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Multimedia Systems

Algorithms, Standards, and Industry Practices

Chapter 2

Digital Data Acquisition

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Digital Data Acquisition

- One of first task in creating multimedia content using text, audio, video and image is to record these individual media types into a digital form.
- Physical world around us is in a continuous form.
- We sense the environment by sensing light, sound energy, pressure, temperature, motion.

1. Analog and Digital Signal

- Understanding analog to digital conversion is very important to know on how digital multimedia data is produced.
- We cant ignore this area because it will determine how the quality of multimedia data that being process.
- Understanding analog to digital conversion helps in the design on end-to-end system with desired quality.

Analog and Digital Signal

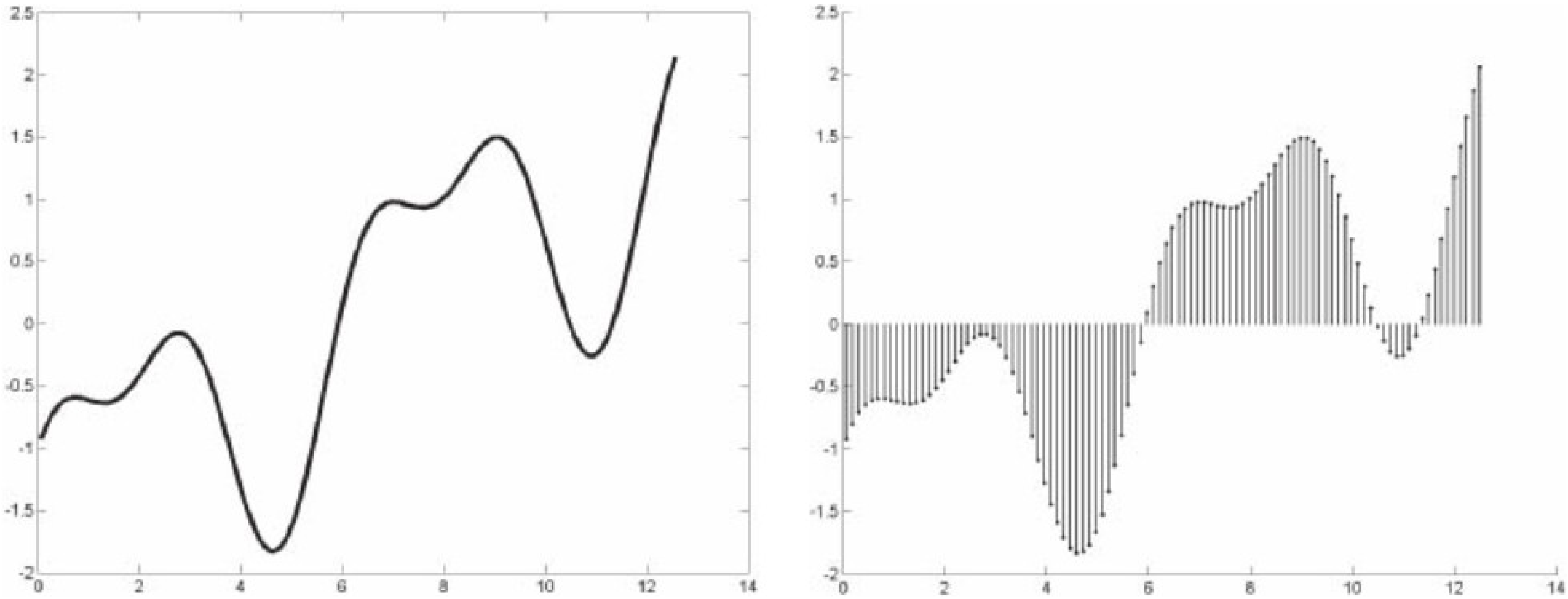

Figure 2.1 Analog vs Digital Signal

Advantage of Digital over Analog Signal

- With digital signal,it is possible to create complex,interactive content.
- In digital medium,we can access each unit of information.
- Stored digital signals do not degrade over time or distance as analog signal do.
- Digital data can be efficiently compressed and transmitted across digital networks.
- Digital data easy to store.

2. Analog to Digital Conversion

- 2 processes that involved, sampling and quantization
- The reverse process of converting digital signals to analog is known as interpolation.
- The desirable property of analog to digital conversion is no artifact are created in the digital data. It means that when signal is converted back to analog domain, it will look the same as original.
- Multimedia data is digital but the end device can be analog, such as CRT monitor.

Sampling

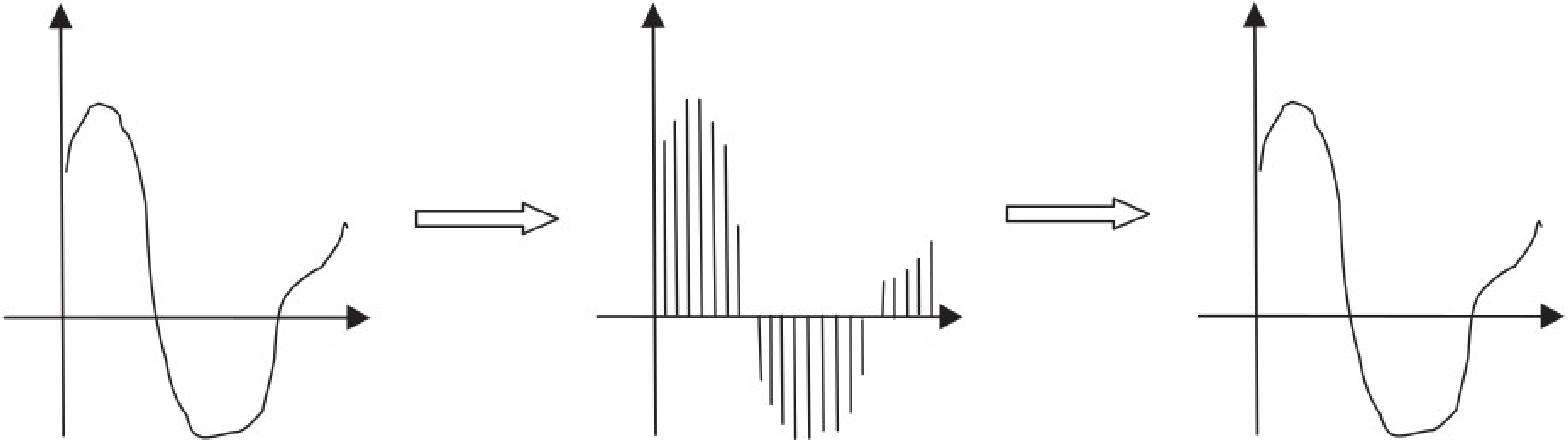

Figure 2.2 Analog to Digital Conversion and Interpolation

Sampling

- If the sampling is increase (reduce T), the storage requirement is also increase and maybe redundant
- Vice versa, if T increase (f decrease), It will lead to artifact, where the signal is under sampled.
- Sampling is done across one,two and three dimension signal.

Quantization

- Quantization deals with encoding the signal value at every sampled location.
- It is a representation of every sample slice in bits.
- The amplitude of the signal will be represented by a binary value with rounding function.

