Digital Signal Processing

- **先修课程：信号与系统**
- **后续课程： DSP原理及应用、语音信号处理、图像信号处理等**

Digital Signal Processing

教材：Sanjit K. Mitra. 数字信号处理－－基于计算机的方法（第四版）（英文改编版）

参考书：

[1] S. J. Orfanidis. Introduction to Signal Processing. 北京：清华大学出版社/ Prentice Hall 出版公司，2003

电子版：http://www.ece.rutgers.edu/~orfanidi/intro2sp/

[2] A.V.奥本海姆, R.W.谢弗.（离散时间信号处理）（第三版）. 北京：电子工业出版社，2011 (或：西安交通大学出版社的第二版中译本)

[3] R. G. Lyons. Understanding Digital Signal Processing, 北京：科学出版社，2003

[4] 程佩青．数字信号处理教程（第三版）. 北京：清华大学出版社，2007

[5] 其他数字信号处理教材

实验参考书：

[1]S.K.Mitra（著）,孙洪（译）.数字信号处理实验指导书（MATLAB版）. 北京：电子工业出版社，2005

[2]维纳.K.恩格尔 约翰.G.普罗克斯 ,刘树棠译，数字信号处理----使用MATLAB. 西安交通大学出版社

[3] 其它Matlab书籍

Content of the Course

- 1 Signals and Signal Processing
- 2 Discrete-Time Signals and Systems
- 3 Discrete-Time Fourier Transform
- 4 Digital Processing of Continuous-Time Signals
- 5 Finite-Length Discrete Transforms
- 6 z-Transform
- 7 LTI Discrete-Time Systems in the Transform Domain
- 8 Digital Filter Structures
- 9 IIR Digital Filter Design
- 10 FIR Digital Filter Design
- 11 DSP Algorithm Implementation

Chapter 1
Signals and Signal Processing

Chapter 1
Signals and Signal Processing

- 1.1 Characterization and Classification of Signals
- 1.2 Typical Signal Processing Operations
- 1.3 Examples of Typical Signals 3
- 1.4 Typical Signal Processing Applications
- 1.5 Why Digital Signal Processing?

知识概述和分析：

Signals and Signal Processing

- Signals play an important role in our daily life
- A signal is a function of independent variables such as time, distance, position, temperature, and pressure
- Most signals we encounter are generated naturally
- However, a signal can also be generated synthetically or by a computer

Signals and Signal Processing

- A signal carries information
- Objective of signal processing: Extract the useful information carried by the signal
- Method information extraction: Depends on the type of signal and the nature of the information being carried by the signal
- This course is concerned with the discrete-time representation of signals and their discrete-time processing

Characterization and Classification of Signals

- **Types of signal**: Depends on the nature of the independent variables and the value of the function defining the signal
- For example, the independent variables can be continuous or discrete
- Likewise, the signal can be a continuous or discrete function of the independent variables

Characterization and Classification of Signals

- Moreover, the signal can be either a real-valued function or a complex-valued function
- A signal generated by a single source is called a scalar signal
- A signal generated by multiple sources is called a vector signal or a multichannel signal

Characterization and Classification of Signals

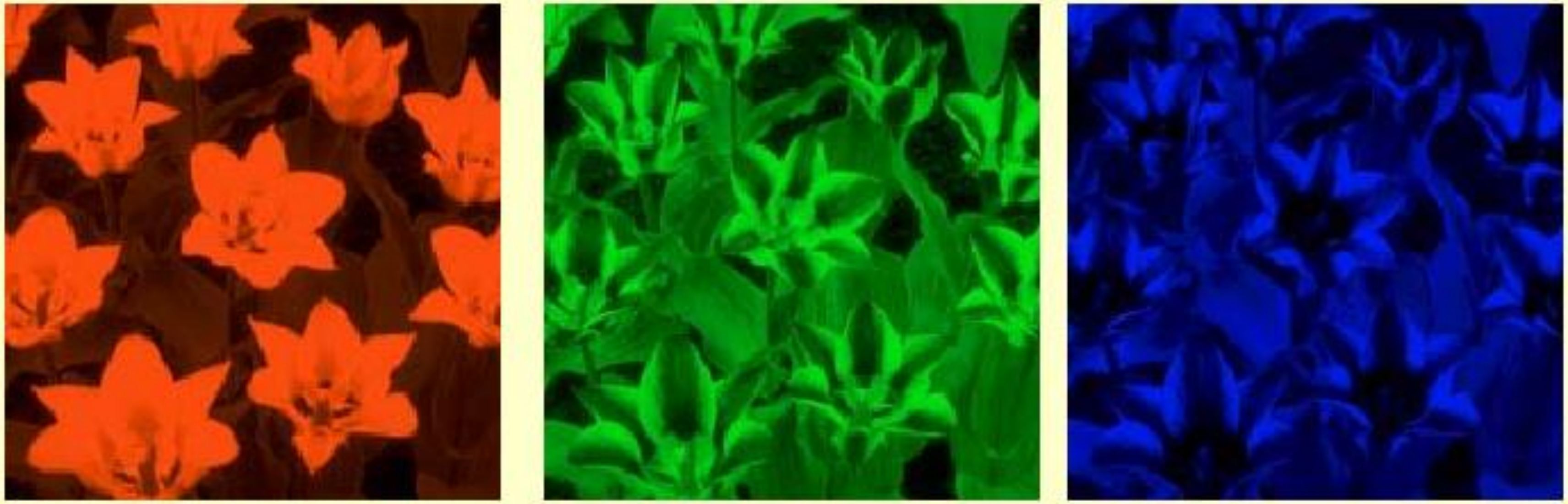
- A one-dimensional (1-D) signal is a function of a single independent variable
- A multidimensional (M-D) signal is a function of more than one independent variables
- The speech signal is an example of a 1-D signal where the independent variable is time

Characterization and Classification of Signals

- An image signal, such as a photograph, is an example of a 2-D signal where the 2 independent variables are the 2 spatial variables
- A color image signal is composed of three 2-D signals representing the three primary colors: red, green and blue (RGB)

Characterization and Classification of Signals

- The 3 color components of a color image are shown below



Characterization and Classification of Signals

- The full color image obtained by displaying the previous 3 color components is shown below

Characterization and Classification of Signals

- A color video signal is a vector signal composed of 3 signals representing the 3 primary colors: red, green, and blue

$$\mathbf{u}(x,y,t)=\begin{bmatrix} r(x,y,t) \\ g(x,y,t) \\ b(x,y,t) \end{bmatrix}$$

Characterization and Classification of Signals

- For a 1-D signal, the independent variable is usually labeled as time
- If the independent variable is continuous, the signal is called a continuous-time signal
- If the independent variable is discrete, the signal is called a discrete-time signal

Characterization and Classification of Signals

- A continuous-time signal is defined at every instant of time
- A discrete-time signal is defined at discrete instants of time, and hence, it is a sequence of numbers
- A continuous-time signal with a continuous amplitude is usually called an analog signal
- A speech signal is an example of an analog signal

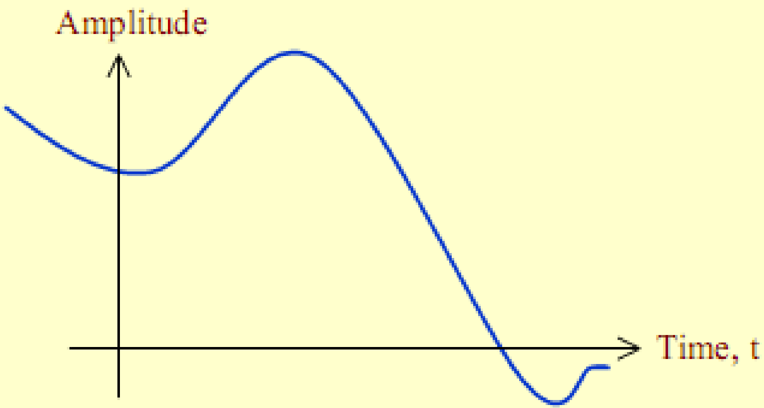
Characterization and Classification of Signals

- A discrete-time signal with discrete-valued amplitudes represented by a finite number of digits is referred to as the digital signal
- An example of a digital signal is the digitized music signal stored in a CD-ROM disk
- A discrete-time signal with continuous-valued amplitudes is called a sampled-data signal

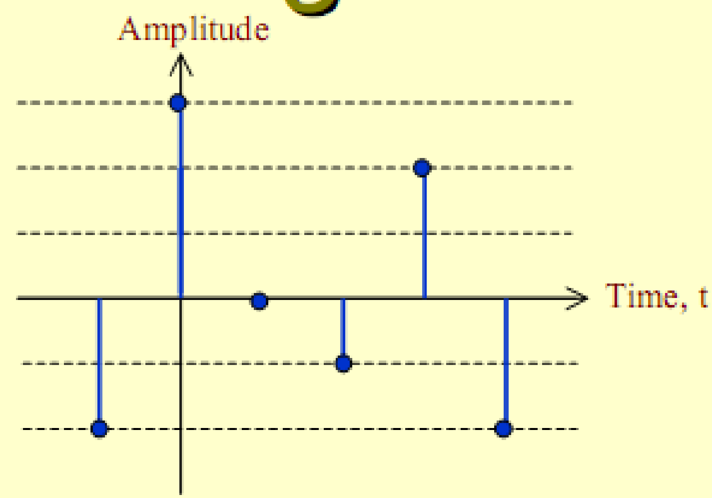
Characterization and Classification of Signals

