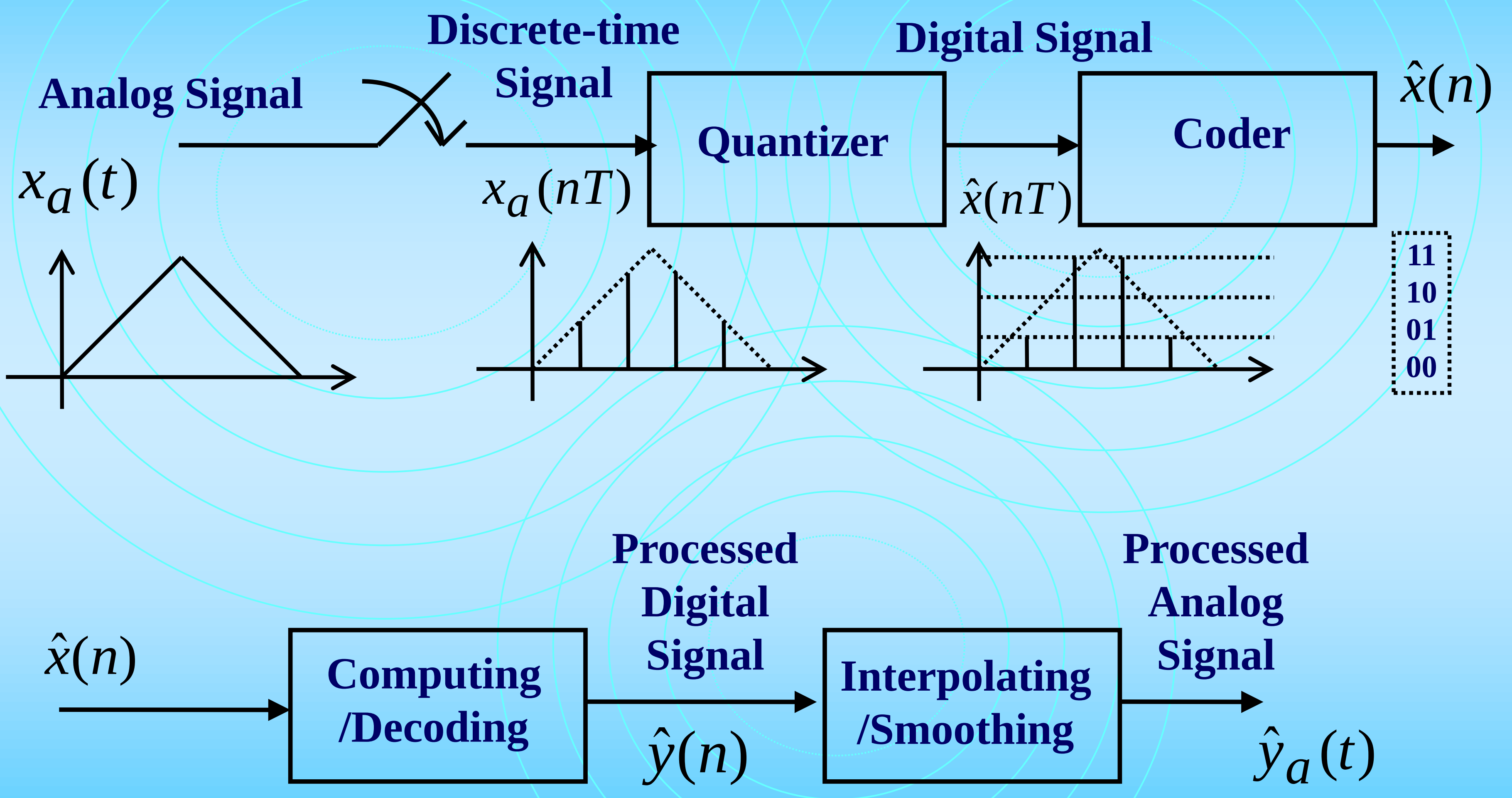
Digital Representation of Audio Information