Quantization

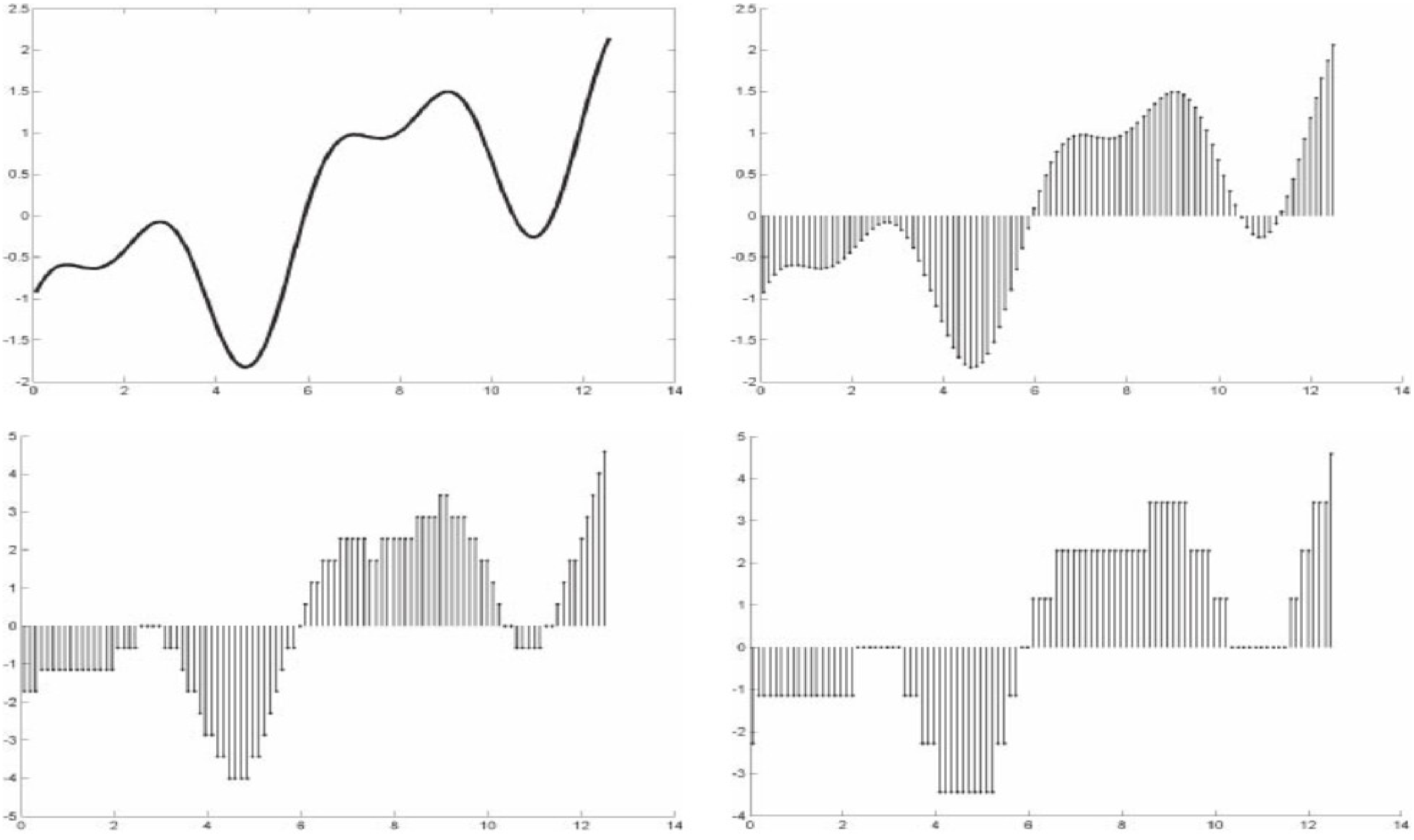

Figure 2.3 Original Analog Signal (upper left) is sampled and quantized at different quantization level. 8 bit (256 levels), 4 bit (16 levels), and 3 bit (8 levels) were used to produce the digital signals on the top right, bottom left and bottom right, respectively.

Quantization

- The quantized value will differ from the original signal, thus introduce error.
- The error decrease as the number of bits used to represent the sample increase.
- The is irreversible loss, so the selection of quantization bits need to be right.
- Audio signal which represent music, must be quantized on 16 bits, whereas speech only required 8 bits.

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Quantization on Image

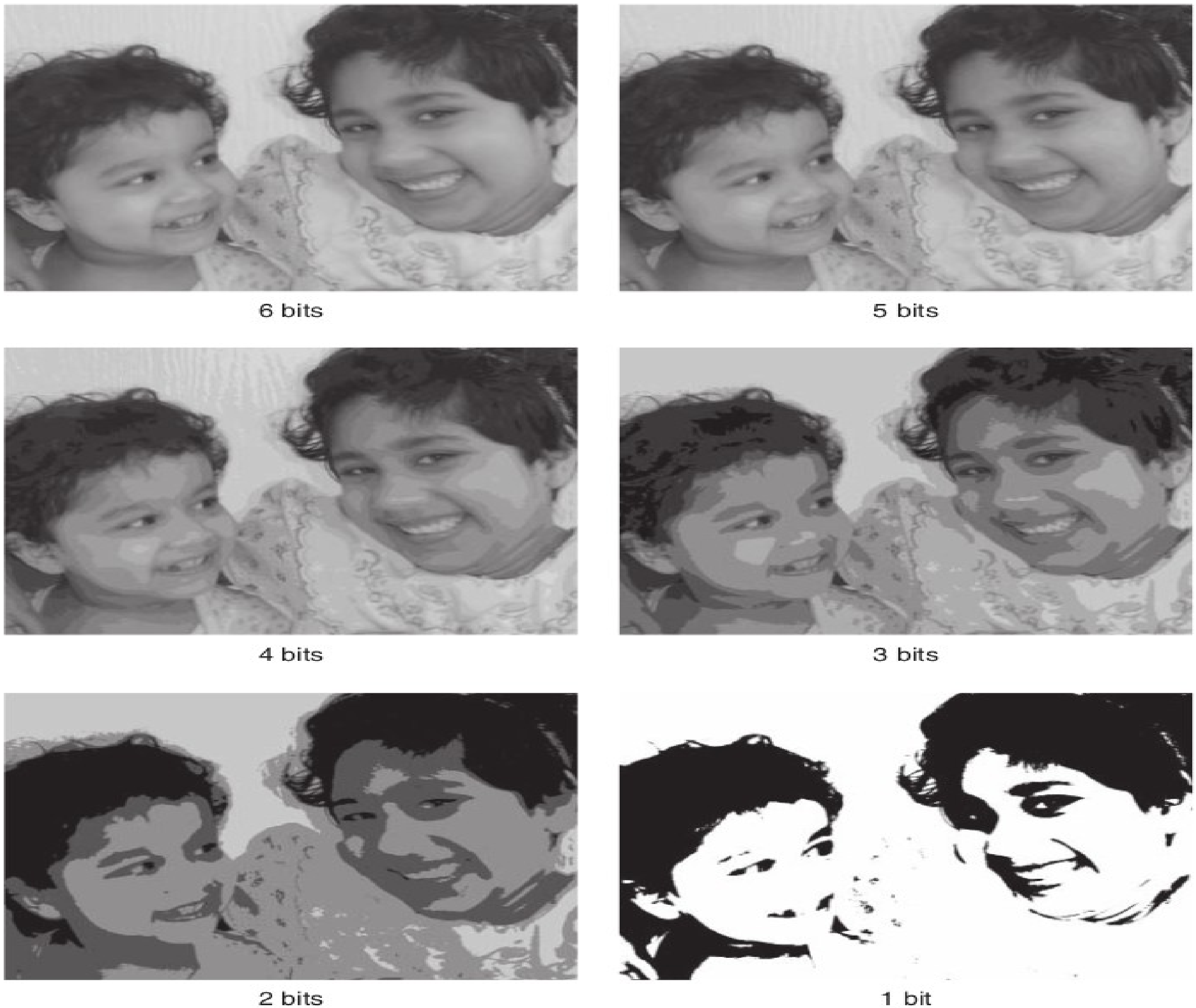

Figure 2.4 Example of Quantization from original 8 bit image to 6 down to 1 bit quantization.

