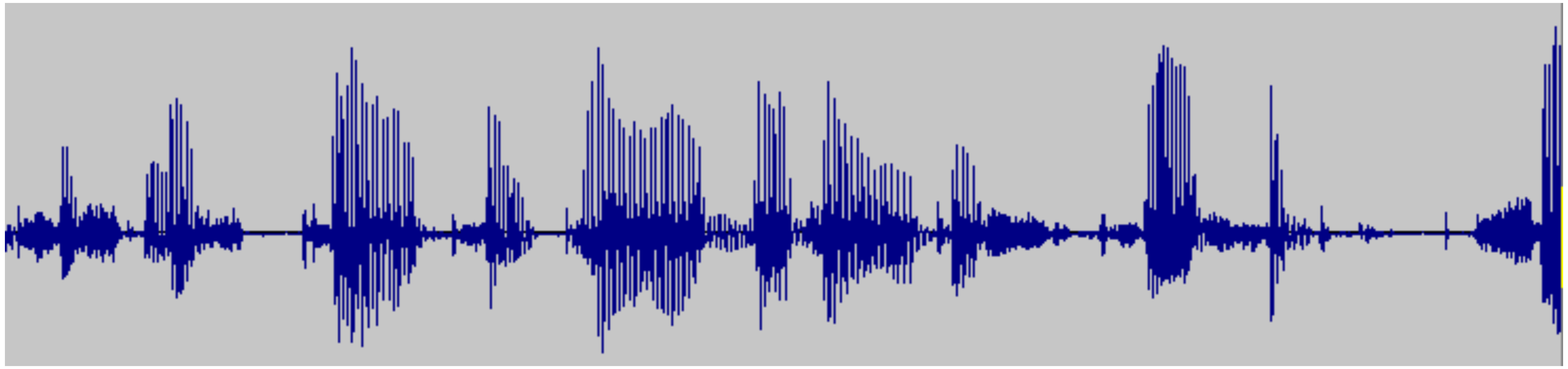
Lecture 10: Signals

ENAE 380 Flight Software Systems
October 7, 2024

Signal Processing

Signals: functions of one or more independent variables that carry information



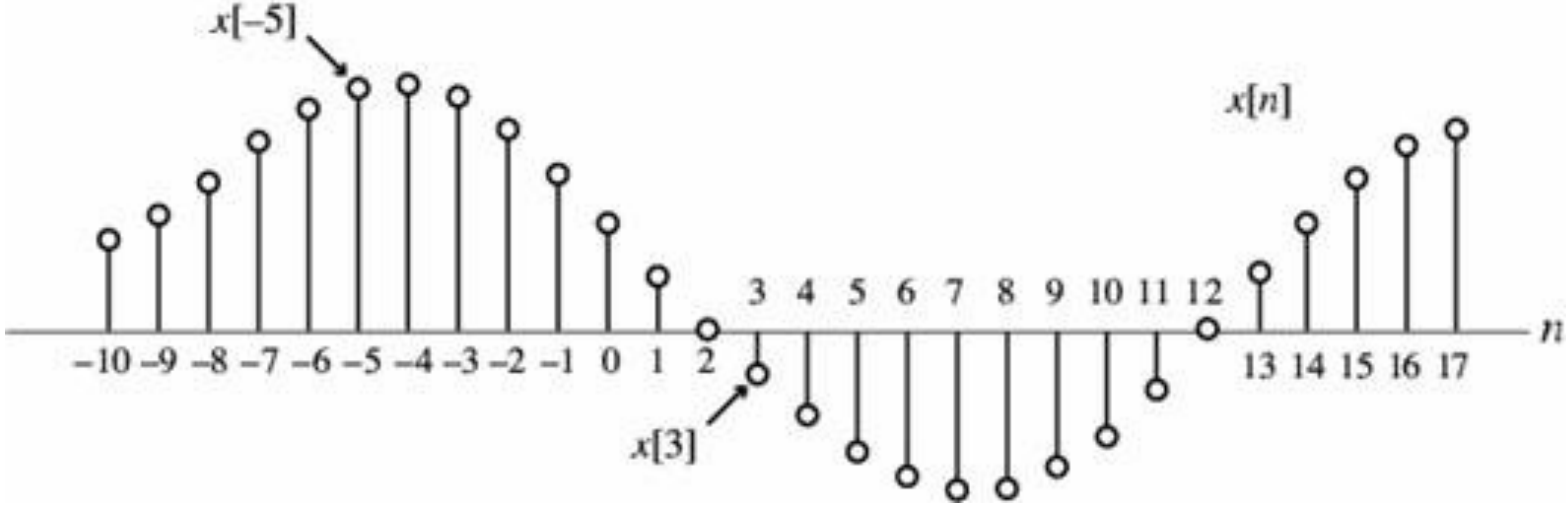
Independent variable time is continuous —> ***continuous time signal***

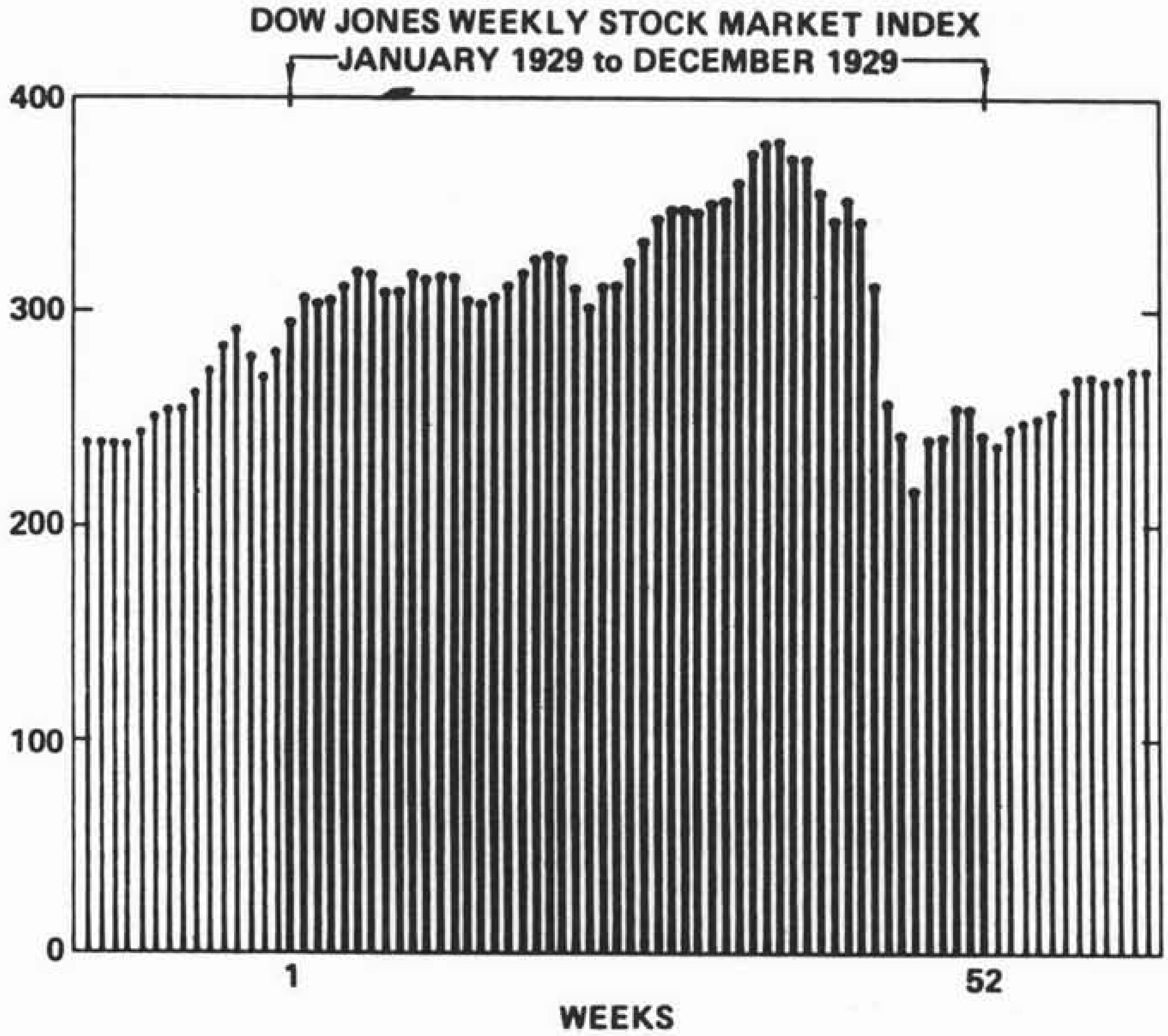
Function of one independent variable —> ***one-dimensional signal***

Signals can be multi-dimensional, and
independent variable does not have to be time
—> images

1 sample point: Black and white (or grayscale)
3 sample points: Color (red, green, blue)
4 sample points: color and opacity (alpha)

Discrete time signals: function of an integer variable

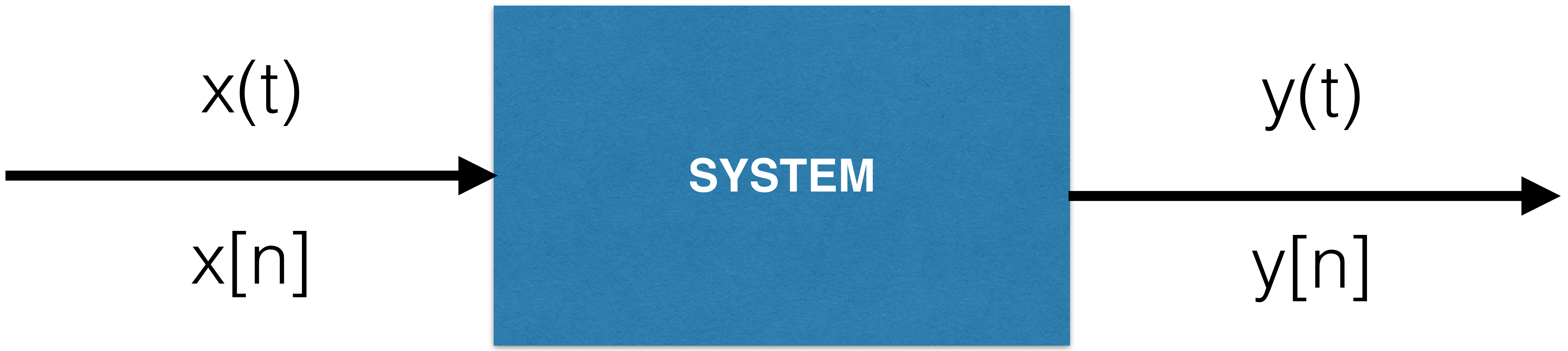




DOW JONES WEEKLY STOCK MARKET INDEX
JANUARY 1929 to DECEMBER 1929
400
300
200
100
0
1
52
WEEKS

Current and emerging technologies permit processing of continuous-time signals by first converting them to discrete signals and processing them with discrete-time systems.

A system processes signals. They have inputs and outputs



linear

non-linear

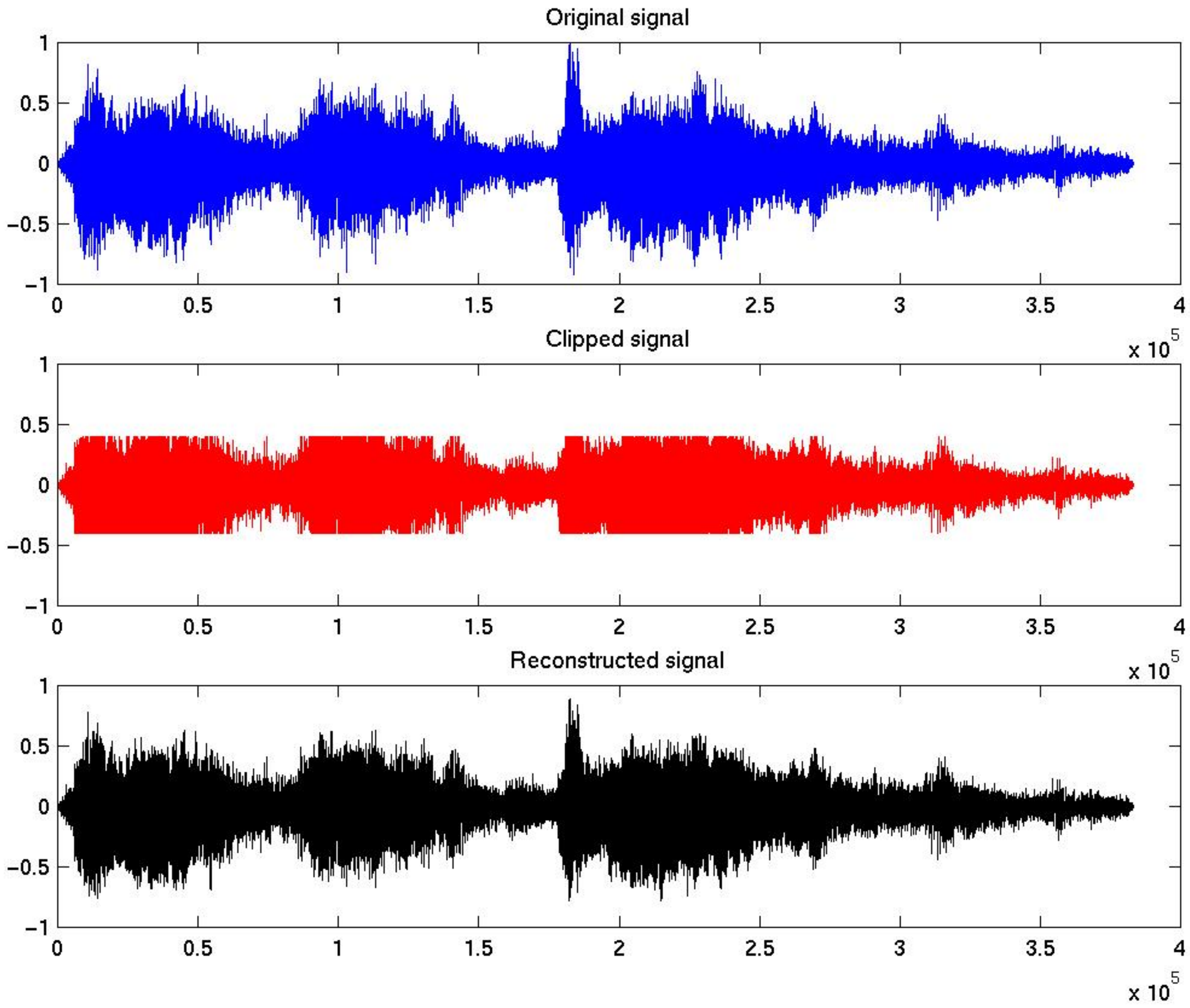
time-invariant

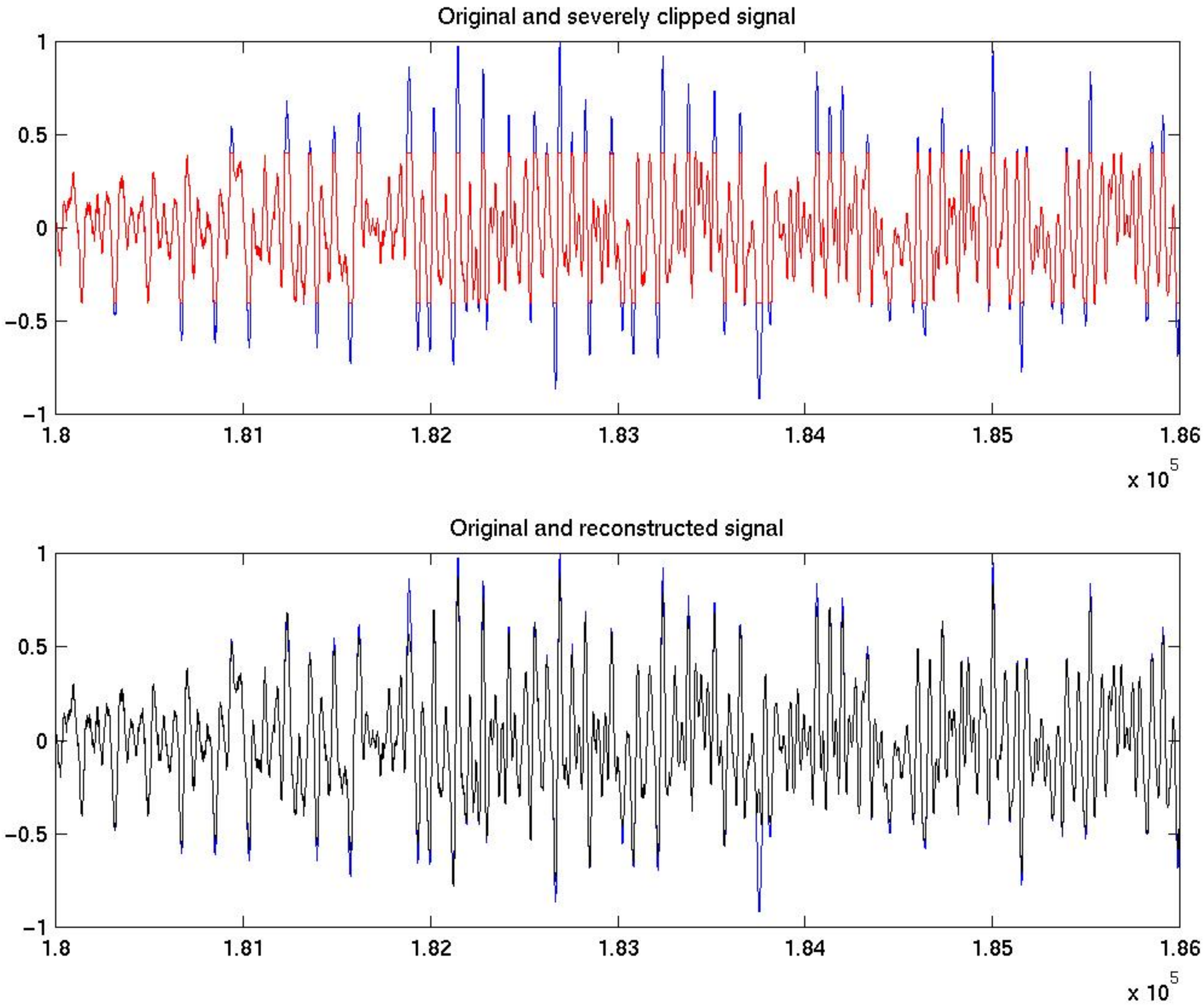
time-varying

Linear, Time-Invariant Systems (LTI)

