Digital Signal Processing

Introduction

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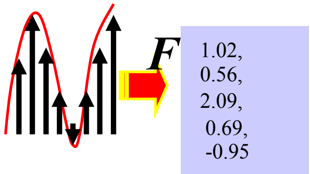
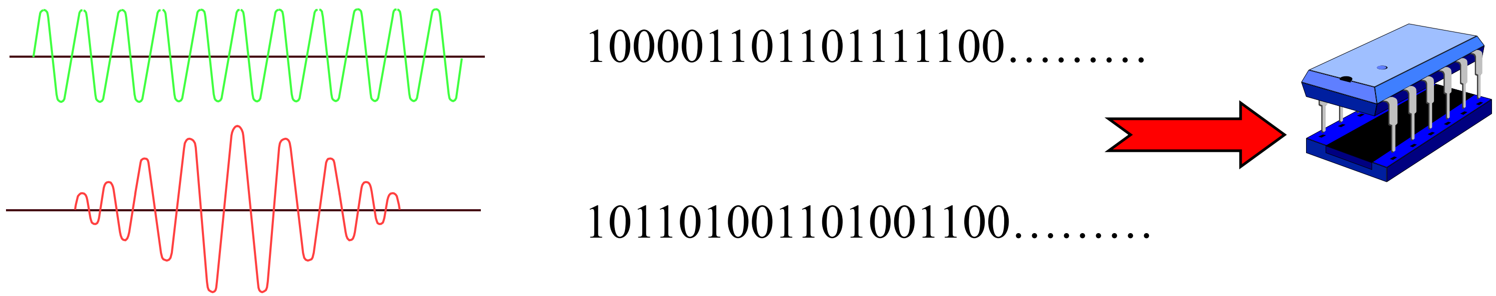
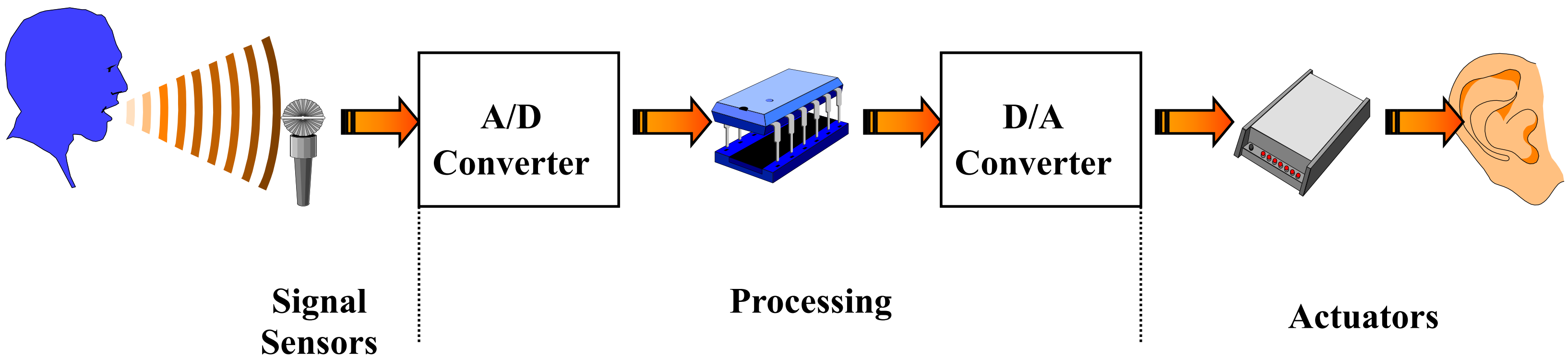


What is Digital Signal Processing ?

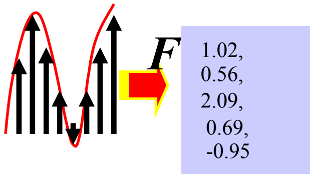


The representation of signals in terms of sequences of digital numbers and the use of digital computers to process these signals to either analyze or modify the original signal is termed as *Digital Signal Processing* and in short *DSP*.

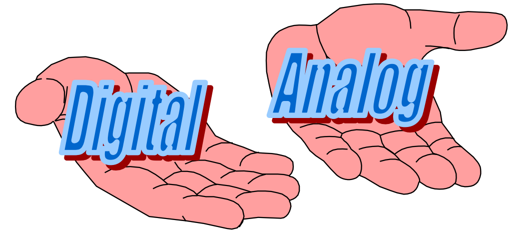
Digital Signal Processing System



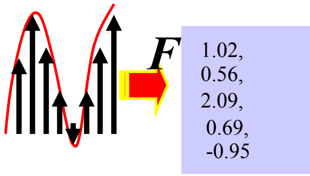
Analog signals are the most popular signals used for DSP, which are discretized by sampling at regular intervals and converted into a digital format for processing.



Advantages of DSP

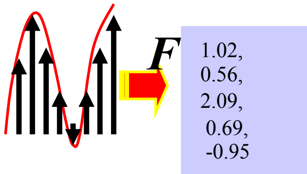


- Guaranteed Accuracy (determined by the number of bits)
- Superior Performance (Than analog signal processing)
- Perfect Reproducibility (no variations due to component tolerances)
- No Drift in performance with temperature & age
- Greater Flexibility (wider applications with minimal changes in hardware)
- Immunity from Noise

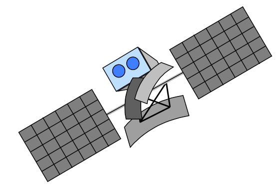


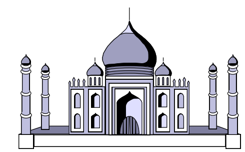
Disadvantages of DSP

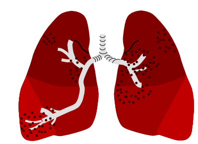
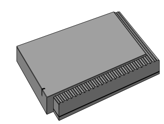
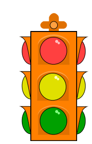
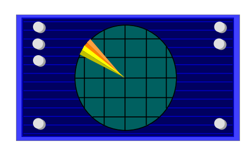
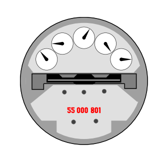
- Speed : The speed at which DSP takes place is still in the 100MHz range which means that signals having a very large frequency component in them are not suitable for DSP applications.
- Finite Word-length : In some real time simulations, cost considerations limit the DSP implementation with fewer number of bits which may create degradation in system performance.



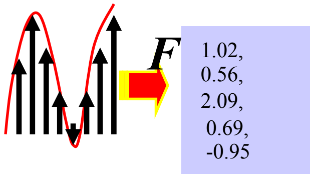
Applications of DSP





- Communication Systems
- Speech & Music Processing.
- Image Processing
- Medical Imaging
- Biomedical Signal Processing
- High speed Modems
- Closed Loop Control systems
- Radar/Sonar signal analysis
- Real Time Measurement & Instrumentation



Applications contd..