COM 429: MULTIMEDIA TECHNOLOGIES

Patrick Theuri

Elements of a DSP System



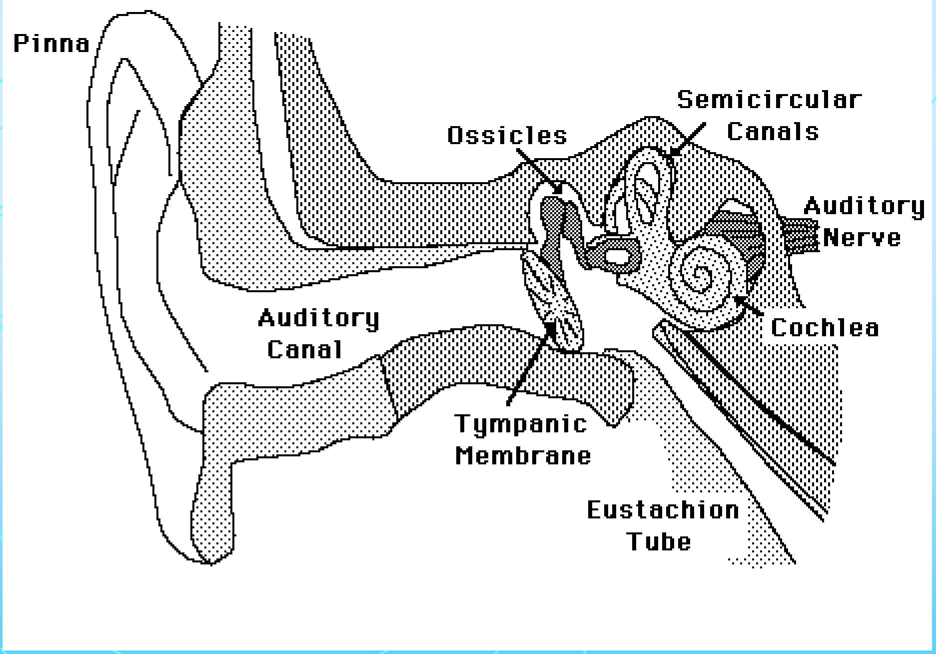
Critical Audio Issues

Trade-off between resources to store/transmit and quality of audio information

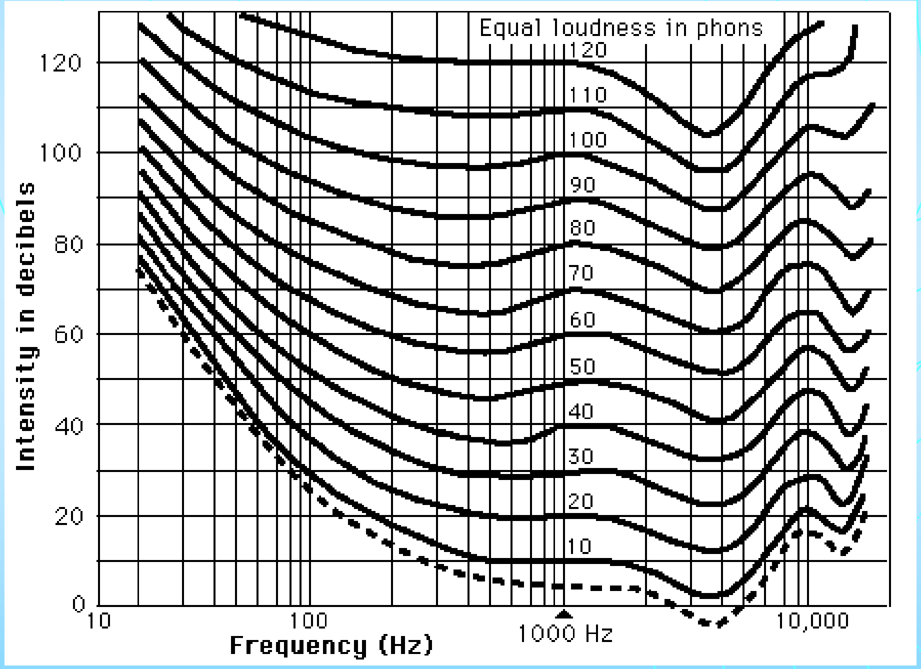
- Sampling rate
- Quantization level
- Compression techniques

