

Lecture 4

Signal Classification and Filters

- Classification of Signals
- Signal Transmission via Linear Systems
- Filters

Contents

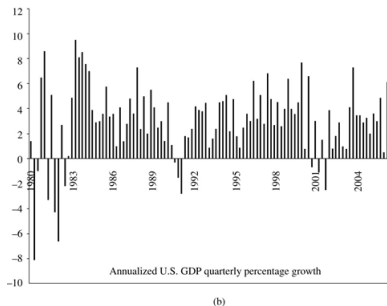
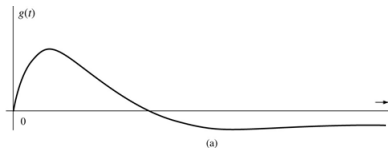
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Classification of Signals

- Continuous time and discrete time signals
- Analog and digital signals
- Periodic and aperiodic signals
- Energy and power signals
- Deterministic and probabilistic signals

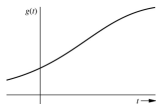
Continuous vs. Discrete Time

- **Continuous-Time Signal:** is specified for every value of time t (e.g., audio and video recording)
- **Discrete-Time Signal:** is specified only at discrete points of $t = nT$ (e.g., daily stocks value, quarterly GDP)

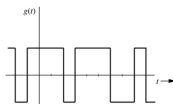


Analog vs. Digital

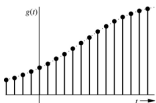
- **Analog Signal:** A signals whose amplitude can have any value in a continuous range
- **Digital Signal:** A signals whose amplitude can take only a finite number of values
- (a) analog/continuous time (mathematics) (b) digital/continuous time (c) analog/discrete time (digital signal processing) (d) digital/discrete time (digital switching)



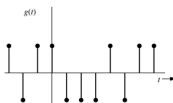
(a)



(b)



(c)



(d)

Don't mix up analog signals with continuous time signals; the two concepts are not the same.

Energy vs. Power Signals

Measures of size of a signal

- **Signal Energy:** the energy of a signal $g(t)$ is the area under $|g(t)|^2$

$$E_g = \int_{-\infty}^{\infty} |g(t)|^2 dt$$

- **Signal Power:** mean squared value of $|g(t)|$

$$P_g = \lim_{T \rightarrow \infty} \frac{1}{T} \int_{-\frac{T}{2}}^{\frac{T}{2}} |g(t)|^2 dt$$

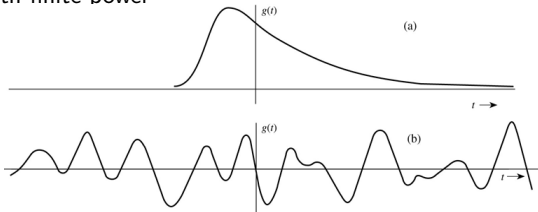
- $\sqrt{P_g}$ is the familiar **rms** (root mean square) value
- **Energy vs. Power Signals:**
 - A signal with finite energy ($E_g < \infty$) is an *energy signal*
 - A signal with finite non-zero power ($0 < P_g < \infty$) is a *power signal*

- A signal cannot be both an energy signal and a power signal
- Certain signals are neither energy nor power signals ($P_g = \infty$).

Example

A necessary condition for the energy to be finite is that the signal amplitude must $\rightarrow 0$ when $|t| \rightarrow \infty$
(otherwise the energy integral will not converge)

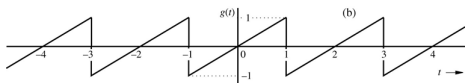
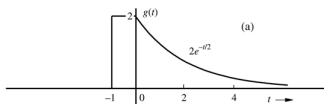
- (a) signal with finite energy
- (b) signal with finite power



- Can you come up with a neither energy and nor power signal?
Examples: $g(t) = e^t$ and $g(t) = t$

Example

- Determine a suitable measure (energy or power) for the below signals



- (a) $E_g = \int_{-\infty}^{\infty} g(t)^2 dt = \int_{-1}^0 2^2 dt + \int_0^{\infty} 4e^{-t} dt = 4 + 4 = 8 \Rightarrow$ energy signal
- (b) obviously not an energy signal (why?)

