Data Encoding

- Digital signaling
  - Data source $g(t)$ encoded into digital signal $x(t)$
    - $g(t)$ itself may be analog or digital
    - Actual form of $x(t)$ dependent on encoding technique, chosen to optimize use of transmission medium – conserve bandwidth or minimize errors
- Analog signaling
  - Based on a continuous constant-frequency signal, called the *carrier signal*
  - Carrier signal frequency chosen to be compatible with transmission medium
  - Data transmitted by carrier signal modulation
  - Modulation
    - Process of encoding source data onto a carrier signal with frequency f_c
    - Operation on one or more of three fundamental frequency-domain parameters – amplitude, frequency, and phase
- Input signal $m(t)$
  - Can be analog or digital
  - Also called *modulating signal* or *baseband signal*
  - Modulated signal $s(t)$ – result of modulating carrier signal
    - Also called bandlimited or bandpass signal
  - Location of bandwidth on the spectrum related to carrier frequency f_c
- Different possible combinations
  - Digital data, digital signal
    - Simple and inexpensive equipment
  - Analog data, digital signal
    - Data needs to be converted to digital form
  - Digital data, analog signal
    - Needed to take advantage of existing transmission media that only allows the transmission of analog signal
  - Analog data, analog signal
    - Transmitted as baseband signal easily and cheaply
    - Modulation to shift the bandwidth of baseband signal to another portion of spectrum
    - Multiple signals on different position on the spectrum can share the same transmission medium (frequency-division multiplexing)

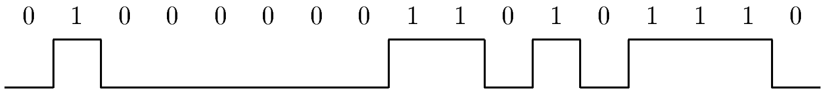
Digital Data, Digital Signals

- Digital signal
  - Sequences of discrete, discontinuous voltage pulses
  - Binary data transmitted by encoding each data bit into signal elements

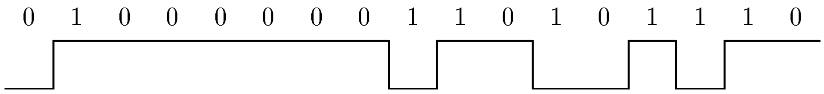
- Unipolar signal
  - All signal elements have the same algebraic sign
  - Binary logic states represented by a positive and negative voltage level
- Data rate
  - Number of bits per second transmitted
  - Duration or length of a bit
    * Amount of time required to emit a bit
    * For a data rate R bits per second, duration of each bit is $\frac{1}{R}$
- Modulation rate
  - Rate at which signal level is changed
  - Depends on the nature of signal encoding
  - Expressed in *bauds*, or signal elements per second
- *Mark* – Binary digit 1
- *Space* – Binary digit 0
- Interpreting digital signal at the receiver
  - Receiver must know the timing of each bit (start and end)
  - Determine if the signal level for bit is high (1) or low (0)
  - Sampling and comparison with a threshold value
  - Important factors: SNR, data rate, and bandwidth
  - Important facts:
    1. Increase in data rate increases bit-error-rate (BER)
    2. Increase in SNR decreases BER
    3. Increase in bandwidth allows for an increase in data rate
- Encoding scheme
  - Mapping from data bits to signal elements
- Evaluation and comparison of various schemes
  - Signal spectrum
    * Lack of high frequency components implies less bandwidth requirement
    * Lack of DC component is desirable
    * DC component requires direct physical attachment between transmission components
    * No DC component means that AC coupling via transformer is possible, providing electrical isolation and reducing interference
    * Magnitude of the effect of signal distortion and interference depend on the spectral properties of transmitted signal
    * Good signal design should concentrate in the middle of transmission bandwidth to avoid band edges
    * Code can be designed with the aim of shaping the spectrum of transmitted signal
  - Clocking
    * Beginning and end of each bit needs to be synchronized
    * Provide synchronization based on transmitted signal
  - Error detection

