ECE442: Wireless Communications

Lecture 4: Information Sources and Analog-to-Digital (A/D) Conversion

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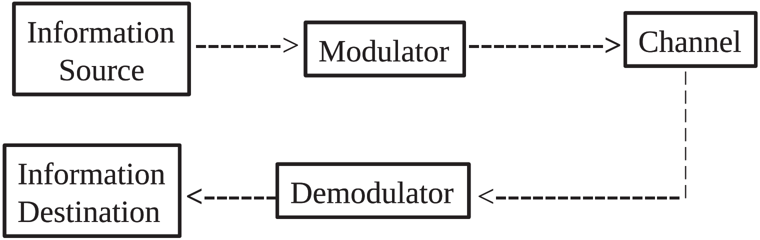
Exam Dates

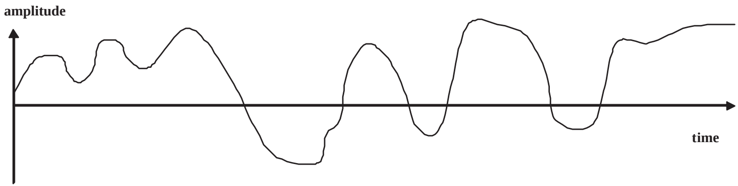
- Midterm Exam: Tuesday, Oct. 14th (in-class)
- Final Exam: Tuesday, Dec. 16th. 7.30am - 9.30am (check the UNM schedule for changes)

An Abstract Model for a Communications System



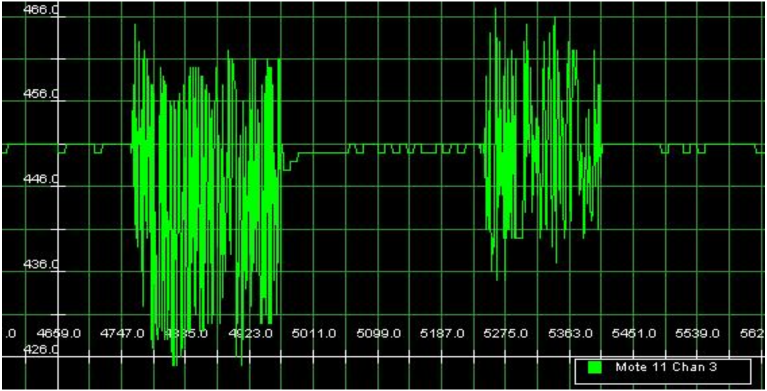
Everything starts with a source

Information Sources



- Information source generates messages to be transmitted over the channel to a destination
  - A message could be speech, music, data, images, video etc.
- Information sources can be classified into two basic types depending on the messages they produce
  1. Analog sources
  2. Digital sources

- Generated information messages are continuous functions of time (or space)
  - voice, music, photographs and pictures, video etc.



- Generated information messages are sequences (files) of discrete values (often 0's and 1's)

. . . 0101001110110011101011000110011010000 1111 . . .

text, compact discs, MP3, DVD, HDTV, Satellite TV, MPEG, JPEG, GIF, HTML files etc.

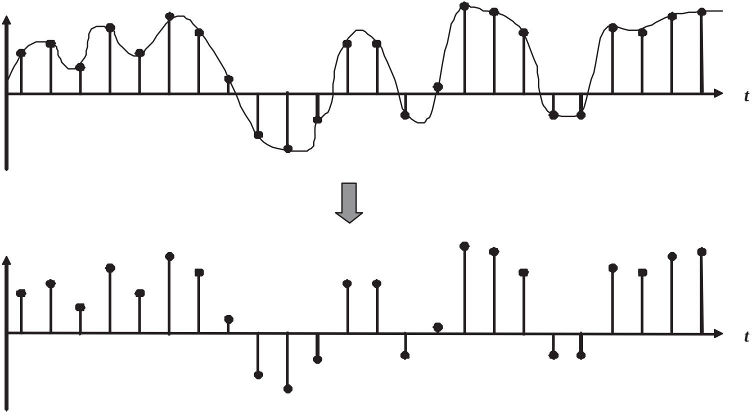
Digitization of Analog Sources

- Most information sources in nature are inherently analog (e.g. voice, music etc)
- But transmission of information in digital form is desirable for various reasons:
  - Encoding (to protect against channel noise and fading)
  - Compression (to reduce required transmission rates and thus resources)
  - Encryption (to prevent eavesdropping and interception)
- Thus, analog messages are often needed to be converted into digital form
  - This process is called Analog-to-Digital conversion (ADC)

