# B. Signal Encoding and Decoding Techniques

## B1. Introduction

- Studying the various encoding/decoding methods for digital data/signals and analog data/signals.

- Encoding: process of converting data into signals.

- Decoding: process of converting signals into data.

- Four different ways to encode data into signals/decode signals into data:
  See Figure 3.14 (p.74).

## B2. Digital Data ⇔ Analog Signal

- The device used is called a **modem** (**mo**dulation/**dem**odulation).

- Modulation:
  the process, or result of the process, of varying certain characteristics of a signal, called a **carrier**, in accordance with a message signal. The message signal can be digital or analog.

  (A **carrier** is a continuous/analog, constant and high frequency signal.)



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- Demodulation:
  the process of restoring a signal that has been modulated to its original form.

  See Figure 5.1(b) (p.130),
  where,
  m(t): digital data (or modulating signal, baseband signal)
  x(t): analog signal (or modulated signal, bandlimited signal, bandpass signal)
  $f_c$: frequency of carrier signal

- Modulation techniques:

  – Basic techniques:
    (1) **A**mplitude-**s**hift **K**eying (**ASK**)
    (2) **F**requency-shift **K**eying (**FSK**)
    (3) **P**hase-**s**hift **K**eying (**PSK**)

  – Variations:
    (4) Multi-level signaling
    (5) Hybrid modulation

  – Recall key data transmission terms:
    See Table 5.1 (p.132).

### (1) ASK

- Coding rules for binary data:
  See Figure 5.7(a) (p.142).

Given that the carrier signal is $A\cos(2\pi f_c t + \theta_c)$,
select two amplitude values, A and 0,
represent 1 and 0 (or, 0 and 1) by

$$x(t) = \begin{cases} A\cos(2\pi f_c t + \theta_c) & \text{for binary 1 (or 0)}, \\ 0 & \text{for binary 0 (or 1)}. \end{cases}$$

Then, x(t) is the modulated signal.

- ASK is susceptible to sudden gain changes, and is a rather inefficient modulation technique on voice-grade lines.

- ASK is suitable for transmitting digital data over optical fibre cables (with repeaters).

**(2) FSK (or BFSK: Binary FSK or Bi-level FSK)**

- Coding rules for binary data:
  See Figure 5.7(b) (p.142).

Given that the carrier signal is $A\cos(2\pi f_c t + \theta_c)$,
select two frequencies, $f_1$ and $f_2$ near the carrier frequency $f_c$, such that
(i) $f_1 < f_c < f_2$, and
(ii) $f_d$ (difference frequency) = $f_2 - f_c = f_c - f_1$;
represent 1 and 0 (or, 0 and 1) by

$$x(t) = \begin{cases} A\cos(2\pi f_1 t + \theta_c) & \text{for binary 1 (or 0)}, \\ A\cos(2\pi f_2 t + \theta_c) & \text{for binary 0 (or 1)}. \end{cases}$$

Then, x(t) is the modulated signal.



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- FSK is less susceptible to error than ASK.

- FSK is used
  on voice-grade lines up to 1200 bps;
  for high-frequency radio transmission (3-30 MHz);
  at even higher frequencies on LANs with coaxial cables.

- An application:

  – a full-duplex operation on a telephone line for a bandwidth of about 300-3400 Hz.

  – dividing the available bandwidth into two halves and using half for transmission in one direction and the other half for the reverse direction.
  See Figure 5.8 (p.144).

  – from transmitter to receiver,
  use a carrier frequency, $f_c$ = 1170 Hz.
  choose $f_d$ = 100 Hz.
  represent a 1 using $f_1 = f_c - f_d$ = 1070 Hz.
  represent a 0 using $f_2 = f_c + f_d$ = 1270 Hz.

  – from receiver to transmitter,
  use a carrier frequency, $f_c$ = 2125 Hz.
  choose $f_d$ = 100 Hz.
  represent a 1 using $f_1 = f_c - f_d$ = 2025 Hz.
  represent a 0 using $f_2 = f_c + f_d$ = 2225 Hz.

## (3) PSK (or BPSK: Binary PSK or Bi-level PSK)

- Coding rules for binary data:
  See Figure 5.7(c) (p.142).

  Given that the carrier signal is $A\cos(2\pi f_c t + \theta_c)$,
  select two phase angles:
  $\theta_c = 0$ and $\pi$ radians,
  or, $\theta_c = \pi/2$ and $3\pi/2$ radians;
  represent 1 and 0 (or, 0 and 1) by

  $$x(t) = \begin{cases} A\cos(2\pi f_c t + 0) & \text{for binary 1 (or 0),} \\ A\cos(2\pi f_c t + \pi) & \text{for binary 0 (or 1).} \end{cases}$$

  Then, x(t) is the modulated signal.

