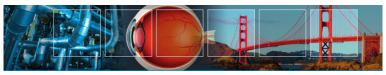
Chapter 5: Physical Layer



Outline

- Basic Components
- Source Encoding
  - The Efficiency of a Source Encode
  - Pulse Code Modulation and Delta Modulation
- Channel Encoding
  - Types of Channels
  - Information Transmission over a Channel
  - Error Recognition and Correction
- Modulation
  - Modulation Types
  - Quadratic Amplitude Modulation
  - Summary
- Signal Propagation

Physical Layer

- One of the desirable aspects of WSNs is their ability to communicate over a wireless link, so
  - mobile applications can be supported
  - flexible deployment of nodes is possible
  - the nodes can be placed in areas that are inaccessible to wired nodes
- Once the deployment is carried out, it is possible to
  - rearrange node placement - optimal coverage and connectivity
  - the rearrangement can be made without disrupting the normal operation

Physical Layer

- Some formidable *challenges*:
  - limited bandwidth
  - limited transmission range
  - poor packet delivery performance because of interference, attenuation, and multi-path scattering
- therefore, it is vital to understand their properties and some of the mitigation strategies
- this chapter provides a fundamental introduction to *point-to-point* wireless digital communication

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Basic Components

- The *basic components* of a digital communication system:
  - transmitter
  - channel
  - receiver
- Here, we are interested in *short range communication* - because nodes are placed close to each other

Basic Components

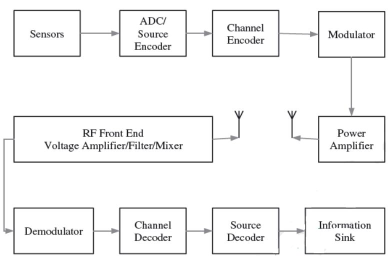

Figure 5.1 provides a block diagram of a digital communication system

Basic Components

- *The communication source* represents one or more sensors and produces *a message signal* - *an analog signal*
  - the signal is a *baseband* signal having dominant frequency components *near zero*
  - the message signal has to be converted to a discrete signal (*discrete* both in *time* and *amplitude*)
- The conversion requires sampling the signal at least at *Nyquist rate* - no information will be lost
  - the Nyquist rate sets a lower bound on the sampling frequency
  - hence, the minimum sampling rate should be twice the bandwidth of the signal

Basic Components

- *Source encoding:* the discrete signal is converted to a binary stream after sampling
- An efficient source-coding technique can satisfy the channel's bandwidth and signal power requirements
  1. by defining a probability model of the information source
  2. *channel encoding* - make the transmitted signal robust to noise and interference
     - transmit symbols from a predetermined codebook
     - transmit redundant symbols
- *Modulation* - the baseband signal is transformed into a bandpass signal
  - main reason is to transmit and receive signals with *short* antennas

Basic Components

- Finally, the modulated *signal* has to *be amplified* and the *electrical energy* is *converted into electromagnetic energy* by the transmitter's antenna
- The signal is *propagated* over a wireless link to the desired destination
- The receiver block carries out the reverse process to retrieve the message signal from the electromagnetic waves
  - the receiver antenna *induces a voltage* that is similar to the modulated signal

Basic Components

- The magnitude and shape of the signal are changed because of losses and interferences
- The signal has to pass through a series of *amplification* and *filtering processes*
- It is then transformed back to a baseband signal through the process of *demodulation* and *detection*
- Finally, the baseband signal undergoes *a pulse-shaping process* and *two* stages of *decoding* (channel and source)
  - extract the sequence of symbols - the original analog signal (the message)

Fundamentals of Wireless Sensor Networks: Theory and Practice
Waltenegus Dargie and Christian Poellabauer © 2010 John Wiley & Sons Ltd.

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Source Encoding

- A source encoder transforms an *analog signal* into a *digital sequence*
- The process consists of: *sampling*, *quantizing*, *encoding*
  - Suppose a sensor produces an analog signal $s(t)$
  - $s(t)$ will be sampled and quantized by the analog-to-digital converter (ADC) that has a resolution of Q distinct values
  - as a result, a sequence of samples, $S = (s[1], s[2], ..., s[n])$ are produced
  - the difference between the sampled $s[j]$ and its corresponding analog value at time t_j is the *quantization error*
  - as the signal varies over time, the quantization error also varies and can be modeled as a random variable with a probability density function, $P_s(t)$

Source Encoding

- The *aim* of the source encoder is to map each quantized element, $s[j]$ into a corresponding binary symbol of length r from a codebook, C
- *Block code*: if all the binary symbols in the codebook are of *equal* length
- Often, the symbol length and the sampling rate are *not uniform*
- It is customary to assign:
  - *short-sized symbols* and *high* sampling rates to the most probable sample values
  - *long-sized symbols* and *low* sampling rates to less probable sample values

Source Encoding

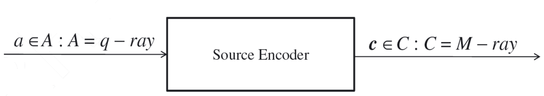

Figure 5.2 illustrates the input – output relationship of a source encoder

Source Encoding

- A codebook, C, can be uniquely decoded, if each sequence of symbols, $(C(1), C(2), ...)$ can be mapped back to a corresponding value in $S = (s[1], s[2], ..., s[n])$
- A binary codebook has to satisfy Equation (5.1) to be uniquely decoded
  - $$\sum_{i=1}^{u}\left(\frac{1}{r}\right)^{l_i} \leq 1$$ Equation (5.1)
  - where u is the size of the codebook
  - l_i is the size of the codeword $C(i)$

Source Encoding

- A codebook can be instantaneously decoded
  - if each symbol sequence can be extracted (decoded) from a stream of symbols *without* taking into consideration previously decoded symbols
- This will be possible
  - *iff* there does *not exist* a symbol in the codebook, such that the symbol $\mathbf{a} = (a_1, a_2, ..., a_m)$ is not a prefix of the symbol $\mathbf{b} = (b_1, b_2, ..., b_n)$, where $m < n$ and $a_i = b_i$, $\forall i = 1, 2, ..., m$ within the same codebook

