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GLOSSARY

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Aliasing may occur during analogue-to-digital conversion or during downsampling. If components in the sound are higher than the Nyquist frequency of the output digital signal, then these components will be "aliassed" in that signal. The components are too high to be encoded properly using the sampling frequency in question, and will consequently appear in the form of spurious "alias" components which have lower frequencies, below the Nyquist frequency. In D/A conversion these components may be removed using an anti-aliasing filter. A digital low-pass filter is used when downsampling.

The **amplitude** of waveform is the magnitude of its values, either the individual samples values, or the peak value within the waveform, as in, for instance, the amplitude of a sinusoid.

The **amplitude spectrum** of a waveform is the set of amplitudes of the sinusoidal cmoponents into which a fourier transform would analyse it.

Analogue-to-digital conversion (A/D), is the overall process of taking an analogue signal, such as the fluctuating voltage from a microphone and turning it into a digital signal, consisting of a stream of numbers. It is useful to think of A/D conversion as composed of two processes, quantisation and sampling.

An **Anti-aliasing filter** is a low-pass filter which has a cut-off just below the Nyquist frequency . The filter thus removes all frequencies above the Nyquist frequency and prevents aliasing.

Attenuation is the reduction in level of a signal.

AutoCAD window is the drawing area, its surrounding menus, and the command line

Autocorrelation **is the convolution of a signal with itself.**

A band-pass filter **attenuates frequency components which lie outside a particular band of frequencies.**

A band-stop filter **attenuates frequency components which lie within a particular band of frequencies.**

Bits
Digital signals are generally stored in a binary-integer format. The number of bits used to encode each sample specifies the degree of amplitude quantisation.

Bitmap
The digital representation of an image having bits referenced to pixels. In color graphics, a different value represents each red, green, and blue component of a pixel.

The **coefficients** of a filter are the weights which are used when summing input values in order to produce each output value.

A **Complex Fourier transform** is a Fourier transform, whose input is an array of complex numbers rather than real numbers. A complex Fourier transform is not used frequently in digital signal processing.

Complex numbers are numbers that are composed of a "complex" of two parts. These parts are a real number and an imaginary number. Complex numbers may be converted to phase and amlpitude values via Cartesian-to-polar coordinate conversion.

Any waveform may be considered as the sum a set of **components**, which are each sinusoidal waveforms. A Fourier analysis will decompose any sound into a fixed set of such components. However, real sounds, particularly those produced by resonating bodies, such as musical instruments, vocal chords, and so forth, tend to show prominent peaks in the power spectrum indicating that these sources produce certain components predominantly. These components often form harmonic series.

The **Continuous Fourier transform** is the mathematical form of the Fourier transform, in which the input is a function, rather than a list of numbers, as in the discrete Fourier transform. The input and output are consequently infinite in extent and in resolution on the time axis.

Convolution is an operation on two waveforms in which one waveform is multipled with the other waveform at a number of different sample offsets. The values in each product waveform are summed to produce each sample in the output waveform, which is, therefore, the sum-of-products waveform as a function of waveform offset. In FIR filtering, one waveform is an impulse response of some sort, and this waveform will need to be time-reversed before convolution.

Cross-correlation is the convolution of one signal by another (e.g. the left- and right-channels of a stereo signal).

The **cut-off frequency** of a low- or high-pass filter the frequency above or below which a filter attenuates an input signal by a specified amount

Digital-to-analogue conversion (D/A) is the process of taking a digital dignal and converting it into an analogue signal. The initial ouput of the D/A converter is a squared-off waveform that jumps between the different quantised values that are represented by the digital signal during each sampling period. This signal contains many unwanted frequency components above the Nyquist frequency, which should by removed by a reconstruction filter.

The **discrete Fourier transform** is the Fourier transform of a digital signal. A digital signal has a fixed resolution (the sampling period) and a limited extent on the time axis (duration). As a result the output frequency spectrum also has a limited frequency resolution (caused by the limited duration of the input) and limited frequency range, caused by the limited temporal resolution of the input. The frequency resolution is the reciprocal of the input duration and the frequency range is half the sampling frequency. Thus, frequency spectra produced by the dft have just over half as many values (N/2+1) as the input waveform. The extra value is for the amplitude at zero frequency and its (arbitrary) phase. This value represents the DC-offset of the waveform combined with its overall power. The discrete Fourier transform may be executed with less computation by using a more efficient algorithm called the fast Fourier transform or FFT.

