ECE 6640
Digital Communications

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Chapter 2

2. Formatting and Baseband Modulation.
   2. Formatting Textual Data (Character Coding).
   4. Formatting Analog Information.
   5. Sources of Corruption.
   7. Uniform and Nonuniform Quantization.
   8. Baseband Modulation.
   9. Correlative Coding.
Sklar’s Communications System

Signal Processing Functions

Notes and figures are based on or taken from materials in the course textbook:
Bernard Sklar, Digital Communications, Fundamentals and Applications,
Formatting

• Insure that the message is compatible with digital processing
• Transmit formatting is where the source information is translated into digital symbols
• When data compression is also employed, the process is called source coding. (see Chap. 13)
Baseband Signaling

• Generation of the baseband waveform from the digital symbols provided by formatting or source coding.
• This could take the form of pulse modulation or pulse code modulation (PCM).
• The baseband signal may be sent using a wired connection or network to a receiver.
  – applicable for wired applications and wireless
Formatting and Transmission

Digital info.

Textual info.

Analog info.

Sample → Quantize → Encode → Pulse modulate → Transmit

Bit stream → Pulse waveforms → Channel → Receive

Digital info.

Textual info.

Analog info.

Source

Sink

Format

Low-pass filter → Decode → Demodulate/Detect → Receive

Format
Textual Data

• 5-bit coding – Baudot: 32 characters, alphabet plus 6
• 7-bit coding – ASCII: American Standard Code for Information Interchange
  – Originally designed for telegraph; therefore, extra fields
• 8-bit coding – EBCDIC: Extended Binary Coded Decimal Interchange Code
  – IBM system
• 16-bit coding – Unicode

• Code may be sent serially with start, parity and stop bits
• Code may be structured as words/symbols
Data Format for Asynchronous Data Communication

- Data is transmitted character by character bit-serially.
- A character consists of
  - one start bit (0-level)
  - 7 to 8 data bits (often, an ASCII character plus a parity bit)
  - an optional parity bit
  - one, or one and a half, or two stop bits (1-level)
  - least significant bit is transmitted first
  - most significant bit is transmitted last

<table>
<thead>
<tr>
<th>Start bit</th>
<th>0</th>
<th>1</th>
<th>2</th>
<th>3</th>
<th>4</th>
<th>5</th>
<th>6</th>
<th>7</th>
<th>Stop bit 1</th>
<th>Stop bit 2</th>
</tr>
</thead>
</table>

Figure 9.4 The format of a character

The transmission is a burst occurring at an unknown time but with known bit periods.
The EIA-232E Electrical Specifications

Electrical signal on a pair of wires … signal and ground.
Negative logic often used in physical layer transmissions.
Messages, Characters and Symbols

• Message is encoded into a sequence of bits
  – The bit stream may be a basedband signal
  – ASCII can generate a continuous bit stream if
    “idle characters are 1’s”

• Grouping of k-bits can be formed into symbols
  – M-ary systems use symbols sets where \( M=2^k \)
  – For \( k=1 \), the bit rate and symbol rate are the same
  – Defined waveforms represent each of the symbols

• Therefore a message based bit streams can be represented as
  a string of Octal or Hex characters in sequences!
  – See Text Figure 2.5
A Review of Sampling Theorem

• We use digital signal processing to transmit and receive all forms of communications.

• Digital communications inherently describes bit values and symbol values that “conceptually” exist for a defined period of time and then “instantaneously” switch to another value.
  – The transmitted signals can not physically do this!
  – Transmitted signals must exist at defined frequencies and within defined bandwidths … limited bandwidths often start at baseband.

• We may not discuss or simulate all the “real world” effects, even in this class.
Analog to Digital Conversion

• Sampling
  – Sampling Theorem
    • Nyquist rate $\Rightarrow f_s \geq 2 \cdot f_{\text{max}}$
  – Sample Rate
  – Sample Period

• Impulse Sampling Function

$$f_s = \frac{1}{T} = \frac{w_s}{2\pi}$$

$$T = \frac{1}{f_s}$$

$$x_\delta(t) = \sum_{k=-\infty}^{\infty} \delta(t - n \cdot T_s)$$

$$x_s(t) = x(t) \cdot x_\delta(t) = \sum_{n=-\infty}^{\infty} x(t) \cdot \delta(t - n \cdot T_s)$$

$$x_s(t) = \sum_{n=-\infty}^{\infty} x(n \cdot T_s) \cdot \delta(t - n \cdot T_s)$$
Fourier Domain – Replicated Spectra

\[ X_\delta(f) = \frac{1}{T_s} \cdot \sum_{n=-\infty}^{\infty} \delta(f - n \cdot f_s) \]

