Unit 3

Baseband Transmission
Syllabus

Baseband Transmission

• The digital signal used in baseband transmission occupies the entire bandwidth of the network media to transmit a single data signal.

• Baseband communication is bidirectional, allowing computers to both send and receive data using a single cable.
Baseband Modulation

- An information bearing-signal must conform to the limits of its channel
- Generally modulation is a two-step process
  - baseband: shaping the spectrum of input bits to fit in a limited spectrum
  - passband: modulating the baseband signal to the system rf carrier
- Most common baseband modulation is Pulse Amplitude Modulation (PAM)
  - data amplitude modulates a sequence of time translates of basic pulse
  - PAM is a linear form of modulation: easy to equalize, BW is pulse BW
  - Typically baseband data will modulate in-phase [cos] and quadrature [sine] data
- streams to the carrier passband
- Special cases of modulated PAM include
  - phase shift keying (PSK)
  - quadrature amplitude modulation (QAM)
Need for Baseband Modulation

• An analog signal has a finite bandwidth.
• · A digital stream or signal, with sharp transitions, has an infinite bandwidth.
• · Due to the limited available system bandwidth, only the major portion of
a digital signal spectrum can be transmitted and restored. Even if there is
• no loss or noise in the communication system, the received signal will
• have distortion due to the limited channel bandwidth.
• To avoid or to reduce this signal distortion, we
• use baseband modulation techniques
Line Codes

• In telecommunication, a **line code** (also called digital baseband modulation or digital baseband transmission method) is a **code** chosen for use within a communications system for baseband transmission purposes.

• **Line** coding is often used for digital data transport.

• Line coding consists of representing the digital signal to be transported by an amplitude- and time-discrete signal that is optimally tuned for the specific properties of the physical channel (and of the receiving equipment).

• The waveform pattern of voltage or current used to represent the 1s and 0s of a digital data on a transmission link is called **line encoding**.
Common types of Line Codes

- The common types of line encoding are
  - unipolar
  - polar
  - bipolar
  - Manchester encoding
Need for Line Codes

• Various Techniques
• Other Way: From Computers
• Information: Inherently discrete in nature
• Transmitted over band-limited channel: Signal gets Dispersed
• Causes: Overlap and Distortion
• Distortion: Intersymbol Interference (ISI)
Properties of Line Codes

- Transmission Bandwidth: as small as possible
- Power Efficiency: As small as possible for given BW and probability of error
- Error Detection and Correction capability: Ex: Bipolar
- Favorable power spectral density: dc=0
- Adequate timing content: Extract timing from pulses
- Transparency: Prevent long strings of 0s or 1s
Unipolar coding

• **Unipolar encoding** is a line code. A positive voltage represents a binary 1, and zero volts indicates a binary 0. It is the simplest line code, directly encoding the bitstream, and is analogous to on-off keying in modulation.

• Its drawbacks are that it is not self-clocking and it has a significant DC component, which can be halved by using return-to-zero, where the signal returns to zero in the middle of the bit period.

• With a 50% duty cycle each rectangular pulse is only at a positive voltage for half of the bit period.

• This is ideal if one symbol is sent much more often than the other and power considerations are necessary, and also makes the signal self-clocking.
• **NRZ(Non-Return-to-Zero)** - Traditionally, a unipolar scheme was designed as a non-return-to-zero (NRZ) scheme, in which the positive voltage defines bit 1 and the negative voltage defines bit 0.

• It is called NRZ because the signal does not return to zero at the middle of the bit.

• Compared with its polar counterpart, Uni Polar NRZ, this scheme is very expensive.

• The normalized power (power required to send 1 bit per unit line resistance) is double that for polar NRZ.

