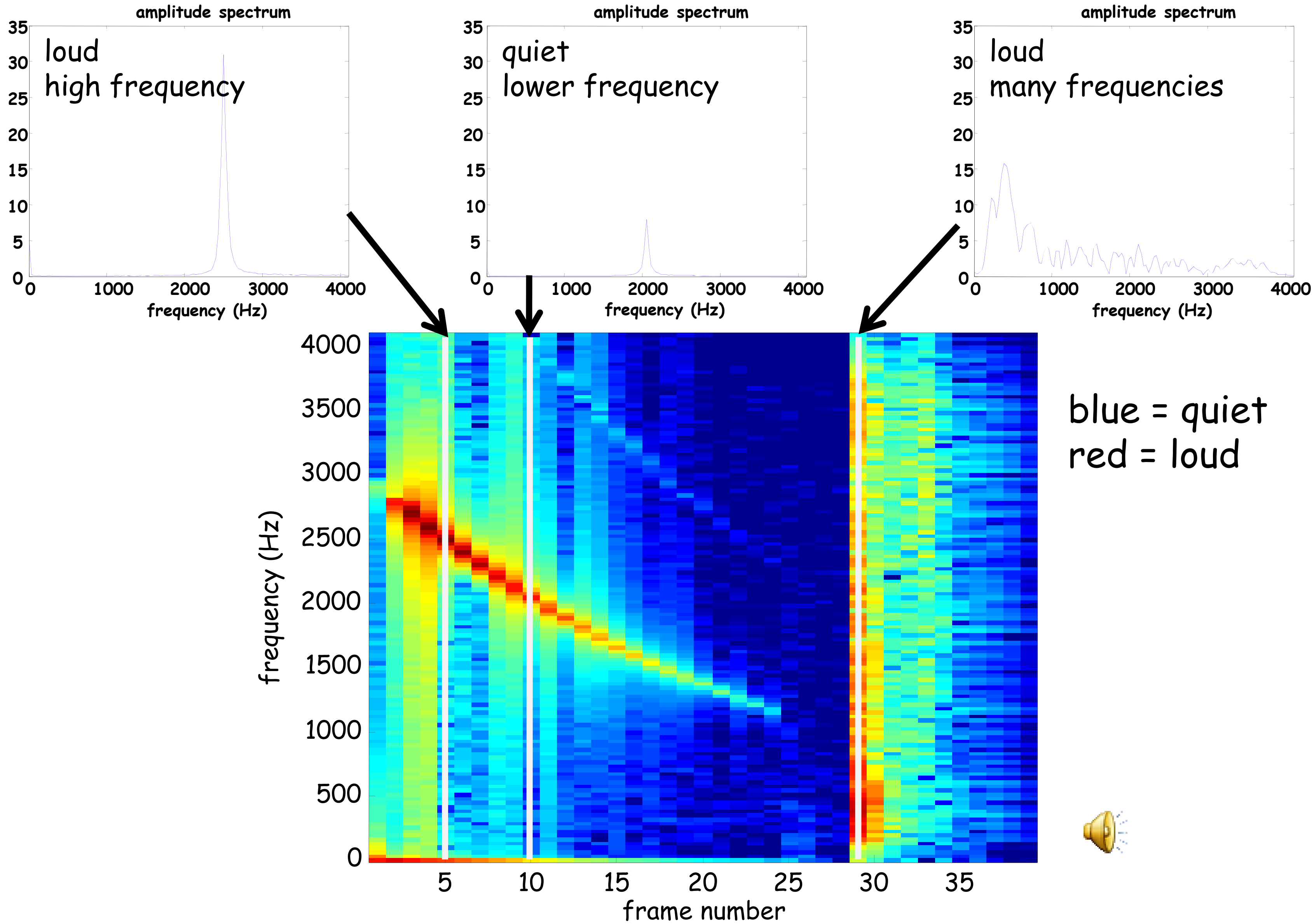
ELEC1200: A System View of Communications: from Signals to Packets Lecture 14

- Time-Frequency Analysis
  - Analyzing sounds as a sequence of frames
  - Spectrogram
- Lossy Encoding
  - MP3 encoding

Time-Frequency Analysis

- For many complex signals (like speech, music and other sounds), short segments are well described by a sinusoidal representation with a few important frequency components, but long segments are not.
- Time-frequency analysis refers to the analysis of how short-term frequency content changes over time.
- The spectrogram of a signal is a picture of how its amplitude spectrum of a signal changes over time.
  - The vertical axis represents frequency
  - The horizontal axis represents time
  - The image color represents Fourier amplitude
    - red = large amplitude, blue = small amplitude