Analog-to-Digital (A/D) Conversion

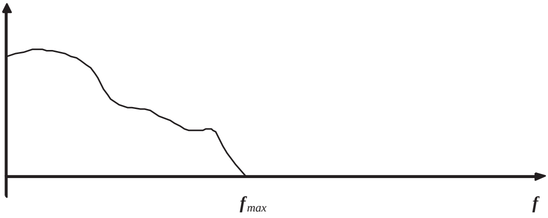
- A/D conversion involves three main steps:
  1. Sampling (time digitization)
  2. Quantization (amplitude digitization)
  3. Compression (removes redundancy)

Sampling of Analog Sources



- Sampling converts a continuous signal into a sequence of numbers
- If the sampling rate is chosen appropriately, then the sampling does not make any information loss

Shannon Sampling Theorem



Theorem (Shannon Sampling Theorem)

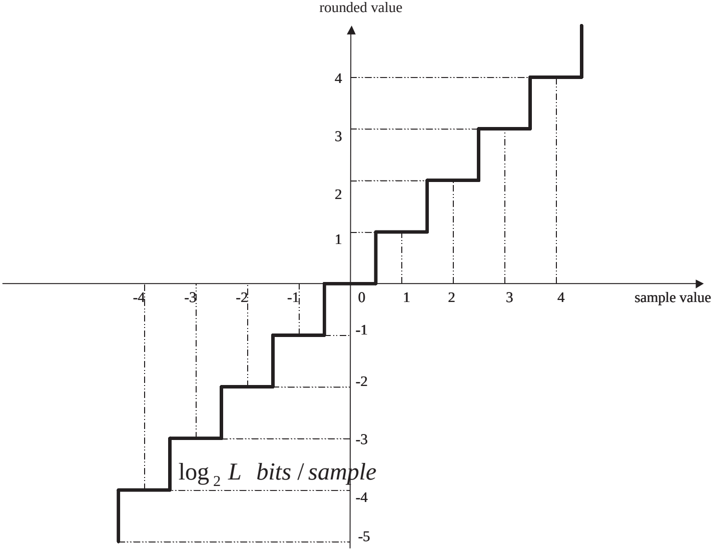
Suppose, the maximum frequency in the source message spectrum is f_{max}. Then a sampling rate of $2f_{max}$ is sufficient to capture all the information in the source (i.e. to perfectly reconstruct the original signal from the sampled values).

$$Nyquist\ Rate = 2f_{max}$$

Quantization

- A sampled signal is called a discrete-time signal.
- A discrete-time signal can take a continuum of values
- But digital signals are only allowed to take some (finite) set of values:
  - Sampled signals are quantized (converted into discrete values)
- Quantized output is truncated at a maximum level
  - e.g. round off the sampled output value to the nearest integer etc. and clip-off at a maximum level
- Number of allowed quantizer levels determines how many bits (binary digits) are needed to represent each sample:
  - For example, if the quantizer can have maximum of L levels then $\log_2 L$ bits per sample is required

Quantization: An Example



PCM: Pulse Code Modulation

- A sampled and quantized signal is called a Pulse Code Modulated (PCM) signal
  - A message is represented by a sequence of coded pulses by discreteizing it in both time and amplitude
- PCM is the most basic digital pulse modulation method
  - If the sampling rate of the analog signal is f_s samples per second, and if the quantizer has L levels, then the bit rate of the digital PCM signal is $f_s \log_2 L$ bits per second