Sound and Human Perception

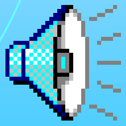
- Signal fidelity does not need to exceed the sensitivity of the auditory system



Audible Frequency Range and Sampling Rate



- Frequency range - 20 to 20,000 Hz
- Audible intensities - threshold of hearing (1 Pico watt/meter^2 corresponds to 0 db
- Sample sweep constant intensity – 0 to 20 kHz in 10 seconds



Sampling Requirement

- **A bandlimited signal can be completely reconstructed from a set of discrete samples by low-pass filtering (or interpolating) a sequence of its samples, if the original signal was sampled at a rate greater than twice its highest frequency.**

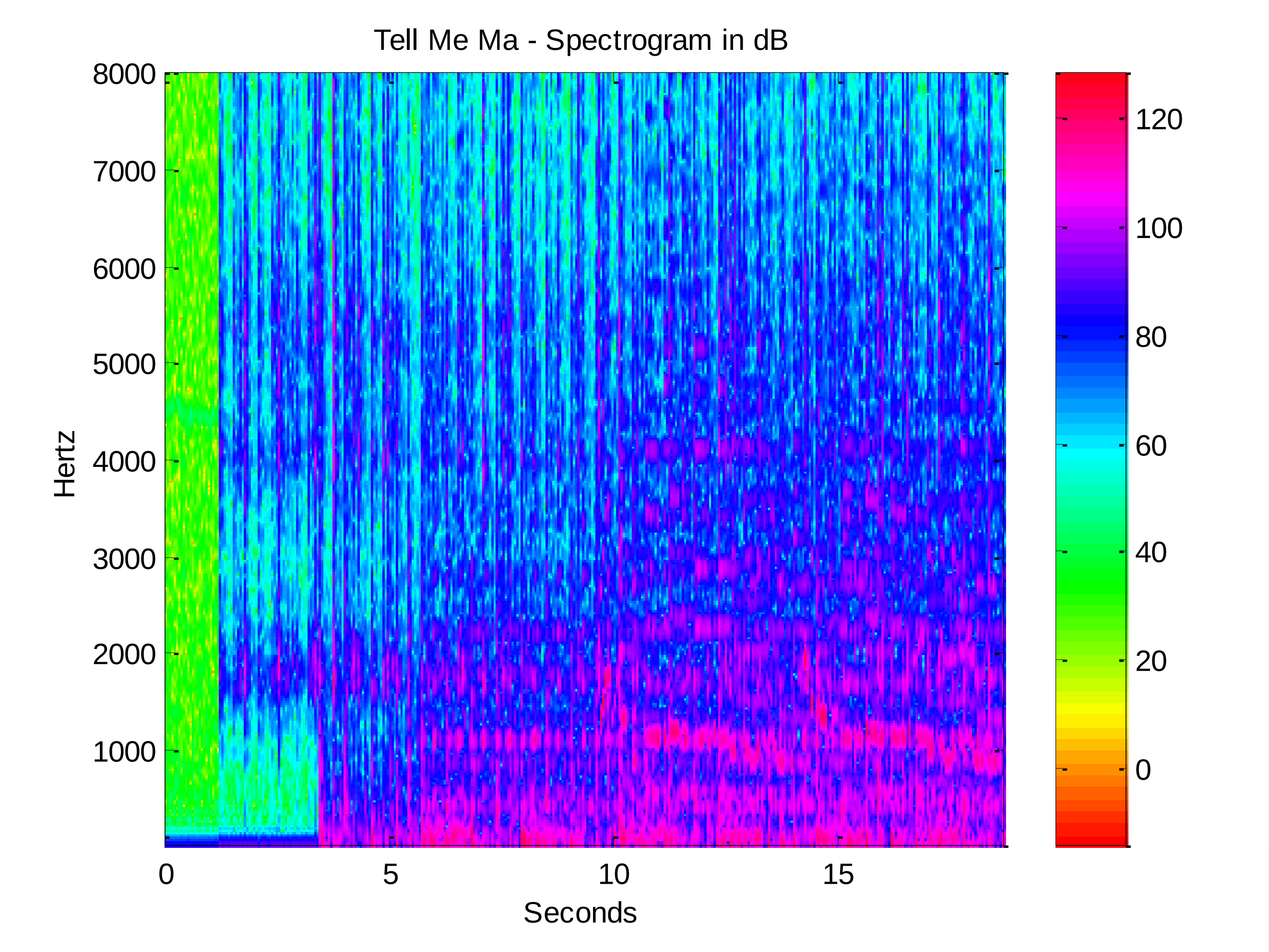
- ***Aliasing* errors occur when original signal contains frequencies greater than or equal to half the sampling rate.**