- Define the reference signal:

  (i) the phase of the current signal is shifted wrt a constant reference signal - known as **non-differential PSK** (**NPSK**).
  See Figure 5.7(c) (p.142).

  (ii) the phase of the current signal is shifted wrt the previous signal - known as **differential PSK** (**DPSK**).
  See Figure 5.10 (p.146).

- PSK is susceptible to random phase changes.



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## (4) Multi-level signalling

- With multi-level signalling, use more than two values of amplitude, frequency, or phase angles so that each signal element may contain two or more bits of encoded data.
  i.e., data rate (bps) > baud rate (baud).

- An example - **QPSK** (**Q**uadrature **P**hase-**s**hift **K**eying) or **Four-level PSK**:

  – based on a 4-phase DPSK scheme using phase shifts of multiples of 45° (45°, 135°, 225°, and 315° ).

  – each signal element represents 2 bits.

  – the resulting signal is

  $$x(t) = \begin{cases} A\cos(2\pi f_c t + 45^o) & \text{for binary 11} \\ A\cos(2\pi f_c t + 135^o) & \text{for binary 10} \\ A\cos(2\pi f_c t + 225^o) & \text{for binary 00} \\ A\cos(2\pi f_c t + 315^o) & \text{for binary 01} \end{cases}$$

- In general,
  higher bit rates can be achieved by associating more bits per signal element and using more signal definitions.

But,

using more signal definitions reduces the differences among them and increases the probability that a small amount of noise can make one signal element look like another.

## (5) Hybrid modulation

- One practical solution is –

  use a combination of amplitude and phase modulation.

This method is termed

**Q**uadrature **A**mplitude **M**odulation (**QAM**),

and is used in ADSL and some wireless standards.

- A diagram, called **signal constellation** or **signal-space diagram**, uses

(i) points plotted on a coordinate system to define all the different phase/amplitude states for the bit values;

and (if any);

(ii) circles to indicate the maximum limits of noise levels permissible at each phase/amplitude state before an error can occur (called **noisy regions**).



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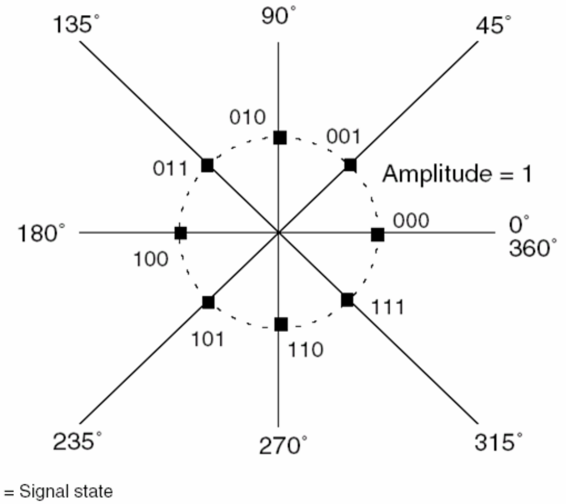
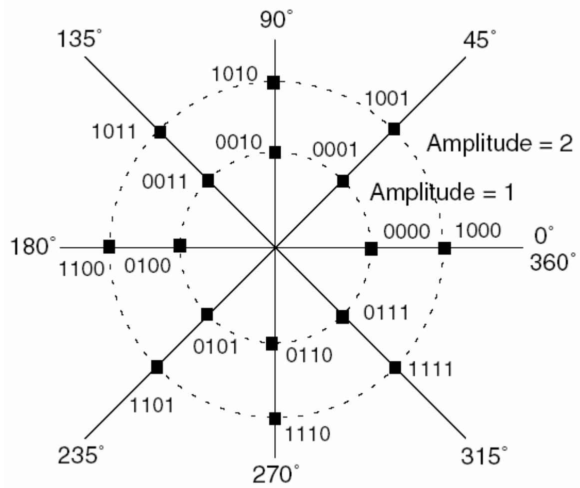
- Examples:

(i) 8-level PSK (with tribits per signal element):




(ii) A variation of QAM (with quadbits per signal change):



## (6) Modem standards

- Other considerations:

  – A modified version of QAM called **TCM** (**T**rellis-**C**oded **M**odulation) incorporates extra bits for error correction.

