Digital Theory

Analog vs. Digital

- Digital recordings are “cleaner” because they add almost no noise or distortion to the input signal.
- Analog recorders add a little “warmth” to the sound (tape compression, bass boost, tape hiss, and print-through)
- Digital is less expensive, smaller
- Random-access to locate a particular section of the recording quickly

A/D and D/A

- Analog to Digital
- Digital to Analog

Digital Recording - How it works

1. Signal from the mixer, preamp, or audio interface is run through a lowpass filter (anti-aliasing filter), which removes all frequencies above 20kHz.
2. Next, the filtered signal passes through an analog-to-digital (A/D) converter. This converter measures (samples) the voltage of the audio waveform several thousand times a second.

Digital Recording - How it works

3. Each time the waveform is measured, a binary number (made of 1's and 0's) is generated that represents the voltage of the waveform at the instant it's measured. This process is called quantization. Each 1 and 0 is called a bit, which stands for binary digit. The more bits that are used to represent each measurement (the higher the bit depth), the more accurate the measurement.

4. These binary numbers are stored on the recording medium as a modulated square wave recorded at maximum level. Numbers can be stored magnetically on a hard disk.

Digital Playback - How it works

1. The binary numbers are read from the recording medium, such as a hard disk.
2. A digital-to-analog (D/A) converter translates the numbers back into an analog signal made of voltage steps.
3. An anti-imaging filter (lowpass filter, smoothing filter, reconstruction filter) smoothes the steps in the analog signal, resulting in the original analog signal.

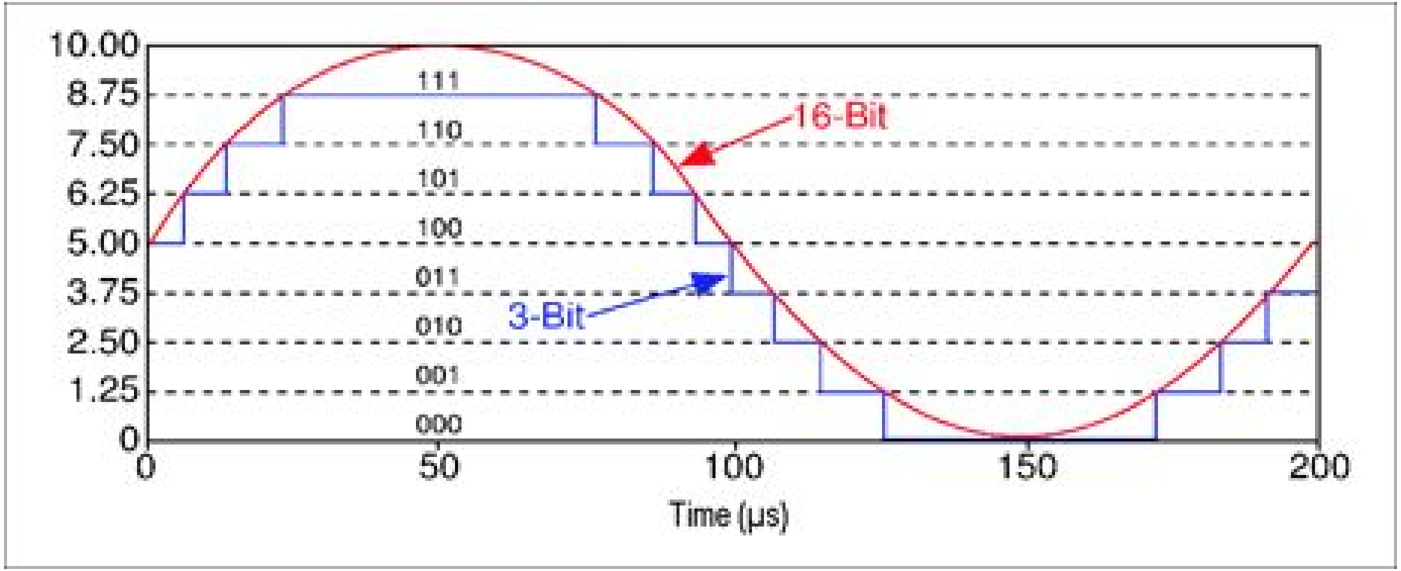
Bit Depth

- Audio signal measured many thousand times a second to generate a string of binary numbers (called words).
- The longer each word is (the more bits it has), the greater the accuracy of each measurement.
- Short words give poor resolution of the signal voltage (high distortion); long words give good resolution (low distortion).
- Bit depth or resolution are other terms for word length.
- 8 bit, 16 bit, 24 bit, 32 bit
- More bits sound smoother and more transparent, but need more disk storage space.