• For this reason, this scheme is not normally used in data communications today.
Unipolar Non-Return to Zero (NRZ):

In unipolar NRZ the duration of the MARK pulse ($T_m$) is equal to the duration ($T_o$) of the symbol slot.
Power Spectral Density of Line Codes

- In general, the PSD of a line code is
  \[ S_y(f) = |P(f)|^2 S_x(f) \]

  where
  - \( S_x(f) \) is the power spectral density of the digital sequence \( \{a_k\} \)
    (The PSD of a digital sequence is periodic in frequency.)
  - \( P(f) \) is the PSD of the basic pulse

- Polar signaling.
  \[ S_y(f) = \frac{|P(f)|^2}{T_b} R_0 = \frac{|P(f)|^2}{T_b} \]

- AMI (bipolar) signaling for full-width pulses.
  \[ S_y(f) \frac{|P(f)|^2}{2T_b} (1 - \cos 2\pi T_b f) = \frac{|P(f)|^2}{T_b} \sin^2(\pi T_b f) \]

  For half-width pulses: AMI (bipolar) signaling for full-width pulses.
  \[ S_y(f) = \frac{|P(f)|^2 T_b}{4} \text{sinc}^2 \left( \frac{\pi f T_n b}{2} \right) \sin^2(\pi T_b f) \]
Return-to-zero

- **Return-to-zero** (RZ) describes a line code used in telecommunications signals in which the signal drops (returns) to zero between each pulse.
- This takes place even if a number of consecutive 0s or 1s occur in the signal.
- The signal is self-clocking. This means that a separate clock does not need to be sent alongside the signal, but suffers from using twice the bandwidth to achieve the same data-rate as compared to non-return-to-zero format.
• The waveforms for the line code may be further classified according to the rule that is used to assign voltage levels to represent the binary data. Some examples include:
  
  - **Unipolar Signalling:** In positive–logic unipolar signalling, the binary 1 is represented by a high level (+A volts) and a binary 0 by a zero level. This type of signalling is also called on–off keying (OOK).

  - **Polar Signalling:** Binary 1’s and 0’s are represented by equal positive and negative levels

  - **Bipolar (Pseudo ternary) Signaling:** Binary 1’s are represented by alternating positive or negative values. The binary 0 is represented by a zero level. The term *pseudo ternary* refers to the use of 3 encoded signal levels to represent two–level (binary) data. This is also called **alternate mark inversion (AMI)** signaling.

  - **Manchester Signaling:** Each binary 1 is represented by a positive half–bit period pulse followed by a negative half–bit period pulse. Similarly, a binary 0 is represented by a negative half–bit period pulse followed by a positive half–bit period pulse. This type of signalling is also called split–phase encoding.
Polar Return to Zero (RZ):

Advantages:
- Simplicity in implementation.
- No DC component.

Disadvantages:
- Continuous part is non-zero at 0 Hz. Causes “Signal Droop”.
- Does not have any error correction capability.
- Does not possess any clocking component for easy synchronisation. However, clock can be extracted by rectifying the received signal.
- Occupies twice as much bandwidth as Polar NRZ.
BiPolar Signalling

Bipolar Signalling is also called “alternate mark inversion” (AMI) uses three voltage levels (+V, 0, -V) to represent two binary symbols. Zeros, as in unipolar, are represented by the absence of a pulse and ones (or marks) are represented by alternating voltage levels of +V and −V.

Alternating the mark level voltage ensures that the bipolar spectrum has a null at DC and that signal droop on AC coupled lines is avoided.

The alternating mark voltage also gives bipolar signalling a single error detection capability.

Like the Unipolar and Polar cases, Bipolar also has NRZ and RZ variations.
In Manchester encoding, the duration of the bit is divided into two halves. The voltage remains at one level during the first half and moves to the other level during the second half.

A 'One' is +ve in 1st half and -ve in 2nd half.

