Multimedia System

Digital Acquisition

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ANALOG AND DIGITAL SIGNALS

- Analog signals are captured by a recording device, which attempts to record a physical signal. A signal is analog if it can be represented by a continuous function.
- For instance, it might encode the changing amplitude with respect to an input dimension(s).
- Digital signals, on the other hand, are represented by a discrete set of values defined at specific (and most often regular) instances of the input domain, which might be time, space, or both.
- An example of a one-dimensional digital signal is shown in Figure 2-1, where the analog signal is sensed at regular, fixed time intervals.
- Although the figure shows an example in one dimension (1D), the theory discussed can easily be extended to multiple dimensions.

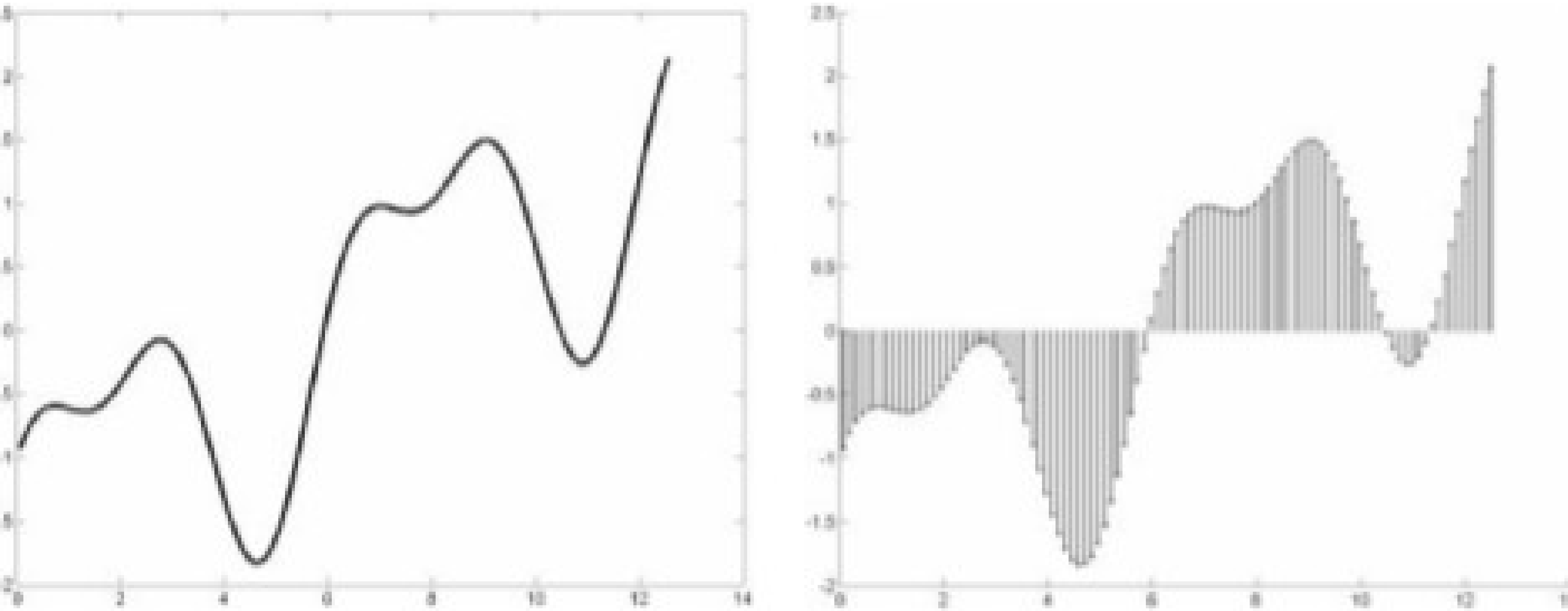


Figure 2-1 Example of an analog signal (left) and a digital signal (right) in one dimension

The advantages of digital signals over analog ones

- When media is represented digitally, it is possible to create complex, interactive content.
- Stored digital signals do not degrade over time or distance as analog signals do.

  One of the most common artifacts of broadcast VHS video is ghosting, as stored VHS tapes lose their image quality by repeated usage and degradation of the medium over time. This is not the case with digital broadcasting or digitally stored media types.
- Digital data can be efficiently compressed and transmitted across digital networks.

  This includes active and live distribution models, such as digital cable, video on demand, and passive distribution schemes, such as video on a DVD.
- It is easy to store digital data on magnetic media such as portable 3.5 inch, hard drives, or solid state memory devices, such as flash drives, memory cards, and so on.

  This is because the representation of digital data, whether audio, image, or video, is a set of binary values, regardless of data type. As such, digital data from any source can be stored on a common medium. This is to be contrasted with the variety of media for analog signals, which include vinyl records and tapes of various widths.

ANALOG-TO-DIGITAL CONVERSION

- The conversion of signals from analog to digital occurs via two main processes: **sampling and quantization**.
- The reverse process of converting digital signals to analog is known as *interpolation*.
- One of the most desirable properties in the analog to digital conversion is to ensure that no artifacts are created in the digital data.
- That way, when the signal is converted back to the analog domain, it will look the same as the original analog signal.

Sampling

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

$$x_s(n) = x(nT), \text{ where } T \text{ is the sampling period and}$$
$$f = 1/T \text{ is the sampling frequency.}$$

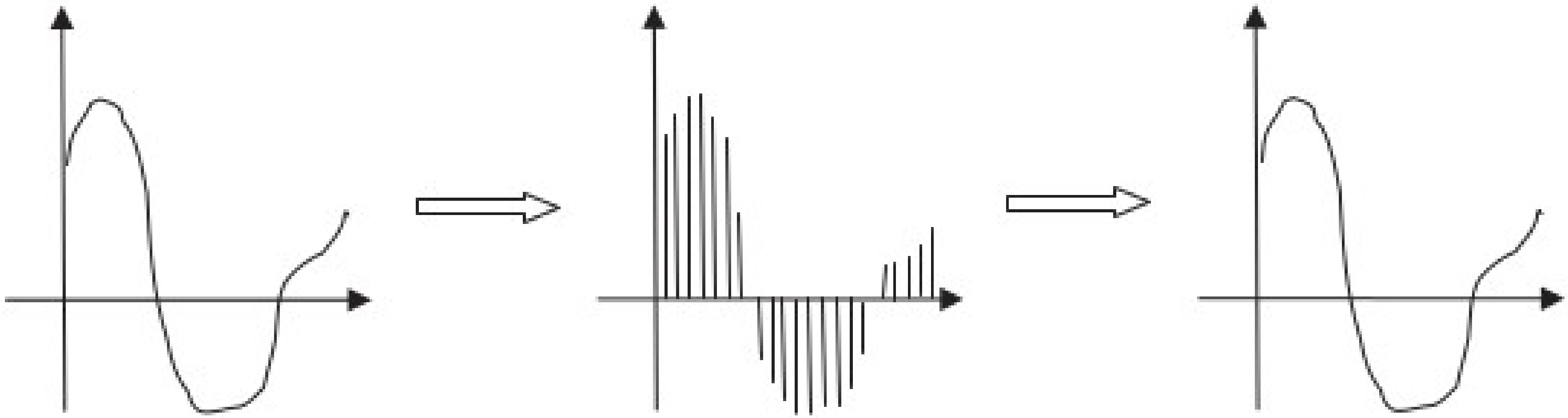


Figure 2-2 Analog-to-digital conversion and the corresponding interpolation from the digital-to-analog domain

Sampling

- Hence, $X_s(1) = x(T)$; $X_s(2) = X(2T)$; $X_s(3) = X(3T)$; and so on.
- If you reduce T (increase f), the number of samples increases; and correspondingly, so does the storage requirement.
- Vice versa, if T increases (f decreases), the number of samples collected for the signal decrease and so does the storage requirement.
- T is clearly a critical parameter. Should it be the same for every signal? If T is too large, the signal might be under sampled, leading to artifacts, and if T is too small, the signal requires large amounts of storage, which might be redundant.
- For commonly used signals, sampling is done across one dimension (time, for sound signals), two dimensions (spatial x and y, for images), or three dimensions (x, y, time for video, or x, y, z for sampling three-dimensional ranges).
- It is important to note that the sampling scheme described here is theoretical. Practical sampling involves averaging, either in time or space.

