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
Data Encoding

- 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
Data Encoding

- 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

\[ 0 \ 1 \ 0 \ 0 \ 0 \ 0 \ 0 \ 1 \ 1 \ 0 \ 1 \ 0 \ 1 \ 1 \ 1 \ 0 \]

* 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

\[ 0 \ 1 \ 0 \ 0 \ 0 \ 0 \ 0 \ 1 \ 1 \ 0 \ 1 \ 0 \ 1 \ 1 \ 1 \ 0 \]

- 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
Data Encoding

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)

- 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 \( \frac{2}{t_B} \)

- 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

<table>
<thead>
<tr>
<th>Polarity of preceding pulse</th>
<th>Number of bipolar pulses (ones) since last substitution</th>
</tr>
</thead>
<tbody>
<tr>
<td>−</td>
<td>000 −</td>
</tr>
<tr>
<td>+</td>
<td>000 +</td>
</tr>
<tr>
<td>−</td>
<td>+00+</td>
</tr>
<tr>
<td>+</td>
<td>−00−</td>
</tr>
</tbody>
</table>

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 light-wave 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 ± 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
Data Encoding

- 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 bit rates on voice grade lines using complex modulation schemes
  * In general

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

where

\[
\begin{align*}
D & \quad \text{modulation rate in baud} \\
R & \quad \text{data rate in bps} \\
L & \quad \text{number of different signal elements} \\
b & \quad \text{number of bits per signal element}
\end{align*}
\]

- 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 \( \Delta F = f_2 - f_1 = f_c - f_1 \)
  * For high frequencies, \( \Delta F \) term dominates
  * FSK signaling on coaxial cable multipoint local network uses \( \Delta F = 1.25 \text{ MHz} \), \( f_c = 5 \text{ MHz} \), and \( R = 1 \text{ Mbps} \), giving \( 2\Delta F = 2.5 \text{ MHz} \) as the dominant term
  * For low frequency (voice grade line modem), we have \( \Delta F = 100 \text{ Hz} \), \( f_c = 1170 \text{ Hz} \) in one direction, and \( R = 300 \text{ bps} \), 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:
        \[
        \text{SNR} = 20 \log 2^n + 1.76 \text{ dB}
        \]
        \[
        = 6.02n + 1.76 \text{ dB}
        \]
      - 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
  - Comanding:
    * 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 ($\delta$) at each sampling interval $T_s$.

Figure 5.13

A bit stream is produced by approximating the derivative of an analog signal rather than 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 $\delta$
   - 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 $\delta$
   - If analog waveform changes rapidly than the staircase can keep up with, there is slope overhead noise that increases with a decrease in $\delta$

Principal advantage of DM is its simplicity of implementation

PCM has better SNR characteristics at the same data rate

Performance

- PCM can produce 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 \( \text{am} \) signal \( \cos 2\pi f_m t \)
    \[ s(t) = [1 + n_a \cos 2\pi f_m t] \cos 2\pi f_c t \]
    \[ = \cos 2\pi f_c t + \frac{n_a}{2} \cos 2\pi (f_c \pm f_m) t \]
  - 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
Data Encoding

* 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