Source Encoding

	C^1	C^2	C^3	C^4	C^5	C^6
S_1	0	0	0	0	0	0
S_2	10	01	100	10	01	10
S_3	00	10	110	110	011	110
S_4	01	11	11	1110	111	111
Block code	No	Yes	No	No	No	No
Uniquely decoded	No	Yes	No	Yes	Yes	Yes
$\sum_{i=1}^{u}\left(\frac{1}{2}\right)^{l_i}$	$1\frac{1}{4}$	1	1	$\frac{15}{16} < 1$	1	1
Instantly decoded	No	Yes (block code)	No	Yes (comma code)	No	Yes

Table 5.1 Source-encoding techniques

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The Efficiency of a Source Encoder

- Quantity that expresses the average length
- Sampled analog signal: $L(C) = E\,[l_i(C)]$
- Suppose the probability of a q-ary source
  - i.e., it has q distinct symbols
  - producing the symbol s_i is P_i and the symbol C_i in a codebook is used to encode s_i
  - the expected length of the codebook is given by:

$$L(C) = \sum_{i=1}^{q} P_i \cdot l_i(C)$$

Equation (5.2)

The Efficiency of a Source Encode

- To express efficiency in terms of the information entropy or *Shannon's entropy*
  - defined as *the minimum message length* necessary to communicate information
  - related to the *uncertainty* associated with the information
  - if the symbol s_i can be expressed by a binary symbol of n bits, the information content of s_i is:
    - $I(s_i) = -\log_2 P_i = \log_2 \frac{1}{P_i}$ Equation (5.3)
    - the entropy (in bits) of a *q*-ary memoryless source encoder is expressed as:

$$H_r(A) = E[l_r(s_i)] = \sum_{i=1}^{q} P(s_i) \cdot l_r(s_i) = \sum_{i=1}^{q} P(s_i) \cdot \log_2 \frac{1}{P(s_i)}$$

Equation (5.4)

The Efficiency of a Source Encode

- The efficiency of a source encoder in terms of entropy reveals the unnecessary redundancy in the encoding process. This can be expressed by:

$$\eta(\mathrm{C}) = \frac{H(S)}{L(C)}$$

Equation (5.5)

  - The redundancy of the encoder is:

$$\frac{L - H(S)}{L} = 1 - \eta$$

Equation (5.6)

Example

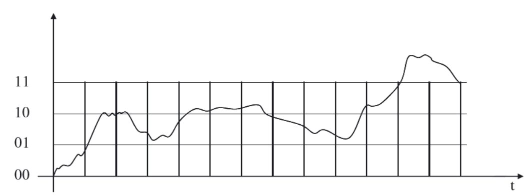

Figure 5.3 An analog signal with four possible values

Example

- In Figure 5.3, it is quantized into four distinct values, $0, 1, 2, 3$
  - some values (2) occur more frequently than others (0 and 3)
  - if the *probability* of occurrence of these values is
    - $P(0) = 0.05, P(1) = 0.2, P(2) = 0.7, P(3) = 0.05$, then,
    - it is possible to compute the efficiency of two of the codebooks given in Table 5.1, namely C^2 and C^3
  - for $P_1 = 0.05$, $\log_2(\frac{1}{0.05}) = 4.3$. Because l_i has to be a whole number and there should be no loss of information, l_1 must be 5. Likewise, $l_2 = 3$; $l_3 = 1$; and $l_4 = 5$. Hence:

$$E[L(C^2)] = \sum_j l_j \cdot P_j = (5 \times 0.05) + (3 \times 0.2) + (1 \times 0.7) + (5 \times 0.05) = 1.8$$

Equation (5.7)

Example

- Using Equation (5.4), the entropy of C^2 is calculated as:

$$H(C^2) = 0.05\log_2(\frac{1}{0.05}) + 0.2\log_2(\frac{1}{0.7}) + 0.7\log_2(\frac{1}{0.7}) + 0.05\log_2(\frac{1}{0.05}) = 1.3$$

Equation (5.8)

- Therefore, the encoding efficiency of the codebook, C^2 (see Table 5.2) is:

$$\eta(C^2) = \frac{1.3}{1.8} = 0.7$$

Equation (5.9)

- The redundancy in C^2 is:

$$rdd_{C^2} = 1 - \eta = 1 - 0.67 = 0.3$$

Equation (5.10)

  - in terms of energy efficiency, this implies that *30% of the transmitted bits are unnecessarily redundant*, because C^2 is not compact enough

Example

j	a_j	P_j	l_j
1	00	0.05	5
2	01	0.2	3
3	10	0.7	1
4	11	0.05	5

Table 5.2 Description of the compactness of C^2

j	a_j	P_j	l_j
1	100	0.05	3
2	11	0.2	2
3	0	0.7	1
4	110	0.05	3

Table 5.3 Description of the compactness of C^3

Example

- In the same way l_j is computed for C^2, the expected symbol length (in bits) for C^3 (see Table 5.3) is given as:

$$E[L(C^3)] = \sum_j l_j \cdot P_j = (3\times 0.05) + (2\times 0.2) + (1\times 0.7) + (3\times 0.05) = 1.4$$

Equation (5.11)

- Because the probabilities of the symbols are unchanged, entropy also remains unchanged. The encoding efficiency of C^3 is therefore:

$$\eta(C^3) = \frac{1.3}{1.4} = 0.9$$

Equation (5.12)

- The redundancy, rdd, in C^3 is:

$$rdd_{C^3} = 1 - \eta = 1 - 0.9 = 0.1$$

Equation (5.13)

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Pulse Code Modulation and Delta Modulation

- PCM and DM are the two predominantly employed *source encoding* techniques
- *In digital pulse code modulation*
  - the signal is quantized first
  - each sample is represented by a binary word from a finite set of words
- The resolution of a PCM technique and the source encoder bit rate are determined by
  - the size of the individual words
  - the number of words in the set

Pulse Code Modulation and Delta Modulation

- In PCM, information is conveyed in the presence or absence of *pulses*
  - greatly enhances the transmission and regeneration of *binary words*
  - the associated cost with this form of source encoding is
    - the quantization error, the energy and bandwidth required to transmit the multiple bits for each sampled output
  - Figure 5.4 illustrates a PCM technique that uses two bits to encode a single sample
  - four distinct levels are permissible during sampling