Quantization

- Quantization deals with encoding the signal value at every sampled location with a predefined precision, defined by a number of levels.
- In other words, now that you have sampled a continuous signal at specific regular time instances, how many bits do you use to represent the value of signal at each instance?
- The entire range R of the signal is represented by a finite number of bits b.

$$x_q(n) = Q[x_s(n)], \text{ where Q is the rounding function.}$$

Q represents a rounding function that maps the continuous value $x_s(n)$ to the nearest digital value using b bits. Utilizing b bits corresponds to $N = 2^b$ levels, thus having a quantization step delta $= R/2^b$. Figure 2-3 shows an analog signal, which is sampled at a common frequency, but quantized using different number of bits, 4 or 2.

- Because each sample is represented by a finite number of bits, the quantized value will differ from the actual signal value, thus always introducing an error.
- The maximum error is limited to half the quantization step.
- The error decreases as the number of bits used to represent the sample increases.
- This is an unavoidable and irreversible loss, as the sample would otherwise need to be represented with infinite precision, which requires an infinite number of bits.

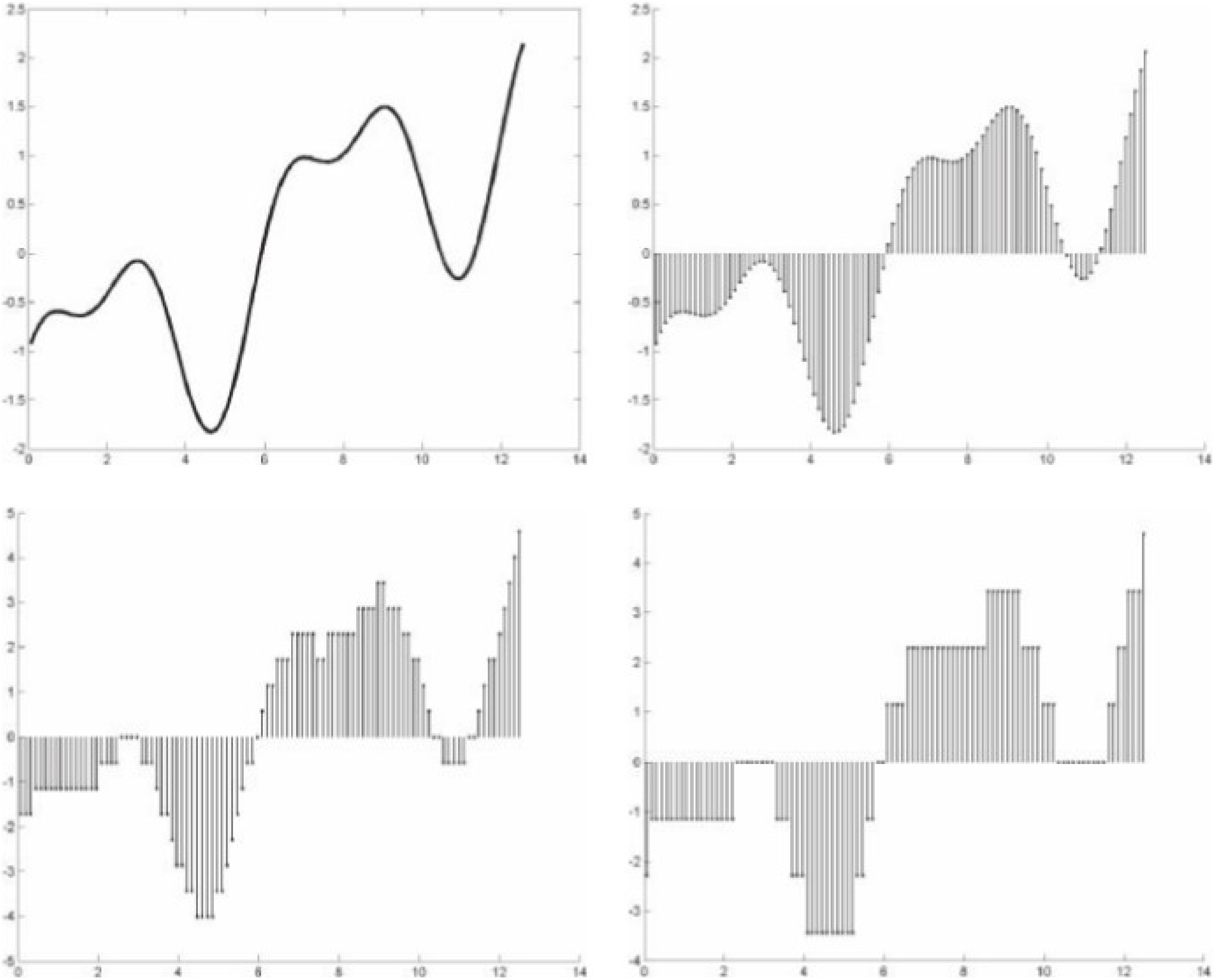


Figure 2-3 Original analog signal (upper left) is shown sampled and quantized at different quantization levels. For quantization, 8 bits (256 levels), 4 bits (16 levels), and 3 bits (8 levels) were used to produce the digital signals on the top right, bottom left, and bottom right, respectively.

This actually depends on the type of signal and what its intended use is. Audio signals, which represent music, must be quantized on 16 bits, whereas speech only requires 8 bits. Figure 2-4 illustrates quantization effects in two dimensions for images. The results show that the error increases as the number of quantization bitsused to represent the pixel samples decreases.

