We can see the output signal using 16-bit / 44.1 kHz becomes a sine wave. Even 24-bit / 96 kHz changes the signal into a sine wave.

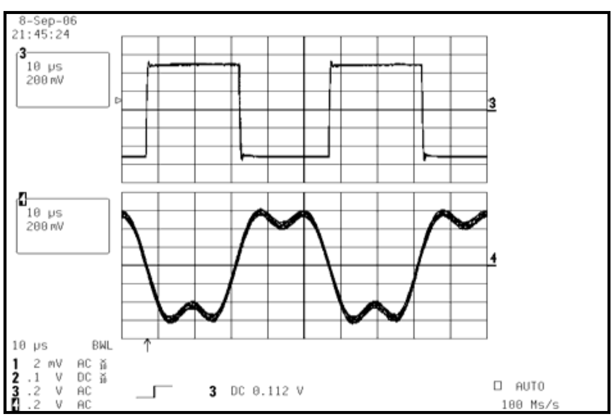


16-bit / 44.1 kHz

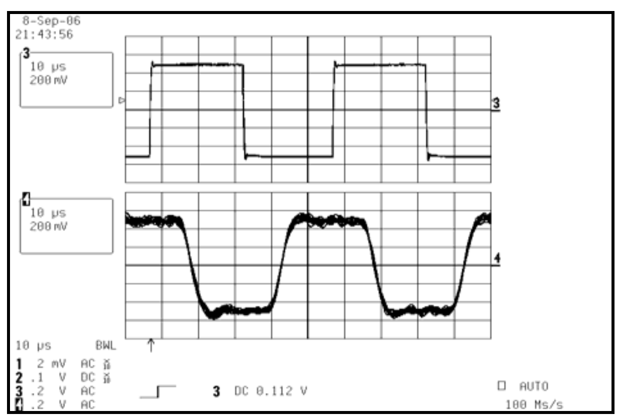


24-bit / 96 kHz

24-bit / 192 kHz got closer to the original, but as you can see, the closest was 1-bit / 5.6 MHz which most accurately captured the instant transient of the square wave.



24-bit / 192 kHz



1-bit / 5.6 MHz

This is a notoriously difficult “torture test”, but it clearly shows the beneficial performance of high-speed 1-bit technology.

KORG®

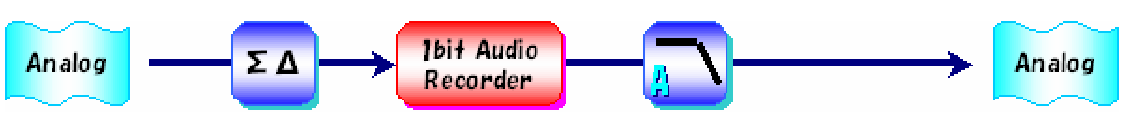
The True Importance of 1-bit Technology – Future Proof Recording

We have spent all this time getting you familiar with this science not to convince you that your current recording systems are no good. As we have stated earlier, current 24-bit / 192 kHz recording produces a very good result. And while the benefits of 1-bit are real, our primary contention is that it is the first format since analog tape worthy of being used as a final mixing and archiving solution. Most experts agree that tape is still the desired final mixing medium when possible. But it is not a perfect archiving format, since it degrades over time, and the potential for players to remain in existence for decades to come is unlikely.

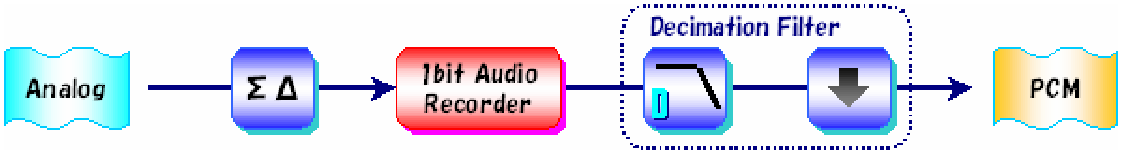
Multi-bit PCM produces good results, but it has not caught up to the 5 Hz - 50 kHz performance of tape. And it is not easily transportable to other formats when considering future possibilities. Taking a current 16-bit / 44.1 kHz file for use in remastering the project in 24-bit / 192 kHz format will produce little perceived benefit, or improvement. You can't improve on the quality of the archived material unless it has captured ALL the nuances and accuracy of the original audio. That is why serious remastering of classic recordings always goes back to the original master tapes – they have the best dynamic range and frequency response and don't require sample rate conversion of already manipulated data. If they were more "rugged" they'd be fine, but they are degrading with each pass across the heads of the machine, and with each year that passes.

Data reduction and filtering had been the main goal in PCM up until now. But the times have changed. Disk storage and memory are very inexpensive, with hundreds of Gigabytes the norm nowadays. Data rate capability and chipsets are readily available that can handle 1-bit data streams with ease. So the only reason we are so complacent with PCM is that there have not been any reasonably-priced 1-bit recorders. Until now.

The Benefits of 1-Bit Recording



- The ability to record and reproduce sound which is very close to the original analog signal

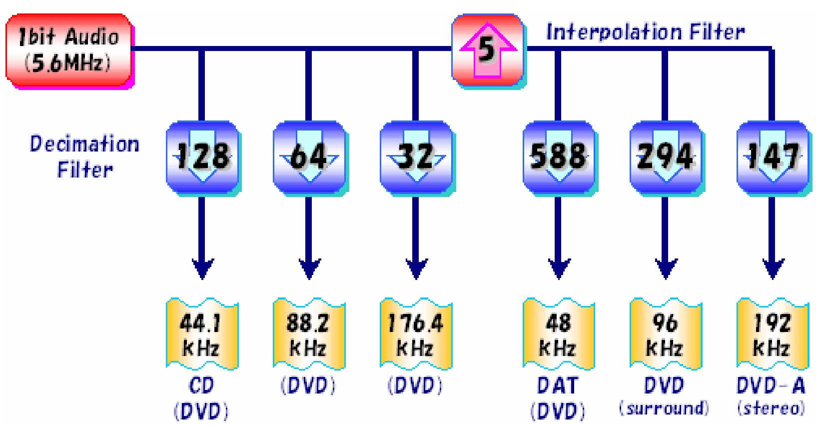


- Useful as an archive whose output can be any format of PCM

Now, with affordable 1-bit recording, you can choose to capture the audio with the best resolution and accuracy available, and archive it without the manipulations that must occur in the decimation and later interpolation processes. Your archive is not colored by these steps, and you can then choose the conversion process of your choice for your current project. As converter technologies continue to improve you can go back to this archive and remaster the project starting from a “pure” source. In this way you’ll only be manipulating the audio one time, through the system of your choosing for the given project.

Both MR recorders use state-of-the-art converters; the Burr Brown PCM4202 for A/D conversion, and the Cirrus Logic CS4398 for D/A conversion. These are well-respected audio devices worthy of your most important recordings.

1-Bit Audio as an Archive Format



- Capable of being converted to any current PCM standard

1-bit audio can more easily be converted into all the current possible formats needed for projects and use today than multi-bit PCM. If you record your final mix or master archive in 1-bit, then moving to any of these current formats is possible, with currently acceptable results. And if the industry moves more towards 1-bit as format your archived projects will be ready. No other available archiving format past and present can make this claim of data reliability and performance.