Type of Quantization, Uniform and Non-Uniform

- Uniform quantization intervals is in which the output range of the signal is divided into fixed and uniformly separated intervals depending on the number of bits used.
- With this quantization error is equally distributed.
- However, for some signal that the distribution of all output values is nonuniform,it is more correct to distribute the quantization intervals nonuniformly.
- Eg:the output intensity values of many audio signals such as human speech are more likely to be concentrated at lower intensity levels, rather than at higher intensity levels in the dynamic audio range.

Bit Rate

- Bit Rate describe the number of bits being produce per second.
- Bit Rate is of critical importance when it comes to storing a digital signal, or transmitting it across networks, that might have high,low or even varying bandwidth.

Bit Rate

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 2.5 Sampling Rate, Quantization and Bit Rate for Typical Signals

3. Signal and Systems

- This part discuss some of fundamental elements in the field of digital signal processing.
- This is to understand signals, how they can be sampled, and the limitation that signals impose on the sampling of signals.
- First distinction need to be made regarding the type of signal under consideration.

Type of Signal

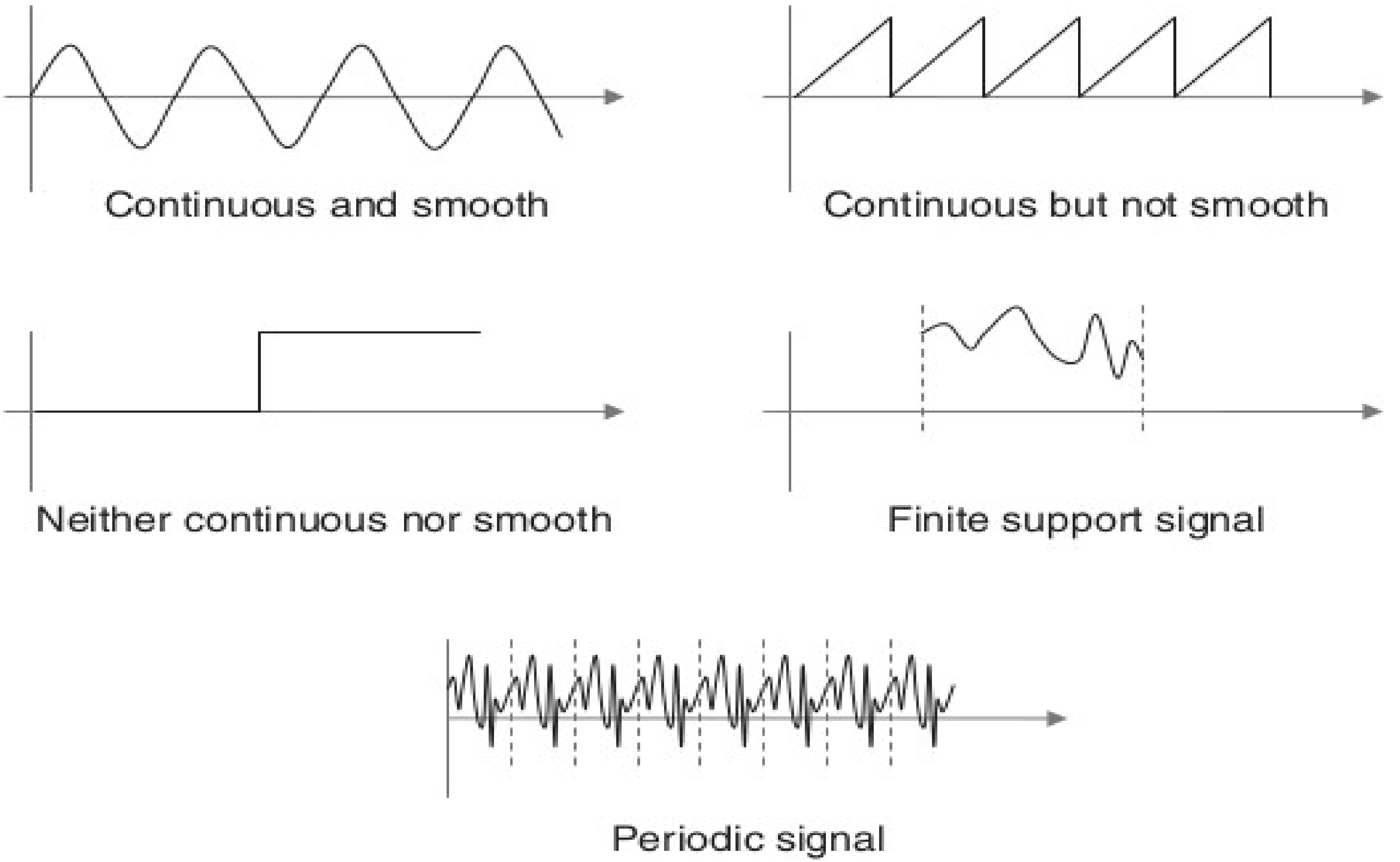

Figure 2.7 Sample Signal with different kind of properties.

Linear Time Invariant Systems

- Any operation that transforms a signal is called a systems.
- Time invariant of a system is define 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.
- If an input is affected by a time delay, it should produce a corresponding time delay in the output.

Linear Time Invariant Systems

- 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 is termed as the time domain point of view of the system.
- 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.
- This transfer function works in the frequency domain, and expresses the systems operation on the input signal in terms of its frequency representation to produce an output signal in the frequency domain is then the product of the transfer function and the Fourier transform of the input.

