01076566 Multimedia Systems

Chapter 2: Digital Data Acquisition

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Outline

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- Analog and Digital Signals
- Analog-to-Digital Conversion
- Signals and Systems
- Sampling Theorem and Aliasing
- Filtering
- Fourier Theory

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- Multimedia systems involve three major components
  - Multimedia content creation
  - Compression/storage of multimedia content
  - Delivery or distribution of multimedia content

Analog and Digital Signal

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- Analog signals are captured by a recording device
- Analog – if it can be represented by a continuous function
- Digital – is represented by a discrete set of values defined at specific instances of input domain, which might be time, space, or both

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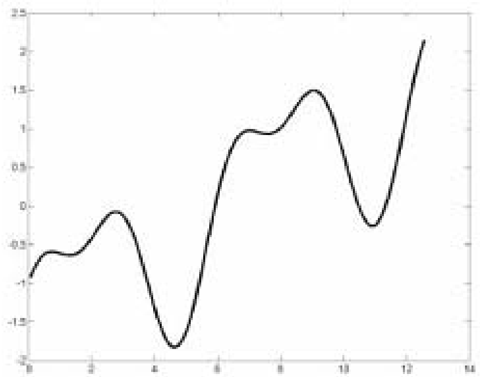
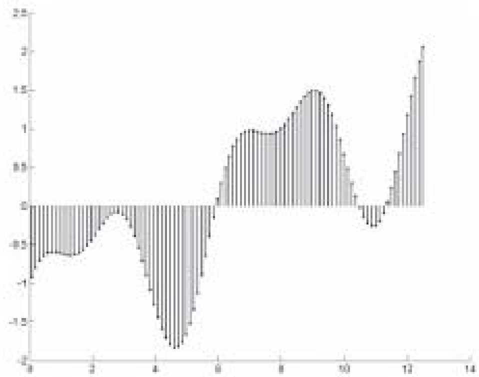


Analog signal



Digital signal

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- Advantages of digital signals over analog signals
  - Digital medium is possible to create complex, interactive content
    - Easy to access a pixel in an image, or a group of pixels in a region
    - Easy to access section of a sound track
    - Different digital operations can be applied to each region
  - Stored digital signals do no degrade over time or distance as analog signals do
  - Digital data can be efficiently compressed and transmitted across-digital networks
  - Easy to store digital data on magnetic media or solid state memory devices

Analog-to-Digital Conversion

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- The conversion of signals from analog to digital occurs via two processes: sampling and quantization.
- The reverse process of converting digital-to-analog is known as interpolation

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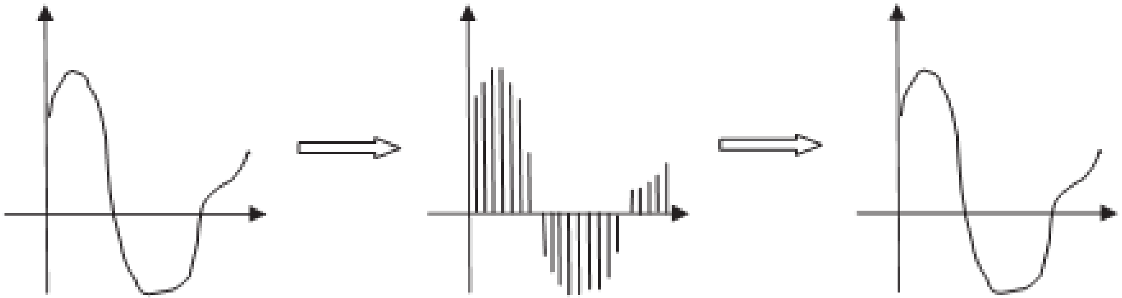

Figure 2-2 Analog-to-digital conversion and the corresponding interpolation from the digital-to-analog domain

- Must ensure that no artifacts are created in the digital data, so that when the signal is converted back to analog domain, it will look the same as the original analog signal.

Sampling

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- Assume that a one-dimensional analog signal in the time *t* domain, with an amplitude given by *x(t)*.
- The sampled signal is given by

$$x_s(n) = x(nT), \text{ where } T \text{ is the sampling period and}$$

$$f = 1/T \text{ is the sampling frequency}$$

- Hence

$$x_s(1) = x(T); x_s(2) = x(2T); x_s(3) = (3T)$$

- If T is reduced (f is increased), the number of samples increases – so does the storage requirement …
- T is a critical parameter
  - If T is too large, the signal might be under sampled, leading to artifacts
  - If T is too small, the signals requires large amounts of storage

Quantization

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- Quantization = encoding the signal value at every sampled location with a predefined precision, defined by a number of levels.
- Formally,

$$x_q(n) = Q[x_s(n)]$$

  , where *Q* is the rounding function
- *Q* represents a rounding function that maps the continuous values to the nearest digital value using *b* bits.
- Utilizing *b* bits corresponds to $N = 2^b$ levels

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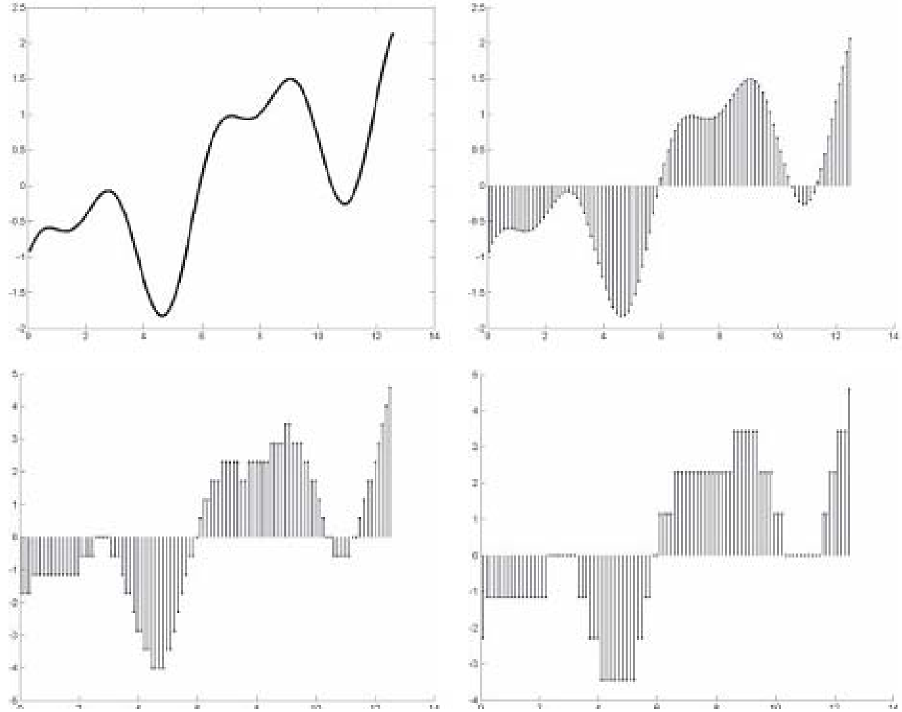

