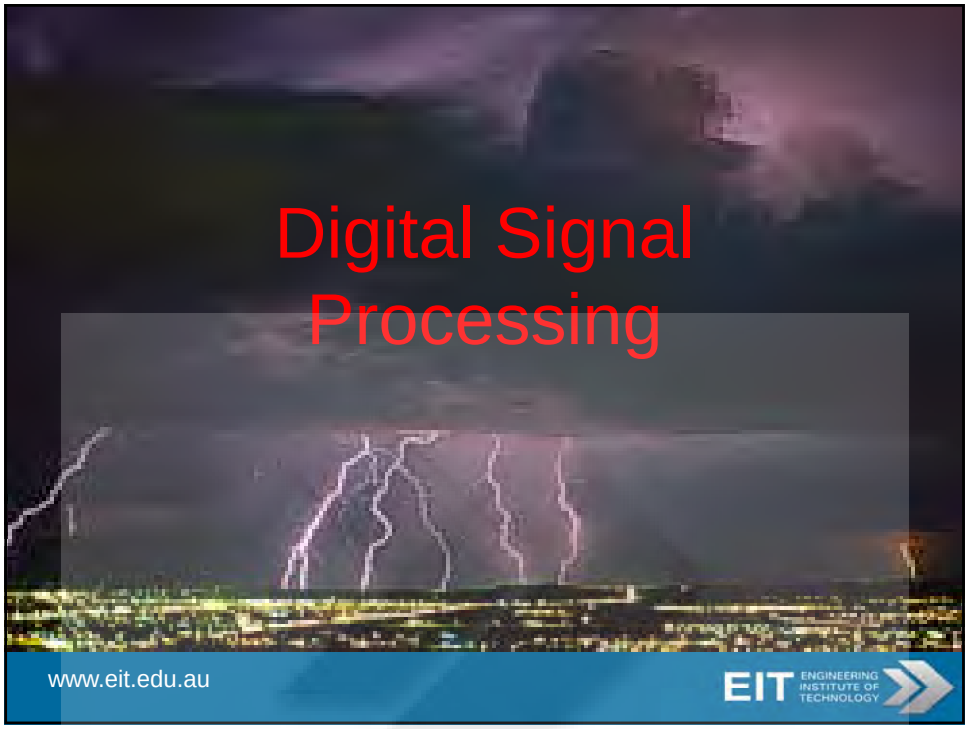
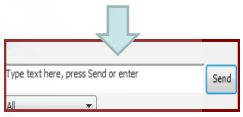



## Welcome to this webinar with Dr Steve Mackay

RH

There are at least 3 ways to interact with your presenter today.

1. Use the ***Text*** tab, near bottom left of your screen. Type the message in the space next to the "Send" button, then enter or click Send. (Ensure "All" is selected in the drop-down menu under the typing area)




2. Use the ***Emoticons*** selection in the text window – choose an emoticon then click send or enter.


3. ***Talk!*** Click the **Push To Talk** button. The button will change to yellow when you are queued, then green to show that your microphone is live.

