Communications Engineering refers to all engineering aspects of designing and analyzing systems for communicating information electronically.

"Signals and systems", or signal processing, plays many important roles in communications engineering.

EX: radio station 1 wants to transmit signal $x_1(t)$ and radio station 2 wants to transmit signal $x_2(t)$.

$X_1(\omega)$ $\rightarrow$ $\omega$ $X_2(\omega)$ $\rightarrow$ $\omega$

→ How can both transmit to the public at the same time?

Answer: The signals $x_1(t)$ and $x_2(t)$ are frequency shifted by different amounts for transmission:

$X_1$ $X_2$

→ Listeners can now receive the signal they want, but they must undo the frequency shift before they can listen to the information.
- The frequency shift is accomplished by embedding the information signals $x_1(t)$ and $x_2(t)$ into other signals.

- The act of combining two signals like this is called **modulation**.

- The act of extracting the information signal at the receiver is called **demodulation**.

- Information bearing electronic signals are transmitted through a medium.

  **Examples:**
  - Atmosphere (broadcast radio and TV)
  - Coaxial cable (cable TV)
  - Twisted pair copper wire (plain old telephone system – POTS)
  - Fiber optic (digital telephone, internet trunk)
  - Satellite Satellite Link (telephone, TV).

- The transmission medium is called the **channel**.

- Generally, the channel is not an all-pass filter.
  - It distorts the signal.
  - Prior to transmission, the signal is often passed through the inverse system of the channel.
  - Then the desired signal is received.
  - This is called **channel equalization**.
- We will focus on modulation, which is probably the most important signal processing in communications engineering.

Amplitude Modulation (AM)

- $x(t)$ is the information signal, also called the modulating signal.

- $x(t)$ is embedded in another signal $c(t)$, called the carrier signal.

- $c(t)$ is a complex exponential or a real sinusoid.

- The modulated signal $y(t)$ is

  $$y(t) = x(t)c(t).$$

  $\rightarrow$ The frequency shifting property of the Fourier transform tells you that $y(t)$ is a frequency shifted version of $x(t)$.

  $\rightarrow$ This is one way of accomplishing the frequency shifting discussed on page 8.1.

  $\rightarrow$ This way of doing it is called amplitude modulation, or AM.
- For a complex carrier, \( c(t) = e^{j(\omega_c t + \theta_c)} \)
- For a real carrier, \( c(t) = \cos(\omega_c t + \theta_c) \)

\( \omega_c \) is called the **carrier frequency**.

- For simplicity, suppose the carrier phase \( \theta_c \) is zero.
- Then \( Y(t) = x(t)e^{j\omega_c t} \) for a complex carrier.
- Then \( Y(\omega) = \frac{1}{2\pi} X(\omega) * C(\omega) \).

\( \Rightarrow \) but \( C(\omega) = 2\pi \delta(\omega - \omega_c) \).

\( \Rightarrow \) so \( Y(\omega) = X(\omega - \omega_c) \) (frequency shift)
- Amplitude modulation shifts the frequency band of the information signal \( X(t) \).

- The information signal \( X(t) \) is also called the baseband signal.

- To recover \( X(t) \) from \( Y(t) \) at the receiver, we simply multiply by \( e^{-j\omega_c t} \), the conjugate of the carrier signal:

\[
X(t) = Y(t) e^{-j\omega_c t}
\]

- This is called demodulation.

- Block diagram of complex carrier AM transmitter:

- For a real carrier, the block diagram becomes:
- With a real carrier \( c(t) = \cos(\omega_c t + \theta_c) \), we have

\[
c(\omega) = \pi \left[ \delta(\omega - \omega_c) + \delta(\omega + \omega_c) \right].
\]

- So,

\[
Y(\omega) = \frac{1}{2} \left[ X(\omega - \omega_c) + X(\omega + \omega_c) \right] \quad (\theta_c = 0)
\]

---

For demodulation to work, the two copies of \( X(\omega) \) cannot overlap.