    - Part of data link control layer
    - Useful to build into physical signal encoding scheme
  - Signal interference and noise immunity
    - Expressed in terms of BER
  - Cost and complexity
    - Cost is proportional to data rate

- Nonreturn to zero (NRZ)
  - Use two different voltage levels for two binary digits
  - Voltage level constant during bit interval
    - No transition, no return to zero voltage level
  - Nonreturn to zero-level (NRZ-L)
    - Negative voltage level represents one bit value while positive voltage level represents the other bit value



    - Generally the code used to generate or interpret digital data by terminals and other devices
    - NRZ-L is $g(t)$ (source signal) while the encoded form is $x(t)$
  - Nonreturn to zero, invert on ones (NRZI)
    - Variation of NRZ
    - Maintains a constant voltage pulse for bit duration (same as NRZ-L)
    - Data encoded as presence or absence of signal transition at bit start time
    - Transition indicates 1 while no transition indicates 0

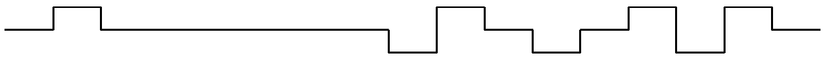


  - *Differential encoding*
    - Exemplified by NRZI
    - Information represented as changes between successive data symbols rather than signal elements
    - Encoding of current bit is determined as
      - Current bit 0? Encoded with the same signal as preceding bit
      - Current bit 1? Encoded with different signal as preceding bit
    - More reliable to detect transition in the presence of noise than to compare the value to a threshold
    - Impervious to reversal of polarity in NRZ-L
  - Limitations
    - Presence of DC component
    - Lack of synchronization capability
    - Any drift between timing of transmitter and receiver during a long sequence of 0's or 1's can result in loss of synchronization
  - Applications
    - Commonly used for digital magnetic recording
    - Not attractive for signal transmission applications

- Multilevel binary
  - Addresses some of the deficiencies of NRZ codes

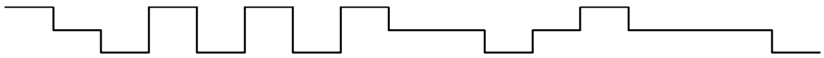
- Use more than two signal levels
- Bipolar-AMI (Automatic Mark Inversion)
  - Binary 0 is represented by no line signal
  - Binary 1 is represented by positive or negative pulse
  - Binary 1 pulses must alternate in polarity

0 1 0 0 0 0 0 0 1 1 0 1 0 1 1 1 0



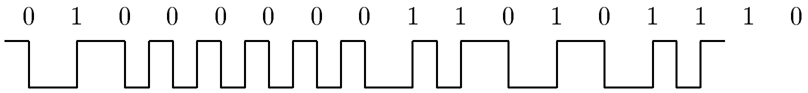
  - Advantages
    - No loss of synchronization with a long string of 1's
    - Each 1 introduces a transition and receiver can resynchronize on transition
    - Long string of 0's still a problem
    - Because 1 signals alternate in voltage from positive to negative, there is no net DC component
    - Bandwidth of resulting signal considerably less than bandwidth for NRZ
    - Pulse alternation provides a simple means of error detection; any isolated error (addition or deletion of pulse) causes a violation
- Pseudoternary
  - Binary 1 represented by absence of line signal
  - Binary 0 represented by alternating positive and negative pulses
  - Almost opposite representation compared to bipolar-AMI

0 1 0 0 0 0 0 0 1 1 0 1 0 1 1 1 0



- Long string of 0's in bipolar-AMI and a long string of 1's in pseudoternary still presents a problem
  - May insert additional bits to force transitions
    - Used in ISDN for relatively low data rate transmission
    - Expensive in high data rate because it results in an increase in already high signal transmission rate
    - Possible to apply at high data rate using data scrambling
  - Line signal may take one of three levels
    - Each signal element can represent $\log_2 3 = 1.58$ bits of information but actually represents only one bit
    - Loss of efficiency compared to NRZ
    - Receiver has to discriminate between three levels $(+A, 0, -A)$
    - Signal requires approximately 3dB more power than the corresponding 2-valued signal for the same BER
    - BER for NRZ codes is significantly less than multilevel binary for the same SNR

