EECE233 Signals and Systems

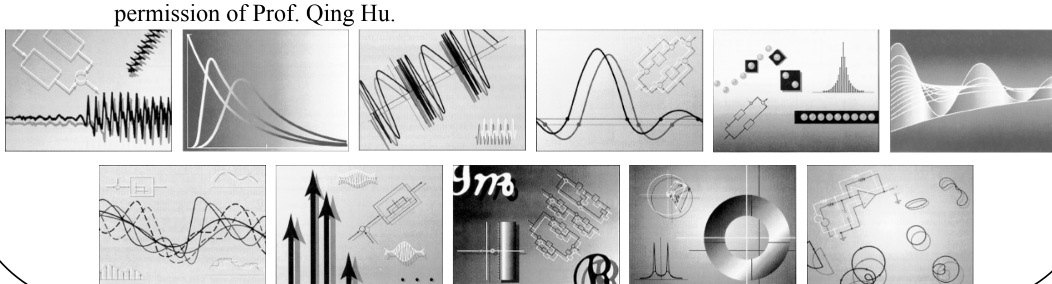
Lecture Note #22

Prof. Joon Ho Cho

(Slides thanks to Qing Hu*, D. Boning, D. Freeman, T. Weiss, J. White, and A. Willsky)

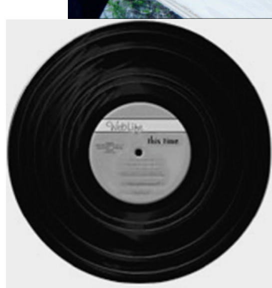
- CD Recording and Playback
- Course wrap-up

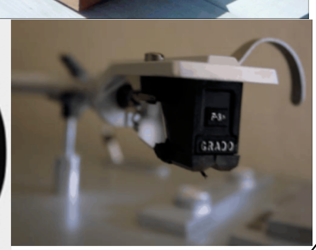
* These lecture notes are based on the lecture notes used in 6.003 taught by Prof. Qing Hu at M.I.T. These notes are adopted for EECE233 under the permission of Prof. Qing Hu.



Brief History of Sound Reproduction

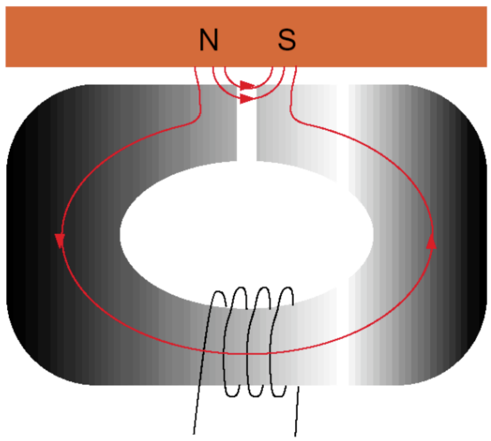
- Mechanical — gramophone (Thomas Edison 1877)





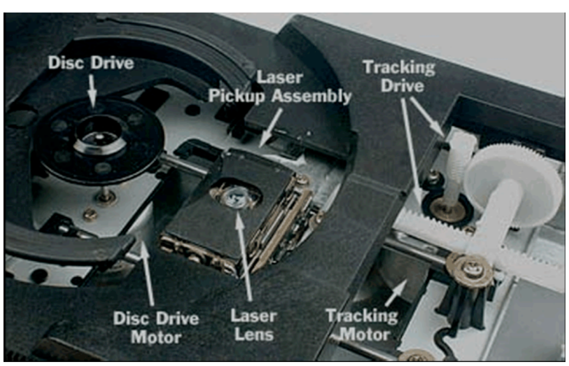
Brief History of Sound Reproduction (cont.)