Pulse Code Modulation and Delta Modulation

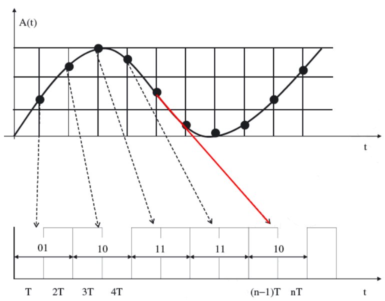

Figure 5.4 A PCM based source encoding

Pulse Code Modulation and Delta Modulation

- *Delta modulation* is a *digital pulse modulation* technique
  - it has found widespread acceptance *in low bit rate* digital systems
  - it is a *differential* encoder and transmits bits of information
  - the information describes *the difference between successive* signal values, as opposed to the actual values of a time-series sequence
  - the difference signal, $V_d(t)$, is produced by first estimating the signal's magnitude based on previous samples ($V_i(t_0)$) and comparing this value with the actual input signal, $V_{in}(t_0)$

Pulse Code Modulation and Delta Modulation

- The polarity of the difference value indicates the polarity of the pulse transmitted
- The difference signal is a *measure* of the *slope of the signal*
  - first, *sampling* the analog signal
  - then, *varying* the amplitude, width, or the position of the digital signal in accordance with the amplitude of the sampled signal
- Figure 5.5 illustrates delta modulation

Pulse Code Modulation and Delta Modulation

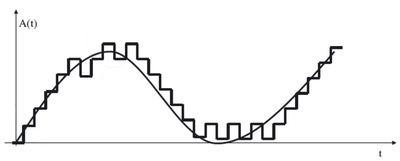

Figure 5.5 Delta encoding

Outline

Channel Encoding

- The main purpose is
  - to produce a sequence of data that is *robust to noise*
  - to provide *error detection*
  - to forward *error correction mechanisms*
- The physical channel sets *limits* to
  - the *magnitude*
  - the rate of signal transmission
- Figure 5.6 illustrates these restrictions

Channel Encoding

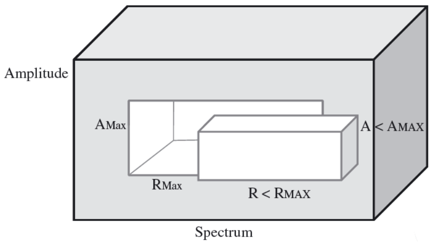

Figure 5.6 Stochastic model of a channel

Channel Encoding

- According to the *Shannon – Hartley theorem*, the capacity of a channel to transmit a message without an error is given as:

$$C = B \cdot \log_2(1 + \frac{S}{N}) \quad \text{Equation (5.14)}$$

  - where C is the channel capacity in bits per second
  - B is the bandwidth of the channel in hertz
  - S is the average signal power over the entire bandwidth, measured in watts
  - N is the average noise power over the entire bandwidth, measured in watts
- Equation (5.14) states that for data to be transmitted free of errors, its transmission rate should be *below* the channel's capacity
- It also indicates how the signal-to-noise ratio (*SNR*) can improve the channel's capacity

Channel Encoding

- The equation reveals *two* independent *reasons* why errors can be introduced during transmission:
  1. information will be lost if the message is transmitted at a rate higher than the channel's capacity - *equivocation* (subtractive error)
  2. information will be lost because of noise, which adds irrelevant information into the signal
- A stochastic model of the channel helps to quantify the impact of these two sources of errors

Channel Encoding

- Suppose an input sequence of data x_l that can have j distinct values, $x_l \in X = (x_1, x_2, ..., x_j)$, is transmitted through a physical channel
- Let $P(x_l)$ denote $P(X = x_l)$
- The channel's output can be decoded with a k-valued alphabet to produce $y_m \in Y = (y_1, y_2, ..., y_k)$
- Let $P(y_m)$ denotes $P(Y = y_m)$
- At time t_i , the channel generates an output symbol y_i for an input symbol x_i

Channel Encoding

- Assuming that the channel distorts the transmitted data, it is possible to model distortion as a stochastic process:

$$P(y_m + x_l) = P(Y = y_m \mid X = x_m) \quad \text{Equation (5.15)}$$

where, $l = 1, 2, ..., j$ and $m = 1, 2, ..., k$

- In the subsequent analysis of the stochastic characteristic of the channel, the following assumptions hold:
  - the channel is discrete, namely, X and Y have finite sets of symbols
  - the channel is stationary, namely, $P(y_m|x_l)$, are independent of the time instance, I
  - the channel is memoryless, namely, $P(y_m|x_l)$, are independent of previous inputs and outputs

Channel Encoding

- One way of describing transmission distortion is by using a channel matrix, P_C

$$P_c = \begin{bmatrix} P(y_1 \mid x_1) & \cdots & P(y_k \mid x_1) \\ \cdot & & \cdot \\ \cdot & & \cdot \\ \cdot & & \cdot \\ P(y_1 \mid x_j) & \cdots & P(y_k \mid x_j) \end{bmatrix} \quad \text{Equation (5.16)}$$

where

$$\sum_{m=1}^{k} p(y_m \mid x_j) = 1 \forall j \quad \text{Equation (5.17)}$$

Moreover:

$$P(y_m) = \sum_{l=1}^{j} P(y_m \mid x_l) \cdot P(x_l) \quad \text{Equation (5.18)}$$

Or, more generally:

$$(\vec{P}_y) = (\vec{P}_x) \cdot [P_C] \quad \text{Equation (5.19)}$$

where both $(\vec{P}_y)$ and $(\vec{P}_x)$ are row matrices

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Types of Channels

- *Binary Symmetric Channel*
  - a channel model
  - bits of information (0 and 1) can be transmitted through it
  - the channel transmits a bit of information
    - *correctly* (regardless of whether information is 0 or 1) with a probability p
    - *incorrectly* (by flipping 1 to 0 and 0 to 1) with a probability $1-p$

$$P(y_0 \mid x_0) = P(y_1 \mid x_1) = 1 - p \qquad \text{Equation (5.20)}$$

$$P(y_1 \mid x_0) = P(y_0 \mid x_1) = p \qquad \text{Equation (5.21)}$$

- the channel matrix of a binary symmetric channel:

$$P_{BSC} = \begin{bmatrix} (1-p) & p \\ p & (1-p) \end{bmatrix} \qquad \text{Equation (5.22)}$$

Types of Channels

- Binary Symmetric Channel

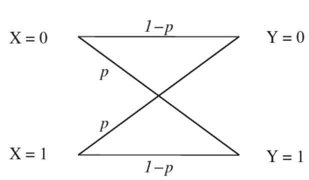

Figure 5.7 A binary symmetric channel model

Types of Channels

- *Binary Erasure Channel*
  - in a BEC, there is *no guarantee* that the transmitted bit of information can be received at all (correctly or otherwise)
  - a binary input - a ternary output channel
  - the probability of erasure is p and the probability that the information is correctly received is $1-p$
  - the probability of error is zero

$$P_{BSC} = \begin{bmatrix} (1-p) & p & 0 \\ 0 & p & (1-p) \end{bmatrix}$$

Equation (5.23)