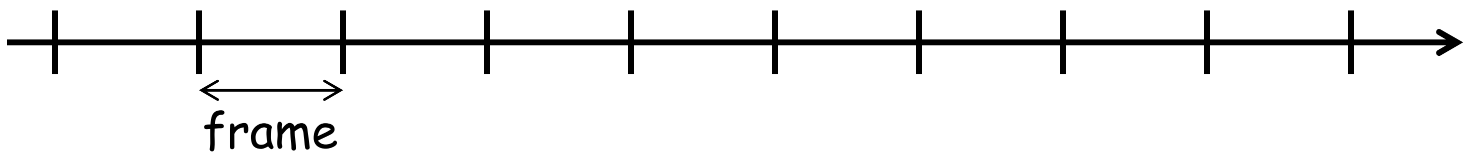
Spectrogram Example



blue = quiet
red = loud

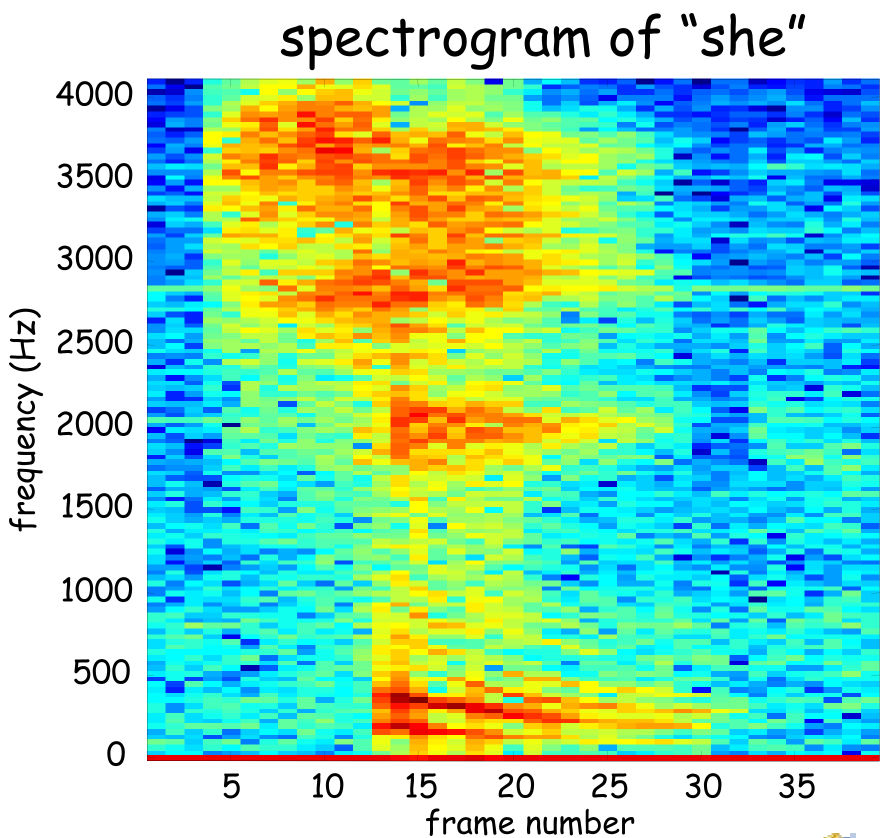
Computation of the Spectrogram

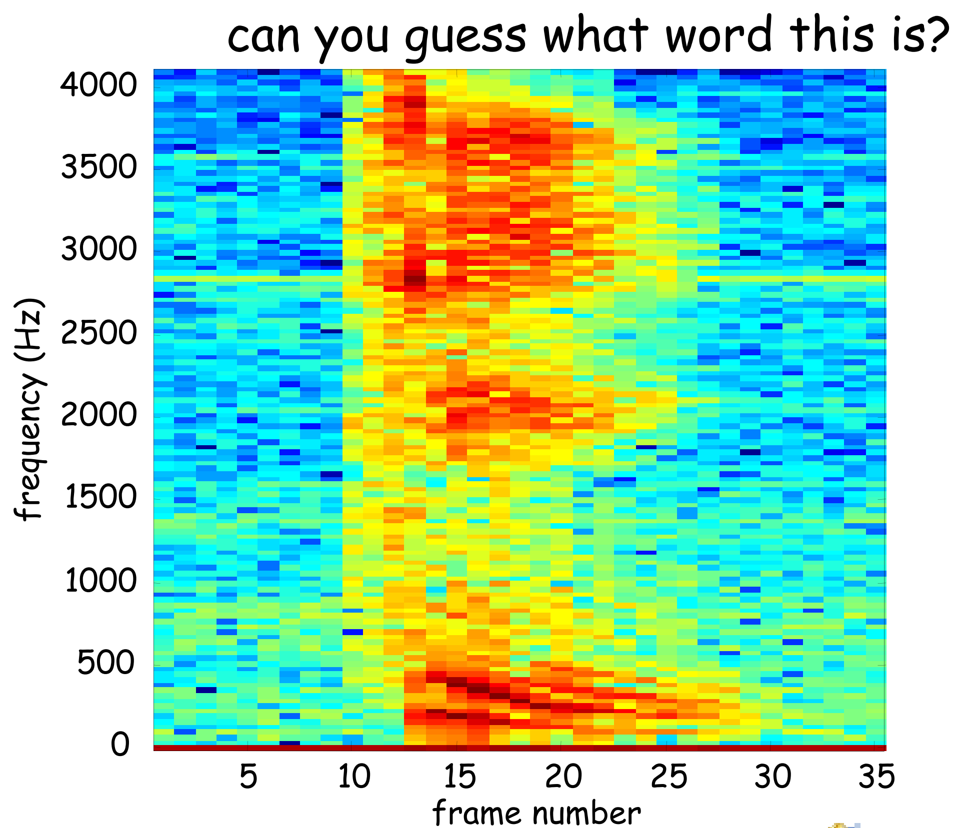
- **Divide the signal into a set of frames, typically about 20-50ms long.**