- Magnetic — tape cassette (~1927—)



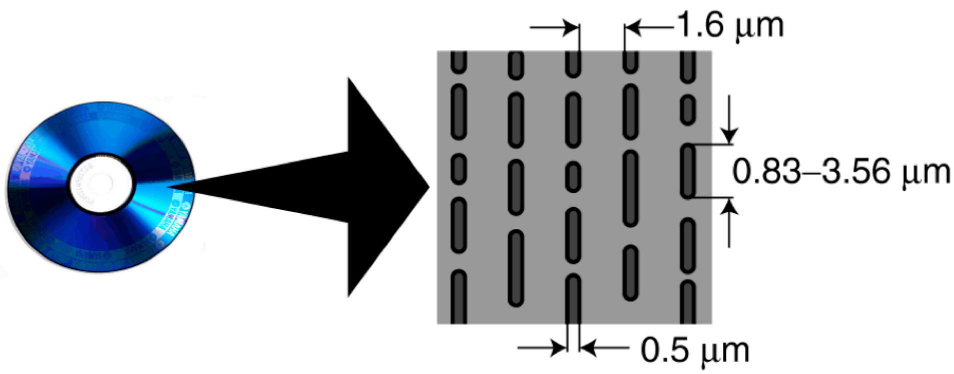
- Both the mechanical and magnetic recordings are *analog* — The recorded *CT* signals are proportional to the original sound signals.

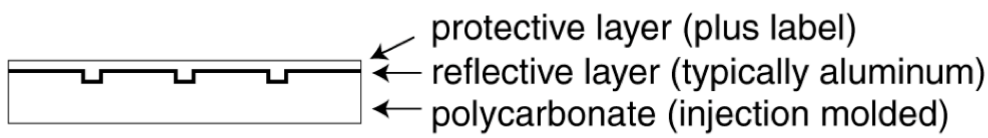
Digital Recording — Compact Disk (CD)



Digitally recorded medium with optical read-out.

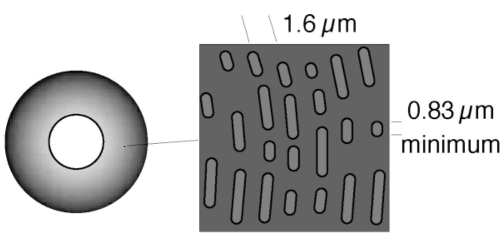
What is on a CD?



