Signal: A function of time that contains in it some sort of desired information.
Communications: The signal is sent by a human and received by a human.
Signal Processing: The signal is sent by *nature* and received by a human.
Most real-world signals are *continuous* time: Audio, radio, bioelectric, ECG, EKG.

Discrete: *Sample* continuous-time signal $x(t)$ by setting $t = nT_s$ for some small T_s.
Example: $x(t) = \cos(2\pi 60t) \rightarrow x[n] = x(t = 0.001n) = \cos(0.12\pi n)$ (discrete-time).

Q: WHAT do we want to do with a signal $x(t)$? **A:** *Filter* it. **Q:** What does that mean?
A: Alter its *spectrum.* **Q:** What's a spectrum? **A:** It's same as the output of a prism:
If $x(t)$ is periodic with period T (this means that $x(t) = x(t + T)$ for all t), then:
$x(t) = c_o + c_1 \cos(\frac{2\pi}{T}t + \theta_1) + c_2 \cos(\frac{4\pi}{T}t + \theta_2) + c_3 \cos(\frac{6\pi}{T}t + \theta_3) + \ldots$

This is the *Fourier series expansion* of $x(t)$. *Filtering* means to alter the $\{c_n\}, \{\theta_n\}$.
The *spectrum* of $x(t)$ is the sinusoids in the Fourier series (like output of a prism).

Q: WHY do we want to filter $x(t)$? **A:** To do any of the following (all in EECs 206):

1. To *reduce noise* in the signal by decreasing $\{c_n\}$ for large n (low-pass filter);
2. To *detect edges* in the signal by increasing $\{c_n\}$ for large n (high-pass filter);
3. To *eliminate interference* by decreasing $\{c_n\}$ for ONE n (notch or band-reject filter);
4. To *dereverberate* $x(t)$ (eliminate undesired echoes present in $x(t)$) (echo cancellation):
   Examples: Multipath in cellular phones, water column in seismic processing.
5. To *reverberate* $x(t)$ (add in desired echoes of $x(t)$ for a fuller sound) (reverberator):
   Example: Make a lousy singer sound better (the "singing in bathroom" effect).

Q: HOW do we filter $x(t)$? **A:** By sampling it and passing $x[n]$ through a *system.*
Example: Input samples $x[n]$ through this system to get output samples $y[n]$:
$x[n] \rightarrow \boxed{y[n] - 1.84y[n-1] + 0.98y[n-2] = x[n] - 1.86x[n-1] + x[n-2]} \rightarrow y[n]$.
This eliminates 60 Hz interference in a signal $x(t)$ sampled with $T_s = 0.001$ second.

Q: Where did THAT equation come from?! **A:** Stay tuned to EECS 206!
Q: How do we know what this filter does? **A:** Compute its *frequency response.*
Q: How do we *design* this filter? **A:** By using its *transfer function (poles &zeros).*

Topics in EECS 206 and Why We Will Be Studying Them:

1. **Signal properties:** Energy, power, mean, variance, statistics; for detection.
2. **Sinusoids:** Building blocks of signal spectrum and Fourier series; musical tones.
3. **Complex numbers:** Math tool for manipulating sinusoids–easier than trigonometry!
4. **Line spectra:** Graphical picture of spectrum of sum of sinusoids. Simple version of:
5. **Fourier Series:** Writing a periodic signal $x(t)$ as sum of sinusoids (see above).
6. **Sampling:** Can't process $x(t)$ itself; CAN process the samples $x[n]$ of $x(t)$.
7. **Discrete Fourier Transform:** Writing a periodic signal $x[n]$ as a sum of sinusoids.
8. **Systems:** What we use to perform signal processing on $x[n]$ (see example above).
9. **Frequency Response:** What a given system will do to the spectrum of $x[n]$.
10. **z-transforms:** Math tool for systems–easier than doing things directly.
11. **Transfer functions:** Best way to represent systems–get other ways from it.
12. **Poles and zeros:** Use to design systems having a desired frequency response.