- a bit of information
  - either transmitted successfully with $P(1|1) = P(0|0) = 1-p$
  - or erased altogether by the channel with a probability of p
- the probability that 0 is received by transmitting 1 or vice versa is 0

Types of Channels

- Binary Erasure Channel

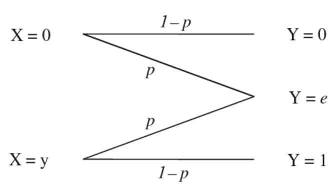

Figure 5.8 A stochastic model of a binary erasure channel

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Information Transmission over a Channel

- Given the input message, $X:(X, \xrightarrow{P_X}, H(X))$, the channel matrix, [*PC*]
- The output message, $Y:(Y, \xrightarrow{P_Y}, H(Y))$ describe
  - the impact of *irrelevance* and *equivocation*
  - the percentage of information be transmitted over the channel without an error ---- *transinformation* or *mutual information*
- *Irrelevance*
  - the content of information that can be introduced into the channel due to noise is described as the conditional information content, $I(y|x)$
    - the information content of y that can be observed provided that x is *known*. The conditional entropy is given as:

$$\mathrm{H(y \mid x)} = \mathrm{E}_y[I(y \mid x)] = \sum_{y \in Y} P(y \mid x) \cdot \log_2\left(\frac{1}{P(y \mid x)}\right) \quad \text{Equation (5.24)}$$

Information Transmission over a Channel

- $P(y|x)$ can be known from the channel matrix $[PC]$. The average conditional entropy over all input message symbols, $x \in X$, is given by:

$$H(Y \mid X) = E_x[H(Y \mid x)] = \sum_{x \in X} P(x) \cdot \sum_{y \in Y} P(y \mid x) \cdot \log_2\left(\frac{1}{P(y \mid x)}\right) \quad \text{Equation (5.25)}$$

  - which is also equal to:

$$H(Y \mid X) = E_x[H(Y \mid x)] = \sum_{x \in X} \sum_{y \in Y} P(y \mid x) \cdot P(x) \cdot \log_2\left(\frac{1}{P(y \mid x)}\right) \quad \text{Equation (5.26)}$$

  - from Baye's law, it is clear that:

$$p(x, y) = P(y \mid x) \cdot P(x) \quad \text{Equation (5.27)}$$

  - according to Equation (5.26), a good channel encoder is one that *reduces the irrelevance entropy*

Information Transmission over a Channel

- *Equivocation*
  - the content of information that can be lost because of the channel's inherent constraints can be quantified by observing the input x given that the output y is known:

$$H(Y \mid X) = \sum_{x \in X} \sum_{y \in Y} P(x \mid y) \cdot P(y) \cdot \log_2\left(\frac{1}{P(y \mid x)}\right) \quad \text{Equation (5.28)}$$

  - once again, applying *Baye's conditional probability*:

$$P(y \mid x) = \frac{P(y \mid x)P(x)}{P(y)} = \frac{P(y \mid x)P(x)}{\sum_{x \in X} P(y \mid x)P(x)} \quad \text{Equation (5.29)}$$

  - a good channel encoding scheme has a high inference probability
    - by introducing redundancy during channel encoding

Information Transmission over a Channel

- *Transinformation*
  - the information content $I\ (X;Y)$ is called transinformation
    - overcomes the channel's constraints
    - reaches the destination (the receiver)
  - Given the input entropy, $H(X)$ and equivocation, $H(X|Y)$
    - the transinformation is computed as:
    - $I(X;Y) = H(X) - H(X \mid Y)$ Equation (5.30)
  - Expanding Equation (5.30) yields:
    - $\sum_{x \in X} P(x) \cdot \log_2\left(\frac{1}{P(x)}\right) - \sum_{x \in X}\sum_{y \in Y} P(x \mid y) \cdot \log_2\left(\frac{1}{P(x \mid y)}\right)$ Equation (5.31)
  - Rearranging the terms in Equation (5.31) also yields:
    - $H(Y) - H(X \mid Y) = I(Y;X)$ Equation (5.32)

Information Transmission over a Channel

- Summarize:

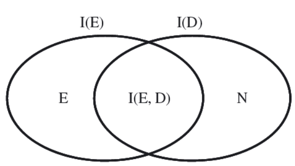

Figure 5.9 Irrelevance, equivocation, and transinformation

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  - Quadratic Amplitude Modulation
  - Summary
- Signal Propagation

Error Recognition and Correction

- Error recognition
  - by permitting the transmitter to transmit *only specific types of words*
  - if a channel decoder recognizes unknown words
    - it corrects the error or requests for retransmission (automatic repeat request, ARQ)
    - a decoder can correct only m number of errors
      - where m depends on the size of the word
- Error correction
  - by sending n bits of information together with r control bits
  - problem: *it slows down transmission*

Outline

- Basic Components
- Source Encoding
  - The Efficiency of a Source Encode
  - Pulse Code Modulation and Delta Modulation
- Channel Encoding
  - Types of Channels
  - Information Transmission over a Channel
  - Error Recognition and Correction
- ***Modulation***
  - Modulation Types
  - Quadratic Amplitude Modulation
- Signal Propagation

Modulation

- Modulation is a process where
  - characteristics (amplitude, frequency, and phase) of a carrier signal are modified according to the message (a baseband) signal
- Modulation has several *advantages*:
  - the message signal will become *resilient to noise*
  - the channel's spectrum can be *used efficiently*
  - signal detection will *be simple*

Outline

- Basic Components
- Source Encoding
  - The Efficiency of a Source Encode
  - Pulse Code Modulation and Delta Modulation
- Channel Encoding
  - Types of Channels
  - Information Transmission over a Channel
  - Error Recognition and Correction
- Modulation
  - ***Modulation Types***
  - Quadratic Amplitude Modulation
  - Summary
- Signal Propagation