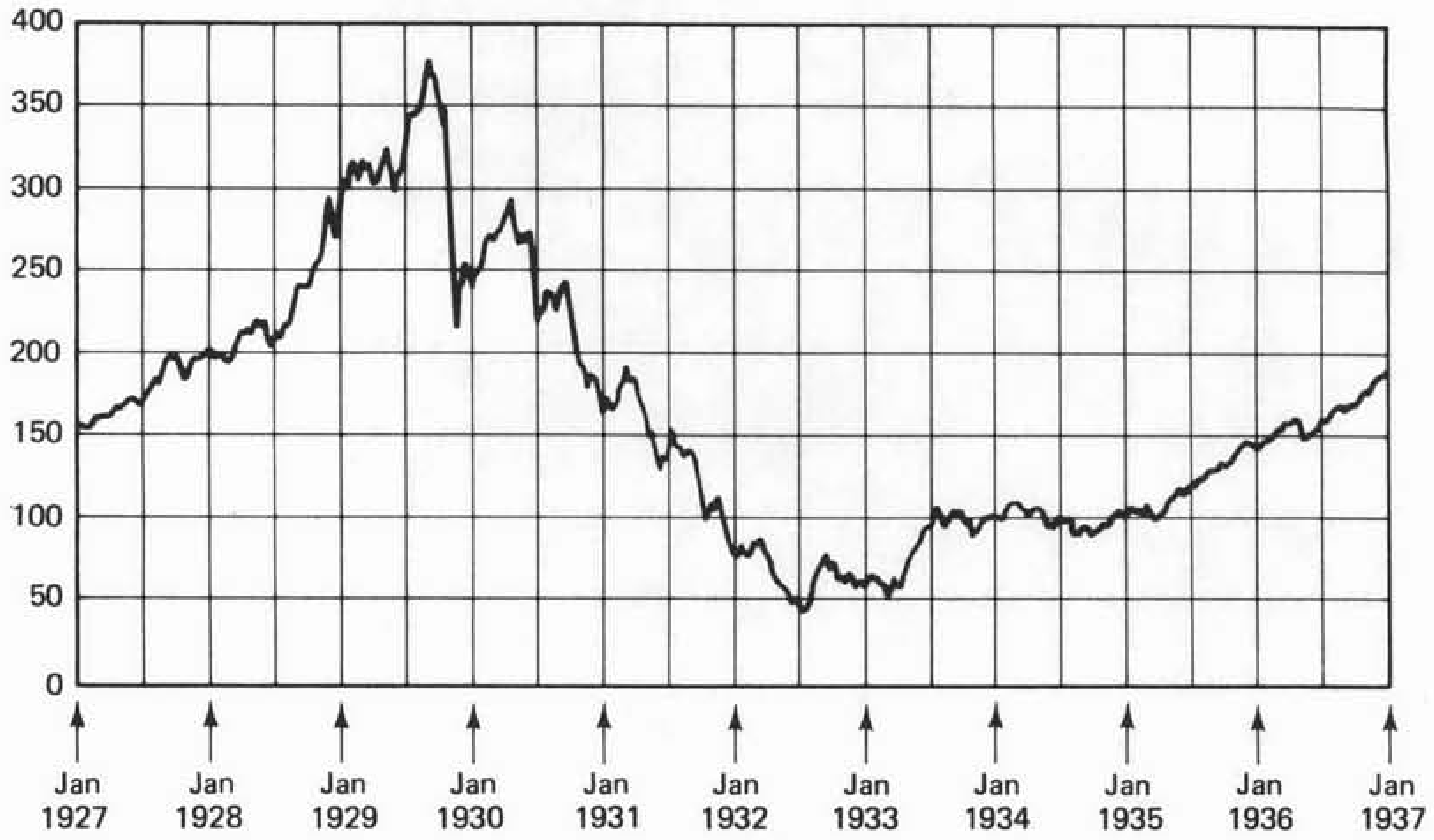
Continuous Time Processing

https://www.math.ucdavis.edu/~strohmer/research/audio/audio.html

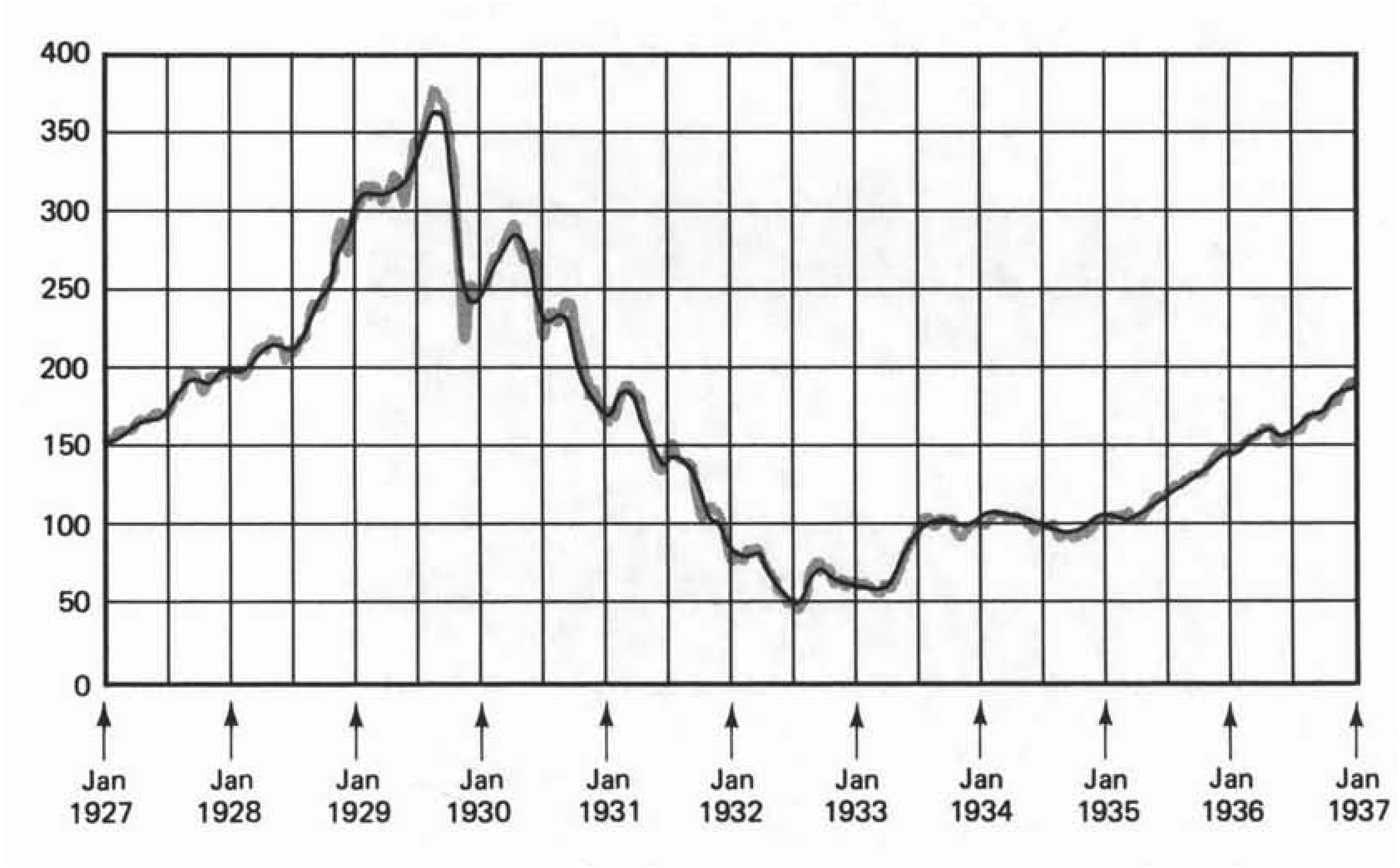



Original and severely clipped signal
1
0.5
0
−0.5
−1
1.8
1.81
1.82
1.83
1.84
1.85
1.86
x 10^5
Original and reconstructed signal
1
0.5
0
−0.5
−1
1.8
1.81
1.82
1.83
1.84
1.85
1.86
x 10^5

How to process discrete signal?



51 Day Moving Average

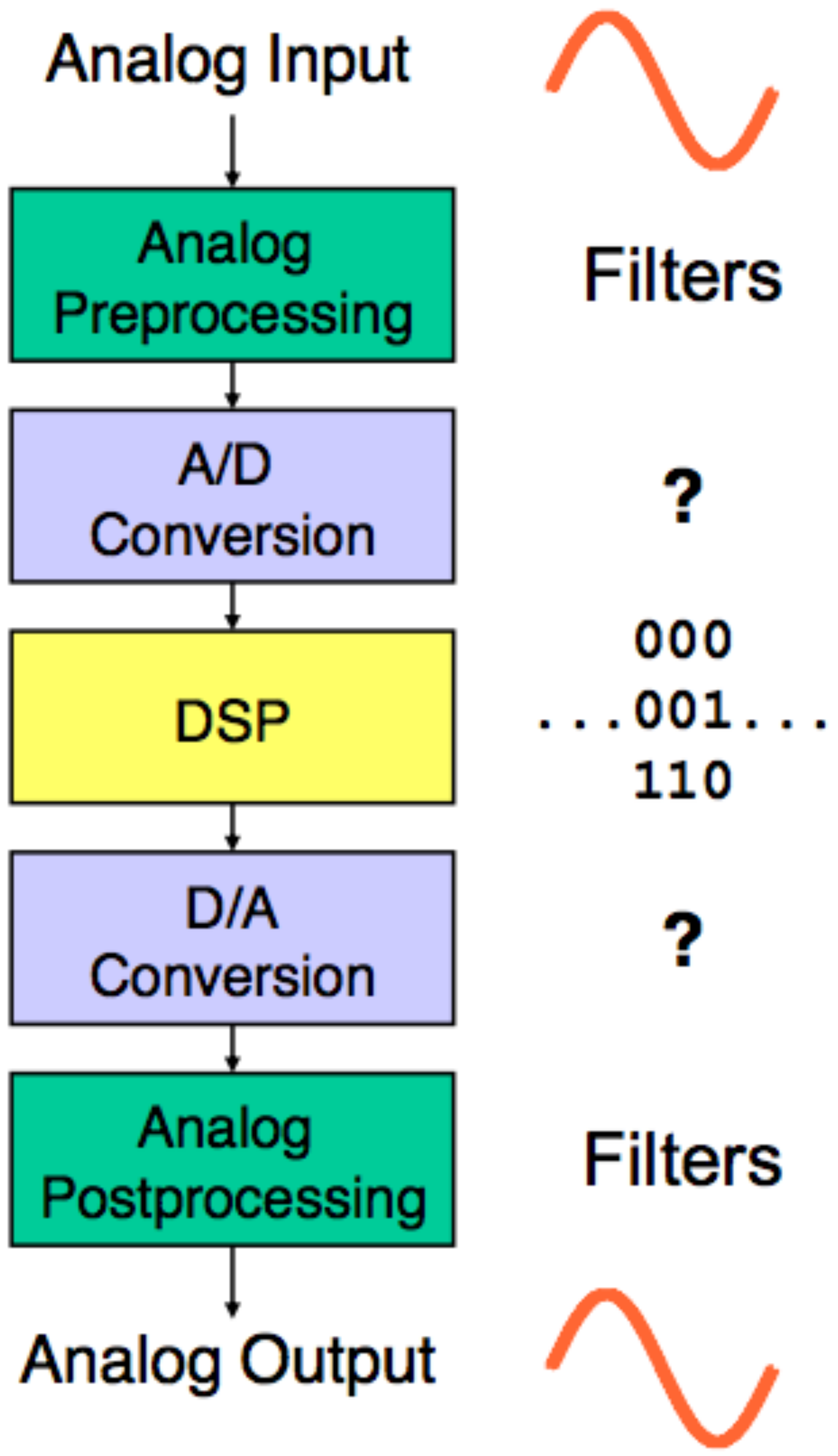


Digital Signal Processing

- Digital Signal Processor (DSP) is an integrated circuit designed for high speed data manipulations.
- Used in audio, communications, image manipulation, and other data acquisition and data control applications
- How it works
  - Analog: sound, pressure, light intensity
  - Digital: computers

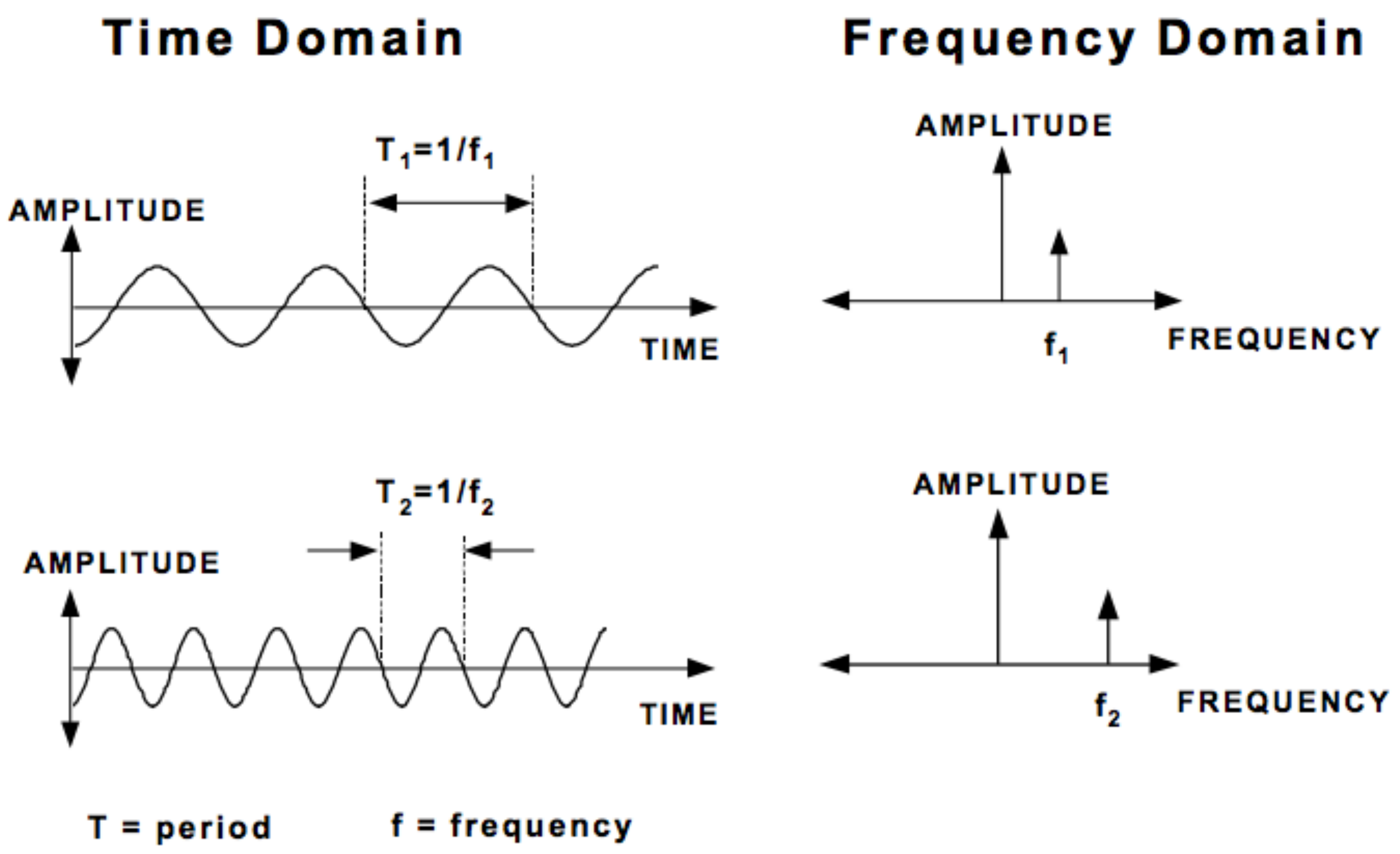
Digital Signal Processing

- How do they work together?
  - World is analog, but computers are digital. So, to utilize tremendous processing power
    - Convert analog to electrical signals using transducer
      - Microphone
    - Digitize signals (ADC)
    - DSP
    - Convert back to analog (DAC)
      - Loudspeaker



Signals

All electric signals can be visualized in
time domain or frequency domain

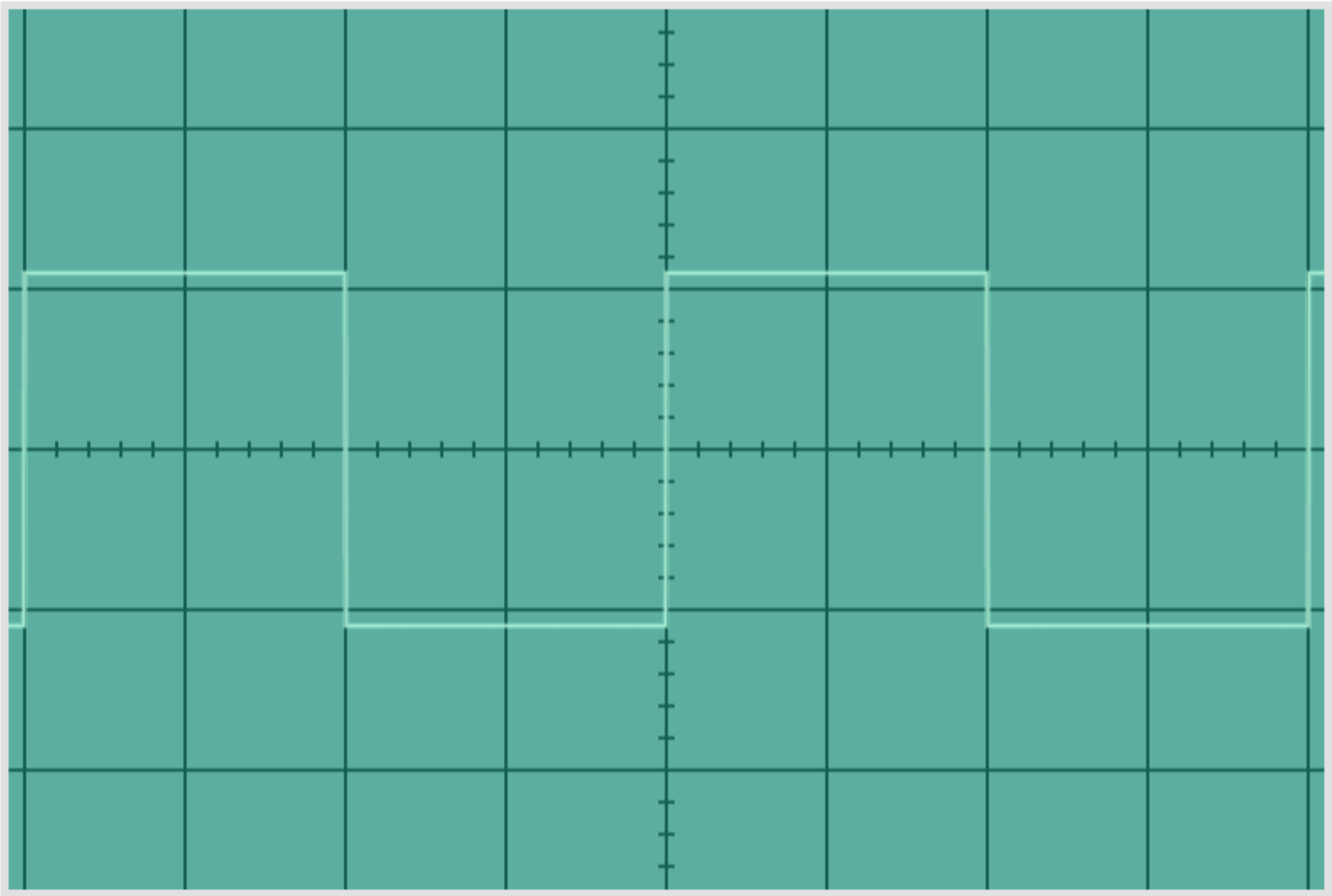


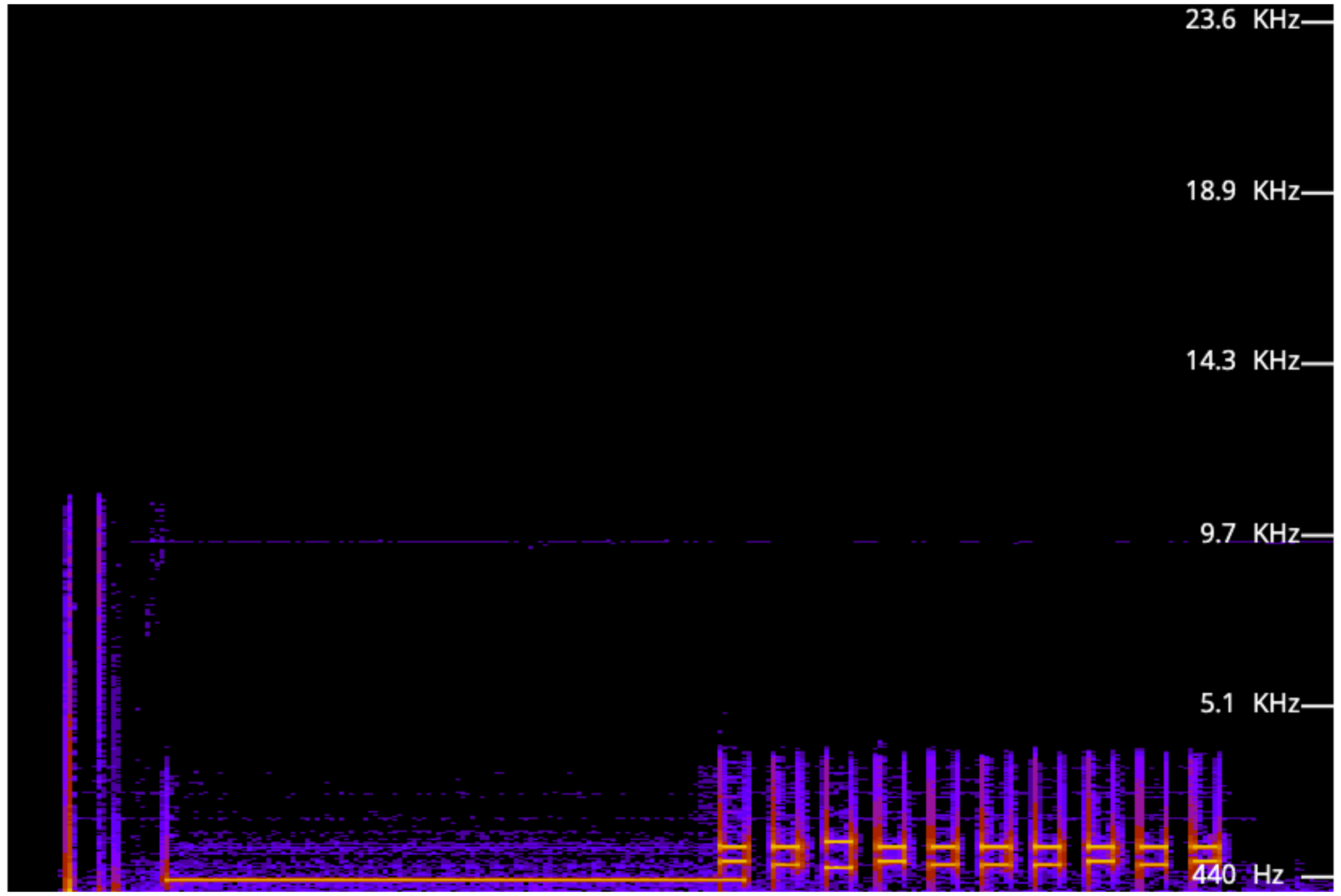
- Time domain: variations of signal over time (oscilloscope)
  - Frequency: number of times signal repeats in 1 second
  - Period: time of 1 cycle
- Frequency domain: variation of signal with respect to frequency (music spectrum analyzers)

Domains: Time and Frequency

https://academo.org/demos/virtual-oscilloscope/

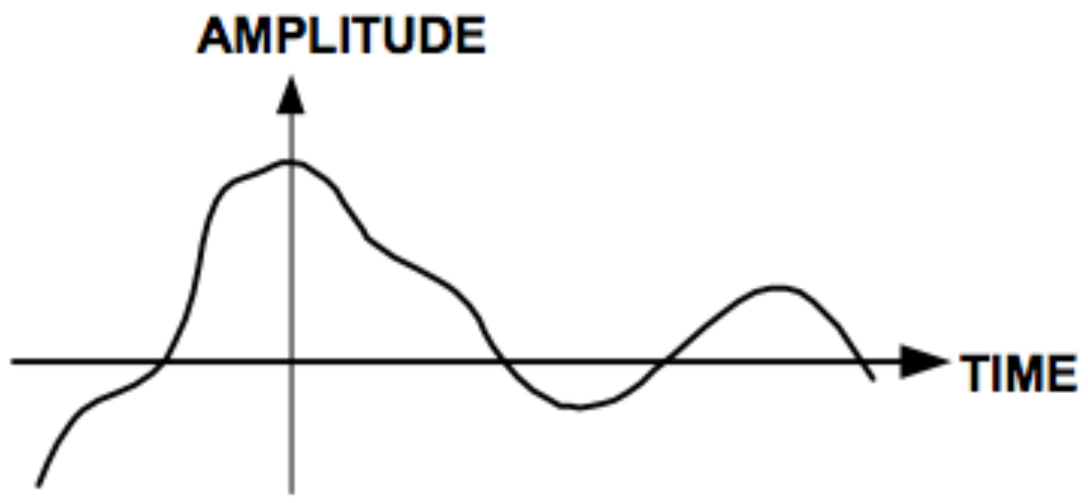
https://academo.org/demos/spectrum-analyzer/