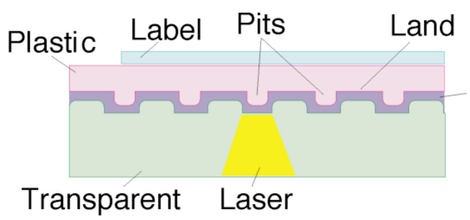


Read out from a CD

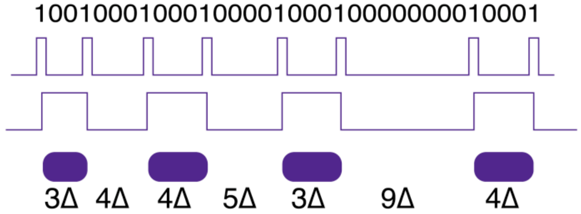
Picture of the surface of a CD



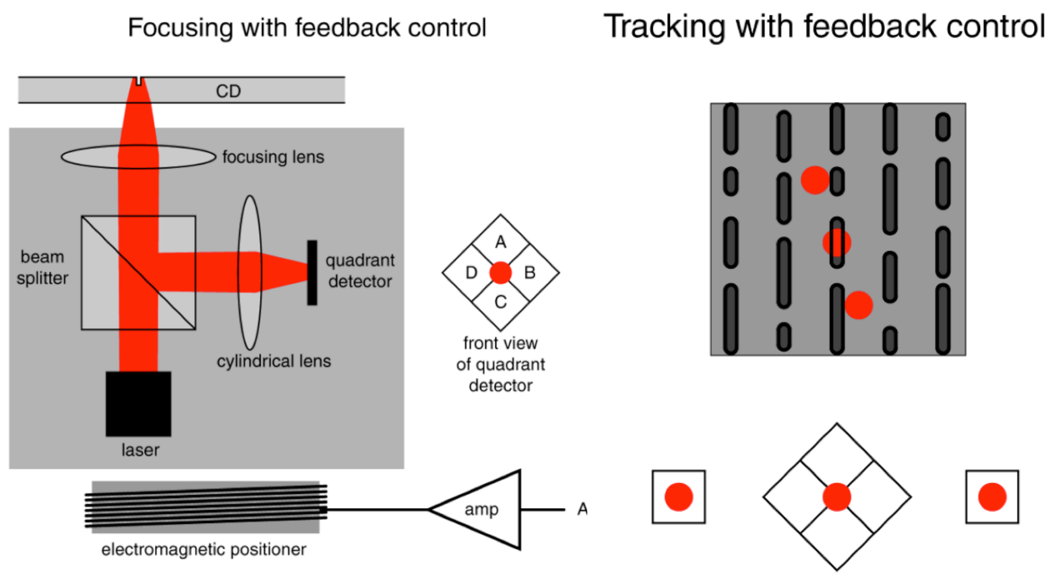
Schematic of read-out



- From pits to bits — the length of the pits and lands represents a binary number



Feedback control to stay focused with a constant linear speed



Advantages of Digital Over Analog Recording

- A much better signal/noise ratio, — just like all other digital devices over their analog counterparts (***E.g.*** Digital vs. analog wireless phones, HDTV vs. analog TV, *etc.*).

  E.g. For a standard 16-bit dynamic range of CD recording, assume the noise is mainly due to the quantization error in *LSB*, then the signal/noise ratio is:

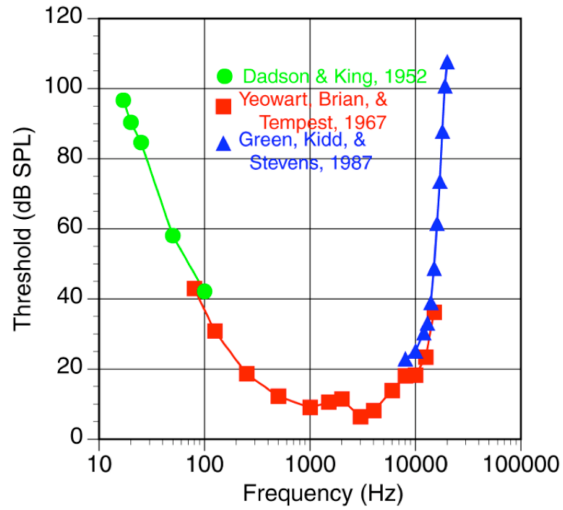
$$2^{16} \Rightarrow 20\log_{10}(2^{16}) \approx 96 \text{ dB!}$$

$$\Rightarrow \frac{\text{Power of signal}}{\text{Power of noise}} \approx 10^{10}\,!$$

- Can perform error corrections to further improve fidelity of recording and playback
  E.g. Minor scratches will not affect the sound quality.

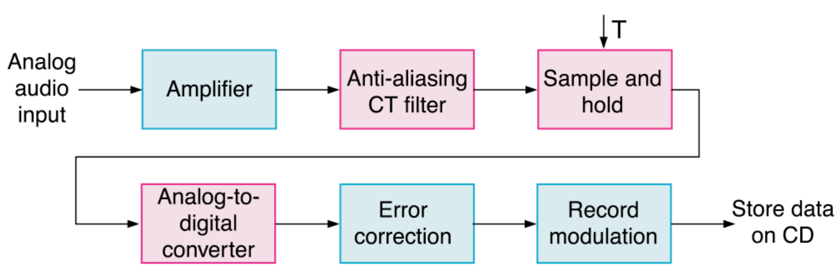
“Boundary Conditions” for Recording — Human Hearing

- Experimentally measured human hearing threshold as a function of frequency $f = \omega/2\pi$.

