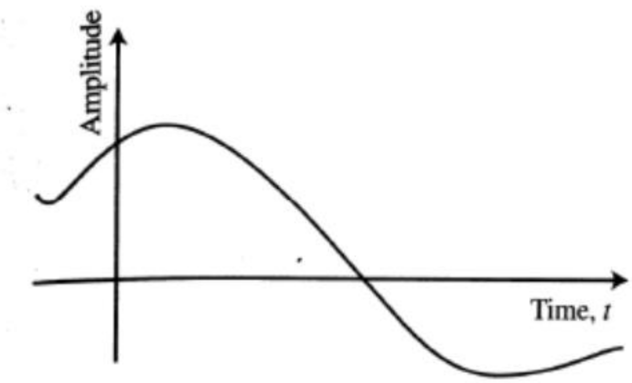
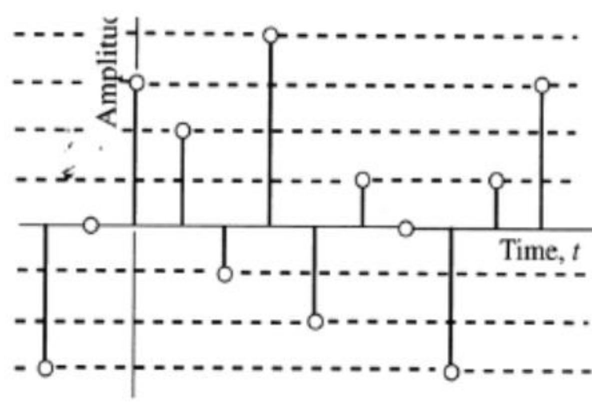
ANALOG VERSUS DIGITAL SIGNALS

A continuous-time signal with a continuous amplitude is usually called an **analog signal**. A speech signal is an example of an analog signal.
A discrete time signal with discrete valued amplitudes represented by a finite number of digits is referred to as a **digital signal**





CONVOLUTIONS

The convolution of *f* and *g* is written *f*∗*g*, using an asterisk or star. It is defined as the integral of the product of the two functions after one is reversed and shifted. As such, it is a particular kind of integral transform:

$$(f * g)(t) \stackrel{\text{def}}{=} \int_{-\infty}^{\infty} f(\tau)\, g(t-\tau)\, d\tau$$
$$= \int_{-\infty}^{\infty} f(t-\tau)\, g(\tau)\, d\tau.$$

While the symbol *t* is used above, it need not represent the time domain. But in that context, the convolution formula can be described as a weighted average of the function $f(\tau)$ at the moment *t* where the weighting is given by $g(-\tau)$ simply shifted by amount *t*. As *t* changes, the weighting function emphasizes different parts of the input function.

For functions *f*, *g* supported on only $[0,\infty)$ (i.e., zero for negative arguments), the integration limits can be truncated, resulting in

$$(f*g)(t)=\int_0^t f(\tau)\,g(t-\tau)\,d\tau \quad \text{for} \quad f,g:[0,\infty)\to\mathbb{R}$$

In this case, the Laplace transform is more appropriate than the Fourier transform below and boundary terms become relevant.

Circular convolution

When a function g_T is periodic, with period *T*, then for functions, *f*, such that $f*g_T$ exists, the convolution is also periodic and identical to:

$$(f*g_T)(t)\equiv\int_{t_0}^{t_0+T}\left[\sum_{k=-\infty}^{\infty} f(\tau+kT)\right] g_T(t-\tau)\,d\tau,$$

where t_o is an arbitrary choice. The summation is called a periodic summation of the function *f*.

When g_T is a periodic summation of another function, *g*, then $f*g_T$ is known as a *circular* or *cyclic* convolution of *f* and *g*.
And if the periodic summation above is replaced by f_T, the operation is called a *periodic* convolution of f_T and g_T

SAMPLING AND QUANTIZATION

Nearly all data acquisition systems sample data with uniform time intervals. For evenly sampled data, time can be expressed as:
$T = (N\ 1)\Delta t$
where N is the sampling index which is the number of equally spaced samples. For most Fourier analyzers N is restricted to a power of 2.
• The sample rate or the sampling frequency is:
$f = 1 = (N-1)\Delta f$
Sampling frequency is the reciprocal of the time elapsed Δt from one sample to the next.
• The unit of the sampling frequency is cycles per second or Hertz (Hz), if the sampling period is in seconds.
• The sampling theorem asserts that the uniformly spaced discrete samples are a complete representation of the signal if the bandwidth *fmax* is less than half the sampling

rate. The sufficient condition for exact reconstructability from samples at a uniform sampling rate *fs* (in samples per unit time) ($fs \geq 2fmax$).

Aliasing

One problem encountered in A/D conversion is that a high frequency signal can be falsely confused as a low frequency signal when sufficient precautions have been avoided.