• Biphase
- Overcome the limitations of NRZ codes
- Require at least one transition per bit time and may even have two transitions
- Manchester code
  - Transition at the middle of each bit period
  - Serves as clocking mechanism as well as data
  - Transition from low to high indicates 1 while transition from high to low indicates 0



- Differential Manchester
  * Midbit transition is used only to provide clocking
  * 0 is represented by presence of transition at the beginning of bit period
  * 1 is represented by the absence of transition at the beginning of bit period
  * Advantage of employing differential encoding (no DC)

0

- Maximum modulation rate is twice that for NRZ due to one transition per bit time
  * Bandwidth required is correspondingly greater
- Advantages
  * Synchronization
    · Predictable transition during each bit time
    · Can be used for synchronization
    · *Self-clocking codes*
  * No DC component
  * Error detection
    · Can be detected by absence of expected transition
    · Impervious to noise as noise will have to invert the signal both before and after the expected transition
- Bandwidth reasonably narrow with no DC component
  * Bandwidth wider than that for multilevel binary codes
- Popular for data transmission
  * Manchester used for baseband coaxial cable and twisted pair bus LANs
  * Differential Manchester used for token ring LAN using shielded twisted pair

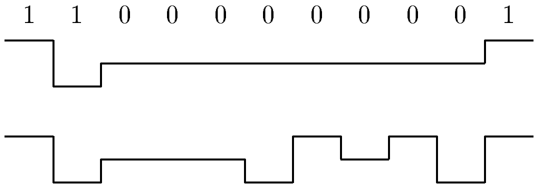
• Modulation rate

- Data rate
  * Bits per second, or bit rate
  * $\frac{1}{t_B}$ where t_B is bit duration
- Modulation rate
  * Baud
  * Rate at which signal elements are generated
- Manchester code
  * Minimum size signal element is a pulse of one-half the duration of bit interval
  * All 0's or all 1's generate a continuous stream of such pulses
  * Maximum modulation rate is $\frac{2}{t_B}$
- Modulation rate $D = \frac{R}{b}$ where R is the data rate in bps and b is the number of bits per signal element

• Scrambling techniques

- Biphase techniques require high signal rate relative to data rate limiting their use in long distance applications

- Scrambling can remove the above limitation
- Constant voltage level sequences are replaced by fillers to provide synchronization for receiver clock
- Filler must be recognized by receiver and replaced by original data sequence
- Filler sequence is the same length as original sequence, implying no increase in data rate
- Design goals
  * No DC component
  * No long sequences of zero-level line signals
  * No reduction in data rate
  * Error-detection capability
- Bipolar with 8-zeros substitution (B8ZS)
  * Commonly used in North America
  * Based on bipolar-AMI
  * Encoding amended with following rules
    · If there is an octet of all zeros, and last voltage preceding this octet was positive, then the octet is encoded as $000 + -0 - +$.
    · If there is an octet of all zeros, and last voltage preceding this octet was negative, then the octet is encoded as $000 - +0 + -$.
  * Forces two code violations of the AMI code, using patterns not allowed in AMI
    · Receiver recognizes the pattern and interprets the octet as all zeros



- High-density bipolar-3 zeros (HDB3)
  * Commonly used in Europe and Japan
  * Based on bipolar-AMI
  * Replaces strings of four zeros with one or two pulses
    · In each case, fourth zero is replaced with a code violation
  * A rule is added to ensure that successive violations are of alternate polarity to avoid DC component

	Number of bipolar pulses (ones) since last substitution	
Polarity of preceding pulse	Odd	Even
−	000−	+00+
+	000+	−00−