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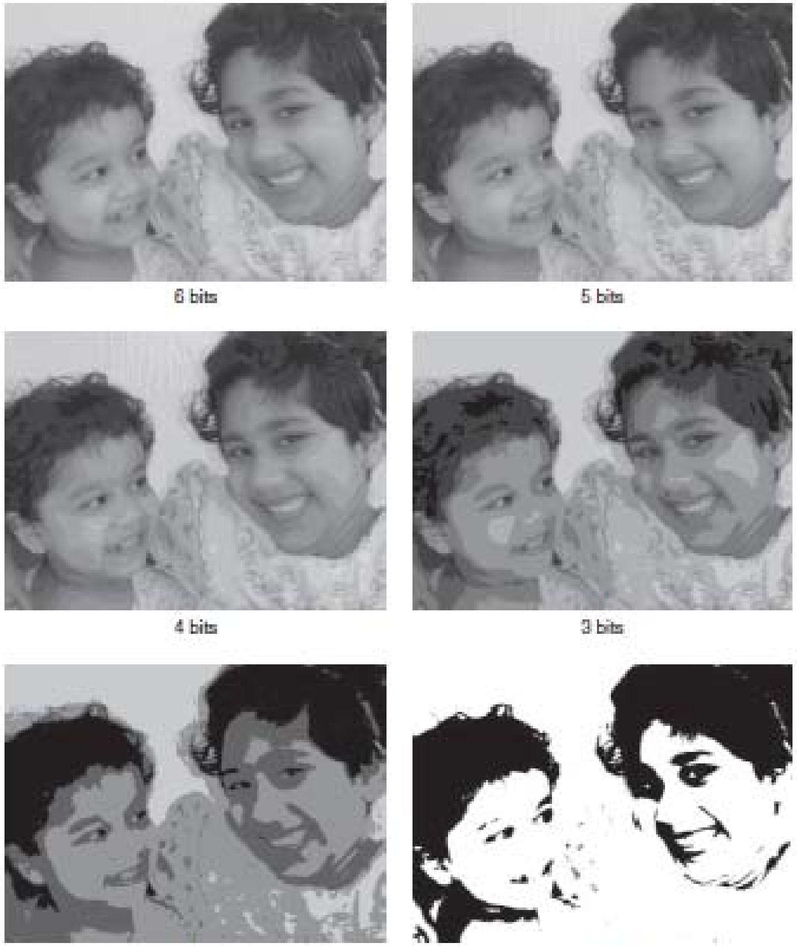

Figure 2-4 Examples of quantization; initial image had 8 bits per pixel, which is shown quantized from 6 bits down to 1 bit per pixel

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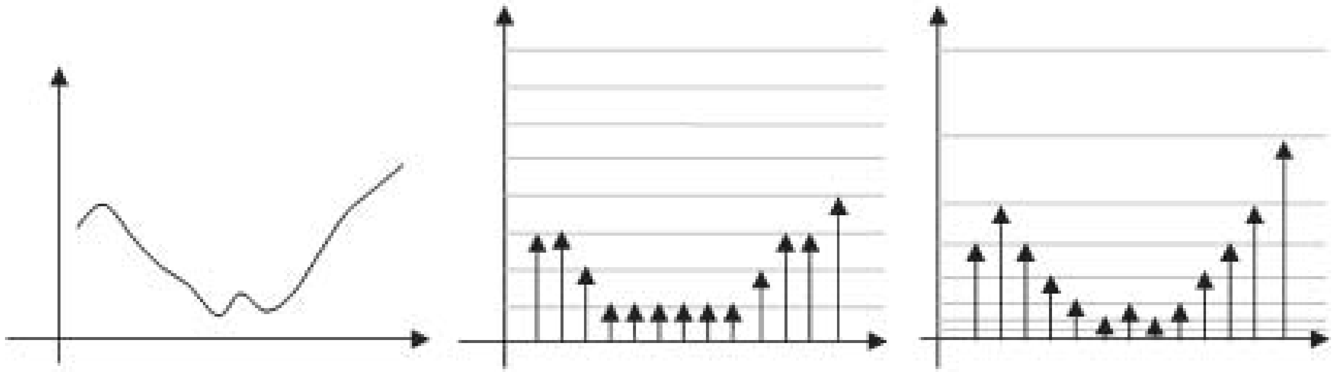

Figure 2-5 Nonlinear quantization scales. The left signal shows the original analog signal. The corresponding digitized signal using linear quantization is shown in the center. The right signal is obtained by a logarithmically quantized interval scale.

Bit Rate

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- Bit rate describes the number of bits being produced per second.

$$
\begin{aligned}
\textit{Bit rate} &= \frac{\textit{Bits}}{\textit{Second}} = \left(\frac{\textit{Samples produced}}{\textit{Second}}\right) \times \left(\frac{\textit{Bits}}{\textit{Sample}}\right) \\
&= \textit{Sampling rate} \times \textit{Quantization bits per sample}
\end{aligned}
$$

- Ideally, the bit rate should be just right to capture or convey the necessary information with minimal perceptual distortion, while also minimizing storage requirements.

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Signal	Sampling rate	Quantization	Bit rate
Speech	8 KHz	8 bits per sample	64 Kbps
Audio CD	44.1 KHz	16 bits per sample	706 Kbps (mono) 1.4 Mbps (stereo)
Teleconferencing	16 KHz	16 bits per sample	256 Kbps
AM Radio	11 KHz	8 bits per sample	88 Kbps
FM Radio	22 KHz	16 bits per sample	352 Kbps (mono) 704 Kbps (stereo)
NTSC TV Image frame	Width – 720 Height – 486	16 bits per sample	5.6 Mbits per frame
HDTV (1080)	Width – 1920 Height – 1080	12 bits per pixel on average	24.88 Mbits per frame

Signals and Systems

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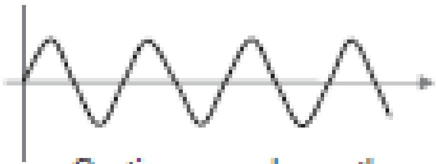
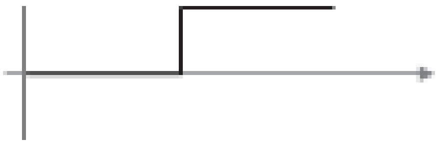
- Continuous and smooth; e.g., a sinusoid
- Continuous but not smooth; e.g., a saw tooth
- Neither smooth nor continuous; e.g., a step edge
- Symmetric – odd ($y = \sin(x)$) or even ($y = \cos(x)$)
- Finite support signals – a finite interval and zero outside of that interval
- Periodic signal

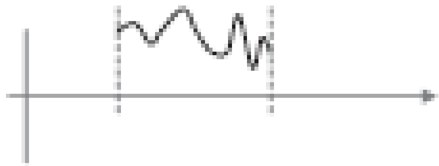


Continuous and smooth

Continuous but not smooth



Neither continuous nor smooth



Finite support signal

Periodic signal

Linear Time Invariant (LTI) Systems

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- Any operation that transforms a signal is called a *system*
- Let a system transform an input signal *x(t)* into an output *y(t)*.
- A system is a linear if the output and input obey the following

$$\text{If} \quad x(t) = c_1 x_1(t) + c_2 x_2(t)$$

$$\text{then} \quad y(t) = c_1 y_1(t) + c_2 y_2(t)$$

where $y_k(t)$ is the sole output resulting from $x_k(t)$

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- Another important operation is convolution.
- Convolution of two signals *f* and *g* is mathematically represented by *f* * *g*.
- It is the result of taking the integral of the first signal multiplied with the other signal reversed and shifted

$$
\begin{aligned}
(f * g) &= \int_{-\infty}^{\infty} f(\tau) \cdot g(t-\tau) d\tau \\
&= \int_{-\infty}^{\infty} f(t-\tau) \cdot g(\tau) d\tau
\end{aligned}
$$