Remember to click it again when you have spoken


**This 45 minute session will commence shortly**

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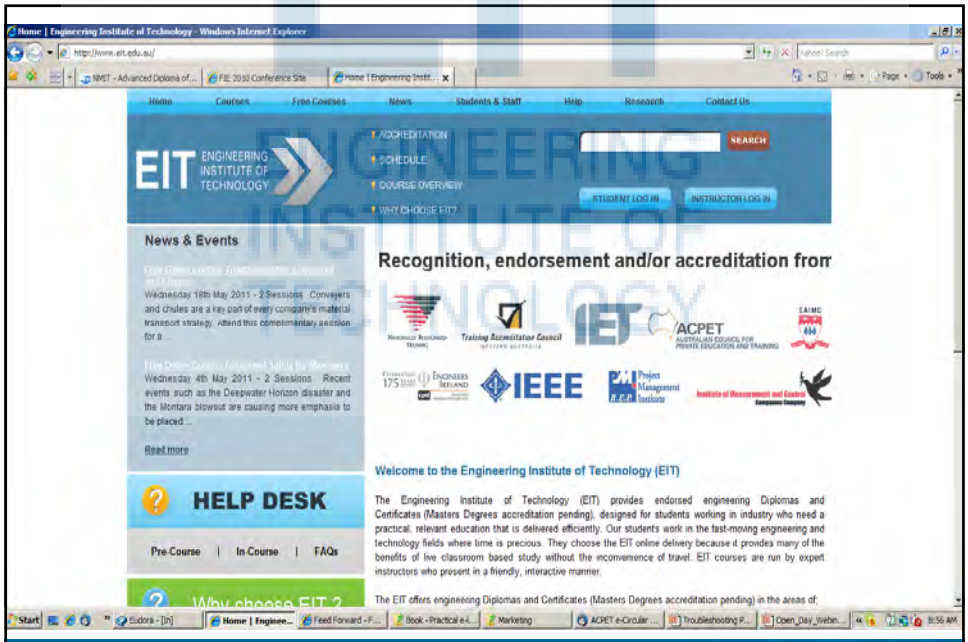
# Can't hear us ?

- **Please configure your audio by going to the TOOLS menu, then AUDIO SETUP WIZARD – and follow the steps ☺**
- If you are having problems with your microphone, don't worry – as long as you can hear the presentation that's OK – you can communicate with the text box. However, after the session we recommend you chat let us know so that we can help you.
- If you are having trouble with your speakers, and can't hear anything – and have tried the Audio Wizard – don't worry. We will be recording the session, so you will get to view a recording.
- We hope you enjoy the session, we will be starting in the next slide



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# Steve Mackay

- Dean of Engineering
- Worked for 30 years in Industrial Automation
- 30 years experience in mining, oil and gas, electrical and manufacturing industries

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RH

*Start recording!*

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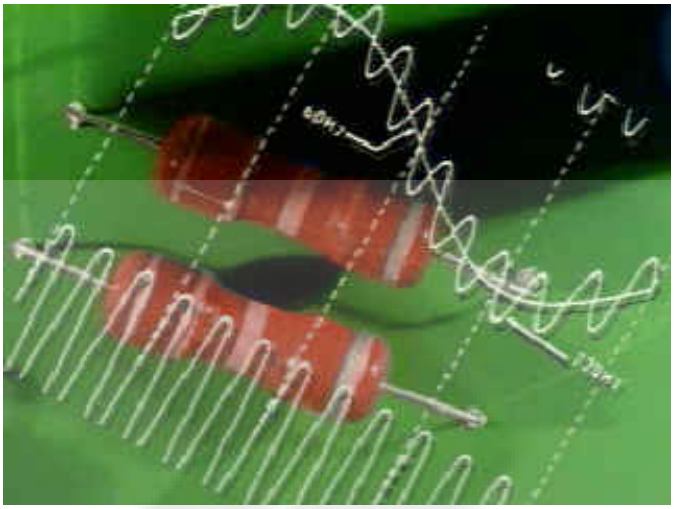
# Introduction



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## DSP

- Digital - finite set of distinct values.
- Analog - continuous range of values.

- DSP Processing means processing of signals which are:
  - Discrete in time.
  - Discrete in amplitude.

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## Two sources of discrete signals

- Ones that are discrete in nature.
- Sampled version of a continuous-time signal.



## Why Process Signals Digitally?

- The signals are inherently discrete in nature.
- Consistency -- compared with analog methods.

# Definition of Terms

- Signals. A quantity that can be measured over time.
- Frequency. A signal is said to have a frequency of 50 or 60 Hz.
- Spectrum. Some signals have a combination of frequencies.