- **Compute the amplitude spectrum of each frame.**
- **This gives you a two dimensional array of real numbers, indexed by frame number and frequency index.**
- **Plot this as an image.**
  - **It is generally more informative to plot the logarithm of the amplitude, as this compresses large amplitudes allowing the smaller details to show up.**
  - **To avoid problems at zero, apply a small positive floor to the values (i.e. replace each amplitude by P if it is smaller than P, where P is small).**

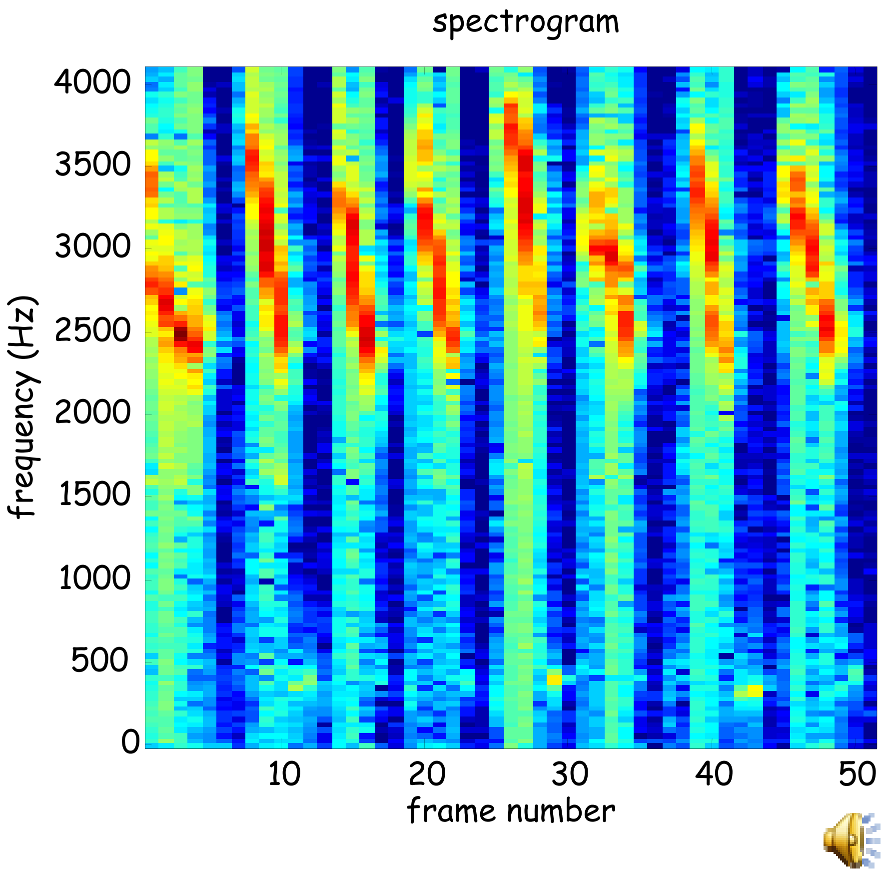
Speech Spectrogram

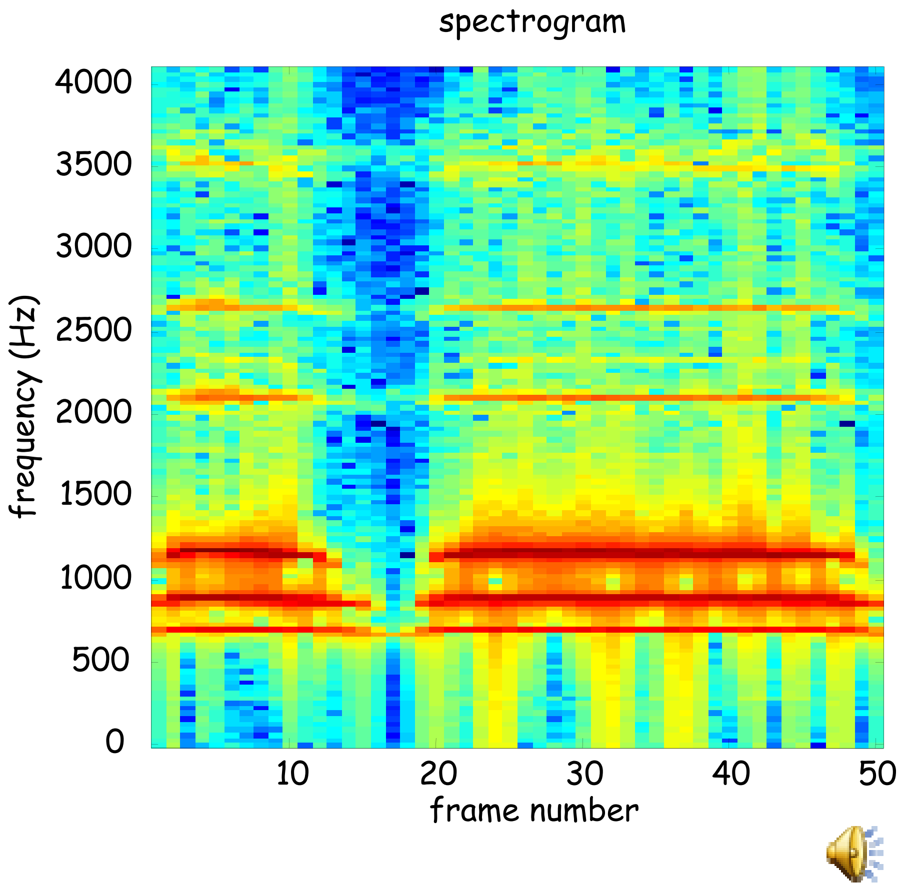




Train Whistle vs. Bird Chirps

- **Can you figure out which is which?**

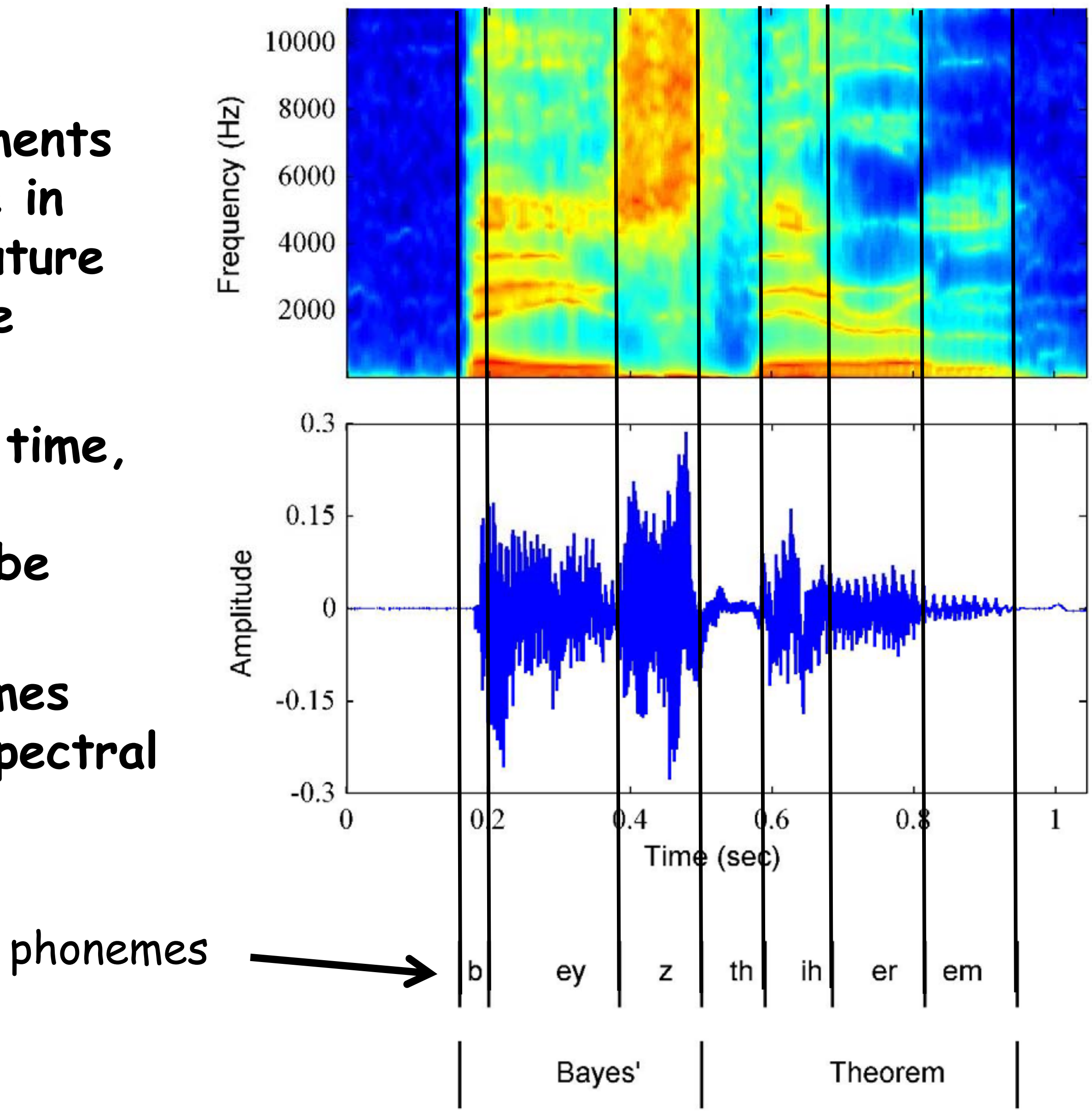




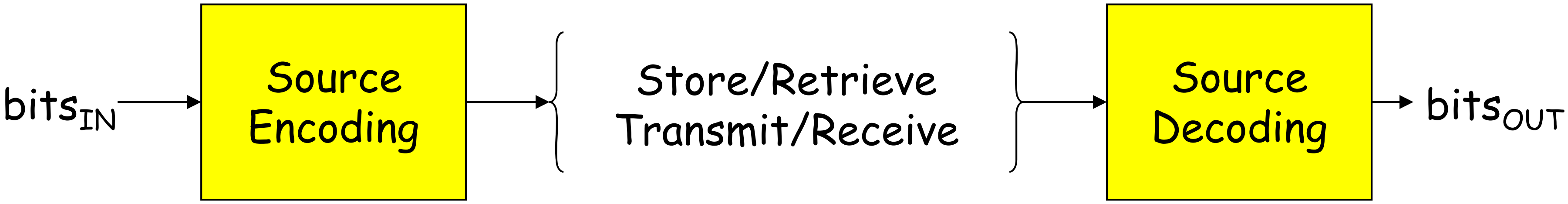
Speech Data

red = high energy
blue = low energy

- **Characteristics**
  - **Recent measurements more informative in predicting the future than those in the distant past.**
  - **At each point in time, different sounds (phonemes) may be pronounced.**
  - **Different phonemes have different spectral content.**

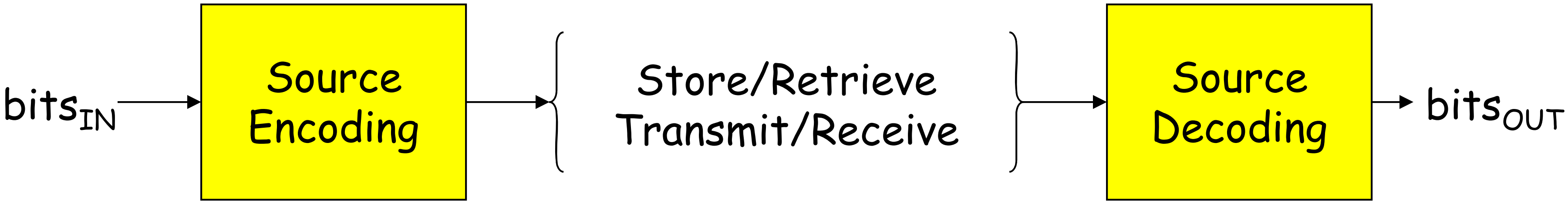