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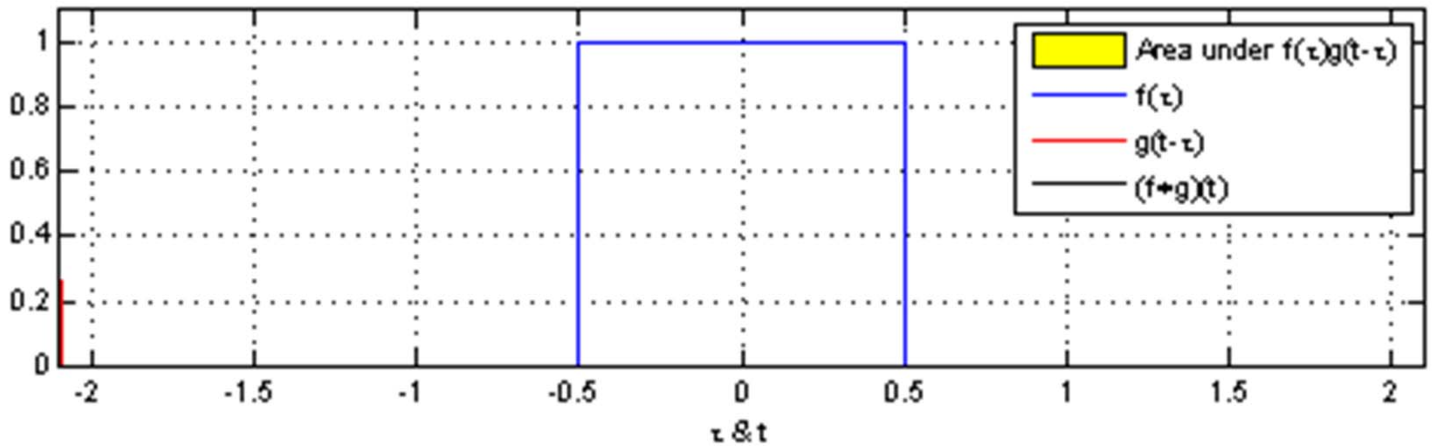

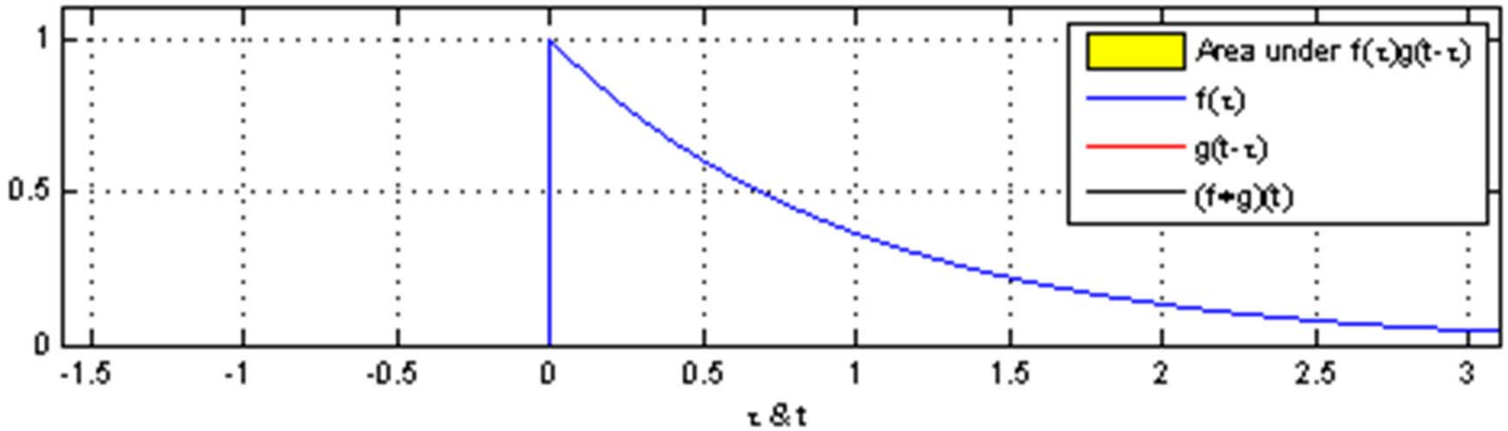
Area under f(τ)g(t-τ)
f(τ)
g(t-τ)
(f*g)(t)
1
0.5
0
-1.5
-1
-0.5
0
0.5
1
1.5
2
2.5
3
τ & t

Fundamental Results in LTI Systems

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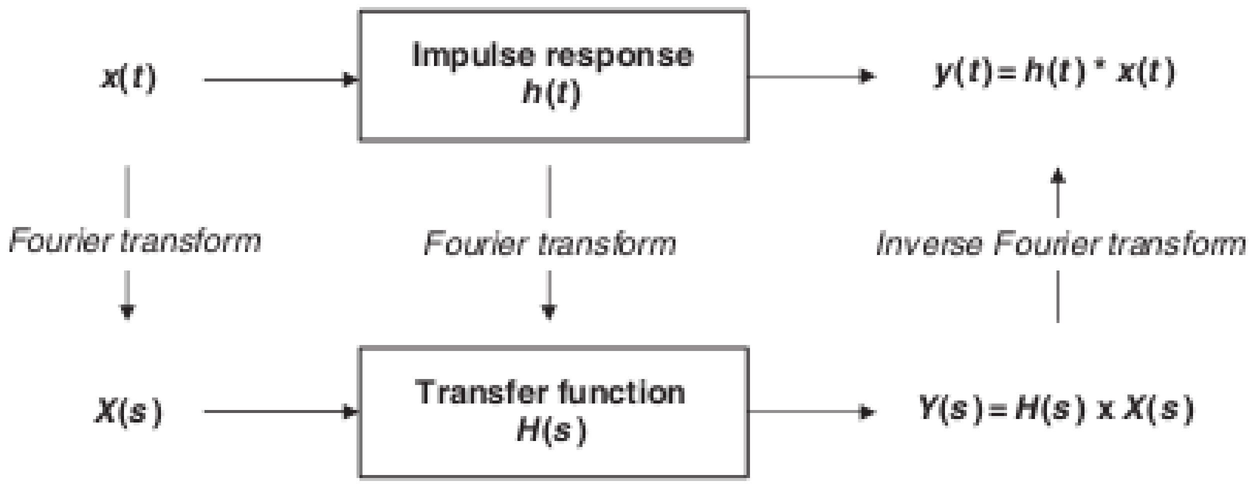

Figure 2-8 Relationship between the impulse response function in the time domain and the transfer function in the frequency domain