\[ X_s(f) = X(f) \ast X_\delta(f) \]

\[ X_s(f) = \int_{-\infty}^{\infty} X_\delta(\lambda) \cdot X(f - \lambda) \cdot d\lambda \]

\[ X_s(f) = \int_{-\infty}^{\infty} \frac{1}{T_s} \cdot \sum_{k=-\infty}^{\infty} \delta(\lambda - n \cdot f_s) \cdot X(f - \lambda) \cdot d\lambda \]

\[ X_s(f) = \frac{1}{T_s} \cdot \sum_{k=-\infty}^{\infty} \int_{-\infty}^{\infty} \delta(\lambda - n \cdot f_s) \cdot X(f - \lambda) \cdot d\lambda \]

\[ X_s(f) = \frac{1}{T_s} \cdot \sum_{k=-\infty}^{\infty} X(f - n \cdot f_s) \]
Fourier Domain

- Spectral replication at steps of fs
- Appears as the convolution of the original spectrum by a comb waveform spaced as fs
  - If the Nyquist rate is not maintained, the convolved elements will overlap and become distorted

- See Figure 2.6 on p. 64 (next slide)

- Note: Signals are not typically band limited; therefore, there will be some aliasing whenever sampling is performed
Sampling Spectral Replication: Perfect Impulse Sampling

Figure 2.6  Sampling theorem using the frequency convolution property of the Fourier transform.
Sampling Pulses and Filters

- While Nyquist Theory and Impulse Sampling is mathematically wonderful ….
  - Sampling rates above Nyquist are more practical (Fig. 2.7)
    - 2.2 fmax for audio example (20 kHz vs. 44.1 ksps CD rate)
  - Impulses must be approximated by signals with real duration and magnitude (Section 2.4.1.2 Natural Sampling and Fig. 2.8)
    - Sample by infinite sequence of rects
    - Math equivalent of convolving sampling impulses with rect in time
      In frequency, convolve infinite replicas with sinc … amp mod impulses
    - When this sampling signal is used (mult. in time, conv. In freq.) you get Fig. 2.8 (next slide)

\[
X_s(f) = \frac{1}{T_s} \cdot \sum_{k=-\infty}^{\infty} c_k \cdot X(f - n \cdot f_s)
\]
Sampling Spectral Replication: Non-ideal Rect Sampling

\[ x(t) = \sum_{n=-\infty}^{\infty} c_n e^{j2\pi n f_0 t} \]

\[ X(f) = \begin{cases} 1/T & \text{for } |f| < f_m \\ 0 & \text{otherwise} \end{cases} \]

\[ Y(f) = X(f) \cdot X_s(f) \]

\[ |X_s(f)| \]

\[ |Y(f)| \]

Figure 2.8 Sampling theorem using the frequency shifting property of the Fourier transform.
Sample and Hold, Zero Order Hold

- Typical ADCs use a “sample and hold” prior to the ADC
- Sampling is typically an integration of the signal for a fixed sampling period
- Hold is to insure the ADC has a stable signal for a defined period of time (conversions time)

\[
x_{sp}(t) = p(t) \ast [x(t) \cdot x_\delta(t)] = p(t) \ast \left[ \sum_{n=-\infty}^{\infty} x(t) \cdot \delta(t - n \cdot T_s) \right]
\]

\[
p(t) = \text{rect} \left( \frac{t}{T_s} \right)
\]
ZOF Spectral Domain

\[ X_{sp}(f) = P(f) \cdot \left[ \frac{1}{T_s} \cdot \sum_{k=-\infty}^{\infty} X(f - n \cdot f_s) \right] \]

\[ X_{sp}(f) = T_s \cdot \text{sinc}(f \cdot T_s) \cdot \left[ \frac{1}{T_s} \cdot \sum_{k=-\infty}^{\infty} X(f - n \cdot f_s) \right] \]

\[ X_{sp}(f) = \text{sinc}(f \cdot T_s) \cdot \sum_{k=-\infty}^{\infty} X(f - n \cdot f_s) \]

• The spectrum is shaped by the sinc function.
  – Note that if spectral analysis is being performed, an inverse sinc weighting should be applied to “correct” the output.
Filter and Aliasing

• If a signal is “under-sampled” the output will have spectral content that is not desired.
  – “Engineering Nyquist” 2.2 x fmax
  – With digital post filtering, sample at 4 x fmax (or 4.4) and then use a half-band filter decimator (may be less expensive)

• If additional digital filtering will be employed, aliased regions of the spectrum may be digitally removed.
  – This allows the transition bands to overlap and the stopband to be placed nearer to the passband edge.
Filter and Aliasing