• This happens when the sample rate is not fast enough for the signal and one speaks of **aliasing**.

• Unfortunately, this problem can not always be resolved by just sampling faster, the signal's frequency content must also be limited.

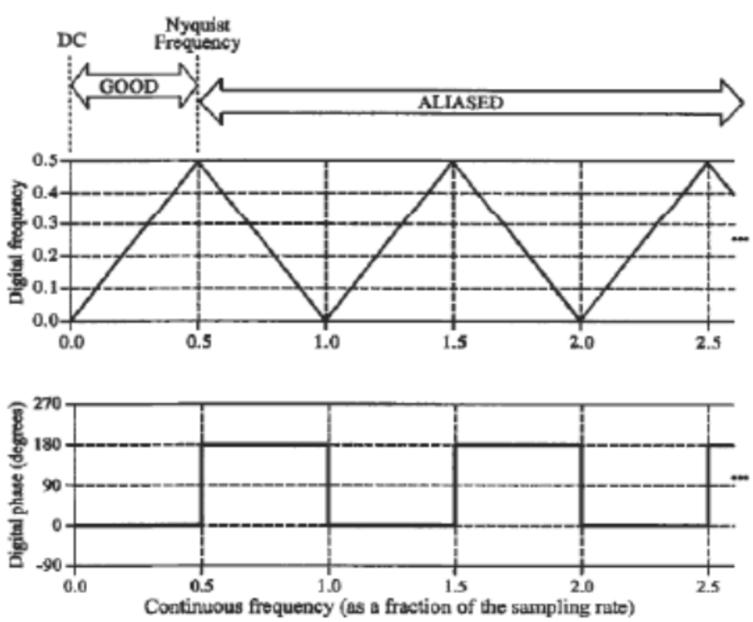
• Furthermore, the costs involved with postprocessing and data analysis increase with the quantity of data obtained. Data acquisition systems have finite memory, speed and data storage capabilities. Highly oversampling a signal can necessitate shorter sample lengths, longer time on test, more storage medium and increased database management and archiving requirements The central concept to avoid aliasing is that the sample rate must be at least twice the highest frequency component of the signal

($fs \geq 2fmax$).

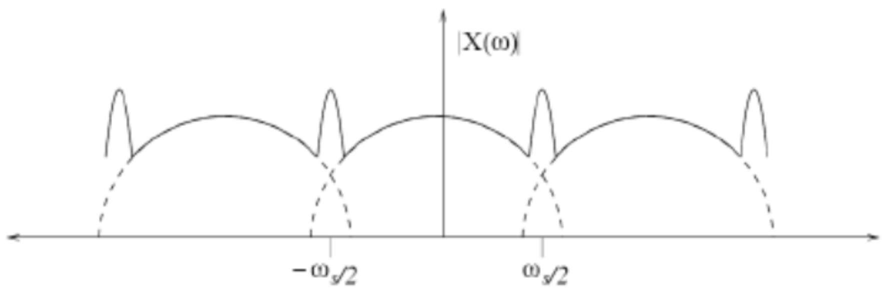
We define the Nyquist or cut-off frequency

• The concept behind the cut-off frequency is often referred to as $2\Delta t$

Shannon's sampling criterion. Signal components with frequency content above the cut-off frequency are aliased and can not be distinguished from the frequency components below the cut-off frequency. Conversion of analog frequency into digital frequency during sampling is shown in the figure. Continuous signals with a frequency less **than** one-half of the sampling rate are directly converted into the corresponding digital frequency. Above one-half of the sampling rate, aliasing takes place, resulting in the frequency being misrepresented in the digital data. Aliasing always changes a higher frequency into a lower frequency between 0 and 0.5. In addition, aliasing may also change the phase of the signal by 180 degrees.



If any energy in the original signal extends beyond the Nyquist frequency, it is folded back into the Nyquist interval in the spectrum of the sampled signal. This folding is called aliasing fs≥2fmax



Spectrum of sampled signal $x_s(t)$ with overlap - aliasing.

Quantization

Quantization is involved to some degree in nearly all digital signal processing, as the process of representing a signal in digital form ordinarily involves rounding. Quantization also forms the core of essentially all lossy compression algorithms. The difference between an input value and its quantized value (such as round-off error) is referred to as **quantization error**. A device or algorithmic function that performs quantization is called a **quantizer**. An analog-to-digital converter is an example of a quantizer.

Because quantization is a many-to-few mapping, it is an inherently non-linear and irreversible process (i.e., because the same output value is shared by multiple input values, it is impossible in general to recover the exact input value when given only the output value).