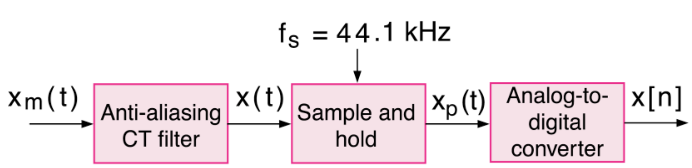


The frequency range of human hearing is approximately *20Hz-20kHz*

Block Diagram of CD Recording



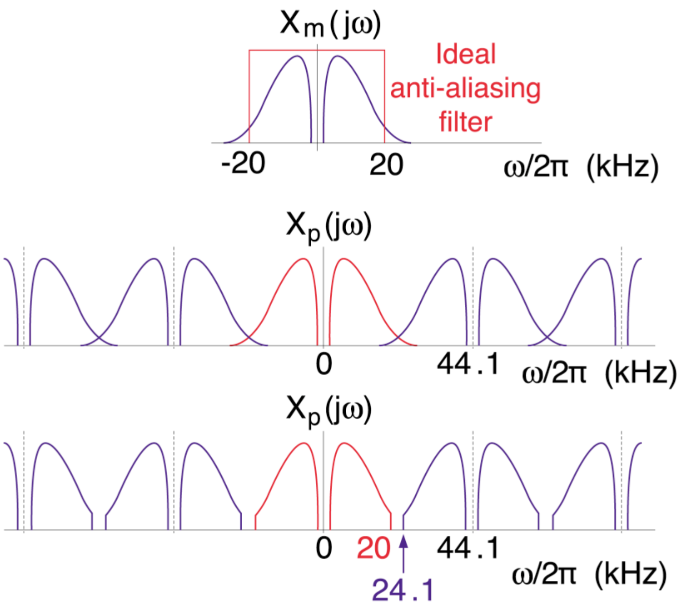
- A straightforward and simplistic approach — 1 × sampling



- $f_s = \omega_s/2\pi = 44.1$ kHz > 2 × maximum frequency of human hearing
  The specific value was chosen to synchronize with video

  f_s = (3 samples/line)×(490 lines/frame) ×(30 frames/s) = 44.1 kHz

1 × Sampling Using an Ideal Anti-aliasing Filter (AAF)

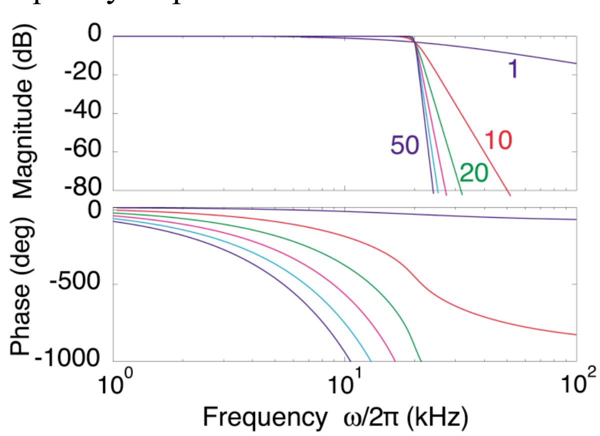


Problem of 1× sampling: Analog AAF with sharp cut-off difficult to implement.

Problem of Analog AAF

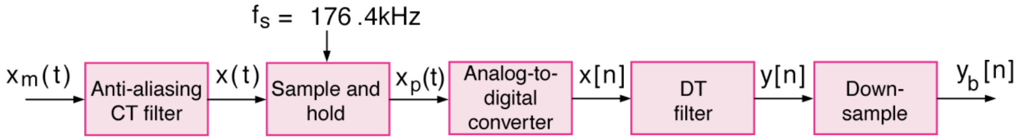
- In order to avoid aliasing, need an analog LPF with a very sharp cut-off. Specifically, need a 80-dB roll-off over the frequency range from 20 *kHz* to 24.1 *kHz*.

