ECE4634 Digital Communications Fall 2007

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Lecture #7: Digital Pulse

Modulation: PCM



Analog and Digital Communications

Overview

- We are currently studying baseband communication systems
- Previously we looked at
 - Sampling continuous time waveforms
 - Nyquist sampling theorem
 - Representing time samples using discrete baseband communication systems where samples are used to modulate a pulse train
- Digital systems are discrete in time and have a finite number of messages
- Pulse Code Modulation is one form of digital baseband communications
 - Often called analog-to-digital conversion since it is so common
 - However, this is only one form of analog-to-digital conversion
- What to read 5.4-5.6





- In this lecture we will
 - Introduce the digital communication concept of Pulse Code Modulation
 - Examine (uniform scalar) quantization in detail
 - Derive the SNR of quantization
 - Demonstrate the trade-off that exists in quantization between fidelity and bit rate (i.e., bandwidth)



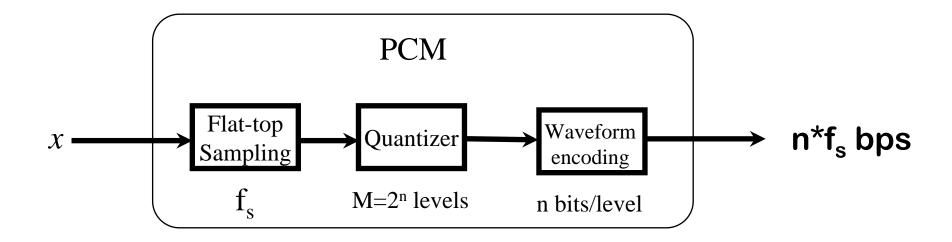


- Modulation is changing a parameter of a carrier in order to convey information
- Two common carriers are
 - Sinusoid (bandpass systems)
 - Pulse stream (baseband systems)
- In PCM (Pulse Code Modulation) we modulate a stream of pulses (typically we assume square pulses for convenience)

Pulse Code Modulation (PCM)



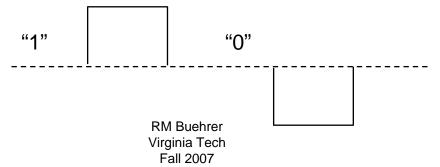
 Pulse Code Modulation refers to a system that creates a baseband signal that is generated directly from the binary quantizer output





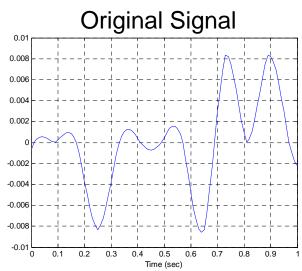


- Once the information is converted to bits, it must be mapped onto waveforms (i.e., modulation)
- If each bit is mapped to one of two different waveforms, we term this binary encoding
- If m bits are mapped to $M = 2^m$ waveforms, we term this M-ary encoding
- We will discuss specific waveforms next class
- For now assume that we use the mapping

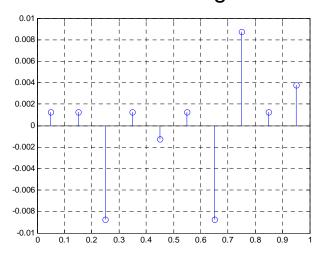




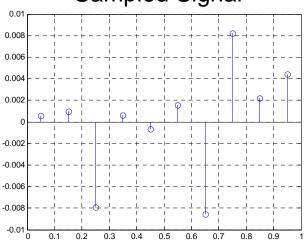




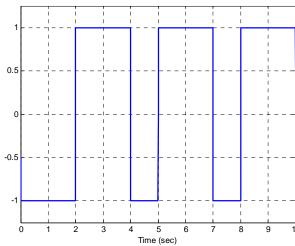
Quantized Signal



Sampled Signal



Resulting Digital Signal



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Bandwidth of PCM Signals



- Sample rate: f_s samples/second.
- Bit rate out of the quantizer:

$$f_S \log_2 M = f_S \cdot n \text{ bits / second}$$

- Bandwidth of the resulting digital signal depends on the pulses used
- Minimum theoretical bandwidth (with optimal pulse shaping which we will discuss later): $f_s \cdot n/2$ Hz
- First null bandwidth (with rectangular pulse waveforms):

$$f_{S} \cdot n \text{ Hz}$$

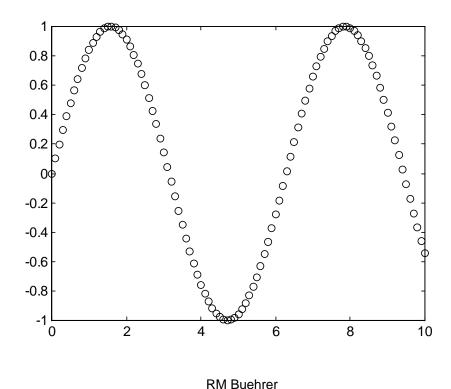
We'll talk more about the effects of pulse shaping in a few days

Note that the resulting bandwidth depends on the digital waveform (i.e., pulse shape) that is used.