Useful Signals

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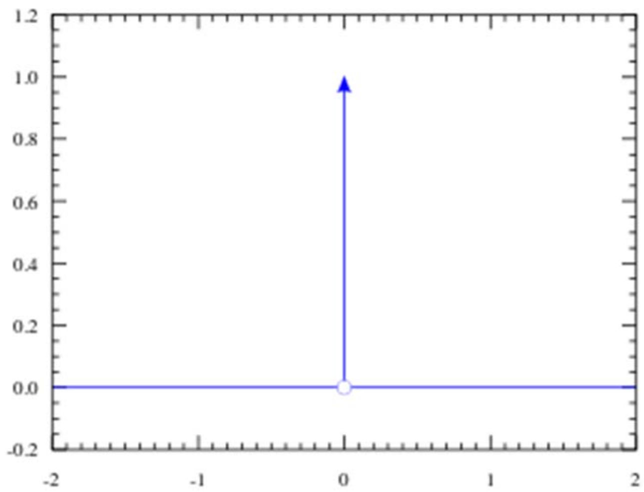
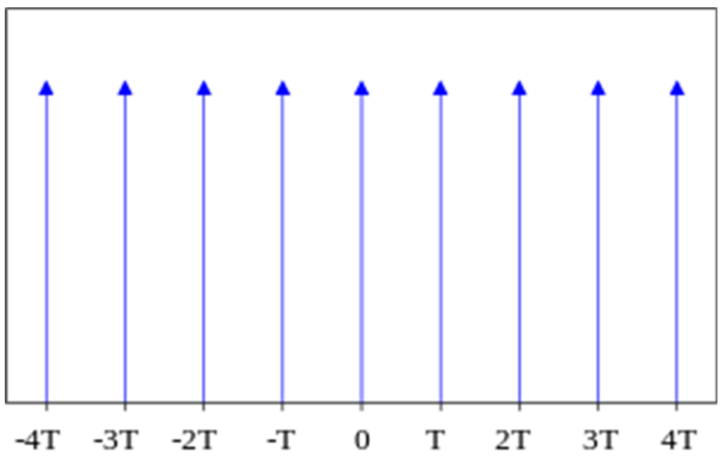


Delta function



Comb function

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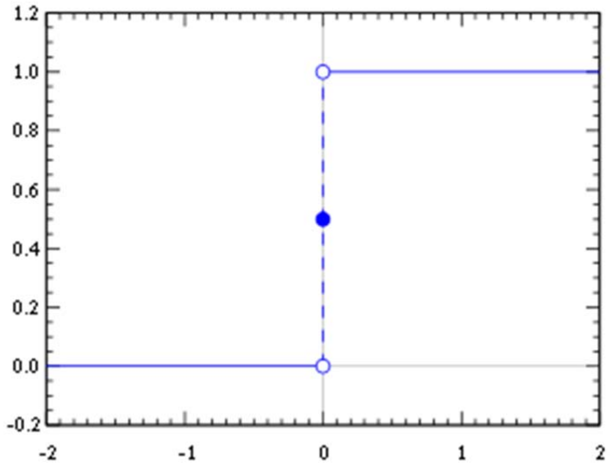
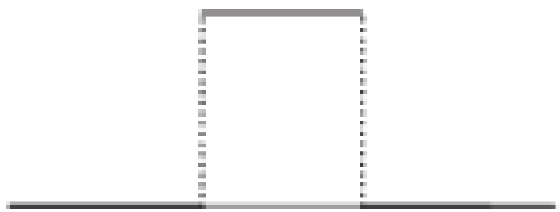


Step function



Box function

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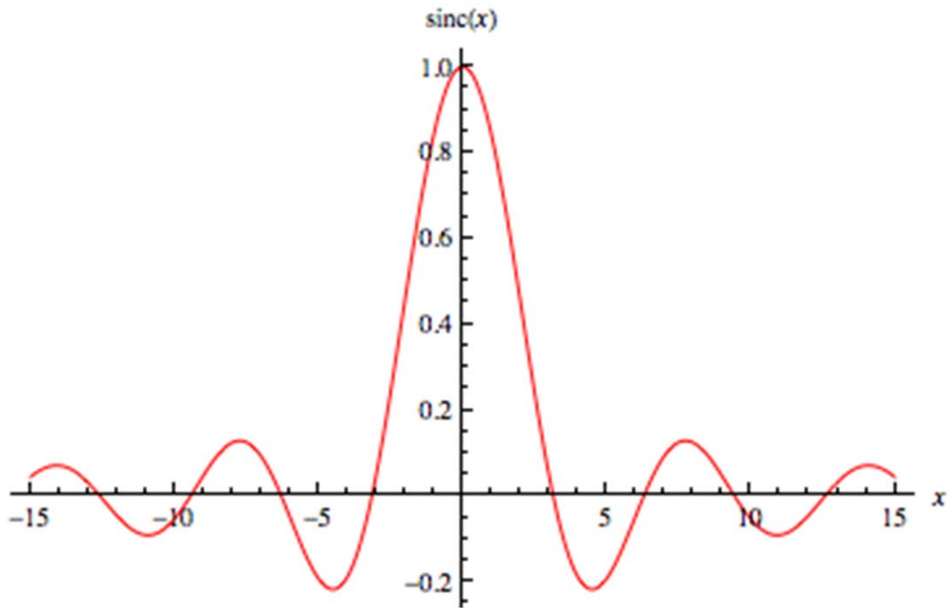

Sinc function

The Fourier Transform

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- Joseph Fourier (1768 – 1830)
- Any periodic, continuous signals can be represented as a sum of individual complex sinusoids (a Fourier series expansion).
- *Duality* property
  - Given a function and its Fourier transform, one can interchange the labels signal and spectrum for them

Sampling Theorem and Aliasing

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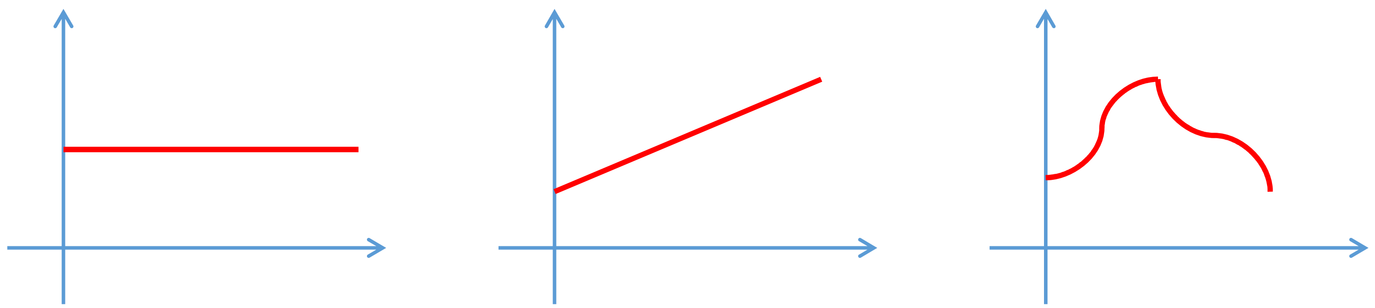
- The number of samples required to ensure that both starting and ending analog signals are the same is different



Examples of simple one-dimensional functions.
Different numbers of samples are required to digitize them.