- **Signal energy beyond 20 kHz is not audible, ∴ sampling rates beyond 40 kHz should capture almost all audible detail (no perceived quality loss).**

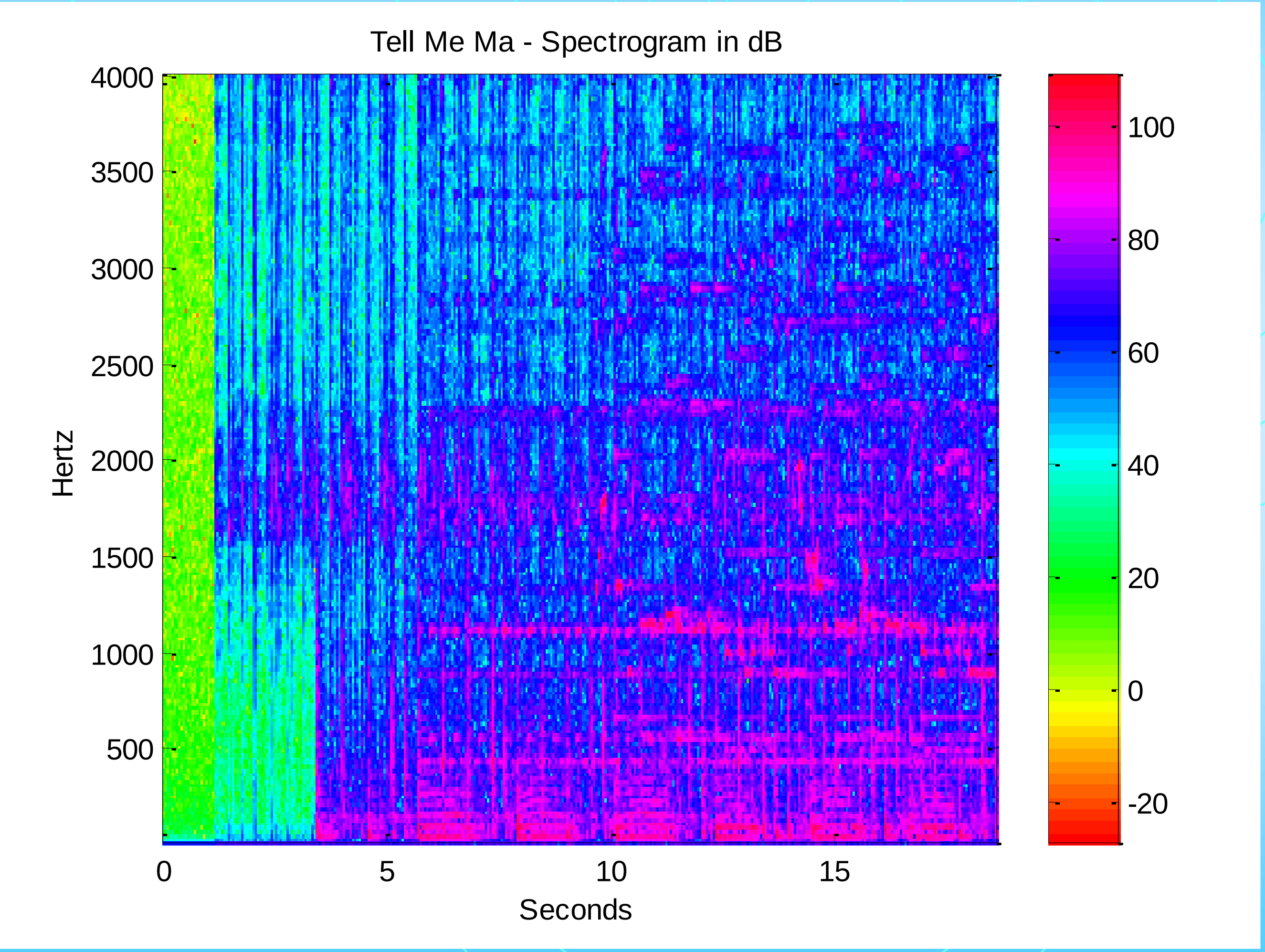
Sampling Standards

- **CD quality samples at 44.1 kHz**
- **DVD quality samples at 48 kHz**
- **Telephone quality 8 kHz.**

Spectogram of CD sound

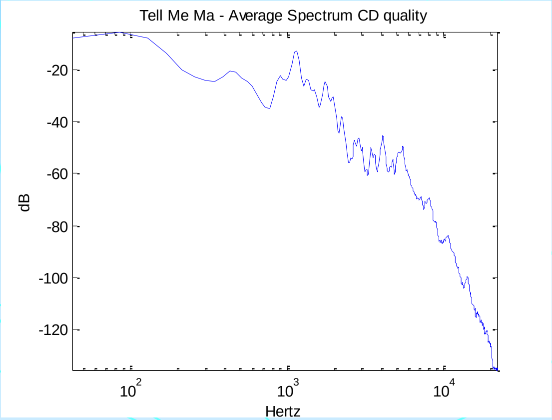


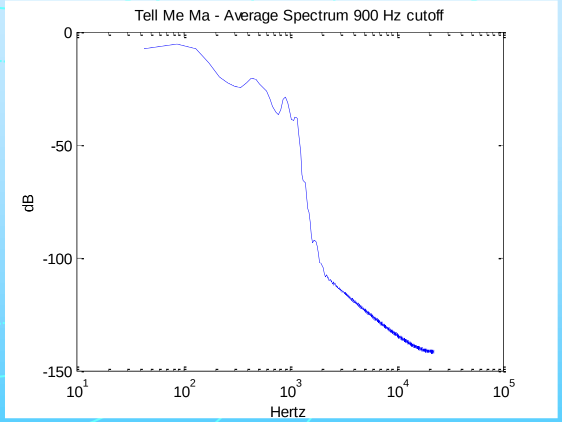
Spectrogram at Telephone Rate Sound