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

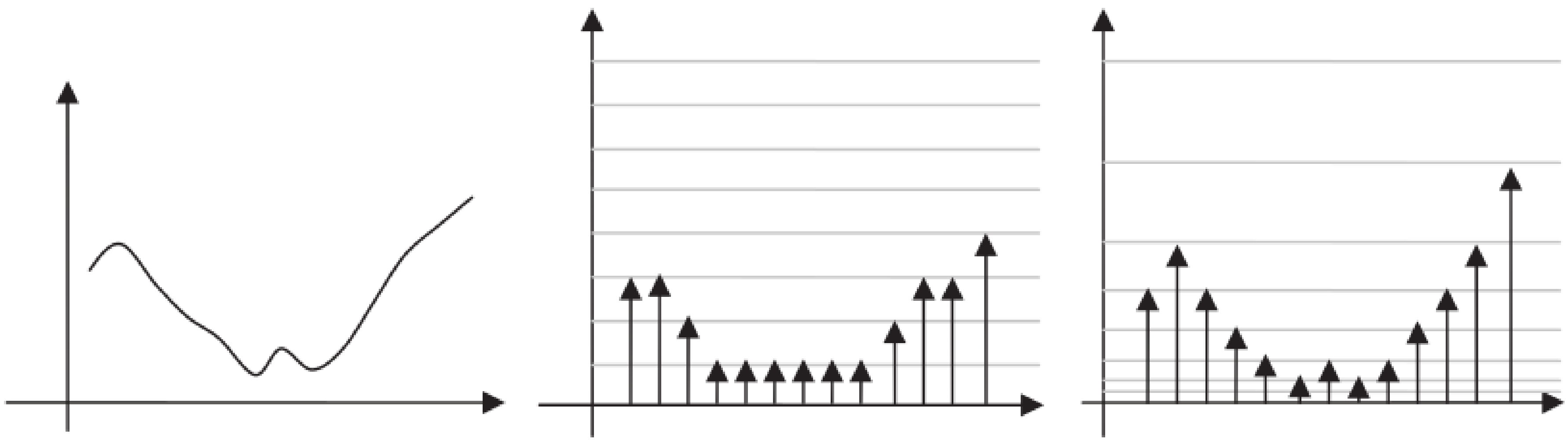


Figure 2-5 Nonlinear quantization scales. The left signal shows the original analog signal. The corresponding digitized signal using linear quantization is shown in the center. The right signal is obtained by a logarithmically quantized interval scale.

Bit Rate

- Understanding the digitization process from the previous two subsections brings us to an important multimedia concept known as the bit rate, which describes the number of bits being produced per second.
- Bit rate is of critical importance when it comes to storing a digital signal, or transmitting it across networks, which might have high, low, or even varying bandwidths.
- Bit rate, which is measured in terms of bits per second, consists of the following:

$$\text{Bit rate} = \frac{\text{Bits}}{\text{Second}} = \left(\frac{\text{Samples produced}}{\text{Second}}\right) \times \left(\frac{\text{Bits}}{\text{Sample}}\right)$$

$$= \text{Sampling rate x Quantization bits per sample}$$

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 – 486 Height – 720	16 bits per sample	5.6 Mbits per frame
HDTV (1080i)	Width – 1920 Height – 1080	12 bits per pixel on average	24.88 Mbits per frame

Figure 2-6 Table giving the sampling rate, quantization factor, and bit rates produced for typical signals

Signal

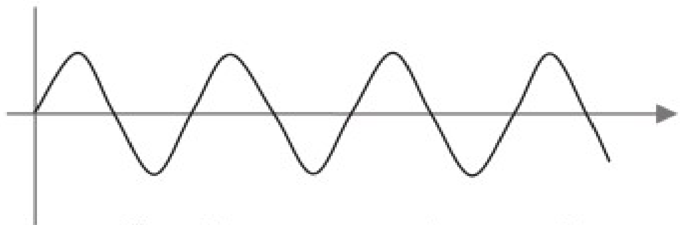
A signal is a set of data, usually a function of time. We denote a signal as $x(t)$, where t is nominally time. We have already seen two types of signals:

- discrete signals, where t takes integer values, and continuous signals, where t takes real values. In contrast, digital signals are those where the signal $x(t)$ takes on one of a quantized set of values, typically 0 and 1, and
- analog signals are those where the signal $x(t)$ takes on real values.

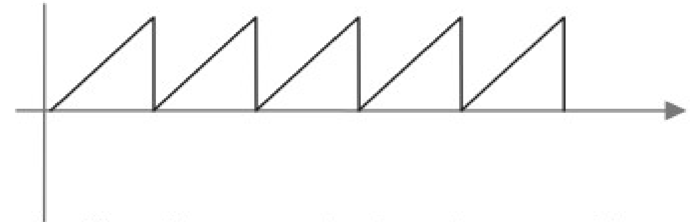
Type of Signal

- Continuous and smooth-such as a sinusoid
- Continuous and not smooth-such as a saw tooth
- Neither smooth nor continuous-for example, a step edge
- Symetric-which can be further describe either as odd (y=sin(x)) or even(y=cos(x))
- Finite support signals
- Periodic signal-a signal that repeats it self over a time period.

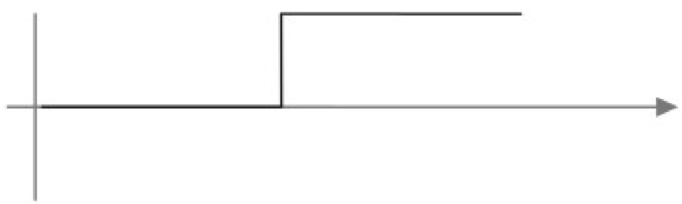
Type of Signal



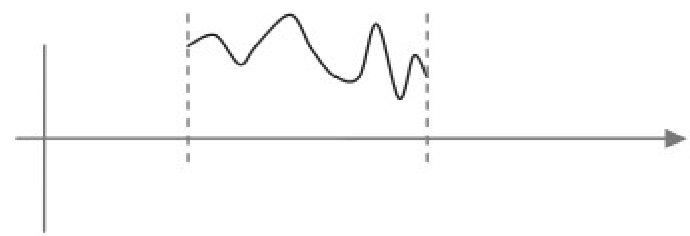
Continuous and smooth



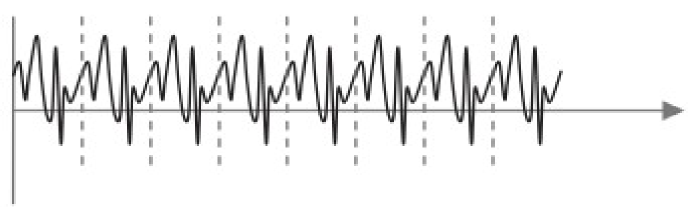
Continuous but not smooth



Neither continuous nor smooth



Finite support signal



Periodic signal

Linear Time Invariant System

- Any operation that transforms a signal is called a system.
- Time invariance of a system can be defined by the property that the output signal of a system at a given instant in time, depends only on the input signal at that instant in time.

Linear Time Invariant System

- Let a system transform an input signal x(t) into an output y(t). We call this system a linear if the output and input obey the following:

$$\begin{aligned} &\textit{If} && x(t) = c_1 x_1(t) + c_2 x_2(t) \\ &\textit{then} && y(t) = c_1 y_1(t) + c_2 y_2(t) \end{aligned}$$

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

Case For Linear System

Is the system defined by the transfer function H(x) = x^2 linear?

Solution:

$H(k_1x_1 + k_2x_2) = (k_1x_1 + k_2x_2)^2$, which is not the same as $k_1y_1 + k_2y_2 = k_1^2x_1^2 + k_2^2x_2^2$. Hence, the system is not linear.

Fundamental Result in Linear Time Invariant System

Any LTI system is fully characterized by a specific function, which is called the impulse response of the system.