  * If last violation was positive, this violation must be negative, and vice versa

Digital data, analog signals

- Transmission of digital data through public phone network
  - Phone network works with voice frequency range of 300 to 3400 Hz
  - Not currently suitable for handling digital signals from subscriber location
  - Digital devices attached to network via modems, to produce signals in voice frequency range
- Encoding techniques

- Based on modifying one of the three characteristics of carrier signal: amplitude, frequency, and phase, as described below
  * In each case, resulting signal occupies a bandwidth centered on carrier frequency
- Amplitude-shift keying (ASK)
  * Two binary values are represented by two different amplitudes of carrier signal
  * One of the amplitudes is zero
    · One binary digit is represented by the presence of the carrier, the other by the absence of the carrier
  * Resulting signal is

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

    where the carrier signal is given by $A\cos(2\pi f_c t)$
  * Susceptible to sudden gain changes
  * Typically used only up to 1200 bps on voice grade lines
  * Used to transmit digital data over optical fiber
    · For LED transmitters, presence of light pulse indicates a 1 while the absence of light pulse indicates a zero
    · For laser transmitters, a low light level indicates one signal element while a high amplitude lightwave represents the other signal element
- Frequency-shift keying (FSK)
  * Two binary values are represented by two different frequencies near the carrier frequency
  * Resulting signal is

    $$s(t) = \begin{cases} A\cos(2\pi f_1 t) & \text{binary 1} \\ A\cos(2\pi f_2 t) & \text{binary 0} \end{cases}$$

    where f_1 and f_2 are offset from carrier frequency f_c by equal but opposite amounts
  * Full duplex transmission over voice grade line
    · Voice grade line passes frequencies between 300 to 3400 Hz
    · In one direction, f_c is 1170 Hz with f_1 and f_2 given by ± 100 Hz
    · In other direction, f_c is 2125 Hz $\pm$ 100 Hz
    · There is little overlap and hence, little interference
  * Less susceptible to error compared to ASK
  * Used up to 1200 bps on voice grade lines
  * Also used for high frequency (3 to 30 MHz) radio transmission
  * Can be used at even higher frequencies in LANs on coaxial cables
- Phase-shift keying (PSK)
  * Data is represented by carrier signal phase-shift to indicate different values
  * Two-phase system with differential PSK
    · Phase shift is with respect to the previous bit transmitted rather than some constant reference signal
    · Binary zero is represented by sending a signal burst of the same phase as the previous signal burst
    · Binary one is represented by sending a signal burst of the opposite phase as the previous signal burst
    · Resulting signal is

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

      with the phase measured relative to previous bit interval
  * Bandwidth can be used more effectively by representing more than one bit in each signal element

    - Instead of a phase shift of 180° as in differential PSK, quadratic PSK, or QPSK uses phase shifts of multiples of 90°
    - Resulting signal is

$$s(t) = \begin{cases} A\cos(2\pi f_c t + \frac{\pi}{4}) & 11 \\ A\cos(2\pi f_c t + \frac{3\pi}{4}) & 10 \\ A\cos(2\pi f_c t + \frac{5\pi}{4}) & 00 \\ A\cos(2\pi f_c t + \frac{7\pi}{4}) & 01 \end{cases}$$

    - Each signal element represents 2 bits instead of 1
    - Can be extended to transmit three bits at a time using eight different phase angles
    - Each phase angle can have more than one amplitude
    - A standard 9600 bps modem uses 12 phase angles, four of which have two amplitude values
  * Difference between R (data rate in bps) and D (modulation rate in bauds)
    - Assume the scheme being used with NRZ-L digital input
    - Data rate $R = \frac{1}{t_B}$ where t_B is width of each NRZ-L bit
    - Encoded signal contains 4 bits in each signal element using $L = 16$ different combinations of amplitude and phase
    - Modulation rate $D = \frac{R}{4}$, because each change of signal element communicates four bits
    - Line signaling speed is 2400 baud but data rate is 9600 bps, allowing higher bt rates on voice grade lines using complex modulation schemes
  * In general

$$D = \frac{R}{b} = \frac{R}{\log_2 L}$$

where

D	modulation rate in baud
R	data rate in bps
L	number of different signal elements
b	number of bits per signal element