Bit Depth

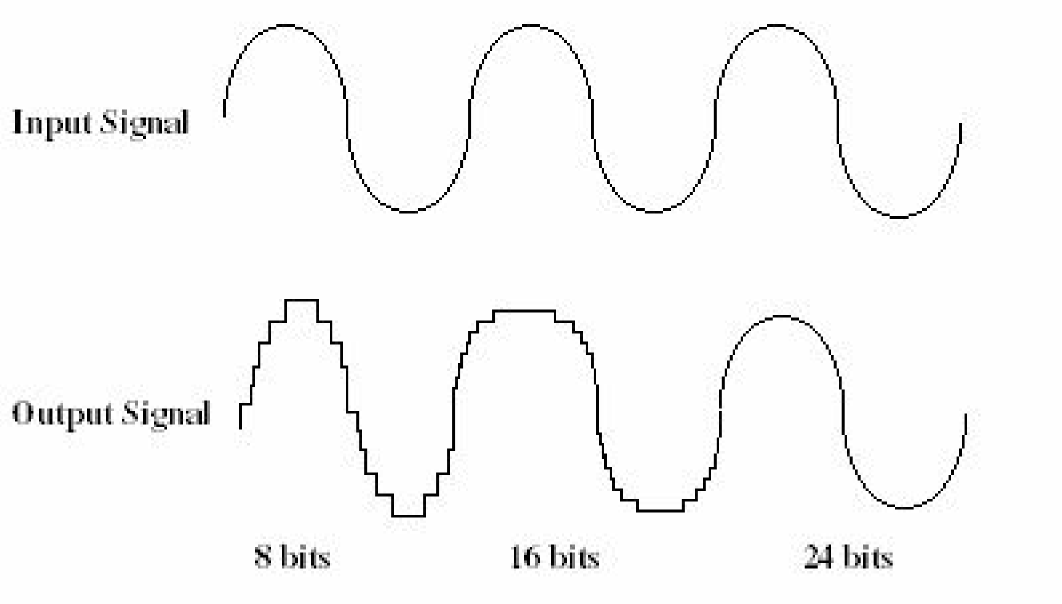
- Even though CDs are a 16-bit format, they sound better when made from 24-bit recordings
- Dither: added during mastering to a 24-bit recording before exporting to a 16-bit recording. The dither helps the 16-bit recording sound more like the 24-bit recording in other words, dither lets you retain most of the quality and resolution that you recorded at 24 bits, even though the recording ends up on a 16-bit CD.
- Bit depth affects the dynamic range, noise and distortion