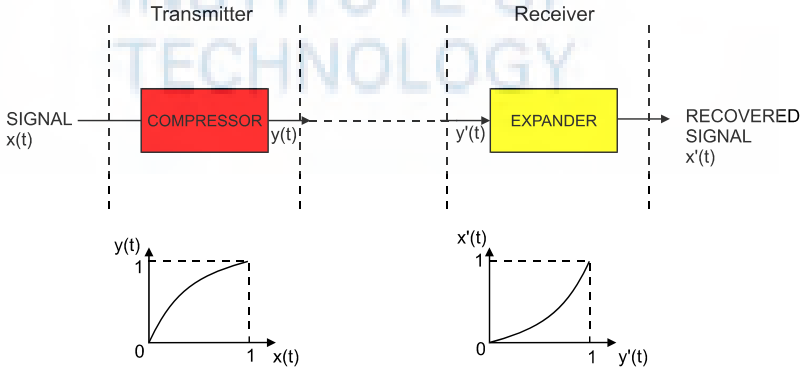
- Low Pass Filter. This lets through the low pass component.
- Bandpass Filter. Only a range of frequencies is passed through intact.
- High-pass Filter. Allows frequencies above a certain frequency to pass through intact.

# Some Application Areas

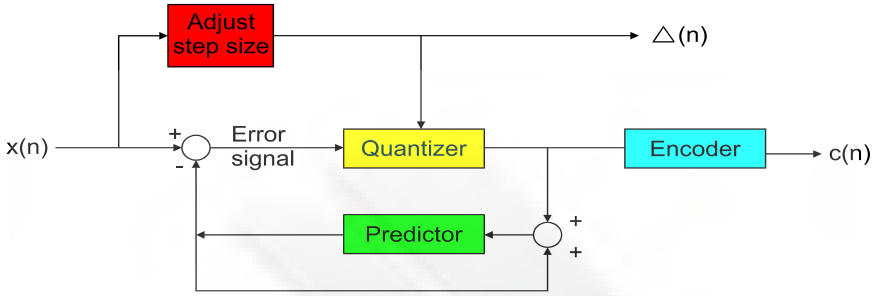
- Speech and Audio Processing.
  - Coding.
  - Synthesis.
  - Recognition.



# Companding Process



# Adaptive Differential Pulse Code Modulator

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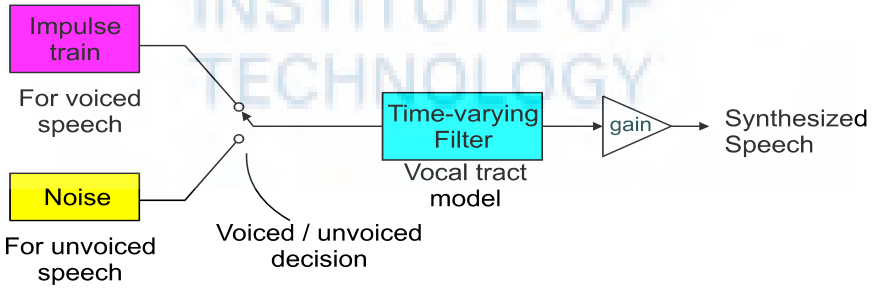
# Linear Predictive Coding

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## Image and Video Processing

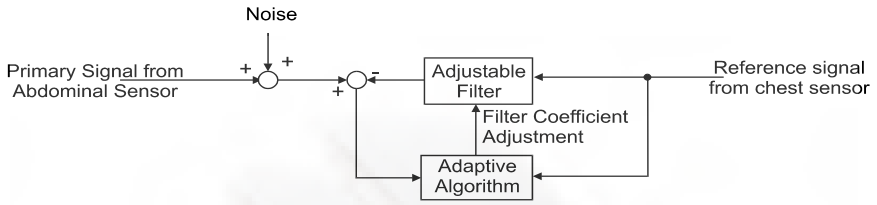
- Image Enhancement.
- Image Restoration.
- Image Compression and Coding.