Modulation Types

- The message signal is a *baseband signal*
  - its dominant frequency components are in the vicinity of *zero*
  - if without any modulation
    - the size of receiver antenna should equal to one-fourth of the size of the signal's wavelength
      - such an antenna is very *long* and it is *impractical* to deploy
    - or, superimpose the message signal on a *bandpass carrier signal*
      - wavelength of carrier signal is very much *smaller* than the baseband signal
      - *sinusoidal* carrier signals are used for modulation

$$s_c(t) = S_C \sin(2\pi f t + \phi(t)) \qquad \text{Equation (5.33)}$$

      - where S_C is the peak amplitude of the signal
      - f is the frequency; and $\varphi(t)$ is the phase

Modulation Types

- A radio frequency signal can also be described in terms of its *wavelength*
  - a function of the propagation speed and the frequency
  - Figure 5.10 shows two sinusoidal signals that have the same frequency and amplitude, but are also out of phase by φ degrees
- Figure 5.11 shows the how to use *polar presentation* to describe the relationship between two sinusoidal signals that have the same frequency

Modulation Types

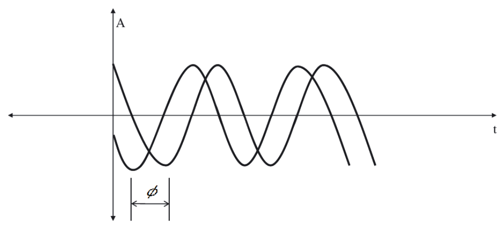

Figure 5.10 Two signal having a phase difference of ϕ

Modulation Types

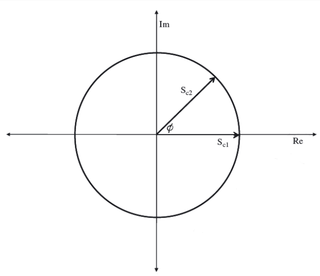

Figure 5.11 Representation of a relationship between signals with a polar diagram

Modulation Types

- A message signal, $s_m(t)$, can change
  - either the amplitude, the phase or frequency of $s_c(t)$
    - if $s_m(t)$ changes the *amplitude* of $s_c(t)$, the modulation is known as amplitude modulation *(AM)*
    - if $s_m(t)$ changes the *frequency* of $s_c(t)$, the modulation is known as frequency modulation *(FM)*
    - if $s_m(t)$ changes the *phase* of $s_c(t)$, the modulation is known as phase modulation
- $s_m(t)$ can be a digital (binary) signal
  - amplitude shift keying (ASK)
  - frequency shift keying (FSK)
  - phase shift keying (PSK)

Modulation Types

- A modulation process can further be classified into
  - *coherent* or *non-coherent*
  - *binary* or *q-ary*
  - *power-efficient* or *spectrum-efficient*
- In a coherent modulation technique
  - *a carrier signal* of the same frequency (and ideally, of the same phase) *is required* to demodulate (detect) the received signal
- In a non-coherent modulation technique
  - *no* additional *carrier signal is required* to demodulate the received signal

Modulation Types

- In a binary modulation
  - the modulating (message) signal is *binary*
- In a q-ary modulation
  - the modulating signal can have m discrete values
- In a power-efficient modulation technique
  - the aim is to optimize the *power* of the modulated signal
- In a spectrum-efficient modulation technique
  - the aim is to optimize the *bandwidth* of the modulated signal

Modulation Types

- ***Amplitude Modulation***
  - considering that both the carrier and the modulating signals are analog sinusoidal signals, an amplitude modulation can be described as follows:

$$s_{\text{mod}}(t) = [S_C \times S_M \cos(2\pi f_m t + \phi_m)]\cos(2\pi f_c t + \phi_c) \quad \text{Equation (5.34)}$$

  - the amplitude of $s_c(t)$ is varied according to the modulating signal, $s_m(t)$. To simplify the analysis, assume that the two signals are in phase $(\varphi_m = \varphi_c = 0)$ and thus, Equation (5.34) reduces to:

$$s_{\text{mod}}(t) = [S_C \times S_M \cos(2\pi f_m t)]\cos(2\pi f_c t) \quad \text{Equation (5.35)}$$

  - applying Euler's formula ($e^{j\omega t} = \cos(\omega t) + j\sin(\omega t)$), Equation (5.35) reduces to:

$$s_{\text{mod}}(t) = \frac{S_C \times S_M}{2}[\cos(2\pi(f_c + f_m)t) + \cos(2\pi(f_c - f_m)t)]$$

Equation (5.36)

Modulation Types

- In reality, the message signal is a baseband signal
  - it has a *bandwidth* of B
  - in B the amplitude and frequency change as functions of time
- The *Fourier transformation* of such a baseband signal resembles the one displayed in Figure 5.12
- The *Fourier transformation of the carrier signal* is displayed in Figure 5.13
- Hence, the spectrum of the amplitude modulated signal based on Figure 5.12 and Figure 5.13 looks like the one displayed in Figure 5.14.

Modulation Types

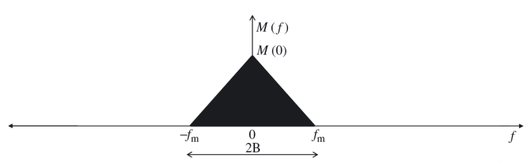

Figure 5.12 The spectrum of a baseband signal having a bandwidth of *B*

Modulation Types

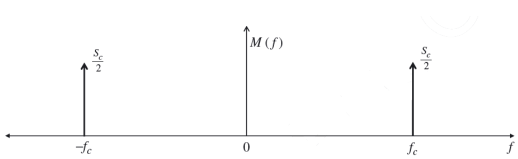

Figure 5.13 The Fourier transformation of a carrier signal having a frequency of f_c

Modulation Types

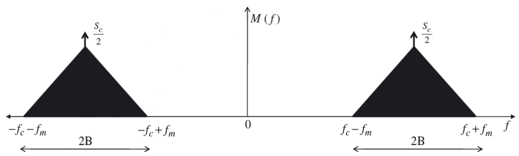

Figure 5.14 The Fourier transformation of an amplitude modulated signal

Modulation Types

- Figure 5.15 illustrates *amplitude modulation*
  - the baseband signal and the carrier signal are *mixed* by using a mixer (an amplifier having a bandwidth *greater* than the bandwidth of the baseband signal)

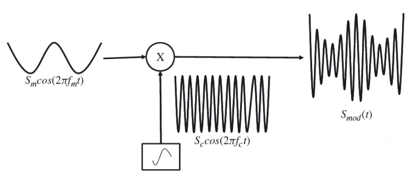

Figure 5.15 Amplitude modulation

Modulation Types

- The demodulation process
  - the extraction of the message signal from the modulated signal
  - first, the received modulated signal is mixed with a carrier signal that has the same frequency as the original carrier signal, $S_C(t)$

$$s_{demo}(t) = S_C \cos(2\pi f_c t) \times s_{\text{mod}}(t)$$ Equation (5.37)

