Glossary

A-Law - a logarithmic companding scheme which is used with PCM. A-Law encoding follows the equation

$$F_A(x) = \frac{A|x|}{1+\ln(A)} \qquad 0 \le |x| \le 1/A$$

$$F_A(x) = \frac{1+\ln(Ax)}{1+\ln(A)} \qquad 1/A \le |x| \le 1$$

A = 87.6

A-Law encoding is primarily used in Europe. Also see μ -Law encoding.

Algorithm - A set of step by step instructions leading to a desired result without requiring an understanding of the operations.

Aliasing - When a continuous time signal is sampled at a frequency f_s, all frequencies in the signal that are greater than $f_s/2$ will produce the same sample points as a some frequency below $f_s/2$. The higher frequency signal is said to be an alias of the lower frequency signal.

Analog signal - a continuous time signal.

Analog to digital converter - see A to D converter.

Anti-Aliasing Filter - A low pass analog filter on the front end of a sampled system which greatly attenuates continuous time frequencies greater than $f_s/2$ where f_s is the sampling rate. These higher frequencies, if not excluded, masquerade as lower frequencies (aliasing) and cause errors in the sampled system. An anti-aliasing filter is sometimes referred to as a "pre-filter".

Anti-Imaging Filter - A low pass analog filter which receives the staircase output from a DSP system and eliminates the "images" of the frequency spectrum of the desired output which are produced by the sampling process. An ideal anti-imaging filter would have a gain of 1 and a cutoff frequency of $f_s/2$. An anti-imaging filter is sometimes referred to as a "post-filter".

A to D - This term refers to an analog to digital converter. An A to D converter changes a continues amplitude signal into one that has discrete amplitudes and is represented by a binary number. Also see *Flash converter, integrating A to D, Sigma delta converter,* and *Successive Approximation converter.*

Attenuation - A reduction in gain value. Generally a gain less than 1. Attenuation is often expressed in decibels.

Audio Frequency Band - That band of frequencies that are audible to the human ear. Generally this band is taken to be from 20Hz to 20,000Hz.

Autocorrelation sequence – The product of the terms of a sequence with a shifted version of the same sequence. For a sequence $x(n)$ the autocorrelation sequence may be defined as

$$Autocorrelation \quad x(n) = \sum_{k=-\infty}^{\infty} x(k)x(k+n)$$

See also *cross correlation sequence.*

Autoregressive difference equation - A difference equation in which the output variable y(k) is a function of only the past values of the output variable. The general form of the equation is given by $y(n)=\sum_{k=1}^{N} b_k y(n-k)$

Autoregressive/Moving average difference equation - A difference equation in which the output variable y(k) is a function of the input variable x(k) , its past values and the past values of the output variable. The general form of the equation is given by

$$y(n)=\sum_{k=0}^{M} a_k x(n-k)+\sum_{k=1}^{N} b_k y(n-k)$$

Band limited - A signal is said to be band limited if it's amplitude is zero (or approximately so) outside a particular band of frequencies. Most continuous time signals are not band limited.

Band pass filter - A filter which passes a range of continuous frequencies and attenuates frequencies above and below this range.

Band reject filter - see band stop filter.

Band stop filter - A filter which attenuates a range of continuous frequencies and has gain for frequencies above and below this range. A band stop filter is sometimes called a *band reject* filter.

Bandwidth - The width in Hz or radians of a particular band of frequencies whose value falls above a prescribed reference value. When the term is applied to a filter (such as bandwidth of a low pass filter) it usually refers to the width of the pass band. The term *half power bandwidth* is sometimes used when the reference values are 3DB below the peak value in a band of frequencies.

Bartlett window - Another name for the triangular window function.

Bessel function - A mathematical function named after the German astronomer Friedrich Wilhelm Bessel. Bessel functions are the particular solution to the second order differential equation given by

$$x^2\frac{d^2y}{dy^2}+x\frac{dy}{dx}+(x^2-n^2)y=0$$

For the special case where n is zero or a positive integer a Bessel function of order n is given by

$$J_n(x)=\sum_{k=0}^{\infty}(-1)^k\,\frac{x^{n+2k}}{2^{2k+1/2}\,k!(n+k)!}$$

In signal processing, Bessel functions can be used to design analog filters.