Downsampling is the process of reducing the sampling frequency of a digital signal. If the signal contains components which are higher than the new Nyquist frequency, then the signal will first need to be low-pass filtered using a cut-off below that frequency in order to avoid aliassing. The signal may then be decimated.

The **Fast Fourier transform** (FFT) is a form of discrete Fourier transform which executes more efficiently. The only constraint is that the input waveform must have a number of samples which is an integer power of two. Although this constraint may seem limiting, the FFT can be applied to any input through the use of zero padding.

A **finite impulse response (FIR) filter**, is a digital filter for which each output sample is a weighted sum of a finite set of input samples. The array of weights, known as coefficients or taps have the same form as the impulse response of the filter. Such filters are computationally expensive, since they work by convolution, but can perform any transformation of the amplitude or phase spectrum.

Filtering is a general term for transforming a signal in some way. It usually means an alteration of its amplitude spectrum, but may also be a manipulation of its phase spectrum, or a non-linear operation such as logarithmic compression. Digital filtering may be achieved using a finite impulse response (FIR) filter, with an infinite impulse response (IIR) filter, or in the frequency domain.

The **Fourier transform** is the most important operation in digital signal processing. The transform is used for analysing the frequency content of signals, the frequency response of digital filters, and for performing digital filtering in the frequency domain. A fourier transform may be described as real or complex, continuous or discrete. However, of particular importance to digital signal processing is the form of discrete Fourier transform known as the fast Fourier transform (FFT). All forms of Fourier transform convert a signal into a representation which represents the input signal as the sum of a number of sinusoidal components, which each posess a fixed amplitude and phase. The original signal can be reconstructed by adding together sinusoidal functions with the frequencies and phases derived by the transform. However, the inverse Fourier transform is a more computationally efficient way of reversing the transform. The output of the transform is *not* usually in the form of and amplitude spectrum and a phase spectrum, but as pair of complex spectra. The output of the transform commonly also contains negative frequencies

To perform operations in the **frequency domain**, such as filtering, is to perform them upon the Fourier spectra. Filtering is particularly straightforward in the frequency domain, as the amplitude and phase values may be directly changed and the filtered signals reconstructed using inverse Fourier transformation.

The **frequency response** of a filter is the gain of the filter at a range frequencies.

The **gain** of a system, usually experssed in dB, is the ratio by which the power of an input waveform is increased in the output waveform. It is the opposite of attenuation

A **Gaussian distribution,** also known as normal distribution, is the distribution of (instantaneous) amplitude values found in white noise, which is (consequently) also known as **Gaussian noise.**

The **Hanning window** is frequently used in two contexts, in each case prior to Fourier transformation. First, it is used in periodic signal analysis when the segment of signal being analysed is not an integer multiple of the period of the signal. In this case the spectrum may appear ragged. The Hanning window is a single cycle of a rasied sinusoidal function whose period matches the duration of the signal. The window removes the ragged features. Second, the Hanning window is used in the overlap-and-add technique for digital filtering.

Harmonics are prominent frequency components which are equally spaced in frequency.

A **high-pass filter** is a filter which attenuates low frequencies, but leaves high frequencies relatively unaffected.

An **ideal low-pass filter** is a low-pass filter which has an infinitely steep cut-off. It eliminates all freqeuncies above the cut-off and has no effect on either the amplitude of phase of frequencies below the cut-off. The impulse response of an ideal low-pass filter is a sinc function (sin(x)/x).

An **IIR filter**, or infinite impulse response filter, is a digital filter which has internal registers that contain past output of the filter. Each output sample is the weighted sum of these registers added to the input sample. Input to the filter consequently circulates around the registers indefinitely. Hence the filter has a theoretically infinite impulse response.