$$\begin{aligned} P_g &= \lim_{T \rightarrow \infty} \frac{1}{T} \int_{-\frac{T}{2}}^{\frac{T}{2}} |g(t)|^2 dt \\ &= \frac{1}{T} \int_{-\frac{T}{2}}^{\frac{T}{2}} |g(t)|^2 dt \quad \text{because } g(t) \text{ is periodic} \\ &= \frac{1}{2} \int_{-1}^1 t^2 dt = \frac{1}{3} \Rightarrow \text{power signal} \end{aligned}$$

Units of Power

1. Watts

- power level may vary from megawatts (TV transmitter) to 10^{-12} watts (satellite receiver)

2. dBW and dBm

- powers ratio can be expressed in decibels (dB). dB is logarithmic and relative to some standard power¹

$$\left(\frac{P}{P_0} \right)_{dB} = 10 \log_{10} \left(\frac{P}{P_0} \right)$$

- if P is measured in watts, then
 - power in **dBW** = $10 \log_{10} \left(\frac{P}{1W} \right)$ (power relative to $P_0 = 1W$)
 - power in **dBm** (or dBmW) = $10 \log_{10} \left(\frac{P}{10^{-3}W} \right) = 30 + 10 \log_{10} P$;
 - thus

$$\text{dBm} = \text{dBW} + 30$$

¹Equivalently, number of decibels (dB) in terms of voltage is equal to $20 \log_{10} \left(\frac{V}{V_0} \right)$.

Units of Power

- dBW and dBm are power units
- dB is a relative, unit-less number; it represents gain/loss in a logarithmic scale

Examples:

- 3dB loss means the signal strength has halved
- 10dB gain means the signal strength has increased 10 times
- **Q:** typical cell-phone sensitivity is -107 dBm. What is the sensitivity in watts?

$$10 \log_{10} \left(\frac{P}{10^{-3}W} \right) = -107 \Rightarrow P = 1.995 \times 10^{-14}W$$

Find your-cell phone's received signal strength



- typical macro base station (cell tower) transmission power is 20W-69W. Express this in dBW and dBm. (Answer: 13-18 dBW or 43-48 dBm)

Examples of Communication Channels

- optical fiber (attenuation 4dB/km)
- broadcast TV (50 kW transmit)
- voice telephone line (under -9 dBm or 110 μ W)
- low-power not-for-profit FM (10W or 100W transmitter) (range of approximately 3.5 miles)
- walkie-talkie: 500 mW, 467 MHz
- Bluetooth: 20 dBm, 4 dBm, 0 dBm
- minimal received signal of wireless network (802.11 variants): 0.1 pW (-100 dBm)
- typical received signal power from a GPS satellite: -127.5 dBm

Gain and Power

Gain as a ratio	Gain in dB (= $10 \log_{10}[\text{gain as ratio}]$)
2	3 dB
4	6 dB
10	10 dB
50	17 dB
100	20 dB
1000	30 dB
0.5 (a loss of a factor of 2)	-3 dB
0.1 (a loss of a factor of 10)	-10 dB

Power in milliwatts	Power in dBm (= $10 \log_{10}[\text{power in mW}]$)
2	3 dBm
10 mW	10 dBm
100 mW	20 dBm
1000 mW	30 dBm
20 W = 20000 mW	43 dBm
1 μ W	-30 dBm

Energy Conservation

Parseval's Theorem: $\int_{-\infty}^{\infty} |g(t)|^2 dt = \int_{-\infty}^{\infty} |G(f)|^2 df$

- Parseval's theorem says that energy in time domain is equal to energy in the frequency domain
- **Example:**
Find the energy of the sinc pulse $g(t) = 3 \text{sinc}(2t)$.

$$\begin{aligned} E &= \int_{-\infty}^{\infty} 3^2 \text{sinc}^2(2t) dt \\ &= \left(\frac{3}{2}\right)^2 \int_{-\infty}^{\infty} \text{rect}^2\left(\frac{f}{2}\right) df = \left(\frac{3}{2}\right)^2 \int_{-1}^1 df = \frac{9}{2} \end{aligned}$$

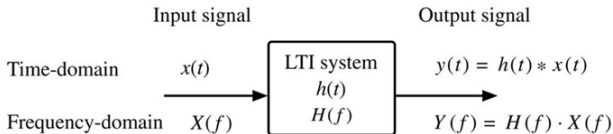
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Contents

- Classification of Signals
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Linear Systems

- A *linear time-invariant* (LTI) can often be used to characterize communication channels.



