**Signal Encoding Techniques**

Chapter 6

**Reasons for Choosing Encoding Techniques**

- Digital data, digital signal
  - Equipment less complex and expensive than digital-to-analog modulation equipment
- Analog data, digital signal
  - Permits use of modern digital transmission and switching equipment

**Signal Encoding Criteria**

- What determines how successful a receiver will be in interpreting an incoming signal?
  - Signal-to-noise ratio
  - Data rate
  - Bandwidth
  - An increase in data rate increases bit error rate
  - An increase in SNR decreases bit error rate
  - An increase in bandwidth allows an increase in data rate

**Factors Used to Compare Encoding Schemes**

- Signal spectrum
  - With lack of high-frequency components, less bandwidth required
  - With no dc component, ac coupling via transformer possible
  - Transfer function of a channel is worse near band edges
- Clocking
  - Ease of determining beginning and end of each bit position

**Factors Used to Compare Encoding Schemes**

- Signal interference and noise immunity
  - Performance in the presence of noise
- Cost and complexity
  - The higher the signal rate to achieve a given data rate, the greater the cost
Basic Encoding Techniques
- Digital data to analog signal
  - Amplitude-shift keying (ASK)
    - Amplitude difference of carrier frequency
  - Frequency-shift keying (FSK)
    - Frequency difference near carrier frequency
  - Phase-shift keying (PSK)
    - Phase of carrier signal shifted

Amplitude-Shift Keying
- One binary digit represented by presence of carrier, at constant amplitude
- Other binary digit represented by absence of carrier
  \[ s(t) = \begin{cases} 
  A \cos(2\pi f_c t) & \text{binary 1} \\
  0 & \text{binary 0} 
\end{cases} \]
  - where the carrier signal is \( A \cos(2\pi f_c t) \)

Binary Frequency-Shift Keying (BFSK)
- Two binary digits represented by two different frequencies near the carrier frequency
  \[ s(t) = \begin{cases} 
  A \cos(2\pi f_1 t) & \text{binary 1} \\
  A \cos(2\pi f_2 t) & \text{binary 0} 
\end{cases} \]
  - where \( f_1 \) and \( f_2 \) are offset from carrier frequency \( f_c \) by equal but opposite amounts

Figure 6.2: Modulation of Analog Signals for Digital Data

Amplitude-Shift Keying
- Susceptible to sudden gain changes
- Inefficient modulation technique
- On voice-grade lines, used up to 1200 bps
- Used to transmit digital data over optical fiber

Binary Frequency-Shift Keying (BFSK)
- Less susceptible to error than ASK
- On voice-grade lines, used up to 1200 bps
- Used for high-frequency (3 to 30 MHz) radio transmission
- Can be used at higher frequencies on LANs that use coaxial cable
Multiple Frequency-Shift Keying (MFSK)

- More than two frequencies are used
- More bandwidth efficient but more susceptible to error

\[ s_i(t) = A \cos 2\pi f_i t \quad 1 \leq i \leq M \]

- \( f_i = f_c + (2i - 1 - M) f_d \)
- \( f_c \) = the carrier frequency
- \( f_d \) = the difference frequency
- \( M \) = number of different signal elements = \( 2^L \)
- \( L \) = number of bits per signal element

To match data rate of input bit stream, each output signal element is held for:

\[ T_s = L T \text{ seconds} \]

where \( T \) is the bit period (data rate = \( 1/T \))

So, one signal element encodes \( L \) bits

Total bandwidth required \( 2Mf_d \)

Minimum frequency separation required \( 2f_d 1/T_s \)

Therefore, modulator requires a bandwidth of

\[ W_d = 2^L / LT = M / T_s \]

Phase-Shift Keying (PSK)

- Two-level PSK (BPSK)
  - Uses two phases to represent binary digits

\[ s(t) = \begin{cases} 
A \cos(2\pi f_c t) & \text{binary 1} \\
A \cos(2\pi f_c t + \pi) & \text{binary 0} 
\end{cases} \]

Phase-Shift Keying (PSK)

- Differential PSK (DPSK)
  - Phase shift with reference to previous bit
    - Binary 0 – signal burst of same phase as previous signal burst
    - Binary 1 – signal burst of opposite phase to previous signal burst
Phase-Shift Keying (PSK)

- Four-level PSK (QPSK)
  - Each element represents more than one bit

\[
S(t) = \begin{cases} 
A \cos \left(2\pi f t + \frac{\pi}{4}\right) & \text{11} \\
A \cos \left(2\pi f t + \frac{3\pi}{4}\right) & \text{01} \\
A \cos \left(2\pi f t - \frac{3\pi}{4}\right) & \text{00} \\
A \cos \left(2\pi f t - \frac{\pi}{4}\right) & \text{10}
\end{cases}
\]

QPSK and OQPSK

Phase-Shift Keying (PSK)

- Multilevel PSK
  - Using multiple phase angles with each angle having more than one amplitude, multiple signals elements can be achieved

\[
D = R = \frac{R}{L} = \frac{R}{\log_2 M}
\]

- \( D \) = modulation rate, baud
- \( R \) = data rate, bps
- \( M \) = number of different signal elements = \( 2^L \)
- \( L \) = number of bits per signal element