  - expanding Equation (5.37) yields:

$$s_{demo}(t) = S_C \cos(2\pi f_c t) \times \frac{KS_C \times S_M}{2}\left[\cos(2\pi(f_c + f_m)t) + \cos(2\pi(f_c - f_m)t)\right]$$

Equation (5.38)

  - where $K1<<1$, ---- the modulated signal is attenuated. Applying properties of trigonometry, Equation (5.38) can be simplified into:

$$s_{demo}(t) = \frac{KS^2{}_C \times S_M}{4}\left[\cos(2\pi(2f_c - f_m)t) + \cos(2\pi(2f_c + f_m)t) + 2\cos(2\pi f_m t)\right]$$

Equation (5.39)

Modulation Types

- Equation (5.39) contains:
  - the message signal
  - a carrier signal
  - the two components can very easily be separated by
    - a envelope detector consisting of *a half-wave rectifier* and *a low-pass filter*
- Figure 5.16 shows how a modulated signal is mixed with a carrier signal generated by the local oscillator of the receiver
  - the result passes through a bandpass filter (not shown here) to remove the f_c component
  - afterwards, a simple half-wave rectifier and a lowpass filter are used to retrieve the message (baseband) signal

Modulation Types

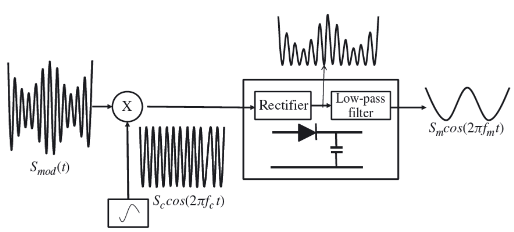

Figure 5.16 Demodulating an AM carrier signal

Modulation Types

- ***Frequency and Phase Modulation***
  - the amplitude of the carrier signal, $s_c(t)$, remains intact
  - but its frequency changes according to the message signal, $s_m(t)$
  - here, it is essential to restrict the amplitude of the modulating signal such that $|s_m(t)| \leq 1$
  - hence, the modulated signal is described as follows:

$$s_{FM}(t) = S_C \cos\left(2\pi \int_0^t f(\tau) d\tau\right) \quad \text{Equation (5.40)}$$

  - where $\int_0^t f(\tau) d\tau$ is the instantaneous variation of the local oscillator's frequency

Modulation Types

- Expressing this frequency variation as a function of the modulating signal yields:

$$s_{FM}(t) = S_C \cos\left(2\pi\int_0^t \left[f_c + f_\delta s_m(\tau)\right]d\tau\right) \quad \text{Equation (5.41)}$$

  - where f_δ is the maximum frequency deviation of the carrier frequency, f_c
- Rearranging the terms in Equation (5.41) yields:

$$s_{FM}(t) = S_C \cos\left(2\pi f_c t + 2\pi f_\delta \int_0^t s_m(\tau)d\tau\right) \quad \text{Equation (5.42)}$$

- In phase modulation, the phase of the carrier changes in accordance with the message signal

Modulation Types

- ***Amplitude Shift Keying***
  - a digital modulation technique - the amplitude is a binary stream
  - the frequency and phase of the carrier signal remain unchanged
- *The on – off modulation system*
  - the mixer produces an output - multiplication of the *two* input signals
    - one is the message stream
    - another one is the output of the local oscillator
    - the sinusoidal carrier signal having a frequency of f_c (Figure 5.17)
  - it requires a mixer with an *excessive bandwidth* - *expensive* to afford

Modulation Types

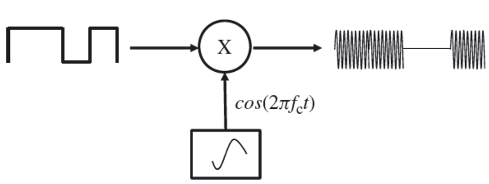

Figure 5.17 Amplitude shift-keying technique using an on – off switch

Modulation Types

- *Pulse-shaping filter (PSF)*
  - removes high-frequency components from the square wave signal
  - approximates it with a low-frequency signal
  - then modulate the carrier signal
- The demodulation process
  - employs a *mixer*, a local *oscillator*, a *PSF*, and a *comparator*
  - to remove the high-frequency component from the modulated signal
  - the comparator changes the analog wave form into a stream of bits

Modulation Types

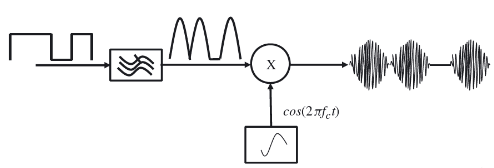

Figure 5.18 An amplitude shift-keying process using a pulse-shaping filter

Modulation Types

- ***Frequency Shift Keying***
  - the *frequency* of a carrier signal *changes* in accordance with the message bit stream *between two values*
    - because the message bit stream will have either 0 or 1
  - Figure 5.19 demonstrates how a simple switching amplifier and two local oscillators with carrier frequencies f_1 and f_2 can be used in frequency shift-keying modulation
  - the switching amplifier is controlled by the message bit stream
  - the *demodulation process* requires two local oscillators (with frequency f_1 and f_2), two PSFs and a comparator (Figure 5.20)

Modulation Types

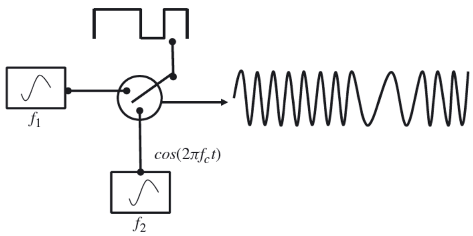

Figure 5.19 A frequency shift-keying modulation

Modulation Types

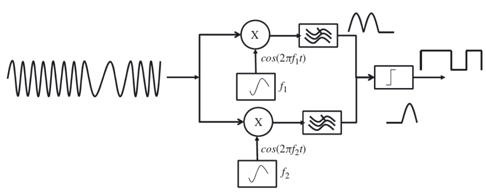

Figure 5.20 Demodulation in a frequency shift-keying process

Modulation Types

- ***Phase Shift Keying***
  - a *carrier signal* is *changed* according to the message bit stream
  - *make a phase shift of 180°* when the bit stream changes from 1 to 0 or vice versa (Figure 5.21)
- The modulation process requires
  - *a local oscillator*, *an inverter*, *a switching amplifier*, and *a PSF*
  - the inverter is responsible for inverting the carrier signal by 180°
  - alternatively, *a PSF*, *a mixer*, and *a local oscillator* (Figure 5.22)
- The demodulation process uses
  - *a local oscillator*, *a mixer*, *a PSF*, and *a comparator* (Figure 5.23)