Encoding and Decoding



- **Encoding**
  - **Auditory signal (from a recording) is coded into an mp3 file containing carefully stored spectral information**
- **Decoding**
  - **mp3 file is turned back into an auditory file that can be output to your speakers**

Lossless vs. Lossy Compression



- For *lossless* data compression: $bits_{OUT} = bits_{IN}$
  - We can reconstruct the original bit stream exactly
  - $bits_{OUT} = bits_{IN}$
  - Usually used for "naturally digital" bit streams, e.g. documents, messages, datasets, …
  - Examples: Huffman encoding, LZW, zip files, rar files
- For *lossy* encodings: $bits_{OUT} \approx bits_{IN}$
  - "Essential" information preserved
  - Appropriate for sampled data streams (audio, video) intended for human consumption via imperfect sensors (ears, eyes).

MP3

- **MPEG is moving pictures experts group**
  - **set up by ISO (international standards organization)**
  - **every few years issues a standard**
    - **MPEG1 (1992)**
    - **MPEG2(1994)..**
- **MP3 stands for MPEG audio layer III**
- **MP3 achieves a 10:1 compression ratio!**
- **This enables**
  - **bit-streaming**
  - **compact audio storage**

Bad ways to compress an audio file

- Reduce the total number of bits per sample
  - e.g. 32 bit to 16 or 16 to 8 bit
  - Gives you a factor of 2 in compression
  - However, perceptual quality of signal is poor
- Reduce the sampling rate
  - 44kHz to 22kHz or 22kHz to 10kHz
  - Again only a gain of a factor of 2 in size.
  - Total loss of all high frequency information.
  - Equivalent to a high pass filter.
- A factor 10:1 in compression cannot be achieved using linear compression schemes

Perceptual Coding

- Start by evaluating input response of bitstream consumer (eg, human ears or eyes), i.e., how consumer will perceive the input.
  - Frequency range, amplitude sensitivity, color response, …
  - Masking effects
- Identify information that can be removed from bit stream without perceived effect, e.g.,
  - Sounds outside frequency range
  - Masked sounds
- Encode remaining information efficiently
  - Use frequency-based transformations
  - Quantize coefficients of frequency (loss occurs here)
  - Add lossless coding (e.g. the Huffman encoding to be studied later)