Digital Representation of Analog Signals



Sampling analog signals makes them discrete in time:

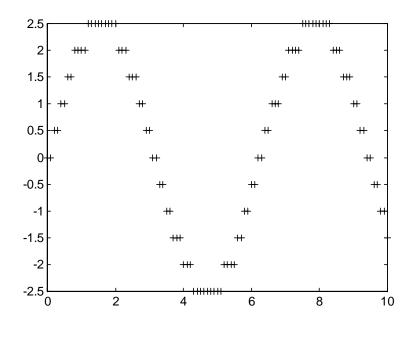


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Digital Representation of Analog Signals



 Quantization of sampled analog signals makes the samples discrete in amplitude:



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Quantization



- Continuous time signals are sampled at discrete time intervals
- Sampling may be performed without distortion provided signal is sampled at Nyquist rate
- Continuous-valued samples of data require an infinite # of bits to represent with perfect precision.
- Quantization is the process of approximating continuousvalued samples with a finite # of bits.
- Quantization always introduces some distortion.

Notation Associated with Quantization



- Let X be a random variable representing a sample of data.
- Then $\widetilde{X} = f_O(X)$ is the quantized value of X.
- A quantizer has M quantization levels:

$$\widetilde{X} \in \{\widetilde{x}_1, \widetilde{x}_2, \dots, \widetilde{x}_M\}$$

- The M levels correspond to M quantization regions.
- The endpoints of the quantization regions are specified by M+1 values: $\{x_0,x_1,\ldots,x_M\}$, where $x_0=-\infty,x_M=\infty$
- Then:

$$x_{k-1} \le x < x_k \implies \widetilde{x} = f_Q(x) = \widetilde{x}_k$$

Graphical Description of Quantization



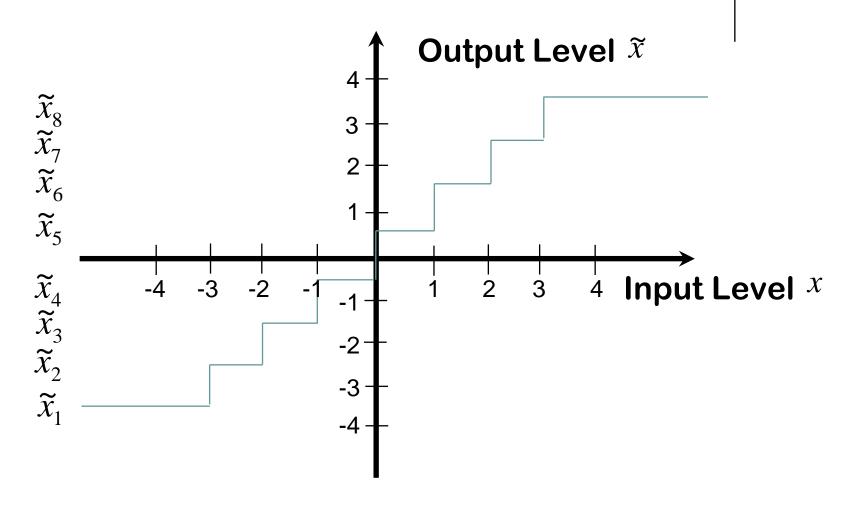


Table Representation of Quantizer



k	x_{k-1}	x_k	\widetilde{x}_k	Output
				Bits
1	.—∞	-3	-3.5	000
2	-3	-2	-2.5	001
3	-2	-1	-1.5	010
4	-1	0	-0.5	011
5	0	1	0.5	100
6	1	2	1.5	101
7	2	3	2.5	110
8	3	∞ .	3.5	111

Concise Representation of Quantizer



- Usually, it is sufficient just to list the quantization levels.
- Example: {-3.5, -2.5, -1.5, -0.5, 0.5, 1.5, 2.5, 3.5}
- Why?
 - We assume that all points are quantized to the nearest quantization level
 - This determines where the borders of the quantization regions are
 - Any other borders would increase the error introduced by the quantizer