Modulation Types

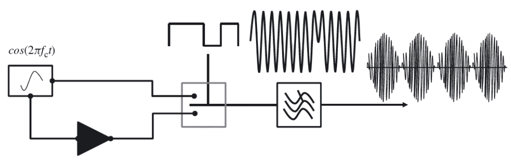

Figure 5.21 A phase shift-keying modulation process

Modulation Types

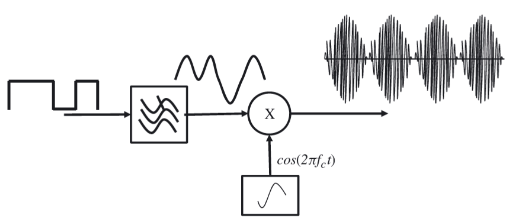

Figure 5.22 A phase shift-keying modulation with a PSF

Modulation Types

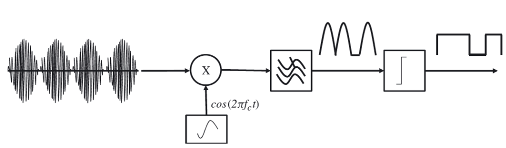

Figure 5.23 A demodulation scheme for a phase shift keying

Outline

- Basic Components
- Source Encoding
  - The Efficiency of a Source Encode
  - Pulse Code Modulation and Delta Modulation
- Channel Encoding
  - Types of Channels
  - Information Transmission over a Channel
  - Error Recognition and Correction
- Modulation
  - Modulation Types
  - Quadratic Amplitude Modulation
  - ***Summary***
- Signal Propagation

Quadratic Amplitude Modulation

- *A single* message source is used to modulate *a single* carrier signal - *not efficient enough*
- Employ *orthogonal signals* to effectively exploit the channel's bandwidth
- In the QAM process
  - *two* amplitude-modulated, orthogonal carriers are combined as a composite signal
  - achieving *double bandwidth efficiency* compared to the normal amplitude modulation
- QAM is used with *pulse amplitude modulation (PAM)* in digital systems
  - the modulated bit stream is divided into two parallel sub-streams each of which independently modulates the two orthogonal carrier signals

Quadratic Amplitude Modulation

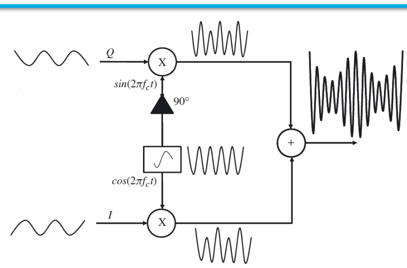

Figure 5.24 A quadratic amplitude modulation process

Quadratic Amplitude Modulation

- The carrier signals have the same frequency, f_c, but they are out of phase by 90°
- Since the signals are orthogonal, they *do not interfere* with each other
- One of the carriers is called the I carrier (*in-phase signal*) and the other is called the Q signal (*quadrature signal*)
  - recall that:

$$s_Q(t) = S_C \cos(2\pi f t + 90^\circ) = S_C \sin(2\pi f t) \quad \text{Equation (5.43)}$$

- At the receiver side, the composite modulated signal will be mixed with *two* demodulating signals
  - they are *identical in frequency* but *out of phase with each other by 90°*

Quadratic Amplitude Modulation

- The demodulation process of a QAM signal (Figure 5.25)
  - the composite signal arrives at the receiver
  - the input signal
    - one has a reference zero phase
    - while the other has a 90° phase shift
  - The composite input signal is thus split into an in-phase, I , and a quadrature, Q, components
    - they are *independent* and *orthogonal* ---- One can be changed
    - *without* affecting the other
  - Digital modulation is *easy* to accomplish with I/Q modulators
    - map the data to *constellation points* (a number of discrete points) on the I/Q plane

Quadratic Amplitude Modulation

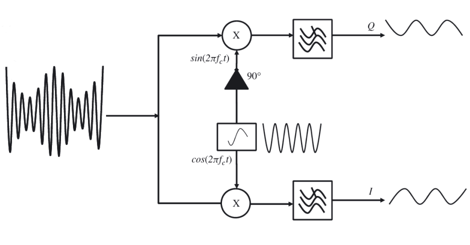

Figure 5.25 Demodulating a QAM signal

Quadratic Amplitude Modulation

- ***Modulation Efficiency***
  - the modulation efficiency refers to
    - *the number of* bits of information that can be conveyed in a single symbol
    - in a QAM, the composite carrier signal contains two orthogonal signals
      - a receiver is sensitive enough to detect the differences between these two signals
      - much information can be conveyed with a single state of the composite carrier signal
    - however, a *tradeoff* between the compactness of the modulated technique and the receiver's complexity

Quadratic Amplitude Modulation

- *Bit rate* vs. *Symbol rate*
  - *bit rate* refers to the frequency of a system's bit stream
  - a *symbol rate* (baud rate)
    - refers to the bit rate divided by the number of bits that can be transmitted with each symbol
  - *for example*, a 10-bit ADC that samples an accelerometer sensor at a rate of 1 KHz has a bit stream of 10 bits multiplied by 1 KHZ samples per second, or 10 kbps
  - *quadrature phase shift keying* (*QPSK*) digital modulation
    - a phase difference of 90° between the *I* and *Q* carrier signals indicates a message of 1 or 0
    - in Figure 5.26, The four states can be represented by two bits: 00, 01, 10, 11. Subsequently, the symbol rate is half of the bit rate
    - for the ADC example, the symbol rate is 5 kbps

Quadratic Amplitude Modulation

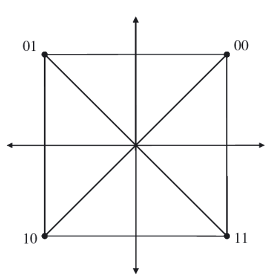

Figure 5.26 Binary phase shift keying: 2 bits per symbol

Quadratic Amplitude Modulation

- An eight-state phase shift-keying modulation
  - it can be mapped into *eight distinct symbols* by the demodulator
  - the eight symbols can be represented by *3 bits*, the symbol rate is *one-third* of the bit rate
  - the 8PSK modulator should be able to discriminate *eight different transitions* in phase of the composite carrier signal - the efficiency in spectrum is not achieved without a cost