Bandwidth and Sampling Errors

- Original Sound
- Limited Bandwidth (LPF with 900 Hz cutoff) and sampled at 2 kHz
- Original Sound sampled at 2 kHz (aliasing)





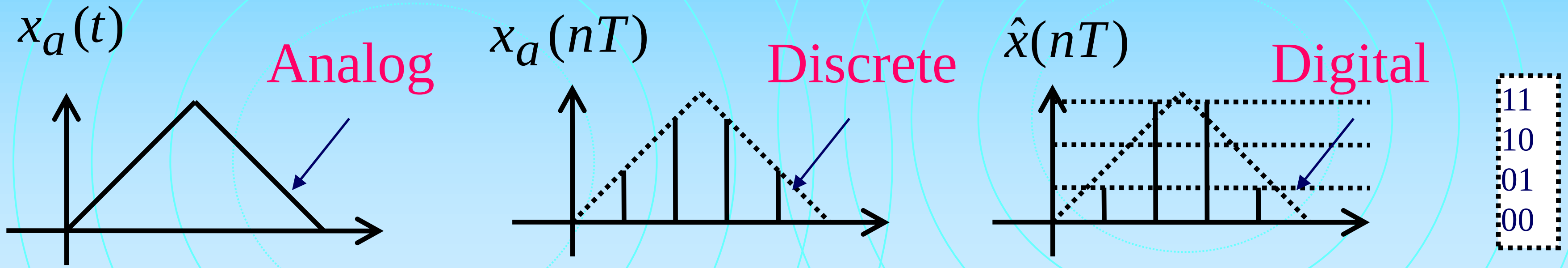
Dynamic Range and Audible Sound

- Intensity changes less than 1 dB in intensity typically are not perceived by the human auditory system.
- 25 tones at 1 kHz, decreasing in 3 dB increments
- The human ear can detect sounds from $1x10^{-12}$ to 10 watts / meter^2 (130 dB dynamic range)