The **inverse Fourier transform** reverses the action of the Fourier transform in order to produce a signal in the time domain from phase and amplitude information in the frequency domain. The amplitude and phase information will normally be required in the form of complex spectra and negative frequency values may also be required.

Interpolation is the process of inserting extra samples into a waveform, between those which are already present.

The **Inverse Hilbert transform** is a form of sine-wave synthesis. A sine-wave is generated whose amplitude, for each sample, is the instantaneous amplitude and whose phase, for each sample, is the instantaneous phase.

An **imaginary number** is a number which is some multiple of the square root of minus one. It is represented in the machine by that multiple.

An **impulse**, also known as a **delta-function** is a signal in which only one sample has a non-zero value.

The **impulse response** of a filter (or system or room) is the output of the filter when the input is an impulse or "delta-funtion". The Fourier transform of the impulse response can be used to find the frequency response of the filter. For an FIR filter the impulse response has the same form as the coefficients.

A **Laplace transform** is similar to a Fourier transform, except that the signal is decomposed into ramped and damped sinusoids, which have an exponential envelope.

Layer
A logical grouping of data that are like transparent acetate overlays on a drawing. You can view layers individually or in combination. (LAYER)

A **low-pass filter** is a filter that attenuates high frequencies, but leaves low frequencies relatively unaffected. See also ideal low-pass-filter.

Mechanical looseness produces a strongly distorted signal. The inter harmonics (½, 1/3 etc.) are attributable to the fact that the loose part bounces and thus does get excited every 2nd or 3rd revolution of the shaft.

Model Space is One of the two primary spaces in which AutoCAD objects reside. Typically, a geometric model is placed in a three-dimensional coordinate space called model space. A final layout of specific views and annotations of this model is placed in paper space. See also paper space. (MSPACE)

Negative frequencies are a second set of complex spectra produced by a fourier transform. For a real Fourier transform the real negative frequency values are identical to those of the positive frequencies and the imaginary values are the positive imaginary values multiplied by minus one. These values are clearly somewhat redundant, but for filtering in the frequency domain it may be neccessary to modify both positive and negative spectra in a consistent way before inverse Fourier transformation

A **non-recursive filter** is the same thing as an FIR filter.

The **Nyquist frequency** is half of the sampling frequency. It is the highest frequency which is represented properly at a given sampling frequency.

Object Snap Mode
Methods for selecting commonly needed points on an object while you create or edit an AutoCAD drawing. See also running object snap and object snap override

Oversampling may be used in analogue-to-digital conversion. The waveform is initially sampled at a high sampling frequency and is downsampled before being stored. This technique permits the use of an anti-aliassing filter whose cut-off frequency is well above the Nyquist frequency. The phase distortion introduced by the anti-aliassing filter, which tends to be greatest close to the cut-off frequency is consequently outside the frequency range of the recorded signal.

The **overlap-and-add** method of digital filtering involve taking a series of overlapping Hanning windowed segments of the input waveform (50% overlapping) and filtering each segment separately in the frequency domain. After filtering the segments are recombined by adding the the overlapped sections together. This method permits frequency domain filtering to be performed on continuous signals in real time, and without excessive memory requirements.

Paper Space
One of two primary spaces in which AutoCAD objects reside. Paper space is used for creating a finished layout for printing or plotting, as opposed to doing drafting or design work. You design your paper space viewports using a layout tab. Model space is used for creating the drawing. You design your model using the Model tab. See also model space and viewport. (PSPACE)

The **phase spectrum** of a waveform is the set of phases of the sinusoidal cmoponents into which a fourier transform would analyse it.

Polyline
An AutoCAD object composed of one or more connected line segments or circular arcs treated as a single object. Also called pline. (PLINE, PEDIT)

The **power spectrum** is the square of the ampltiude spectrum

Two signals are said to be in **quadrature phase** when they differ in phase by 90 degrees.

Quantisation is the process of turning a continuous (analogue) variable into one which has discrete numerical values at each point in (for instance) time. In audio applications, a voltage is typically quantised into a 16-bit number, that can represent 65536 different levels of voltage. The numbers are usually proportional to the input voltages, but logarithmic relationships have also been used. It is important to note that quantisation is a separate process from sampling; after quantisation the amplitude of the signal is still represented continuously in time.