Quadratic Amplitude Modulation

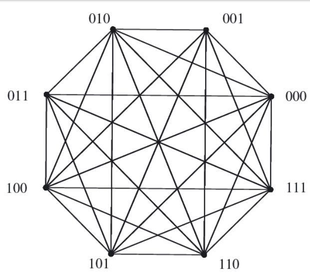

Figure 5.27 8PSK modulation with 3 bits per symbol

Summary

- The choice of a modulation technique depends on
  - the design goals of the communication subsystem
- There is a tradeoff between
  - *power consumption*, *spectrum efficiency*, and *cost*
  - *a power efficient modulator* enables a communication system to reliably transmit information at the lowest practical power cost
  - *a spectrally efficient modulator* enables a communication subsystem to send as many bits of information as possible within a limited bandwidth
  - power and spectrum efficiency *cannot be achieved at the same time*

Summary

- For terrestrial links, the concern is *bandwidth efficiency with low bit-error-rate*
  - power efficiency, the receiver's cost or complexity are not prior concerns
- In wireless sensor networks, *power efficiency and* the cost of the transceivers (in large-scale deployments) are prime concern
  - bandwidth is not prior concerns
- Subsequently, the communication subsystems *sacrifice bandwidth efficiency* to *achieve power and cost efficiency*

Outline

Signal Propagation

Spectrum	Center frequency	Availability
6.765–6.795MHz	6.780MHz	Subject to local regulations
13.553–13.567MHz	13.560MHz	
26.957–27.283MHz	27.120MHz	
40.66–40.70MHz	40.68MHz	
433.05–434.79MHz	433.92MHz	Europe, Africa, the Middle East west of the Persian Gulf including Iraq, the former Soviet Union and Mongolia
902–928MHz	915MHz	The Americas, Greenland and some of the eastern Pacific Islands
2.400–2.500 GHz	2.450 GHz	
5.725–5.875 GHz	5.800 GHz	
24–24.25 GHz	24.125 GHz	
61–61.5 GHz	61.25 GHz	Subject to local regulations
122–123 GHz	122.5 GHz	Subject to local regulations
244–246 GHz	245 GHz	Subject to local regulations

Table 5.4 The Industry, Scientific and Medical (ISM) spectrum as defined by the ITU-R

Signal Propagation

- Wireless sensor networks
  - must *share the spectrum* with and *accept interference from devices* that operate in the same spectrum
    - such as cordless phones, WLAN, Bluetooth, Microwave oven
- A simple channel model (Figure 5.28)
  - *ignores* the effect of *interference*
  - *considers* the surrounding *noise* as the predominant factor that affects the transmitted signal
  - the noise can be modeled as an *additive white Gaussian noise (AWGN)*
    - has a constant spectral density over the entire operating spectrum
    - has a normal amplitude distribution
    - the noise distorts the amplitude of the transmitted signal

Signal Propagation

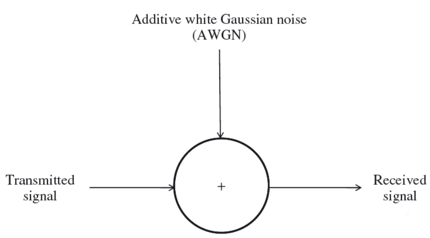

Figure 5.28 An additive white Gaussinan noise channel

Signal Propagation

- One can also use *a spread spectrum technique* to distribute the energy of the transmitted signal
  - a wider effective bandwidth can be achieved
- The received power can be improved by *adjusting* a number of *parameters*
  - the relationship between the received power and the transmitted power can be expressed using Figure 5.29
    - suppose the power amplifier outputs a constant transmission power, P_t, to transmit the signal over a distance of ρ
    - the relationship between the transmitter's antenna gain, g_t, and the antenna's effective area, A_t, is expressed as:

$$A_t = g_t \frac{\lambda^2}{4\pi}$$ Equation (5.44)

- *where* λ is the wavelength of the carrier signal

Signal Propagation

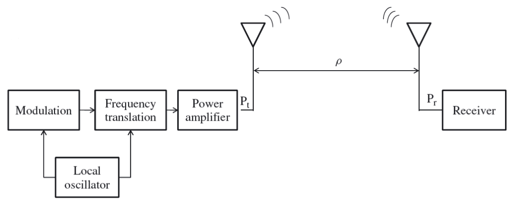

Figure 5.29 Relationship between the transmitted power and the received power

Signal Propagation

- At the receiver's side, the transmitted signal will be received and the received power is a function of
  - the distance
  - the path loss index
  - the receiver's antenna gain and effective area
- A *line-of-sight (LOS)* communication link
  - the path loss index is *2*
- A *non-LOS* communication link
  - it lies *between 2 and 4*

Signal Propagation

- The relationship between the received power and the transmitted power for a LOS link is expressed as:

$$P_r = \frac{P_t}{4\pi\rho^2} g_t \times A_r \qquad \text{Equation (5.45)}$$

  - where ρ is the distance that separates the transmitter and the receiver. Since the receiver's antenna gain, g_r, and the effective area, A_r, are related, Equation (5.45) can be reformulated:

$$P_r = \frac{P_t}{4\pi\rho^2} g_t \times g_r \frac{\lambda^2}{4\pi} \qquad \text{Equation (5.46)}$$

  - the ratio of the transmitted power to the received power, P_t / P_r is the propagation loss and it is customary to quantify this ratio in decibels (dBs)

$$a(t) = \frac{P_t}{P_r} = \left(\frac{4\pi\rho}{\lambda}\right) \times \frac{1}{g_r g_t} \qquad \text{Equation (5.47)}$$

Signal Propagation

- Hence, the propagation loss expressed in dBs is:

$$a(t)/dB = 20\log\left(\frac{4\pi\rho}{\lambda}\right) - 10\log(g_r g_t)$$ Equation (5.48)

  - the term $20\log(4\pi\rho/\lambda)$ is called the basic transmission loss and is independent of the transmitter and receiver antennas

$$a(t)/dB = 20\log\left(\frac{4\pi\rho}{\lambda}\right) - 10\log(g_r g_t)$$