A 'Zero' is -ve in 1st half and +ve in 2nd half.
Inter Symbol Interference (ISI)

- In telecommunication, **inter symbol interference (ISI)** is a form of distortion of a signal in which one symbol interferes with subsequent symbols.
- This is an unwanted phenomenon as the previous symbols have similar effect as noise, thus making the communication less reliable.
- ISI is usually caused by multipath propagation or the inherent non-linear frequency response of a channel causing successive symbols to "blur" together.
Nyquist criterion for distortionless transmission

- To design $h_T(t)$ and $h_d(t)$ under the following two conditions:

  (a). There is no ISI at the sampling instants (Nyquist criterion).
  (b). A controlled amount of ISI is allowed (correlative coding).
Design of Bandlimited Signals for Zero ISI - Nyquist criterion

• Recall the output of the receiving filter, sampled at $t = kT$, is given by

$$y(kT) = \alpha b_k + \sum_{n\neq k} b_n p(kT - nT) + n_o(kT)$$

• Thus, in time domain, a sufficient condition for $\mu p(t)$ such that it is ISI free is

$$p(nT) = \begin{cases} 1 & n = 1 \\ 0 & n \neq 0 \end{cases}$$  \hspace{1cm} (1)
• Theorem: (Nyquist) A necessary and sufficient condition for \( p(t) \) to satisfy (1) is that the Fourier transform \( P(f) \) satisfies

\[
\sum_{n} P(f - \frac{n}{T}) = T
\]  

(2)

• This is known as the Nyquist pulse-shaping criterion or Nyquist condition for zero ISI.
**Proof.** When we sample $p(t)$ at $t = kT$, $k = 0, \pm 1, \pm 2, \cdots$, we have the following pulses

$$ p_\delta(t) = p(t) \sum_k \delta(t - kT) $$

$$ = \sum_k p(kT) \delta(t - kT) $$

The Fourier transform of $p_\delta(t)$ is given by

$$ P_\delta(f) = \mathcal{F}(p_\delta(t)) $$

$$ = \mathcal{F}\left( \sum_k p(kT) \delta(t - kT) \right) $$

$$ = \sum_k p(kT) \exp(-j2\pi f kT) $$

( $p(kT)$ is constant for $t$.)

$$ = 1 \quad \text{(from (1))} \quad (4) $$

From (3) and (4), ISI free $\Leftrightarrow$

$$ \frac{1}{T} \sum_k P\left(f - \frac{k}{T}\right) = 1 $$

which gives the result in (2).
Investigate possible pulses which satisfy the Nyquist criterion.

Suppose that the channel has a bandwidth of $W$, then

$$H_c(f) = 0 \text{ for } |f| > W$$

Since

$$P(f) = H_T(f)H_c(f)H_d(f)$$

we have

$$P(f) = 0 \text{ for } |f| > W$$

We write

$$Z(f) = \sum_n P(f - n/T)$$

and distinguish the following three cases:
Fig. 4.1 $Z(f)$ for the case $T < 1/(2W)$

Fig. 4.2 $Z(f)$ for the case $T = 1/(2W)$

$Z(f)$ for the case $T > 1/(2W)$
1. \( T < \frac{1}{2W} \), or \( \frac{1}{T} > 2W \) (i.e., bit rate > 2\( W \), impossible!) No choices for \( P(f) \) such that \( Z(f) = 0 \).

2. \( T = \frac{1}{2W} \), i.e., \( W = \frac{1}{2T} \) (the Nyquist rate)

In this case, if we choose

\[
P(f) = \begin{cases} 
T & |f| \leq W \\
0 & \text{otherwise}
\end{cases}
\]

i.e.,

\[
P(f) = T \cdot \text{rect}\left(\frac{f}{2W}\right)
\]

which results in

\[
p(t) = \sin c\left(\frac{t}{T}\right)
\]

This means that the smallest value of \( T \) for which the transmission with zero ISI is possible is

\[
T = \frac{1}{2W} \quad (R = \frac{1}{T} = 2W, \text{ bit rate })
\]

This is called the ideal Nyquist channel.
In other words,

**Ideal Nyquist channel:**

\[ R = 2B_o = \frac{1}{T} \]

\[ W = B_o \quad (R = R_b, \ T = T_b) \]
Disadvantages:
(a) an ideal LPF is not physically realizable.
(b) Note that

\[ p(t) = \sin c \left( \frac{t}{T} \right) \propto \frac{1}{|t|} \]

Thus, the rate of convergence to zero is low since the tails of \( p(t) \) decay as \( 1/|t| \).