This means we must have \( \omega_c > \omega_M \), which is almost always the case.

**Ex:** AM radio: \( \omega_M \sim \text{kHz} \)

\( \omega_c \sim \text{MHz} \)
Synchronous AM Demodulation

- The received signal is \( y(t) = x(t) \cos \omega_c t \)
  (assuming a lossless channel).
- The first thing the receiver does is multiply \( y(t) \) by the carrier signal to get
  \[ w(t) = y(t) \cos \omega_c t = y(t) c(t) \]

\( Y(w) \) contains two frequency shifted copies of \( X(w) \).
- Each of these copies is frequency shifted a second time in \( W(w) \). Two of them add together to make a new copy of \( X(w) \).
- As shown below, \( X(w) \) and thus \( x(t) \), can be recovered by applying an ideal low-pass filter (dashed line):
The filter must have a gain of 2.

Another way to see this is to realize that

\[ w(t) = y(t) \cos \omega_c t = x(t) \cos^2 \omega_c t \]

→ use identity \( \cos^2 \omega_c t = \frac{1}{2} + \frac{1}{2} \cos 2\omega_c t \)

\[ \implies w(t) = \frac{1}{2} x(t) + \frac{1}{2} x(t) \cos 2\omega_c t \]

This term is removed by the low-pass filter.

Recall that no low-pass filtering is required when a complex carrier is used, since we can simply multiply \( y(t) \) by the conjugate of the carrier.

Block diagram for complex carrier:

- Modulation
- Demodulation
- In the preceding block diagrams, note that the transmitter and receiver must both know the carrier signal \( c(t) = e^{j\omega_c t} \) or \( c(t) = \cos(\omega_c t) \).

- In particular, the demodulation does not work correctly if there is a phase offset between the carrier signals used in the transmitter and receiver.

- Since the receiver must know the phase of the carrier signal used at the transmitter, this type of demodulation is called synchronous demodulation.

- Suppose now that there is a phase error in the carrier signal used at the receiver (as compared to the one used at the transmitter).
This type of error is called a synchronization error.

Specifically, for the case of a complex carrier, suppose that

\[ c_T(t) = e^{j(w_c t + \theta_c)} \] at the transmitter
\[ c_R(t) = e^{-j(w_c t + \phi_c)} \] at the receiver.

We have

\[ w(t) = c_R(t) y(t) \]
\[ = c_R(t) [c_T(t) x(t)] \]
\[ = e^{j(w_c t + \phi_c)} e^{j(w_c t + \theta_c)} x(t) \]
\[ = e^{j(\theta_c - \phi_c)} x(t). \]

The demodulated signal is off by a multiplicative complex factor.

For the case of the real-valued carrier, suppose that

\[ c_T(t) = \cos(w_c t + \theta_c) \]
\[ c_R(t) = \cos(w_c t + \phi_c). \]
- The demodulated signal is

\[ w(t) = x(t) \cos(w_c t + \Theta_c) \cos(w_c t + \phi_c) \]

- Apply the trig identity

\[
\cos(w_c t + \Theta_c) \cos(w_c t + \phi_c) = \frac{1}{2} \cos(\Theta_c - \phi_c) + \frac{1}{2} \cos(2w_c t + \Theta_c + \phi_c)
\]

- The demodulated signal is

\[
w(t) = \frac{1}{2} \cos(\Theta_c - \phi_c) x(t) + \frac{1}{2} \cos(2w_c t + \Theta_c + \phi_c)
\]

Discarded by low-pass filter.

- If \( \Theta_c = \phi_c \), the output of the low-pass filter is \( x(t) \).