- A stable LTI is characterized in time domain by its *impulse response* $h(t)$
- Impulse response is the system response to a *unit impulse* input $\delta(t)$ that is $y(t) = h(t)$ when $x(t) = \delta(t)$

The system response to a bounded input $x(t)$ is the following convolution relationship: $y(t) = h(t) * x(t)$

Transfer Function

- $H(f)$, the FT of the impulse response $h(t)$, is generally referred to as **transfer function** or **frequency response**
- $H(f)$ is generally a complex number that can be written as

$$H(f) = |H(f)|e^{j\theta_h(f)}$$

- Signal **distortion** via transmission

$$Y(f) = H(f)X(f)$$

$$\Rightarrow |Y(f)|e^{j\theta_y(f)} = |H(f)|e^{j\theta_h(f)}|X(f)|e^{j\theta_x(f)}$$

or, equivalently,

$$\begin{cases} |Y(f)| &= |H(f)| |X(f)| \\ \theta_y(f) &= \theta_h(f) + \theta_x(f) \end{cases}$$

During transmission

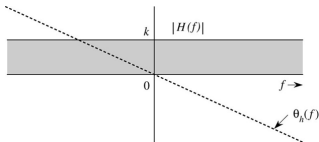
- the input amplitude spectrum $|X(f)|$ has changed to $|H(f)| |X(f)|$
- the input phase spectrum $\theta_x(f)$ has changed to $\theta_h(f) + \theta_x(f)$

Distortionless Transmission

- In signal transmission over a communication channel, we require the output waveform is required to be a *replica* of the input waveform
- Transmission is said to be **distortionless** if the output has the same waveform as input except for a multiplicative constant and/or a delay

$$y(t) = k \cdot x(t - t_d)$$

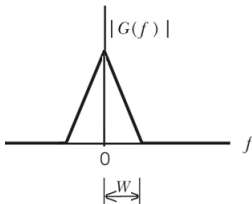
$$\Rightarrow Y(f) = kX(f)e^{-j2\pi ft_d} \quad \text{or, equivalently,} \quad \begin{cases} |H(f)| &= k \\ \theta_h(f) &= -2\pi ft_d \end{cases}$$



- Hence, the distortionless system's **magnitude is constant** and its **phase is linear with frequency**

Low-Pass Signals

- A signal is said to be **low-pass** if its non-zero frequency content is limited by $|f| \leq W$, where W is the signal bandwidth and is a constant.



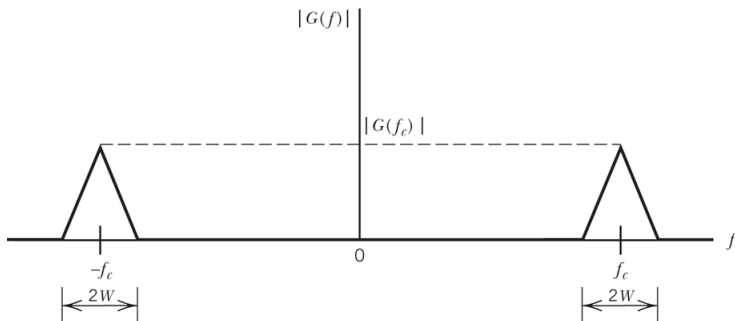
Example: spectrum of a low-pass signal

- A communication using low-pass signals is referred to as **baseband communication**

Band-Pass Signals

- **Band-Pass Signal:**

- A **bandpass signal** is a signal containing a band of frequencies not adjacent to zero frequency ($|f - f_c| \leq W$)
- f_c is referred to as carrier frequency
- if the bandwidth ($2W$) is small compared to f_c , the signal is called *narrow-band*