Linear Time Invariant Systems

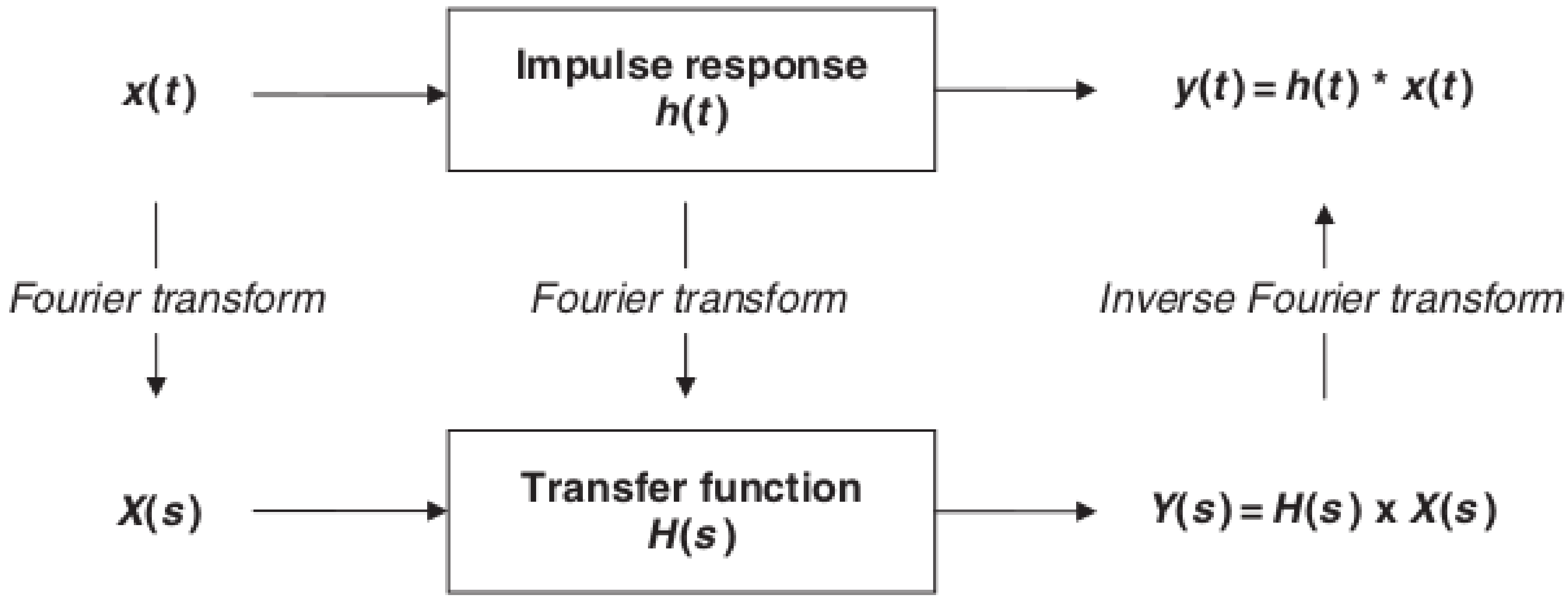

Figure 2.7 Relationship Between Impulse Response Function
in Time domain and Transfer Function in Frequency Domain

Linear Time Invariant Systems

- Useful signal such as delta function,box function and step function, play an important role in many signaling operations.
- Fourier transform : Fourier proposed to represent any periodic, continuous signal as a sum of individual complex sinusoids.
- In other words every periodic,continous signal can be expressed as a weighted combination of sinusoid waves.
- The fourier formulation represent a transformation of the signal into frequency space.

Fourier Transform

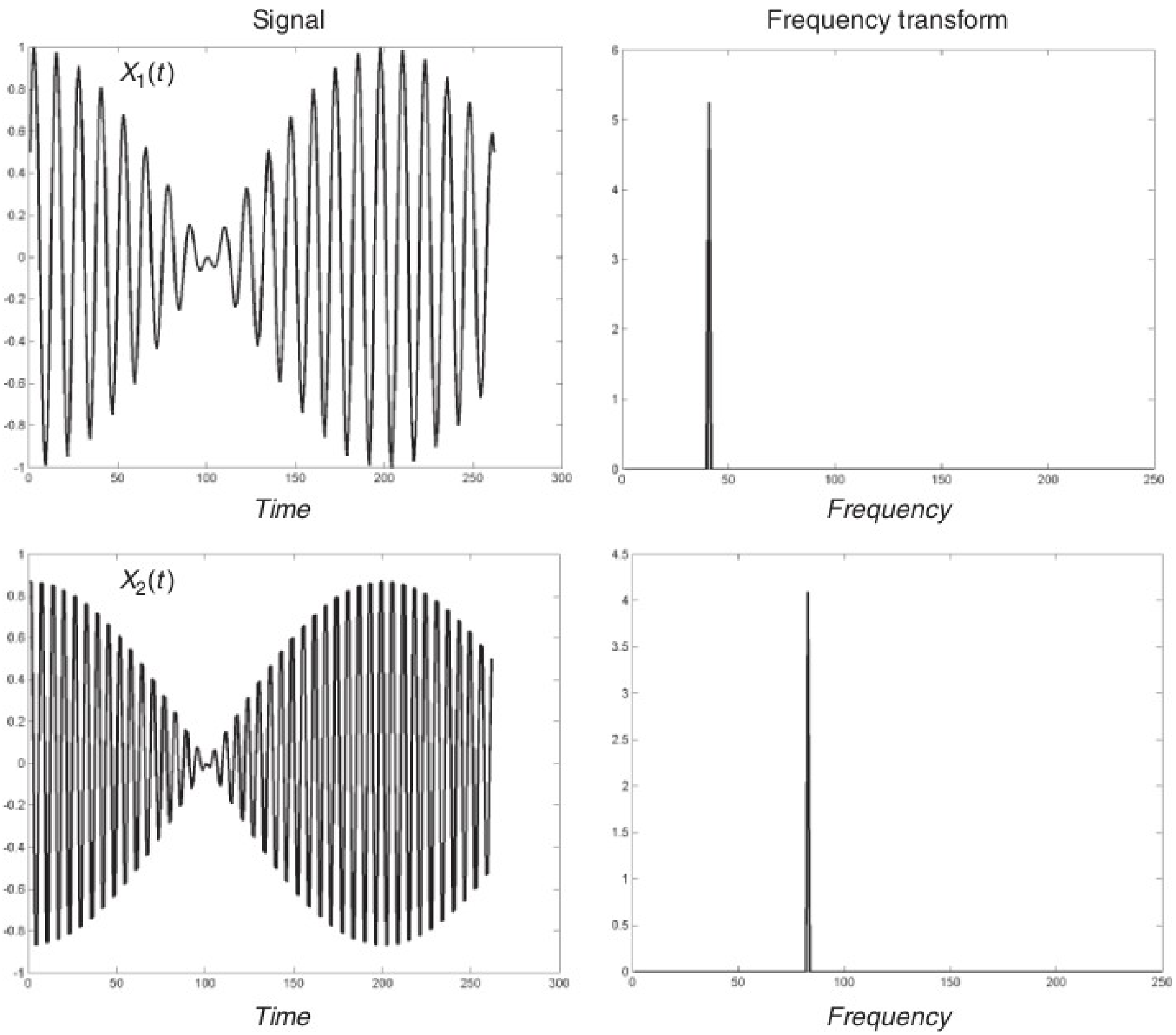

Figure 2.8 Two signal having different frequencies and shown in left and their fourier transforms on the right. The transform show the amount of energy at all frequencies.

Sampling Theorem and Aliasing

- Here we discuss on the rate of sampling that should occur.
- Below is different type of signal that need different sampling treatment.

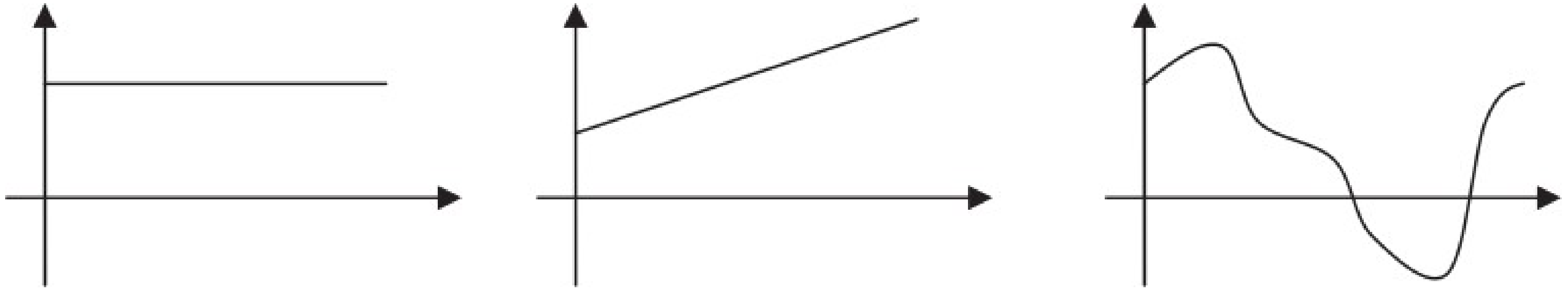

Figure 2.9 Example of simple one-dimensional function.
Different numbers of samples are required to digitize them

Sampling Theorem and Aliasing

- The number of sample that require to digitize signals in figure 2.9 are clearly different.
- The first signal where the signal is not changing, using one sample is suffice.
- Two sample are required for the second signal.
- As we go from left to right, the frequency is increase, hence the number of sample to digitize also goes up.
- This define Nyquist theorem that state the the sampling rate must be 2 times of the maximum frequency of the signal.
- Eg : if the signal has a maximum frequency of 10kHz, it should be sampled at a frequency greater than 20kHz. 20KHz is known as the Nyquist sampling frequency for that signal.