- If \( \Theta_c - \phi_c = \frac{\pi}{2} \), the output of the low-pass filter is zero.
Asynchronous AM Demodulation

- This approach does not require synchronization between signals at the transmitter and receiver.
- Suppose that $X(w)$ is band limited to $-\omega_m < w < \omega_m$, so that $X(w) = 0$ for $|w| > \omega_m$.
- Suppose further that the carrier frequency $\omega_c \gg \omega_m$.
- Finally, suppose that $X(t) > 0$ for all $t$.
- In this case, $X(t)$ is approximately equal to the envelope of $y(t)$.

- A demodulator that tracks the envelope of $y(t)$ is called an envelope detector.
Example of a simple circuit for envelope detection (half-wave rectifier).

![Circuit Diagram](image)

**Figure 8.11** Demodulation by envelope detection: (a) circuit for envelope detection using half-wave rectification; (b) waveforms associated with the envelope detector in (a): $r(t)$ is the half-wave rectified signal, $x(t)$ is the true envelope, and $w(t)$ is the envelope obtained from the circuit in (a). The relationship between $x(t)$ and $w(t)$ has been exaggerated in (b) for purposes of illustration. In a practical asynchronous demodulation system, $w(t)$ would typically be a much closer approximation to $x(t)$ than depicted here.

Usually, the output of a demodulation circuit of this type is then processed with a low-pass filter to smooth out the differences between $w(t)$ and $x(t)$. 

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The two main requirements for asynchronous demodulation to work are:

1. \( x(t) \geq 0 \)

2. \( \omega_c \gg \omega_m \)

The second condition is often satisfied automatically.

For example, in AM radio \( \frac{\omega_m}{2\pi} \approx 20 \text{ kHz} \),
while \( \frac{\omega_c}{2\pi} \approx 0.5 \text{ MHz} \) to \( 2 \text{ MHz} \).

The first condition can always be satisfied by adding a constant \( A \) to \( x(t) \). If \( A \) is picked large enough, then \( x(t) + A > 0 \).

- Let \( K = \text{maximum amplitude of } x(t) \), so that \( |x(t)| \leq K \).
- For \( x(t) + A \) to be positive, we need \( A > K \).

The ratio \( M = \frac{K}{A} \) is called the "modulation index."

If \( M \) is expressed in percent, it is called the "percent modulation."
Examples of modulated waveforms with

\[ m = 0.5 \quad (50\% \text{ modulation}) \quad \text{and} \quad m = 1.0 \quad (100\% \text{ modulation}) : \]

\[ y(t) = (x(t) + A) \cos \omega t \]

Figure 8.13: Output of the amplitude modulation system of Figure 8.12: (a) modulation index \( m = 0.5 \); (b) modulation index \( m = 1.0 \).

- When synchronous demodulation is used, the transmitted signal is:
  \[ y(t) = x(t) \cos \omega t \]

- When asynchronous demodulation is used, the transmitted signal is:
  \[ y(t) = [x(t) + A] \cos \omega t \]

- The Fourier transform of \( y(t) \) is not the same in these two cases.
- Spectrum $Y(w)$ of transmitted signal for the case of synchronous and asynchronous demodulation.

$X(j\omega)$

- $Y(w)$ for synchronous demodulation
- $Y(w)$ for asynchronous demodulation

- The dirac deltas in the asynchronous demodulation case come from the term $A\cos\omega_c t$ in $y(t) = [x(t)+A]\cos\omega_c t$.

→ Usually, you want to make $A$ small to minimize the power required to transmit the signal.

→ This is equivalent to wanting a large value of the modulation index $m = \frac{\omega}{\Delta}$.
- But, the smaller the modulation index, the better the simple envelope detector is able to track the envelope and approximate $X(t)$ at the receiver.

- Thus, there is always a tradeoff between transmission efficiency and quality of the demodulated signal when asynchronous demodulation is used.

**Frequency Division Multiplexing (FDM)**

- In many communications engineering applications, more than one information signal is transmitted through the channel simultaneously.

**EX:** Cable television: the transmitted and received signals going through the channel (coaxial cable) contain many channels.