  – Digital signal processing and data compression techniques can be used to increase the apparent speed of a modem.
  e.g. V.42 bis standard (from 9,600 to 128,000 bps).

- Some ITU-T standards for modems:

**Selected V.x Modem Protocols**

| Protocol | Description |
|---|---|
| V.21 | Standard for 300-bps modems using full-duplex transmission over a dialup line. |
| V.22 | Standard for 600-bps and 1200-bps full-duplex modems over dialup and two-wire leased lines. Compatible with the Bell 212A standard used in the United States. |
| V.22 *bis* | Standard for 2400-bps full-duplex modems over dialup and two-wire leased lines; cycles to 1200- and 600-bps operation. |
| V.23 | Standard for 600-bps or 1200-bps synchronous or asynchronous half-duplex modems used on dialup lines. Used in the United Kingdom. |
| V.29 | Standard for 9600-bps facsimile service. |
| V.32 | Standard for 9600-bps modems, cycles to 4800 bps when line quality degrades, and cycles forward when line quality improves. |
| V.32 *bis* | Standard that extends V.32 to 7200, 12,000, and 14,400 bps; cycles to lower rate when line quality degrades; cycles forward when line quality improves. |
| V.32 *ter* | Pseudostandard that extends v.32 *bis* to 19,200 bps and 21,600 bps. |
| V.34 | Standard for 28,800-bps modems. (Some V.34 modems were enhanced with new software that provided them with the capability to achieve data rates of 31,200 bps or 33,600 bps.) |
| V.FAST | Proprietary, pseudostandard from Hayes and Rockwell for modems transmitting at data rates up to 28,800 bps; served as a migration path for V.34. |
| V.42 | Standard for error correction instead of for a modem. Uses LAPM as the primary error-correcting protocol, with MNP Classes 1 through 4 as an alternative (see Table 16.2). |
| V.42 *bis* | Standard that enhances V.42 by incorporating the British Telecom Lempel Ziv data compression technique to V.42 error correction. Most V.32, V.32 *bis*, and V.34-compliant modems come with V.42 or V.42 *bis* or MNP. |
| V.44 | A compression technology originally developed for the satellite industry to maximize available bandwidth. Approved by ITU as an industry standard in mid-2000, V.44 can generally yield download speed improvements of 20% to 60% when compared to V.42bis. In cases where data are highly compressible, a 200% improvement in download speeds can be expected. The actual amount of improvement is dynamic and depends on the data content. |
| V.90 | Standard for 57,600-bps modems (commonly called "56K modems") in which asymmetric data rates apply (i.e., the send and receive rates are different). Depending on telephone line conditions, upstream rates (send) are restricted to 33,600 bps, and downstream rates (receive) are restricted to 57,600 bps. V.90 modems are designed for connections that are digital at one end and have one digital-to-analog conversion. |
| V.92 | A new 56K modem standard approved by ITU in mid-2000. V.92 offers a maximum upstream rate of 48,000 bps, faster connection time, and call-waiting. The 48k upstream rate is an improvement over V.90's 33.6k, although achieving it might be as problematic as achieving a 56k downstream rate. The faster connection time feature will reduce the time it takes a modem to complete its handshaking process by one-half. The call-waiting feature will allow the modem to place an Internet connection on hold while you take another call on the same line. The total wait-time will be controlled by your ISP and can range from 0 to 16 minutes, or indefinitely. |



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# B3. Analog Data ⇔ Digital Signal

- The device used is called **codec** (**co**der/**dec**oder):

  – coder converts analog data into digital form for transmission.

  – decoder recovers the (original) analog data from the digital signal.

  See Figure 5.1(a) (p.130).

- In practice,
  See Figure 5.15 (p.153).

- Two main techniques used in codecs:

  **P**ulse **C**ode **M**odulation (**PCM**) and **D**elta **M**odulation (**DM**).

  We only discuss PCM here.

- PCM is based on the sampling theorem.

- **Sampling** is the process in which a (continuous) signal is sampled by measuring its amplitude at discrete instants (usually spaced at equal time intervals).

- Sampling theorem:

  It states that a band-limited signal which has no frequency components higher than $f_h$ Hz. can be recovered completely from a set of samples taken at the rate of $f_s$ ( $\geq 2 \times f_h$ ) samples per second.

  Or,

  Given:
  x(t) is a band-limited signal with bandwidth W,
  p(t) is a sampling signal consisting of pulses at intervals $T_s = 1 / f_s$, where $f_s$ is the sampling frequency,
  and, $x_s(t) = x(t)\,p(t)$ is the sampled signal.



