## Communication Systems

**Analog communication**
- Transmit and receive analog waveforms
- Amplitude Modulation (AM)
- Phase Modulation (PM)
- Frequency Modulation (FM)
- Quadrature Amplitude Modulation (QAM)
- Pulse Amplitude Modulation (PAM)

**Digital communication**
- Treat transmission and reception as digitized
- Transmission and reception with analog waveforms
- Amplitude Shift Keying (ASK)
- Phase Shift Keying (PSK)
- Frequency Shift Keying (FSK)
- QAM
- PAM

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## Communication Systems

**Basic structure**

![Basic structure diagram](image)

**Information sources**
- Message signal $m(t)$ is the information source to be sent
- Possible information sources include voice, music, images, video, and data, which are baseband signals
- Baseband signals have power concentrated near DC
Transmitter

- Signal processing conditions the message signal
  - Lowpass filtering to make sure that the message signal occupies a specific bandwidth, e.g., in AM and FM radio, each station is assigned a slot in the frequency domain.
  - In a digital communications system, we might add redundancy to the input bit stream $m[n]$
- Carrier circuits
  - Convert baseband signal into a frequency band appropriate for the channel
  - Uses analog and/or digital modulation

Channel

- Transmission media:
  - Wireline (twisted pair, coaxial, fiber optics)
  - Wireless (indoors/air, outdoor/air, underwater, space)
- Propagating signals experience a gradual degradation over distance
- Boosting improves signal and reduces noise, e.g., repeaters
**Wireline Channel Impairments**

- **Attenuation:** linear distortion that is dependent on the frequency response of the channel.
- **Spreading:** the finite extent of each transmitted pulse increases, i.e. pulse widens
  - Transmit pulse length $T_s$
  - Channel impulse response length $T_h$
  - Resulting waveform due to convolution has duration $T_s + T_h$
- **Phase jitter:** the same sinusoid experiences different phase shifts in the channel
- **Additive noise:** arises from many sources in the transmitter, channel, and receiver

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**Wireless Channel Impairments**

- Same as wireline channel impairments plus others
- **Fading:** multiplicative noise
  - Example: talking on a cellular phone while driving a car when the reception fades in and out
- **Multiple propagation paths**
  - Multiple ways for transmitted signal to arrive at receiver
Receiver and Information Sinks

- **Receiver**
  - Carrier circuits undo effects of carrier circuits in transmitter, e.g. demodulate from a bandpass signal to a baseband signal
  - Signal processing subsystem extracts and enhances the baseband signal
- **Information sinks**
  - Output devices such as computer screens, speakers, and TV screens

Hybrid Communication Systems

- **Mixed analog and digital signal processing in the transmitter and receiver**
  - Ex: message signal is digital but broadcast over an analog channel (compressed speech in digital cell phones)
- **Signal processing in the transmitter**
  - **A/D Converter** → **Error Correcting Codes** → **Digital Signalling** → **D/A Converter**
- **Signal processing in the receiver**
  - **A/D** → **Equalizer** → **Detection** → **Decoder** → **Waveform Generator** → **D/A**
Power Spectra

- Deterministic signal \( x(t) \) w/ Fourier transform \( X(f) \)
  - Power spectrum is square of absolute value of magnitude response (phase is ignored)
    \[ P_s(f) = |X(f)|^2 = X(f) X^*(f) \]
  - Multiplication in Fourier domain is convolution in time domain
  - Conjugation in Fourier domain is reversal and conjugation in time
    \[ X(f) X^*(f) = F\{ x(t) * x^*(\tau) \} \]

- Autocorrelation of \( x(t) \)
  - Maximum value at \( R_x(0) \)
  - \( R_x(\tau) \) is even symmetric, i.e. \( R_x(\tau) = R_x(-\tau) \)

\[ R_x(\tau) = E\{ n(t) n^*(t+\tau) \} = \int_{-\infty}^{\infty} n(t) n^*(t+\tau) \, dt \]

- Estimating noise power spectrum in Matlab
  \[ P_n(f) = \sigma^2 \, \delta(\tau) \]

\[ N = 16384; \quad \text{gaussianNoise} = \text{randn}(N,1); \quad \text{plot( abs(ft(gaussianNoise)), .^2 );} \]
Transmit One Bit
- Transmission over communication channel (e.g. telephone line) is analog
- Here is one way to do it
- 2-level digital PAM

Transmit Two Bits (Interference)
- Transmitting two bits (pulses) back-to-back will cause overlap (interference) at the receiver
- Sample $y(t)$ at $T_b, 2T_b, \ldots$, and threshold with threshold of zero
- How do we prevent intersymbol interference (ISI) at the receiver?
Transmit Two Bits (No Interference)

- Prevent intersymbol interference by waiting $T_h$ seconds between pulses (called a guard period)

![Diagram showing pulse transmission and convolution]

- Disadvantages?