Example: spectrum of a band-pass signal

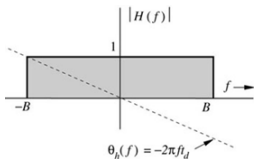
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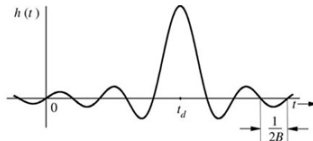
Ideal Low-Pass Filters

- *Ideal* filters allow distortionless transmission of a certain frequency band and suppress all other frequencies

Low-Pass Filter:



(a)

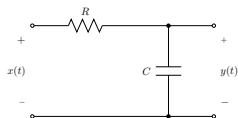


(b)

$$H(f) = |H(f)|e^{j\theta_h(f)} = \Pi\left(\frac{f}{2B}\right)e^{-j2\pi f t_d}$$
$$\Rightarrow h(t) = F^{-1}[H(f)] = 2B\text{sinc}[2B(t - t_d)]$$

- Fig. (b) shows that this “ideal” filter is practically unrealizable. (why?)

Non-Ideal Low-Pass Filters

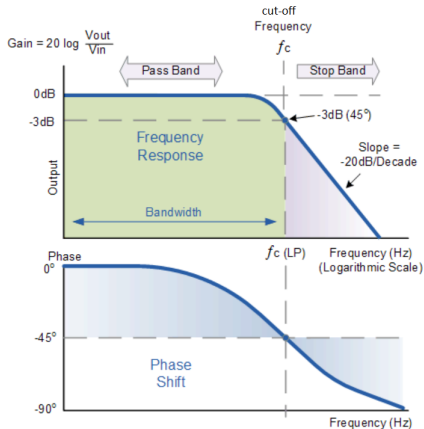


Example: first-order Butterworth filter

- $H(j\omega) = \frac{Y(j\omega)}{X(j\omega)} = \frac{1/jC\omega}{R+1/jC\omega} = \frac{1}{1+jRC\omega}$
 - magnitude of frequency response: $|H(j\omega)| = \frac{1}{\sqrt{1+(RC\omega)^2}}$
 - phase of frequency response: $\angle H(j\omega) = -\tan^{-1} RC\omega$
- for $RC\omega \ll 1$
 - $|H(j\omega)| \approx 1$
 - $\angle H(j\omega) \approx -RC\omega$
- therefore, for $RC\omega \ll 1$ (equivalently $f \ll \frac{1}{2\pi RC}$) the magnitude is constant and the phase is linear with frequency (as required in a distortionless transmission)
- for $RC\omega \ll 1$, the RC circuit approximates the ideal low-pass filter
- cut-off frequency (bandwidth) is where $|H(j\omega)| = \frac{1}{\sqrt{2}}$ (power halves)

Frequency Response of First Order Low-Pass Filter

- Frequency response curve or Bode Plot
- V_{in} and V_{out} are the input ($x(t)$) and output ($y(t)$)



Filters Example

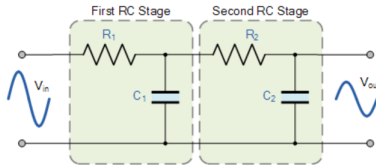
Example: A Low-pass filter circuit consisting of a resistor of $4.7k\Omega$ in series with a capacitor of $47nF$ is connected across a $10v$ sinusoidal supply. Calculate the output voltage at a frequency of $100Hz$ and again at frequency of $10kHz$. Which one passes the filter? What is the cut-off frequency?

- $f = 100Hz \Rightarrow \omega RC = 2\pi fRC = 0.1388 \Rightarrow$
 $|H(j\omega)| = \frac{1}{\sqrt{1+(RC\omega)^2}} = \frac{1}{\sqrt{1+0.0193}} = 0.9905 \Rightarrow V_{out} = 0.99 * V_{in} = 9.9V$
- $f = 10,000Hz \Rightarrow \omega RC = 2\pi fRC = 13.88 \Rightarrow$
 $|H(j\omega)| = \frac{1}{\sqrt{1+(RC\omega)^2}} = \frac{1}{\sqrt{1+192.64}} = 0.0719 \Rightarrow V_{out} = 0.0719 * 10 = 0.72V$
- **cut-off frequency:** $2\pi fRC = 1 \Rightarrow f = 720.5Hz$
- We say $100Hz$ passes the filter but $10kHz$ does not.