Quantization Levels and Dynamic Range

- An N bit word can represent 2^N levels
- For audio signal an N bit word corresponds to: $N \times 20 \times \log_{10}(2)$ dB dynamic range
- 16 bits achieve a dynamic range of about 96 dB. For every bit added, about 6 db is added to the dynamic range.

Quantization Error and Noise



- Quantization has the same effects as adding noise to the signal:

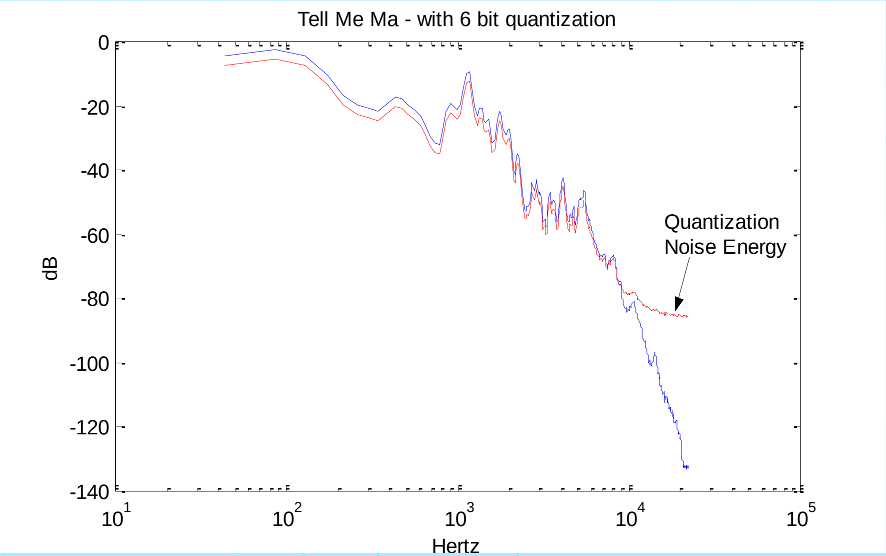
$$n_q(nT) = x_a(nT) - \hat{x}(nT) \qquad x_a(nT) - n_q(nT) = \hat{x}(nT)$$

- Intervals between quantization levels are proportional to the resulting quantization noise.

- For uniform quantization, the interval between signal levels is the maximum signal amplitude value divided by the number of quantization intervals.

Quantization Noise

- Original CD clip quantized with 6 bits at original sampling frequency
- 6 bit quantization at 2 kHz sampling



Encoding and Resources

- Pulse code modulation (PCM) encodes each sample over uniformly spaced N bit quantization levels.

- Number of bits required to represent *C* channels of a *d* second signal sampled at *Fs* with *N* bit quantization is:

  *d*C*N*Fs* + bits of header information

- A 4 minute CD quality sound clip uses Fs=44.1 kHz, C=2, N=16 (assume no header):
  - File size = (4*60)*2*16*44.1k = 338.688Mb (or 42.336MBytes)
  - Transmission in real time requires a rate greater than 1.4 Mb/s

Compression Techniques

- Compression methods take advantage of signal redundancies, patterns, and predictability via:
  - Efficient basis function transforms (wavelet and DCT)
  - LPC modeling (linear predictive coding)
  - CLPC (code excited linear prediction)
  - ADPCM (adaptive delta pulse code modulation)
  - Huffman encoding

File Formats

- Critical parameters for data encoding describe how samples are stored in the file
  - signed or unsigned
  - bits per sample
  - byte order
  - number of channels and interleaving
  - compression parameters