- A digital signal is thus a quantized sampled-data signal
- A continuous-time signal with discrete-value amplitudes is usually called a quantized boxcar signal
- The figure in the next slide illustrates the 4 types of signals

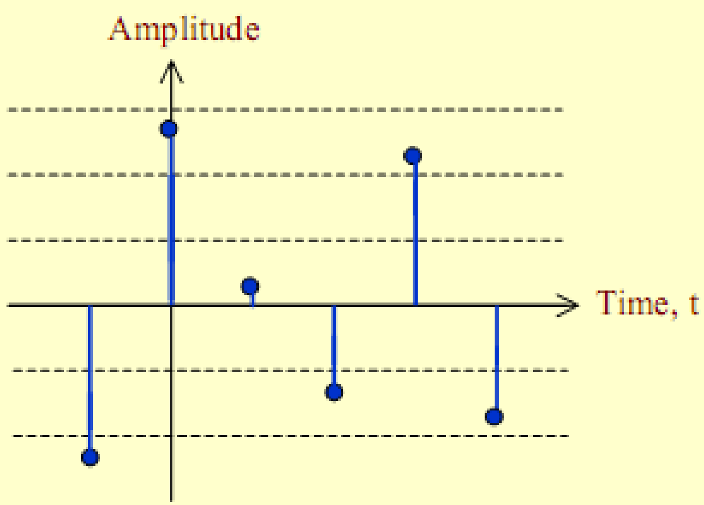
Characterization and Classification of Signals



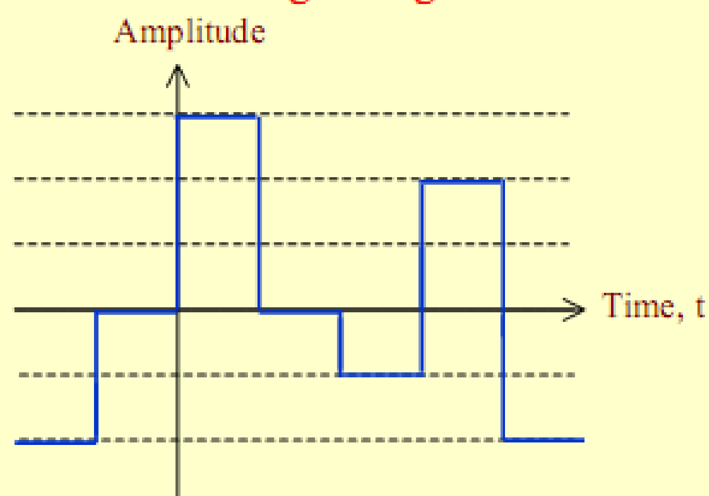
A continuous-time signal



A digital signal



A sampled - data signal



A quantized boxcar signal

Characterization and Classification of Signals

- The functional dependence of a signal in its mathematical representation is often explicitly shown
- For a continuous-time 1-D signal, the continuous independent variable is usually denoted by t
- For example, $u(t)$ represents a continuous-time 1-D signal

Characterization and Classification of Signals

- For a discrete-time 1-D signal, the discrete independent variable is usually denoted by n
- For example, $\{v[n]\}$ represents a discrete-time 1-D signal
- Each member, $v[n]$, of a discrete-time signal is called a sample

Characterization and Classification of Signals

- In many applications, a discrete-time signal is generated by sampling a parent continuous-time signal at uniform intervals of time
- If the discrete instants of time at which a discrete-time signal is defined are uniformly spaced, the independent discrete variable n can be normalized to assume integer values

Characterization and Classification of Signals

- A signal that can be uniquely determined by a well-defined process, such as a mathematical expression or rule, or table look-up, is called a deterministic signal
- A signal that is generated in a random fashion and cannot be predicted ahead of time is called a random signal

§ 1.2 Typical Signal Processing Operations

- Most signal processing operations in the case of analog signals are carried out in the time-domain
- In the case of discrete-time signals, both time-domain or frequency-domain operations are usually employed

§ 1.2.1 Simple Time-Domain Operations

Elementary Time-Domain Operation

- Three most basic time-domain signal operations are scaling, delay, and addition
- Scaling is simply the multiplication of a signal either by a positive or negative constant
- In the case of analog signals, the operation is usually called amplification if the magnitude of the multiplying constant, called gain, is greater than 1

Elementary Time-Domain Operations

- If the magnitude of the multiplying constant is less than 1, the operation is called attenuation
- If $x(t)$ is an analog signal that is scaled by a constant α, then the scaling operation generates a signal $y(t) = \alpha\, x(t)$
- Two other elementary operations are integration and differentiation

Elementary Time-Domain Operations

- The delay operation generates a signal that is a delayed replica of the original signal
- For an analog signal $x(t)$,
$$y(t) = x(t - t_o)$$
is the signal obtained by delaying $x(t)$ by the amount of time t_o which is assumed to be a positive number
- If t_o is negative, then it is an advance operation

Elementary Time-Domain Operations

- Many applications require operations involving two or more signals to generate a new signal
- For example,

$$y(t) = x_1(t) + x_2(t) + x_3(t)$$

is the signal generated by the addition of the three analog signals, $x_1(t)$, $x_2(t)$, and $x_3(t)$

Elementary Time-Domain Operations

- The product of 2 signals, $x_1(t)$ and $x_2(t)$, generates a signal

$$y(t) = x_1(t) \cdot x_2(t)$$

- The elementary operations discussed so far are also carried out on discrete-time signals
- More complex operations operations are implemented by combining two or more elementary operations

Filtering

- Filtering is one of the most widely used complex signal processing operations
- The system implementing this operation is called a filter
- A filter passes certain frequency components without any distortion and blocks other frequency components

Filtering

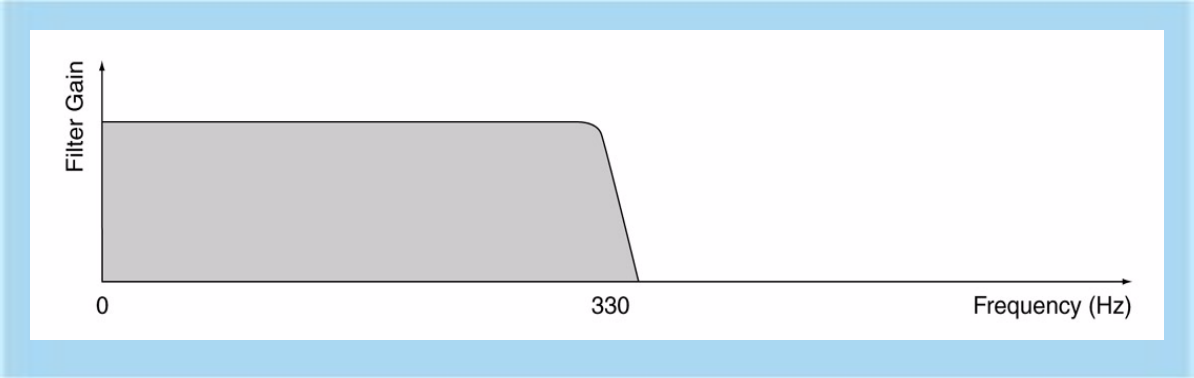
- The range of frequencies that is allowed to pass through the filter is called the passband, and the range of frequencies that is blocked by the filter is called the stopband
- In most cases, the filtering operation for analog signals is linear

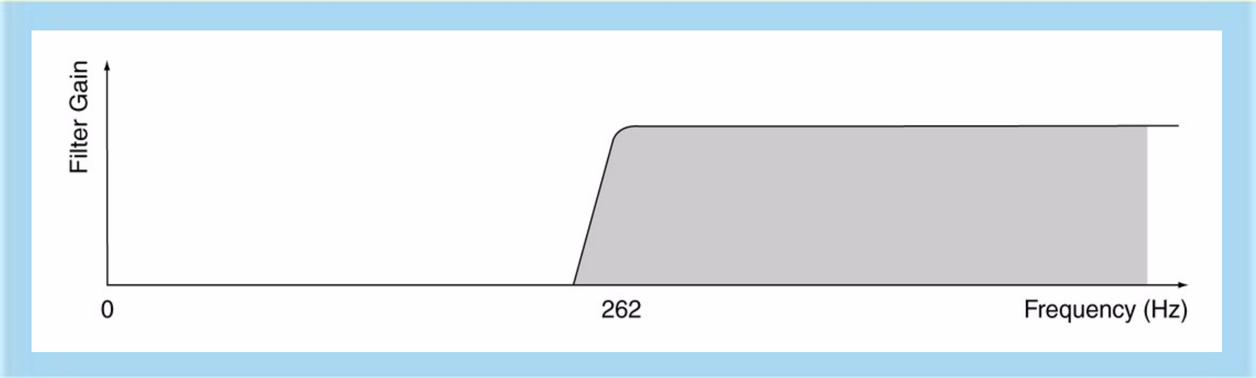
Filtering

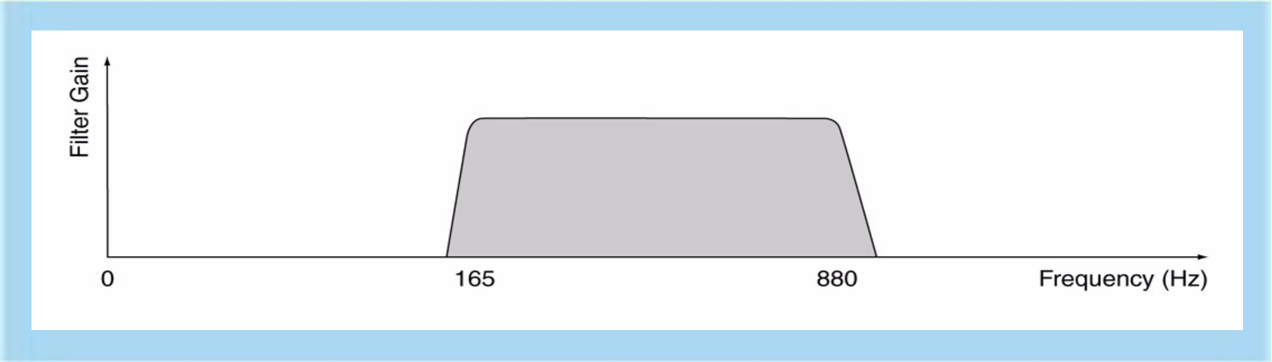
- A lowpass filter passes all low-frequency components below a certain specified frequency f_c, called the cutoff frequency, and blocks all high-frequency components above f_c
- A highpass filter passes all high-frequency components a certain cutoff frequency f_c and blocks all low-frequency components below

