21M.380 · Music and Technology
Recording Techniques & Audio Production

Digital audio

Session 13 · Monday, October 24, 2016

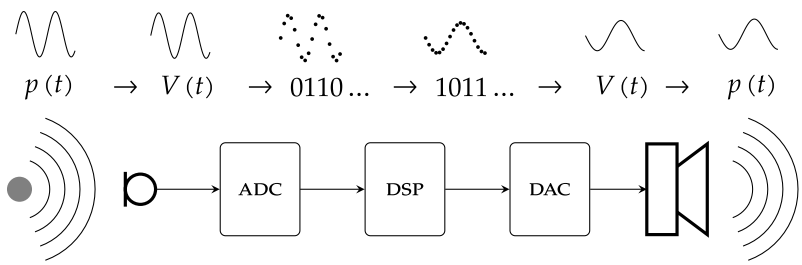

Figure 1. Digital reproduction chain

1 Quiz 2 (qz2)

2 Why digital?

Discussion: What is to be gained from digital audio?

- ________________________
- ________________________
- ________________________
- ________________________
- ________________________

3 Some definitions

- Signal: ____________________
- Analog signal: ____________________
- Digital signal: ____________________
- Binary signal: ____________________
- Noise: ____________________
- Distortion: ____________________
- Harmonic distortion: ____________________
- Inharmonic distortion: ____________________

4 Analog-digital conversion

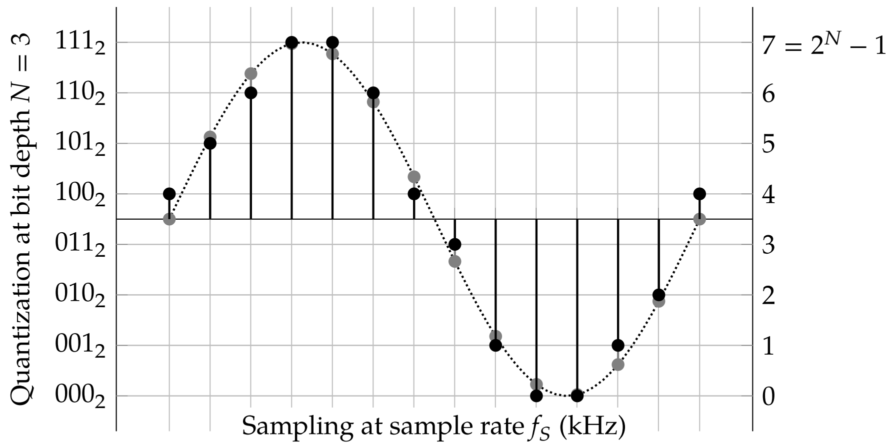

Figure 2. Analog/digital conversion

Two-step process, usually (but not necessarily!) performed in order of:

1. Sampling
2. Quantization

5 Sampling

5.1 Sampling theorem

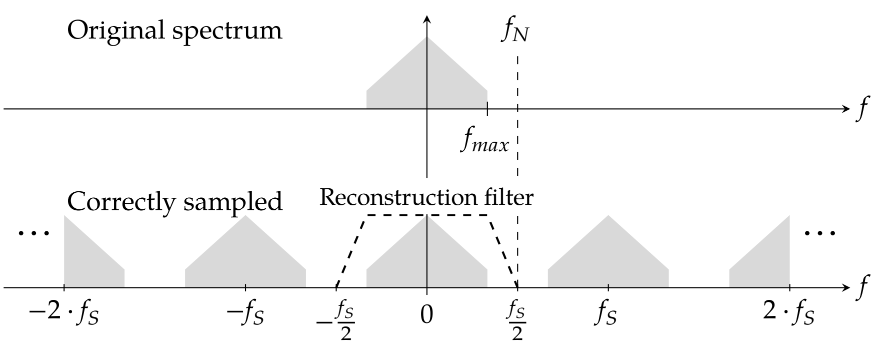

Figure 3. Sampling creates spectral sidebands of the original spectrum that repeat periodically around multiples of f_S (after Lyons 2004, fig. 2.4)