**EX:** Broadcast TV and radio: Each station transmits one modulated signal into the channel (public airwaves). The received signal is the sum of these.
EX: internet or token ring data network:
    many different data signals are being transmitted simultaneously through the network.

EX: T1 telephone trunk:
    - A T1 trunk can hold about \( \frac{256 \times 2}{64} \) individual telephone calls simultaneously.

- One way of achieving the simultaneous transmission of several information signals through the channel is to divide up the channel bandwidth into frequency bands,

  \( \rightarrow \) Each information signal gets one of the frequency bands.

  \( \rightarrow \) Each information signal uses a carrier signal \( c(t) = \cos \omega t \) or \( c(t) = e^{i\omega t} \) that frequency shifts the information signal \( x(t) \) into its assigned frequency band.

- This is called "frequency division multiplexing", or FDM.
- The carrier frequencies have to be spaced far enough apart so that the modulated signals \( y(t) \) have Fourier spectra \( Y(w) \) that do not overlap one another.

\[ \text{EX: } \]

- The information signals are \( x_a(t), x_b(t), x_c(t), \ldots \)

- The carrier signals are \( \cos \omega_a t, \cos \omega_b t, \cos \omega_c t, \ldots \), with carrier frequencies \( \omega_a, \omega_b, \omega_c, \ldots \)

- The modulated signals are \( y_a(t), y_b(t), y_c(t), \ldots \)

- The signal that is transmitted through the channel is \( w(t) = y_a(t) + y_b(t) + y_c(t) + \ldots \)
- In the frequency domain:

\[ X_a(j\omega) \quad X_b(j\omega) \quad X_c(j\omega) \]

\[ Y_a(j\omega) \quad Y_b(j\omega) \]

\[ Y_c(j\omega) \]

\[ W(j\omega) \]

- To recover any one given information signal, say \( x_a(t) \), from \( w(t) \) requires two steps...
- Step 1: **Demultiplex** \( w(t) \) to recover \( y_a(t) \).

- Step 2: Demodulate \( y_a(t) \) to recover \( x_a(t) \).

- **To demultiplex**, you bandpass filter \( w(t) \) to get \( y_a(t) \).

- **To demodulate**, you demodulate the recovered \( y_a(t) \) as before.

![Diagram of demultiplexing and demodulation](image)

**Figure 9.17** Demultiplexing and demodulation for a frequency-division multiplexed signal.

**NOTE**: in this figure, synchronous demodulation is used.

- For AM radio, the tuning dial on your radio controls the center frequency of the bandpass filter. It also controls the carrier frequency used for demodulation.
**NOTE**: For broadcast radio, asynchronous demodulation is used to keep the receivers from becoming too expensive.

**NOTE**: For the AM we have discussed so far, the modulated signal spectrum $Y(w)$ contains two copies of the information signal spectrum $X(w)$:

![Diagram](image)

- This scheme is called "double sideband AM".
- Since both copies must be transmitted, the modulated signal $y(t)$ needs twice as much bandwidth in the channel as the original information signal $x(t)$.
- Thus, double sideband AM is inherently bandwidth inefficient.
Single-Sideband AM

- With a real carrier \( \cos \omega_c t \), we have seen that the modulated signal \( y(t) \) occupies twice as much bandwidth as the information signal \( x(t) \).

- With a complex carrier \( e^{j\omega_c t} \), the modulated signal \( y(t) \) occupied the same bandwidth as \( x(t) \).

\[ \rightarrow \text{But this requires a complex modulated signal and a complex carrier signal, which is undesirable in many applications.} \]

- Single-Sideband AM is a way to get the best of both worlds:

\[ \rightarrow \text{The carrier signal and modulated signal are real.} \]

\[ \rightarrow \text{The modulated signal occupies the same bandwidth as the information signal} \ x(t). \]
- The information signal spectrum \( X(w) \) contains positive frequencies and negative frequencies.