  E.g. Frequency response of *nth*-order Butterworth filters

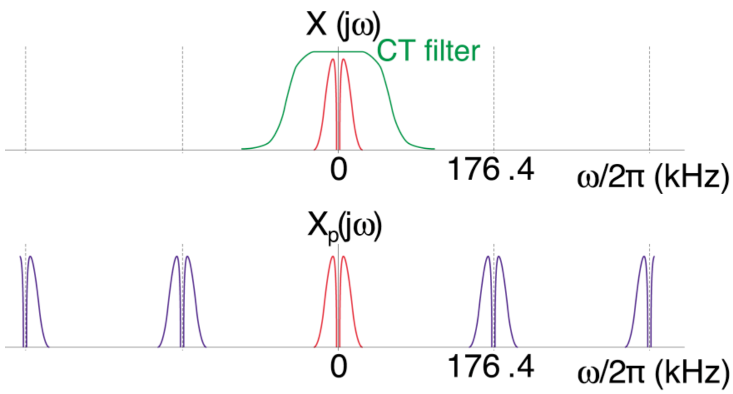


- Problem — Significant *nonlinear* phase distortion in the passband.

Solution: 4× Oversampling

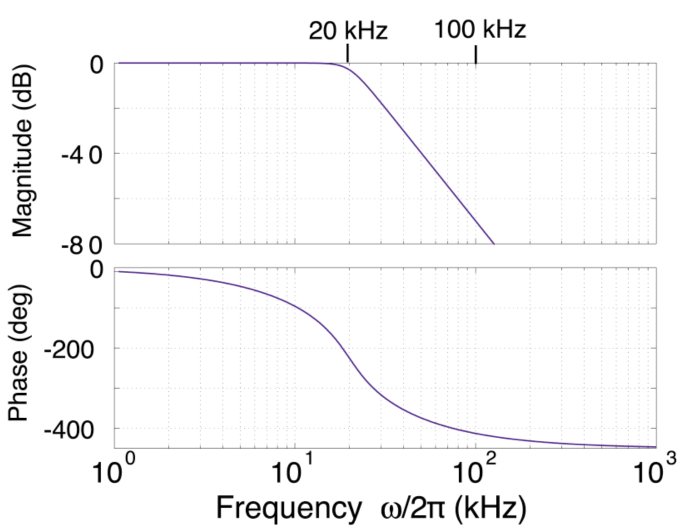


Now the shifted spectra are far apart (176.4 *kHz*), the requirement on the front-end analog AAF is much more relaxed.



4× Oversampling (continued)

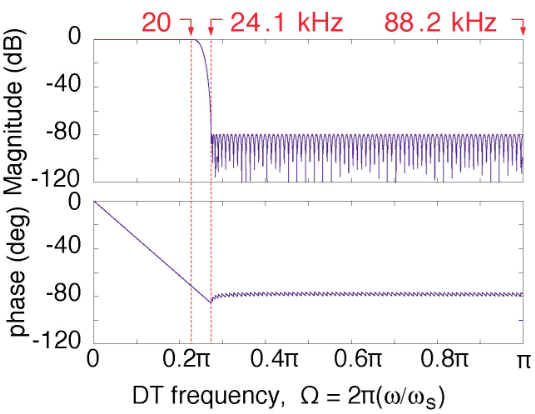
- Now the 80-dB attenuation can be easily achieved over the frequency range of 20-156.4 *kHz*, ***e.g.***, with a fifth-order Butterworth filter.

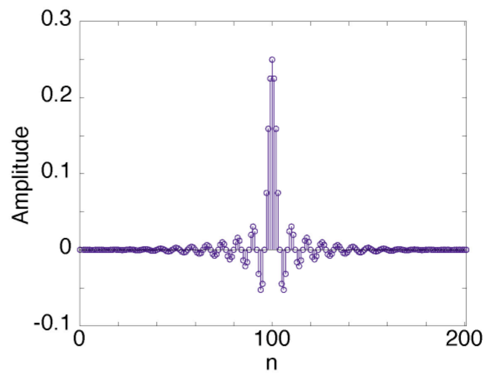


DT Processing of Oversampled Signals ($\Omega = 2\pi\omega/\omega_s$)

- Once the signals are converted into DT signals, it is *much* easier to construct *DT* LPF with a very sharp cut-off.

E.g. A 200-point FIR filter with a *linear* phase.