Filtering

- A bandpass filter passes all frequency components between 2 cutoff frequencies, f_{c1} and f_{c2}, where $f_{c1} < f_{c2}$, and blocks all frequency components below the frequency f_{c1} and above the frequency f_{c2}
- A bandstop filter blocks all frequency components between 2 cutoff frequencies, f_{c1} and f_{c2}, where $f_{c1} < f_{c2}$, and passes all frequency components below the frequency f_{c1} and above the frequency f_{c2}

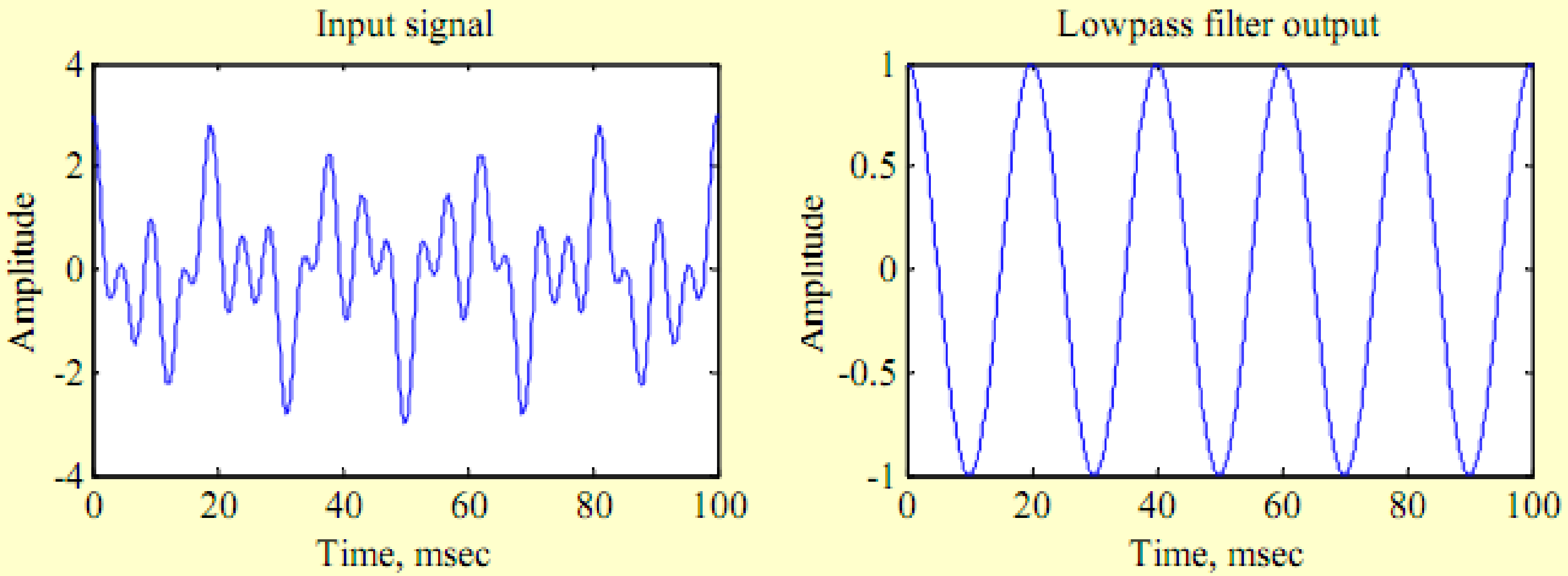

Filter Gain
0
330
Frequency (Hz)


Filter Gain
0
262
Frequency (Hz)


Filter Gain
0
165
880
Frequency (Hz)

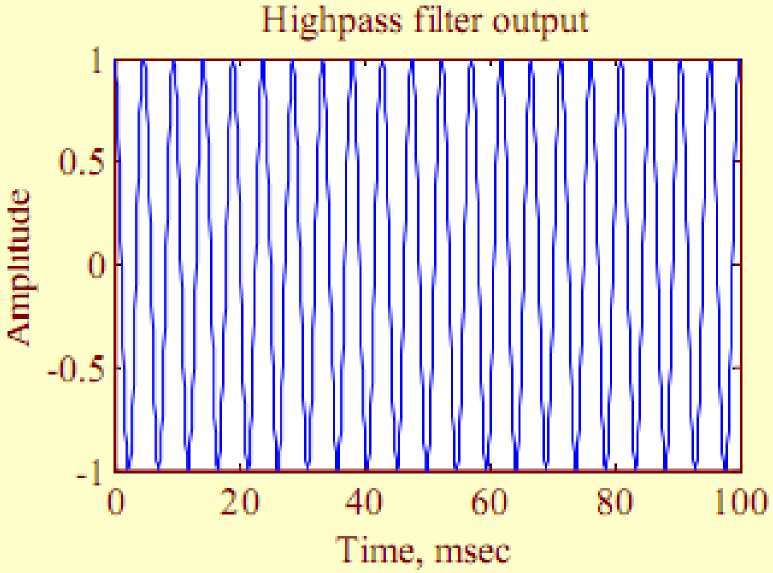
Filtering

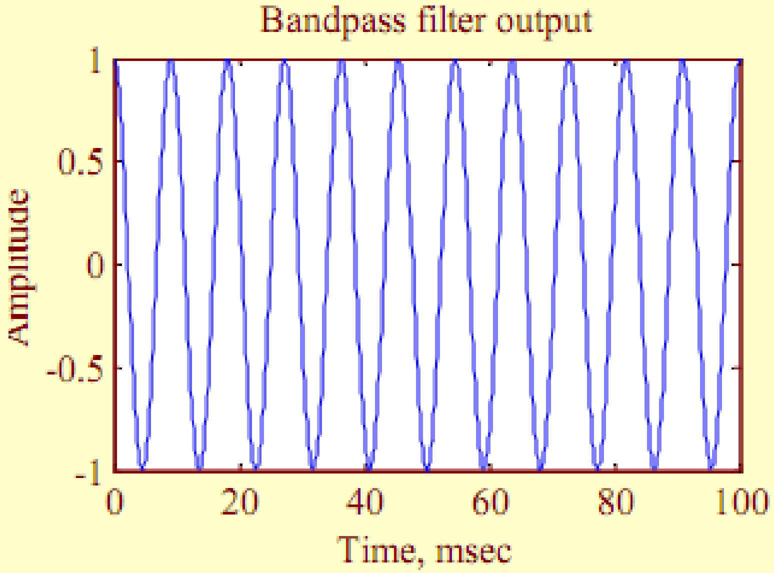
- Figures below illustrate the lowpass filtering of an input signal composed of 3 sinusoidal components of frequencies 50 Hz, 110 Hz, and 210 Hz



Filtering

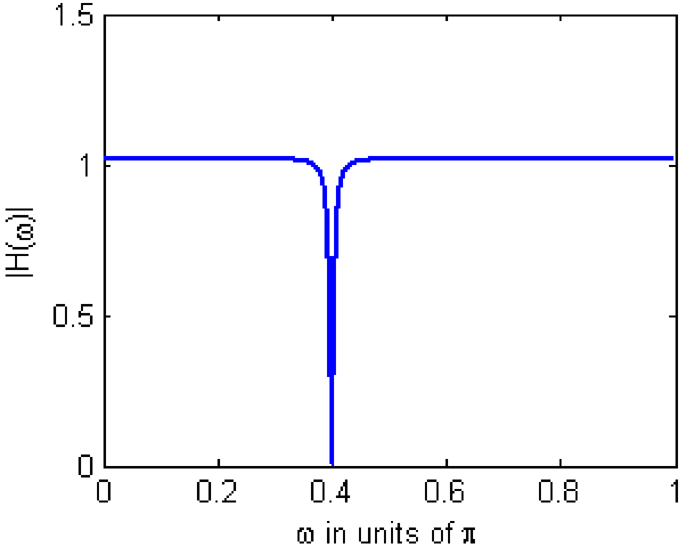
- Figures below illustrate highpass and bandpass filtering of the same input signal