• Hence, a small mistiming error in sampling the output of the matched filter at the demodulator results in an infinite series of ISI components.
3. For \( T > \frac{1}{2W} \), i.e., \( \frac{1}{T} < 2W \), in this case, there exists numerous choices for \( P(f) \) such that \( Z(f) = T \). The important one is so called the raised cosine spectrum.

The raised cosine frequency characteristic is given by

\[
P(f) = \begin{cases} 
\frac{1}{2B_0} 
& 0 \leq |f| < (1-\alpha)B_0 \\
\frac{1}{4B_0} \left[ 1 + \cos \left( \frac{\pi |f| - (1-\alpha)B_0}{2\alpha B_0} \right) \right] 
& (1-\alpha)B_0 \leq |f| < (1+\alpha)B_0 \\
0 
& |f| \geq (1+\alpha)B_0 
\end{cases}
\]

where \( \alpha \in [0,1] \) is called the rolloff factor and \( B_0 = \frac{R}{2} \).

\( B_0 = \frac{1}{2T} \).
\( Z(f) = T \) by the following sum of three terms at any interval of length \( 2B_0 \):

\[
P(f) + P(f - 2B_0) + P(f + 2B_0) = T \quad -B_0 \leq f \leq B_0
\]

\[
P(f) + P(f - 2B_0) + P(f + 2B_0) = T \quad B_0 \leq f \leq 3B_0
\]
The time response $p(t)$, the inverse Fourier transform of $P(f)$, is given by

$$p(t) = \text{sinc}2B_0 t \frac{\cos2\pi\alpha B_0 t}{1-16\alpha^2 B_0^2 t^2}$$

(5)

This function has much better convergence property than the ideal Nyquist channel. The first factor in (5) is associated with the ideal filter, and the second factor that decreases as $1/|t|^2$ for large $|t|$. Thus

$$p(t) \propto \frac{1}{|t^3|}$$
Responses for different rolloff factors. (a) Frequency response. (b) Time response. Note that $B_o = 1/2T_o$. 
Correlative Coding and Equalization

• Correlative Coding
  ❖ For zero ISI, the symbol rate $R = 1/T < 2W$, the Nyquist rate.
  ❖ We may relax the condition of zero ISI in order to achieve $R = 2W$.

• The schemes which allow a controlled amount of ISI to achieve the symbol rate $2W$ are called correlative coding or partial response signaling schemes.
The condition for zero ISI is

\[ p_1(nT) = \begin{cases} 
1, & n = 0 \\
0, & n \neq 0 
\end{cases} \]  

(1)

Suppose that we design the band-limited signal \( p(t) \) to have controlled ISI at one time instant, i.e., to allow one additional nonzero value in the samples \( \{ p(nT) \} \) for example,

\[ p_2(nT) = \begin{cases} 
1, & n = 0 \text{ and } n = 1 \\
0, & \text{otherwise} 
\end{cases} \]  

(2)
$p_2(t)$ has a larger time duration than $p_1(t)$; 
$\Rightarrow P_2(f) = F[p_2(t)]$ has a smaller bandwidth on frequency domain than $P_1(f) = F[p_1(t)]$; 
$\Rightarrow$ Spectral efficiency is increased by using $p_2(t)$.