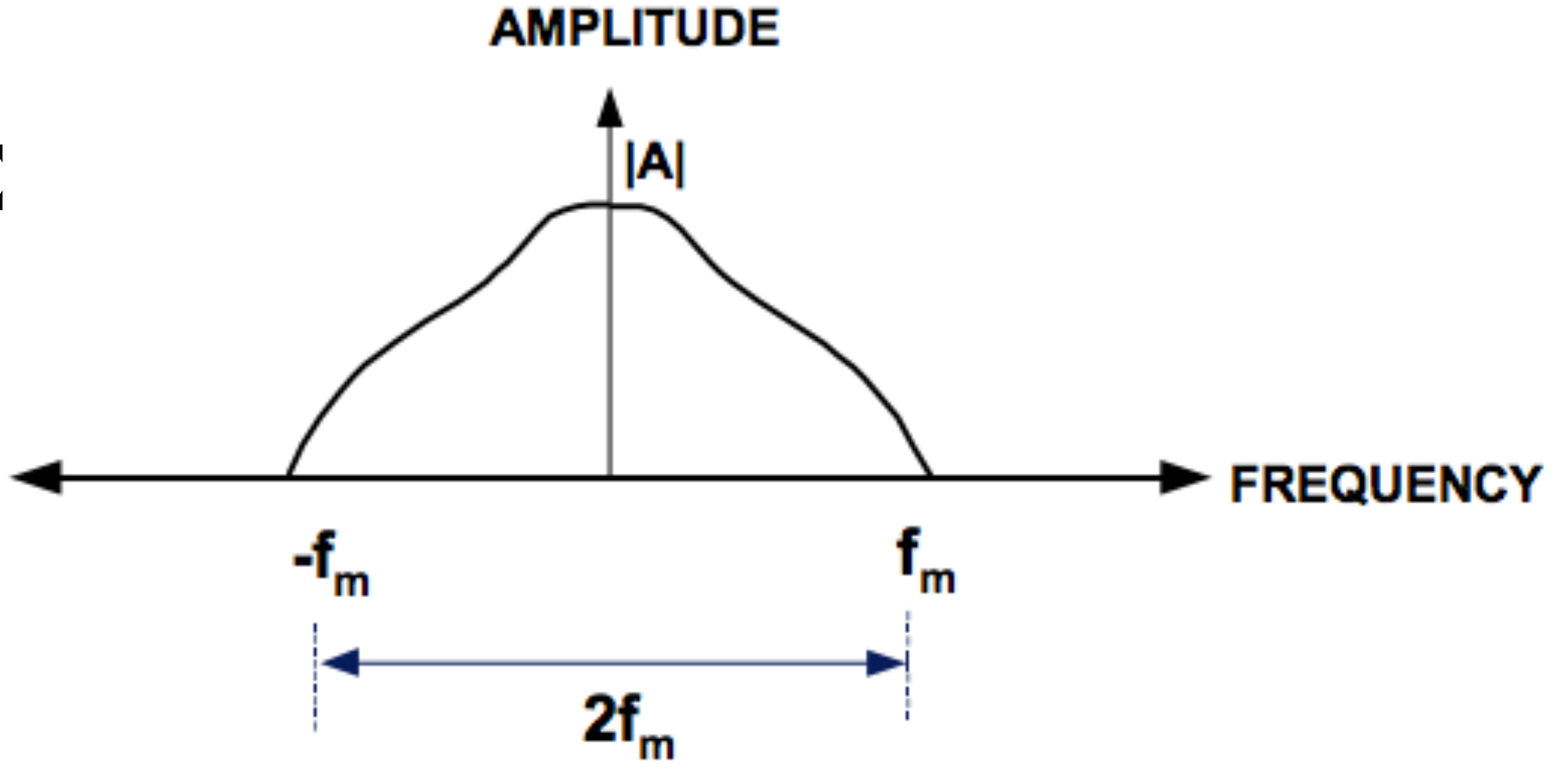




Key Concepts

- Real Signals, such as sound/speech, are more complicated than sine wave.
  - Contains many frequencies and amplitudes to combine in a composite signal
- Bandwidth: difference between ma and min frequency above certain amplitude (f_m)
- Spectrum: shape of signal
- Sampling: One step in process of converting from analog to digital back to analog





Breadth and Depth of DSP

- Roots of DSP: Sensory data from real world
  - Seismic vibrations, images, sound waves
- DSP: math, algorithms, and techniques to manipulate these signals after they have been converted into a digital form
  - Enhancement of images
  - Recognition and speech generation
  - Data compression

Breadth and Depth of DSP

Space	Photograph enhancement, Data Compression, Intelligent sensory analysis by remote space probes
Medical	Diagnostic imaging (CT, MRI, ultrasound), Electrocardiogram analysis, medical image storage, retrieval
Commercial	Image and sound compression, Movie special effects, Video conference calling
Telephone	Voice and data compression, Echo reduction, Signal multiplexing, Filtering
Military	Radar, Sonar, Ordnance guidance, Secure communication
Industrial	Oil and mineral prospecting, Processing monitoring and control, Nondestructive testing, CAD and design tools
Scientific	Earthquake recording and analysis, Data acquisition, Spectral analysis, Simulation and modeling

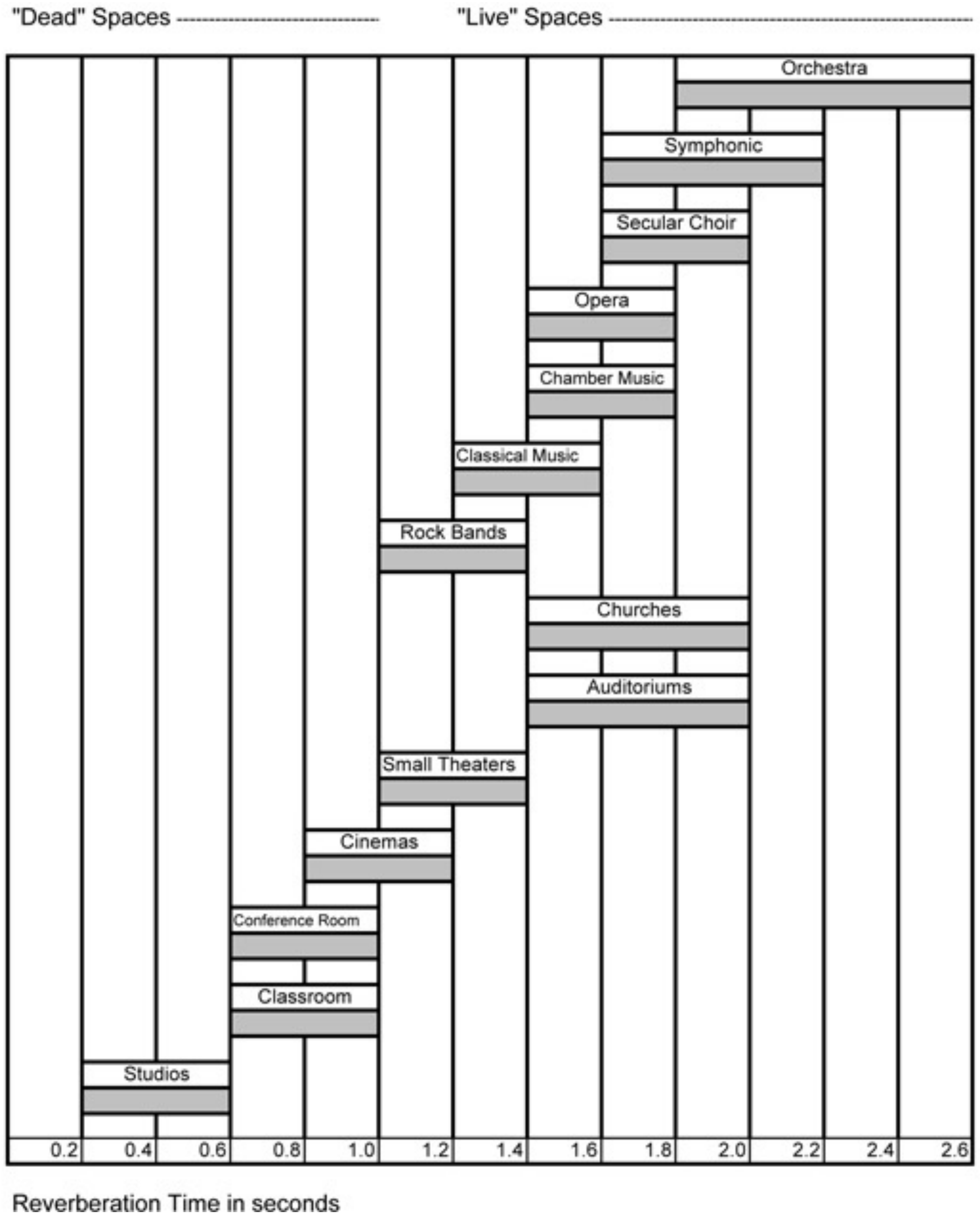
1. Telecommunications

- Transferring information from one location to another
- Multiplexing
  - Multiple conversations over a single channel
- Compression
  - Digitized voice signal is sometimes redundant
- Echo Control
  - Intercontinental delay several hundred milliseconds. DSP measures returned signal, generates anti signal

2. Audio Processing

- Music
  - Microphone to speaker
  - Artificial reverberation
- Speech Generation
  - Digital recording - 1 hour of speech = 3MB
  - Vocal tract simulation - Generate digital signals resembling voiced and fricative sounds
- Speech Recognition
  - Feature extraction, feature matching

https://acousticalsolutions.com/reverberation-examples-and-explanations/



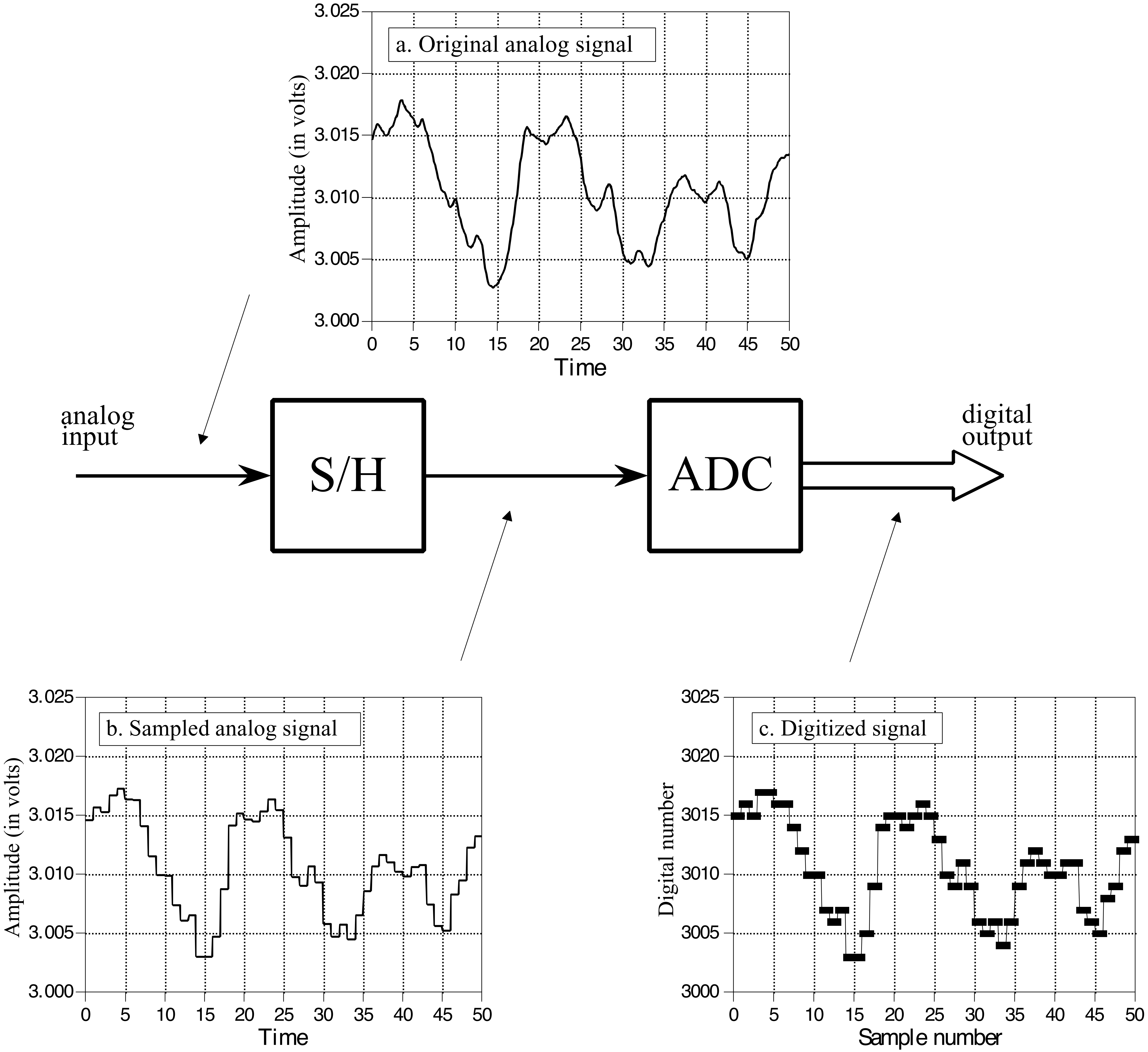
https://www.youtube.com/watch?v=0uIzOUh1t8M

3. Echo Location

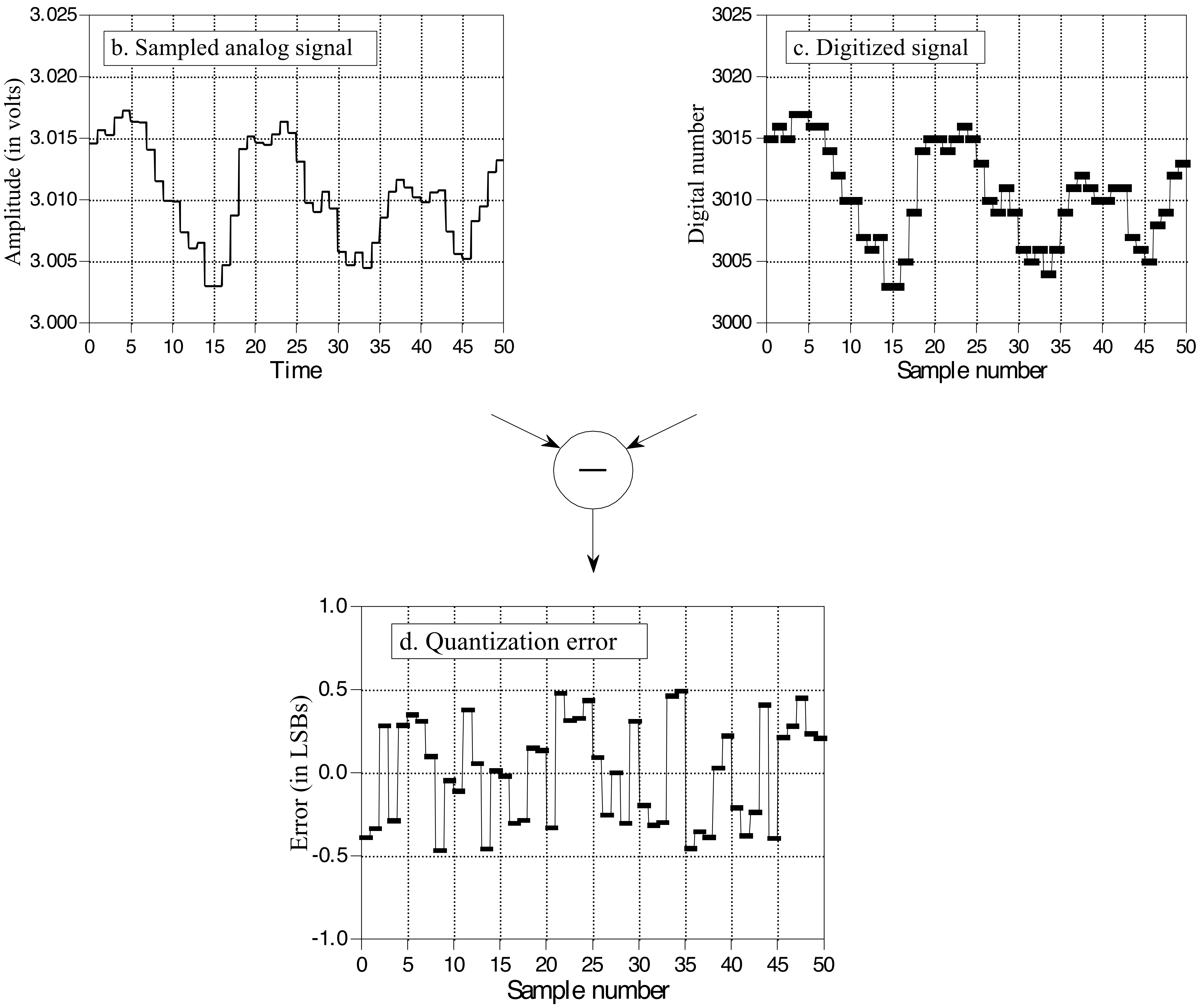
- Radar: Radio Detection and Ranging
  - Compress pulse after received (better distance determination), filter received signal to decrease noise, rapid selection and generation of different pulse shapes
- Sonar: Sound Navigation and Ranging
  - Active and passive
  - Pulse generation, pulse compression, filtering
- Reflection Seismology
  - Structure of earth probed with sound
  - DSP used to isolate primary from secondary echoes

4. Image Processing

- Medical
  - X-rays (1895) limited until DSP:
  - overlapping structures -> CAT or CT scans
  - similar tissues, anatomy not physiology, causes cancer -> Penetrating energy versus x-ray (MRI)
- Space
  - Improve quality of images taken during unfavorable conditions
- Commercial Imaging Products
  - Image compression



a. Original analog signal
Amplitude (in volts)
Time
analog input
S/H
ADC
digital output
b. Sampled analog signal
Amplitude (in volts)
Time
c. Digitized signal
Digital number
Sample number



b. Sampled analog signal
Amplitude (in volts)
Time
c. Digitized signal
Digital number
Sample number
d. Quantization error
Error (in LSBs)
Sample number

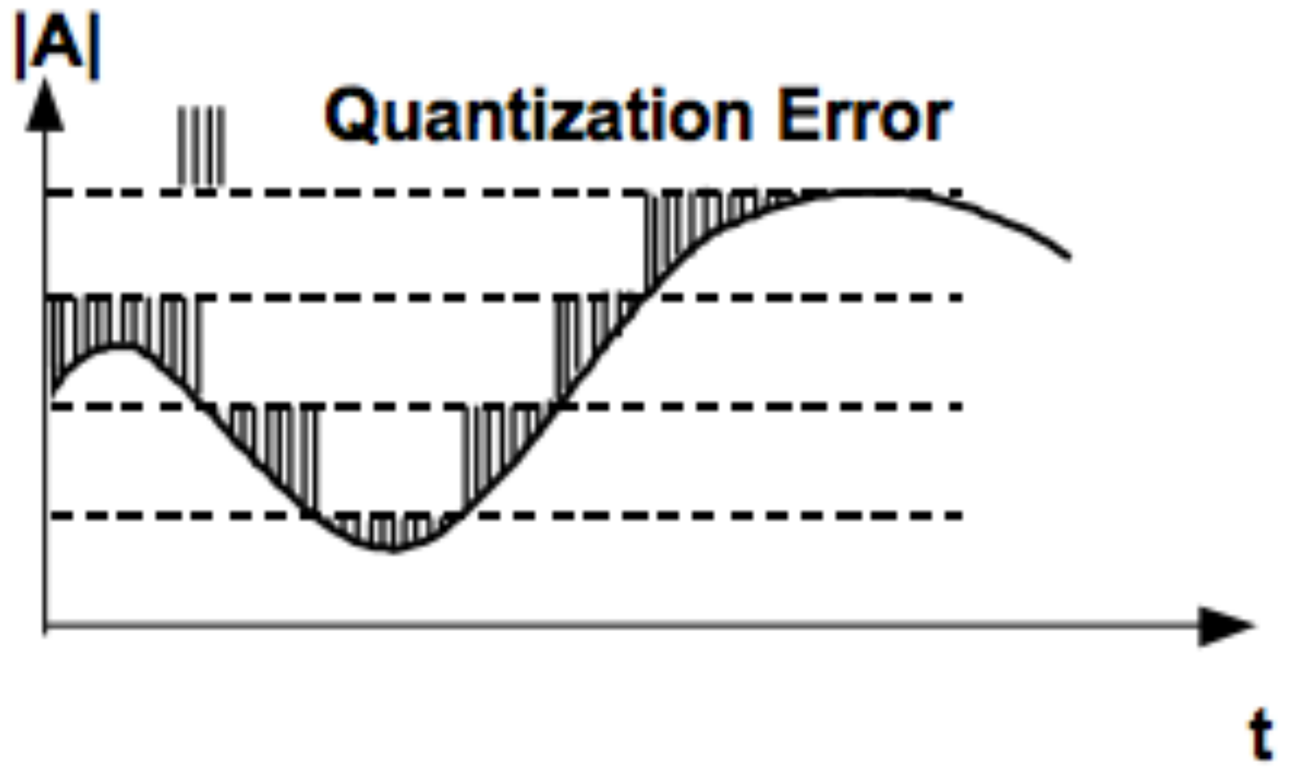
Quantization

- Quantization converts dependent variable from continuous to discrete
  - Any one sample in digitized signal can have max error of 1/2 LSB
  - Quantization error looks like random noise
  - Number of bits determines precision of data
  - Doesn’t work when analog signal remains same value for many consecutive samples -> Dithering