Bilinear Transform - A mapping function between the s-plane and the z-plane which is based on trapezoidal integration. For this function the jω axis in the s-plane is mapped into the unit circle in the z-plane with the left half s-plane mapping inside the circle. The equation for the bilinear transform is $s\leftarrow\dfrac{2(z-1)}{T(z+1)}$, where T is the sample period. The bilinear transform is often referred to as the BLT.

Bitstream converter - see one-bit converter.

Blackman window - A window function used in the design of FIR windowed filters. The Blackman window function is given by

$$W_B(n) = \begin{cases} .42 + .5Cos\left(\dfrac{2\pi n}{L-1}\right) + .08Cos\left(\dfrac{4\pi n}{L-1}\right) & |n| \le (L-1)/2 \\ 0 & otherwise \end{cases}$$

BLT - see Bilinear transform

Boxcar window - Another name used for a rectangular window in FIR windowed filter design.

Brickwall Filter - A filter which has a very steep transition band. A low pass brickwall filter with a cut off frequency near $f_s/2$ may be used as an anti-aliasing filter on the front end of a DSP system.

Butterworth filter - A filter based on the Butterworth polynomial. The low pass Butterworth digital filter is maximally flat in the pass band, has a relatively slow transition band, and is monotonic in the stop band. All of the zeros are located at the z = -1 point.

Butterworth function - A Butterworth polynomial given by $B_n(s)$ may be defined by the following equation:

$B_n(s) \cdot B_n(-s) = 1 + (-1)^n s^{2n}$ where s is a complex number.

Cauer filter - See elliptic filter.

Center frequency - The middle frequency in a pass or stop band.

Chebyshev filter - A filter based on the Chebyshev polynomial. The low pass Chebyshev digital filter has some ripple in the pass band, has a transition band that is faster than that of the same order Butterworth filter, and has a monotonic stop band. All the zeros of a digital low pass Chebyshev filter are at the z = -1 point.

Chebyshev function - A Chebyshev polynomial is given by $C_n(x) = \cos(n \cdot \cos^{-1}(x))$.

Circular addressing – Many specialized DSP processor chips have a base or mod register which can be loaded with a constant n. Addresses in memory can be calculated mod n. Thus the address becomes circular without the need to reset it.

Circular buffer – A set of storage locations which is addressed with circular or modular addressing. This eliminates the need to keep track of beginning an ending pointers and the overhead that goes with it.

Comb filter – A filter with multiple evenly spaced frequency bands. A comb filter's frequency response plot resembles the teeth of a comb.

Companding - Compression of a signal on the transmission side followed by expansion of the signal on the reception side for the purpose of reducing the dynamic range of a signal without sacrificing much in the way of fidelity. For example, in 8-bit PCM, more bits may be assigned to represent low signal levels than are used to represent high signal levels using a logarithmic mapping. See A-Law and μ-Law coding.

Computational complexity - A measure of the relative complexity of a computer algorithm. There is no standard methodology for measuring computational complexity. Typically, an algorithm's computational complexity is measured by counting the number of multiplications, additions, and/or other time consuming operations that must be performed in the implementation.

Convolution filter – A finite impulse response (FIR) filter.

Cross correlation sequence - The product of the terms of one sequence with the shifted terms of a second sequence. The cross correlation sequence of the two sequences $x(n)$ and $y(n)$ may be given by

$$\text{Cross correlation of x(n) and y(n)} = \sum_{k=-\infty}^{\infty} x(k)y(n+k)$$

Also see *Autocorrelation sequence*

Crossover frequency - the frequency at which two units of the same system have an equal gain response. For example, for a low frequency amplifier cascaded with a high frequency amplifier, the cross over frequency is that frequency where the high end of the low frequency amplifier characteristic matches the low end of the high frequency amplifier characteristic. This can also apply to low pass and high pass filters etc.

Cross talk – refers to interference on a signal line which originates from an adjacent signal line.