DSP

- **SPACE**
  - Space photograph enhancement
  - Data compression
  - Intelligent sensory analysis by remote space probes
- **MEDICAL**
  - Diagnostic Imaging (CT, MRI, ultrasound)
  - Electrocardiogram analysis
  - Medical image storage/retireval
- **COMMERCIAL**
  - Image and sound compression
  - Movie special effects
  - Video conference calling
- **TELEPHONE**
  - Voice and data compression
  - Echo reduction
  - Signal multiplexing
  - Filtering
- **MILITARY**
  - Radar
  - Sonar
  - Ordinance guidance
  - Secure communication
- **INDUSTRIAL**
  - Oil and mineral prospecting
  - Pocess monitoring & control
  - Nondestructive testing
  - CAD and desing tools
- **SCIENTIFIC**
  - Earthquake recording and analysis
  - Data acquisition
  - Spectral analysis
  - Simulation and modeling

Applications contd..

- Given the advent of DSP chips, DSP technology can now be found in such devices as mobile phones, multimedia computers, video recorders, CD players, hard disc drive controllers and modems. It will soon even replace analog circuitry in TV sets and telephones. Most of these practical applications exploit two key attributes of DSP technology: signal compression/ decompression and real-time operation.

- Signal compression and decompression is used in a variety of applications. In CD systems, for example, the music recorded on the CD is in a compressed form (to increase storage capacity). It must be decompressed in order for the recorded signal to be reproduced.

Applications contd..

DSP technology enables the signal to be compressed and decompressed resulting in a cleaner, crisper signal.

• Signal compression is also used in digital cellular phones to allow a greater number of calls to be handled simultaneously within each local "cell". This compression technology allows people not only to talk to one another by telephone but also to see one another on the screens of their PCs, using small video cameras mounted on the computer monitors, with only a conventional telephone line linking them together.

Applications contd..

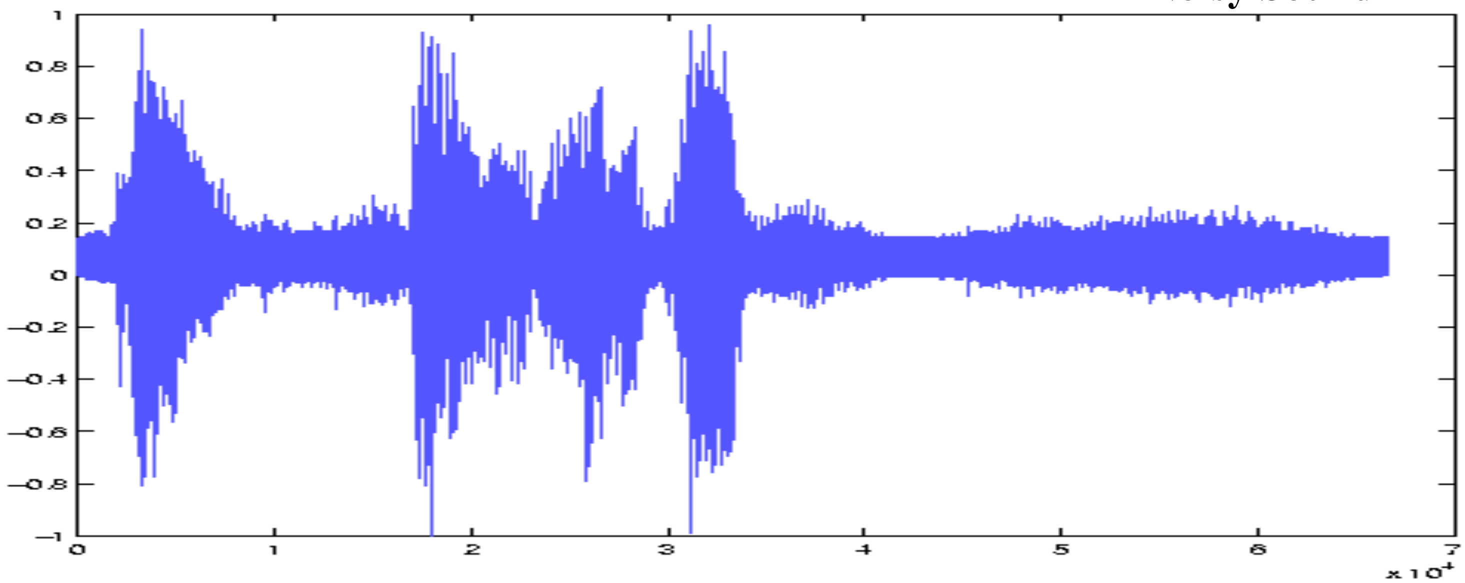
- The architecture of a DSP chip is designed to carry out complex mathematical operations incredibly fast, processing up to tens of millions of samples per second, to provide *real-time* performance. The real-time performance results from the ability to process a signal "live" as it is sampled and then output the processed signal, for example to a loudspeaker or video display. Most of the practical examples of DSP applications, such as hard disc drives and mobile phones, demand real-time operation.

Example

Application of DSP to recover damaged sound tracks :

Sampled Voice of *Darth Vader* from '*Empire Strikes Back'* Saying " *Don't Fail me again* ".

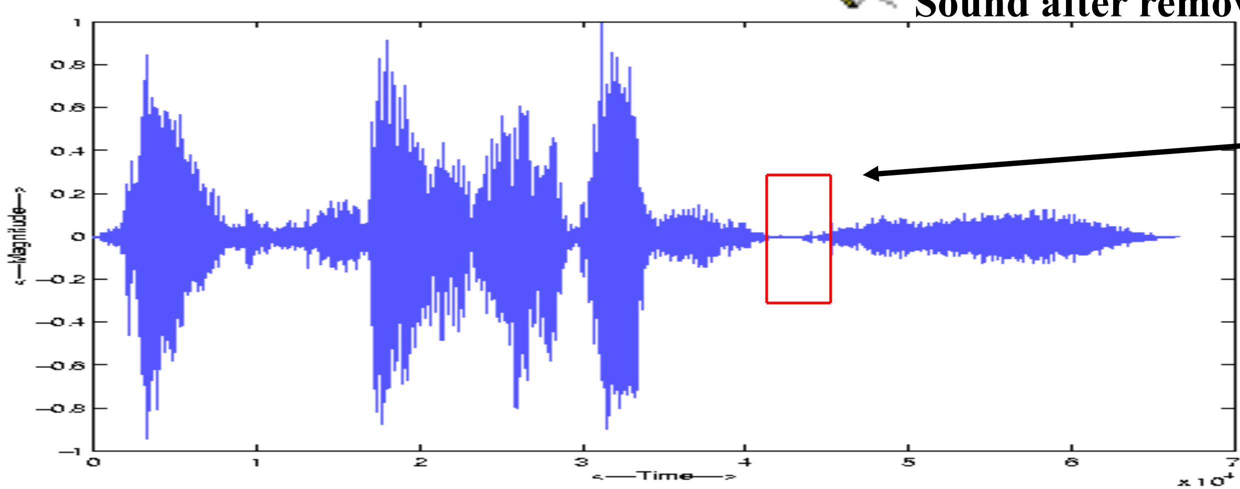
Noisy Sound



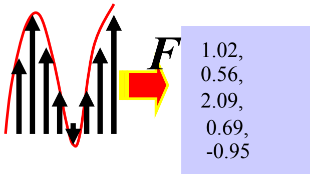
Example Contd..