Matched Filter

- Detection of a pulse in presence of additive noise, ignoring channel memory
- Receiver knows what pulse shape it is looking for

\[ y(t) = p(t) * h(t) + w(t) * h(t) \]

\[ y(t) = g_0(t) + n(t) \]
Matched Filter Derivation

- Design of matched filter
  - Maximize signal power, i.e. power of \( g_0(t) \) at \( t = T \)
  - Minimize noise, i.e. power of \( n(t) \)
- Combine design criteria

\[
\eta = \frac{\left| g_0(T) \right|^2}{E[n^2(t)\/\text{average power}}
\]

Matched Filter

- Given transmitter pulse shape \( p(t) \) of duration \( T \), matched filter is given by \( h_{opt}(t) = k^* p^*(T-t) \) for all \( k \)
  - Duration and shape of impulse response of the optimal filter is determined by pulse shape \( p(t) \)
  - \( h_{opt}(t) \) is scaled, time-reversed, and shifted version of \( p(t) \)
- Optimal filter maximizes peak pulse SNR

\[
\eta_{max} = \frac{1}{\sigma^2} \int_{-\infty}^{\infty} |P(f)|^2 \, df = \frac{1}{\sigma^2} \int_{-\infty}^{\infty} |p(t)|^2 \, dt = \frac{E_b}{\sigma^2} = \text{SNR}
\]

- Does not depend on pulse shape \( p(t) \)
- Proportional to signal energy \( E_b \)
- Inversely proportional to power spectral density of noise
**Digital 2-level PAM System**

- Transmitted signal: $s(t) = \sum_{k} a_k p(t - k T_b)$
- Requires synchronization of clocks between transmitter and receiver

**Eliminating ISI in PAM**

- One choice for $P(f)$ is a rectangular pulse
- $P(f) = \begin{cases} 
\frac{1}{2W}, & -W < f < W \\
0, & |f| > W 
\end{cases}$
- Inverse Fourier transform of a rectangular pulse is a sinc function
- $p(t) = \text{sinc} \left( \frac{2 \pi W t}{2} \right)$
- This is called the Ideal Nyquist Channel
- It is not realizable because the pulse shape is not causal and is infinite in duration
Eliminating ISI in PAM

- Another choice for $P(f)$ is a raised cosine spectrum

\[
P(f) = \begin{cases} 
\frac{1}{4W} \left( 1 - \sin \left( \frac{\pi |f|}{2W} \right) \right) & 0 \leq |f| < f_t \\
\frac{1}{2W} & f_t \leq |f| < 2W - f_t \\
0 & 2W - f_t \leq |f| \leq 2W 
\end{cases}
\]

- Roll-off factor gives bandwidth in excess of bandwidth $W$ for ideal Nyquist channel:

\[
\alpha = 1 - \frac{f_t}{W}
\]

- Raised cosine pulse has zero ISI when sampled correctly

\[
p(t) = \text{sinc}(2\pi W t) \frac{\cos(2\pi \alpha W t)}{1 - 16\alpha^2 W^2 t^2}
\]

- Let $g(t)$ and $c(t)$ be square root of raised cosine

Symbol Clock Recovery

- Critical to sample at correct time instances to have maximum signal power and minimum ISI
- Transmitter and receiver normally have different crystal oscillators
- Digital receiver should try its best to synchronize with transmitter clock
- Receiver must extract the clock information from the received signal and then adjust its A/D timing
Symbol Clock Recovery

- The receive filter output $x(t)$ is first passed through a prefilter with frequency response $B(\omega)$.

- This is a bandpass filter centered at $\omega_s/2$, half the symbol frequency.

\[ B(\omega) \]

\[ -\omega_s/2 \quad 0 \quad \omega_s/2 \]

Symbol Clock Recovery

- Let the combined baseband shaping filter and prefilter frequency and impulse responses be

\[ G_1(\omega) = G(\omega)B(\omega) \quad \text{and} \quad g_1(t) = g(t) * b(t) \]

- The prefilter output

\[ q(t) = s'(t) \otimes g_1(t) = \sum_{k=-\infty}^{\infty} a_k g_1(t-kT) \quad p(t) = q^2(t) = \sum_{k=-\infty}^{\infty} \sum_{m=-\infty}^{\infty} a_k a_m g_1(t-kT) g_1(t-mT) \quad E\{ p(t) \} = \sum_{k=-\infty}^{\infty} \sum_{m=-\infty}^{\infty} E\{ a_k a_m \} - a^2 \delta(k-m) g_1(t-kT) g_1(t-mT) \]

\[ = a^2 \sum_{k=-\infty}^{\infty} g_1^2(t-kT) \]
Symbol Clock Recovery

- $g_1(t)$ is impulse response of composite channel of pulse shaper, channel, and receive lowpass filter
- Received signal $q(t)$

$$q(t) = s^*(t) \otimes g_1(t) = \sum_{k=-\infty}^{\infty} a_k g_1(t-kT)$$

$$p(t) = q^2(t) = \sum_{k=-\infty}^{\infty} \sum_{m=-\infty}^{\infty} a_k a_m g_1(t-kT) g_1(t-mT)$$

$$E\{p(t)\} = \sum_{k=-\infty}^{\infty} \sum_{m=-\infty}^{\infty} E\{a_k a_m\} - a^2 \delta(k-m) g_1(t-kT) g_1(t-mT)$$

$$= a^2 \sum_{k=-\infty}^{\infty} g_1^2(t-kT)$$

Symbol Clock Recovery

- $E\{p(t)\}$ is a periodic function

$$E\{p(t)\} = \sum_{k=-\infty}^{\infty} p_k e^{j k \omega_s t} \quad \text{where} \quad p_k = \frac{1}{T} \int_{0}^{T} E\{p(t)\} e^{-j k \omega_s t} dt$$

- Lowpass filter $E\{p(t)\}$ to pass fundamental frequency $\omega_s$ and filter out harmonics

$$H(\omega)$$

\[ -\omega_s \quad 0 \quad \omega_s \]
Symbol Clock Recovery

- $z(t)$ looks like a sinusoid at the clock frequency with slowly varying amplitude and phase. Its zero crossings cluster together.
- Extract sine wave that with same frequency symbol rate
- Use as reference signal to digital phase locked loop or reshape it as a sampling clock