## Adaptive Filtering

- Noise Cancellation.
- Echo Cancellation.
- Channel Equalisation.

# Adaptive Noise Cancellation System

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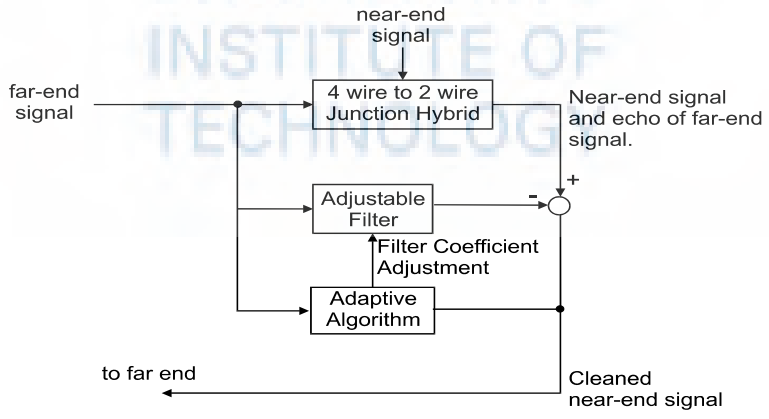
# Adaptive Echo Cancellation System

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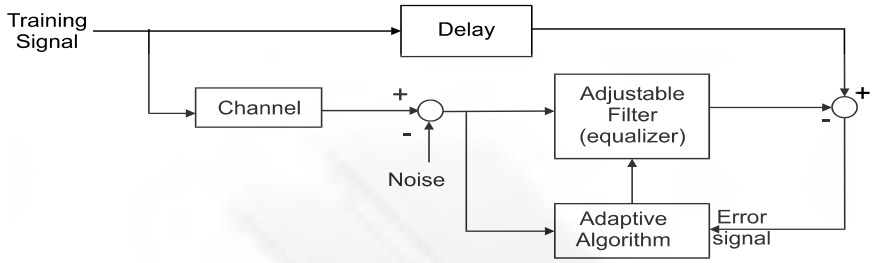
# Adaptive Equalizer in Training Mode

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# Control Applications

- Controlling closed-loop feedback systems.
- Controller implements algebraic algorithms such as filters and compensators.

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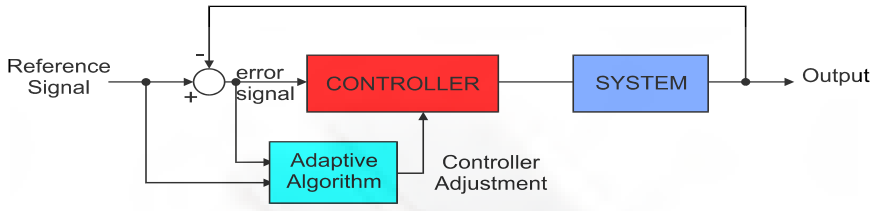
# Digital Closed Loop Control System

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# Sensor or Antenna Array Processing

- Spatially distributed sensors are used for receiving signals from some sources.
- Coherent summing of outputs from these sensors referred to as beamforming.
- One can “listen” preferentially to wavefronts originating from one direction over another.

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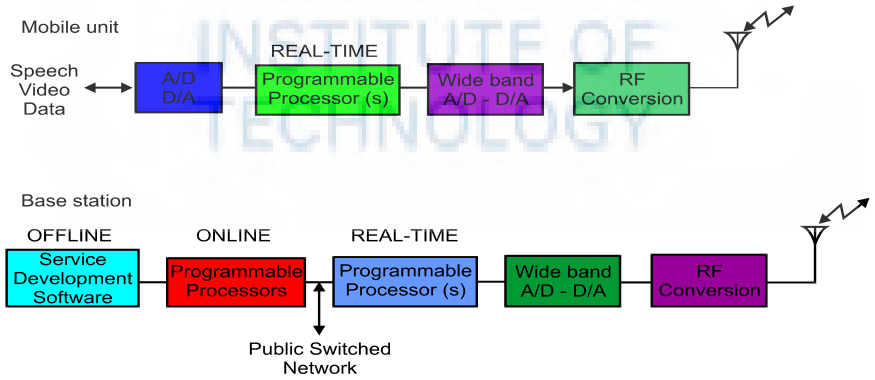
# Digital Communication

- New Architecture called software radio.
- Use of wideband A/D and D/A converters to convert RF or IF signals directly.