A **real Fourier transform** is a Fourier transform of an input signal which is composed only of real numbers, as opposed to complex numbers.

A **Reconstruction filter** filter is an analogue low-pass filter. It smoothes out the squared-off waveform which that emerges from the D/A converter removing all frequencies above the Nyquist frequency. Because the same piece of equipment can be used for both functions, reconstruction filters are often referred to as anti-aliasing filters.

A **Recursive filter** is the same thing as an IIR filter.

The **Rayleigh distribution** is most commonly encountered as the statistical distribution of amplitudes in the amplitude spectum of a Gaussian noise.
The distribution is a Gaussian distribution, multiplied by the values of the amplitudes.

A **sample** of a waveform is a measured value of its amplitude at some instant of time.

Sampling is the process of selecting regularly spaced points in time in which to record the level of a signal. However, the term "sampling" is often used to refer to the whole process of analogue-to-digital conversion.

The **sampling frequency** is the number of times per second that the ampltiude of a waveform is specified in a digital waveform.

The **sampling period** is the reciprocal of the sampling frequency, the interval of time between samples.

The **Slip frequency** (Fs) is the difference between the synchronous speed (Ns) and the rotation frequency of the rotor of the motor.
Ns = (Net frequency in Hz) / (number of pole pairs of the motor)
$Fs = Ns - Rf$
Rf = (RPM of the motor) / 60

A **successive-approximation** digital-to-analogue converter generates an analogue signal from a digital one by adding together voltages which are gated by each bit of the digital signal. Bit two gates a voltage which is twice that gated by bit one, bit three gates a voltage which is twice that gated by bit two, and so on.

Tiled Viewports
A type of display that splits the AutoCAD drawing area into one or more adjacent rectangular viewing areas. See also floating viewports, TILEMODE, and viewport. (VPORTS)

The **time domain** is the normal form of representation for digital signals, as a fluctuating amplitude over time. The term "time domain" is used to distinguish this representation from the frequency domain

Upsampling is the process of increasing the sampling frequency of a signal by an integer factor. There are three stages to the process, interpolation of zeroes, low-pass filtering and scaling. Interpolating zeroes rather than linearly or quadratically interpolated values appears odd to the uninitiated. However, this method *is* the most appropriate. When zeroes are interpolated, the frequency components below the old Nyquist frequency are unchanged by the operation. The added energy in the waveform, caused by the fact that successive values do not follow a smooth curve is all at higher frequencies. Consequently, a digital low-pass filter can remove these added frequencies very effectively leaving a smoothed waveform which is identical in form to the original, but with an increased sampling frequency. The waveform requires scaling up by the integer upsampling factor in order to match the original power.

Viewport
A bounded area that displays some portion of the model space of a drawing. The TILEMODE system variable determines the type of viewport created. 1. When TILEMODE is off (0), viewports are objects that can be moved and resized. (MVIEW) 2. When TILEMODE is on (1), the entire drawing area is divided into nonoverlapping viewports. See also floating viewport, TILEMODE, view, and viewpoint. (VPORTS)

A **window** is a temporal weighting function, which is applied to a signal before some other operation. Different window shapes are used in different contexts. The most commonly used is probably the Hanning window. Others include the Hamming, Blackman and Kaiser windows.

White or **Gaussian noise** is a sound in which successive samples of the signal have random values taken from a Gaussian distribution. The amlpitude spectrum of white noise has random values from a Rayleigh distribution, while the phase spectrum has random values from a rectangular distribution between 0 and 2 pi.

Zero padding is a practice which enables the Fast Fourier transform (FFT) to be used in almost any application which may be performed by the slower discrete Fourier transform. To use an FFT the number of samples in the input waveform must be an integer power of two. Zero-padding increases the number of samples in any input waveform up to the next power of two by adding zero-valued samples to the ends of the waveform. This practice has few undesirable effects in analysing frequency spectra. For filtering in the frequency domain, the power spectrum may be altered correctly, but caution should be used when attempting to alter the phase spectrum.

Zoom
To reduce or increase the apparent magnification of the drawing area.