- Performance
  - Bandwidth of modulated signal depends on factors such as
    * Definition of bandwidth used
    * Filtering technique used to create bandpass signal
  - Transmission bandwidth B_T for ASK and PSK is of the form

$$B_T = (1 + r)R$$

where R is the bit rate and r is related to the technique by which the signal is filtered to establish bandwidth for transmission; typically $0 < r < 1$

    * $B_T \propto R$
  - For FSK, we have

$$B_T = 2\Delta F + (1 + r)R$$

where $\Delta F = f_2 - f_c = f_c - f_1$

    * For high frequencies, ΔF term dominates
    * FSK signaling on coaxial cable multipoint local network uses $\Delta F = 1.25$ MHz, $f_c = 5$ MHz, and $R = 1$ Mbps, giving $2\Delta F = 2.5$ MHz as the dominant term
    * For low frequency (voice grade line modem), we have $\Delta F = 100$ Hz, $f_c = 1170$ Hz in one direction, and $R = 300$ bps, so that the $(1 + r)R$ term dominates

- With multilevel signaling, bandwidth can improve significantly

$$B_T = \left(\frac{1+r}{b}\right) R = \left(\frac{1+r}{\log_2 L}\right) R$$

- Relationship with bandwidth efficiency
  * We have

    $$\frac{E_b}{N_0} = \frac{S}{N_0 R}$$

    where N_0 is the noise power density in watts/Hz
  * Noise in a signal with bandwidth B_T is given by

    $$N = N_0 B_T$$

  * Substituting for N_0, we have

    $$\frac{E_b}{N_0} = \frac{S}{N} \cdot \frac{B_T}{R}$$

  * Thus, bit error rate can be reduced by increasing bandwidth efficiency $\frac{E_b}{N_0}$, or by
    · Increasing the bandwidth, or
    · Decreasing the data rate
- Example: Bandwidth efficiency for FSK, ASK, PSK, and QPSK for a bit error rate of 10^{-7} on a channel with an SNR of 12 dB
  * Efficiency is given by

    $$\frac{E_b}{N_0} = 12 \text{ dB } - \left(\frac{R}{B_T}\right)_{\text{dB}}$$

  * For FSK and ASK, we get from Figure 5.4 (p. 138)

    $$\frac{E_b}{N_0} = 14.2 \text{ dB}$$

    which yields

    $$\left(\frac{R}{B_T}\right)_{\text{dB}} = -2.2 \text{ dB}$$

  * For PSK, we get from Figure 5.4 (p. 138)

    $$\frac{E_b}{N_0} = 11.2 \text{ dB}$$

    which yields

    $$\left(\frac{R}{B_T}\right)_{\text{dB}} = 0.8 \text{ dB}$$

  * For QPSK, baud rate $D = \frac{R}{2}$ so that

    $$\frac{R}{B_T} = 1.6 \text{ dB}$$

Analog data, digital signals

- Digitization: Converting analog data into digital signals
- After digitization, one of the following can happen

  1. Figure 5.9

2. Digital data can be transmitted using NRZ-L, going directly from analog to digital signal
3. Digital data can be encoded as a digital signal using a code other than NRZ-L
4. Digital data can be converted into analog signal, using one of the modulation techniques
   - Allows analog data to be treated as digital data, even though it is transmitted as analog data

- Codec
  - Device used to convert analog data into digital form for transmission and subsequently, recover the analog data from digital signal
  - Uses pulse code modulation or delta modulation
- Pulse code modulation (PCM)
  - Based on sampling theorem