Cut-off frequency - The frequency at which the gain of a low pass filter falls below some predetermined point. In most applications the cut-off frequency is taken to be the frequency at which the power gain is one half of the pass band gain. This is $1/\sqrt{2}$ of the voltage gain and marks the point where the gain is 3 db (decibels) down from the pass band gain.

DAC - See D to A Converter.

Decibel - A unit for measuring the loudness of sound. Used in signal processing and communications applications as a measurement of signal power. The decibel is defined by the equation $db = 10\log_{10}(P_2 / P_1)$. Since power is proportional to voltage squared the decibel can be written as $20\log(V_2 / V_1)$.

Decimation filter – A filter which reduces the sample rate of the incoming signal. The output sample rate is usually equal to the input sample rate divided by an integer. By combining the decimation and interpolation process it is possible to reduce the sample rate by a fractional amount.

Delay - The time, measured in seconds, that is required for a signal to pass through a filter.

Delay line filter - this is another name for an FIR filter.

Delta modulation – a modulation technique in which the output is two level. A high level indicates a positive change in the slope of the input and a low level indicates a negative change in the slope of the input. The sampling rate for this type of modulation is dependent on the slope of the incoming signal.

DFT - see Discrete Fourier transform

Difference equation – An equation relating the terms in a sequence and the differences between them. A linear difference equation with constant coefficients of order N gives the relationship of a discrete time output signal y(k) to a discrete time input signal x(k) through an equation of the form

$$\sum_{k=0}^{N} a_k y(n-k) = \sum_{i=0}^{M} b_i x(n-i)$$

Difference equations may be classified as moving average, auto-regressive, or auto-regressive/ moving average.

Differential nonlinearity error - A static error for A to D (or D to A) converters which measures the ability of the A to D to track a ramp function without missing code steps or inserting code steps that should not be there.

Digital Signal Processing (DSP) - The process of operating on electrical signals using digital techniques to encode or extract information.

Digital Signal Processor - A specialized computer whose architecture is optimized for digital filter implementation and the processing of digital signals.

Discrete Fourier Transform – An approximation to the Fourier transform where both the time and the frequency variable are discrete variables. The discrete Fourier transform is given by

$$F(k) = \sum_{n=0}^{N-1} f(n) \cdot e^{-j\frac{2\pi kn}{N}}$$

The discrete Fourier transform is referred to as the DFT.

Discrete Time Fourier Transform – An approximation to the Fourier transform in which only the time variable is discrete. The frequency variable is continuous. Thus, the discrete time Fourier transform is the dual of the Fourier series. The discrete time Fourier transform is given by

$$F(\omega T) = T \cdot \sum_{n=-\infty}^{\infty} f(nT) e^{-j\omega nT}$$

Discrete time signal - a signal which is defined only at discrete instances of time. The amplitude variable for a discrete time signal is usually understood to be continuous in time.

Discrete signal - a signal which is taken to be discrete in both time and in amplitude.

Distortion - generally, the undesirable deviation of a signal from the ideal signal. Distortion can be measured in terms of gain, phase, delay time, or power. For example, the nonlinear phase characteristic of IIR filters is said to introduce a phase distortion.

Dither - The addition of a small amount of noise to a signal for the purpose of bringing a low level signal above a detection threshold where it can be recovered with the noise removed. Thus, dither causes a signal to vary between adjacent quantization levels thereby representing the signal more accurately.

DTFT - see discrete time Fourier transform.

D to A - This term refers to a digital to analog converter. A D to A converter changes a signal that is discrete in time and amplitude to one that approximates a signal that is continuous in time and amplitude

Dynamic range – The dynamic range of a signal x(n) is the difference between its maximum and minimum values. This term is generally seen in reference to the range of A to D and D to A converters.

Echo - The reflection of sound back to the source. A echo usually refers to a sound wave that is reflective back to the source one time. Multiple echoes are referred to as reverberation.

Elliptic filter - A filter based on elliptic integrals of the first kind. These have the form

$$f(\varphi, k) = \int_0^{\varphi} \frac{d\theta}{\sqrt{(1 - k \cdot \sin^2(\theta))}}.$$

The elliptic filter has ripple in both the pass and stop band but has a transition band that is faster than that of the Chebyshev or Butterworth filter.