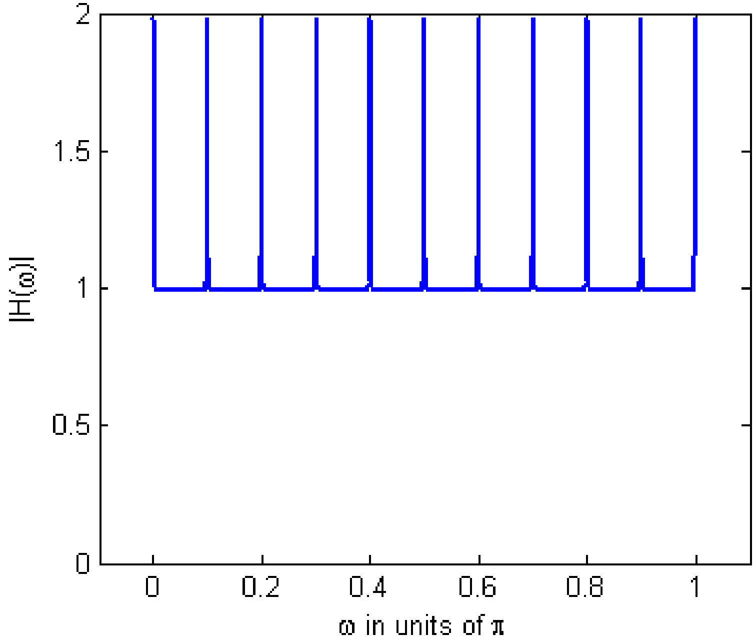


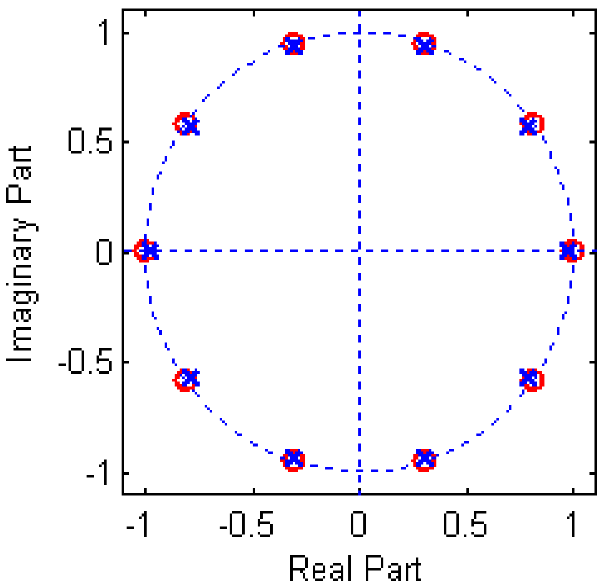


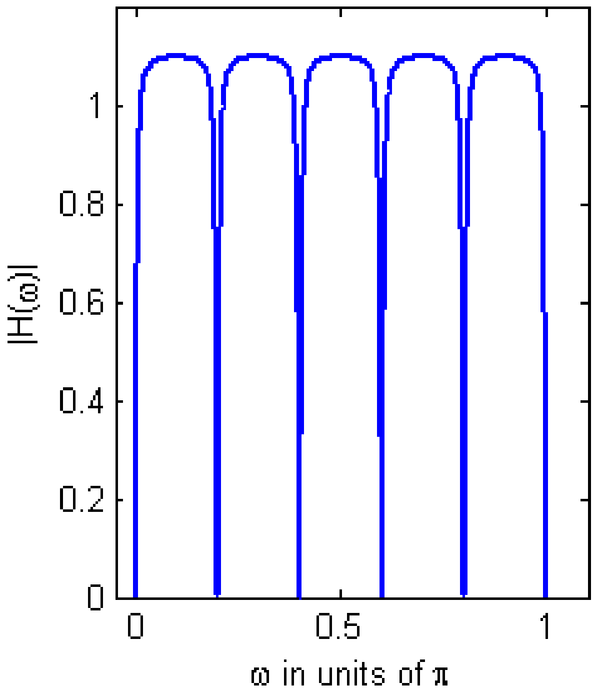
Filtering

- There are various other types of filters
- A filter blocking a single frequency component is called a notch filter
- A multiband filter has more than one passband and more than one stopband
- A comb filter blocks frequencies that are integral multiples of a low frequency


|H(ω)|
ω in units of π


|H(ω)|
ω in units of π


Imaginary Part
Real Part


|H(ω)|
ω in units of π

§ 1.2.4 Amplitude Modulation

Modulation and Demodulation

- For efficient transmission of a low-frequency signal over a channel, it is necessary to transform the signal to a high-frequency signal by means of a modulation operation
- At the receiving end, the modulated high-frequency signal is demodulated to extract the desired low-frequency signal

Modulation and Demodulation

- There are 4 major types of modulation of analog signals:

  (1) Amplitude modulation

  (2) Frequency modulation

  (3) Phase modulation

  (4) Pulse amplitude modulation

Multiplexing and Demultiplexing

- For an efficient utilization of a wideband transmission channel, many narrow-bandwidth low-frequency signals are combined for a composite wideband signal that is transmitted as a single signal
- The process of combining the low-frequency signals is called multiplexing

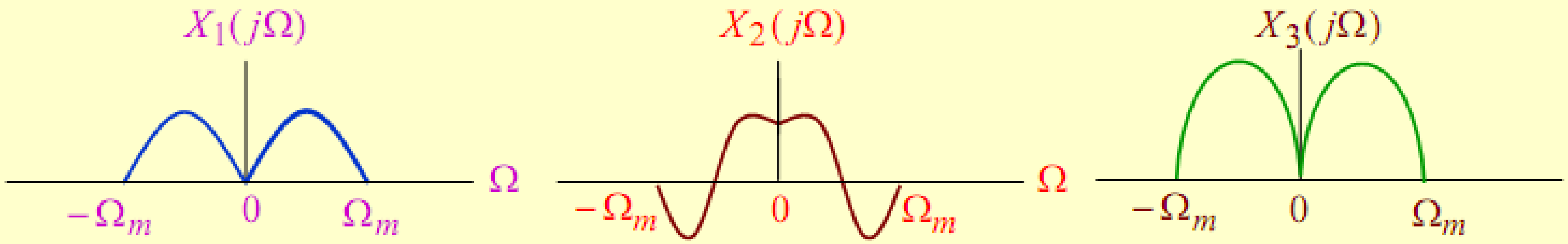
Multiplexing and Demultiplexing

- Multiplexing is implemented to ensure that a replica of each of the original narrow-bandwidth low-frequency signal can be recovered at the receiving end
- The recovery process of the low-frequency signals is called demultiplexing

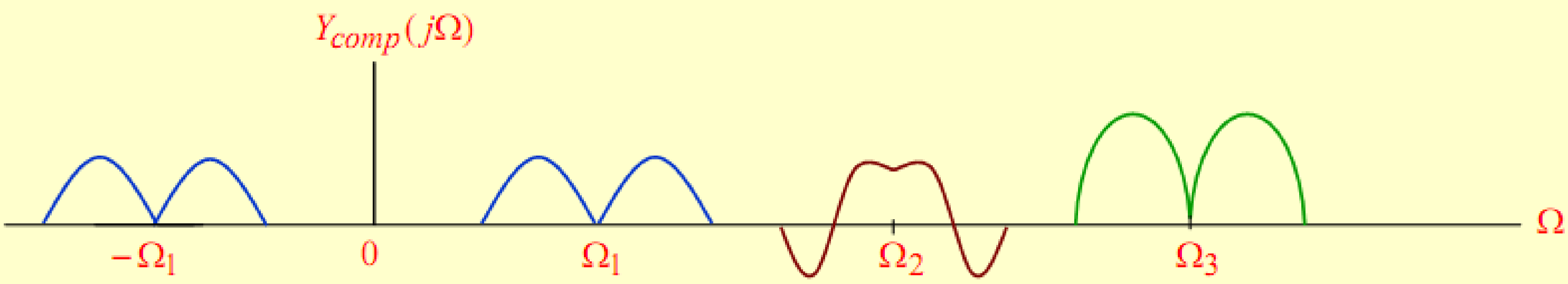
Multiplexing and Demultiplexing

- One method of combining different voice signals in a telephone communication system is the frequency-division multiplexing (FDM) scheme
- Here, each voice signal, typically bandlimited to a low-frequency band of width Ω_m , is frequency-translated into a higher frequency band using the amplitude modulation method

Multiplexing and Demultiplexing



Spectra of the low–frequency signals



Spectra of the modulated composite signal

Multiplexing and Demultiplexing

- At the receiving end, the composite baseband signal is first recovered from the FDM signal by demodulation
- Then each individual frequency-translated signal is demultiplexed by passing the composite signal through a bank of bandpass filters

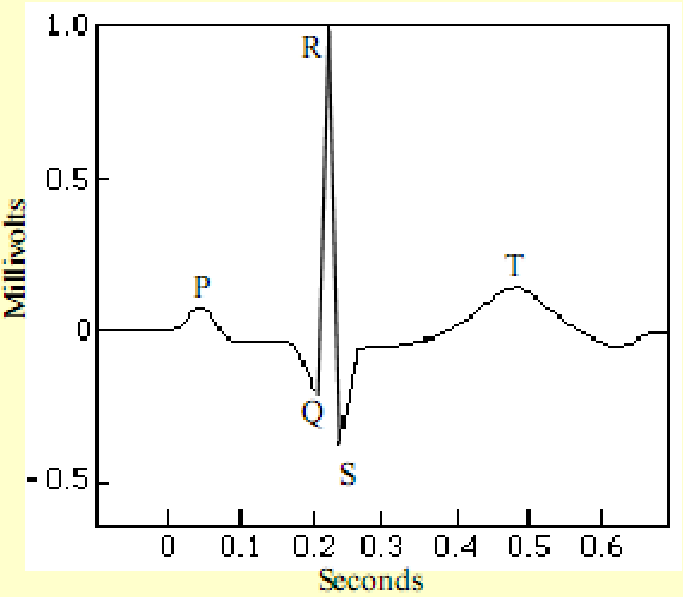
§ 1.3 Examples of Typical Signals

Examples of Typical Signals

- **Electrocardiography (ECG) Signal** - Represents the electrical activity of the heart
- A typical ECG signal is shown below