Bit Depth

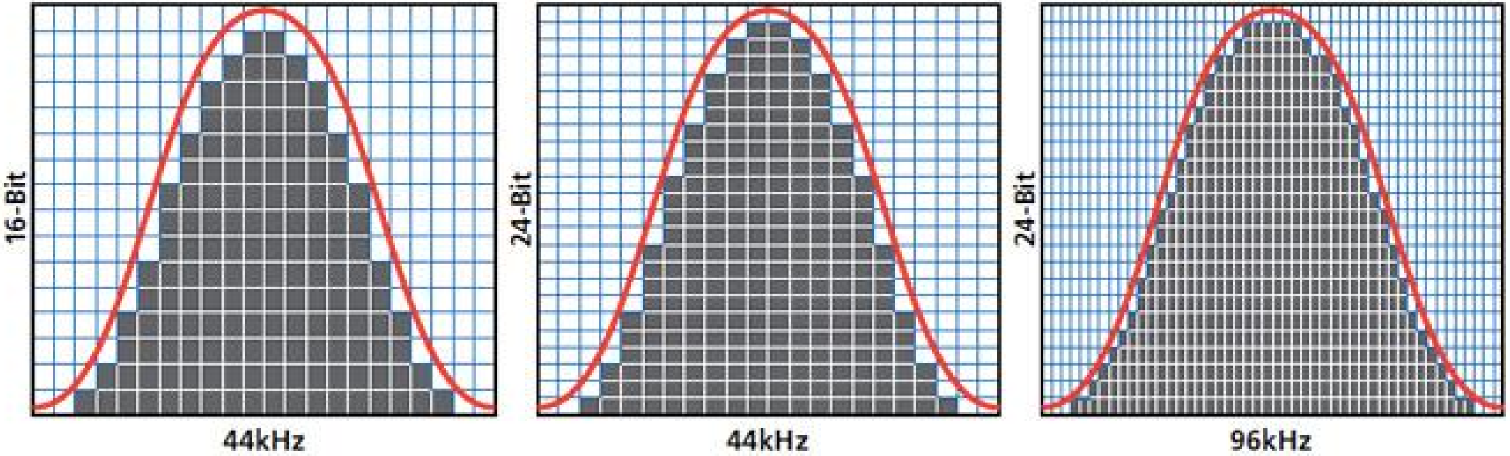


10.00
8.75
7.50
6.25
5.00
3.75
2.50
1.25
0
111
110
101
100
011
010
001
000
16-Bit
3-Bit
0
50
100
150
200
Time (µs)

Bit Depth



Sampling Rate & Bit Depth



As Bit depth and sample rate increase, more information is captured, this resulting in higher quality audio.

Sampling Rate

- The rate at which the A/D converter samples or measures the analog signal while recording
- Example: 48 kHz is 48,000 samples per second; 48,000 measurements are generated for each second of sound
- The higher the sampling rate, the wider the frequency response of the recording
- Compact Discs use a 44.1kHz sampling rate
- 44.1, 48, 88.2, 96, 192
- Sampling Rate affects the high-frequency response
- Higher sample rate requires more storage space

Bit Depth comparison

https://www.youtube.com/watch?v=7Tl9B_hX3cM Mario Brothers 8 bit

https://www.youtube.com/watch?v=52bxoqqgDWM Dubstep 8 bit

https://www.youtube.com/watch?v=BStjuHfP238&list=PL0u92-Pq732uRFXP1NMybMHvzCrSuT-vz&index=2 Zelda 16 bit

https://www.youtube.com/watch?v=4m3E2D8dDpE&index=25&list=PLgCeMwFu0QPsUpMWvk8RwPgs-ZYLW50F9 Supafly 24 bit

Sampling Rate Comparison

https://www.youtube.com/watch?v=96jFvdteqWI

Sampling Rules

- Nyquist Theorem: the upper frequency limit of a digital recording is one-half the sampling rate. CDs use a 44.1 kHz sampling rate, so their frequency response extends to 22.05 kHz.
- Oversampling: sampling an audio signal at a higher rate than needed to reproduce the highest frequency in the signal. Ex. sampling a 20 kHz audio signal at 8 times 44.1 kHz is called “8x oversampling.” This process is followed by a digital lowpass filter and a gentle-slope analog anti-alias filter. Result: less phase shift and less harshness compared to a steep, “brick-wall” analog filter used alone.

Digital Recording

- Reduces noise, distortion, speed variations, and data errors
- Because digital playback head reads only 1's and 0's, it is insensitive to the magnetic medium's noise and distortion
- During recording and playback, numbers are read into a buffer memory and read out at a constant rate, eliminating speed variations in the rotating media.

Recording Medias

- Hard-disk drive records on magnetic hard disk
- Compact disc and DVD recorder record on an optical disc
- Memory recorder records onto a Flash memory card
- Sampler records into computer memory

File Formats

- Recorded as a wave (.wav) file or Audio Interchange File Format (AIFF) file.
- Both formats use linear PCM encoding and no data compression
- CDs (Compact Discs) are 16 bit/44.1kHz --- also referred to as “Red Book Format”

Digital Recording Level

- In a digital recorder, the record-level meter is a peak-reading LED or LCD bar graph metere that reads up to 0 dBF
- In a 16-bit digital recorder, 0dBFS means all 16 bits are on (1) at the waveform peak, and all 16 bits are off (0) at the waveform trough
- In a 24-bit digital recorder, 0 dBFS means that all 24 bits are on or off
- “Over” indication means that the input level exceeded the voltage needed to produce 0 dBFS, and there is some short-duration clipping of the output analog waveform
- Setting recording level: aim for -6 dB maximum so that unexpected peaks don’t exceed 0 dBFS

