

MTE 1001

Fundamentals of Sound Synthesis

Topic Areas:

Digital Signals

Readings:

(Previously, Chapt 3, pp. 87-90
Chapt. 4, pp 139-146)

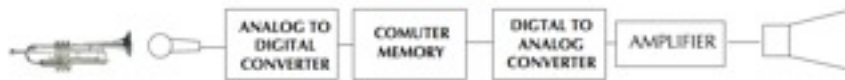
New: Chapt 1, pp 22-32

Analog vs. Digital

Analog

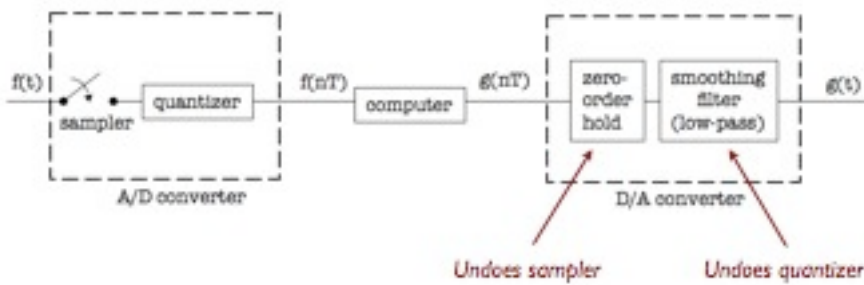


Digital

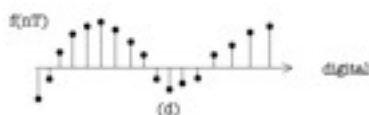
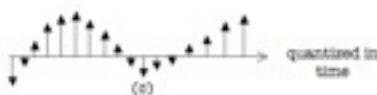


Idealized Model of Sampling and Reconstruction

A/D & D/A converters



Sampling and Signal Graphics



Graphic representation of types of signals:

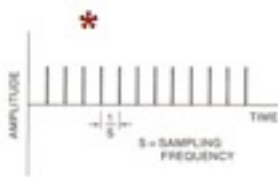
- a) analog signal
- b) quantized signal (continuous time)
- c) sampled data signal
- d) digital signal

Taking Sampling In One Step

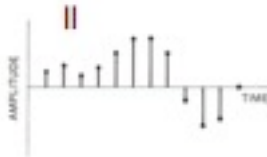


(A) Input waveform (after anti-aliasing filter)

Sampling signal is a series of impulses at the **sampling rate** separated by T seconds where
 $T = 1.0/\text{samplingRate}$



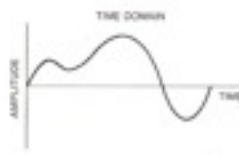
(C) Sampling signal



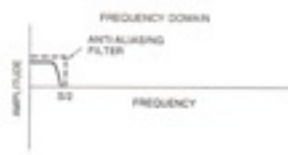
(E) Sampled input waveform

Taking Sampling In One Step

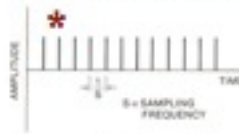
Time Domain & Frequency Domain



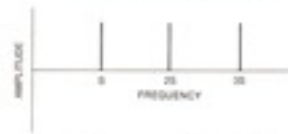
(A) Input waveform (after anti-aliasing filter)



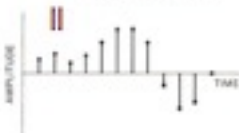
(B) Spectrum of input waveform



(C) Sampling signal



(D) Spectrum of sampling signal



(E) Sampled input waveform



(F) Spectrum of sampled input waveform

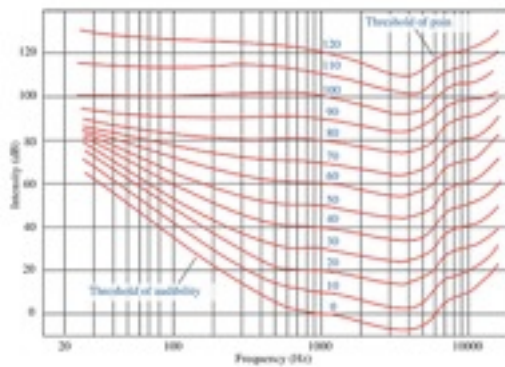
Fidelity

depends on:

- Sampling Rate
- Bit Depth
- (Encoding Scheme)

Human Hearing

20 - 20,000 Hz



Sampling Rate

Sampling Theorem:

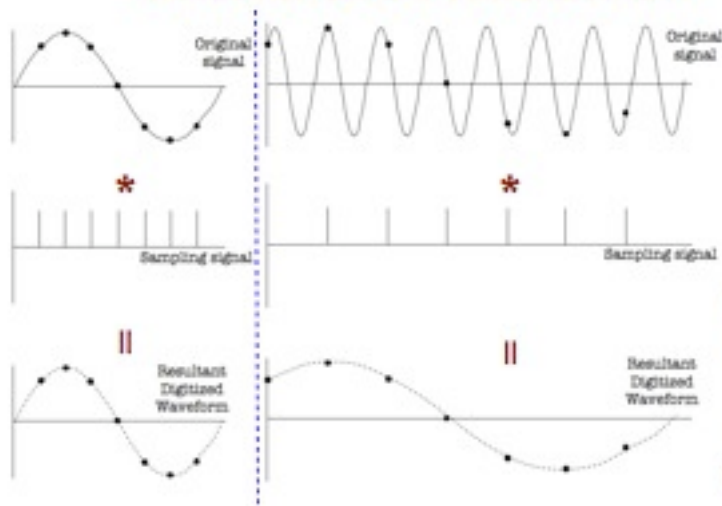
Must sample at a frequency twice as high as the highest sinusoidal component.