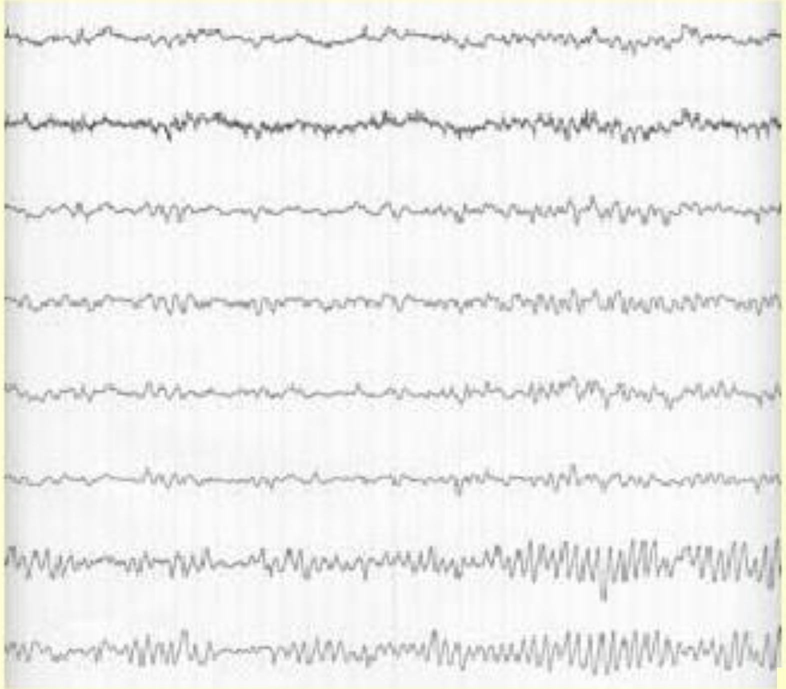
Examples of Typical Signals

- The ECG trace is a periodic waveform
- One period of the waveform shown below represents one cycle of the blood transfer process from the heart to the arteries



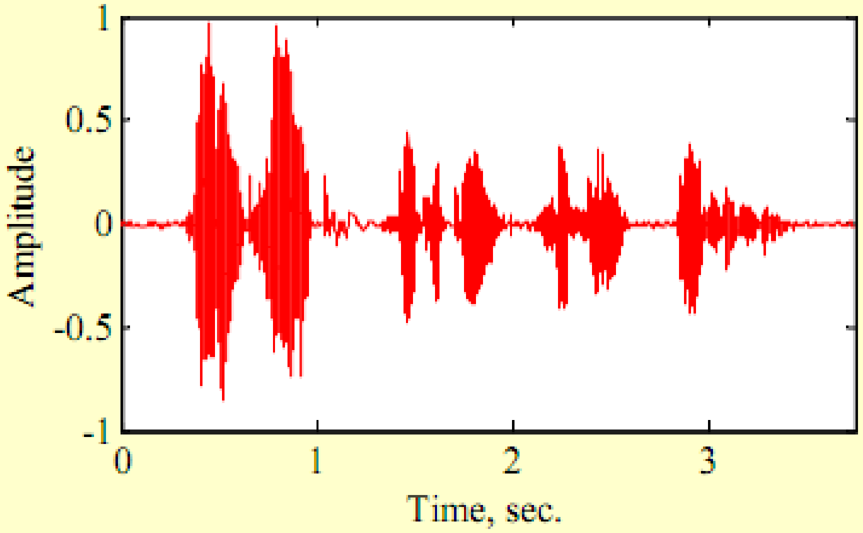
Examples of Typical Signals

- **Electroencephalogram (EEG) Signals** - Represent the electrical activity caused by the random firings of billions of neurons in the brain



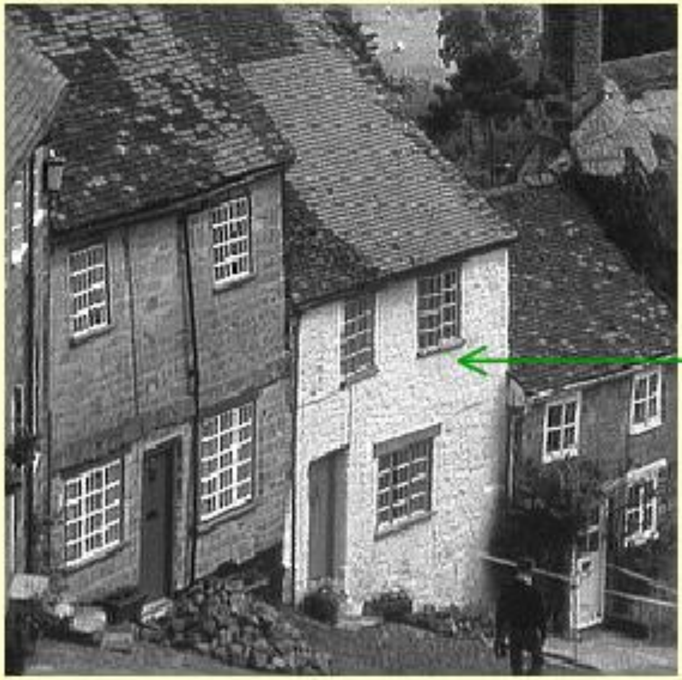
Examples of Typical Signals

- **Speech and music signals** - Represent air pressure as a function of time at a point in space
- Waveform of the speech signal "I like digital signal processing" is shown below



Examples of Typical Signals

- Black-and-white picture - Represents light intensity as a function of two spatial coordinates



$I(x,y)$

Examples of Typical Signals

- Video signals - Consists of a sequence of images, called frames, and is a function of 3 variables: 2 spatial coordinates and time