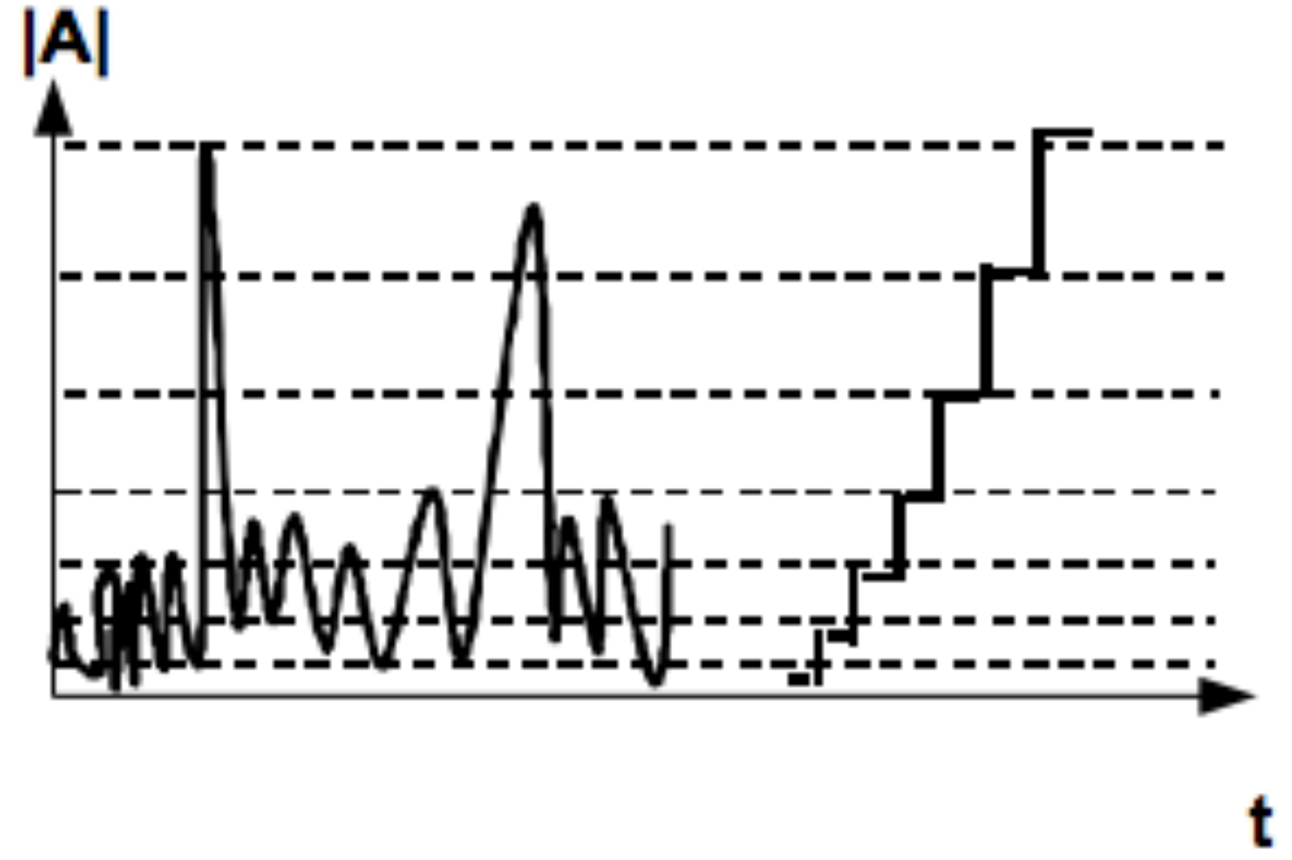
Quantization

- Classifying the signal into certain bands.
- Consider 2-bit quantizer
  - Quantizer determines where the amplitude of a particular sample falls within four levels.
    - 00, 01, 10, 11
    - “Always round up”
- Accuracy: more levels, more accuracy
- Sampling converts independent variable from continuous to discrete.

Quantization Error

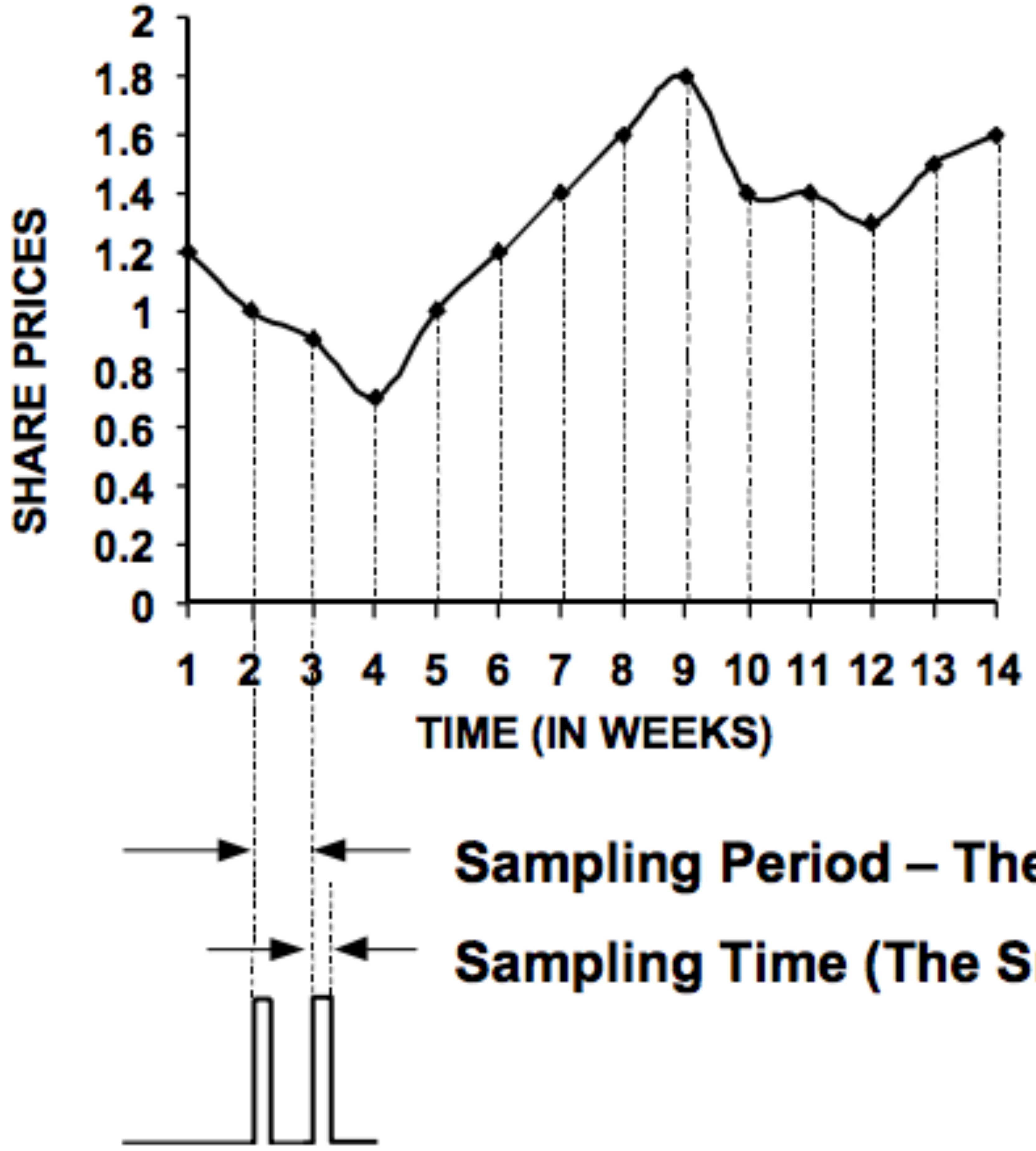


- **Quantization introduces errors**
- **Increasing the number of quantization levels is not always the answer**



- **Non-uniform Quantization**
  - **Use more levels where there are more variations**
  - **Use fewer levels where there are fewer variations**

Sampling: Stock Market



- **Take snapshots of continuously changing data**
- **The Sampling Period is fixed**
- **This makes information understandable**
- **My share price hit its lowest point in week 4**
- **My share price reached its peak in week 9**

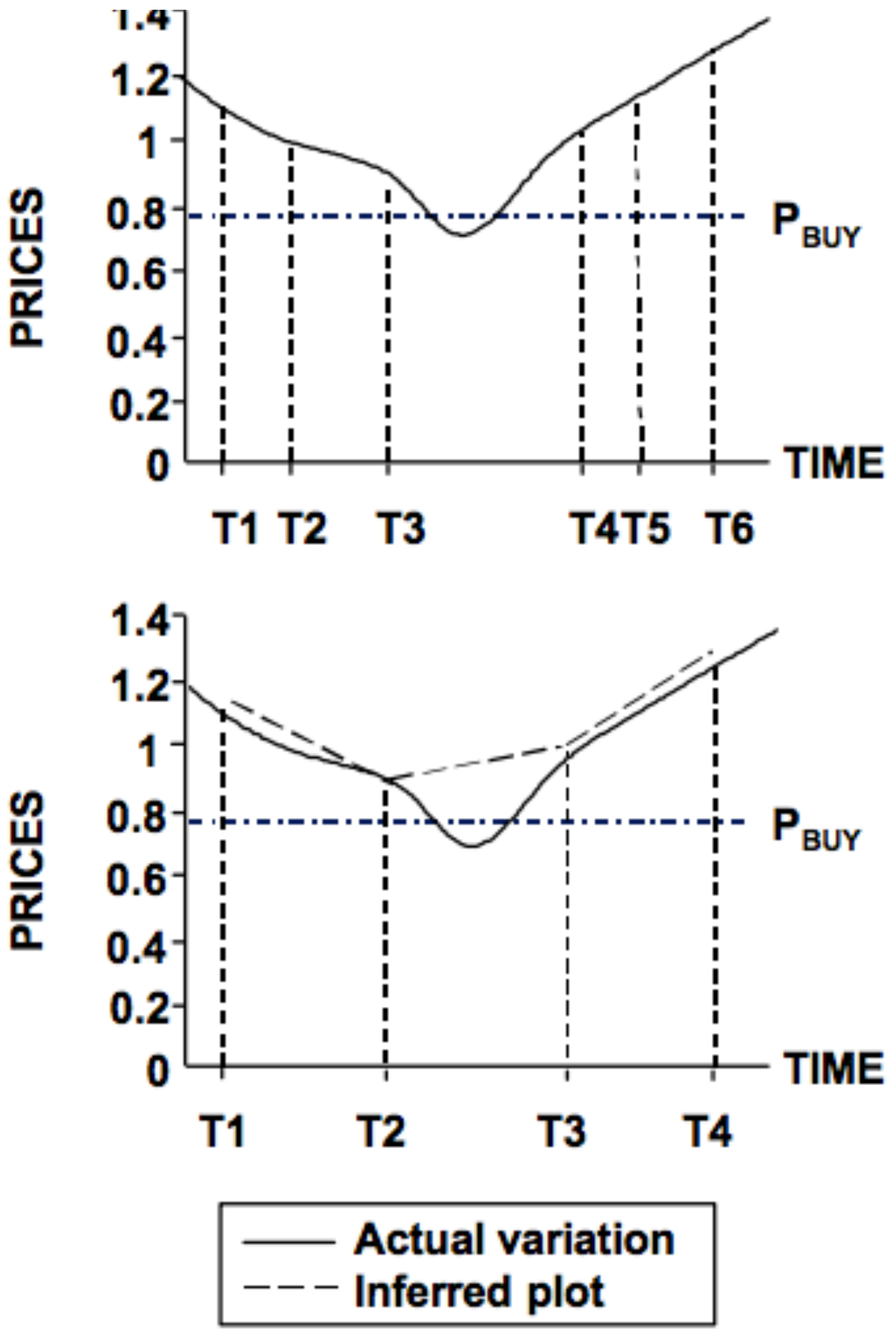
Sampling Period – The period between samples

Sampling Time (The Snapshot) – The time taken to take a sample

Sampling: Stock Market

- Need to decide how often to take samples to gain accurate representation of changes in stock market
  - Sample every day to give more precise representation of fluctuations. But do prices change significantly enough to warrant more frequent sampling rate?
  - Sample every minute, but values could produce unwanted noise
  - Sample every month/year, but miss high and low values
  - Sample sometimes daily, sometimes weekly, or sometimes every 10 days.

Missing Information



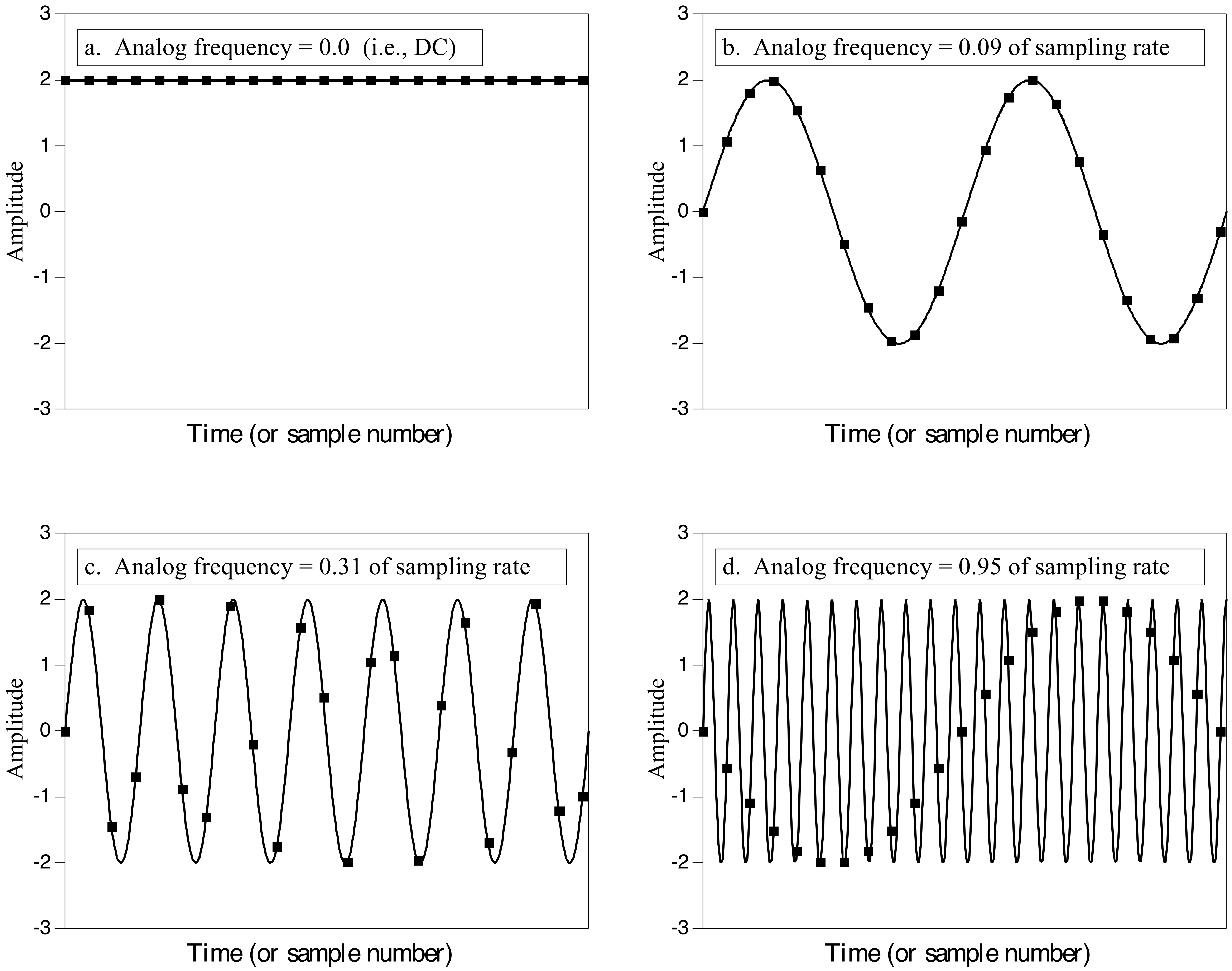
- **Non-periodic Snapshots**
  - **May miss information**

    The dip in prices between T3 and T4 goes unnoticed
  - **Information cannot be interpreted easily**
- **Periodic Snapshots**
  - **May miss information**

    The dip in prices between T2 and T3 goes unnoticed
  - **Easier to interpret**

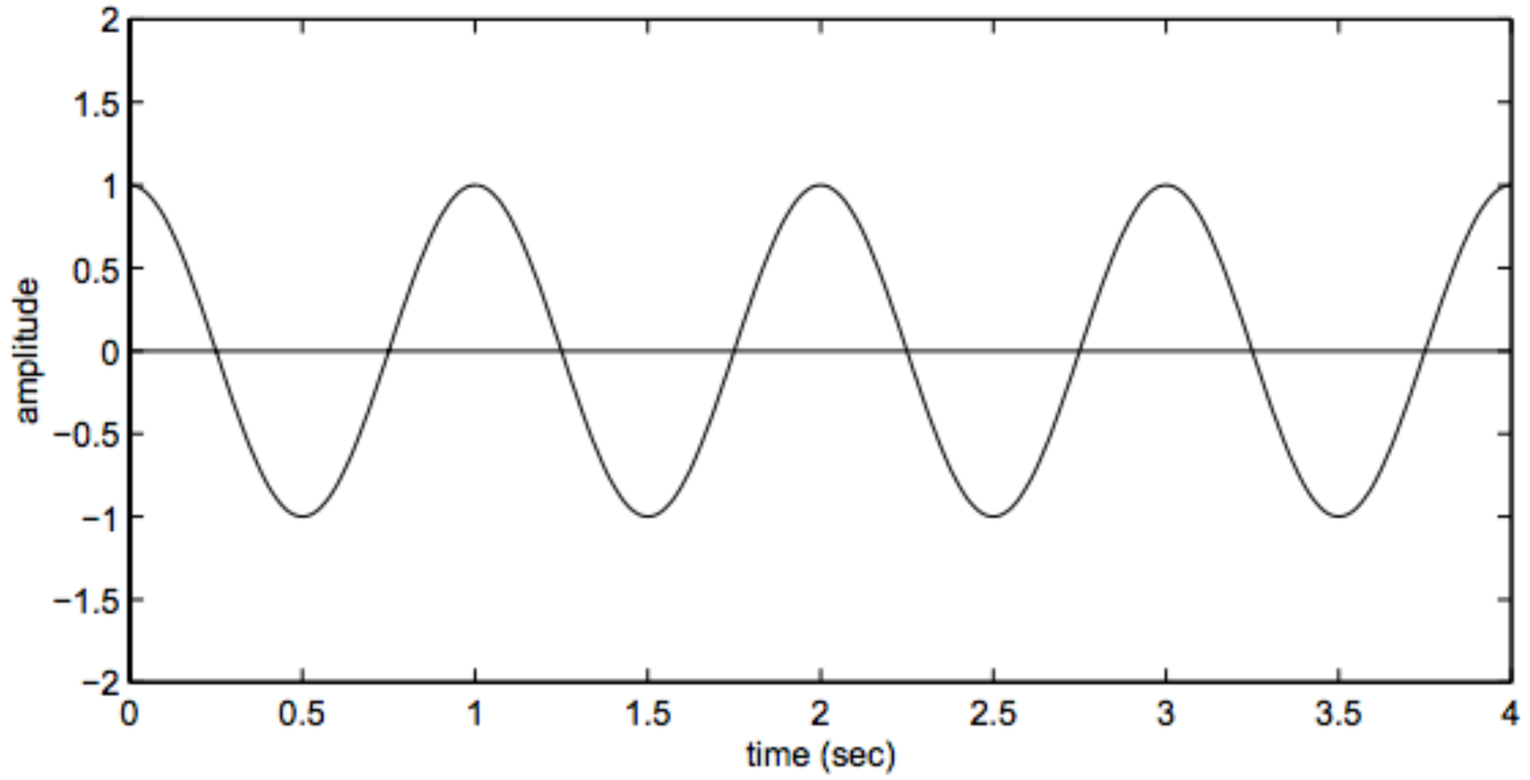
The key is the sampling frequency.