Typically Sounds recorded prior to 1980 were all in analog format, because of which, due to age and wear of the recorded medium, there is a distinguishable loss in quality of sound. Now in-order to recover back as much as the original sound as possible we convert the noisy sound into a digital format and apply DSP techniques for removal of noise. After Removal of Noise ;

Sound after removal of Noise



Here we could clearly notice the effect of filtering

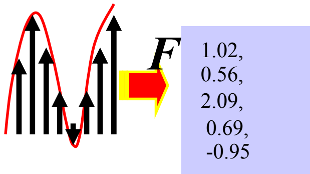


Example Contd..

Another Sound clip from the movie *Return of the Jedi* :

Here we can clearly hear the noise which has crept in to the original soundtrack. Now we process this signal using DSP techniques to get rid of the noise.

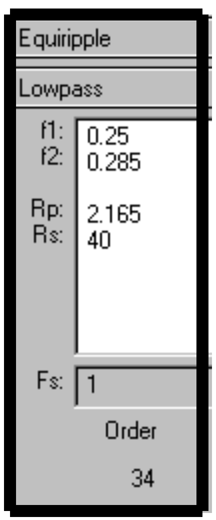
We can clearly notice a marked change in quality of the sound after processing the signal.



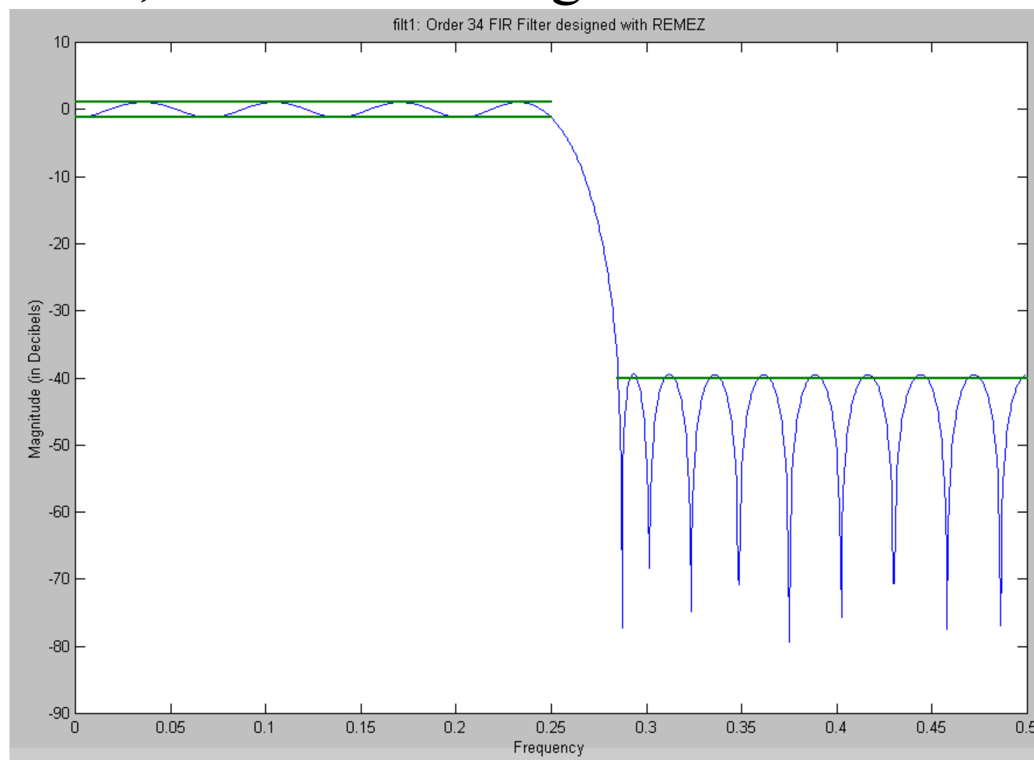
How are we Achieving this ?

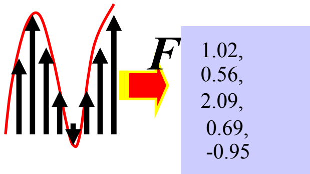
Most of the energy in Speech Signals lies in the frequency band of 0 - 3000 Hz. Utilizing this fact we design a digital filter to remove of all the high frequency components in the signal of our interest, thus eliminating the unwanted noise.