Frame 1

Frame 3

Frame 5

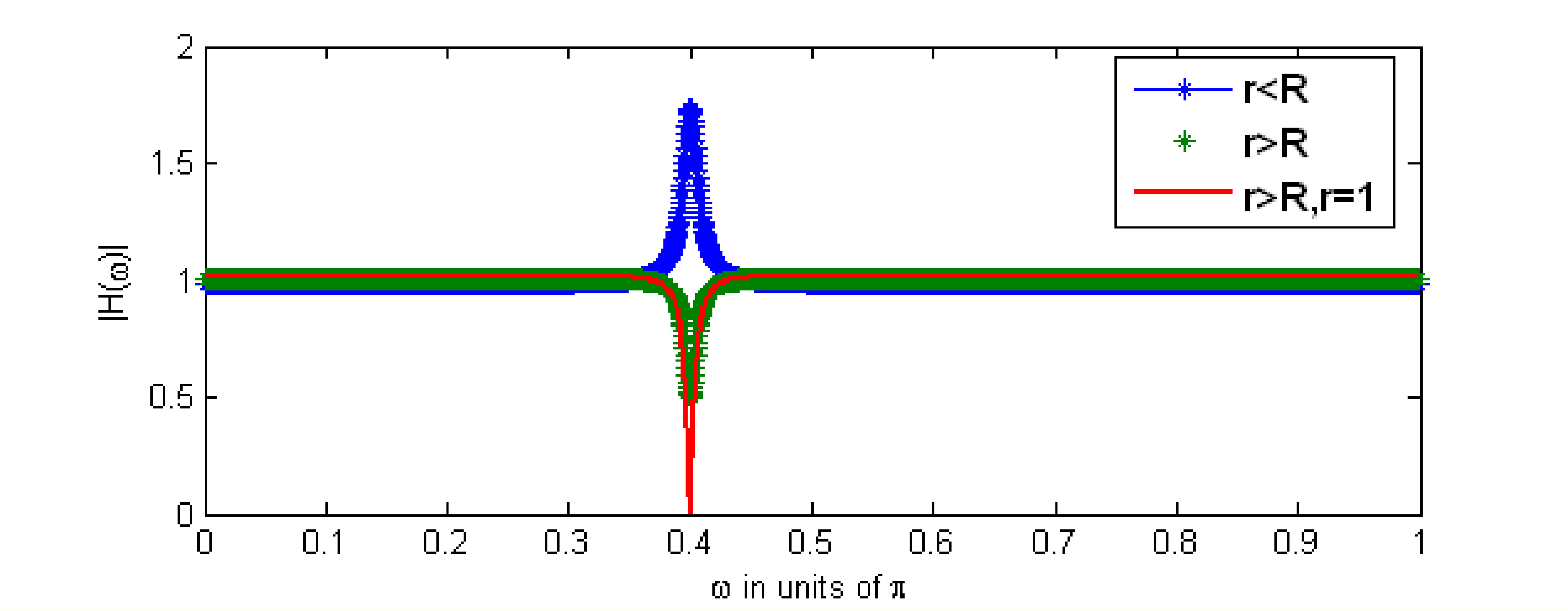
Video

Click on the video

§ 1.4 Typical Signal Processing Applications

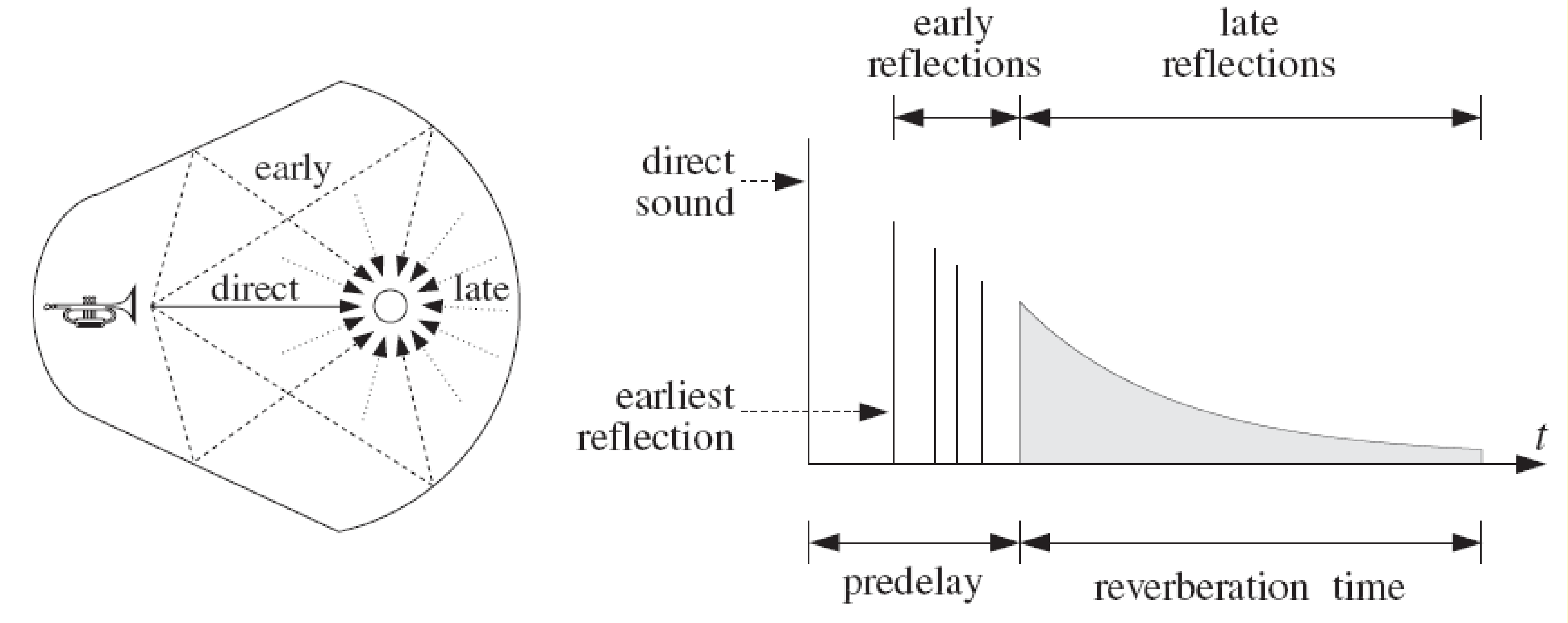
§ 1.4.1 Sound Recording Applications

Examples of Equalizers :

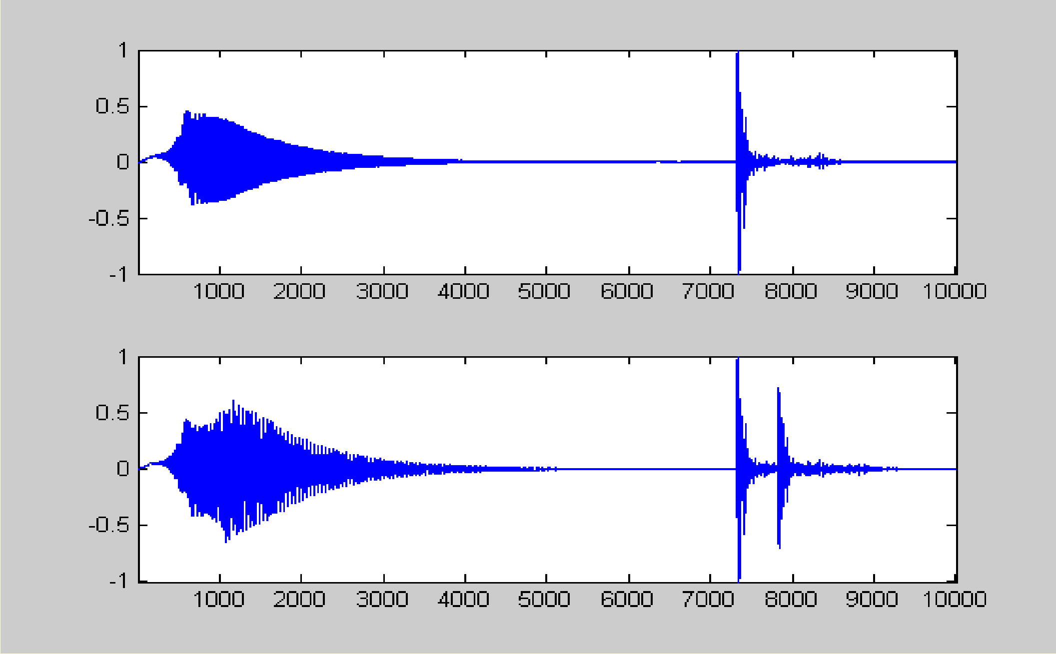


Delay and Reverberation Systems

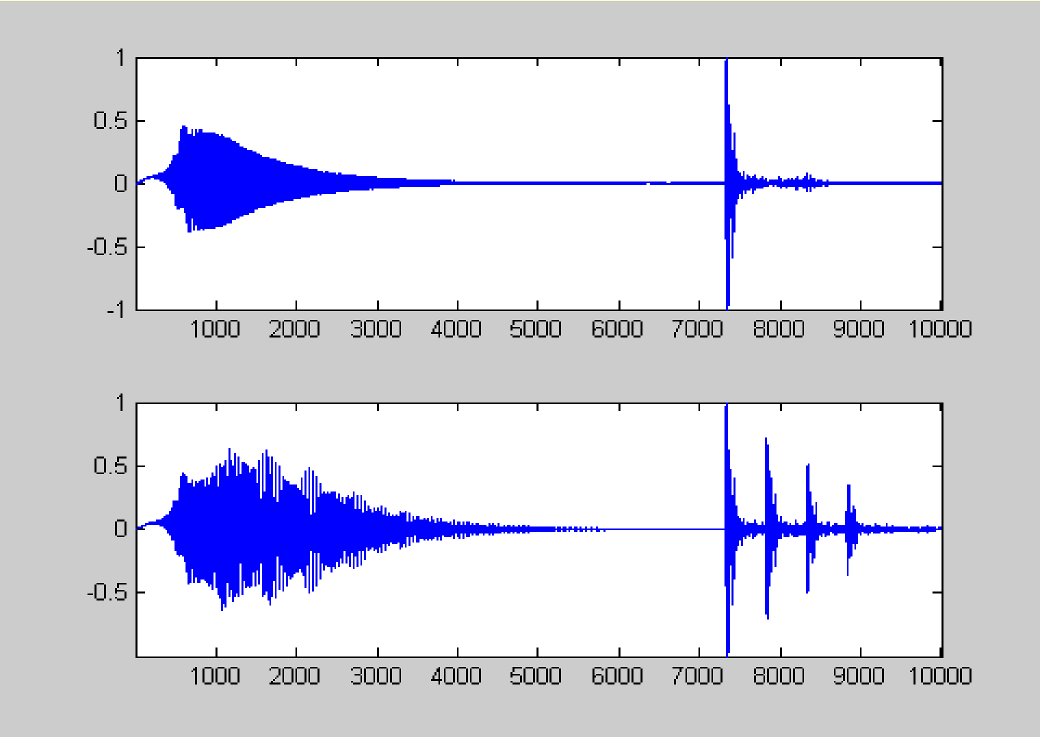
Reverberation impulse response of a listening space:



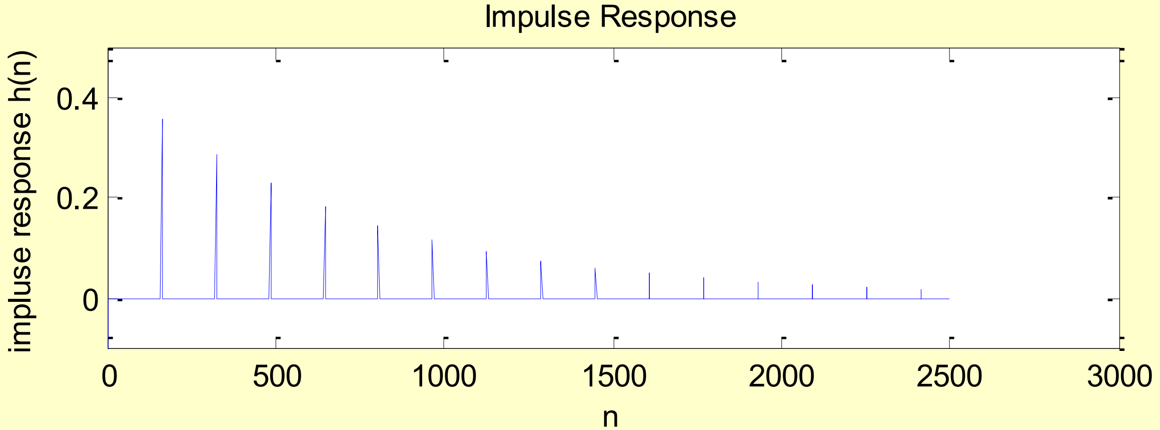
Single echo:



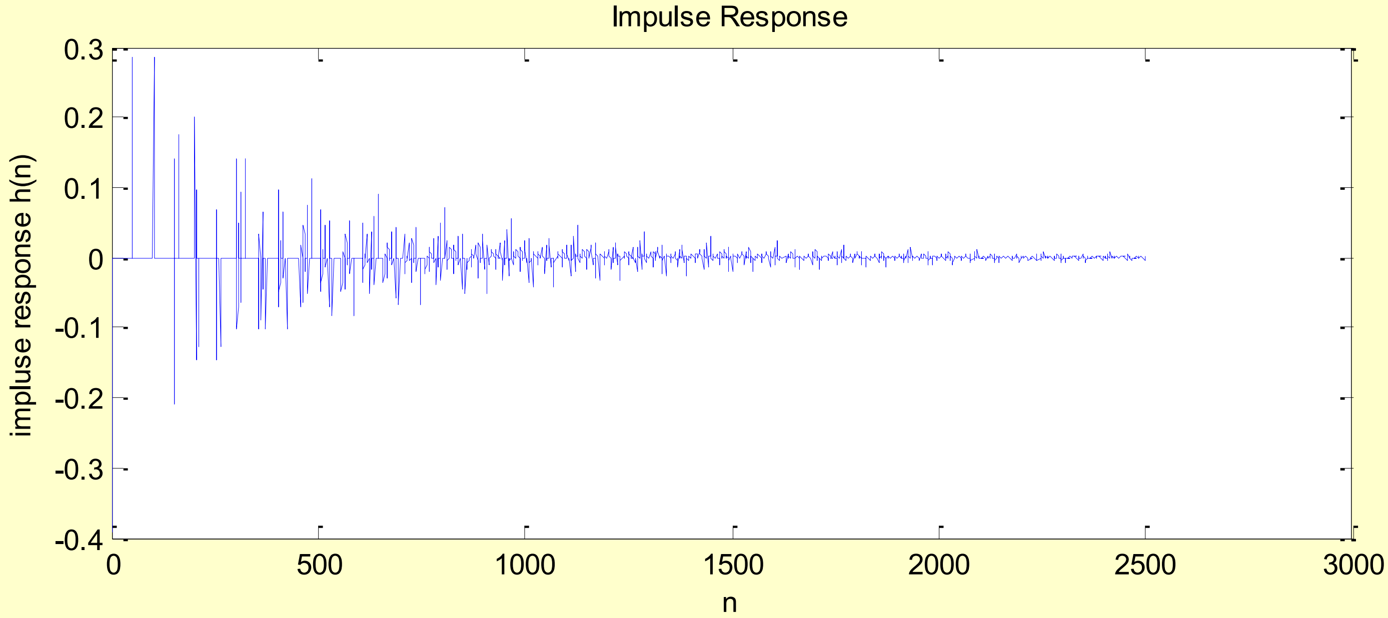
Successive echoes:



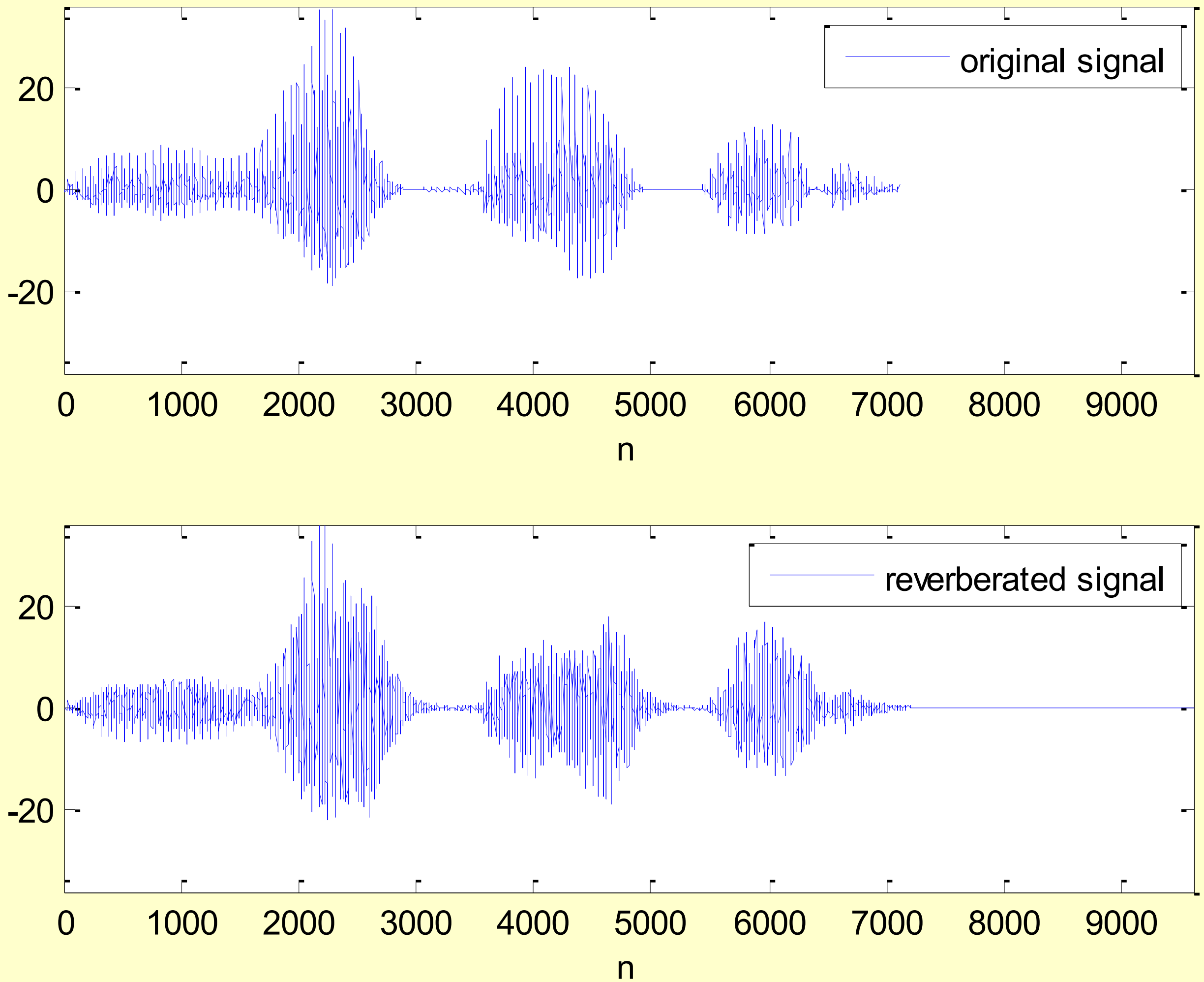
Single allpass reverberator：



Cascade allpass reverberator:



Schroeder's reverb processor.



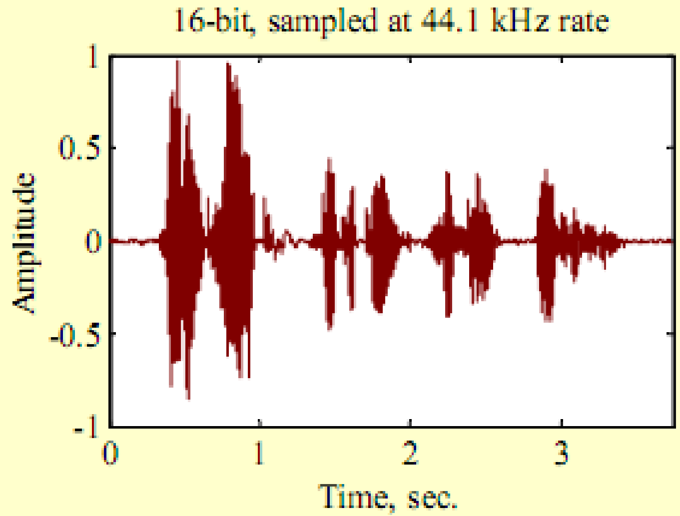
Signal Coding & Compression

- Concerned with efficient digital representation of audio or visual signal for storage and transmission to provide maximum quality to the listener or viewer

Signal Compression Example

- Original speech

  Data size 330,780 bytes



- Compressed speech (GSM 6.10)

  - Sampled at 22.050 kHz, Data size 16,896 bytes

- Compressed speech (Lernout & Hauspie CELP 4.8kbit/s)

  Sampled at 8 kHz, Data size 2,302 bytes

Signal Compression Example

- Original music

  Audio Format: PCM 16.000 kHz, 16 Bit
  (Data size 66206 bytes)

- Compressed music

  Audio Format: GSM 6.10, 22.05 kHz
  (Data size 9295 bytes)

Courtesy: Dr. A. Spanias

Signal Compression Example

Original Lena
8 bits per pixel

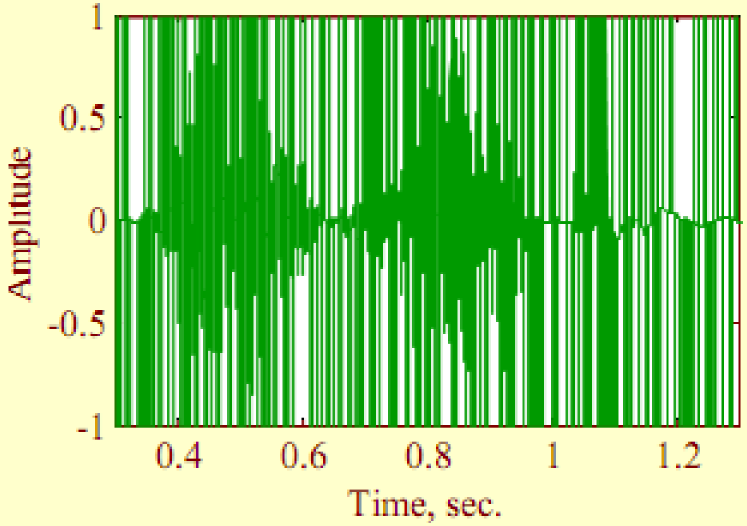
Compressed Image
Average bit rate - 0.5 bits per pixel