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Then,
x(t) can be recovered exactly from $x_s(t)$ if and only if $f_s \geq 2 \times W$.

- PCM operational steps for encoding an analog signal:

  (1) Use a low-pass filter to obtain a bandlimited signal with a bandwidth of B.

  (2) Sample the signal at a frequency **at least twice the value of W** (or **twice the highest frequency** of interest in the signal). This process is called **P**ulse **A**mplitude **M**odulation (or **PAM)**.

  (3) Represent the sampled values of the amplitude (or pulses) by a finite set of levels (e.g. by taking an integer to the nearest whole number of continuous-amplitude values). This process is referred to as **quantization**.

  (4) Generate PCM signals by transforming each quantized amplitude to a binary code.

  See Figure 5.17 (p.154).

- An example of PCM encoding process:
  See Figure 5.16 (p.154).

- Bandwidth requirements of PCM signals:

  e.g. Consider a practical situation –

  Given :
  a voice signal with B = 4 kHz
  and,
  256 quantization levels.

  Then,
  the min. sampling rate = 8000 samples/sec.
  and,
  the data rate = $8 \times 8000 = 64$ kbps.

  Let $B_{PCM}$ be the required minimum bandwidth.

  From the Nyquist theorem,
  $$2 \times B_{PCM} = 64$$
  $$\therefore \quad B_{PCM} = 32 \text{ kHz}.$$

- Decoding of PCM signals:

  – On reception, the PCM encoding process is reversed to reproduce the analog signal.

  – The difference between the discrete value of the signal at a sampling instant and its nearest quantized value becomes the **PCM quantization error** (or **quantization noise**).



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  – With uniform quantization levels, the mean absolute error for each sampled amplitude value is the same regardless of signal level.

    ⇒ small-amplitude values are relatively more distorted.

    Solutions:
    (i) using nonlinear encoding:
    See Figure 5.18 (p.155).

    (ii) using companding functions:
    See Figure 5.19 (p.156).

## B4. Digital Data ⇔ Digital Signal

- Common encoding schemes are:
  (1) Non-return to zero (NRZ) Coding
  (2) Biphase Coding

### (1) Non-Return to Zero (NRZ) Coding

- Coding principle:
  using two voltage levels, each of which is constant during a bit interval.

- Two variations:
  (i) **NRZ-L** (**N**on**R**eturn-to-**Z**ero-**L**evel)
  (ii) **NRZI** (**N**on**R**eturn-to-**Z**ero-Level, **I**nvert on Ones)

  See Figure 5.2 (p.134).

- Advantages:
  a simple coding scheme;
  making efficient use of bandwidth.

- Problems:

  – the presence of a dc component in the encoded waveform.
    ⇒ a problem with transformer connection.

  – no transition during a bit interval.
    ⇒ the timing clock signal cannot be derived.

  – a long string of 1s (or 0s) for NRZ-L coding and a long string of 0s for NRZI coding would be difficult to detect in the presence of time drift at the receiver or transmitter.
    ⇒ lack of synchronization capability.

- Spectral density of various signal encoding schemes:
  See Figure 5.3 (p.136).



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**<u>(2) Biphase Coding</u>**

- Coding principles:
  (i) using two signal levels.
  (ii) having a signal transition in the middle of a bit time.

- Two variations:
  (i) **Manchester coding**
  (ii) **Differential Manchester coding**

  See Figure 5.2 (p.134).

- Biphase coding schemes are widely use in LAN applications at relatively high data rates.
  e.g. the IEEE 802.3 standard for baseband coaxial cable and twisted-pair CSMA/CD bus LANs using Manchester coding scheme;
  the IEEE 802.5 token ring LAN using Differential Manchester coding scheme.

- Because there is a predictable transition during each bit time, the receiver can resynchronize on that transition. (Biphase codes are also called **self-clocking codes**.)

- The absence of an expected transition can be used to detect errors.

- There is no dc component.

- The required effective bandwidth is wider than those for NRZ and other coding schemes.

- The required signal transition rate is higher than the data rate.

- Example:

  Calculate the data rate and the modulation rate for each of the two digital encoding schemes shown in the diagram below.
  See Figure 5.5 (p.139).

- Other signal encoding schemes:
  **Bipolar-AMI**: **Bipolar-A**lternate Mark **I**nversion
  **Pseudoternary**
  **B8ZS**: **B**ipolar with **8-Z**eros **S**ubstitution
  **HDB3**: **H**igh-**D**ensity **B**ipolar-**3** Zeros