File Formats

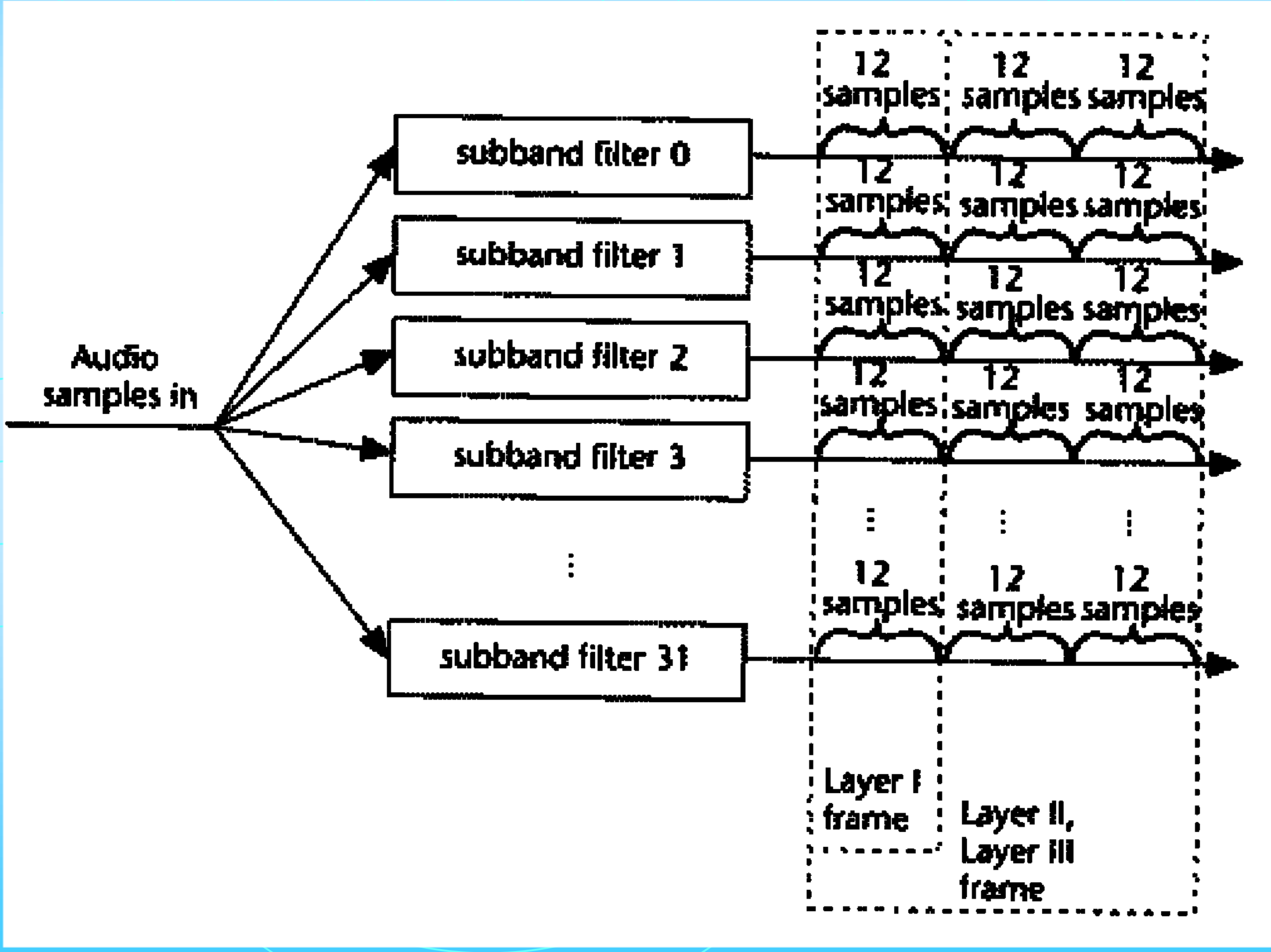
Extension, name	origin	variable parameters (fixed; Comments)
.Au or .snd	next, sun	rate, #channels, encoding, info string
.aif(f), AIFF	apple, SGI	rate, #channels, sample width, lots of info
.aif(f), AIFC	apple, SGI	same (extension of AIFF with compression)
.Voc	Soundblaster	rate (8 bits/1 ch; Can use silence deletion)
.Wav, wave	Microsoft	rate, #channels, sample width, lots of info
.sf	IRCAM	rate, #channels, encoding, info
None, HCOM	Mac	rate (8 bits/1 ch; Uses Huffman compression)