- The output of the system is the convolution of the input with the system’s impulse response. This analysis is termed as the time domain point of view of the system.
- Alternatively, we can also express this result in the frequency domain by defining the system’s transfer function. The transfer function is the Fourier transform of the system’s impulse response.

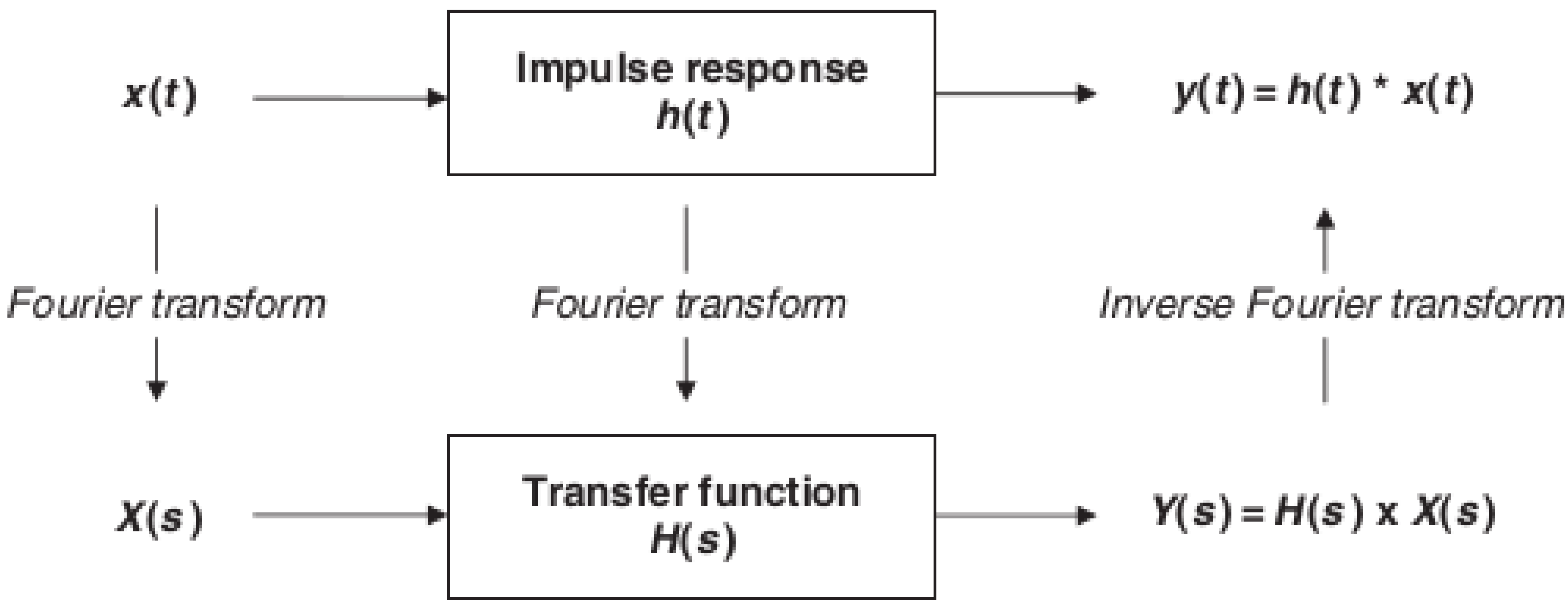
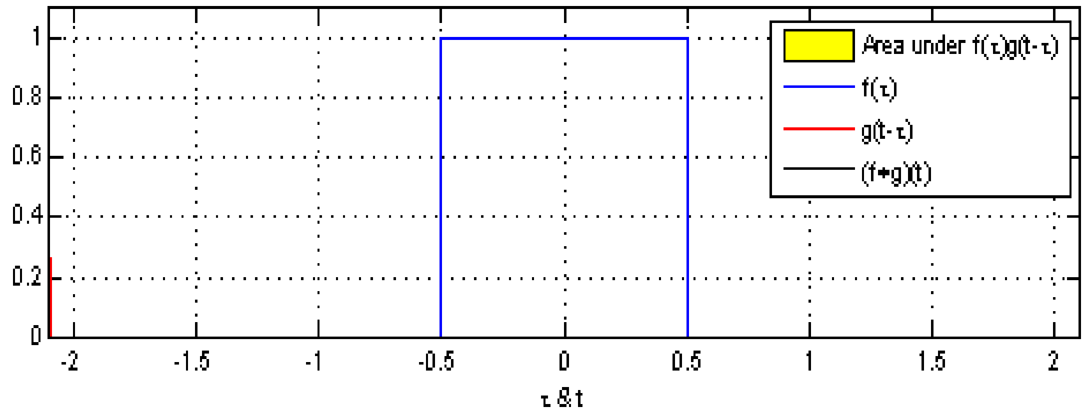


Figure 2-8 Relationship between the impulse response function in the time domain and the transfer function in the frequency domain

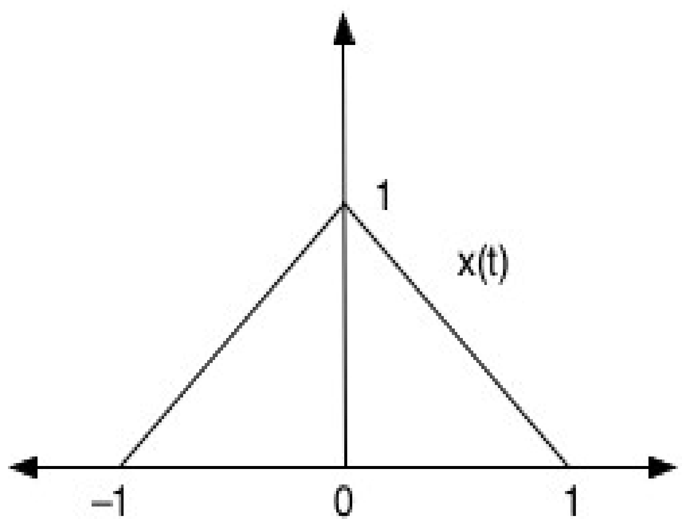
Convolution

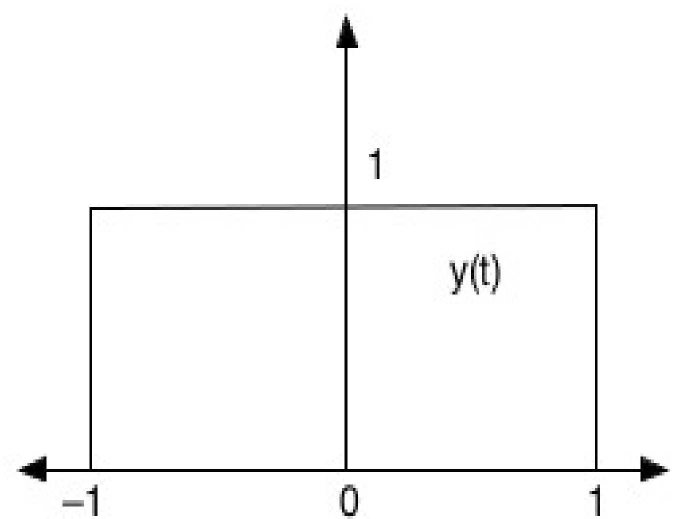
The convolution of two signals f and g is mathematically represented by f *g. It is the result of taking the integral of the first signal multiplied with the other signal reversed and shifted.