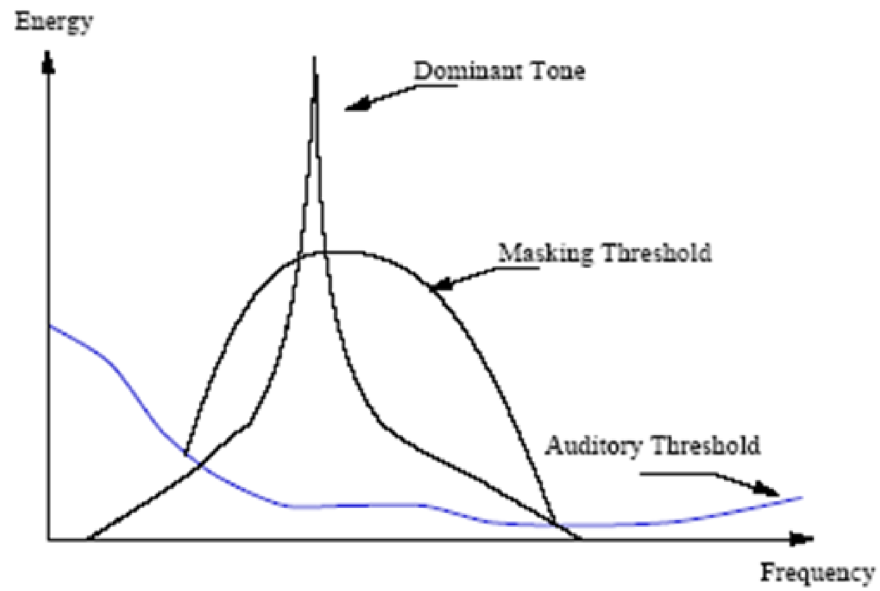
Principles of Auditory Coding

- Time frequency decomposition
  - divide the signal into frames
  - obtain the spectrum of each piece
- Use psycho-acoustic model to determine what information to keep
  - Don't store information outside the range of hearing (40Hz to 15kHz)
  - Stereo info not stored for low frequencies
  - Masking
- Store the information in the most compact way possible
  - minimize the bitrate
  - maximize the audible auditory content

Masking

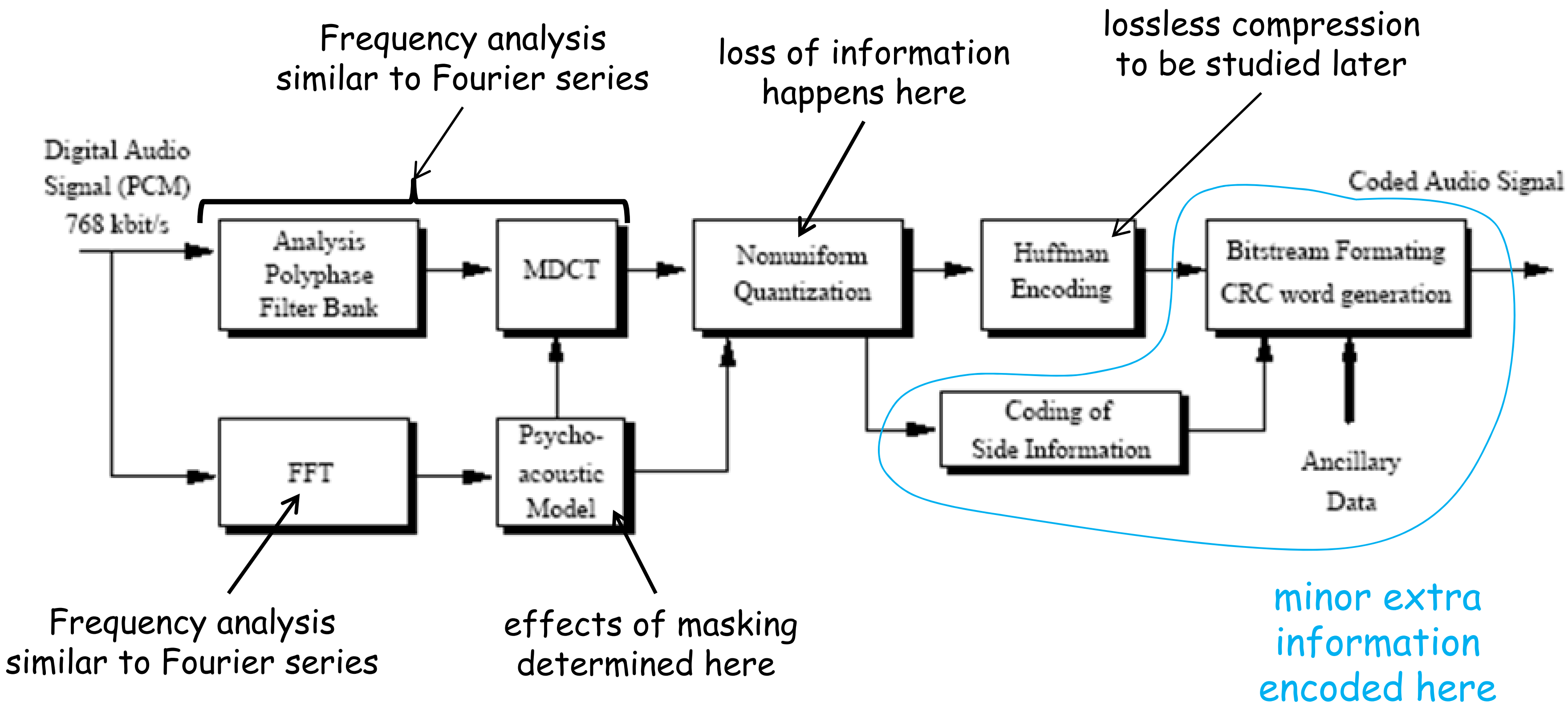
- **If a dominant tone is present then noise can be added at frequencies next to it and this noise will not be heard.**
- **Coding consequences**
  - Less precision is required to store nearby frequencies
  - Less precision = coarser quantization