Over and Under Sampling

- If the sampling rate is bigger than the Nyquist frequency, this is oversampling. It will generate more digital data than needed.
- However, if the sampling rate is lower than Nyquist frequency, this will cause a problem.
- Its because all the frequency content is not well captured.
- It will produce different analog output.
- This called *aliasing.*

Aliasing

- If the sampling rate is bigger than the Nyquist frequency, this is oversampling. It will generate more digital data than needed.
- However, if the sampling rate is lower than Nyquist frequency, this will cause a problem.
- Its because all the frequency content is not well captured.
- It will produce different analog output.
- This called *aliasing.*
- Aliasing can occur in spacial and temporal domain.

Aliasing

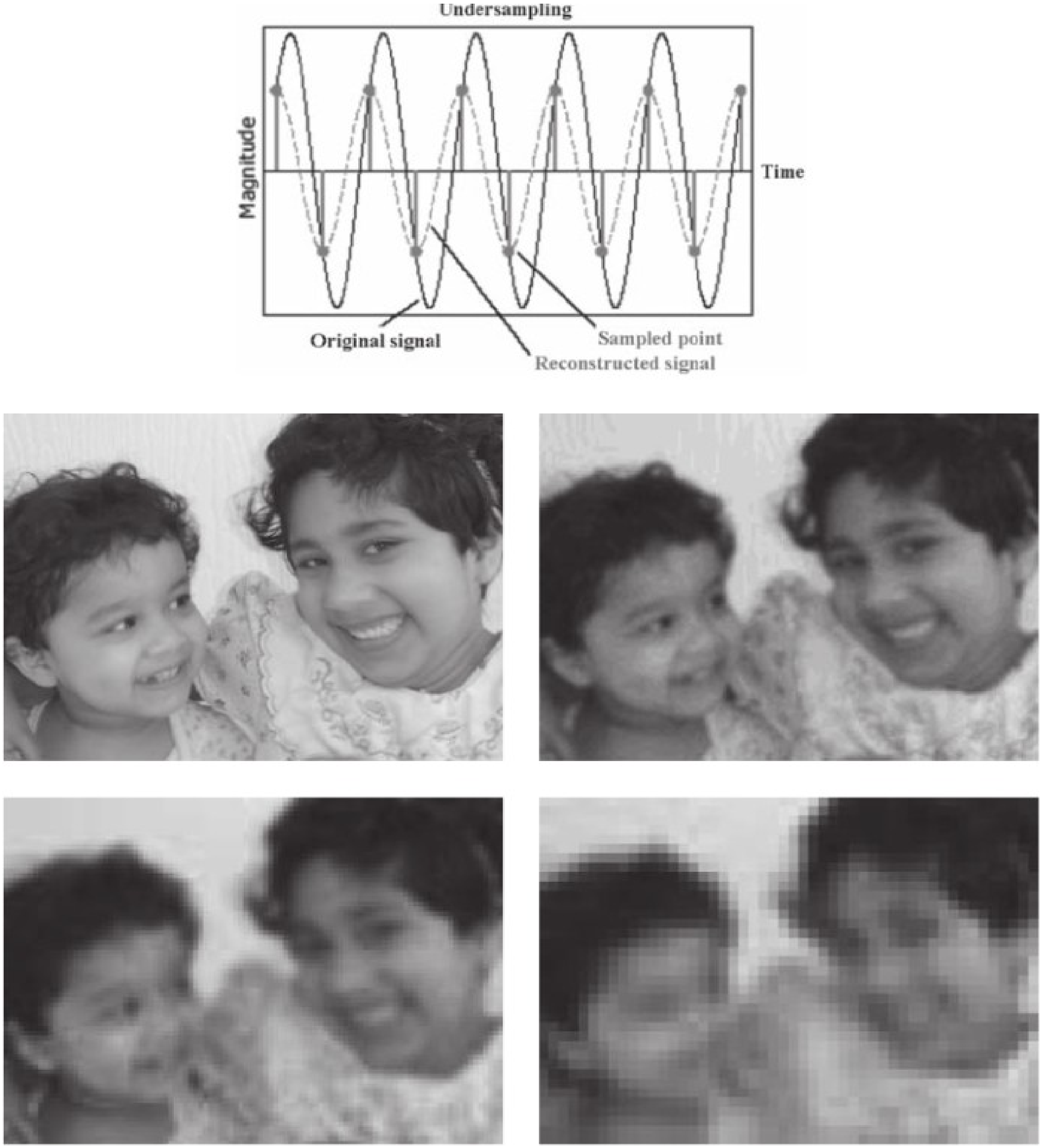

Figure 2.10

Aliasing example in spacial domain. Top figure shows 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 image signal. The top left shows the original signal and the remaining three show example 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.

5. Filtering

- Knowing the input analog frequency range can enable us to correctly convert it to digital.
- However, practically we do not know the range of frequency beforehand hence universal sampling rate cannot be derived.
- For example, human can only hear maximum of 4kHz of frequency. If we have recording of higher frequency, we can apply filter so that the frequencies above cutoff limit are eliminated.
- In practice, sampling is always accompanied by filter.

Filtering

- Function of filter is to remove unwanted parts of the signal, such as random noise, undesired frequencies and to extract useful parts of the signal within range of frequencies.
- There are analog and digital filter.
- Analog filter uses analog electronics circuit made up from components such as registers, capacitors and operational amplifiers.
- Digital filter uses digital numerical computations on sampled,quantized values of the signal.
- The processing can be done using PC or specialized DSP chip.