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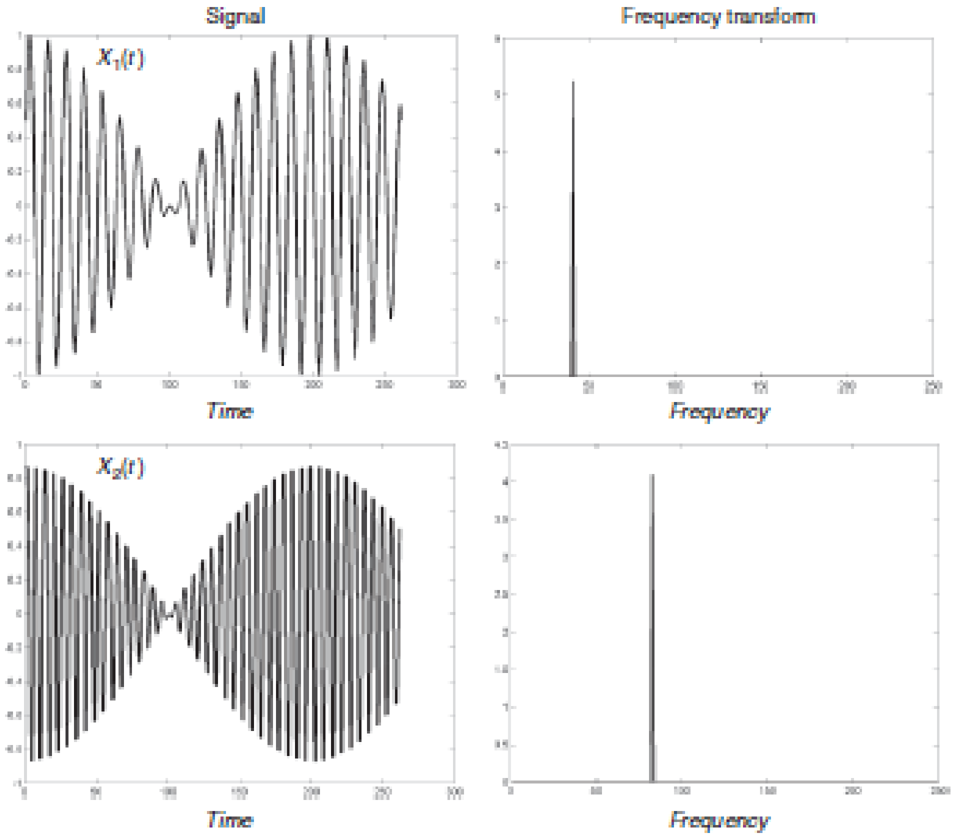

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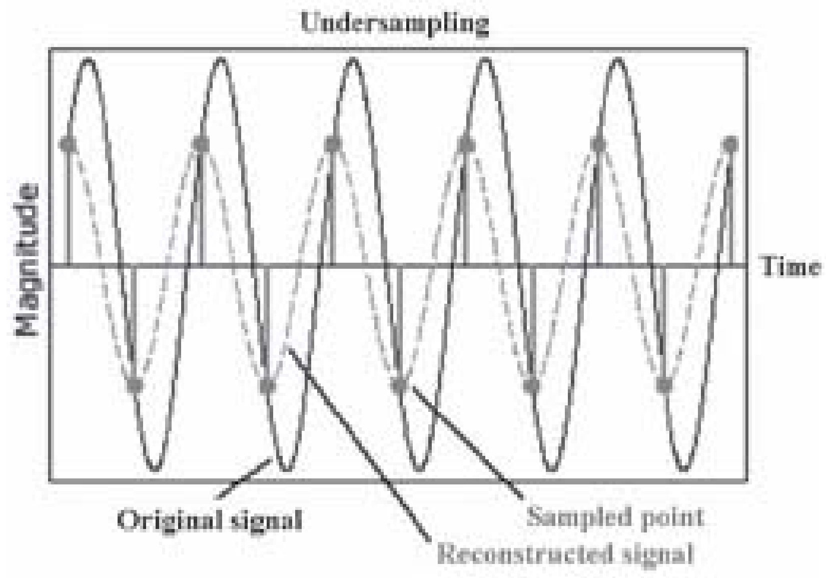
- Signal has to be sampled using a sampling frequency that is greater than twice the maximal frequency occurring in the signal.
- E.g.,
  - If the signal has a maximal frequency of 10 kHz, it should be sampled at a frequency greater than 20 kHz.
  - The 20 kHz is known as the *Nyquist* sampling frequency for that signal

Henry Nyquist (1889 – 1976)

Claude Shannon (1916 – 2001)


Undersampling
Magnitude
Time
Original signal
Sampled point
Reconstructed signal

Aliasing in Spatial Domains

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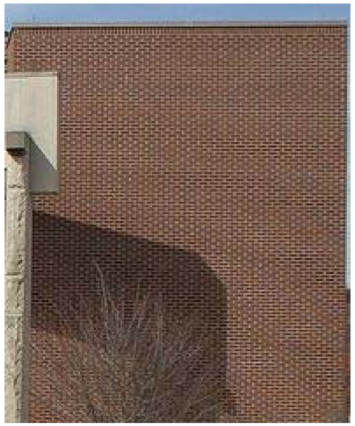
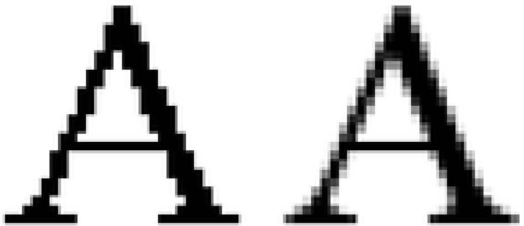
- **Aliasing** refers to an effect that causes different signals to become indistinguishable (or *aliases* of one another) when sampled.