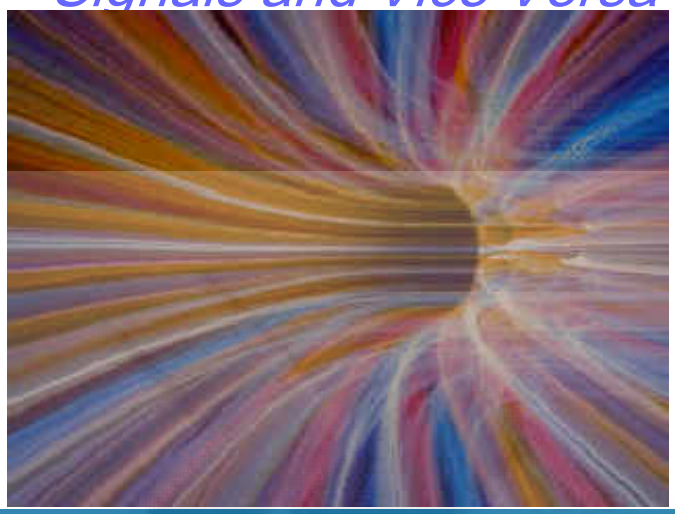


# Software Radio Architecture




Note: Hardware is simple and functions are software-defined

# Converting Analog to Digital Signals and Vice Versa

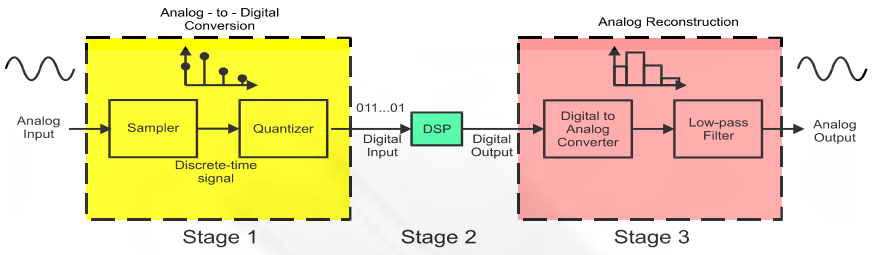




# A Typical DSP System

- Three stages
  - The analog signal is digitized. This involves
    - sampling,
    - quantization.
  - Digital signal processed by DSP algorithms.
  - Outputs converted back into analog signals.

# Analog-Digital-Analog

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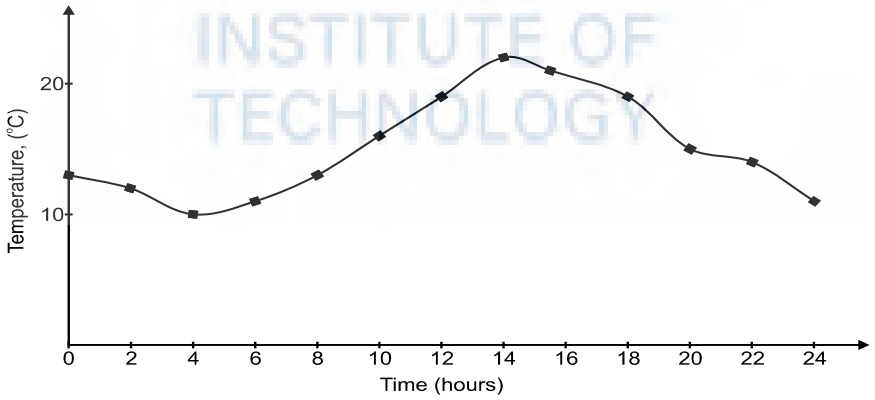
# Temperature Variation in a Day