Digital Filter

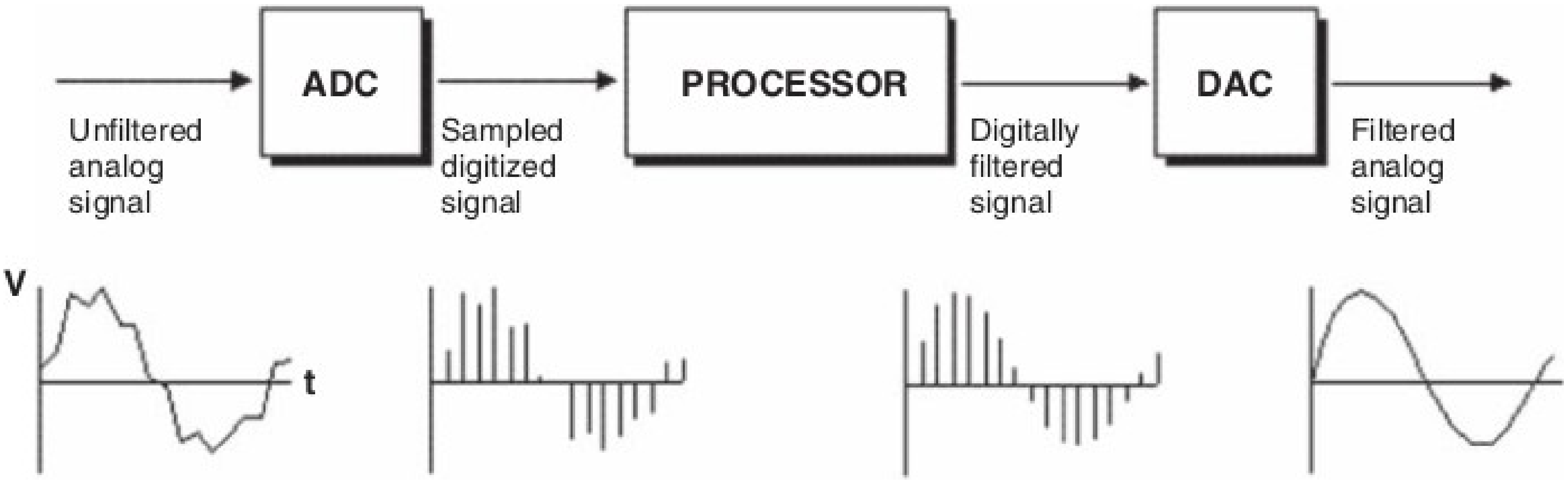

Figure 2.11 Digital Filter – A digital filter takes a digital signal as input and produces another signal with certain characteristics removed. In the bottom row, the noisy analog signal is digitized and filtered.

Advantages of Digital Filter

- Digital Filter is Programmable.
- Digital Filter 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.
- Analog filter are subject to drift and are dependent on temperature.
- Digital Filter can handle low frequency signals accurately.
- Digital Filters are more versatile in their ability to process signal in variety ways.

Categories of Filter

- Low pass, band-pass and high-pass filter.
- Low pass filter remove high frequency. Usually use to avoid aliasing while sampling.
- High pass filter remove low frequency content and are used to enhance edged and sharpen the image.
- Band pass filters output signal containing the frequencies belonging to a defined band. Usually used in sub-band filtering for audio and wavelet theory.

Filtering in 1D

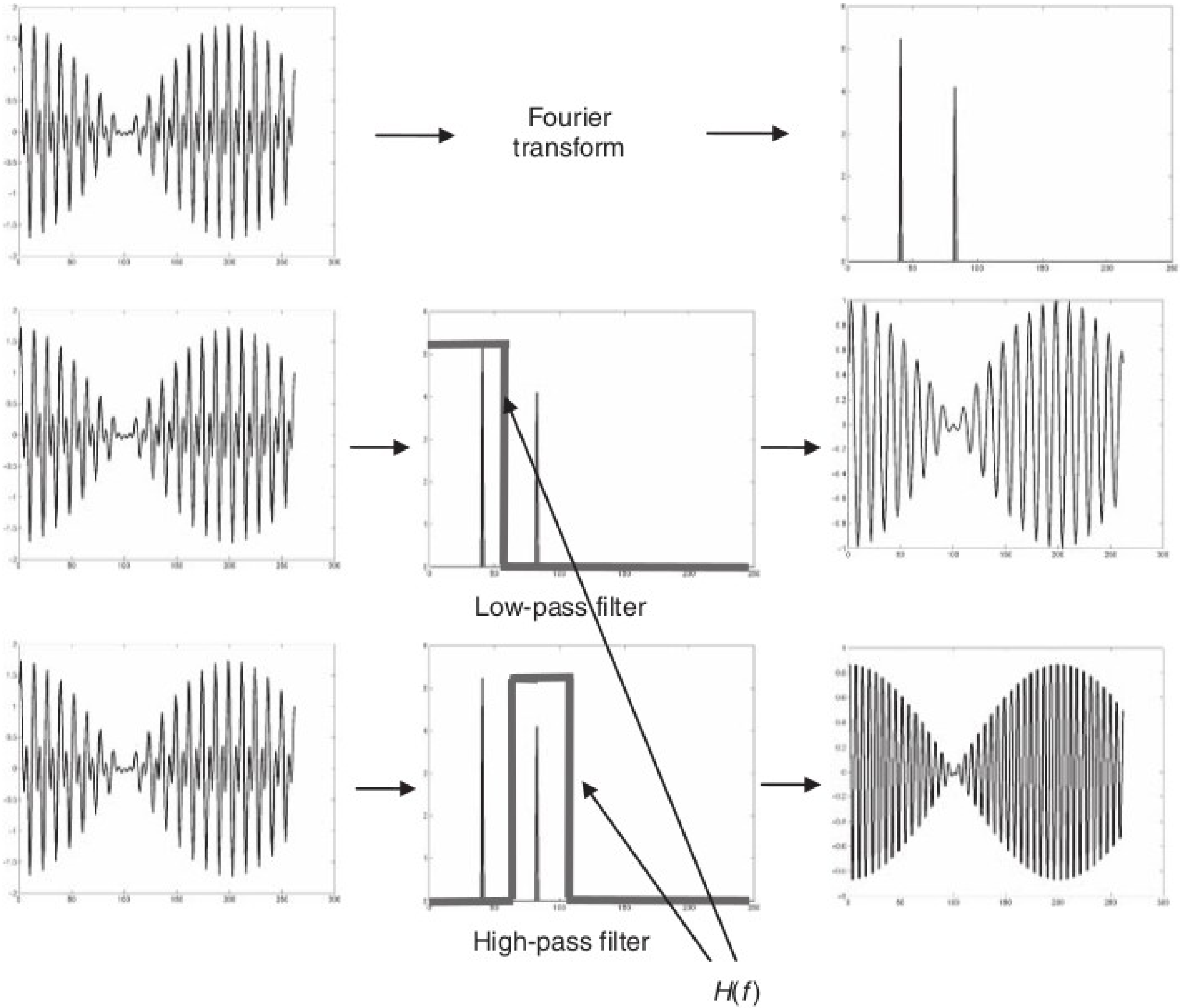

Figure 2.12 The center and bottom row show the effect of low-pass filter and a high-pass filter on the signal. The output produced contains frequencies in accordance with filters response.