- When we modulate with a real carrier \( c(t) = \cos(w_c t) \), \( Y(w) \) contains two copies of the positive frequencies and two copies of the negative frequencies.

- These copies are called upper and lower sidebands, as shown below:

- There are two copies of \( X(w) \), one centered at \( w_c \) and one centered at \(-w_c\).

- Each copy of \( X(w) \) contains a copy of the positive frequencies and a copy of the negative frequencies.
- The upper sideband contains the negative frequencies from the copy of $X(\omega)$ centered at $-\omega_c$ and the positive frequencies from the copy of $X(\omega)$ centered at $\omega_c$.

- The lower sideband consists of the negative frequencies from the copy of $X(\omega)$ centered at $\omega_c$ and the positive frequencies from the copy of $X(\omega)$ centered at $-\omega_c$.

**NOTE:** $X(\omega)$ can be recovered from the upper sideband alone:

![Upper Sideband Diagram]

- Similarly, $X(\omega)$ can be recovered from the lower sideband alone:

![Lower Sideband Diagram]
- So, by transmitting only one sideband, we can still recover $X(\omega)$, but the required transmission bandwidth is cut in half as compared to double sideband AM.

- One way to get just the upper sideband is to apply a high-pass filter to $y(t)$.

$\rightarrow$ The filtered signal is called $y_u(t)$, for “upper sideband.”

**NOTE:** The filter must have a very sharp cutoff frequency.
Another way to keep just one sideband is to use phase shifting to cancel the unwanted sideband.

\[
H(j\omega) = \begin{cases} 
-1 & \omega > 0 \\
+1 & \omega < 0 
\end{cases} \]

\[
x_{p}(t) \quad y_{p}(t) \quad y(t)
\]

Figure 8.21 System for single-sideband amplitude modulation, using a 90° phase-shift network, in which only the lower sidebands are retained.

Figure 8.22 Spectra associated with the single-sideband system of Figure 8.21.
More details about how the single-sideband AM transmitter on page 8.27 works:

$$\cos \omega t \leftrightarrow \frac{\pi}{\omega} \left[ \delta(\omega - \omega_c) + \delta(\omega + \omega_c) \right]$$

$$\sin \omega t \leftrightarrow -j \frac{\pi}{\omega} \left[ \delta(\omega - \omega_c) - \delta(\omega + \omega_c) \right]$$

Suppose $X(\omega) = \begin{cases} \text{Re} & \omega < 0 \\ \text{Im} & \omega > 0 \end{cases}$

Then $X_0(\omega) = \begin{cases} \text{Re} & \omega < 0 \\ \text{Im} & \omega > 0 \end{cases}$

So $Y_1(\omega) = \begin{cases} \text{Re} & \omega < 0 \\ \text{Im} & \omega > 0 \end{cases}$

and $Y_2(\omega) = \begin{cases} \text{Re} & \omega < 0 \\ \text{Im} & \omega > 0 \end{cases}$

$Y(\omega) = Y_1(\omega) + Y_2(\omega) = \begin{cases} \text{Re} & \omega < 0 \\ \text{Im} & \omega > 0 \end{cases}$

→ Only the lower sideband is retained.
- To retain the lower sideband, the frequency response of the phase shifting filter is

\[ H(\omega) = \begin{cases} -j, & \omega > 0 \\ j, & \omega < 0 \end{cases} \]

- To retain the upper sideband, the filter frequency response is

\[ H(\omega) = \begin{cases} j, & \omega > 0 \\ -j, & \omega < 0 \end{cases} \]

- Some acronyms:

**AM-DSB/WC**: double-sideband AM with carrier for asynchronous demodulation.

**AM-DSB/SC**: double sideband AM without carrier (supressed carrier) for synchronous demodulation.

**AM-SSB/WC**: single-sideband AM with carrier for asynchronous demodulation.

**AM-SSB/SC**: single-sideband AM with supressed carrier for synchronous demodulation.
AM with a Pulse Train Carrier

- In this case, the carrier signal $c(t)$ is not sinusoidal, but is instead a sequence of equally spaced rectangular pulses of equal height and equal width:

- Pulse height = $A$
- Pulse width = $\Delta$
- Pulse spacing = $T$

$\rightarrow$ The information signal is $x(t)$.
$\rightarrow$ The transmitted signal is $y(t) = x(t)c(t)$
Fourier transform time multiplication property:

\[ Y(\omega) = \frac{1}{2\pi} X(\omega) \ast C(\omega) \, . \]

Let's find \( C(\omega) \):

- \( c(t) \) is periodic with period \( T \).
- Write \( c(t) \) in a Fourier series (like we did in chap. 7):

\[
c(t) = \sum_{k=-\infty}^{\infty} a_k e^{jk \omega_c t} \quad \omega_c = \frac{2\pi}{T}
\]

\[
a_k = \frac{1}{T} \int_{-T/2}^{T/2} c(t) e^{-jk \omega_c t} \, dt
\]

\[
= \frac{1}{T} \int_{-\Delta/2}^{\Delta/2} e^{-jk \omega_c t} \, dt = \begin{cases} \Delta/T, & k=0 \\ \frac{\sin(k \omega_c \Delta/2)}{\pi k}, & k \neq 0 \end{cases}
\]

So

\[
C(\omega) = \mathcal{F}\{c(t)\} = \mathcal{F}\left\{ \sum_{k=-\infty}^{\infty} a_k e^{jk \omega_c t} \right\}
\]

\[
= 2\pi \sum_{k=-\infty}^{\infty} a_k \delta(\omega - \omega_c)
\]
So the transmitted signal \( y(t) = x(t)c(t) \) has a Fourier transform
\[
Y(\omega) = \frac{1}{2\pi} X(\omega) \ast C(\omega)
\]
\[
= \sum_{k=-\infty}^{\infty} a_k X(\omega - k\omega_c)
\]

\( Y(\omega) \) contains equally spaced copies of \( X(\omega) \) that are weighted by the Fourier series coefficients \( a_k \).

Note: we are assuming that \( x(t) \) is bandlimited to some frequency \( \omega_m \).
As in our discussion of sampling, there will be no aliasing provided that \( \omega_c > 2\omega_m \), so that the copies of \( X(w) \) do not overlap.

→ As long as this is the case, \( x(t) \) can be recovered from the modulated signal \( y(t) \) by applying a low-pass filter.

→ If there is aliasing, then \( x(t) \) cannot be recovered from \( y(t) \).

We have already seen how frequency division multiplexing (FDM) can be used to simultaneously transmit several signals over a single channel.

With pulse-train AM, we can also accomplish this using Time Division Multiplexing (TDM).

→ With FDM, the channel is divided into frequency "slices", or slots. Each signal gets one slice.

→ With TDM, the channel is divided into time slots. The signals share the time slots in round-robin order.
- TDM can be accomplished with a switch.

- Each information signal $X_i(t)$ is multiplied by a pulse-train carrier.

- The pulses for the different carriers do not overlap in time.
Pulse Amplitude Modulation (PAM)

- In pulse-train AM, as in sampling, it is the pulse frequency \( \omega_c = \frac{2\pi}{T} \), not the pulse width \( \Delta \), that determines if we can recover \( x(t) \) from \( y(t) \).

- So, as far as demodulation is concerned, using "time slices" of \( x(t) \) as we did in pulse-train AM is really no better than just using samples \( x(nT) \) of \( x(t) \).