- Definitions
  - Auditory threshold – minimum signal level at which a pure tone can be heard
  - Masking threshold – minimum signal level if a dominant tone is present



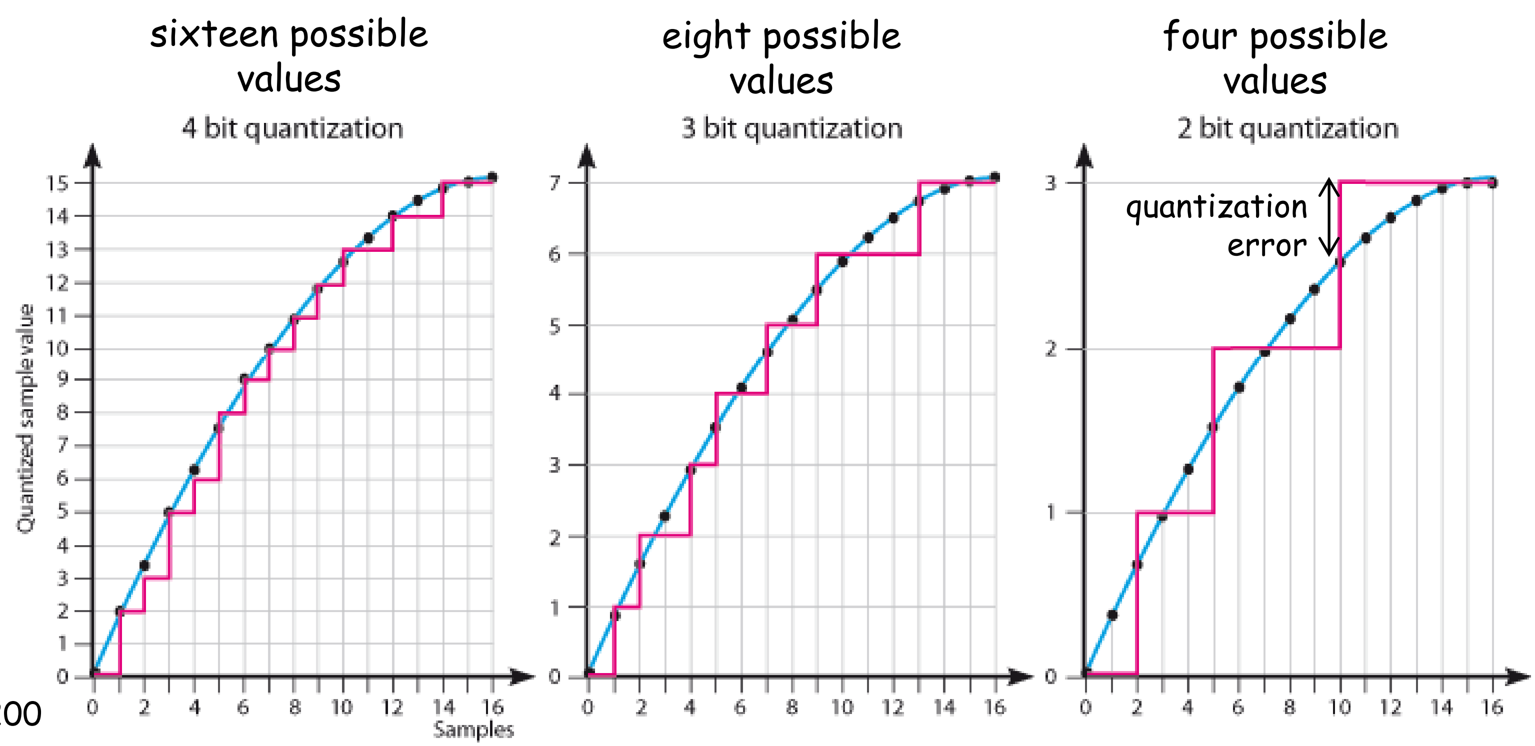
MP3 schematic

- **Input: 16 bit at 44kHz sampling is 768kbit/s**
- **Output: Coded audio signal at ~128kbit/s**