Filtering in 2D

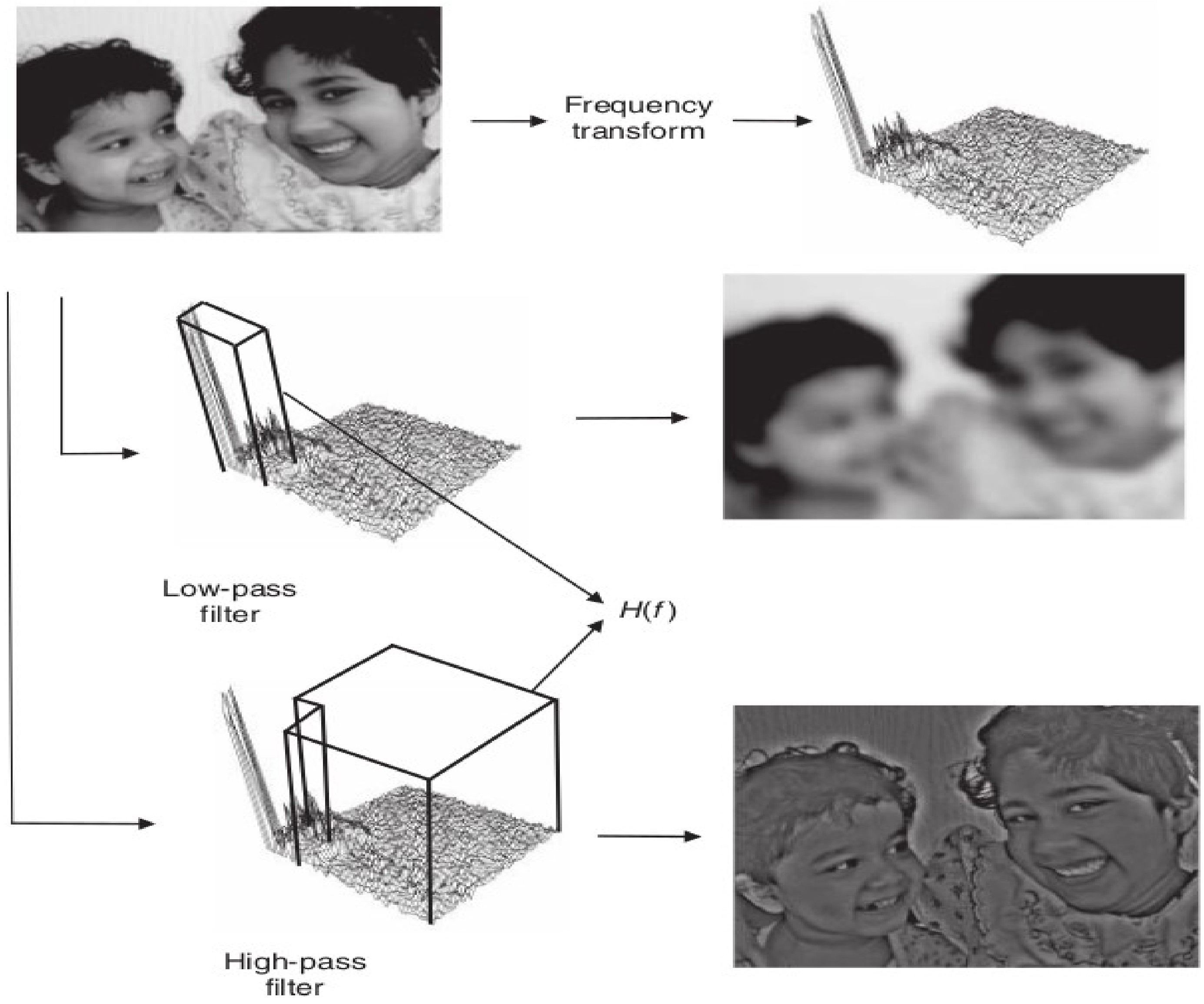

Figure 2.13 Digital Filtering on Images – The middle row shows the effect of a low-pass filter. Here, only frequencies inside the box are used to reconstruct the output image. The bottom row shows the effect of a high-pass filter.

Subsampling

- In Digital domain,there is frequently a need to further decrease (or increase) the number of samples depending on bandwidth requirements, storage capacity and even content requirements.
- Eg: even the image is captured at a high resolution,we might need to downsize it to incorporate it in a document, advertisement and so on.
- Such post digitization sampling adjustment are achieved by a process called subsampling.
- Sub sampling will cause an aliasing if below the Nyquist frequency.

Subsampling on 1D

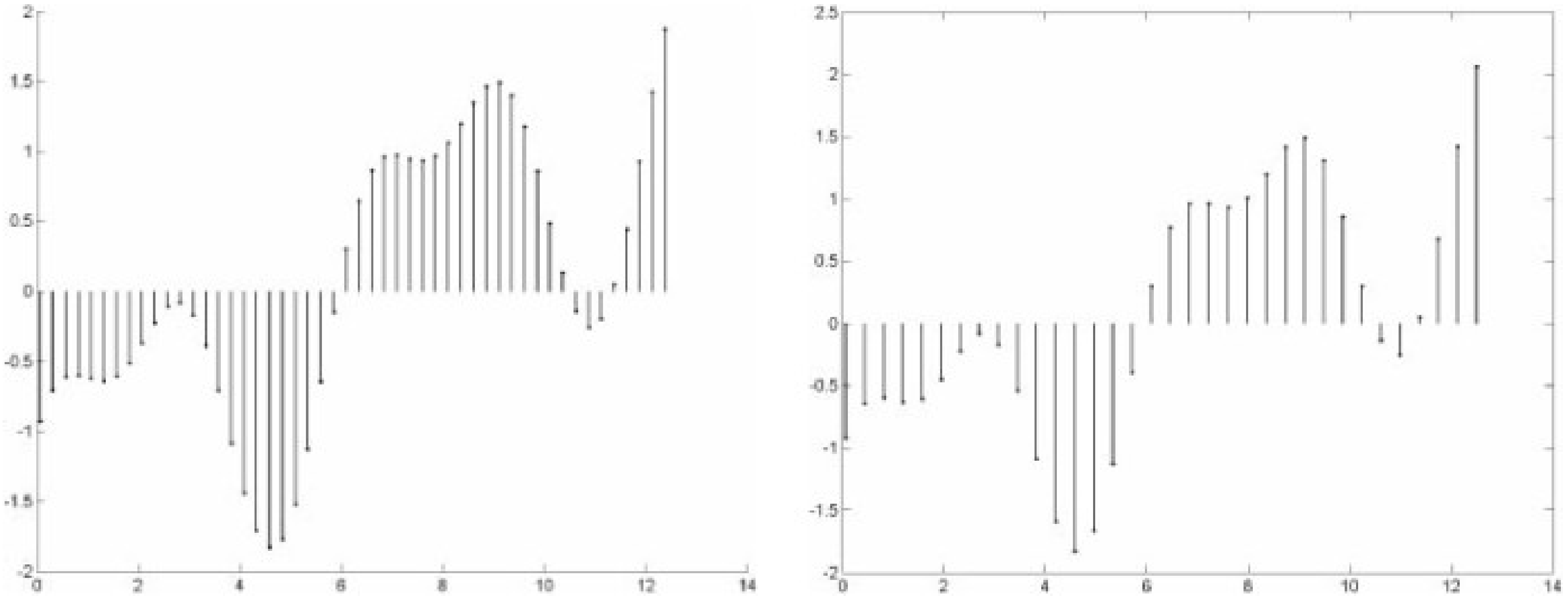

Figure 2.14 Original signal shown on the left, subsampled by 2 producing signal on the right. The number of sampled on the right is half the number of samples on the left.