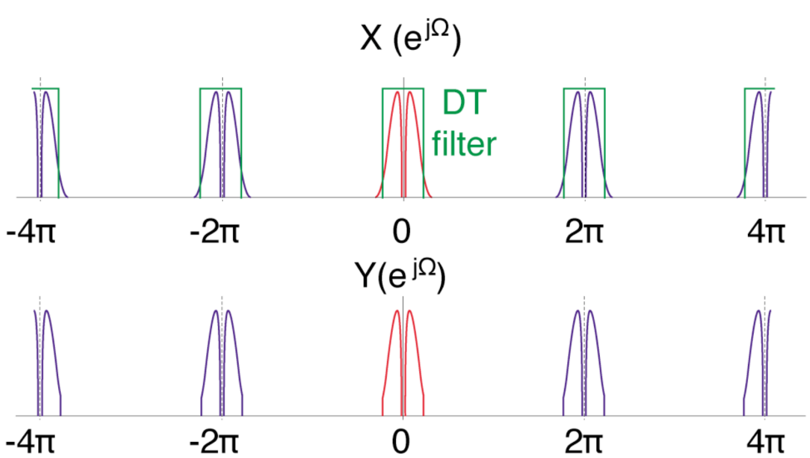




Frequency response
Linear phase in the passband

Unit-sample response

Now We Can Eliminate Signals Beyond Our Hearing Range

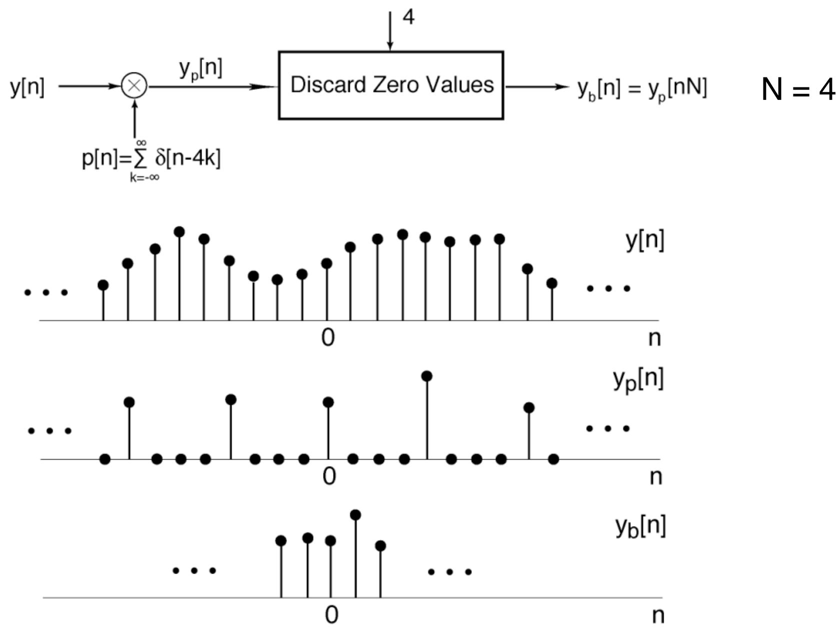


DT signal after sampling at 176 kHz

After filtering with DT filter

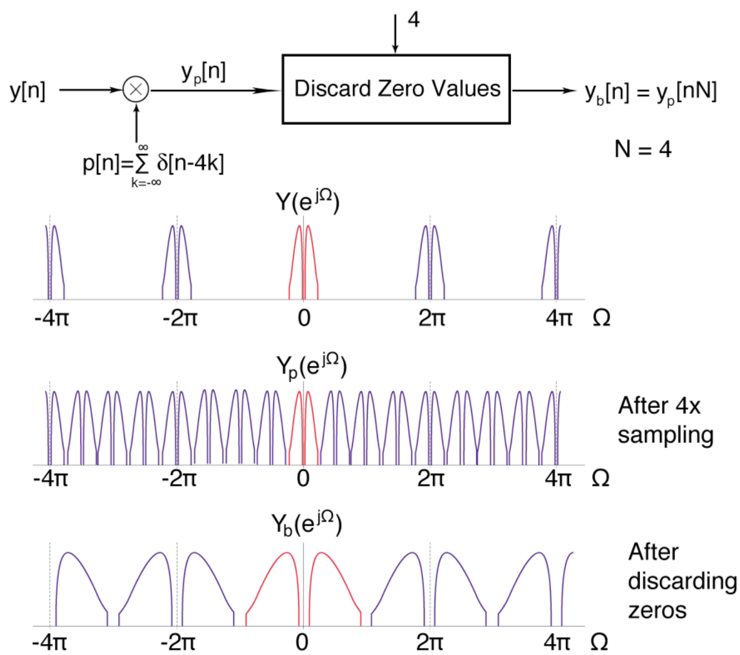
- Now we have solved the problem of aliasing, but ended up with 4× amount of data needed for CT signal reconstruction.
- Solution — 4× downsampling.