Practical Methods for Implementing Analog to Digital Converters



Lower Complexity

- Counting or Ramp ADC
 - test value is incremented in equal steps until it is greater than input sample
- Serial or Successive Approximation ADC
 - uses binary search to narrow range of input sample until desired accuracy is reached
- Parallel or Flash ADC
 - input sample is compared with all possible quantization levels at once

Faster

Distortion



- Quantization introduces distortion into a signal.
- ullet We want to minimize average distortion D , where

$$D = E\left[\left(X - \tilde{X}\right)^{2}\right]$$

$$= \int_{-\infty}^{\infty} \left(x - \tilde{x}\right)^{2} f(x) dx$$

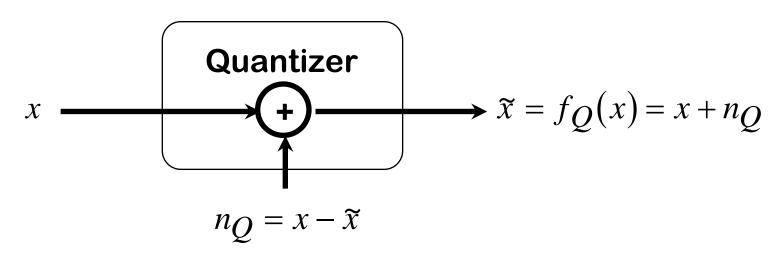
$$= \sum_{k=1}^{M} \int_{x_{k-1}}^{x_{k}} \left(x - \tilde{x}_{k}\right)^{2} f(x) dx$$

- This measure of distortion is sometimes also called mean square error (MSE)
- MSE penalizes large errors more than small errors

Another Way of Viewing Quantization



- Quantization adds a random "noise" to the true value of the sample point___
- Then $MSE = E[n_Q^2]$ may be thought of as noise power
- We can define a signal-to-noise ratio (SNR) to measure performance



Signal to Noise Ratio Calculations for Quantizers



Average SNR

$$\left(\frac{S}{N}\right)_{\text{avg}} = \frac{E\left[X^2\right]}{E\left[n_Q^2\right]} = \frac{E\left[X^2\right]}{D} = \frac{\int_{-\infty}^{\infty} x^2 f_X(x) dx}{\int_{-\infty}^{\infty} (x - \tilde{x})^2 f_X(x) dx}$$

- Sometimes we also define "peak" SNR
 - This is less important.

$$\left(\frac{S}{N}\right)_{\text{peak}} = \frac{X_{\text{peak}}^2}{E[n_Q^2]} = \frac{X_{\text{peak}}^2}{D}$$
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Example 7.1: SNR Calculation



• Let:
$$\{\widetilde{x}_1 = -3.5, \widetilde{x}_2 = -2.5, \widetilde{x}_3 = -1.5, \widetilde{x}_4 = -0.5, \widetilde{x}_5 = 0.5, \widetilde{x}_6 = 1.5, \widetilde{x}_7 = 2.5, \widetilde{x}_8 = 3.5\}$$

• Let:
$$f(x) = \begin{cases} 1/8, & -4 \le x \le 4 \\ 0, & \text{else} \end{cases}$$

$$E\left[X^{2}\right] = \int_{-4}^{4} x^{2} \cdot \frac{1}{8} dx = \frac{x^{3}}{24} \Big|_{-4}^{4} = \frac{128}{24} = \frac{16}{3}$$

Example 7.1 (continued)



Distortion:

$$D = MSE = \sum_{k=1}^{M} \int_{x_{k-1}}^{x_k} (x - \tilde{x}_k)^2 f_X(x) dx$$
$$= \sum_{k=1}^{8} \int_{-5+k}^{-4+k} (x - (-4.5+k))^2 \cdot \frac{1}{8} dx$$

• Examine the k=5 term:

$$\int_{0}^{1} (x - 0.5)^{2} \cdot \frac{1}{8} dx = \frac{1}{8} \cdot \int_{0}^{1} x^{2} - x + 0.25 dx = \frac{1}{8} \cdot \left[\frac{x^{3}}{3} - \frac{x^{2}}{2} + 0.25 x \right]_{0}^{1}$$

$$= \frac{1}{8} \cdot \left[\frac{1}{3} - \frac{1}{2} + \frac{1}{4} \right] = \frac{1}{8} \cdot \frac{1}{12}$$
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Example 7.1 (continued)



• All 8 terms are identical. ∴ MSE = 1/12

$$\left(\frac{S}{N}\right)_{\text{avg}} = 10\log_{10}\frac{E[X^2]}{D} = 10\log_{10}\left(\frac{16/3}{1/12}\right) = 18.1\text{dB}$$

 A standard rule of thumb is that each additional bit adds 6 dB to the SNR of uniform quantizers operating on a uniform pdf.