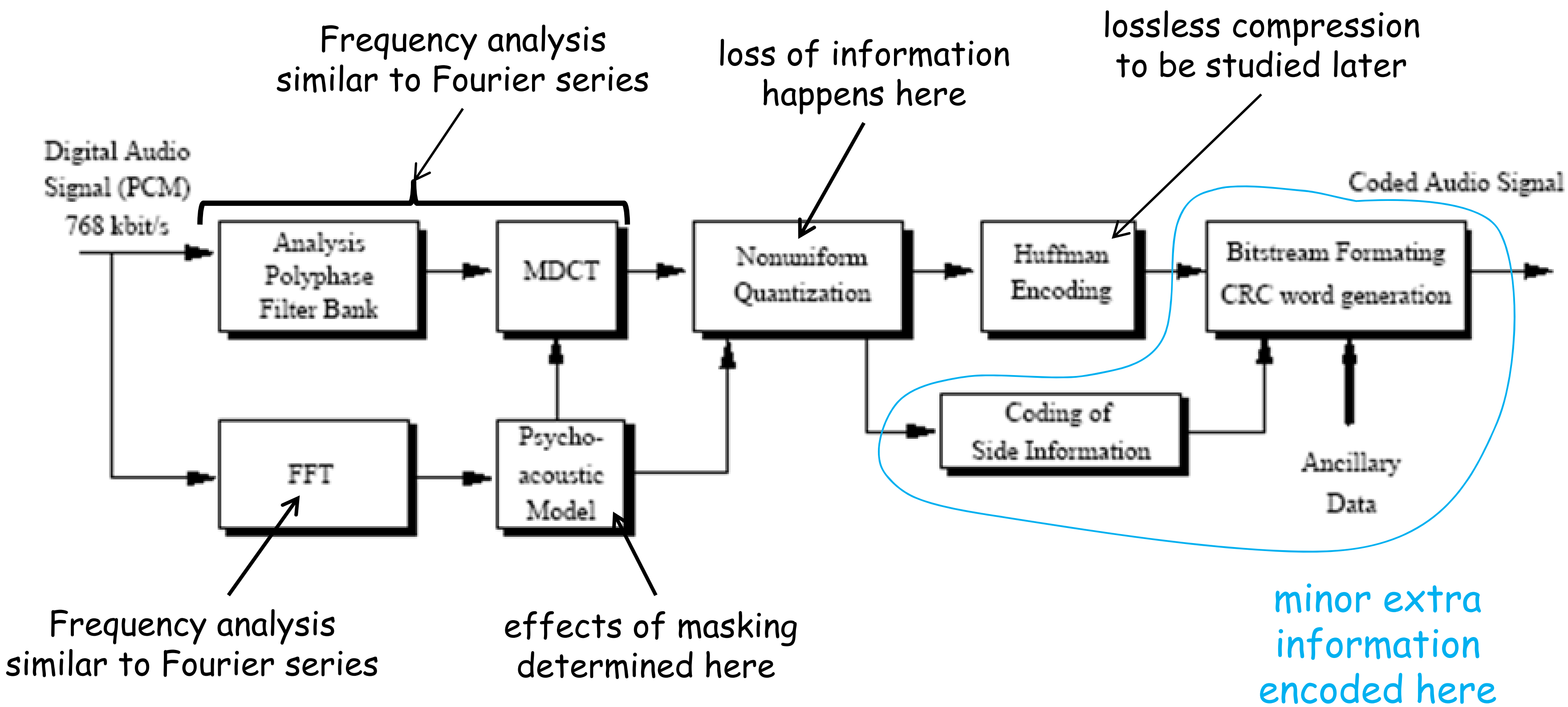
Quantization

- **When creating a digital representation of a sampled value, we must choose the number of bits used to represent the value.**
- **Since fewer bits encode a smaller number of values, using fewer bits results in a larger quantization error**
  - **quantization error = difference between the actual and encoded value**
  - **We want quantization error to be small, i.e. more bits.**
- **On the other hand, fewer bits take up less space.**



Non-uniform quantization

- **MP3 compression quantizes the amplitudes of different frequency components differently, depending upon masking.**
- **Frequency components near a dominant masker are quantized with lower bits.**



Summary

- **Audio waveforms are typically analyzed as a sequence of frames**
  - Within each frame, the signal can be well approximated by a few frequency components
  - The spectrogram can be used to visualize changes in the frequency content over time
  - Framing is used in MP3 audio compression
- **MP3 audio compression combines framing and frequency analysis with a non-uniform quantization based on a perceptual model**
  - Quantization results in loss of information
  - By throwing away "unimportant" (imperceptible) information, we can obtain large compression ratios.
  - We will study a very simple version of this idea in the lab.