Filter Specification



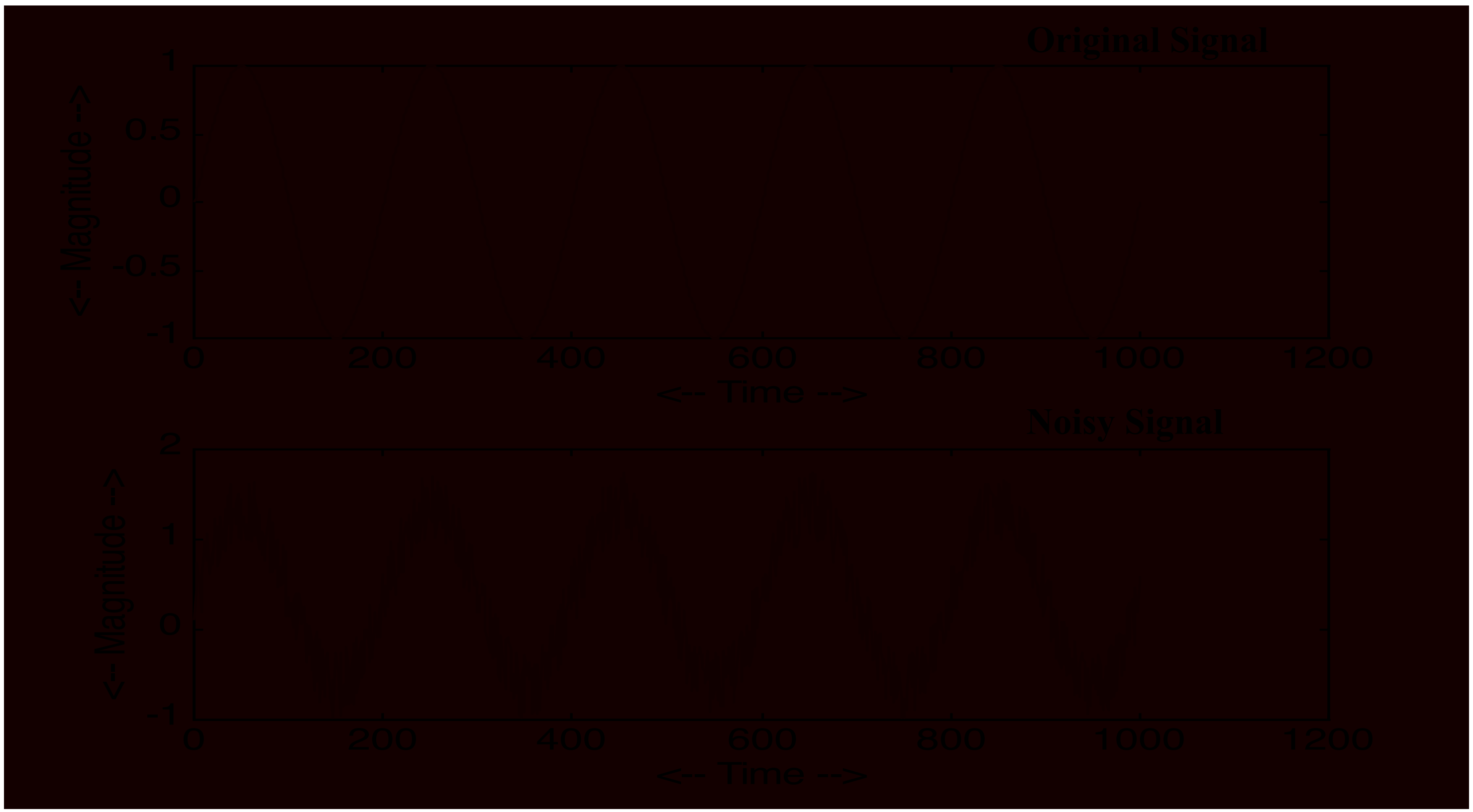
Filter





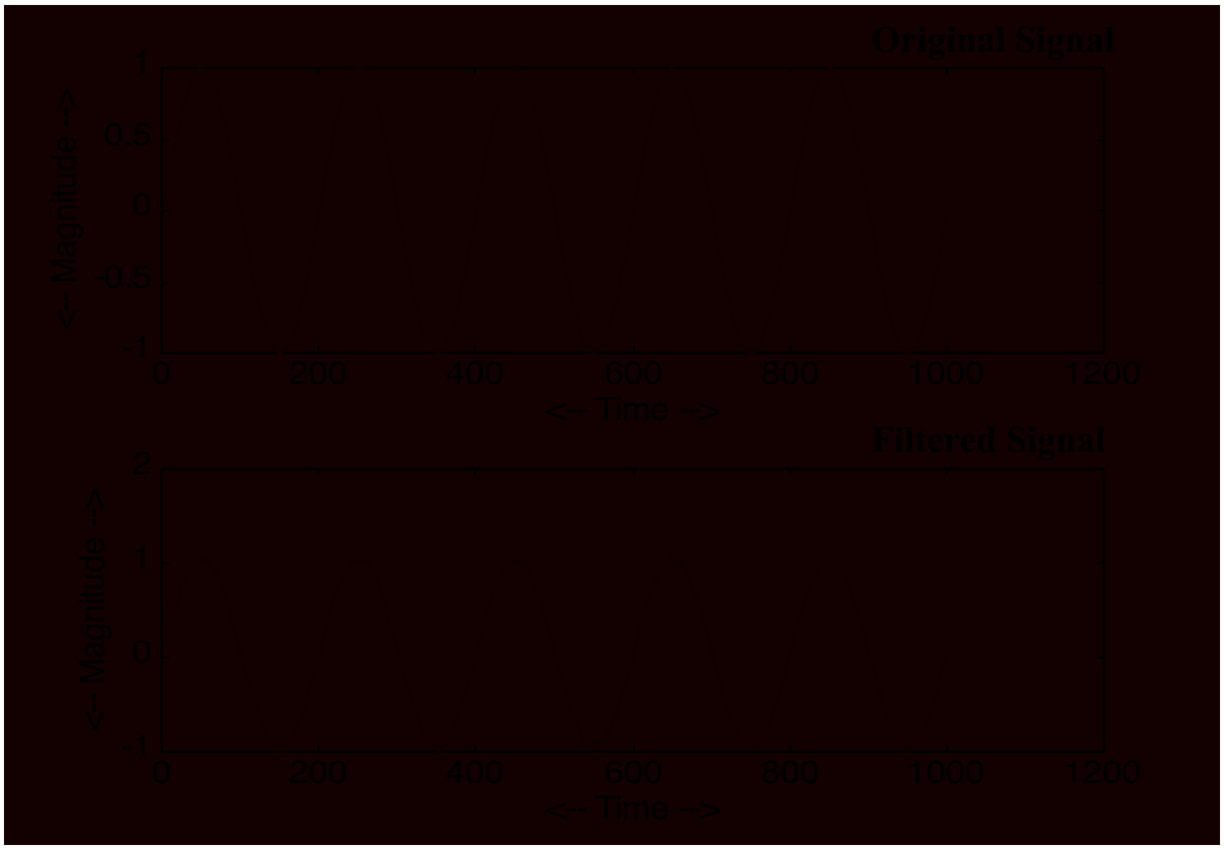
Another Example

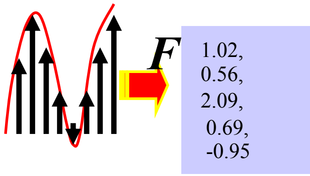
For Easy Visualization purposes let’s consider a simple sine wave corrupted with noise.
Lets consider a simple 50Hz signal.



Example Contd..

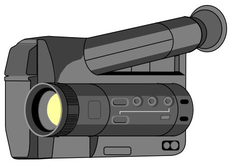
Passing the noisy signal through a Digital filter to remove the noisy portions of the signal.