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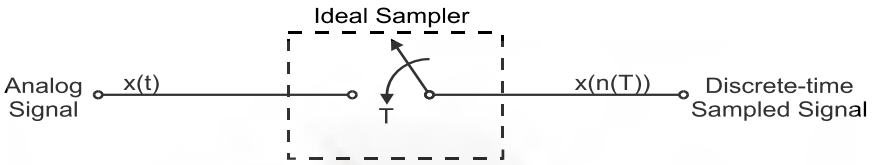
# The Sampling Process



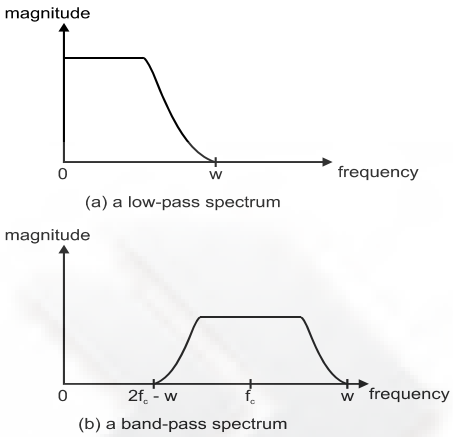



# Uniform Sampling Theorem

- If a continuous time signal contains no frequency components higher than *W* Hz, then it can be completely determined by uniform samples taken at a rate $f_s$ samples per second where $f_s \geq 2W$
- or, in terms of the sampling period

$$T \leq \frac{1}{2W}$$

# Two Bandlimited Spectra

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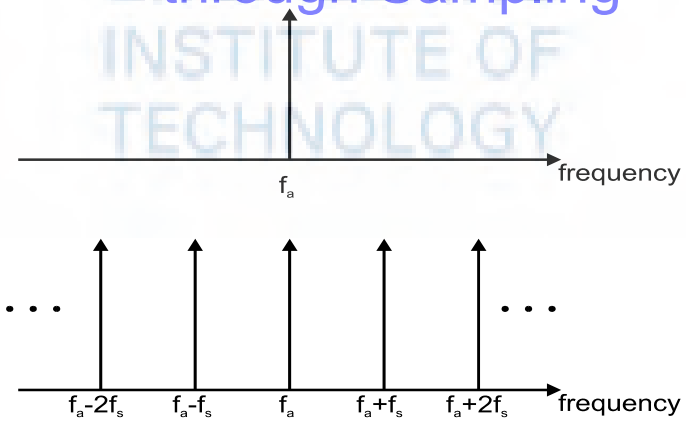
# Replication of Spectrum through Sampling

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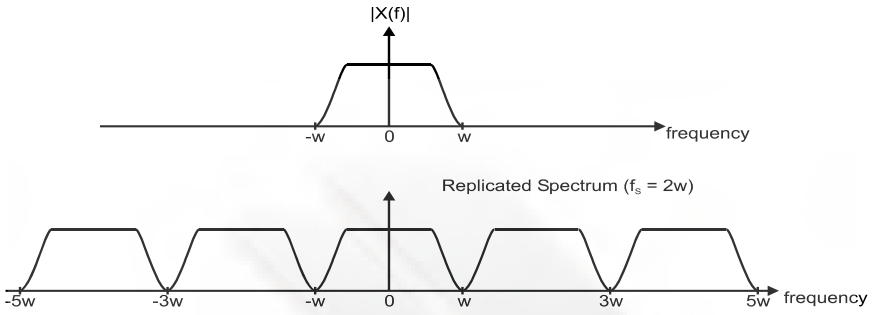
# Replicated Spectrum