Sampling: If you can exactly reconstruct the analog signal from samples, you have done the sampling properly.


a. Analog frequency = 0.0 (i.e., DC)
b. Analog frequency = 0.09 of sampling rate
c. Analog frequency = 0.31 of sampling rate
d. Analog frequency = 0.95 of sampling rate
Amplitude
Time (or sample number)
3
2
1
0
-1
-2
-3

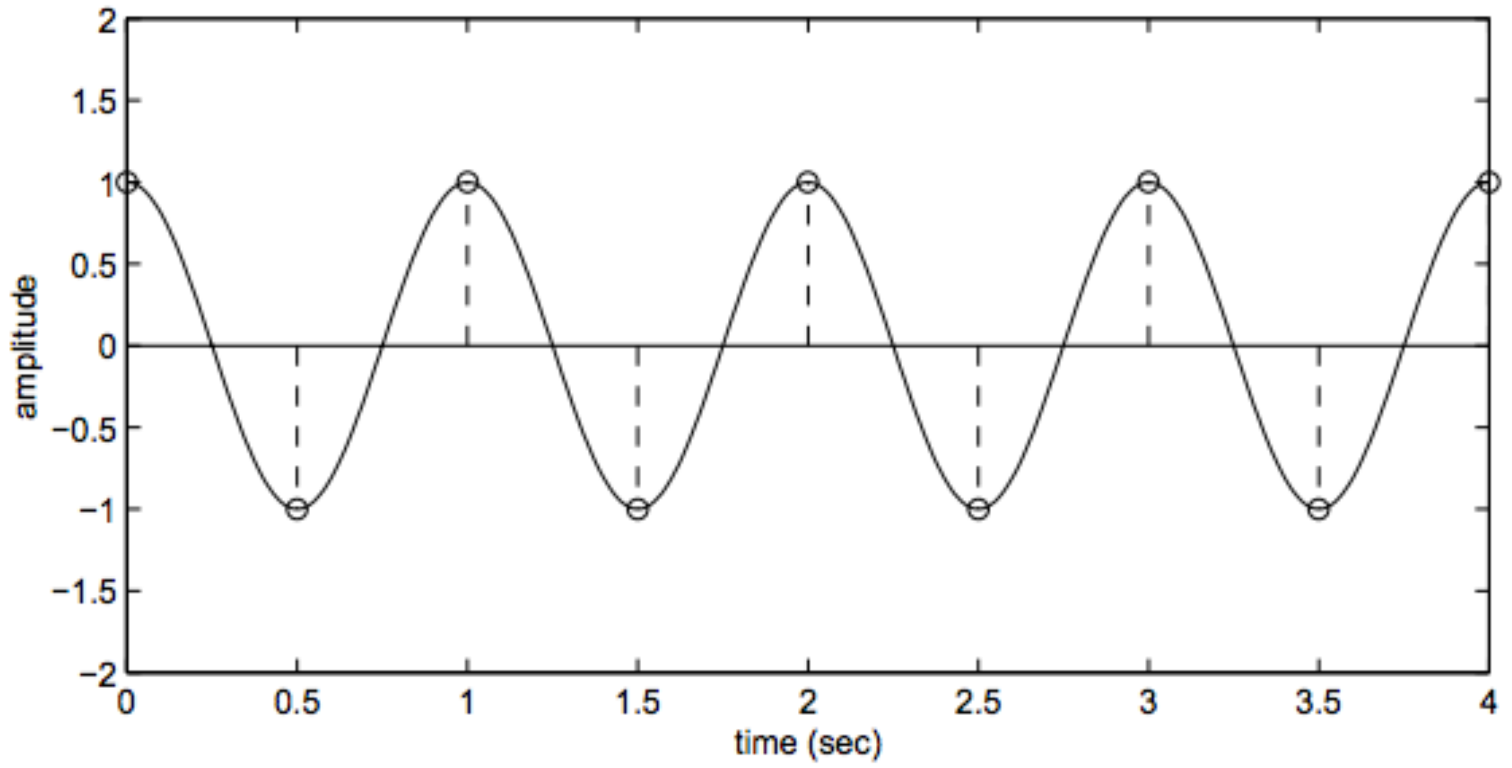
Nyquist Sampling Theorem

- ***The sampling frequency should be at least twice the highest frequency contained in the signal***
  - $f_s >= 2f_m$
  - f_s is the sampling frequency
  - f_m is the highest frequency contained in signal

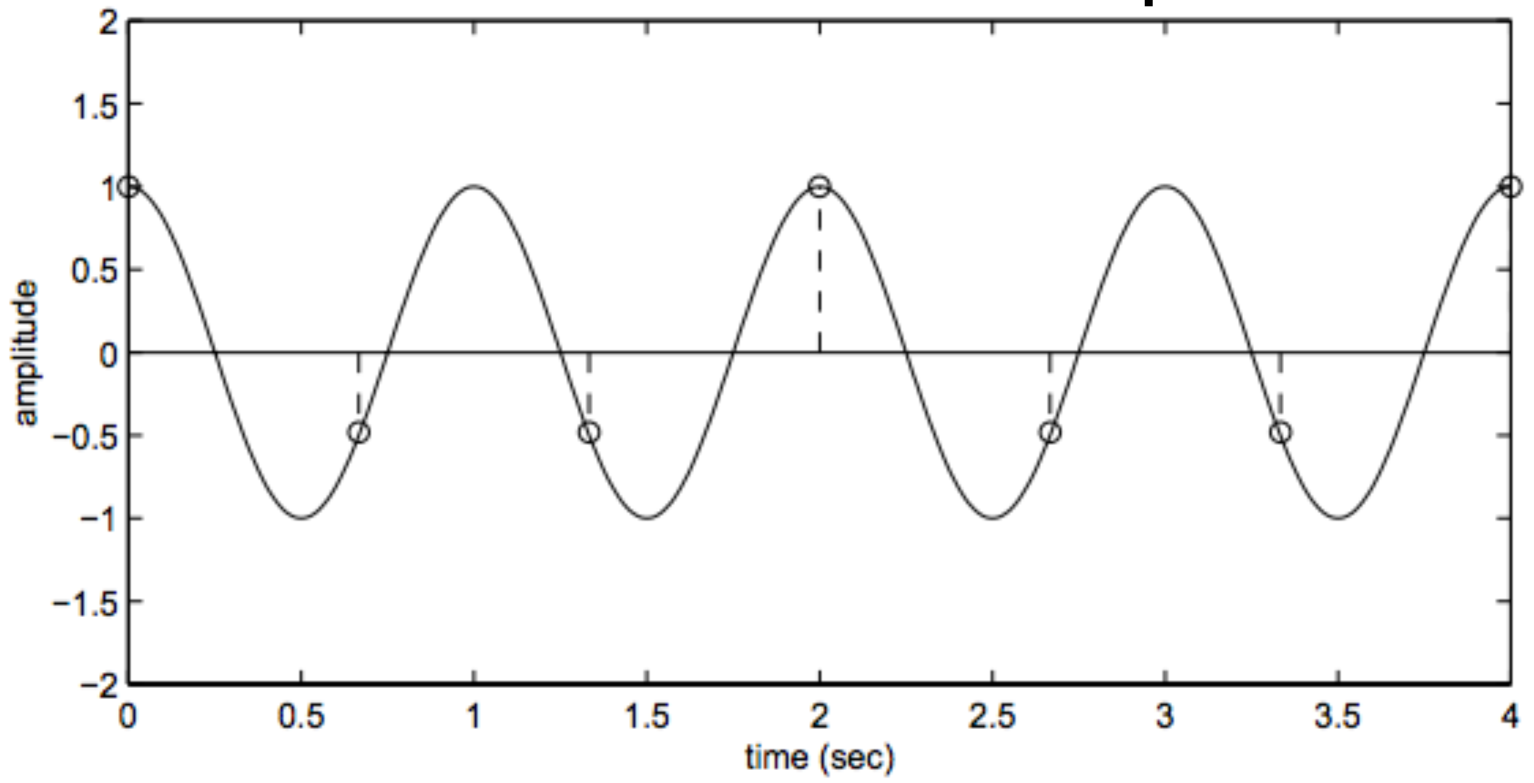


1 Hz Signal

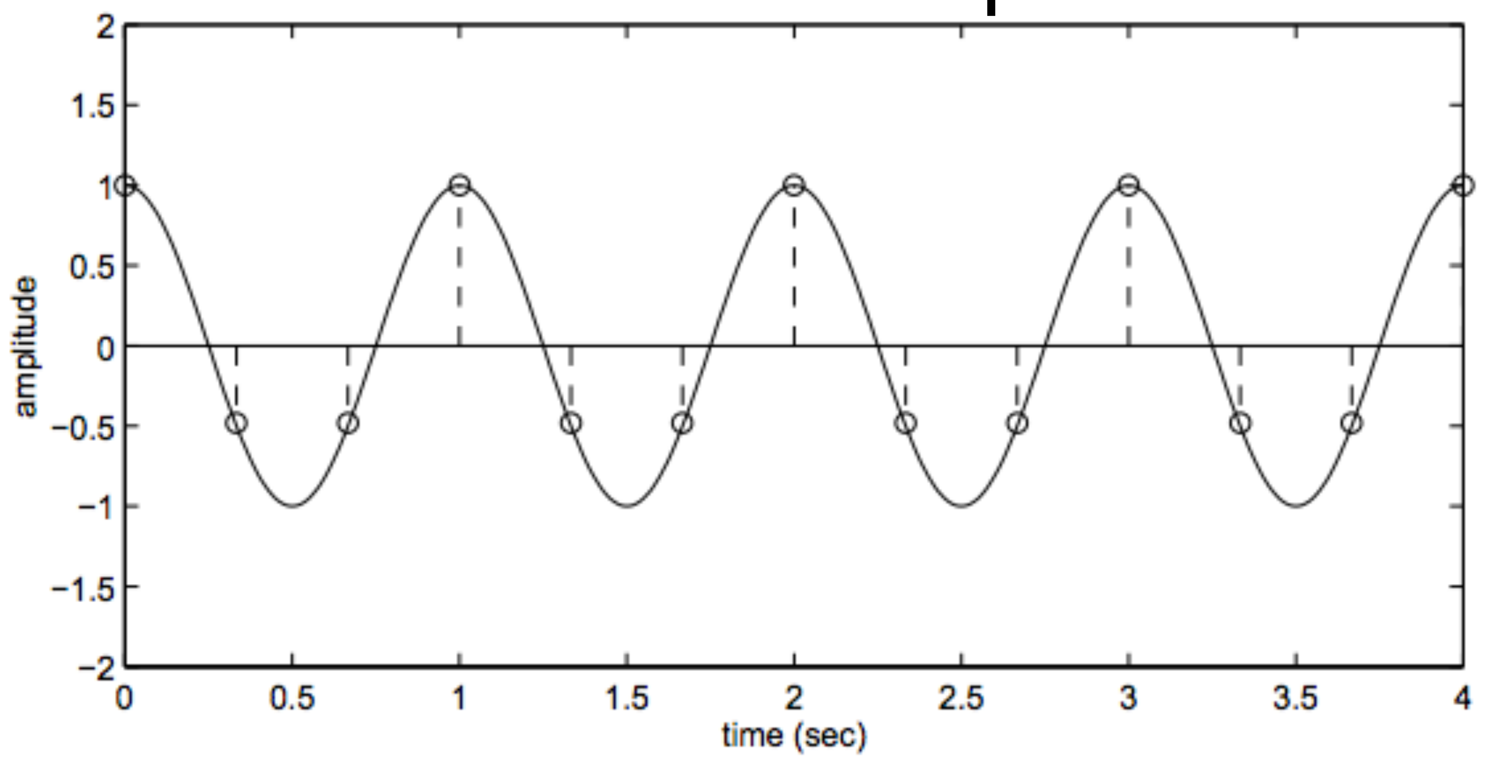
2 Hz sample



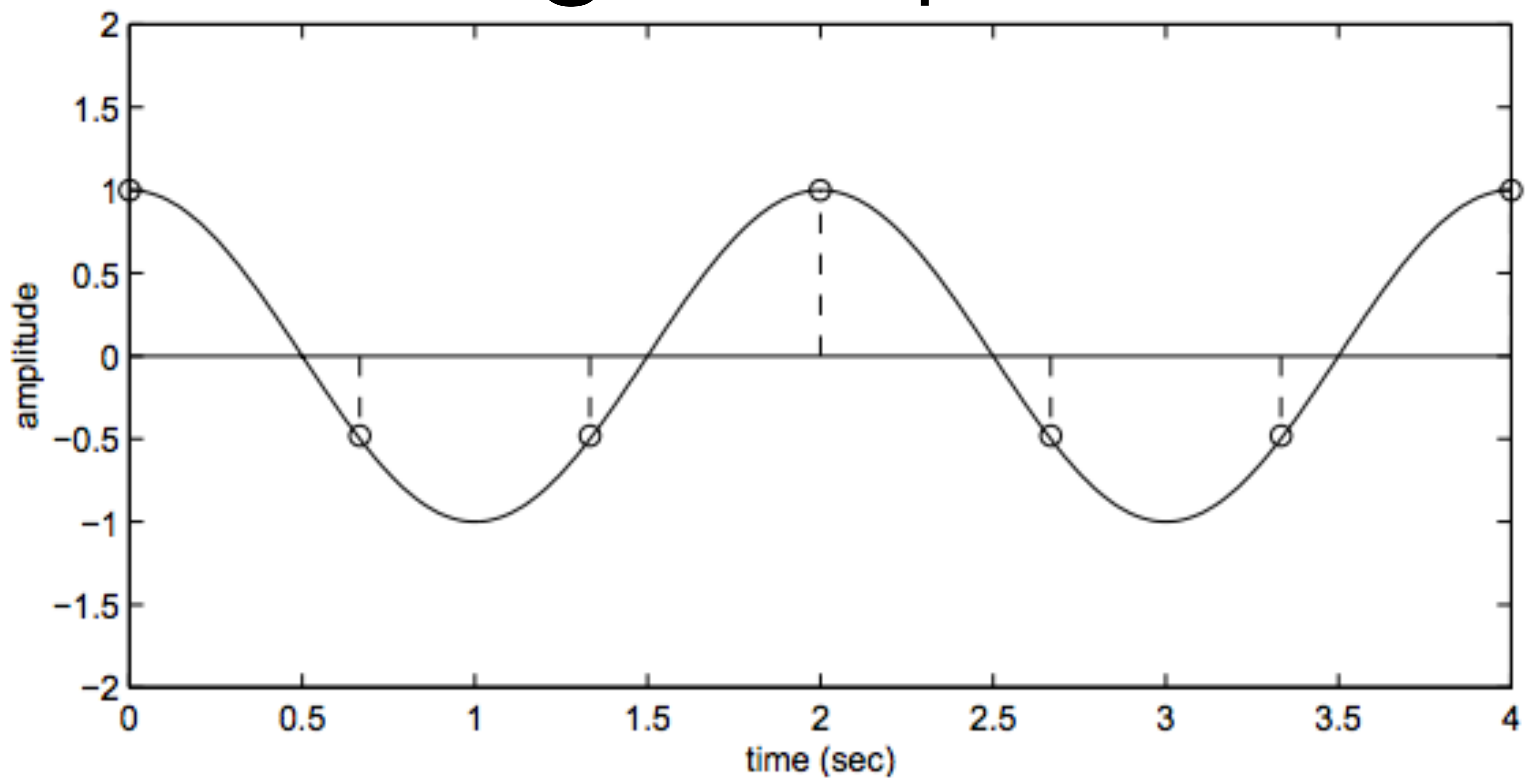
1.5 Hz sample

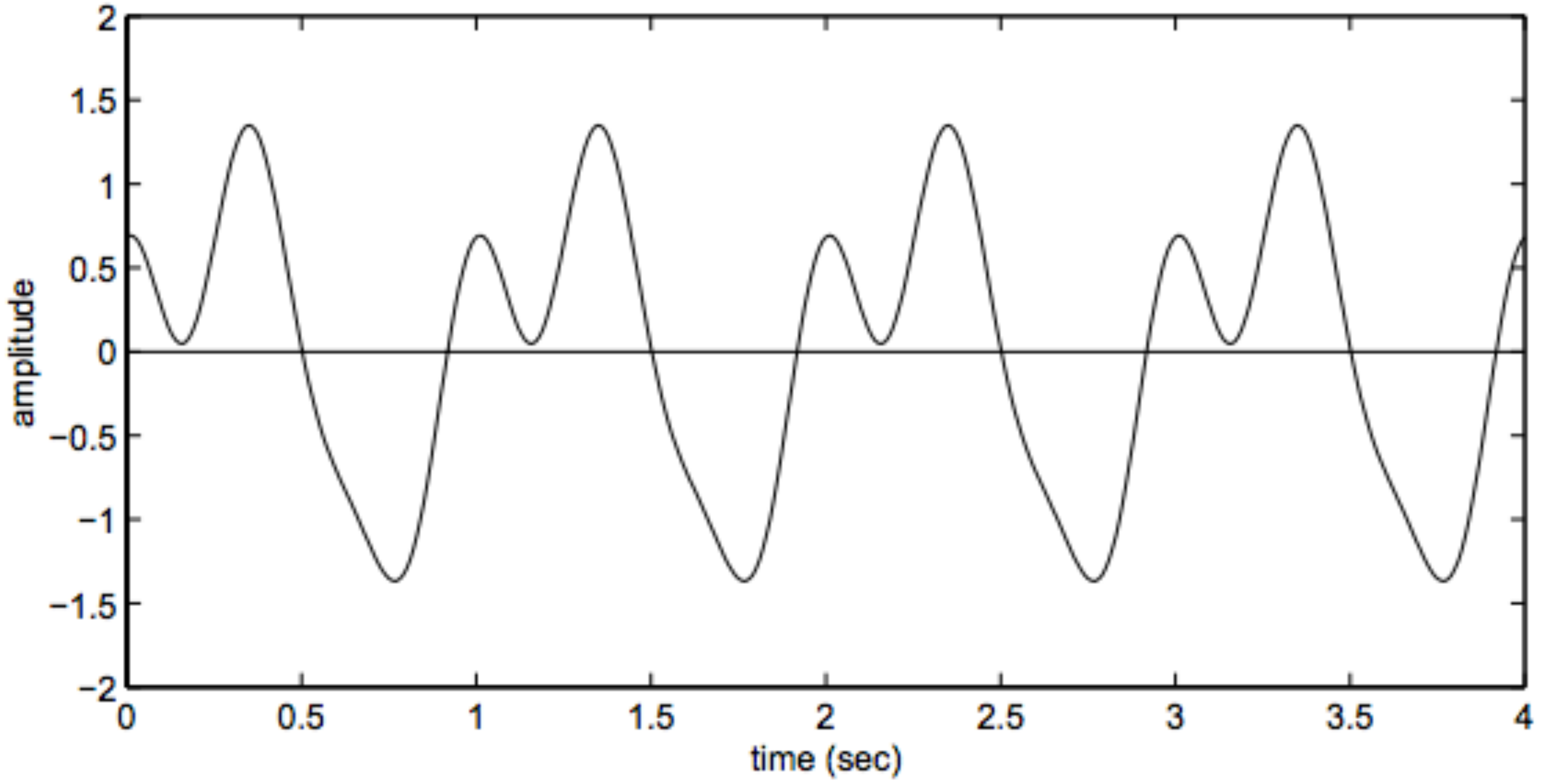


3 Hz sample

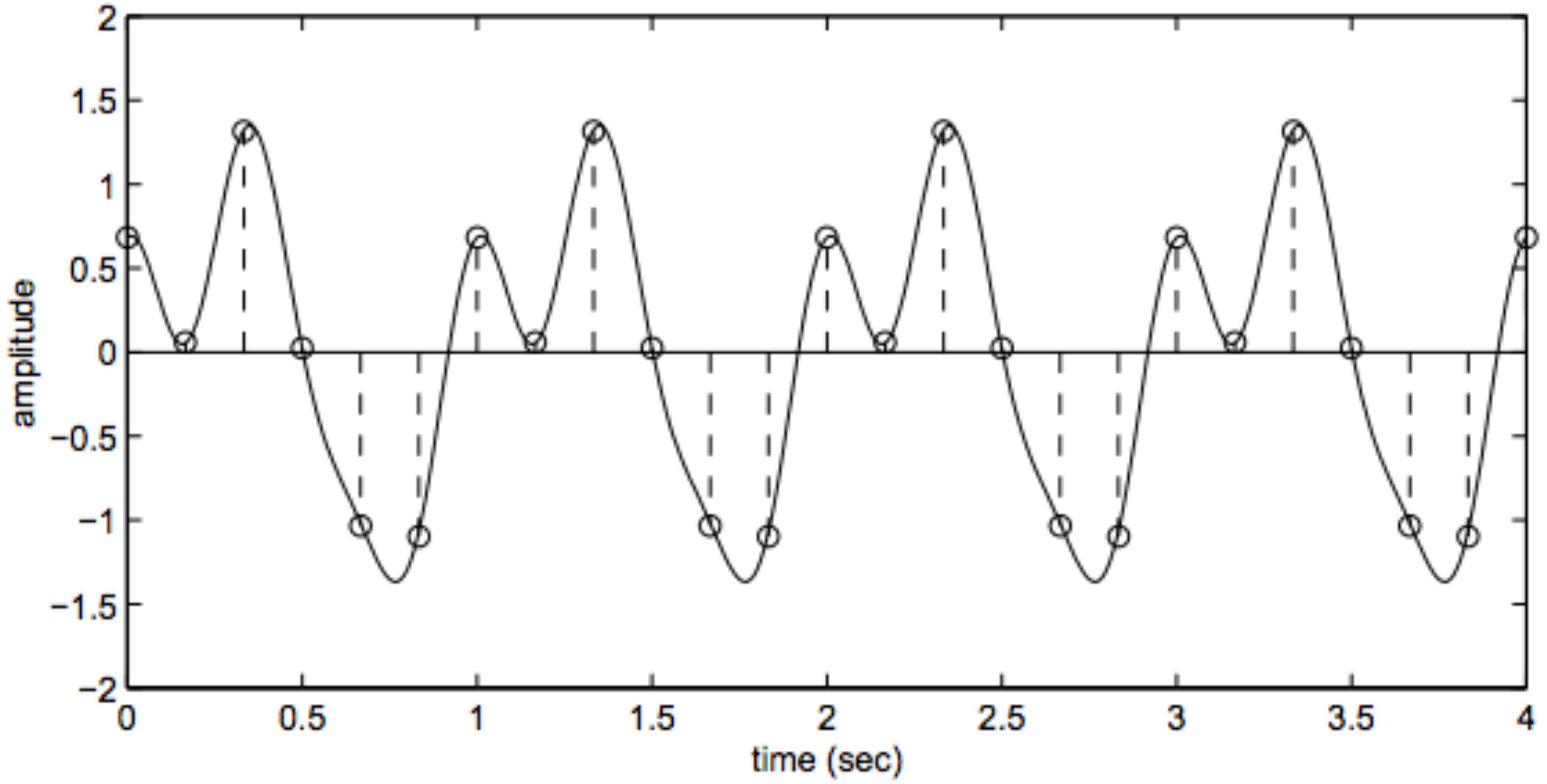


wrong interpretation



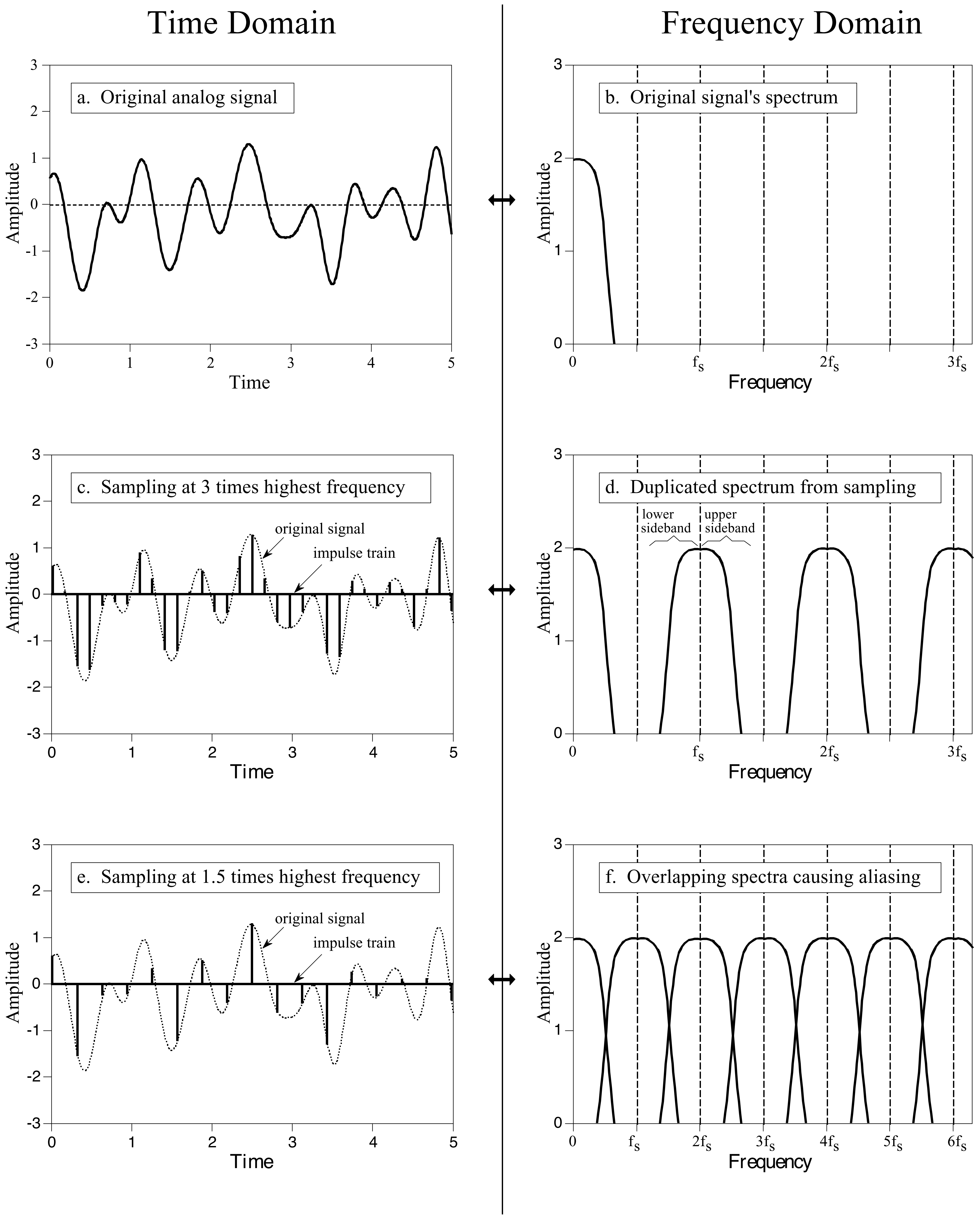


1 Hz, 2 Hz, 3 Hz



What is f_s?