An Example of DSP in Image Processing

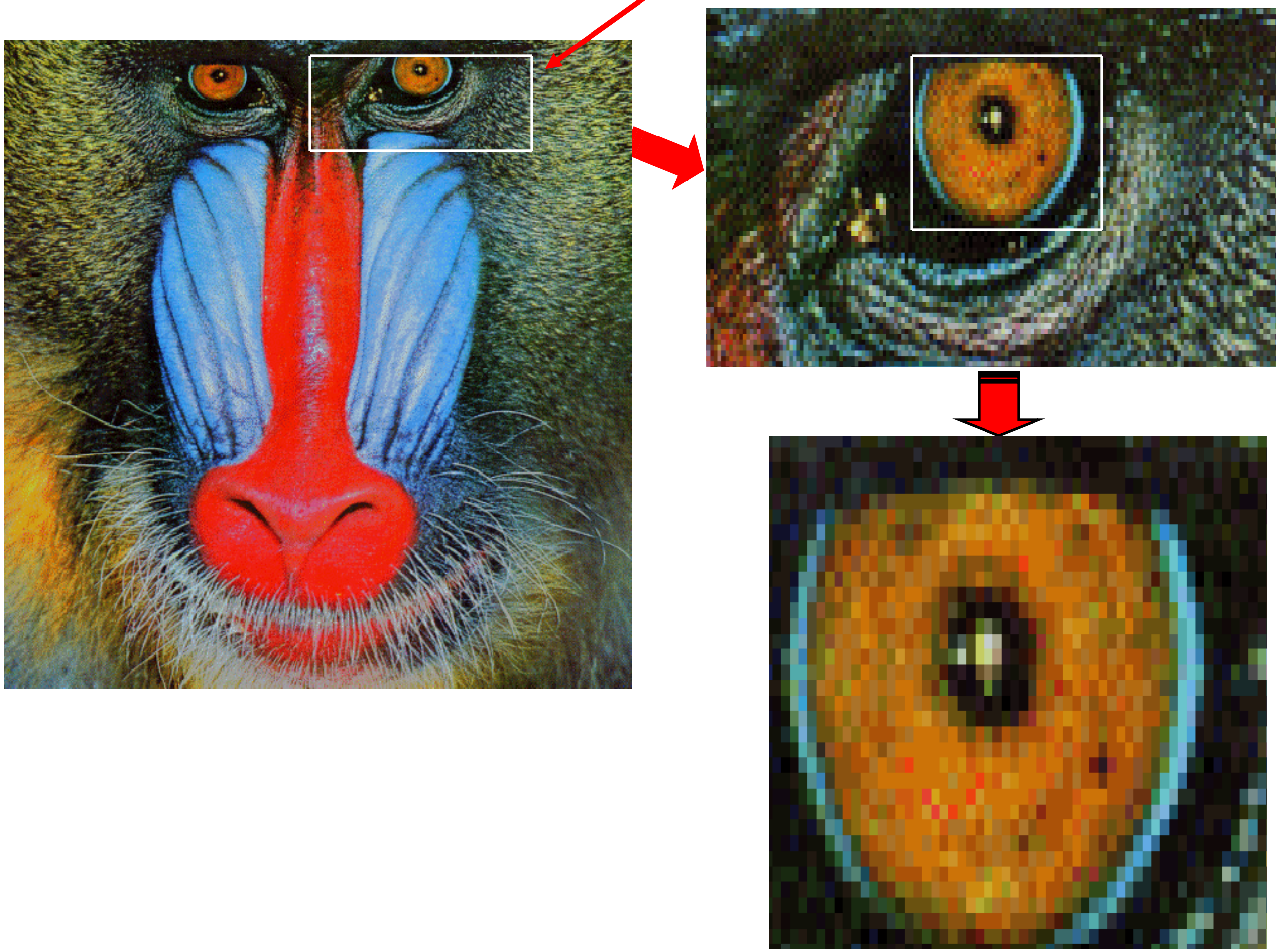
Use of DSP techniques for Zooming instead of using expensive & bulky Zoom Lens



Camcorders use DSP Techniques to achieve a greater zoom then that available on the camera lens. Although this technique deteriorates the image quality at very high zoom factors, but when compared to the cost & weight of lens required for that zoom factor, the cost & weight of the DSP chip used for this application is very marginal.

Image Processing Example

Zooming onto the area shown



In this technique the existing data is interpolated to get the new zoomed in version of the image. This example would compare with the process of up-sampling of 1-dimensional signals.

How is that done ?

Every Image is represented by points known as pixels, each pixel has three values ranging from 0 - 1 for each of the three primary colors “RGB”, for a color image and only by a single value for a Black & White image. Now when an image is zoomed in by a certain factor no new information is added in rather the existing information of the pixels present in the area of interest are replicated by the zoom factor to give us the zoomed version of the image.

Analog I/O interface for real-time DSP systems

The real-world signals are usually in analog form,
the interface Converts the analog signal into a digital one,
the following block Diagram shows the place of the interface in a real system.

Typical real-time DSP system

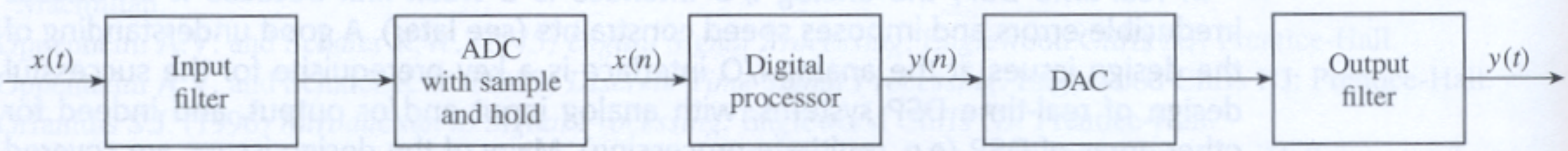
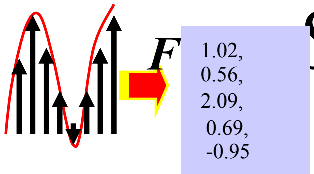


Figure 2.1 Block diagram of a simplified, generalized real-time digital signal processing system. In some applications, the input filter and the ADC, or the DAC and the output filter, will not be necessary.



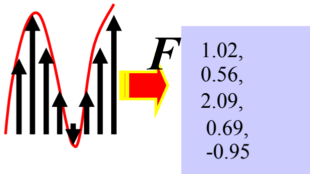
Steps involved,

•The analog input filter limits the bandwidth of the analog input signal.

•The ADC converts the analog input signal into digital form,   for wide band signals ADC is preceded by a sample and hold circuit.

•Digital processor is at the heart of the system, it implements various DSP algorithms. Motorola’s MC68000 , DSP56000 and Texas Instrument’s TMS320C50 are popular Digital processors.

•DAC converts the processed digital data into analog form, followed by an analog filter to give the final output.



ADC: Analog-to-digital conversion

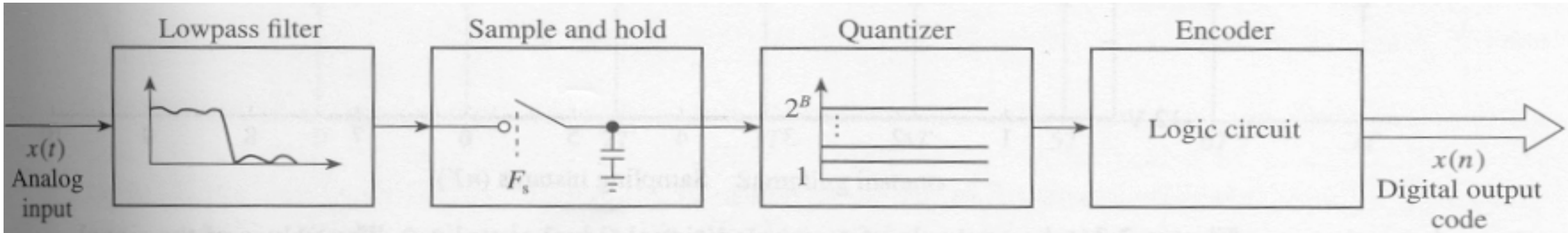
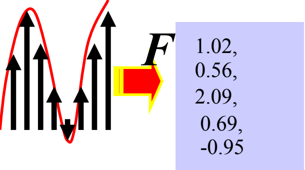


Figure 2.2 A pictorial representation of the analog-to-digital conversion process.