Convolution

- Compute the convolution of the functions x(t) and y(t) defined graphically in below.





Case

- Solution: By convention, we assume that both functions are zero outside their range of definition. Thus,

$$x(\tau) \neq 0 \text{ in the range } -1 \leq \tau \leq 1$$
$$y(t-\tau) \neq 0 \text{ in the range } -1 \leq (t-\tau) \leq 1 \Rightarrow t-1 \leq \tau \leq t+1$$

Therefore, both $x(t)$ and $y(t)$ are simultaneously nonzero only in the range $max(t-1, -1) \leq \tau \leq min(t+1, 1)$, and

$$x(t) \otimes y(t) = \int_{-\infty}^{\infty} x(\tau)y(t-\tau)d\tau = \int_{max(t-1,-1)}^{min(t+1,1)} x(\tau)y(t-\tau)d\tau$$

It is necessary to evaluate this integral separately in the ranges $[-\infty, -2]$, $[-2, -1]$, $[-1, 0]$, $[0,1]$, $[1,2]$, and $[2, \infty]$. In all the ranges, $y(t) = 1$, but the limits of integration are modified as follows:

$$x(t) \otimes y(t) = \begin{cases} 0 & -\infty \le t \le -2 \\ \int_{-1}^{(t+1)} (1+\tau)\,d\tau & -2 \le t \le -1 \\ \int_{0}^{(t+1)} (1-\tau)\,d\tau + \frac{1}{2} & -1 \le t \le 0 \\ \int_{(t-1)}^{0} (1+\tau)\,d\tau + \frac{1}{2} & 0 \le t \le 1 \\ \int_{t-1}^{1} (1-\tau)\,d\tau & 1 \le t \le 2 \\ 0 & 2 \le t \le \infty \end{cases}$$

$$x(t) \otimes y(t) = \left\{ \begin{array}{ll} 0 & -\infty \le t \le -2 \\ \dfrac{(t+2)^2}{2} & -2 \le t \le -1 \\ 1 - \dfrac{t^2}{2} & -1 \le t \le 0 \\ 1 - \dfrac{t^2}{2} & 0 \le t \le 1 \\ \dfrac{(t-2)^2}{2} & 1 \le t \le 2 \\ 0 & 2 \le t \le \infty \end{array} \right\}$$

Useful Signals

The *Delta function* is a mathematical construct introduced by the theoretical physicist Paul Dirac and, hence, is also known as the Dirac Delta function. Informally, it is a sharp peak bounding a unit area: $\delta(x)$ is zero everywhere except at $x = 0$ where it becomes infinite, and its integral is 1. A useful property of the Delta function is *sifting*. The sifting property is used in sampling and can be shown in the following equation:

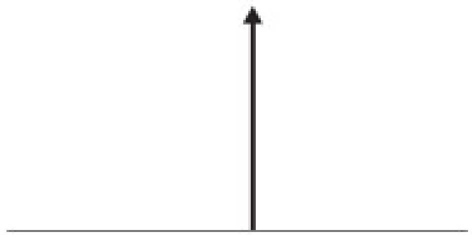
$$
\begin{aligned}
f(t) * \delta(t - T) &= \int_{-\infty}^{\infty} f(\tau) \cdot \delta(t - T - \tau) d\tau \\
&= \int_{-\infty}^{\infty} f(\tau) \cdot \delta(\tau - (t - T)) d\tau \\
&= f(t - T)
\end{aligned}
$$

Fourier Transform

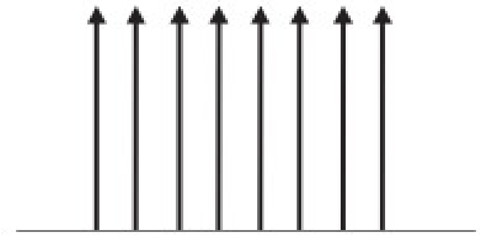
Transforms allow us to achieve three goals.

- The convolution of two functions, which arises in the computation of the out- put of an LTI system, can be computed more easily by transforming the two functions, multiplying the transformed functions, and then computing the inverse transform.
- Transforms convert a linear differential equation into a simpler algebraic equation.
- They give insight into the natural response of a system: the natural frequencies at which it oscillates. This allows us to quickly determine whether there are special frequencies at which the system becomes unstable, whether there are special frequencies that cause the system output to become zero, and the frequencies at which most of the output magnitude can be found.

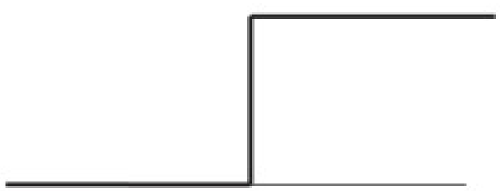
Useful functions in signal-processing theory: Delta function (top left), comb function (top right), step function (middle left), box function (middle right),



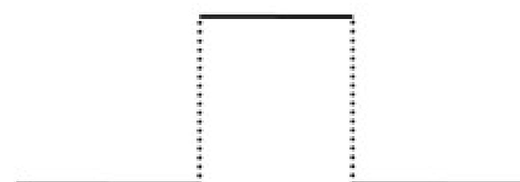
Delta function



Comb function $\Delta_T(t) \stackrel{\text{def}}{=} \sum_{k=-\infty}^{\infty} \delta(t - kT)$