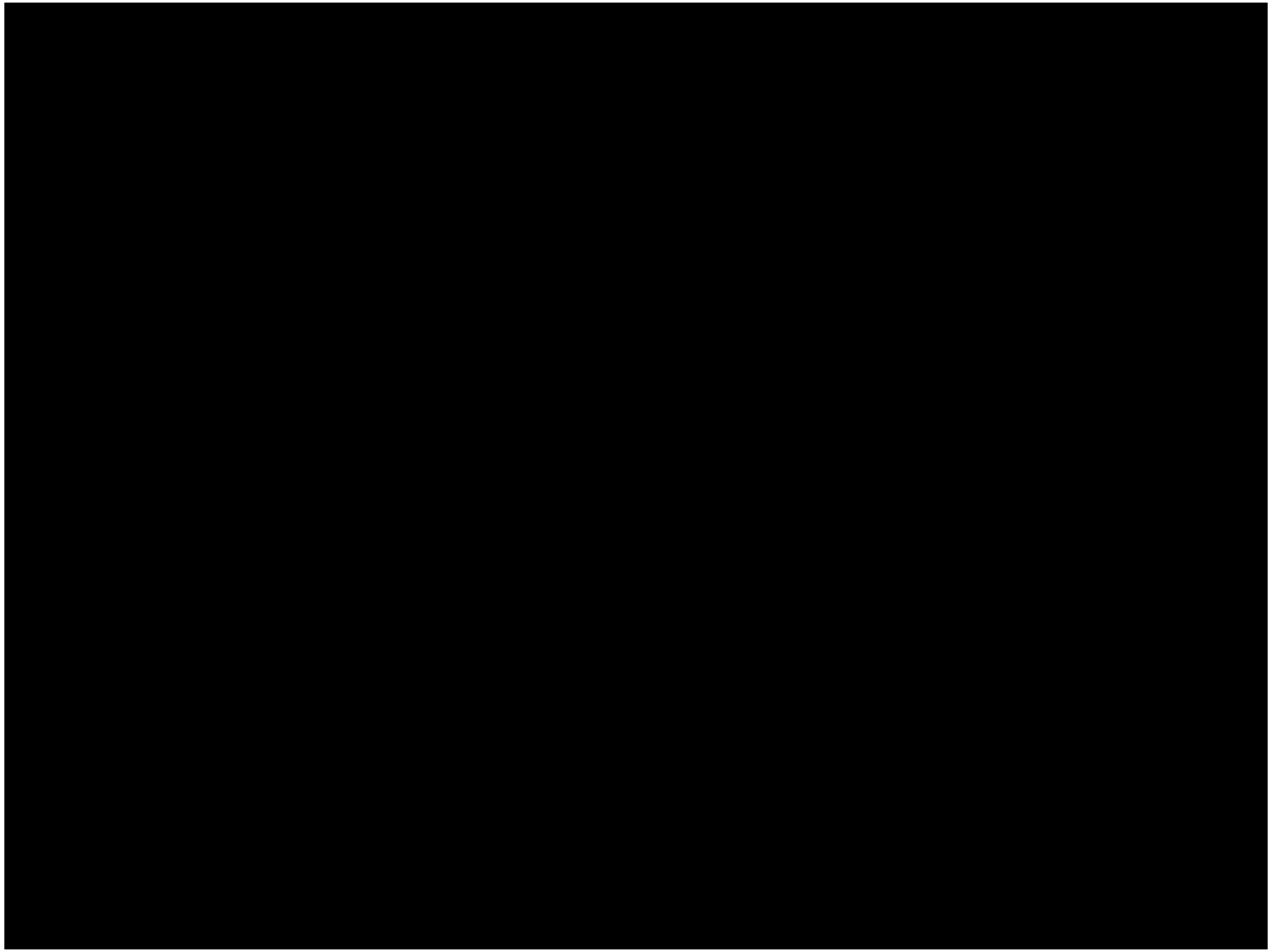
6 Hz



Time Domain
Frequency Domain
a. Original analog signal
b. Original signal's spectrum
c. Sampling at 3 times highest frequency
d. Duplicated spectrum from sampling
e. Sampling at 1.5 times highest frequency
f. Overlapping spectra causing aliasing
original signal
impulse train
lower sideband
upper sideband
Amplitude
Time
Frequency
f_s
$2f_s$
$3f_s$
$4f_s$
$5f_s$
$6f_s$

Aliasing/Quantization Error

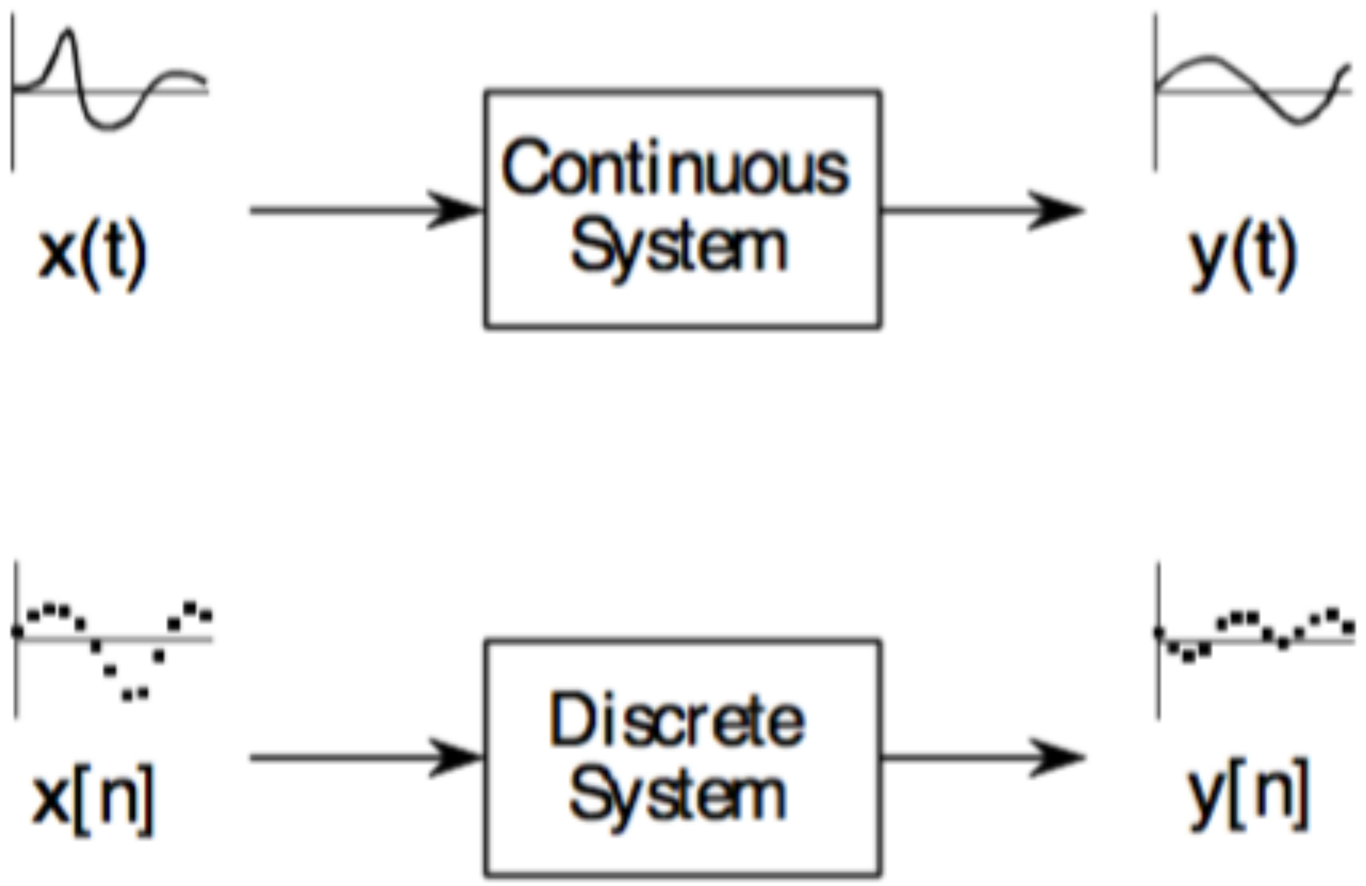
https://www.youtube.com/watch?v=UaKho805vCE



Fundamentals of Linear Systems

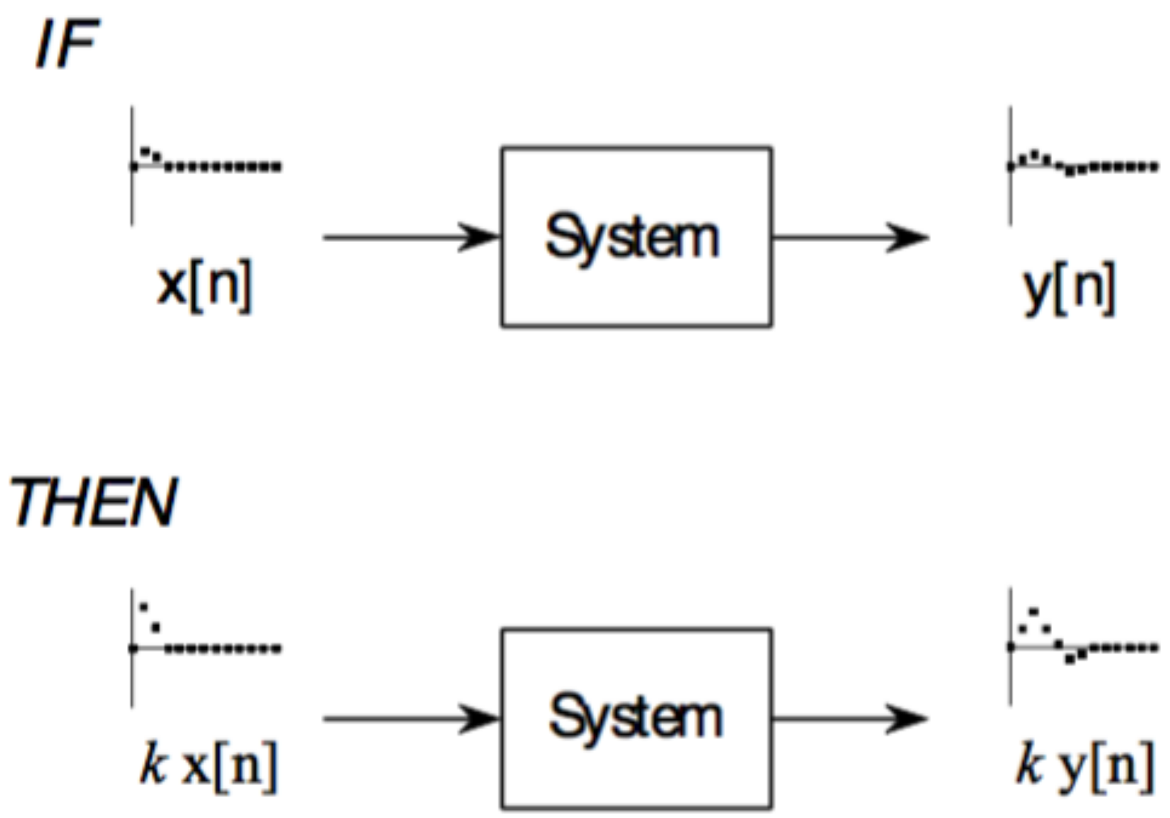
Signals and Systems

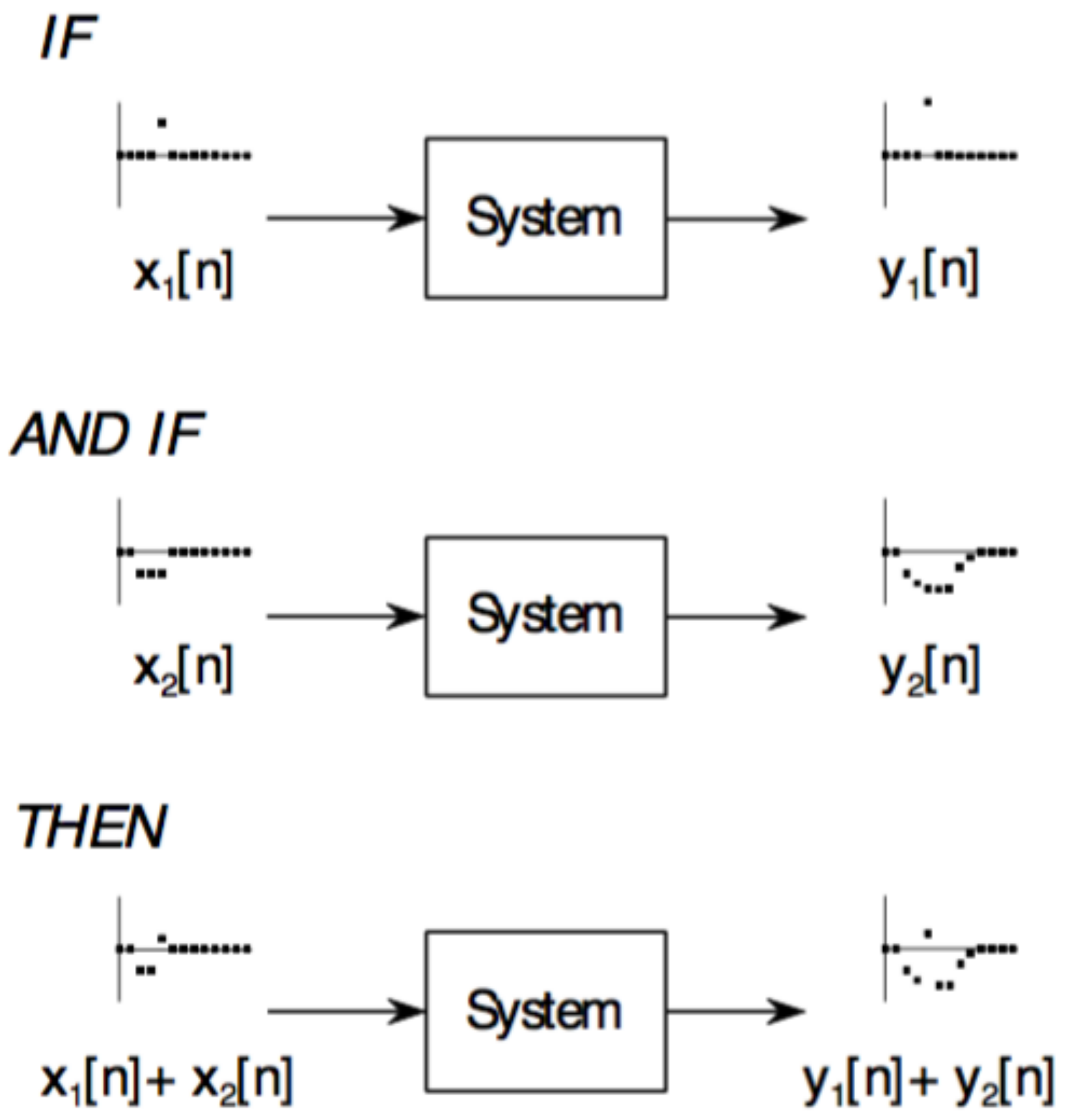
- Signal: description of how one parameter varies with another parameter
- System: process that produces an output signal in response to an input signal
- **If you understand how system is changing, you can compensate for the effect**