1. The analog signal is converted into a discrete-time continuous amplitude signal through sampling.
2. The amplitude of each signal sample is quantized into one Of the 2^B levels, where B is the number of bits used to represent A sample in the ADC.
3. The discrete amplitude levels are encoded into distinct binary Words of length B bits.



Sampling

The acquisition of an analog signal at discrete time Intervals is called sampling.

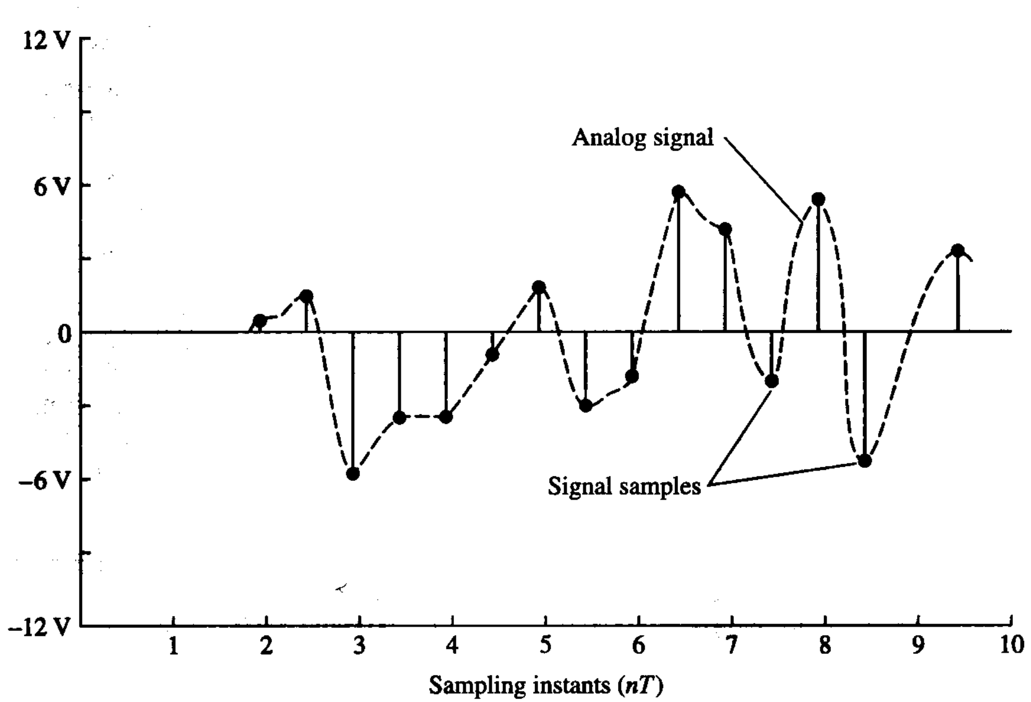
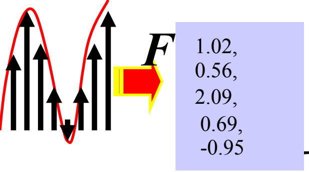


Figure 2.3 An example of a sampled signal (ideal sampling). The values of the signal samples are equal to those of the original analog signal at the sampling instants.



Sampling of lowpass signals

The analog signal should be sampled at the rate of at least $2f_{max}$(the highest frequency component in the signal is f_{max}), for its complete description.

Nyquist rate is the term used to describe the sampling frequency if it is close to $2f_{max}$.

Under Sampling

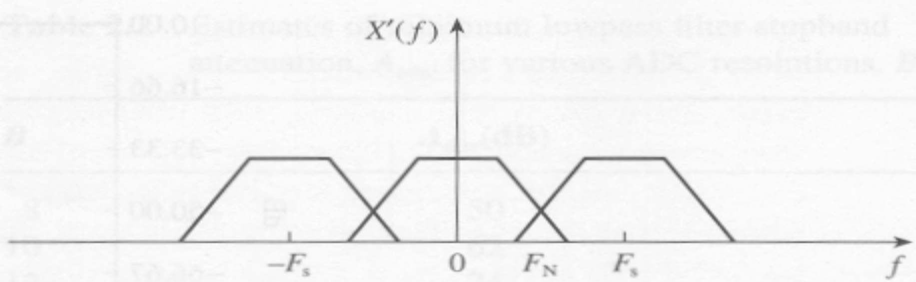


Figure 2.6 Spectrum of an undersampled signal, showing aliasing (foldover region). Signals in the foldover region are not recoverable. F_N is equal to half the sampling frequency and it is often called the Nyquist frequency. To recover all the components of a signal we must sample at a rate greater than (or equal to) twice the highest frequency component.