- If the carrier pulse frequency \( \omega_c = \frac{2\pi}{T} \) is fast enough to recover \( x(t) \), we might as well just sample \( x(t) \) and set the pulse heights in \( y(t) \) equal to the samples \( x(nT) \).

\[ \implies \text{This is called \underline{Pulse Amplitude Modulation}, or "PAM".} \]

\[ \rightarrow \text{It is a way to modulate a pulse-train carrier } c(t) \text{ with a sampled, or "digital" signal } x[n] = x(nT): \]

\[ y(t) = x[nT]c(t). \]

- If the carrier frequency \( \omega_c = \frac{2\pi}{T} \) exceeds the Nyquist rate, then we can recover \( x(t) \) from \( y(t) \).
- PAM for a single information signal $x(t)$:

- As with pulse-train AM, we can use time division multiplexing to transmit several PAM signals simultaneously through a single channel:

- For this PAM TDM system to work, we need for the received signal to look like the transmitted signal $y(t)$.

- In particular, we need the pulses in the received signal to not overlap, so that we can "undo" the TDM and recover the individual samples of each information signal.

- But is this practical?
- Each pulse in the transmitted signal $y(t)$ is a boxcar.
  → each one has a Fourier transform that is a sync.
  → each sync has infinite bandwidth:

- The Fourier transform of $y(t)$ is the sum of these syncs.

  → so the transmitted signal $y(t)$ has infinite bandwidth.

- But any real channel has finite bandwidth.
  → so the transmitted signal will get distorted in the channel.

- Suppose that the frequency response of the channel is $H(w)$:

  - For frequencies where $H(w) \neq 0$, we can correct the distortion by applying the inverse filter $H^{-1}(w) = \frac{1}{H(w)}$.

  - This is called "channel equalization."

- But channel equalization cannot help for the frequencies where $|H(w)| = 0$.

- So there will always be low-pass distortion when $y(t)$ is transmitted through the channel.

- If the received signal is $r(t)$, then $R(w) = Y(w) H(w)$ and $r(t) \neq y(t)$, even if channel equalization is used.
In practice, the best we can do is to make the channel equalizer the pseudo-inverse of \( H(\omega) \):

\[
H_{\text{pseudo}}^{-1}(\omega) = \begin{cases} 
\frac{1}{H(\omega)}, & H(\omega) \neq 0 \\
0, & H(\omega) = 0 
\end{cases}
\]

But some frequencies will always be lost in the received signal \( r(t) \).

This generally has two effects:
- The sharp corners of the pulses will get rounded off.
- The pulses will get spread out or "smeared" in time.

When we attempt to demultiplex the received PAM TDM signal, adjacent pulses will generally interfere with each other... degrading the values of the demultiplexed information signal samples.

This is called **intersymbol interference**.
One way to deal with this is to build the carrier waves out of bandlimited pulses:

Then, the pulses will not be degraded by the channel.

However, we must design the carrier signal $c(t)$ carefully, so that, at the time instant we sample one pulse, all the other pulses are zero.
Digital Pulse Code Modulation (PCM)

In many modern communications systems, the information signals are digital.

→ We begin with an analog signal \( x(t) \).
→ We sample to get a discrete-time signal \( x[n] = x(nT) \).
→ We quantize to get a digital signal \( x[n] \) that only takes integer values in a finite range.

→ For example, with 8-bit quantization, each \( x[n] \) is an integer in the range \( 0 \leq x[n] \leq 255 \).

→ 16 bits is often used for digital speech and telephony.

→ If the samples \( x[n] \) are converted from parallel to serial, we get a PAM TDM system where all the pulse heights are zero or one:

\[
\begin{align*}
X_1[n] &\rightarrow \text{Parallel to Serial} \rightarrow 10011101 \ldots \\
X_2[n] &\rightarrow \text{Parallel to Serial} \rightarrow 01101110 \ldots \\
\vdots \\
X_m[n] &\rightarrow \text{Parallel to Serial} \rightarrow 11111000 \ldots \\
\end{align*}
\]

→ This is called Pulse Code Modulation, or "PCM".

→ With digital PCM, extra bits can be added to implement parity codes or cyclic redundancy checks (CRCs) for error correction.