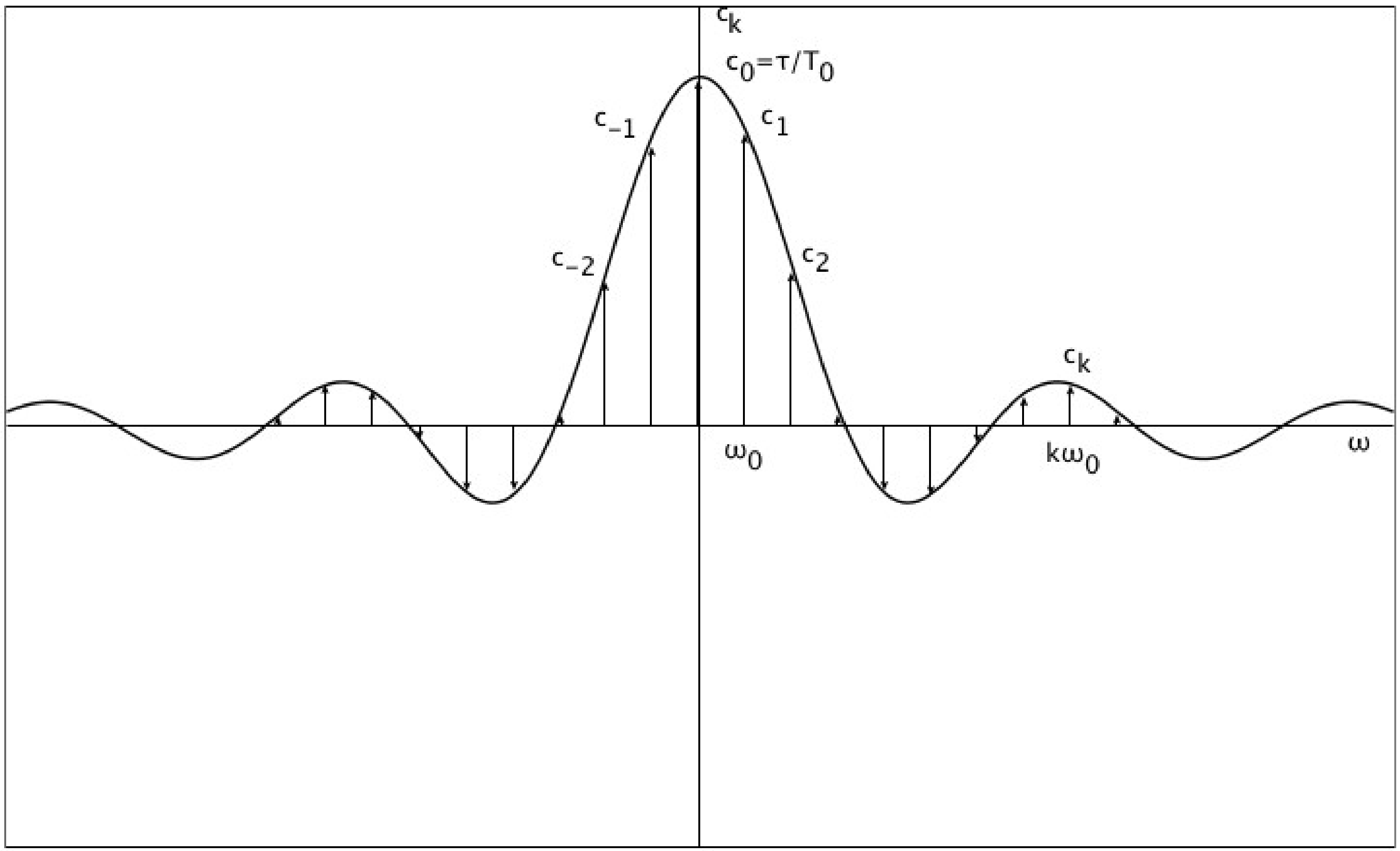
Step function $H(x) = \int_{-\infty}^{x} \delta(t)\mathrm{d}t$



Box function

$$\mathrm{rect}(t) = \sqcap(t) = \begin{cases} 0 & \text{if } |t| > \frac{1}{2} \\ \frac{1}{2} & \text{if } |t| = \frac{1}{2} \\ 1 & \text{if } |t| < \frac{1}{2} \end{cases}$$

Sinc function



Transform	Signal Type	Transform Description
Fourier series	Continuous, periodic	Sum of sinusoids
Fourier	Continuous, periodic or aperiodic	Integral of sinusoids
Discrete-time Fourier	Discrete, periodic or aperiodic	Sum of sinusoids
Laplace	Continuous, periodic or aperiodic	Integral of complex exponentials
Z	Discrete, periodic or aperiodic	Sum of complex exponentials

Sampling Theorem and Aliasing

- If these signals are to be digitized and reproduced back in the analog domain, the number of samples required to ensure that both starting and ending analog signals are the same is clearly different.

Sampling Theorem and Aliasing

The relationship states that the signal has to be sampled using a sampling frequency that is greater than twice the maximal frequency occurring in the signal. Or, it can more formally be stated as follows:

1. A bandlimited signal $f(t)$ with max frequency ωF is fully determined from its samples $f(nT)$ if $2\pi/T > 2\omega F$
2. The continuous signal can then be reconstructed from its samples $f(nT)$ by convolution with the filter

$$r(t) = \text{sinc}(\omega F(t - nT)/2\pi)$$

This is an alternative statement of the Nyquist criterion and can be remembered as: *To prevent aliasing, the sampling function should have a frequency that is at least twice that of the highest frequency component of a signal.*

What happens if your sampling frequency is higher than your Nyquist frequency? The answer is: nothing special. When it comes to reproducing your analog signal, it is guaranteed to have all the necessary frequencies and, hence, the same signal is reproduced.

What happens if your sampling frequency is lower than your Nyquist frequency? In this case, you have a problem because all the frequency content is not well captured during the digitization process. As a result, when the digital signal is heard/viewed or converted back into analog form, it does not correspond to the initial starting analog signal. This results in artifacts, which is termed as aliasing.

Aliasing is the term used to describe loss of information during digitization. Such undesirable effects are experi- enced for 1D signals such as sound, 2D signals such as images and graphics, and even in 3D signals such as 3D graphics. Next, we discuss the aliasing effects commonly observed in the spatial or temporal domains

Aliasing

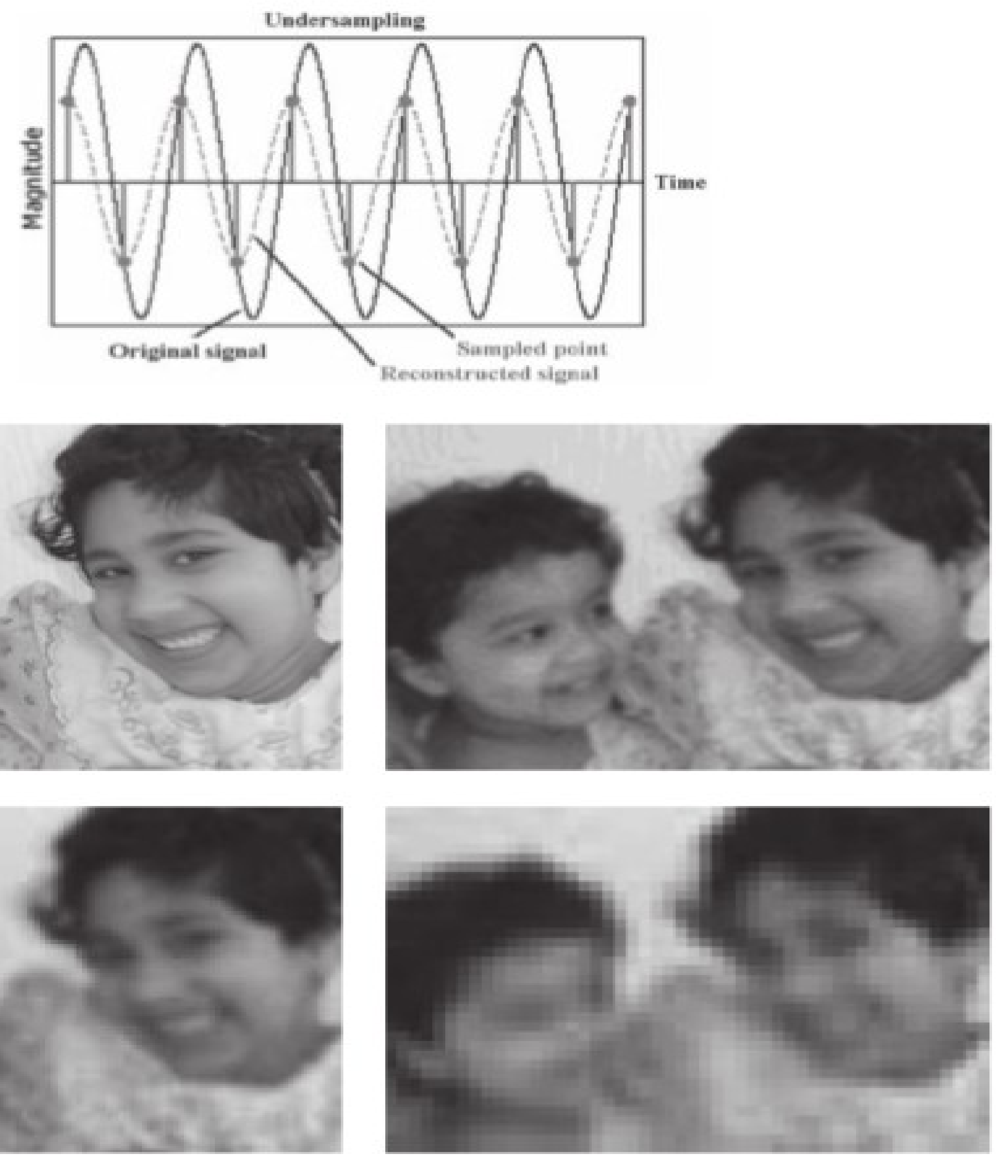
- Aliasing in Spatial Domains

  Aliasing effects in the spatial domain are seen in all dimensions.
- Aliasing in the Temporal Domain