1.02,
0.56,
2.09,
0.69,
-0.95

Over-sampling

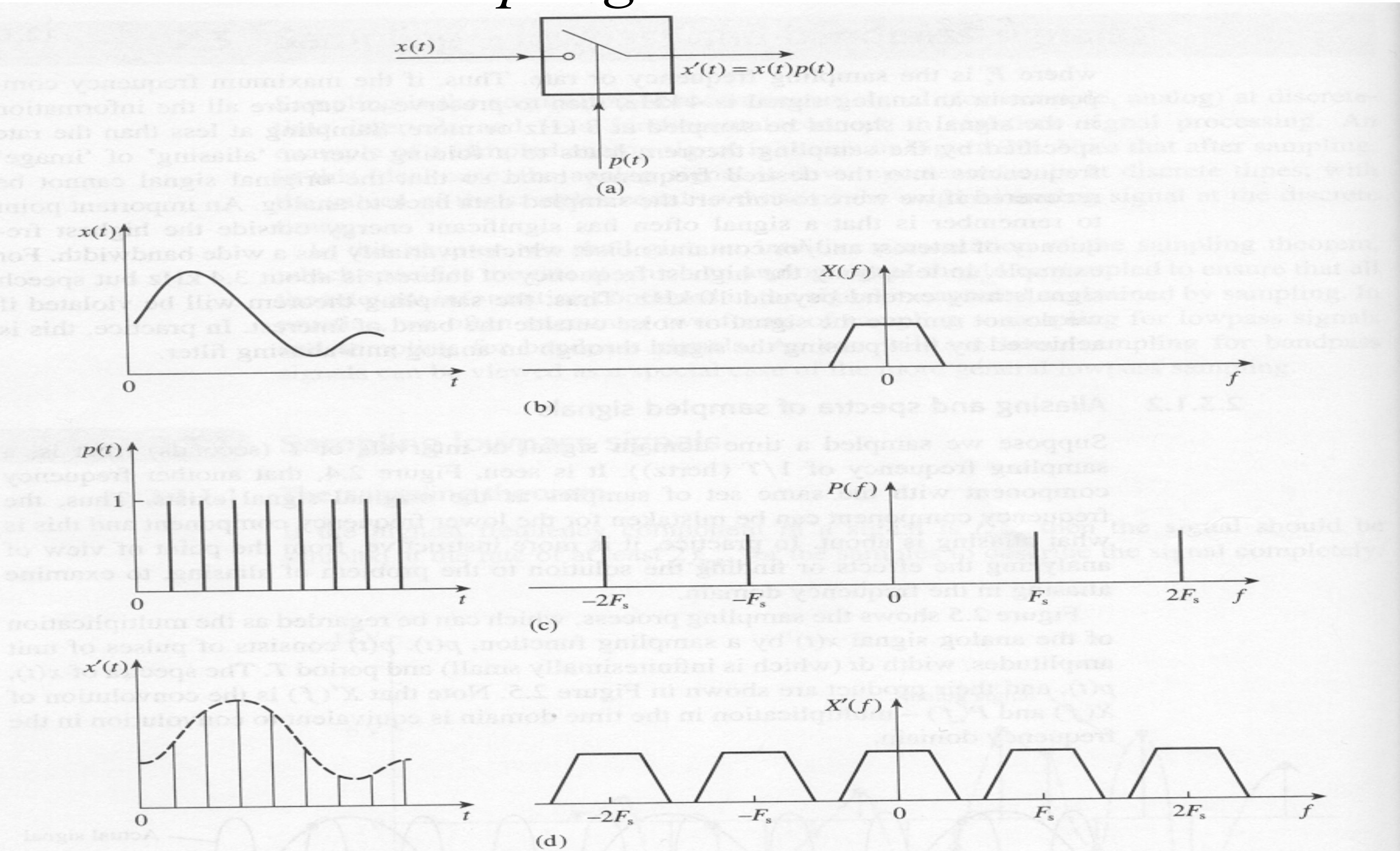
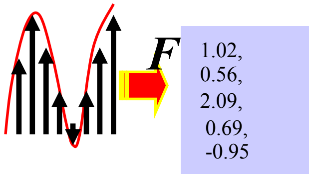


Figure 2.5 Time and frequency domain representations of the sampling process. The spectra of the signal (b) before and (d) after sampling should be compared. (d) Note the changes in the sampled signal and in particular that the spectrum of the sampled signal repeats at multiples of the sampling frequency, F_s.



Sampling of bandpass signals

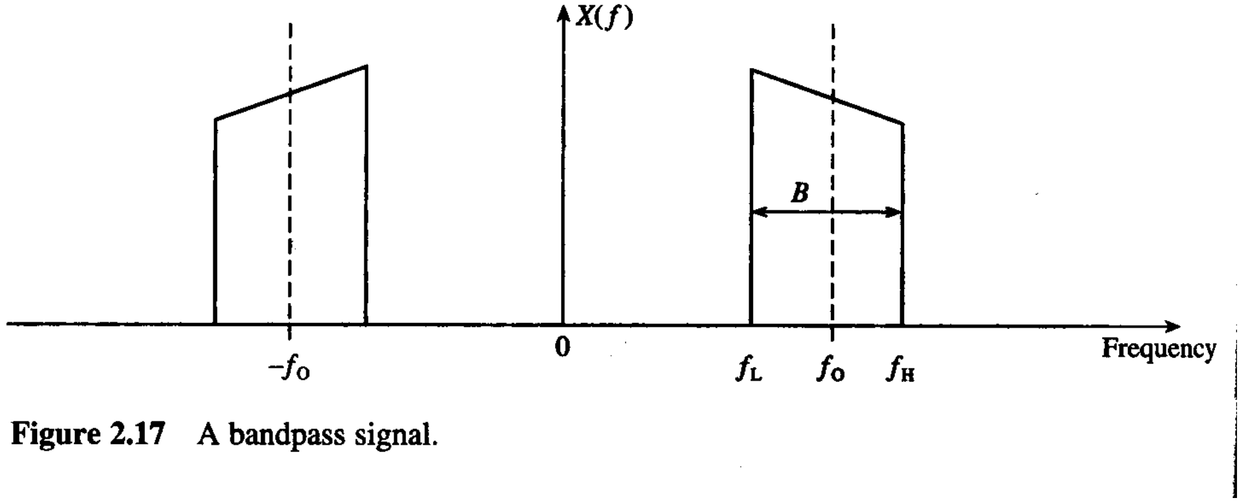


Figure 2.17 A bandpass signal.

If the bandwidth of the signal, B is small compared to the lower and Upper band edge frequencies, it is uneconomical to use the lowpass Sampling theorem. The bandpass sampling theorem states that,

$$\frac{2f_H}{n} \leqslant F_S \leqslant \frac{2f_L}{n-1}$$

where

$$n = \frac{f_H}{B}$$ (n is an integer, rounded up to largest integer)