- As long as we comply with the sampling theorem, we can restore the original signal *without loss of information or quality*.
  - Unless we also quantize (which, in practice, an ADC always does)
  - But no information loss inherent to (correctly performed) sampling!
- Interpreting the sampling theorem from two different angles:

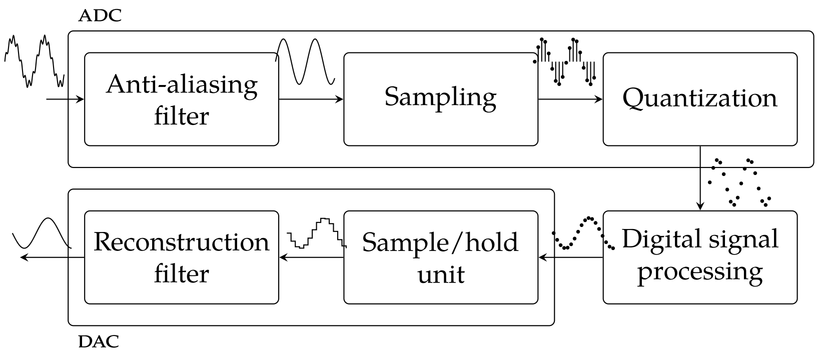

Figure 4. Adc/dac conversion chain (after Smith 1997b, fig. 3.7)

- To faithfully reconstruct a sampled signal whose maximum frequency is f_{max}, we must choose a sample rate $f_S > 2 \cdot f_{max}$
- For a given sample rate f_S, the *Nyquist frequency* $f_N = \frac{f_S}{2}$ represents the highest frequency that can be faithfully reproduced.

- How does reconstruction of original signal come about?
  - Sampled signal's spectrum = copies of original spectrum centered around multiples of f_S
  - *Reconstruction filter* (lp tuned to f_{max}) discards upper sidebands
- Reconstruction filter = last stage of dac (but here quantization has occured, too!)

5.2 Aliasing (undersampling)

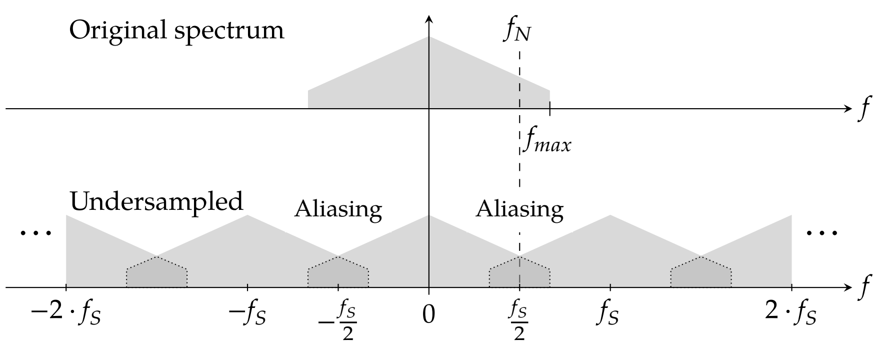

Figure 5. A violation of the sampling theorem (undersampling) results in aliasing (after Lyons 2004, fig. 2.4)

- Result of sampling theorem violation, i.e., when $f_{max} > \frac{f_S}{2}$
- Sidebands overlap and hf components from upper sidebands are 'folded back' down into audible spectrum
- Audible as inharmonic distortion
- Visual analogy: backward-turning wheel in movies

- How to prevent aliasing?
  - *Anti-aliasing filter* (LP tuned to f_N) before sampling
  - Same kind of filter as reconstruction filter, but different purpose