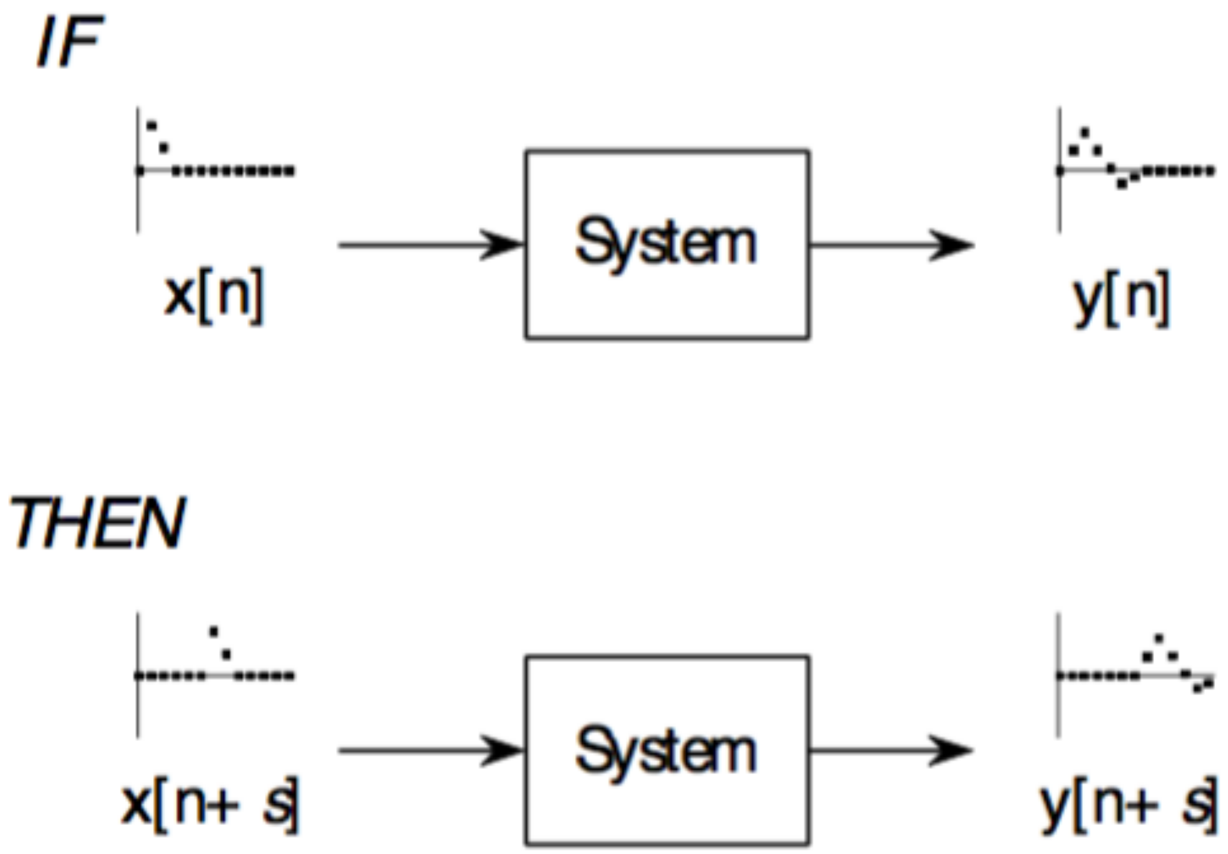


Requirements for Linearity

1. Homogeneity
2. Additivity
3. Shift Invariance

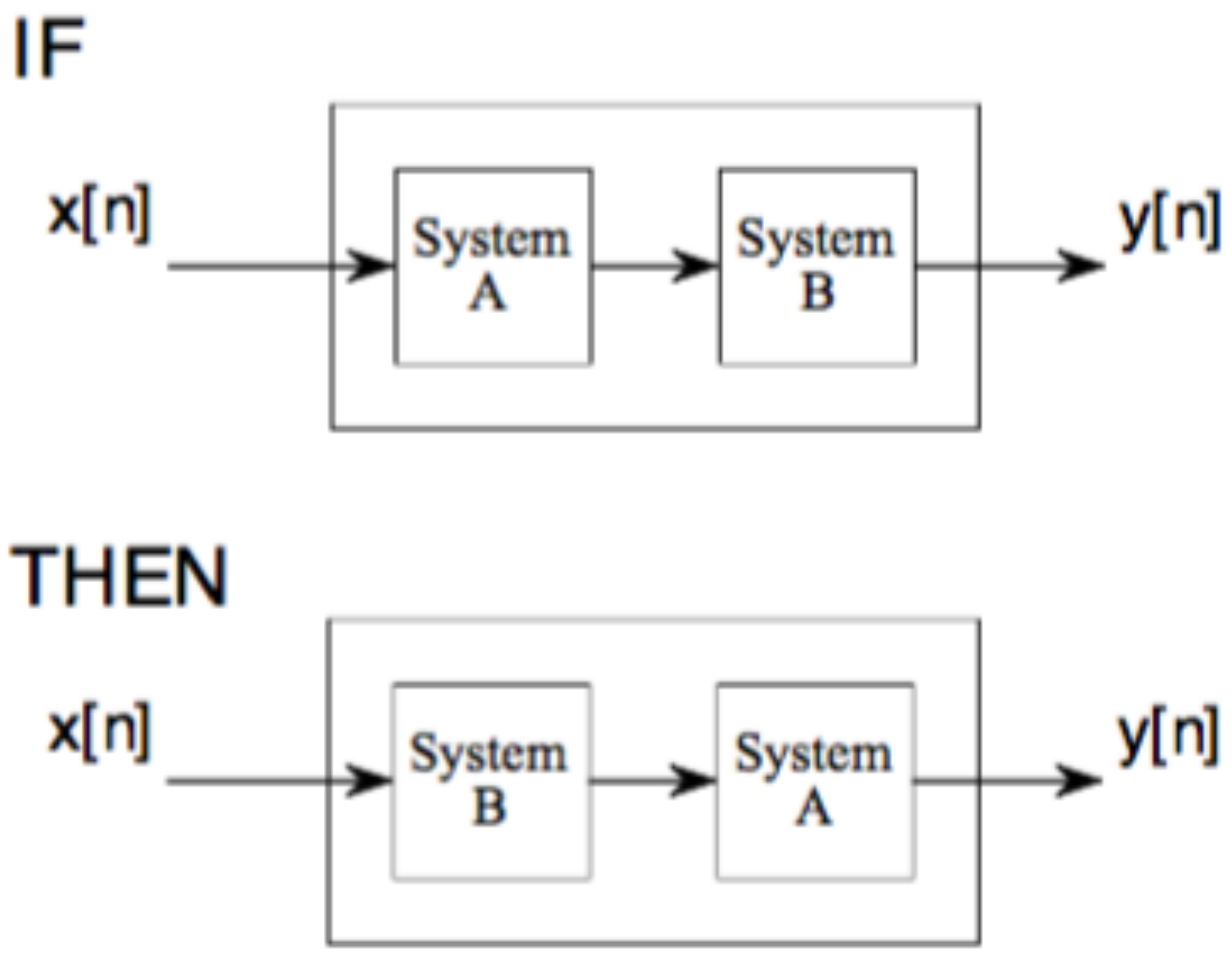






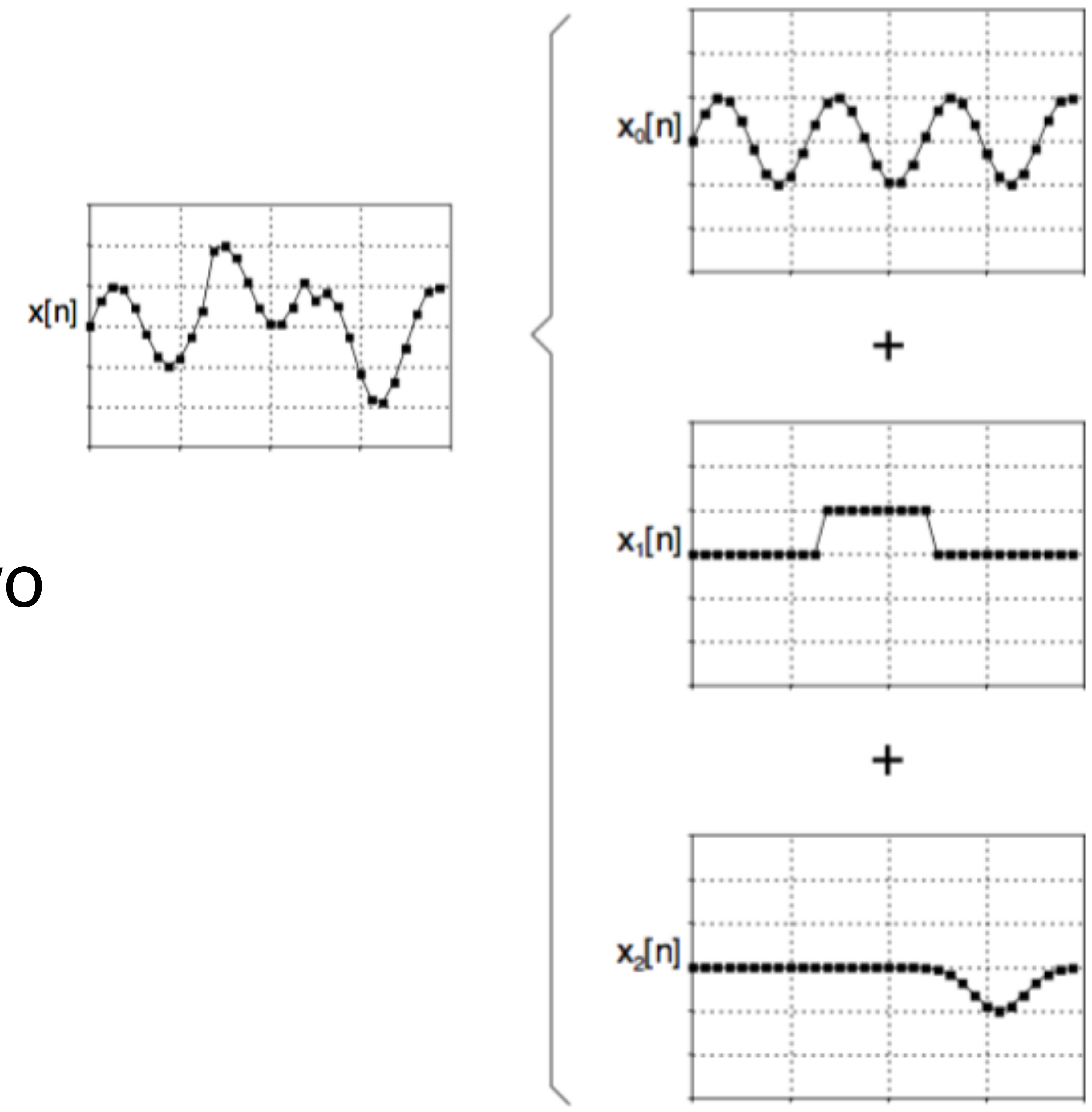
Special Properties of Linearity

- Commutative: order of systems in cascade can be rearranged without affecting overall combination

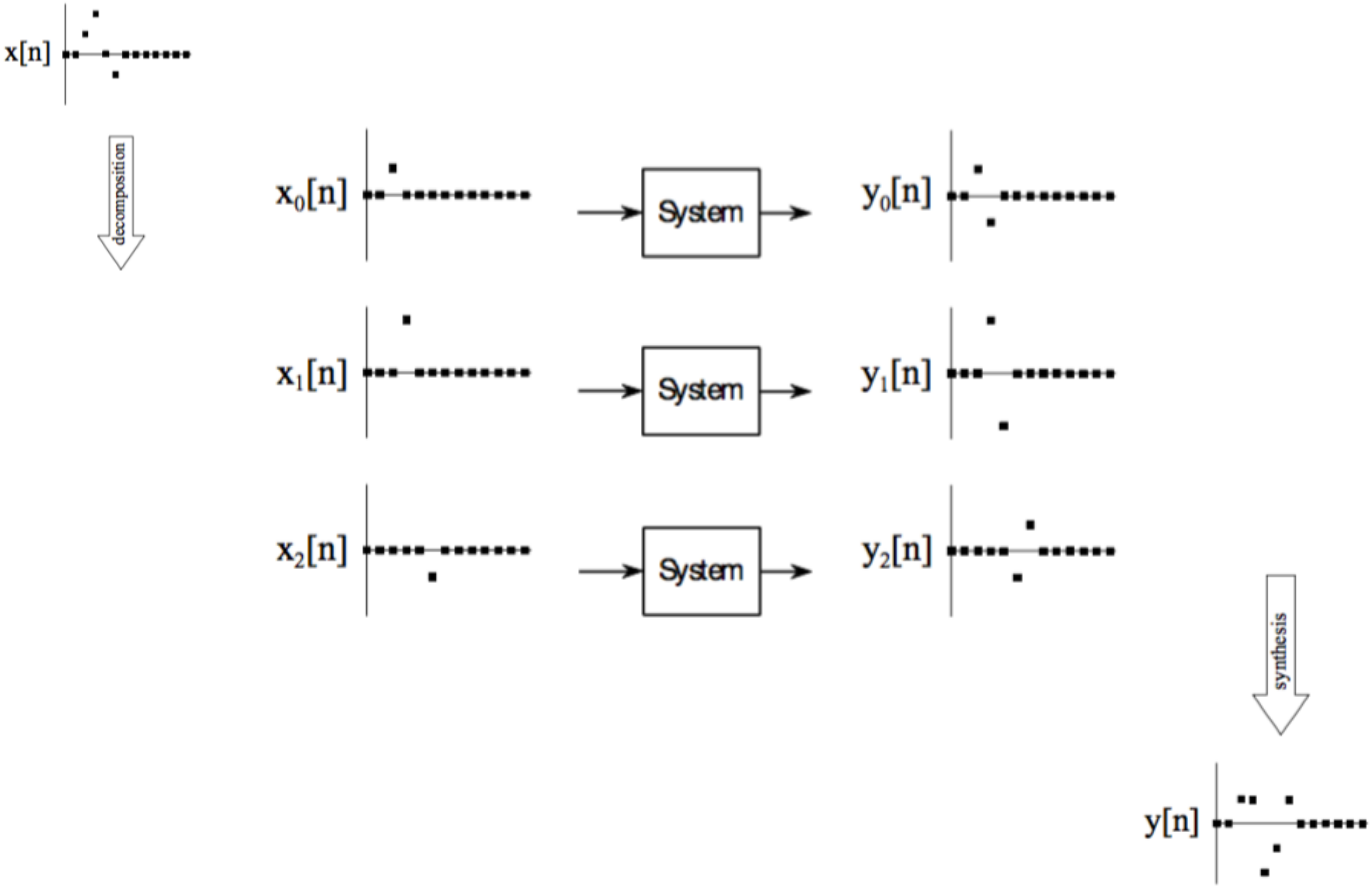


Superposition

- Synthesis: process of combining signals through scaling and addition
- Decomposition: inverse of synthesis
  - Single signal broken into two or more additive components
  - More involved because infinite possible decompositions
- Superposition: overall strategy for understanding how signals and systems can be analyzed



Superposition

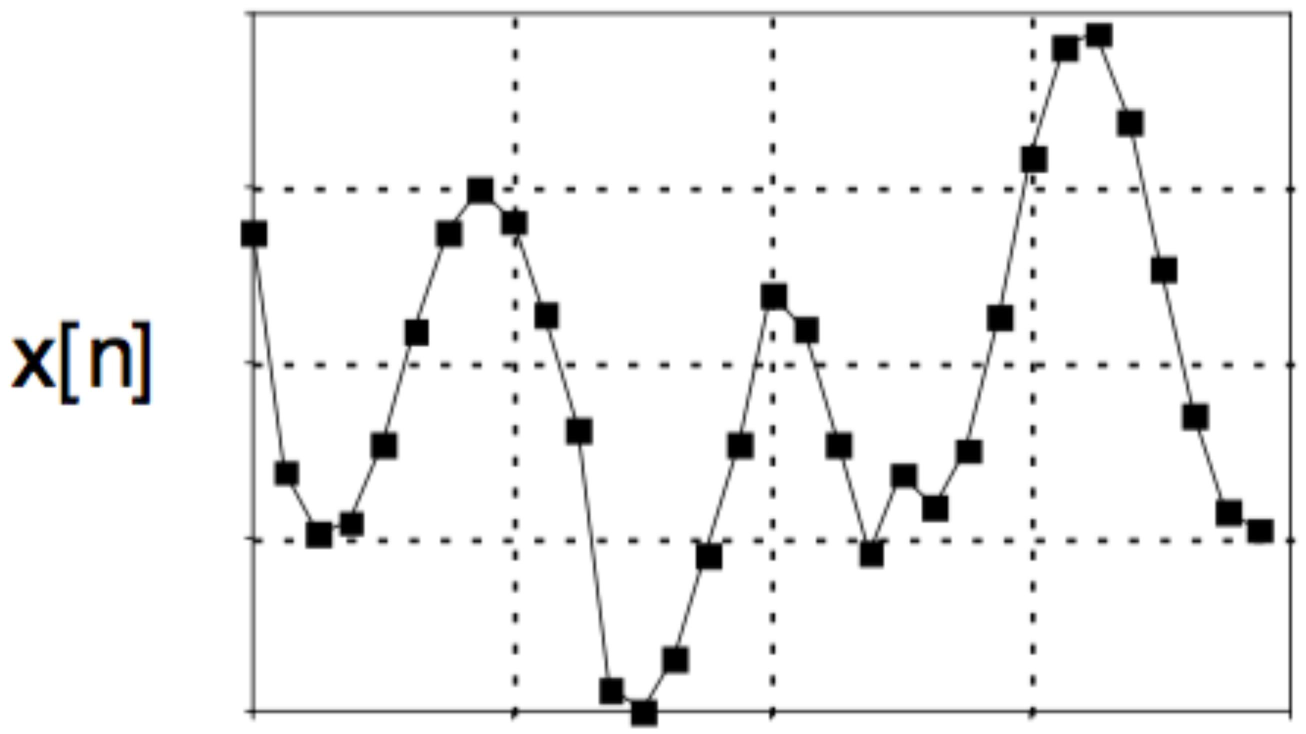


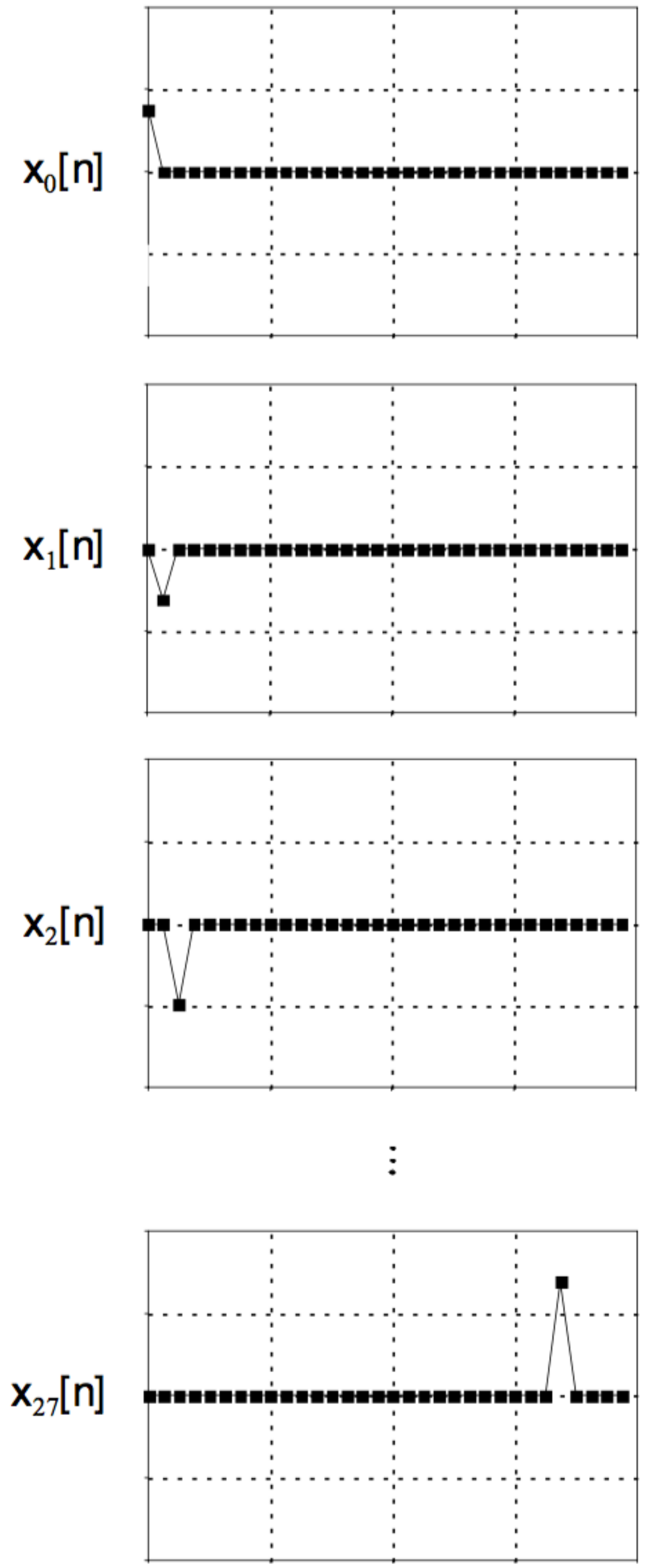
Common Decompositions

- Two main ways to decompose signals
  - Impulse decomposition
  - Fourier decomposition