4× Downsampling — Decimation



Compressed by 4× in the time domain (speed up by a factor of 4).
This is what you have on a CD.

4× Downsampling — Frequency Domain



- Stretched by 4× in the frequency domain.

Decimation in the Frequency Domain

(Slide 19 in Lecture #16)

$$X_b\left(e^{j\omega}\right) = \sum_{k=-\infty}^{+\infty} x_b[k]e^{-j\omega k} \qquad (x_b[k] = x_p[kN])$$

$$= \sum_{k=-\infty}^{+\infty} x_p[kN]e^{-j\omega k} \qquad \text{Let } n = kN \text{ or } k = n / N$$

$$= \sum_{n \text{ a multiple of } N}^{+\infty} x_p[n]e^{-j\omega(n/N)}$$

$$= \sum_{n=-\infty}^{+\infty} x_p[n]e^{-j(\omega/N)n} \qquad (\text{since } x_p[n \neq kN] = 0)$$

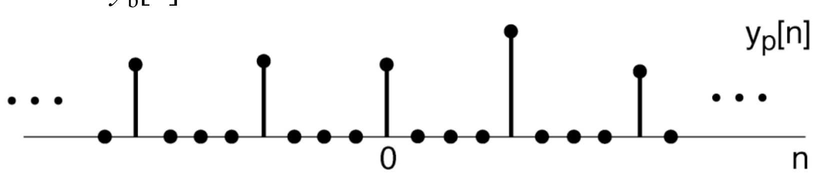
$$= X_p\left(e^{j(\omega/N)}\right)$$

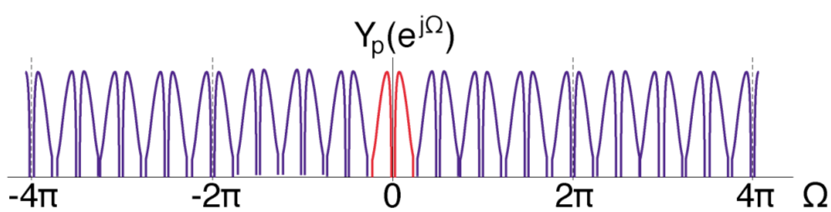
stretched in frequency by N

CD Playback — Reverse Process of Recording 4× Upsampling

Step 1: Inserting 3 zeros between two adjacent samples (need to slow down by a factor of 4).

Data stored on a CD $y_b[n]$ → Insert 3 zeros → $y_p[n]$

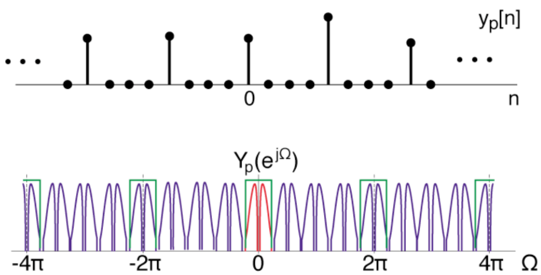




CD Playback — Reverse Process of Recording
4× Upsampling (continued)