**Note.** The ISI we introduce by using $p_2(t)$ is deterministic or “controlled” and, hence, its effect on signal detection at the receiver can be removed, as discussed below.
2nd Nyquist Criterion

• Values at the pulse edge are distortionless
• \( p(t) = 0.5 \), when \( t = -T/2 \) or \( T/2 \); \( p(t) = 0 \), when \( t = (2k-1)T/2 \), \( k \neq 0, 1 \) where \( -1/T \leq f \leq 1/T \)

\[
P_r(f) = \text{Re} \left[ \sum_{n=-\infty}^{\infty} (-1)^n P(f + n/T) \right] = T \cos(fT/2)
\]

\[
P_i(f) = \text{Im} \left[ \sum_{n=-\infty}^{\infty} (-1)^n P(f + n/T) \right] = 0
\]
Nyquist criteria: (a) input signal, (b) output signal satisfying Nyquist's first criterion—values at pulse centers unchanged, (c) output signal satisfying Nyquist's second criterion—values at pulse edges unchanged, (d) output signal satisfying Nyquist's first and second criteria—values at pulse centers and edges unchanged.
3rd Nyquist Criterion

- Within each symbol period, the integration of signal (area) is proportional to the integration of the transmit signal (area)

\[
P(w) = \begin{cases} 
\frac{(wt)/2}{\sin(wT/2)}, & |w| \leq \frac{\pi}{T} \\
0, & |w| > \frac{\pi}{T}
\end{cases}
\]

\[
p(t) = \frac{1}{2\pi} \int_{-\pi/T}^{\pi/T} \frac{(wt/2)}{\sin(wT/2)} e^{j\omega t} \, d\omega
\]

\[
A = \int_{\frac{2n-1}{2}T}^{\frac{2n+1}{2}T} p(t)dt = \begin{cases} 
1, & n = 0 \\
0, & n \neq 0
\end{cases}
\]
Eye Diagram

• Eye diagram is a means of evaluating the quality of a received “digital waveform”
  – By quality is meant the ability to correctly recover symbols and timing
  – The received signal could be examined at the input to a digital receiver or at some stage within the receiver before the decision stage
• Eye diagrams reveal the impact of ISI and noise
• Two major issues are 1) sample value variation, and 2) jitter and sensitivity of sampling instant
• Eye diagram reveals issues of both
• Eye diagram can also give an estimate of achievable BER
• Check eye diagrams at the end of class for participation
**Interpretation of Eye Diagram**

Slope indicates sensitivity to timing error, smaller is better.

Signal excursion or wasted power

Amount of distortion at sampling instant, relates to signal SNR

Amount of noise that can be tolerated by the signal, the larger the better.

Best time to sample

Opening of the eye, Time over which we can successfully sample the waveform
Raised Cosine Eye Diagram

- The larger \( \alpha \), the wider the opening.
- The larger \( \alpha \), the larger bandwidth \( (1 + \alpha)/T_b \)
- But smaller \( \alpha \) will lead to larger errors if not sampled at the best sampling time which occurs at the center of the eye.
Cosine rolloff filter: Eye pattern

2nd Nyquist
1st Nyquist: ✓
2nd Nyquist: 

1st Nyquist: ✓
2nd Nyquist: ✗

2nd Nyquist: 
1st Nyquist: 

1st Nyquist: ✓
2nd Nyquist: ✗
Eye Diagram Setup

- Eye diagram is a retrace display of data waveform
  - Data waveform is applied to input channel
  - Scope is triggered by data clock
  - Horizontal span is set to cover 2-3 symbol intervals
- Measurement of eye opening is performed to estimate BER
  - BER is reduced because of additive interference and noise
  - Sampling also impacted by jitter

Data waveform can be RZ or NRZ, and can be binary or multilevel
Trigger waveform can be any signal synchronous with the data waveform, including the data waveform itself
Partial Response Signals

- Previous classes: $S_y(w) = |P(w)|^2 S_x(w)$
  - Control signal generation methods to reduce $S_x(w)$
  - Raise Cosine function for better $|P(w)|^2$

- This class: improve the bandwidth efficiency
  - Widen the pulse, the smaller the bandwidth.
  - But there is ISI. For binary case with two symbols, there is only few possible interference patterns.
  - By adding ISI in a controlled manner, it is possible to achieve a signaling rate equal to the Nyquist rate (2W symbols/sec) in a channel of bandwidth W Hertz.
Thank you