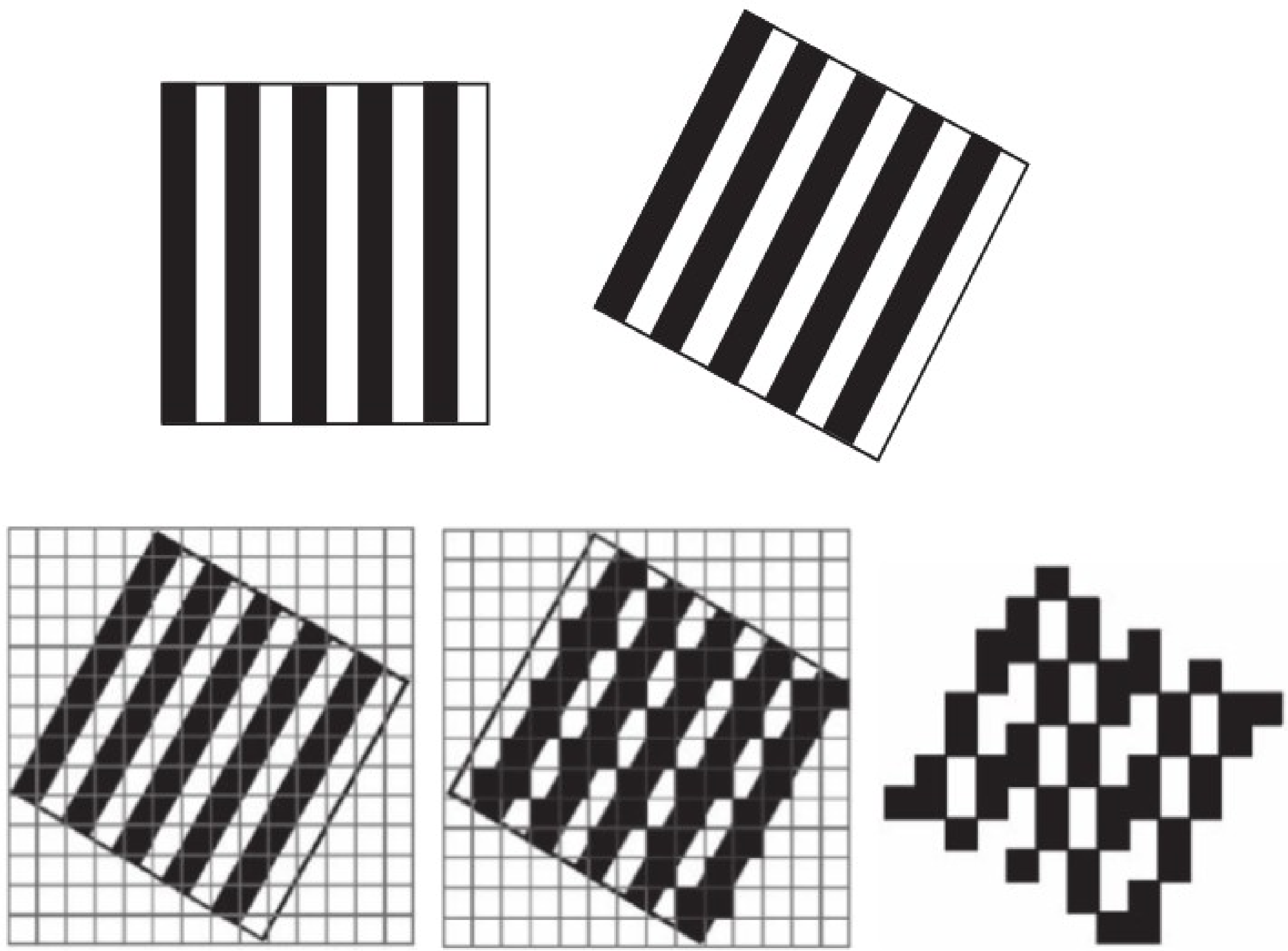
  Examples of temporal aliasing can be seen in western movies, by observing the motion of stage coach wheels.
- Moiré Patterns and Aliasing

  Another interesting example of aliasing, called the moiré effect, can occur when there is a pattern in the image being photographed, and the sampling rate for the digitalimage is not high enough to capture the frequency of the pattern.

Aliasing examples in the spatial domain. The top figure shows an example in one dimension, where original signal is shown along with sampled points and the reconstructed signal. The bottom set of figures show a 2D input image signal. The top left shows the original signal and the remaining three show examples of the signal reconstruction at different sampling resolutions. In all cases, the output does not match the input because the sampling resolution falls below the Nyquist requirement.



Moiré pattern example. The vertical bar pattern shown on the top left is rotated at an angle to form the input signal for sampling. The bottom row illustrates the sampling process. A sampling grid is superimposed to produce a sampled output. The grid resolution corresponds to the sampling resolution. The middle figure in the second row shows the sampled values at the grid locations and the right figure shows just the sampled values. Although the input has a pattern, the output pattern is not like the original.



Filtering

- The sampling theorem states sampling requirements to correctly convert an analog signal to a digital signal.
- Input analog frequency range can enable this conversion
- Analog filtering techniques are commonly used to capture a variety of commonly used signals such as audio and images. However, in the digital world, digital data manipulations also require filters.

Digital Filters

- In signal processing, the function of a filter is to remove unwanted parts of the signal, such as random noise and undesired frequencies, and to extract useful parts of the signal.
- There are two main kinds of filters: analog and digital.
- Analog filter uses analog electronic circuits made up from components such as resistors, capacitors, and operational amplifiers (op-amps) to produce the required filtering effect.