Clock

- each digital audio device has a clock or internal oscillator that sets the timing of its samples
- Clock signal is a series of pulses running the sampling rate
- When you transfer digital audio from one device to another, their clocks must be synchronized; one device must provide the master clock and the other must be the slave
- Clock comes embedded in the digital signal, or comes on a separate wire or connector as a word clock signal
- Internal, External or Word Clock

Digital Audio Signal Formats

- AES/EBU: professional format that transfer 2 channels of digital audio over 110-ohm shielded twisted-pair cable with XLR connectors
- MADI (Multichannel Audio Digital Interface): professional format that transports 28, 56, or 64 channels up to 24 bits/96 kHz on 75-ohm coaxial cable or fiber-optic cable; used to link large mixing consoles to digital multi-track recorders
- S/PDIF (Sony-Philips Digital Interface): 2-channel consumer or semi-pro format; uses a single 75-ohm coaxial cable with RCA or BNC connectors, or a fiber-optic cable with TOSLINK connectors
- ADAT Light pipe: send 8 channels of digital audio on a single optical cable with TOSLINK connectors; data transfer up to 24 bit/48 kHz or 24 bit/96 kHz with half the number of channels

Digital Audio Signal Formats

- TASCAM TDIF (TASCAM digital interface): DB-25 connectors; TDIF sends 8 channels of digital audio in and out ona single cable
- USB (Universal Serial Bus) and FireWire: standard Mac/PC protocols for high-speed serial data transfer between a computer and an external device, such as a hard drive, USB drive, MIDI Interface, or audio interface
- AES/EBU, S/PDIF, and ADAT Lightpipe are self-clocking systems (clock is embedded in the signal and marks the start time of each sample.
- MADI, TDIF, and ADAT Sync carry a separate word-clock signal on a separate connector or wire
- FireWire and USB signals don’t have clock information; reclocked at the receiving device

Dither

- When you save a 24-bit audio file as a 16-bit file to transfer to CD, those last 8 bits are truncated or cut off. Result may be a grainy static sound at very low levels. This distortion can be prevented by adding low-level random noise (dither) to the signal
- Example: 24-bit resolution can accurately capture the quietest parts of a musical concert: very low-level signals such as the end of long fades and reverb tails. Truncation of that signal to 16 bits makes those low-level signals sound grainy or fuzzy, because 16 bits is a less accurate measurement of the analog waveform than 24 bits. The fuzzy sound (called quantization distortion), doesn’t exist at normal high levels.

Jitter

- the unstable timing of samples that occurs in A/D and D/A conversion
- Any change between the sample times creates amplitude errors -- small changes in the audio waveform’s shape -- resulting in a slight veiling of the sound (low-level distortion or noise)
- Accurate A/D and D/A conversions rely on the clock precisely sampling the analog signal at equal time intervals
- One cause of jitter is analog noise and crosstalk in the recording system; they affect the switching times and switching threshold of the clock, causing frequency modulation of the clock; also affect analog filters and oscillators used in the clock’ s phase-locked loops

Jitter

- Also caused by inadequate cables; pick up hum and noise, and introduce phase shift and high-frequency attenuation which degrade the timing of the digital signal
- Jitter doesn’t occur during real-time data transmission over FireWire or USB because they are clockless systems
- To reduce jitter:
  - use high-quality clock sources with low jitter specs
  - use high-quality, well-shielded cables designed for digital signals, and as short as possible
  - keep analog and digital cabling separate
  - use your audio interface or A?D converter as the master clock; don’t drive it from an external source

Digital Transfers

- When sending digital audio signal in real time from one device to another, they must be set to the same sampling rate
- In a digital transfer, the sending device is usually the master, and the receiving device is the slave
- OK to send a lower bit depth signal to a higher bit device
- CDs must be converted down to 16 bit and 44.1 kHz
- WAVES or AIFFs copy a perfect clone of the original file; file transfers much faster than the real-time playback of the signal
- Flawless digital transfers:
  - computer from one hard drive to another
  - Ethernet, USB, FireWire, or the INternet
  - Between computers or flash memory cards

Digital Audio Workstation

- DAW: a computer running recording software with a connected audio interface such as a sound card
- Allows you to record, edit, and mix audio program entirely in digital form

Sources

Practical Recording Techniques Text Book