SNR for Uniform Quantization



• General result:
$$\left(\frac{S}{N}\right)_{avg} = M^2$$

- Assumes uniform quantizer with M levels
- Assumes that input samples have uniform distribution with identical range as quantizer
- Though some texts do not make it clear, this result applies only when these special set of conditions hold. Otherwise, we have to use integral formula
- However, a useful Rule of Thumb:
 - Each additional bit (doubling M) increases SNR by 6 dB

$$\left(\frac{S}{N}\right)_{avg} = 6.02n + \alpha$$

• Where α depends on the distribution of the signal

Example 7.2: PCM Calculation



Problem:

- Suppose that an analog music signal is found to have a bandwidth of 15 kHz and that samples of the signal may be modeled as having a uniform distribution.
- Find the minimum first-null bandwidth (assuming the use of square pulses) at which it would be possible to transmit a PCM signal while maintaining an average SNR of at least 58 dB.

Example 7.2 (continued)



Solution:

$$f_S \ge 2B = 30,000 \text{ samples / sec}$$

$$10 \log_{10} M^2 \ge 58 dB \Rightarrow M \ge 794 \Rightarrow M \ge 1024$$

 $\Rightarrow n \ge 10 \text{ bits/sampl e}$

- We will assume n must be an integer, thus n = 10
- Minimum data rate: $R = f_s n \ge 300kbps$
- First null BW: $B = f_s n = 300kHz$
- Assumes rectangular pulses are used as the waveforms

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Non-Uniform Quantization



A quantizer for which

$$\widetilde{x}_{k-1} - \widetilde{x}_k = \Delta, \forall k \in \{1, \dots, L-1\}$$

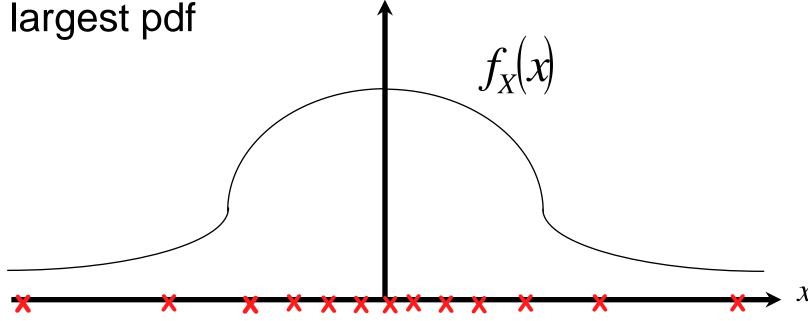
is called a uniform quantizer.

- It is sometimes better to use non-uniform spacing
- Examine the distortion measure: $D = \int_{-\infty}^{\infty} (x \tilde{x})^2 f(x) dx$
- We wish to make $(x-\tilde{x})^2$ small when f(x) is large.
- We can accept larger $(x \tilde{x})^2$ when f(x) is small.
- **Basic principle:** concentrate quantization levels in area of largest pdf.

Non-uniform Quantization



Concentrate quantization levels in areas of



Quantization levels = x

Difference between Uniform and Non-uniform Quantization



• Let:
$$p(x) = e^{-\frac{x^2}{2}} / \sqrt{2\pi}$$

- Let: $\{\tilde{x}_1 = -1.494, \tilde{x}_2 = -0.498, \tilde{x}_3 = 0.498, \tilde{x}_4 = 1.494\}$
- Numerical integration shows that:

$$D = 0.1188, E\left[X^2\right] = 1, \left(\frac{S}{N}\right)_{avg} = 10\log_{10}\left(\frac{1}{0.1188}\right) = 9.25dB$$

- Non-uniform quantization will yield better results
 - The "best" possible quantizer has

$$\left(\frac{S}{N}\right)_{avg} = 12dB$$





• If time....





- Today we have examined Pulse Code Modulation which converts an analog signal to a digital signal.
- Comprised of
 - Sampling
 - Quantization
 - Pulse modulation
- Next class we will examine additional techniques for converting analog signals to baseband digital signals