5.3 Signal reconstruction & oversampling

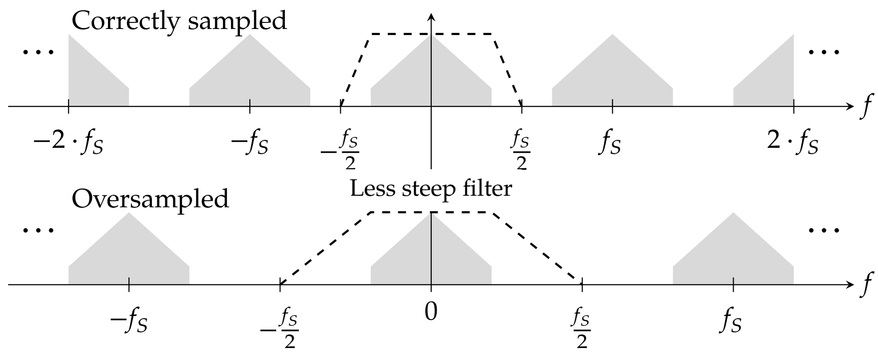

Figure 6. Deliberate oversampling allows the use of less steep reconstruction filters (after Lyons 2004, fig. 2.4)

- Deliberate use of higher sample rate than sampling theorem demands
- Allows the use of (cheaper) reconstruction filters with a less steep slope
- Redistributes quantization noise beyond audible range[1]

[1] Cf., Bohn 1997, p. 8.

5.4 Jitter

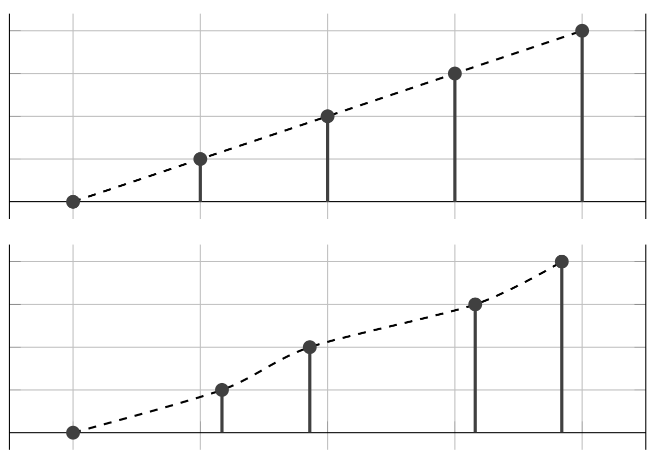

Figure 7. Playing back a digital signal on a jittering sample clock results in distortion.

- ADC and DAC are driven by *sample clock* (quartz crystal oscillator)
- *Jitter* is an irregularity of that sample clock[2]

[2] Cf., Katz 2014b.

- Causes distortion when audio recorded on a sample clock with jitter is played back on a sample-clock without jitter (or vice versa)
- See Katz (2014b) for detailed discussion of jitter

6 Quantization

6.1 Binary numbers

- Digit … decimal (0 or 1 or 2 or … or 9)
- Bit … *b*inary dig*it* (0 or 1)
- Binary-to-decimal conversion:
  $1001_2 = 1 \cdot 2^3 + 0 \cdot 2^2 + 0 \cdot 2^1 + 1 \cdot 2^0 = 8 + 0 + 0 + 1 = 9$
- Same principle as: $975_{10} = 9 \cdot 10^2 + 7 \cdot 10^1 + 5 \cdot 10^0$

6.2 Bit depth

- Can express 10^N numeric values with N digits (base-10 number system)
- Can express 2^N numeric values with N bits (base-2 number system)
- E.g., $2^{16} = 65\,536$ possible amplitude values with 16 bit

6.3 Quantizing error

- Difference between original and quantized signal
- Audible as:
  - Background noise for high-level signals
  - Distortion for low-level signals (where error correlates with signal)
- Determines available dynamic range

☞ *Higher bit depth provides larger dynamic range:* $\Delta L_{dig} \approx (6 \cdot N)$ dB

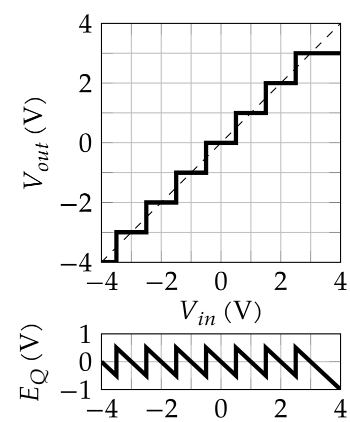

Figure 8. Quantizing error E_Q of a 3 bit ADC

6.4 Dither[3]

- Weird: we deliberately *add* noise to make things sound *better* ☺
- Trade off distortion from quantizing error against background noise
- Reason: Background noise perceptually less annoying than distortion
- Effectively linearizes transfer function
- Goal: Extend perceived dynamic range downwards
- Different types of dither:[4]
  - Rectangular pdf

[3] Montgomery (2012, 11′35″–17′18″) discusses dither in an engaging video. Together with another video by Montgomery (2010), this provides an excellent in-depth introduction to digital audio for especially interested students.