Using a slightly higher sampling rate
Filter Terminology

- **Passband**
  - Frequencies where signal is meant to pass

- **Stopband**
  - Frequencies where some defined level of attenuation is desired

- **Transition-band**
  - The transitions frequencies between the passband and the stopband

- **Filter Shape Factor**
  - The ratio of the stopband bandwidth to the passband bandwidth

\[
SF = \frac{BW_{SB}}{BW_{PB}}
\]

See FilterNotes and FIR_Filter_DSPNotes or
MRSP Chap 4 Nyquist/Raised Cosine Filter
Reducing the Sample Rate

If additional digital filtering will be employed, aliased regions of the spectrum may be digitally removed.

This allows the transition bands to overlap and the stopband to be placed nearer to the passband edge.
Oversampling

- Without Oversampling
  - High performance LPF
  - Nyquist Sampling still includes some aliased components

- With Oversampling
  - Lower performance LPF
  - Aliased components can be significantly reduced
  - High performance digital filter likely to be employed
  - Identical or similar data rate can be achieved
MATLAB Visualization

- Different sample rates
- Average power versus total power scaling
- Frequency scaling
- Filtered pulses … filter bandwidth

- All Chap2 scripts
- StepResponses
- PulseTest1-3
2.5 Sources of Corruption

• Quantization Noise
  – Saturation
  – Timing Jitter

  – See Analog Devices – Radio 101

• Intersymbol Interference (ISI)
  – Symbol filter responses extend in time and “overlap”.
  – A symbol can be “interfered” with by other symbols in time!
Quantization Noise

- **Round-off Error**
  - +/- one half of the LSB
  - Uniform error distribution about the quantized value
  - Error mean $\Rightarrow 0$
  - Error Variance $\Rightarrow q^2/12$

- **Truncation Error**
  - 0 to +1 LSB
  - Uniform error distribution from one quantized value to the next
  - Error mean $\Rightarrow 1/2$
  - Error Variance $\Rightarrow q^2/12$

\[
\sigma^2 = \int_{-q/2}^{q/2} e^2 \cdot p(e) \cdot de = \frac{1}{q} \cdot \int_{-q/2}^{q/2} e^2 \cdot de = \frac{1}{q} \cdot \frac{e^3}{3} \bigg|_{-q/2}^{q/2} = \frac{1}{q} \cdot \frac{q^3}{24} - \frac{-q^3}{24} = \frac{q^2}{12}
\]
Quantization Levels

- The levels defined for a typical L-level \((2^k=L)\) ADC

\[
\text{rms}(V_p) \approx \frac{q \cdot L}{2}
\]
Quantized Peak SNR

- For an L level quantized system, letting power be the square of one half the rms value of a maximum sine wave

\[ \text{rms}(V_p)^2 \approx \left[ \frac{q \cdot L}{2 \cdot \sqrt{2}} \right]^2 \]

- The estimated signal to noise ratio is

\[ \text{SNR}_q = \frac{\text{rms}(V_p)^2}{\sigma^2} \]

\[ \text{SNR}_q \approx \left[ \frac{q \cdot L}{2 \cdot \sqrt{2}} \right]^2 / \frac{q^2}{12} \]

For 8-bits or 256 levels

\[ \text{SNR}_q = 1.5 \cdot 256^2 = 49.9 \text{ dB} \]

Nominally 6 dB per bit
Intersymbol Interference

- Web
  - InterSymbol Interference (ISI)
  - Nyquist ISI Criterion
  - Inter Symbol Interference (ISI) and Raised cosine filtering
    - From C. Langton “Complex to Real” web site

- Other classes (ECE6560 MRSP Chap. 4)
  - A raised cosine window/filter is a form of Nyquist filtering
    - As combined transmitting and receiving filters, each uses a square-root raised cosine filter.

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Intersymbol Interference

- Matlab ISI Example Code
  - The original 16-QAM data in real and imaginary
  - The channel corrupted received data, no filtering
  - The channel corrupted received data, SQRT RC Tx and RX filters
2.6 Pulse Code Modulation

• Use the digital word/symbol generated for each character or ADC value

• Note that the more information or accuracy per symbol, a higher bit rate is required to maintain the symbol rate.
  – Fs demands a fixed, constant communication rate
  – Text may be sent at any rate that is acceptable (non real-time)

• This can also be referred to as Amplitude Shift Keying

• Note that ADC values if sent as pulse would be called PAM or pulse amplitude modulation
  – PAM may be discrete or quantized
2.7 Uniform vs. Non-uniform Quantization