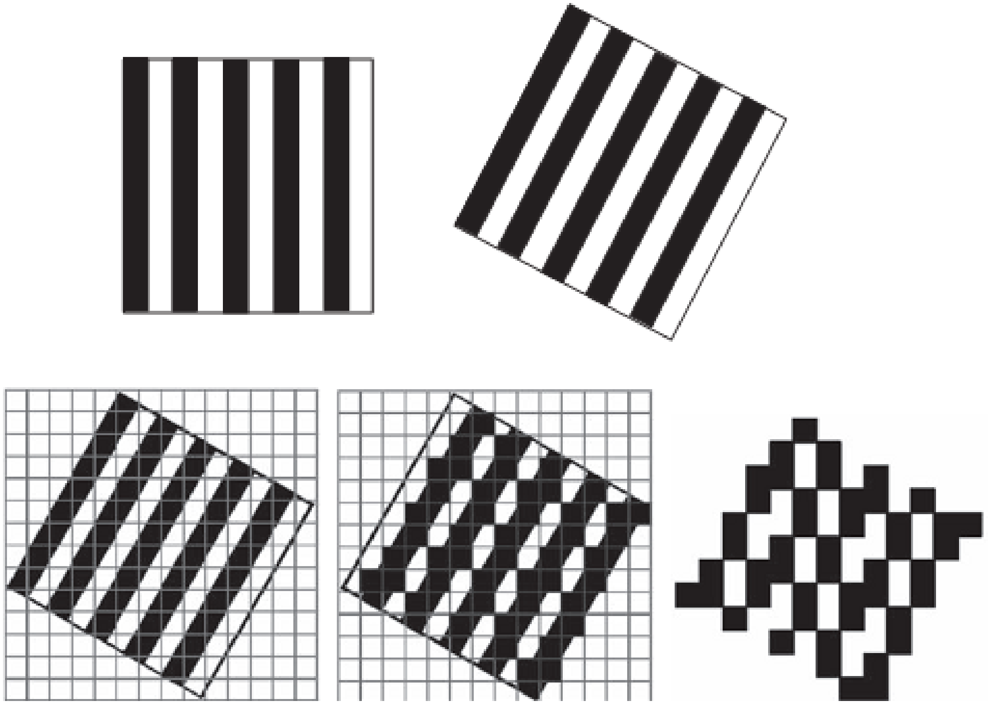


Properly sampled image of brick wall.

Spatial aliasing in the form of a Moiré pattern.



Aliasing example of the A letter in Times New Roman. Left: aliased image, right: *anti-aliased* image.



Filtering

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- Function of a filter
  - Remove unwanted parts of the signal
  - Extract useful parts of the signal
- Analog filter – use analog components
- Digital filter – use digital numerical computations on sampled

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- Digital filters have advantages over analog filters
  - Programmable
  - Easily designed, tested, and implemented
  - Can be combined in parallel or cascaded in series
  - Stable with respect to time and temperature – as opposed to analog filter
  - Can handle low-frequency signals accurately
  - More versatile in their ability to process signals in a variety of ways

Filtering in 1D

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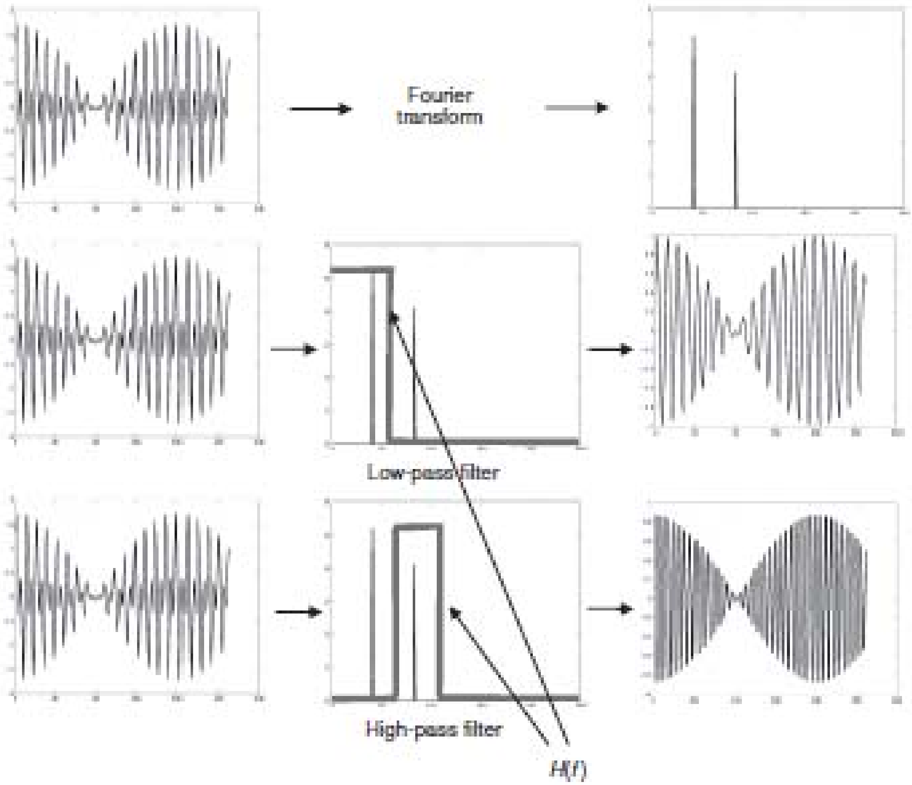
- Normally represented in the time domain with the *x*-axis showing sampled positions and the *y*-axis showing the amplitude values


Fourier transform
Low-pass filter
High-pass filter
H(f)

Filtering in 2D

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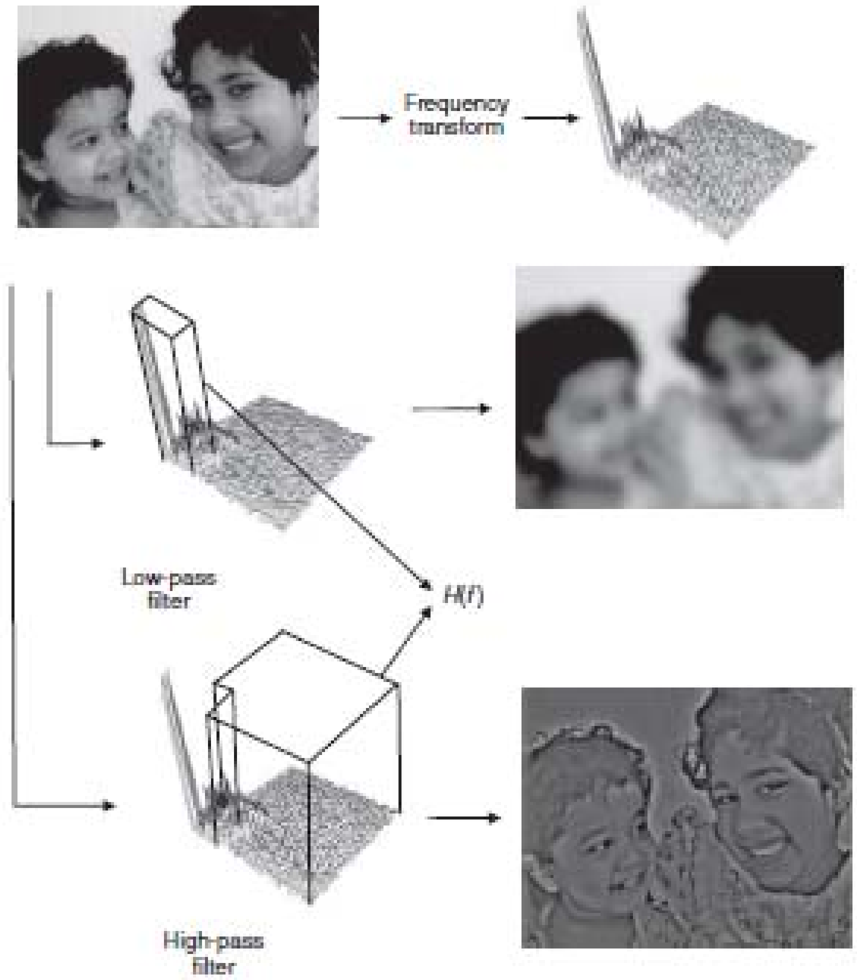