Performance

- Bandwidth of modulated signal \( (B_T) \)
  - ASK, PSK \( B_T = (1+r)R \)
  - FSK \( B_T = 2DF + (1+r)R \)

\( R \) = bit rate
\( 0 < r < 1 \); related to how signal is filtered
\( DF = f_2 - f_1 \)

Performance

- Bandwidth of modulated signal \( (B_T) \)
  - MPSK \( B_T = \left(\frac{1+r}{L}\right)R = \left(\frac{1+r}{\log_2 M}\right)R \)
  - MFSK \( B_T = \left(\frac{(1+r)M}{\log_2 M}\right)R \)

- \( L \) = number of bits encoded per signal element
- \( M \) = number of different signal elements
**Performance**

Quadrature Amplitude Modulation

- QAM is a combination of ASK and PSK
- Two different signals sent simultaneously on the same carrier frequency

\[ s(t) = d_1(t)\cos 2\pi f_c t + d_2(t)\sin 2\pi f_c t \]

**Reasons for Analog Modulation**

- Modulation of digital signals
  - When only analog transmission facilities are available, digital to analog conversion required
- Modulation of analog signals
  - A higher frequency may be needed for effective transmission
  - Modulation permits frequency division multiplexing

**Basic Encoding Techniques**

- Analog data to analog signal
  - Amplitude modulation (AM)
  - Angle modulation
    - Frequency modulation (FM)
    - Phase modulation (PM)

**Amplitude Modulation**

\[ s(t) = \left[ 1 + n_x x(t) \right] \cos 2\pi f_c t \]

- \( \cos 2\pi f_c t \) = carrier
- \( x(t) \) = input signal
- \( n_x \) = modulation index
- Ratio of amplitude of input signal to carrier
- a.k.a double sideband transmitted carrier (DSBTC)
Amplitude Modulation

- Transmitted power
  \[ P_t = P_c \left(1 + \frac{n_a^2}{2}\right) \]
  - \( P_c \) = total transmitted power in \( s(t) \)
  - \( P_t \) = transmitted power in carrier

Single Sideband (SSB)

- Variant of AM is single sideband (SSB)
- Sends only one sideband
- Eliminates other sideband and carrier

Advantages
- Only half the bandwidth is required
- Less power is required

Disadvantages
- Suppressed carrier can’t be used for synchronization purposes

Angle Modulation

- Angle modulation
  \[ s(t) = A_c \cos(2\pi f_c t + \phi(t)) \]

  - Phase modulation
    - Phase is proportional to modulating signal
      \[ \phi(t) = n_p m(t) \]
    - \( n_p \) = phase modulation index

Angle Modulation

- Frequency modulation
  - Derivative of the phase is proportional to modulating signal
    \[ \phi'(t) = n_f m(t) \]
  - \( n_f \) = frequency modulation index

Angle Modulation

- Compared to AM, FM and PM result in a signal whose bandwidth:
  - is also centered at \( f_c \)
  - but has a magnitude that is much different
    - Angle modulation includes \( \cos(2\phi(t)) \) which produces a wide range of frequencies
  - Thus, FM and PM require greater bandwidth than AM
Angle Modulation

- Carson’s rule

\[ B_f = 2(\beta + 1)B \]

\[ \beta = \begin{cases} \frac{n_i A_m}{\Delta F} & \text{for PM} \\ \frac{n_i A_m}{2\pi B} & \text{for FM} \end{cases} \]

- The formula for FM becomes

\[ B_f = 2\Delta F + 2B \]

Basic Encoding Techniques

- Analog data to digital signal
  - Pulse code modulation (PCM)
  - Delta modulation (DM)

Analog Data to Digital Signal

- Once analog data have been converted to digital signals, the digital data:
  - can be transmitted using NRZ-L
  - can be encoded as a digital signal using a code other than NRZ-L
  - can be converted to an analog signal, using previously discussed techniques

Pulse Code Modulation

- Based on the sampling theorem
- Each analog sample is assigned a binary code
  - Analog samples are referred to as pulse amplitude modulation (PAM) samples
- The digital signal consists of block of \( n \) bits, where each \( n \)-bit number is the amplitude of a PCM pulse

Pulse Code Modulation

- By quantizing the PAM pulse, original signal is only approximated
- Leads to quantizing noise
- Signal-to-noise ratio for quantizing noise

\[ \text{SNR}_{dB} = 20\log 2^n + 1.76 \text{ dB} = 6.02n + 1.76 \text{ dB} \]

- Thus, each additional bit increases SNR by 6 dB, or a factor of 4
**Delta Modulation**

- Analog input is approximated by staircase function
  - Moves up or down by one quantization level ($\delta$) at each sampling interval
  - The bit stream approximates derivative of analog signal (rather than amplitude)
    - 1 is generated if function goes up
    - 0 otherwise

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**Delta Modulation**

- Two important parameters
  - Size of step assigned to each binary digit ($\delta$)
  - Sampling rate
- Accuracy improved by increasing sampling rate
  - However, this increases the data rate
- Advantage of DM over PCM is the simplicity of its implementation

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**Reasons for Growth of Digital Techniques**

- Growth in popularity of digital techniques for sending analog data
  - Repeaters are used instead of amplifiers
    - No additive noise
  - TDM is used instead of FDM
    - No intermodulation noise
  - Conversion to digital signaling allows use of more efficient digital switching techniques