PCM Examples: Voice and CD Audio

1. Toll quality voice:
   - Voice is sent over (wire-line) telephone switching systems as PCM
     - Sampling rate = 8000 samples/second
     - $L = 256$ (or 8 bits/sample)
     - Bit rate = 64 kbps
2. CD quality audio:
   - Audio is collected as PCM for CD storage
     - Sampling rate = 44100 samples/second for high fidelity music having a bandwidth of 20kHz
     - $L = 65536$ (or 16 bits/sample) and uniform quantization
     - Bit rate = 705.6 kbps
     - Bit rate for stereo (2 channels) = 1.4112 Mbps
   - A CD has a total track length of 5300m and a scan velocity of 1.2m/s, giving a total playback time of 74 mins

PCM Examples: Images and HDTV

3. Images
   - Image samples are usually referred to as *pixels*
   - In a low resolution image, for example,
     - Pixels per linear inch $= 72$
     - Hence, pixels per square inch $= 5184$
     - Quantizer has 8 Bits/color/smaple $\longrightarrow$ i.e. a total of 24 bits/sample
     - $5'' \times 7''$ image has about 4.4 Mbits of data
4. HDTV:
   - PCM rate of the video part of an HDTV signal is about 1 Gbps

Compression

- Most communication channels are band-limited (e.g. radio channel).

  Above PCM rates are too high for transmission via such band-limited channels
- Compression of the PCM signals are required to reduce the required transmission rate. There are two types:
  - Lossless Compression: removing redundancy in data without loosing any information at all, so that the process is completely reversible (e.g. pzip)
  - Lossy Compression: compress data within some tolerable level of distortion. Not fully reversible
- Practical compression techniques for speech, images, audio and video involve lossy compression to an imperceptible level followed by lossless compression to remove redundancy

Compression Examples: Differential Pulse Code Modulation (DPCM)

- Instead of the actual sample values, the differences between successive samples are quantized
- Allows same quality with only a fewer quantization levels
- Sometimes used in coding voice signals
  - e.g. DPCM provides a 2-to-1 compression in cordless phones (resulting in a rate of only 32 kbps)

Compression Examples: Linear Predictive Coding (LPC)

- Similar to DPCM, but instead of quantizing differences between successive samples, the difference between the sample and a linear prediction of it formed using a number of past samples is quantized
- Various LPC schemes are employed in voice coding
  - In digital cellular, LPC can provide anywhere from 4-to-1 up to 8-to-1 compression (resulting in reasonable quality voice signals with rates from 8 kpbs to 16 kbps)

Compression Examples: MP3 (Sub-band Coding)

- Different frequency bands are quantized with different numbers of quantization levels
- Allows same quality with only a fewer quantization levels
- Used in compressing audio
  - e.g. Can provide about 10-to-1 compression to bring CD quality stereo audio rate down to about 128 kbps!

Compression Examples: JPEG (Image Compression Standard)

- 8×8 pixel blocks are lossy compressed using *transform coding*, and then lossless compressed to remove redundancies
- Uses Discrete cosine transform (DCT)
  - Can provide up to 24-to-1 compression ratios (resulting in 1 bit per pixel)
  - Actual ratio depends on the type of the picture and the desired quality
- JPEG2000 replaces DCT by DWT

Compression Examples: MPEG (Video Compression Standard)

- Similar to JPEG with motion estimation and differential coding
- Various MPEG versions are available: MPEG1, MPEG2, MPEG4
  - MPEG version in HDTV (MPEG2) provides about 50-to-1 compression ratio to bring the original PCM signal down to about 20 Mbps
  - Can also provide lower quality video at about hundreds of kbps rates
  - Even lower rates are possible for low bit rate video such as streaming video
- MPEG standard is based on a hybrid block-based DPCM/DCT coding scheme

References

- S. Haykin and M. Moher *Introduction to Analog and Digital Communications*, Second Edition, John Wiley, 2007. Chapter 5.
- A. B. Carlson, P. B. Crilly and J. C. Rutledge, *Communication Systems*, Fourth Edition, McGraw-Hill: New York, 2002. Chapters 7 and 12.
- J. G.Proakis, *Digital Communication*, Fourth Edition, McGraw-Hill: New York, 2001. Chapter 3.

Next time...

Next time: Digital Modulation and Demodulation

- References: D. P. Agrawal and Q. Zeng, *Introduction to Wireless and Mobile Systems*, Second Edition, Thomson, 2006. Chapter 7.