- **Uniform (linear) quantizing:**
  - No assumption about amplitude statistics and correlation properties of the input.
  - Robust to small changes in input statistic by not finely tuned to a specific set of input parameters
  - Simply implemented

- **Non-uniform quantizing:**
  - Using the input statistics to tune quantizer parameters
  - Larger SNR than uniform quantizing with same number of levels
  - Non-uniform intervals in the dynamic range with same quantization noise variance
Quantization Example

Figure 2.18 Uniform and nonuniform quantization of signals.
Uniform vs. Nonuniform (2)

• Application of linear quantizer:
  – Signal processing, graphic and display applications, process control applications

• Application of non-uniform quantizer:
  – Commonly used for speech
    – u-law in US, A-law in Europe
Non-uniform Quantization

- When some portions of the voltage range are not often used, additional emphasis can be given to those that are.

- $\mu$-law algorithm (North America $\mu=255$)

\[
F(x) = \text{sgn}(x) \frac{\ln(1 + \mu |x|)}{\ln(1 + \mu)} \quad -1 \leq x \leq 1
\]

\[
F^{-1}(y) = \text{sgn}(y) \frac{1}{\mu} ((1 + \mu)^{|y|} - 1) \quad -1 \leq y \leq 1
\]

- A-law algorithm (Standard Value $A=87.6$)

\[
F(x) = \text{sgn}(x) \begin{cases} 
\frac{A|x|}{1+\ln(A)}, & |x| < \frac{1}{A} \\
\frac{1}{1+\ln(A)} \left(1 + \frac{A|x|}{1+\ln(A)} - 1\right), & \frac{1}{A} \leq |x| \leq 1
\end{cases}
\]

\[
F^{-1}(y) = \text{sgn}(y) \begin{cases} 
\frac{|y|(1+\ln(A))}{A}, & |y| < \frac{1}{1+\ln(A)} \\
\frac{\exp(|y|(1+\ln(A))-1)}{A}, & \frac{1}{1+\ln(A)} \leq |y| < 1
\end{cases}
\]
Non-uniform Quantization

- Matlab Example – quantizing integers 0 to 255
  - Small value compression, large value expansion
Baseband Signaling

- Generation of the baseband waveform from the digital symbols provided by formatting or source coding.
- This could take the form of pulse modulation or pulse code modulation (PCM).
- The baseband signal may be sent using a wired connection or network to a receiver.

- Bits to base band symbols … considerations.
PCM Transmission

• Pulse code modulation (PCM) is used when a binary data stream is to be sent.

• In PCM the binary sequence is used to define logical signal levels for transmission.
  – A logical level may map to bits (e.g. 0-High, 1-Low)
  – A bit value may define whether a level changes or not
    Mark : change whenever the bit is a one
    Space: change whenever the bit is a zero
  – Period half-cycles can take on various structures based on a bit value or the sequence of bits

• See Figure 2.22 on p. 87
PCM Transmission

Figure 2.22 Various PCM Waveforms
PCM Common Waveform Types

- Marks (1’s) and Spaces (0’s)
- Non-return-to-zero (NRZ) – Level, Mark, Space
- Return-to-zero (RZ) – unipolar, bipolar, AMI (alternate mark inversion)
- Manchester – biphase level, biphase mark, biphase space
PCM Types

**Biphasic Mark Code**

**AMI-Bipolar Encoding**
(Alternate Mark Inversion)
PCM Type Selection

- Spectral characteristics
  (power spectral density and bandwidth efficiency)
- Bit synchronization capability
- Error detection capability
- Interference and noise immunity
- Implementation cost and complexity
Spectral Attributes of PCM

The graph illustrates the spectral density (W/Hz) of various PCM signaling methods as a function of $WT'$ (normalized bandwidth, where $T$ is the signal pulse width). The methods shown are Duobinary, Delay modulation, NRZ, Dicode NRZ, and Bi-phase.
M-ary Pulse-Modulation Waveforms

• M-ary modulation is used when symbol data stream is to be sent

• M-ary waveform include:
  – PAM: Pulse-Amplitude Modulation
  – PPM: Pulse-Position Modulation
  – PDM: Pulse-Duration Modulation or PWM: Pulse-Width Modulation
  – Multiple “level” can be transmitted as one symbol

• Other M-ary waveforms
  – QAM: Quadrature-Amplitude Modulation
Correlative Codes

• Web site: “Complex technology made real • Complex communications technology made easy” or “Complex to Real” by Charan Langton
  – http://complextoreal.com/

• Tutorial 16 – Partial Response signaling and Quadrature Partial Response (QPR) modulation