Common Decompositions

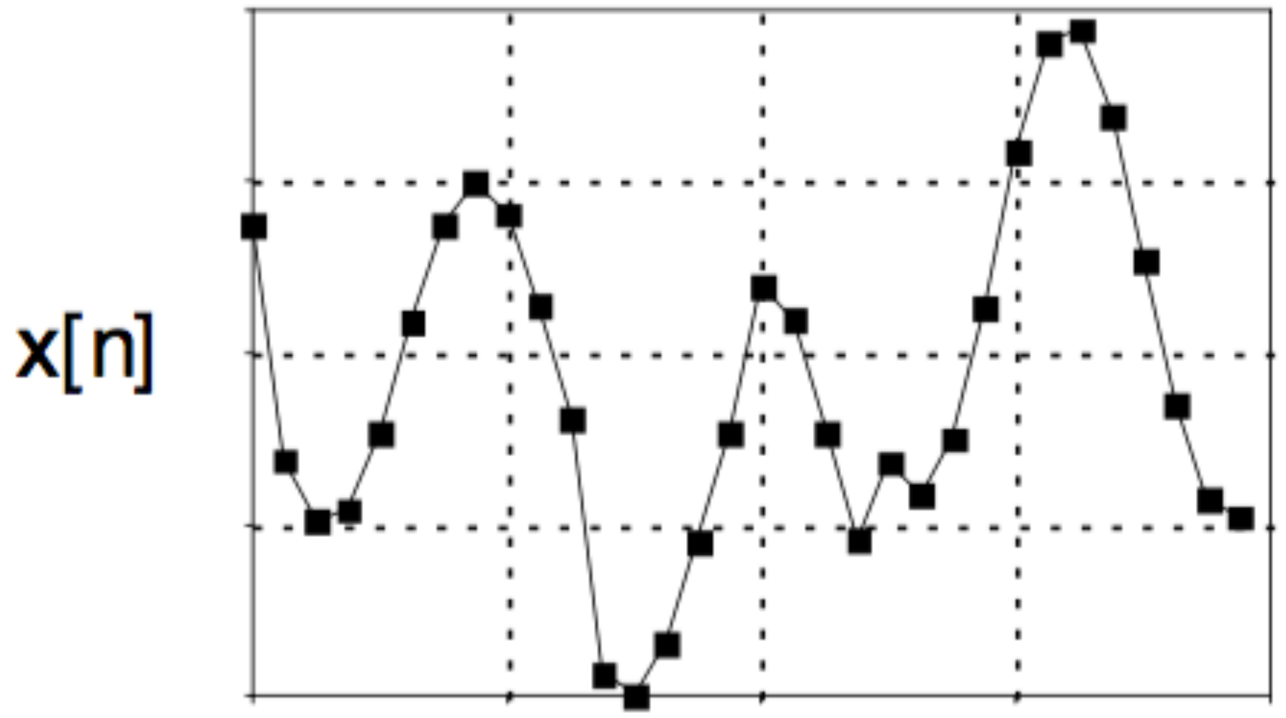
Impulse Decomposition

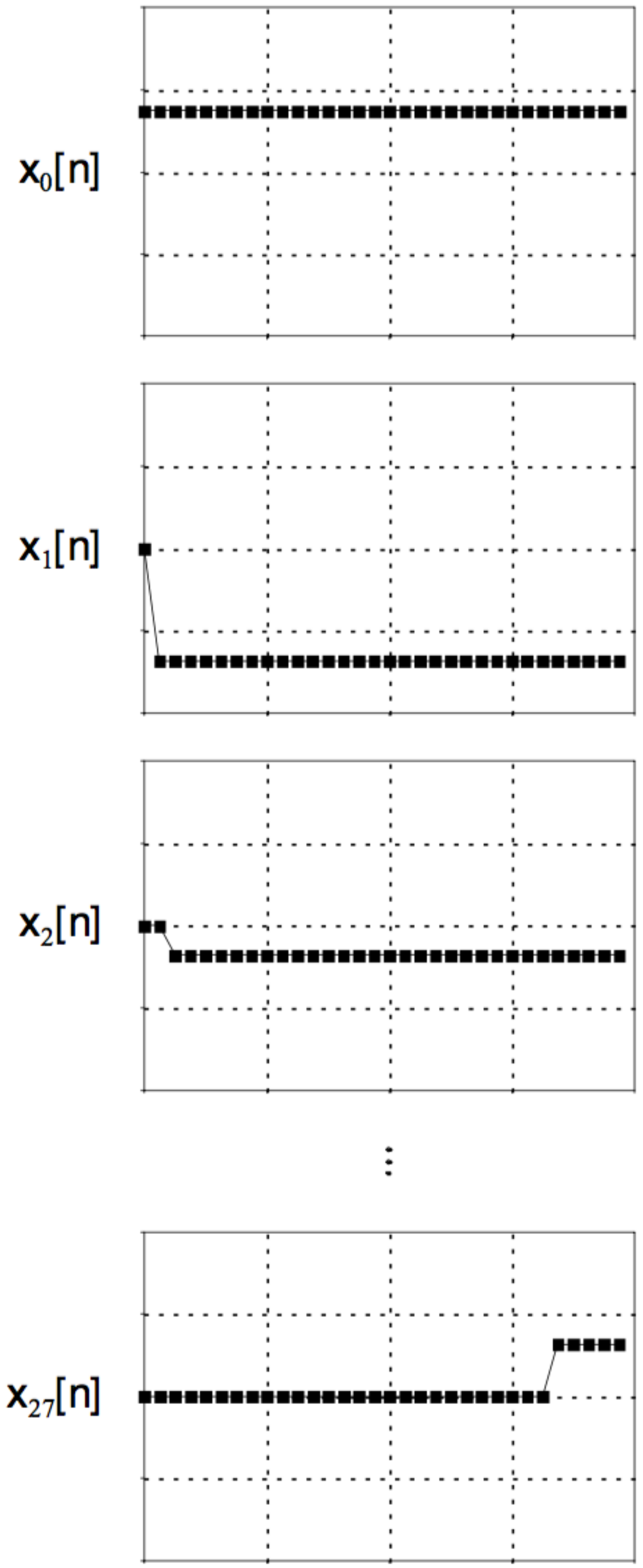




Common Decompositions

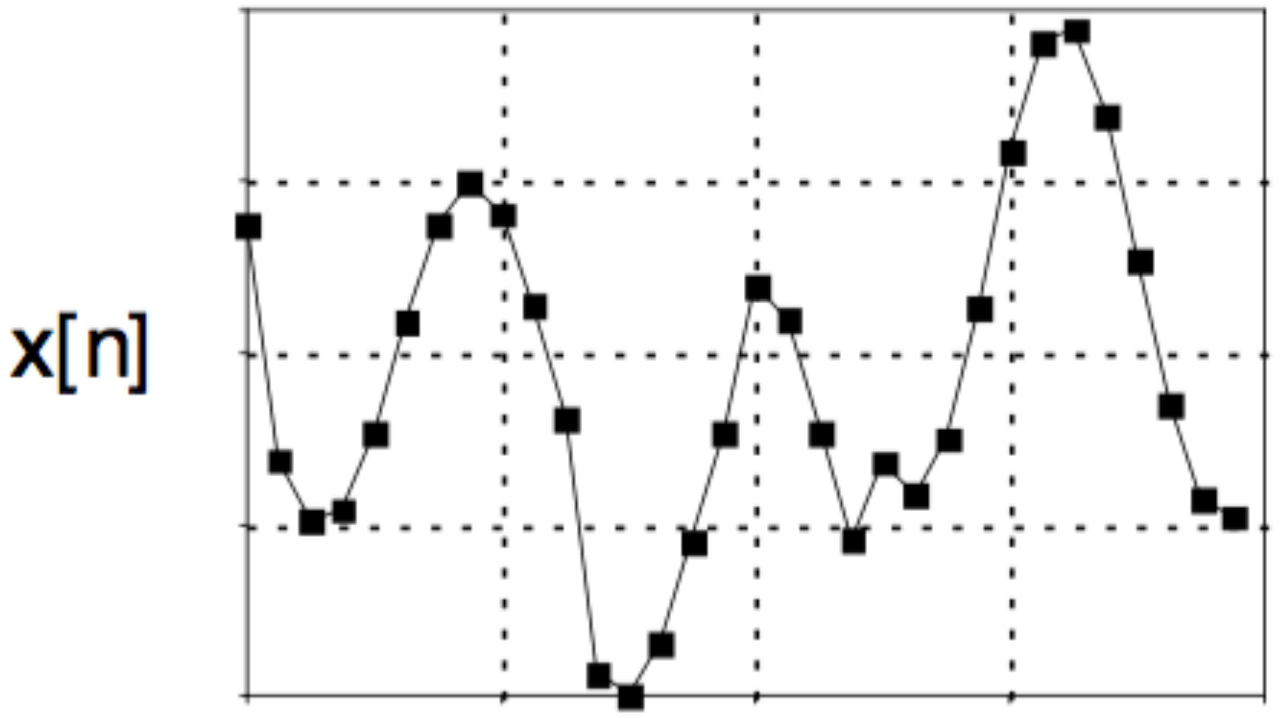
Step Decomposition

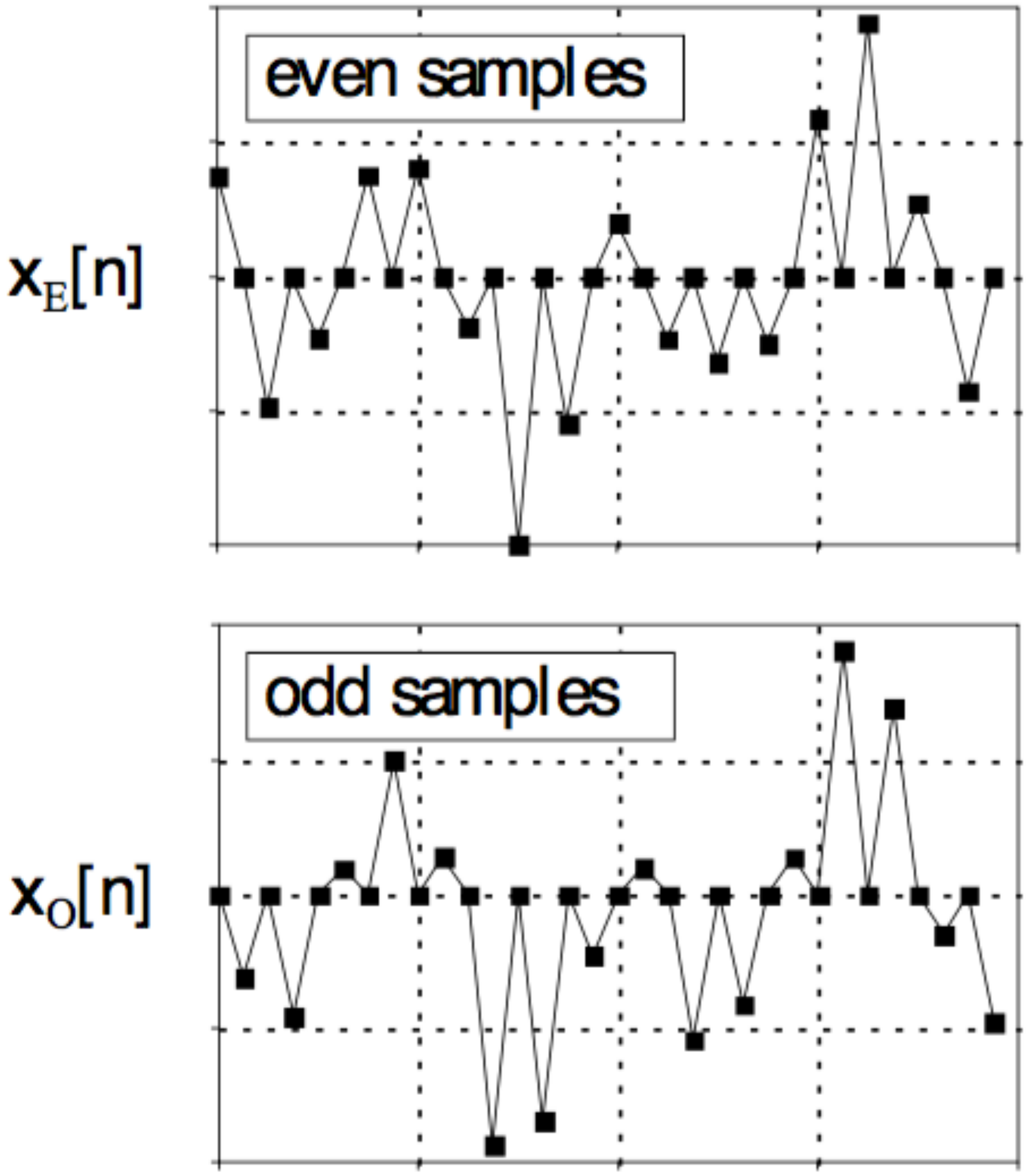




Common Decompositions

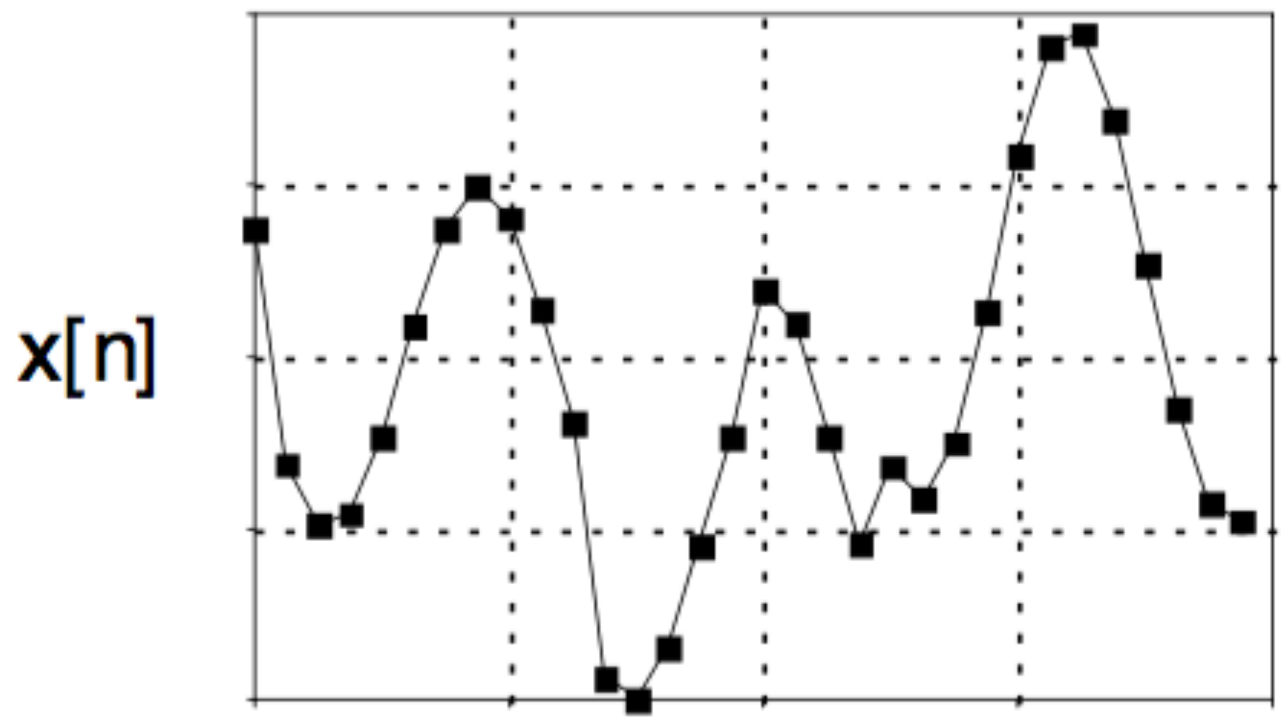
Interlaced Decomposition

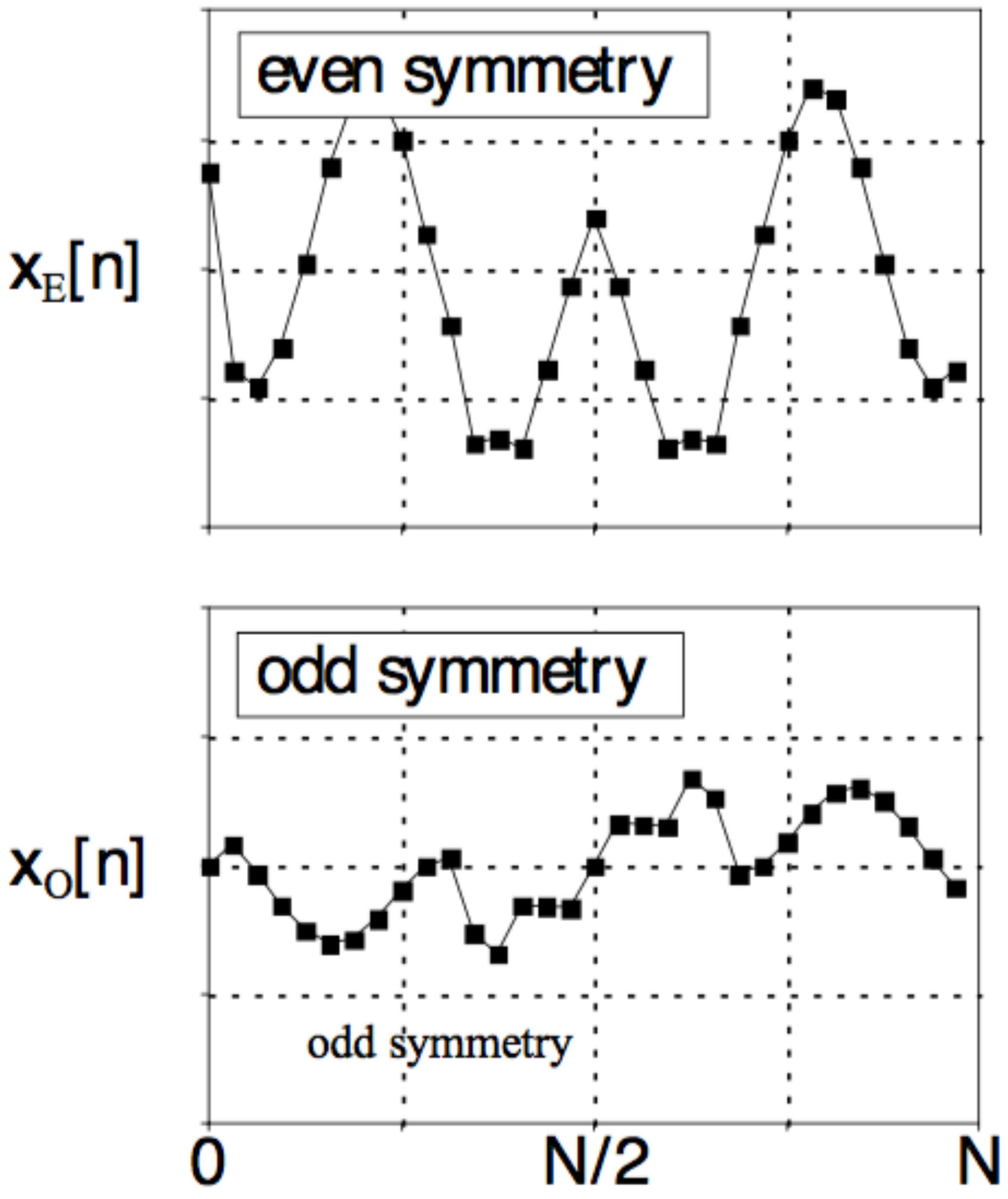




Common Decompositions

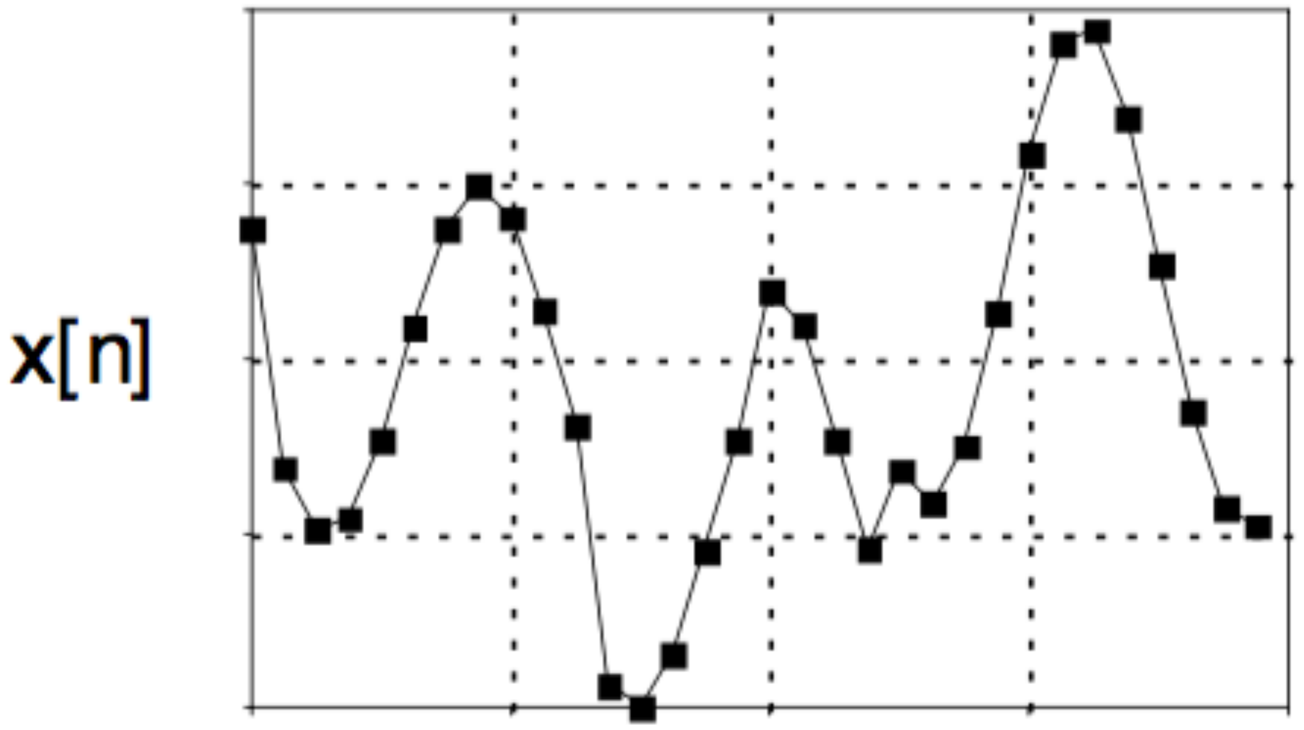
Even/Odd Decomposition

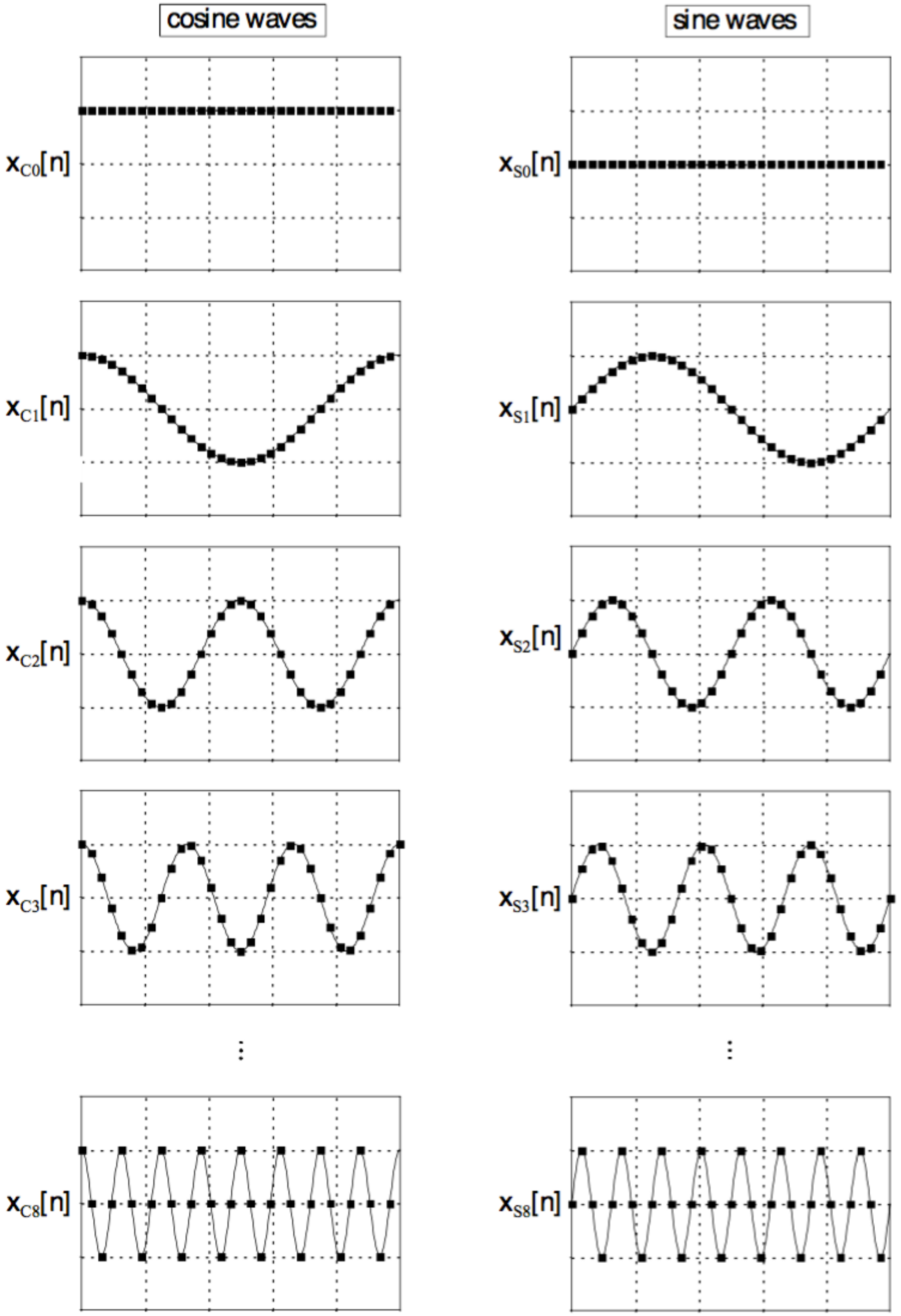




Common Decompositions

Fourier Decomposition





Alternatives to Linearity

- Only one major strategy for analyzing nonlinear systems: make them resemble a linear system
  - Ignore nonlinearity
    - If nonlinearity small enough, approximate as linear. Ignore error or consider it as noise
  - Keep signals small
    - Systems may appear linear if signals have small amplitude (e.g., transistors in mV)
  - Apply a linearizing transform (homomorphic signal processing): log (a[n] = b[n]xc[n])