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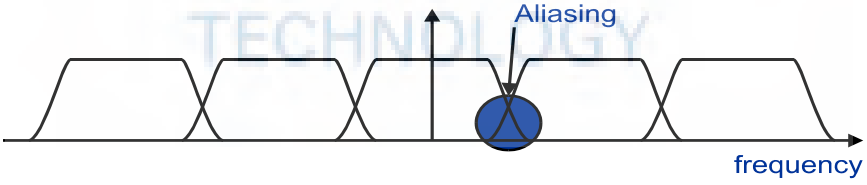
# Aliasing

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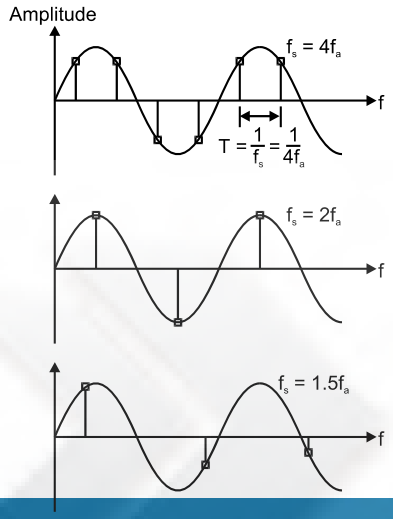
# Sampling at Different Rates

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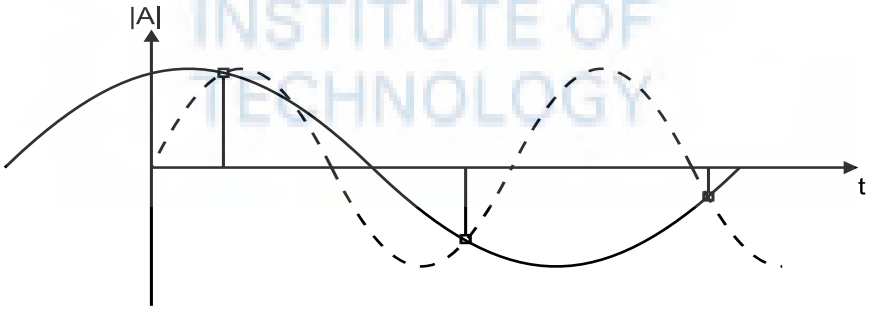
# Effect of Aliasing

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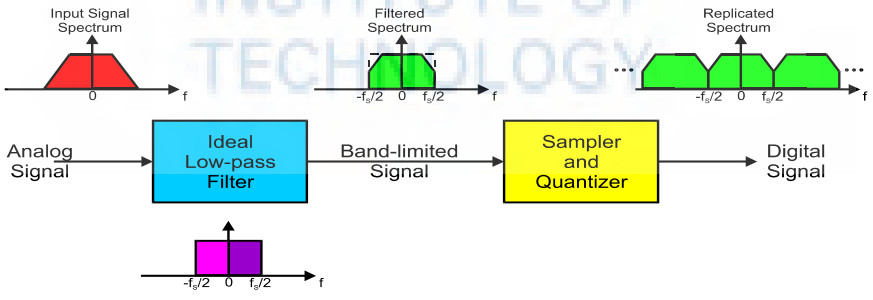
# Anti-aliasing Filters

- Process the signal before it is sampled so they are always analog filters.
- Usually low-pass filters unless bandpass sampling techniques are used.



# Analog-to-Digital Conversion with Anti-alias Filtering



# Implications of Non-ideal Anti-alias Filters

- Sample at higher than Nyquist rate.
- If sampling rate cannot be changed, then a filter with sharper cut-off (implying a higher order filter) has to be used.



# Limits on Sampling Rates

- Practical choice of sampling rate is determined by two factors:
  - Sampling theorem places lower bound on the allowed values of sampling frequency.
  - Economics of hardware imposes upper bound on sampling frequency.