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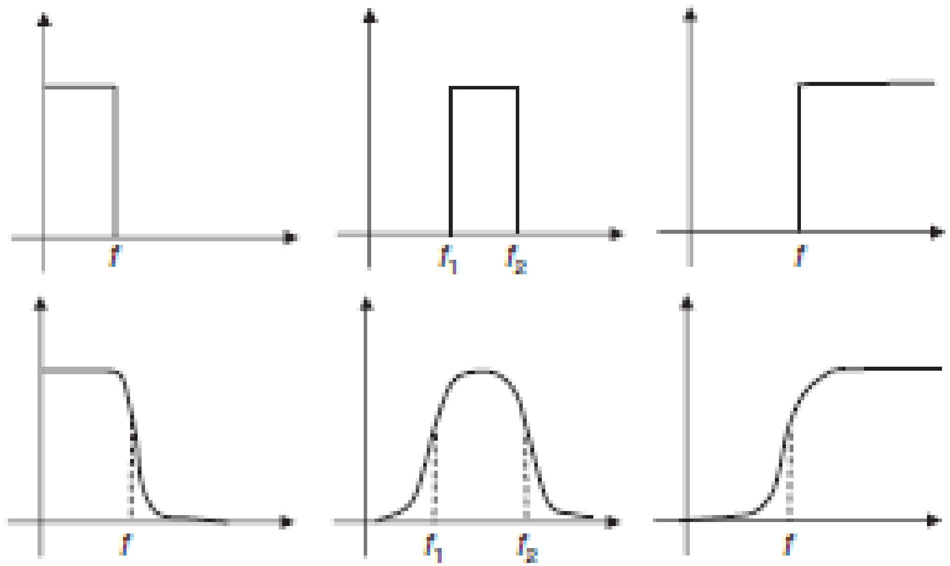

Frequency responses for low-, band-, and high-pass filter

Subsampling

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- Subsampling – post digitization sampling adjustments
- Subsampling an original signal by n corresponds to keeping every n^{th} sample and remove the rest from the original signal. The signal of data is also reduced by a factor of n
- May result in aliasing problem

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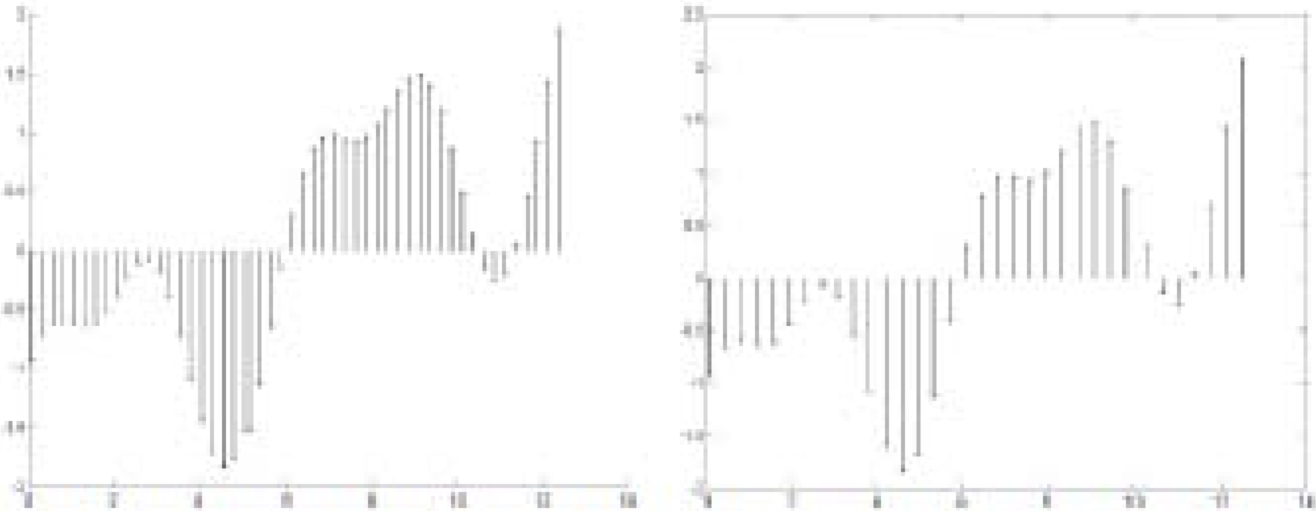

Figure 2-19 Original signal shown on the left, subsampled by 2 producing signal on the right. The number of samples on the right is half the number of samples on the left.

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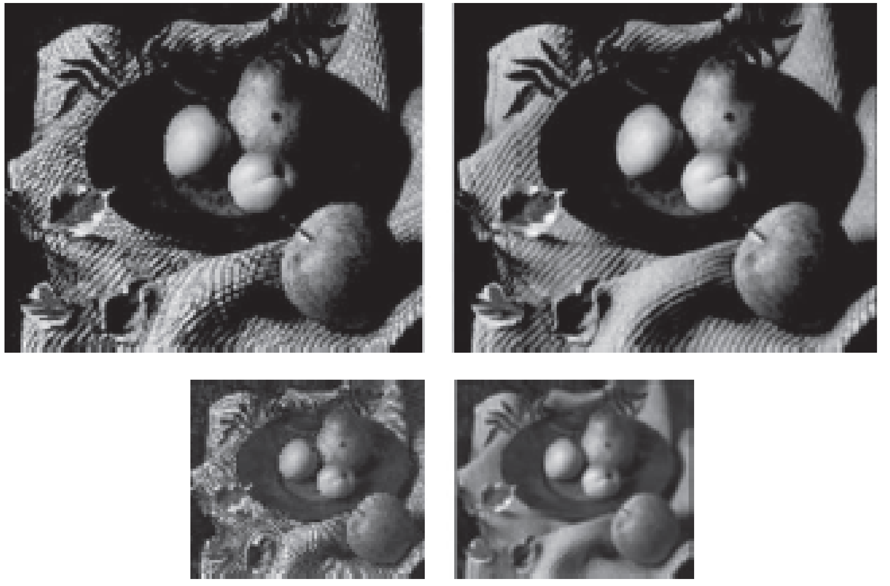
Original image



Subsampled by 2 and subsampled by 4.
The left image is obtained without filtering prior to subsampling.
The right image is obtained by filtering prior to subsampling.

Fourier Theory

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- Any continuous periodic function f(t) can be decomposed into or represented by a weighted combination of sine and cosine wave

$$f(t)=\sum_{i=0}^{i=\infty} A_i \times \sin(i\omega t)+\sum_{j=0}^{j=\infty} B_j \times \cos(j\omega t)$$

- The sine and cosine function defined over the fundamental frequencies are known as basis function.
- A_i and B_j are known as the frequency coefficients.