Step 2: Pass $y_p[n]$ through a DT LPF — DT interpolating

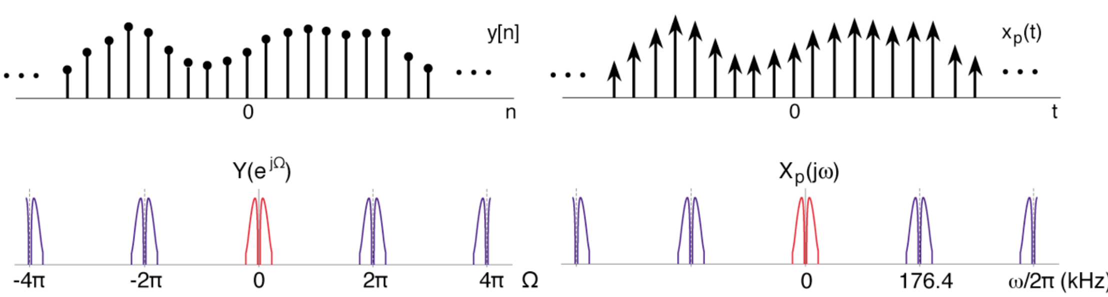
$$y_p[n] \longrightarrow \boxed{\text{DT LPF}} \longrightarrow y[n]$$



CD Playback (continued)

Step 3, DT to CT conversion.

$$y[n] \longrightarrow \boxed{\text{D/C}} \longrightarrow x_p(t)$$



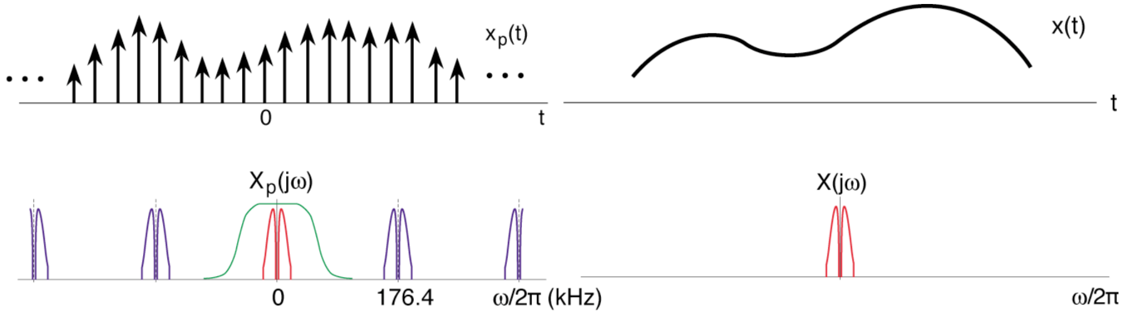
Sequence

Impulse train in time-domain

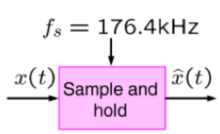
CD Playback (continued)

Finally, CT lowpass → get back the original CT signal

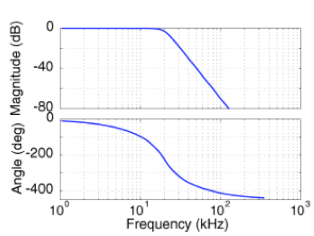
$$x_p(t) \longrightarrow \boxed{\text{CT LPF}} \longrightarrow x(t)$$



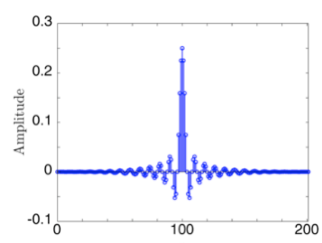
Wrap-up



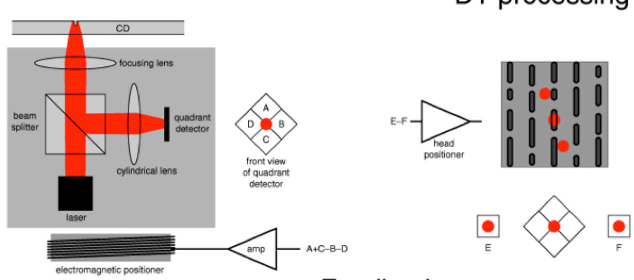
Sampling



Filtering



DT processing of CT signals



Feedback

- CD is said to be one of the most successful electronic instruments ever invented. AND EECE233 enables you to understand how it is recorded and played back, and many other goodies ...