    If a signal $f(t)$ is sampled at regular intervals of time and at a rate higher than twice the highest signal frequency, then the samples contain all the information of the original signal. The function $f(t)$ may be reconstructed from these samples by the use of a low-pass filter.
  - Limiting voice signals to 4000 Hz allows us to characterize them completely by 8000 samples per second
    * Analog samples, or pulse amplitude modulated (PAM) samples
    * Each analog sample is assigned a binary code to convert to digital
    * 8-bit per sample, gives $8000 \times 8 = 64$kbps for each voice channel
    * Figure 5.10
  - PCM starts with a continuous-time, continuous-amplitude (analog) signal
    * Analog signal is converted to digital
      · Blocks of n bits, with each n-bit number representing the amplitude of a PCM pulse
    * On reception, digital signal is converted back to analog
      · Violates the sampling theorem
      · Quantization of PAM pulse only allows the original signal to be approximated and not recovered exactly, leading to *quantizing error* or *quantizing noise*
      · Signal-to-noise ratio for the quantizing noise can be expressed as

$$\begin{aligned} \text{SNR} &= 20 \log 2^n + 1.76 \text{ dB} \\ &= 6.02n + 1.76 \text{ dB} \end{aligned}$$

      · For each additional bit used for quantizing, SNR increases by about 6 dB, or a factor of 4
  - Nonlinear encoding
    * Refinement of PCM
    * Quantization levels are not equally spaced
    * Greater number of quantization steps for lower amplitudes and smaller number of steps for higher amplitudes allows for reduction in overall signal distortion
    * Figure 5.11
  - Companding
    * Compressing-expanding
    * Achieves similar effect to nonlinear encoding
    * Compresses intensity range of a signal by imparting more gain to weak signals than to strong signals on input
    * Reverse operation is performed on output
- Delta modulation (DM)

- Alternative to PCM
- Analog input is approximated by a staircase function that moves up or down one quantization level (δ) at each sampling interval T_s
- Figure 5.13
- A bit stream is produced by approximating the derivative of an analog signal rather then its amplitude
  * Produce a 1 if staircase function is to go up during next interval
  * Produce a 0 otherwise
- Transition at each sampling interval is chosen so that staircase function tracks the original waveform as closely as possible
  * Based on a feedback mechanism
  * Figure 5.14
  * Analog input is compared to the most recent value of approximating staircase function
    · If value exceeds the staircase function, generate a 1
    · Otherwise generate a 0
  * Staircase is always changed in the direction of the input signal
  * Output of DM process is a binary sequence to be used for reconstructing the staircase function
    · Reconstructed staircase function is smoothed by lowpass filter to reconstruct analog approximation of analog input signal
- Two important parameters in DM scheme
  1. Sampling rate
  2. Size of step assigned to each binary digit δ
     * Must be chosen to produce a balance between two types of errors or noise
     * If waveform changes slowly, there is quantizing noise that increases with an increase in δ
     * If analog waveform changes rapidly than the staircase can keep up with, there is slope overhead noise that increases with a decrease in δ
- Principal advantage of DM is its simplicity of implementation
- PCM has better SNR characteristics at the same data rate

• Performance

- PCM can produced good noise reproduction with 128 quantization levels or 7-bit coding
- With a bandwidth of 4 kHz, samples must be taken at a rate of 8000 samples per second (2 samples per cycle), giving a data rate of $8000 \times 7 = 56$ kbps for PCM-encoded signal
- Bandwidth requirement
  * Digital transmission requires 56 kbps for 4 kHz analog signal
  * Using Nyquist criterion, this signal could require on the order of 28 kHz of bandwidth
  * Why do digital signals still grow in popularity?
    · No additive noise because of repeaters instead of amplifiers
    · Digital signals use time division multiplexing instead of frequency division multiplexing used for analog signals; no intermodulation noise
    · More efficient digital switching techniques

Analog data, analog signals

• Modulation

- Process of combining an input signal $m(t)$ and a carrier at frequency f_c to produce signal $s(t)$ with bandwidth centered on f_c

- Reasons for modulation
  * Higher frequency may be needed for effective transmission
  * Modulation permits frequency division multiplexing