Digital Filter

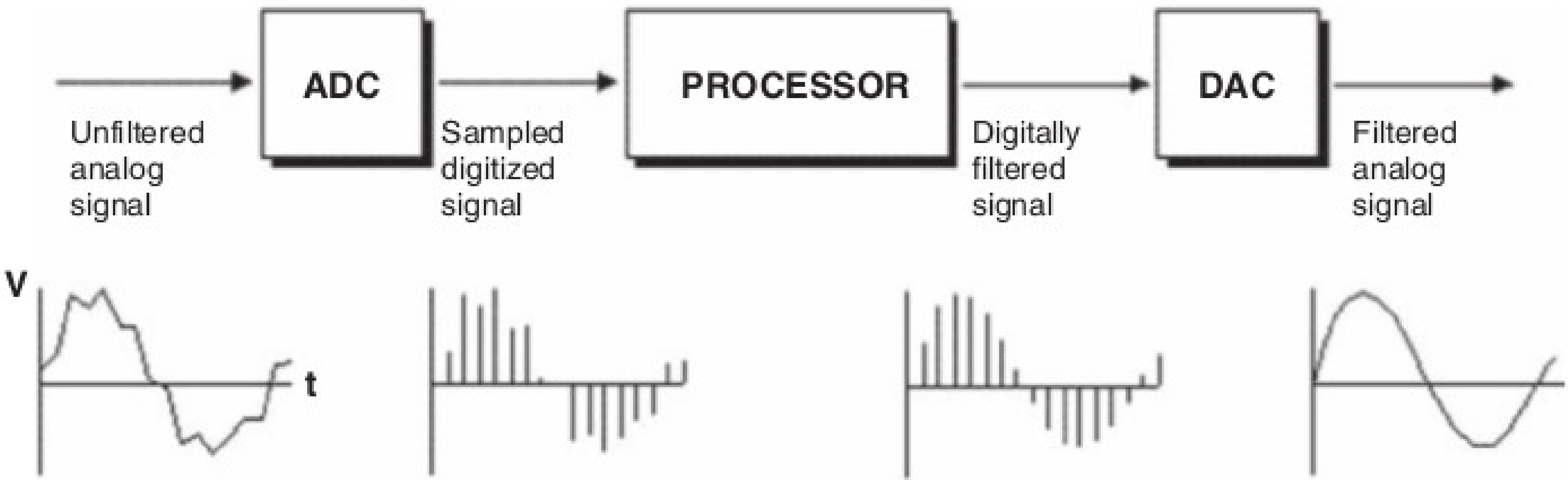


Figure 2-15 Digital filters—A digital filter takes a digital signal as input and produces another signal with certain characteristics removed. In the bottom row, the noisy analog signal is digitized and filtered.

Advantages of Digital Filters

- A digital filter is programmable
- Digital filters are easily designed, tested, and implemented on a general-purpose computer or workstation
- Digital filters can be combined in parallel or cascaded in series with relative ease by imposing minimal software requirements.
- The characteristics of analog filter circuits (particularly those containing active components) are subject to drift and are dependent on temperature.
- Unlike their analog counterparts, digital filters can handle low-frequency signals accurately.
- Digital filters are more versatile in their ability to process signals in a variety of ways.

Filtering Dimension

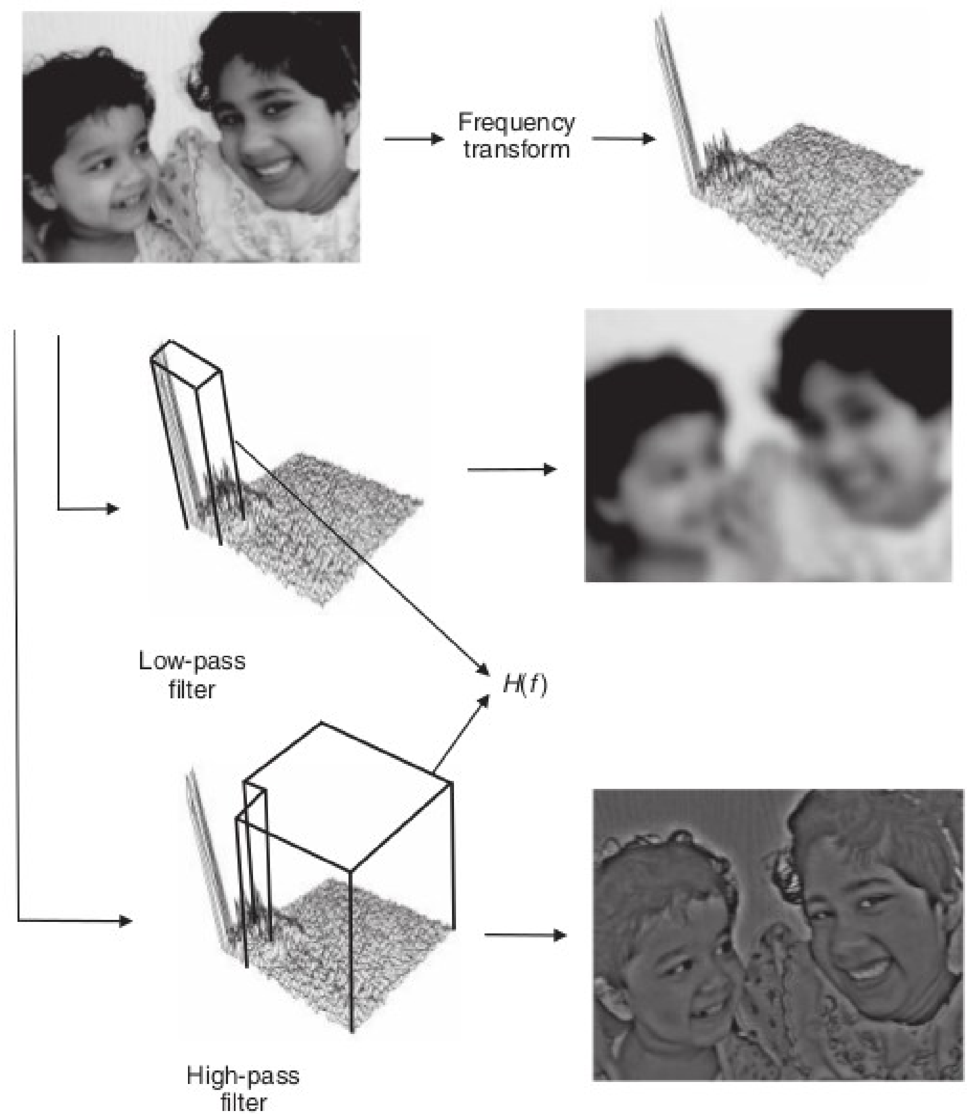
- Filtering in 1D

  A 1D signal is normally represented in the time domain with the x-axis showing sampled positions and the y-axis showing the amplitude values.

- Filtering in 2D

  In the case of 2D signals, such as images, filters are an important part in digital image processing. Digital image signals are contaminated with interference, noise, and aliasing artifacts in the capture process.

Digital filtering on images—The top row depicts an image and its frequency transform, showing frequencies in two dimensions. The middle row shows the effect of a low-pass filter. Here, only the frequencies inside the box are used to reconstruct the output image. The bottom row shows the effect of a high-pass filter.



Subsampling

- Filtering data prior to sampling is, thus, a practical solution to acquiring the necessary quantity of reliable digitized data. The cut off frequency of the filter depends largely on the signal being digitized and its intended usage. However, once in the digital domain, there is frequently a need to further decrease (or increase) the number of samples depending on the bandwidth requirements, storage capacity, and even content creation requirements.

Fourier Theory

Fourier theory is mathematically involved, but here we try to give an intuitive understanding to some simple holistic concepts behind the theory. The sine and cosinefunctions defined over the fundamental frequencies are known as basis functions, while Ai and Bi are known as the frequency coefficients.

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