[4] Cf., Watkinson 2001, pp. 231 ff.

- Triangular pdf
- Gaussian pdf

- *Noise shaping* filters dithering noise to be less intrusive.
- Dither and noise-shaping options in Reaper: File » Render ...
- Rules of 👍:
  - Dither when moving from high to lower bitrate (often 24 to 16)[5]
  - Dither only once, as the very last step![6]

[5] Cf., Ariza 2012, pp. 252 f.

[6] Cf., Izhaki 2011, p. 154.

7 Preview: Ed4 assignment

References & further reading

Ariza, Christopher (2012). *21M.380 Music and Technology. Dithering and Mastering*. URL: `https://ocw.mit.edu/courses/music-and-theater-arts/21m-380-music-and-technology-recording-techniques-and-audio-production-spring-2012/lecture-notes/MIT21M_380S12_lec24.pdf` (visited on 02/05/2017).

Bohn, Dennis (1997). *Digital Dharma of Audio A/D Converters*. URL: `http://www.rane.com/pdf/ranenotes/Digital_Dharma_of_Audio_AD_Converters.pdf` (visited on 09/02/2014). RaneNote 137.

Christopher 'Monty' Montgomery and the Xiph.Org Community (2010). *Episode 1: A Digital Media Primer for Geeks*. URL: `https://xiph.org/video/vid1.shtml` (visited on 11/02/2015).

— (2012). *Episode 2: Digital Show & Tell*. URL: `https://xiph.org/video/vid2.shtml` (visited on 11/02/2015).

Izhaki, Roey (2011). *Mixing Audio. Concepts, Practices and Tools*. 2nd ed. Focal Press. 600 pp. ISBN: 978-0240522227. MIT LIBRARY: 002302617. Hardcopy and electronic resource. On course reserve at the Lewis Music Library. Accompanying sound examples: `http://www.taylorandfrancis.com/cw/izhaki-9780240522227/p/resources/`.

Katz, Bob (2014a). "High sample rates. Is this where it's at?" In: *Mastering Audio. The Art and the Science*. 3rd ed. Burlington, MA: Focal Press. Chap. 23, pp. 311–20. ISBN: 978-0240818962. MIT LIBRARY: 002307049. On course reserve at the Lewis Music Library.

— (2014b). "Jitter. Separating the myths from the mysteries." In: *Mastering Audio. The Art and the Science*. 3rd ed. Burlington, MA: Focal Press. Chap. 24, pp. 321–37. ISBN: 978-0240818962. MIT LIBRARY: 002307049. On course reserve at the Lewis Music Library.

Lyons, Richard G. (2004). *Understanding Digital Signal Processing*. 2nd ed. Prentice Hall. 688 pp. ISBN: 978-0131089891. MIT LIBRARY: 001289139.

Rumsey, Francis and Tim McCormick (2009). "Digital audio principles." In: *Sound and Recording. An Introduction*. 6th ed. Focal Press. Chap. 8, pp. 201–256. MIT LIBRARY: 002147704.

Smith, Steven W. (1997a). "Adc and dac." In: *The Scientist and Engineer's Guide to Digital Signal Processing*. 1st ed. California Technical Pub. Chap. 3, pp. 35–66. url: http://www.dspguide.com/CH3.PDF. © 1997–1998 by Steven W. Smith. For more information visit the book's website at: http://www.DSPguide.com.

— (1997b). *The Scientist and Engineer's Guide to Digital Signal Processing*. 1st ed. California Technical Pub. 640 pp. url: http://www.dspguide.com/.

Watkinson, John (2001). "Conversion." In: *The Art of Digital Audio*. 3rd ed. Chap. 4, pp. 196–274.