Lecture 31

Interpolation

Sample Rate Conversion

Upsampling / Downsampling (Digital)

A/D conversion

Multirate Signal Processing

Why sample rate conversion?

What rates do we record at?

44.1 kHz! Why?

mp3? typically 44.1 kHz

→ lectures? 16 kHz

cellphones? 8 kHz (voice)

app: change the sample rate of a digital signal

ex: 44.1 kHz → 16 kHz

Two things can happen:

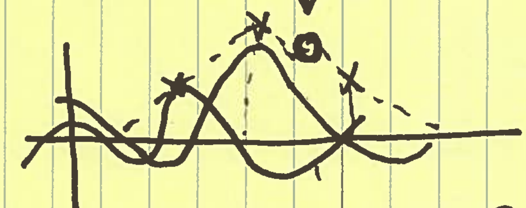
(1) aliasing

(2) signal can be "muffled" due to lowpass filtering

Straightforward Approach:

From the sampling theorem:

perfect reconstruction if $f_s > 2f_{max}$



What is wrong with this?

(1) infinite summation - sums over all samples (non-real-time)

(2) non-causal!

(3) if we limit the sum, we can design a better filter than a ~~[illegible]~~ sync function!

Upsampling by integer ratios:

$$x(n) \longrightarrow \underset{@\,L}{\boxed{\text{zero stuff}}} \longrightarrow \boxed{\text{LPF}} \longrightarrow x(m)$$

$$f_{s_{new}} = L\, f_{s_{old}}$$

Ex: 8KHz ⇒ 48 KHz

$L = 6$

LPF @ 4KHz @ 48KHz

Downsample by integer ratios:

$$\underset{48\text{KHz}}{x(n)} \longrightarrow \underset{\substack{4\text{KHz} \\ @\,48\text{KHz}}}{\boxed{\text{LPF}}} \longrightarrow \boxed{\text{Decimate}} \longrightarrow \underset{8\text{KHz}}{x(m)}$$