The set of possible input values may be infinitely large, and may possibly be continuous and therefore uncountable (such as the set of all real numbers, or all real numbers within some limited range). The set of possible output values may be finite or countably infinite. The input and output sets involved in quantization can be defined in a rather general way. For example, *vector quantization* is the application of quantization to multi-dimensional (vector-valued) input data

CONCEPTS OF SIGNAL PROCESSING

In the case of **analog signals**, most signal processing operations are usually carried out in the **time domain**.

• In the case of **discrete time signals**, **both time domain and frequency domain** applications are employed.

• In either case, the desired operations are implemented by a combination of some **elementary operations** such as:

– Simple time domain operations

– Filtering

– Amplitude modulation

The three most basic time-domain signal operations are:

- **Scaling**
- **Delay**
- **Addition**

Scaling is simply the multiplication of a signal by a positive or a negative constant. In the case of analog signals, this operation is usually called **amplification** if the magnitude of the multiplying constant, called **gain**, is greater than one. If the magnitude of the multiplying constant is less than one, the operation is called **attenuation.** Thus, if $x(t)$ is an analog signal, the scaling operation generates a signal $y(t)=\alpha x(t)$, where α is the multiplying constant.

Delay operation generates a signal that is delayed replica of the original signal. For an analog signal $x(t)$, $y(t)=x(t-t0)$ is the signal obtained by delaying $x(t)$ by the amount $t0$, which is assumed to be a positive number. If $t0$ is negative, then it is an **advance** operation

Addition operation generates a new signal by the addition of signals. For instance, $y(t)=x1(t)+x2(t)-x3(t)$ is the signal generated by the addition of the three analog signals $x1(t)$, $x2(t)$ and $x3(t)$.

TYPICAL APPLICATIONS

The main applications of DSP are

audio signal processing,

sometimes referred to as audio processing, is the intentional alteration of auditory signals, or sound, often through an audio effect oreffects unit. As audio signals may be electronically represented in either digital or analog format, signal processing may occur in either domain. Analog processors operate directly on the electrical signal, while digital processors operate mathematically on the digital representation of that signal.

audio compression

bit-rate reduction involves encoding information using fewer bits than the original representation.[2]Compression can be either lossy or lossless. Lossless compression reduces bits by identifying and eliminating statistical redundancy. No information is lost in lossless compression. Lossy compression reduces bits by identifying unnecessary information and removing it.[3] The process of reducing the size of a data file is referred to as data compression. In the context of data transmission, it is called source coding (encoding done at the source of the data before it is stored or transmitted) in opposition to channel coding.[4]

digital image processing,

is the use of computer algorithms to perform image processing on digital images. As a subcategory or field of digital signal processing, digital image processing has many advantages over analog image processing. It allows a much wider range of algorithms to be applied to the input data and can avoid problems such as the build-up of noise and signal distortion during processing. Since images are defined over two dimensions (perhaps more) digital image processing may be model in the form of multidimensional systems

speech processing

s the study of speech signals and the processing methods of these signals. The signals are usually processed in a digital representation, so speech processing can be regarded as a special case of digital signal processing, applied to speech signal. Aspects of speech processing includes the acquisition, manipulation, storage, transfer and output of speech signals.

speech recognition,

is the inter-disciplinary sub-field of computational linguistics which incorporates knowledge and research in the linguistics, computer science, and electrical engineering fields to develop methodologies and technologies that enables the recognition and translation of spoken language into text by computers and computerized devices such as those categorized as Smart Technologies and robotics. It is also known as "automatic speech recognition" (ASR), "computer speech recognition", or just "speech to text" (STT).

digital communications, radar, sonar, financial signal processing seismology and biomedicine. Specific examples are speech compression and transmission in digitalmobile phones, room correction of sound in hi-fi and sound reinforcement applications, weather forecasting,economic forecasting, seismic data processing, analysis and control of industrial processes, medical imagingsuch as CAT scans and MRI, MP3 compression, computer graphics, image manipulation, hi-fi loudspeakercrossovers and equalization, and audio effects for use with electric guitar amplifiers

ADVANTAGES OF DIGITAL SIGNAL PROCESSING COMPARED WITH ANALOG SIGNAL PROCESSING

Accracy

Implimentation of sophisticated algorithms

Storage

Noise reduction

APPLICATIONS OF SIGNAL PROCESSING IN BIOMEDICAL ENGINEERING

- io signal processing – for electrical signals representing sound, such as speech or music
- Speech signal processing – for processing and interpreting spoken words
- Image processing – in digital cameras, computers and various imaging systems
- Video processing – for interpreting moving pictures
- Wireless communication - waveform generations, demodulation, filtering, equalization
- Control systems
- Array processing – for processing signals from arrays of sensors
- Seismology
- Financial signal processing – analyzing financial data using signal processing techniques, especially for prediction purposes.
- Feature extraction, such as image understanding and speech recognition.

- Quality improvement, such as noise reduction, image enhancement, and echo cancellation.
  - (Source coding), including audio compression, image compression, and video compression