Subsampling on 2D

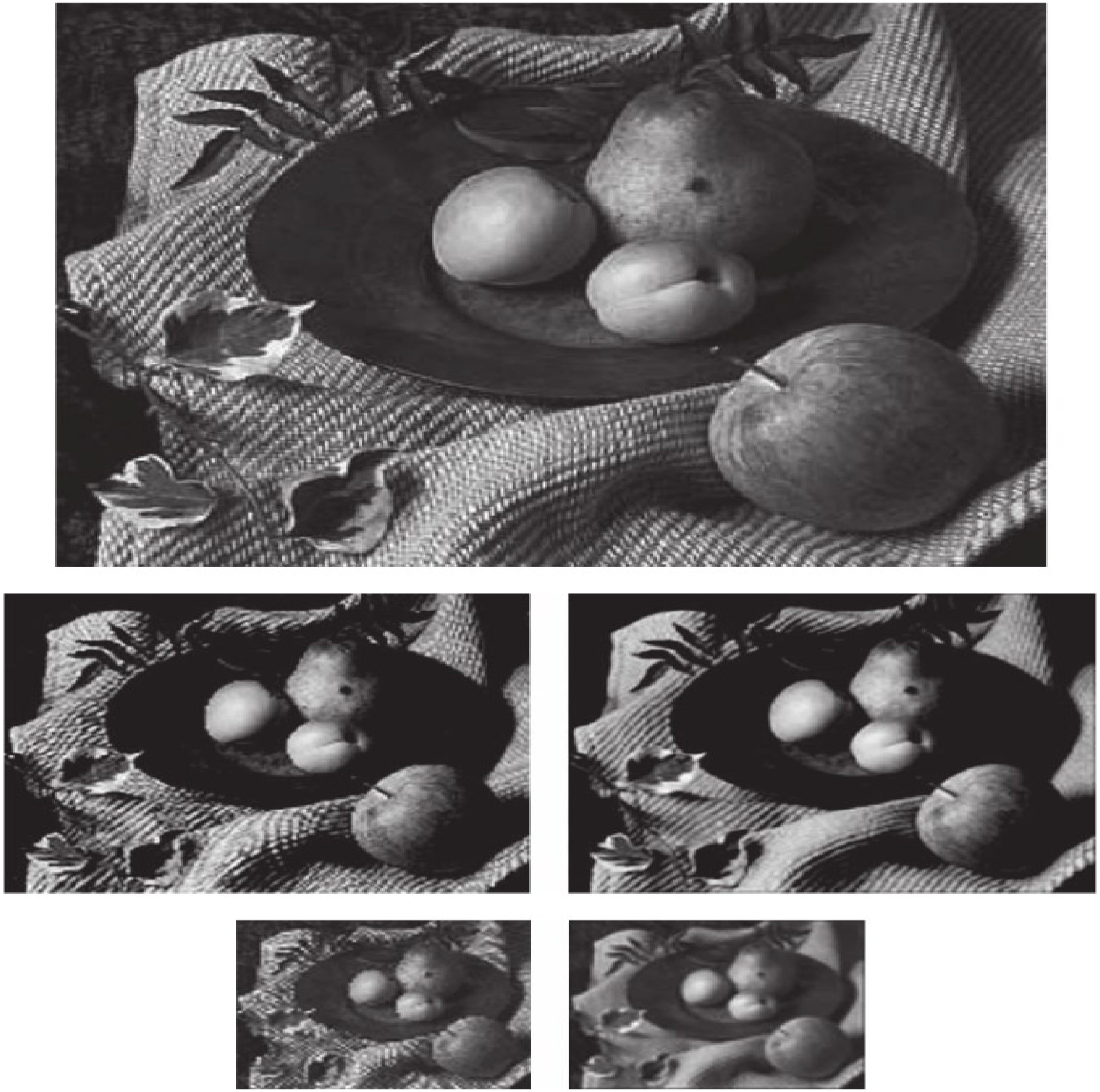

Figure 2.15 The first row shows the original image, the second row shows the image subsampled by 2 and the third row is subsampled by 4. The left image on row second and third is without filtering and the right image is filtered before subsampled.

Fourier Theory

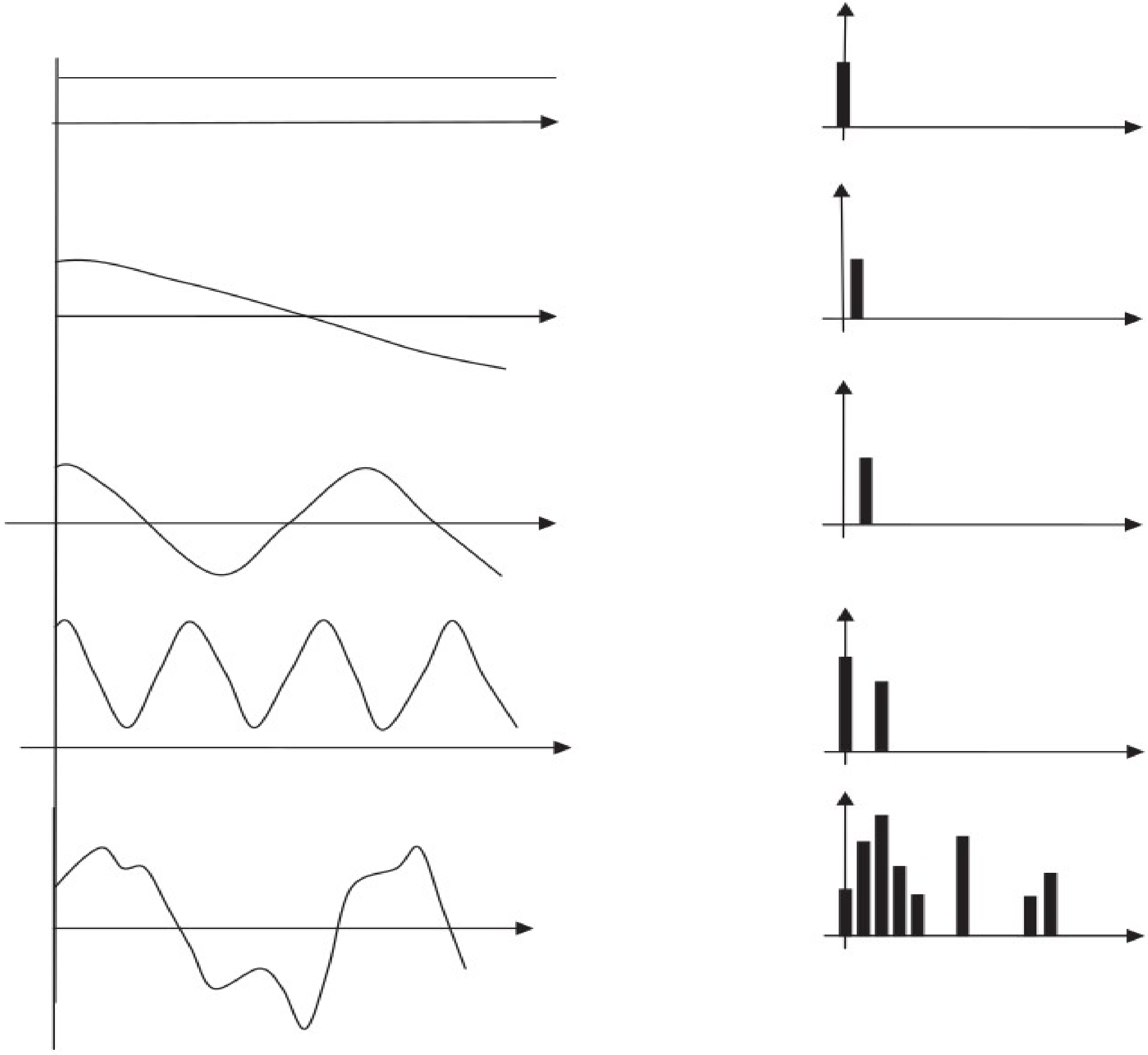

Figure 2.16 The left column shows examples of time domain functions. The right column shows their frequency transforms. The first three functions are simple, and have only one fundamental frequency, the last two are composed of more than one

Conclusion

- This chapter discuss theoretical fundamental in order to understand the background of how multimedia data is digitally produce and process.
- This chapter didn’t provide mathematical fundamental but give overall overview of how digital data is produce and preserve.
- This knowledge will be use throughout the course and will be refer when needed.