An Illustration

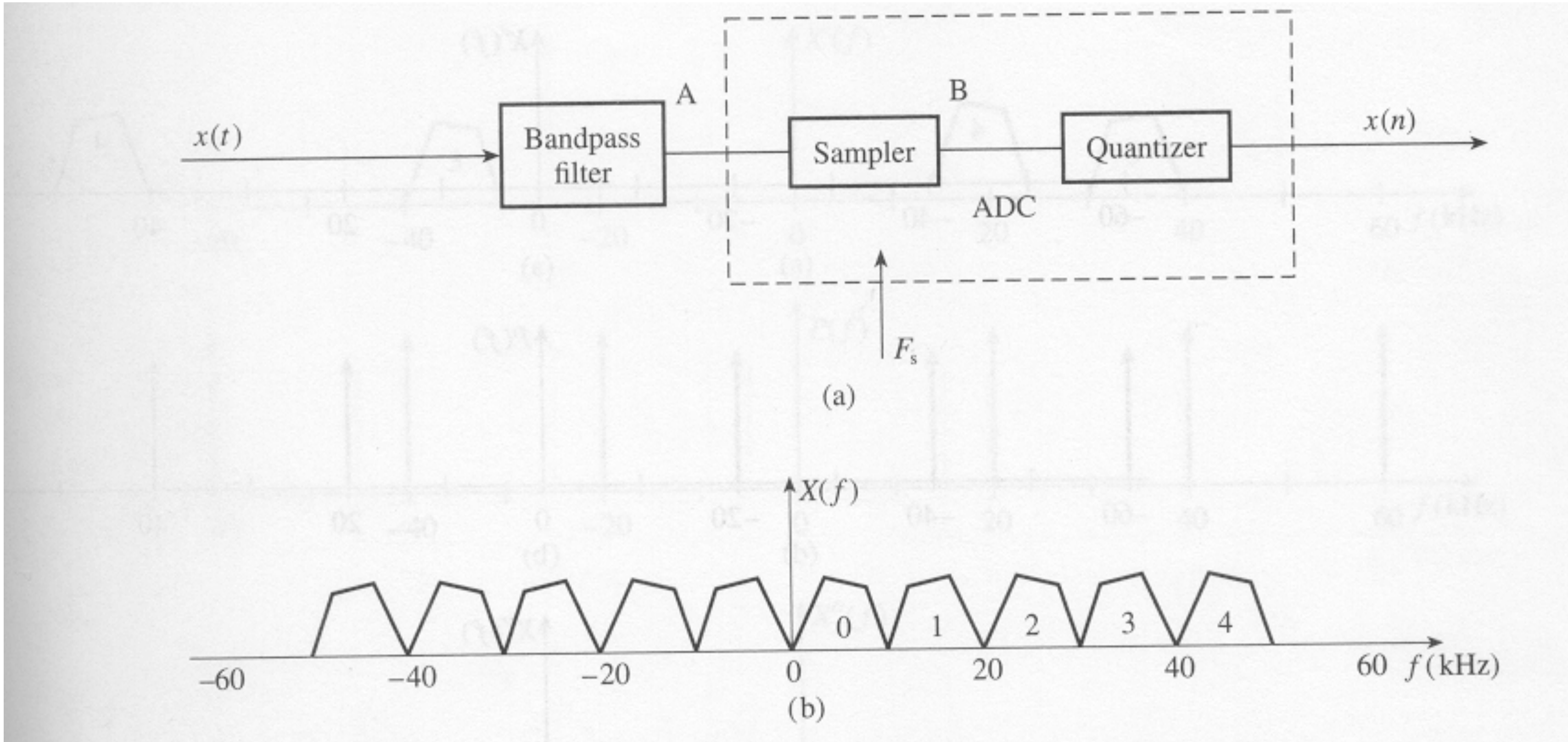
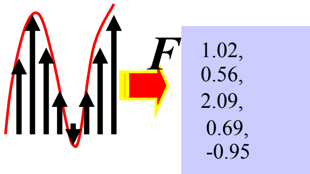


Figure 2.18 (a) Front end of system. (b) Spectrum of received signal. (Example 2.6)

•The band pass filter allows the frequencies between 40 to 50kHz

•Theoretical sampling rate is 20kHz



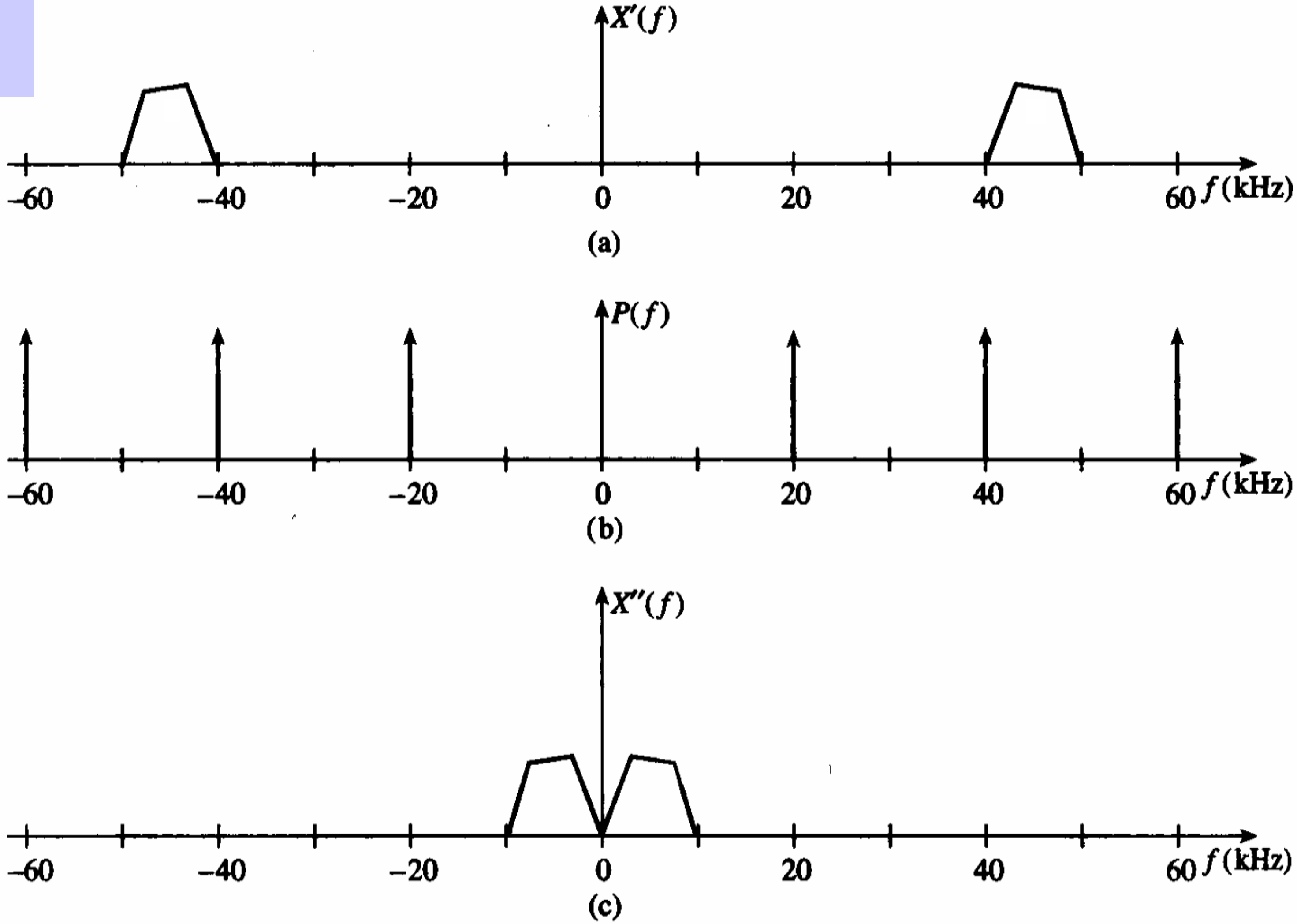
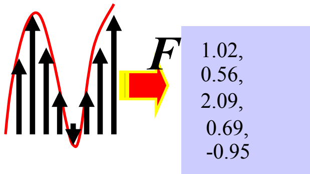


Figure 2.19 (a) Output of the bandpass filter. (b) The sampling function. (c) Output of sampler. (Example 2.6)



Steps involved

1. Fig a, shows the signal after it is passed through the bandpass filter

2. Fig b, shows the process of sampling, the sampling frequency is chosen to be 20kHz, so the samples are seen at the integer multiples of 20kHz.

3. Fig c, illustrates the result of convolution operation in frequency domain between the input spectrum and the sampling impulses.