A consequence of the Sampling Theorem is that the highest frequency reproduced through a D/A converter is one half of its sampling rate.

| Typical rates: | Highest freq.: | Fidelity: |
|----------------|----------------|--------------|
| 8 K | 4 K | dark, dull |
| 11 K | 5.5 K | ↑ |
| 22 K | 11 K | ↕ |
| 44.1 K | 22 K | ↓ |
| 48 K | 24 K | bright/clean |

Why the Sampling Theorem?

Foldover Error: Time Domain



Foldover error is also called **aliasing**

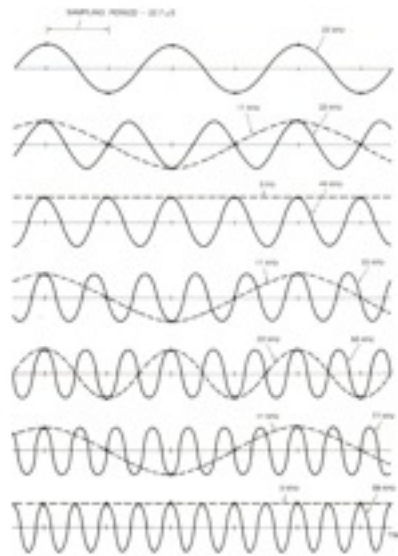
Why the Sampling Theorem?

Foldover Error

If the original signal is *undersampled*, the frequency of the resultant waveform is

$$SR - f$$

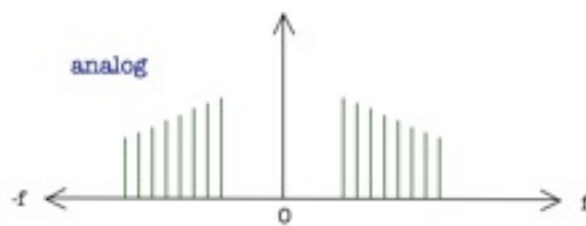
With a SR of 44100, attempting to create a tone of 30,000 Hz would result in an alias of $44100 - 30000 = 14100$!



Why the Sampling Theorem?

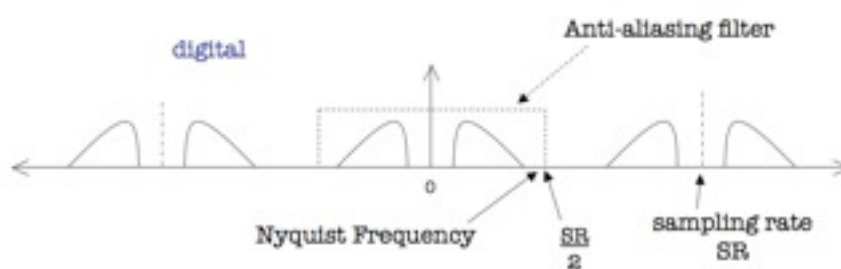
Foldover Error: Frequency Domain

First:
Positive and Negative Frequency



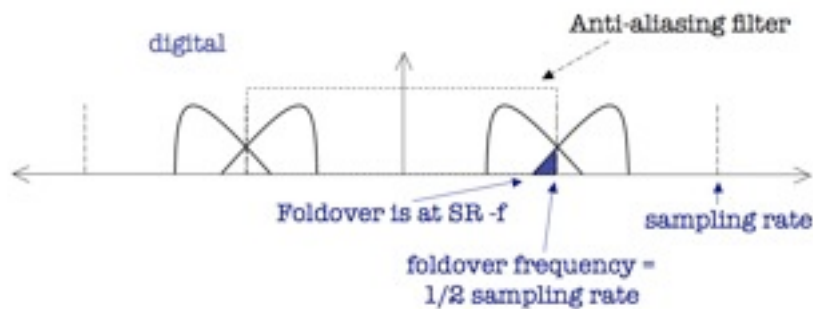
Why the Sampling Theorem?

Foldover Error: Frequency Domain



Why the Sampling Theorem?

Foldover Error: Frequency Domain



What is heard:  error!

Sampling Theorem

Low sampling rates produce low fidelity.

And, if we are not careful, foldover error can create very audible distortion.

This is not a problem for recording any more, because A/D converts always have anti-aliasing filters, but we still have to be careful to avoid foldover in synthesis.

**Next Topic: Digital Signals
continued**