More details can be found at:

- http://www.mcad.edu/guests/ericb/xplat.aud.html
- http://www.intergate.bc.ca/business/gtm/music/sndweb.html#files
- http://www.soften.ktu.lt/~marius/audio.descript.html
- http://www.dspnet.com/TOL/newsletter/vol2_issue1/video_streaming.html

Subband Filtering and MPEG

- Subband filtering transforms a block of time samples (frame) into a parallel set of narrow band signal

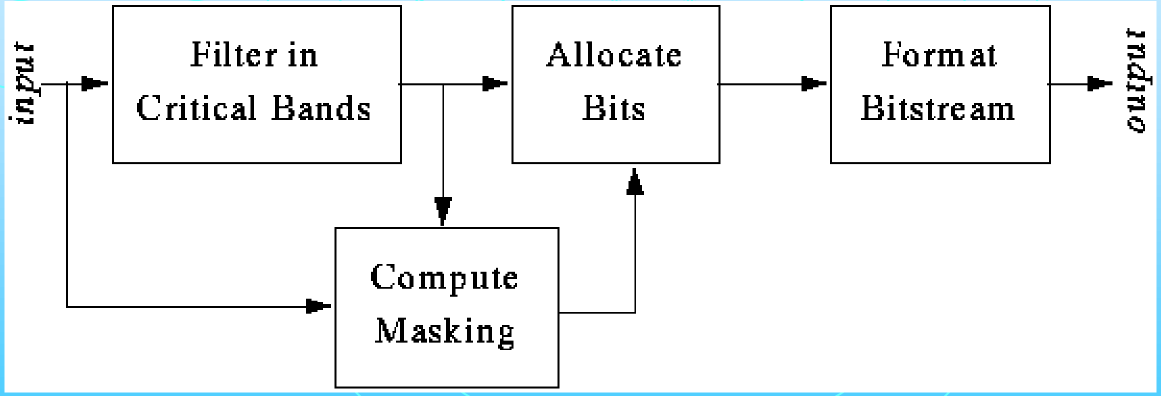


MPEG Layers

- MPEG defines 3 layers for audio. Basic model is same, but codec complexity increases with each layer.
- Divides data into frames, each of them contains 384 samples, 12 samples from each of the 32 filtered subbands.
  - Layer 1: DCT type filter with one frame and equal frequency spread per band. Psychoacoustic model only uses frequency masking (4:1).
  - Layer 2: use three frames in filter (before, current, next, a total of 1152 samples). This models some temporal masking (6:1).
  - Layer 3: better critical band filter is used (non-equal frequencies), psychoacoustic model includes temporal masking effects, takes into account stereo redundancy, and uses Huffman coder (12:1).

MPEG - Audio

- Http://fas.sfu.Ca/cs/undergrad/CourseMaterials/cmpt479/material/notes/chap4/chap4.3/chap4.3.Html
- Steps in algorithm:
  - Filters audio signal (e.g. 48 kHz sound) into frequency subbands that approximate the 32 critical bands --> *sub-band filtering*.
  - Determine amount of masking for each band caused by nearby band (this is called the *psychoacoustic model*).
  - If the power in a band is below the masking threshold, don't encode it. Otherwise, determine number of bits needed to represent the coefficient such that noise introduced by quantization is below the masking effect.
  - Format bitstream



Example

- After analysis, the first levels of 16 of the 32 bands are these:

Band	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16
Level (db)	0	8	12	10	6	2	10	60	35	20	15	2	3	5	3	1

- **If the level of the 8th band is 60db,**
- **It gives a masking of 12 db in the 7th band, 15db in the 9th.**
- **Level in 7th band is 10 db (< 12 db), so ignore it.**
- **Level in 9th band is 35 db (> 15 db), so send it.**
- **--> Can encode with up to 2 bits (= 12 db) of quantization error**