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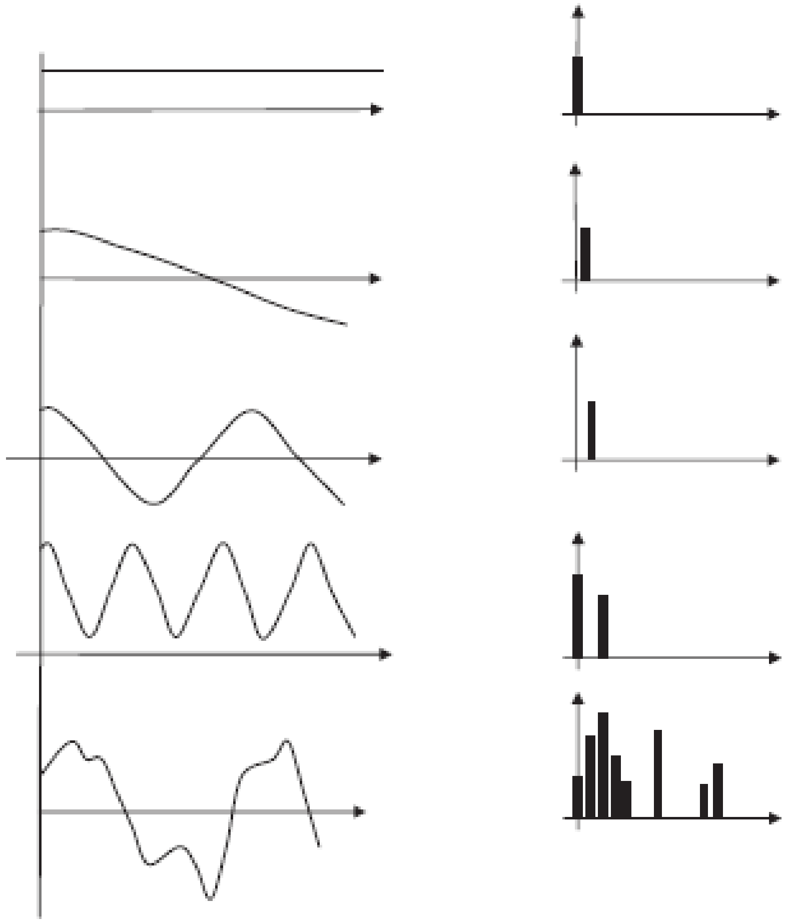

Figure 2-21 The left column shows examples of time domain functions. The right column shows their frequency transforms. For each transform, the nonzero coefficients are shown. Although the first three functions are simple, and have only one fundamental frequency, the last two are composed of more than one.

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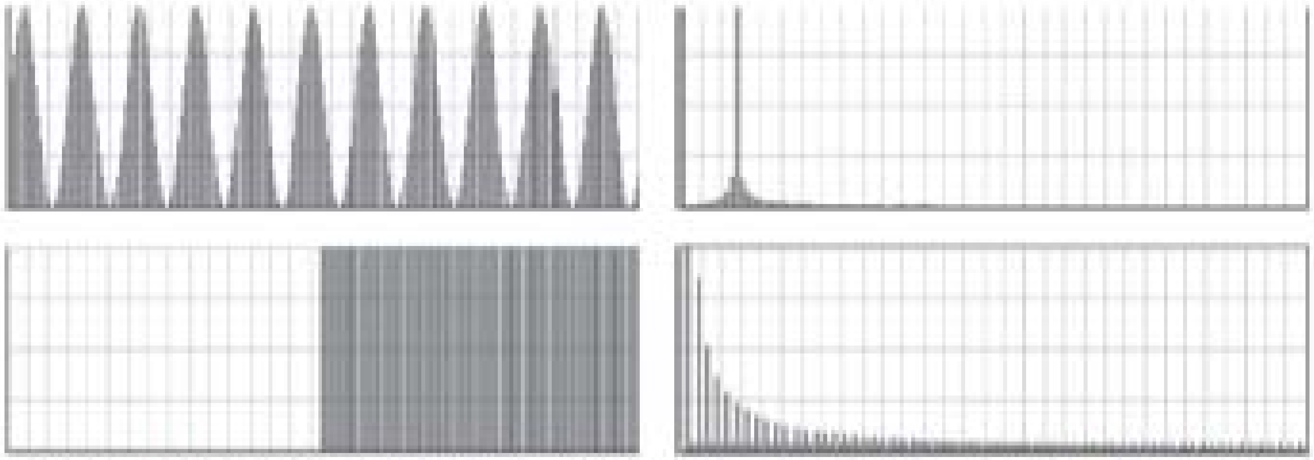

Figure 2-22 The discrete frequency transform operates on discrete signals. The left column shows a sinusoid and a step function. The right column shows the weights of the frequency coefficients. While the sinusoid has a distinct peak, the step function needs to be approximated by all frequencies.

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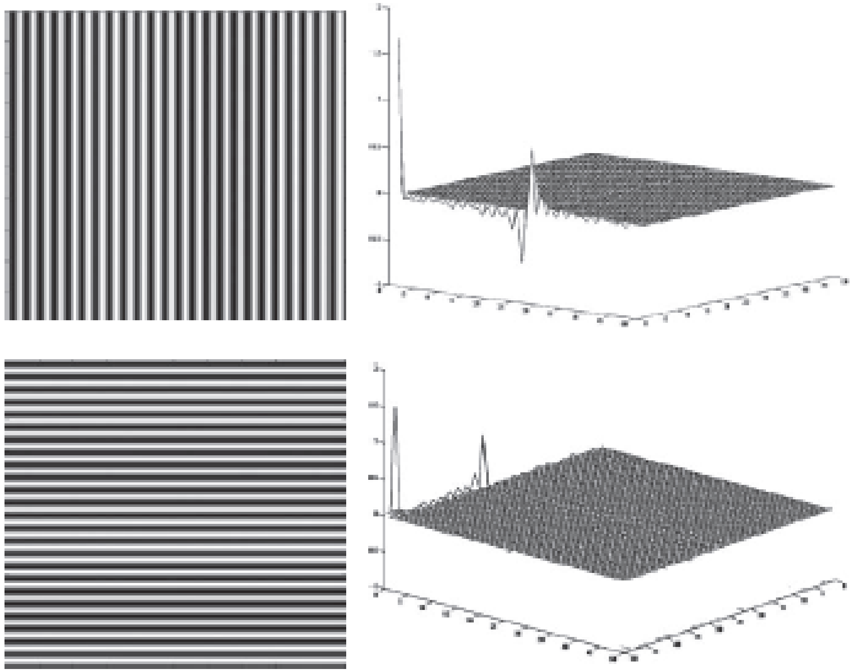

Frequency transforms in 2D

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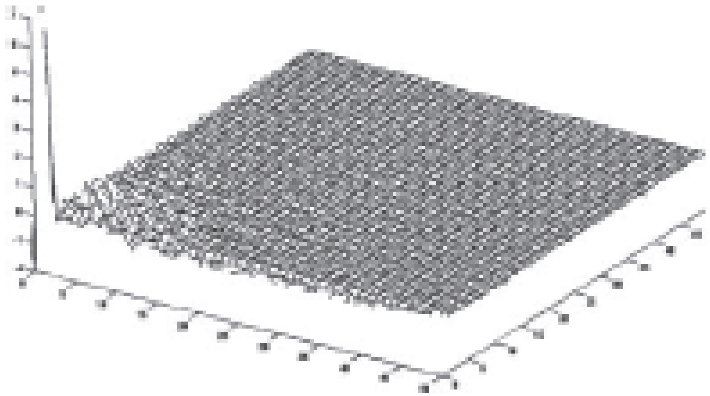

Frequency transforms in 2D

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Q & A