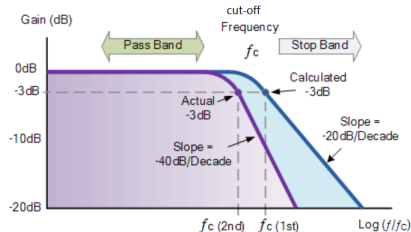
Second Order Low-Pass Filter- An Example

- cascaded two first-order low pass filters

Second-order Low Pass Filter



Frequency Response of a 2nd-order Low Pass Filter

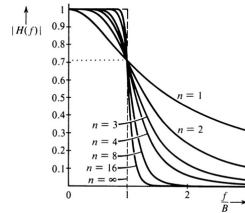


Filters

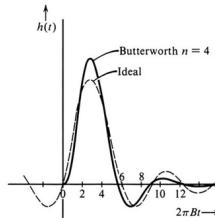
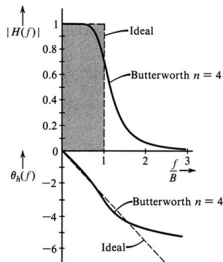
- n th order Butterworth filter

- Magnitude of frequency response:

$$|H(f)| = \frac{1}{\sqrt{1 + (\frac{f}{B})^{2n}}} \text{ where } B \text{ is the bandwidth}$$



- $n = 4$ (notice the causal impulse response)

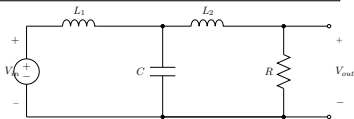


Filters Example

Example: Find the transfer function of the Butterworth filter of the figure below.

- What is the order of the filter? Is it a low-pass or a high-pass?
- Show that with $C = \frac{4}{3}F$, $R = 1\Omega$, $L_1 = \frac{3}{2}H$, and $L_2 = \frac{1}{2}H$ the magnitude of the frequency response is given by $|H(j\omega)| = \frac{1}{\sqrt{1+\omega^6}}$
- For what frequencies the filter is nearly ideal?

$$\frac{V_o(s)}{V_i(s)} = \frac{R}{s^3(L_1CL_2) + s^2(L_1CR) + s(L_1 + L_2) + R}$$



For the given values, the transfer function simplifies to

$$H(s) = \frac{V_o(s)}{V_i(s)} = \frac{1}{s^3 + 2s^2 + 2s + 1}$$

Then, $|H(j\omega)| = \frac{1}{\sqrt{1+\omega^6}}$ (how?)

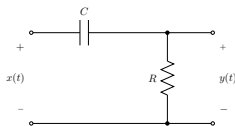
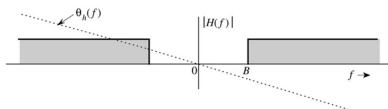
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See the filter notes in blackboard.

High-Pass Filters

- **High-Pass Filter:**

ideal frequency response and a simple circuit



- If the phase is zero

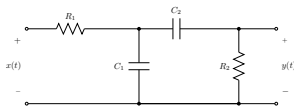
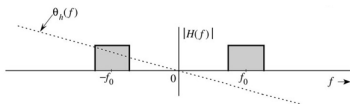
$$H(f) = 1 - \Pi\left(\frac{f}{2B}\right) \Rightarrow$$

$$h(t) = F^{-1}[H(f)] = \delta(t) - 2B \operatorname{sinc}[2Bt]$$

Band-Pass Filters

- **Band-Pass Filter:**

ideal frequency response and a simple circuit



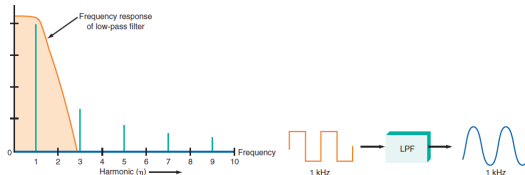
- If the phase is zero

$$h_{BP}(t) = 2 \cos(2\pi f_0 t) \text{sinc}(2Bt)$$

Filtering Examples

Converting a square wave to sine wave

- **Low-Pass Filter:** filtering out all the harmonics with a low-pass filter



- **Band-Pass Filter:** selecting the third harmonic with a bandpass filter