- Amplitude modulation
  - Simplest form of modulation
  - Signal is expressed as
  $$s(t) = [1 + n_a x(t)] \cos 2\pi f_c t$$
  where $\cos 2\pi f_c t$ is carrier and $x(t)$ is the input signal with both normalized to unity amplitude; n_a is modulation index given by the ratio of amplitude of input signal to carrier
  - Input signal is: $m(t) = n_a x(t)$
  - The additive 1 is a DC component to prevent loss of information
  - Scheme is also known as double sideband transmitted carrier
  - Example: expression for $s(t)$ if $x(t)$ is AM signal $\cos 2\pi f_m t$
  $$\begin{aligned} s(t) &= [1 + n_a \cos 2\pi f_m t] \cos 2\pi f_c t \\ &= \cos 2\pi f_c t + \frac{n_a}{2} \cos 2\pi (f_c \pm f_m) t \end{aligned}$$
  - Resulting signal has a component at original carrier frequency as well as a pair of components each spaced f_m Hz from the carrier
  - Effectively, AM involves multiplication of original signal by carrier
    * Envelope of resulting signal is $[1 + n_a x(t)]$
    * With $n_a < 1$, envelope is an exact reproduction of original signal
    * With $n_a > 1$, envelope crosses the time axis and information is lost
  - Spectrum of an AM signal
    * Figure 5.16
    * Spectrum of AM signal is the original carrier plus spectrum of original signal translated to f_c
    * Portion of spectrum for $|f| > |f_c|$ is *upper sideband*
    * Portion of spectrum for $|f| < |f_c|$ is *lower sideband*
    * Consider a voice signal between 30–3000 Hz with f_c being 60 kHz
    * Upper sideband is given by 60.3–63 kHz
    * Lower sideband is given by 57–59.7 kHz
    * Total transmitted power P_t in $s(t)$ is given by
    $$P_t = P_c \left(1 + \frac{n_a^2}{2}\right)$$
    where P_c is the transmitted power in carrier
    * n_a should be maximized (but < 1) to allow most of the signal power to carry information

- Angle modulation
  - Encompasses frequency modulation (FM) and phase modulation (PM) as special cases
  - Modulated signal is given by
  $$s(t) = A_c \cos[2\pi f_c t + \phi(t)]$$
  - Phase modulation

    * Phase is proportional to modulating signal

$$\phi(t) = n_p m(t)$$

where n_p is phase modulation index

  - Frequency modulation
    * Derivative of phase is proportional to modulating signal

$$\phi'(t) = n_f m(t)$$

where n_f is frequency modulation index

Spread spectrum

- Can be used to transmit either analog or digital data, using analog signal
- Spread the information signal over a wider bandwidth to make jamming and interception more difficult
- Figure 5.19
- Channel encoder
  - Receives input and converts it into an analog signal with relatively narrow bandwidth around some center frequency
  - Signal is further modulated using a pseudorandom sequence
    * Modulation spreads the spectrum (increases the bandwidth) of the signal to be transmitted
    * Pseudorandom numbers are dependent on seed and random number generator algorithm
- Channel decoder
  - Uses the same pseudorandom sequence to demodulate the spread spectrum signal
- Frequency hopping
  - Signal is broadcast over seemingly random sequence of radio frequencies, hopping from one to another in split-second interval
  - Receiver also hops on the same frequencies in synchronization with the sender
  - Difficult to catch the signal if you do not know the frequencies
  - Cannot jam the signal if you do not know the frequencies
- Direct sequence
  - Each bit in original code is represented by multiple bits in the transmitted signal
  - Multiple bits are known as *chipping code*
  - Chipping code spreads the signal across a wider frequency band in direct proportion to the number of bits used
    * A 10-bit chipping code spreads the signal across a frequency band that is 10 times greater than the 1-bit chipping code
  - Combine the digital information stream with the pseudorandom bit stream using an exclusive-or
  - Figure 5.21