Signal Enhancement

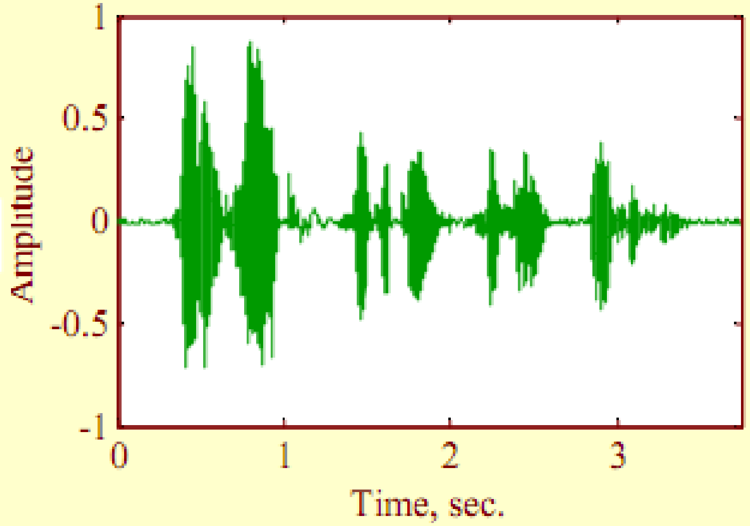
- Purpose: To emphasize specific signal features to provide maximum quality to the listener or viewer
- For speech signals, algorithms include removal of background noise or interference
- For image or video signals, algorithms include contrast enhancement, sharpening and noise removal

Signal Enhancement Example

- Noisy speech signal (10% impulse noise)

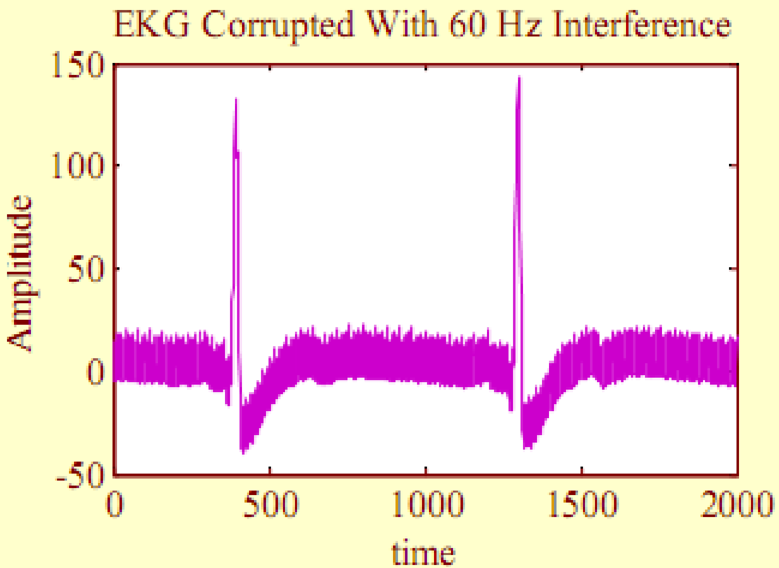


- Noise removed speech

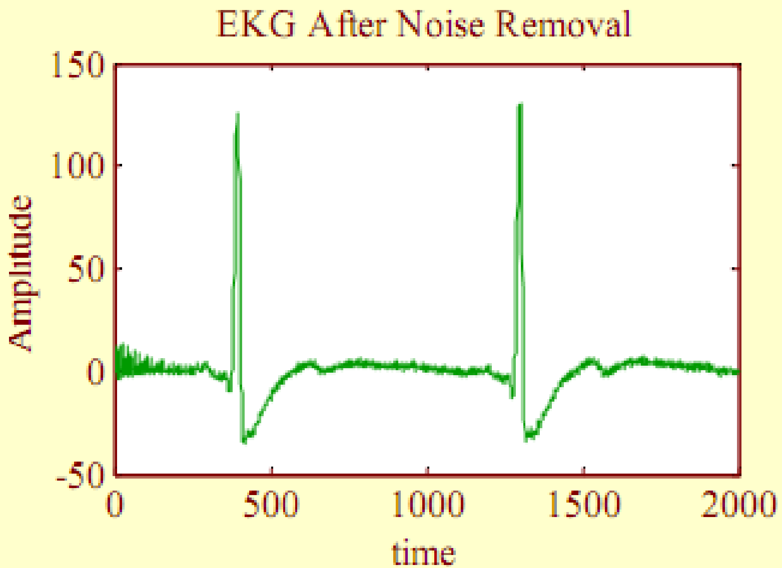


Signal Enhancement Example

EKG corrupted with 60 Hz interference



EKG after filtering with a notch filter



Signal Enhancement Example

- Original image and its contrast enhanced version

Original

Enhanced

Signal Enhancement Example

- Original image and its contrast enhanced version

Original

Enhanced

Signal Enhancement Example

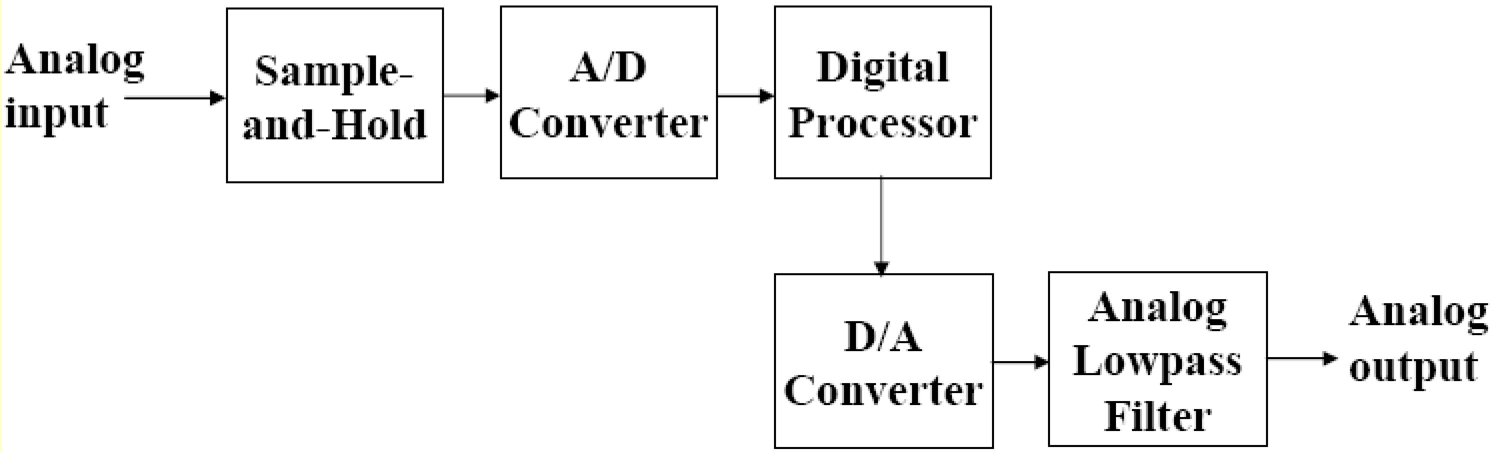
- Noise corrupted image and its noise-removed version

20% pixels corrupted with additive impulse noise

Noise-removed version

§1.5 Why Digital Signal Processing?

Digital processing of an analog signal are shown below



Scheme for the digital processing of an analog signal

Advantages of DSP

- Absence of drift in the filter characteristics
  - Processing characteristics are fixed, e.g. by binary coefficients stored in memories
  - Thus, they are independent of the external environment and of parameters such as temperature
  - Aging has no effect

Advantages of DSP

- Improved quality level
  - Quality of processing limited only by economic considerations
  - Arbitrarily low degradations achieved with desired quality by increasing the number of bits in data/coefficient representation
  - An increase of 1 bit in the representation results in a 6 dB improvement in the SNR

Advantages of DSP

- Reproducibility
  - Component tolerances do not affect system performance with correct operation
  - No adjustments necessary during fabrication
  - No realignment needed over lifetime of equipment

Advantages of DSP

- Ease of new function development
  - Easy to develop and implement adaptive filters, programmable filters and complementary filters
  - Illustrates flexibility of digital techniques

Advantages of DSP

- Multiplexing
  - Same equipment can be shared between several signals, with obvious financial advantages for each function
- Modularity
  - Uses standard digital circuits for implementation

Advantages of DSP

- Total single chip implementation using VLSI technology
- No loading effect

Limitations of DSP

- Lesser Reliability
  - Digital systems are active devices, and thus use more power and are less reliable
- Limited Frequency Range of Operation
  - Frequency range technologically limited to values corresponding to maximum computing capacities that can be developed and exploited

Limitations of DSP

- Additional Complexity in the Processing of Analog Signals
  - A/D and D/A converters must be introduced adding complexity to overall system