Envelope delay - This is another name for the group delay.

FFT - (Fast Fourier Transform) - The FFT is an efficient algorithm for computing the discrete Fourier transform (DFT).

FIR - (Finite Impulse Response) - a category of digital filters that do not have feedback. FIR filters are sometimes referred to as delay line filters because of their architecture. FIR filters can be designed to have a linear phase characteristic but are not generally as computationally efficient as their IIR counterparts.

Fixed point - A number system in which all numbers are represented by a fixed number of bits. This is sometimes referred to as scaled integer arithmetic.

Flash converter – An A to D converter in which the signal is compared to multiple levels of a reference signal simultaneously. Combinational logic accepts the comparator output and converts the result to a binary value. For an n-bit converter, $2^n - 1$ comparators are needed. Flash converters are the fastest A to D converters but because of the numerous comparators and the multilevel reference practical flash converters are limited to about 10 bits. These converters find application in video circuitry.

Floating point - A number system in which all numbers are represented three fields of fixed length. The three fields represent the sign of the number, the mantissa, and the exponent. The value of the number is the mantissa times the base (usually 2) raised to the exponent.

Flutter - high frequency noise introduced by irregularities in the recording or play back mechanism. Low frequency noise of this sort is called wow.

Fourier series - A series which equates an infinite sum of sine and cosine functions to a periodic function. The series can be written in trigonometric form as

$$f(t) = \frac{a_0}{2} + \sum_{k=1}^{\infty} a_k \cdot Cos(\omega_0 kt) + \sum_{k=1}^{\infty} b_k \cdot Sin(\omega_0 kt)$$

where

$$a_k = \frac{2}{T} \cdot \int_T f(t) \cdot Cos(\omega_0 kt) dt \quad \text{and} \quad b_k = \frac{2}{T} \cdot \int_T f(t) \cdot Sin(\omega_0 kt) dt$$

or in exponential form as

$$f(t) = \sum_{k=-\infty}^{\infty} C_k e^{jk\omega_0 t} \quad \text{where} \quad C_k = \frac{1}{T} \int_T f(t) e^{-jk\omega_0 t} dt \quad = \quad |C_k| e^{j\theta_k} \quad \text{and} \quad \omega_0 = \frac{2\pi}{T}$$

Note that f(t) is continuous (mostly) in time and periodic. The series is discrete in frequency.

Fourier transform - a mathematical transformation which gives the frequency composition of a non-periodic continuous (mostly) function of time. The Fourier transform is given by

$$F(\omega) = \int_{-\infty}^{\infty} f(t) \cdot e^{-j\omega t}\, dt$$

The Fourier transform can be thought of as the limit as T times the Fourier series where the period, T goes to infinity. Note that for the Fourier transform both the frequency and the time variable are continuous functions.

Frequency sampling filter - A type of FIR filter which is designed from samples of the desired response taken in the frequency domain. This method relies on the DFT and its inverse to create the filter coefficients and is amenable to computer optimization.

Full duplex – A signal link between two systems in which information can flow in both directions simultaneously. If signal transmission is limited to one direction at a time the link is said to be *half duplex.*

Gain - The ratio of the output amplitude to the input amplitude for a system. The term "gain" by itself usually refers to voltage gain but gain is sometimes measured in terms of power gain.

Gain error - For A to D (or D toA) converters the gain error is a static error which measures the deviation of the output from the ideal that is due to a gain in the A to D circuitry. The gain error is usually proportional to the input voltage.

Graphics equalizer - A control device in audio systems which allows independent adjustment of the amplitude of various frequency bands in an audio signal.

Group delay - The group delay is defined as the negative of the derivative of the phase curve for a filter. In equation form it is given by

$$GroupDelay = \tau_g(\omega) = -\frac{d\theta(\omega)}{d\omega}$$

where $\theta(\omega)$ is the phase function. The group delay is a measure of the delay time in passing data through a filter. A constant group delay implies a linear phase curve which is desirable in applications where information is encoded in the phase of the signal.

Half duplex – A signal link between two systems in which information can flow in only one direction at any one time. If signal transmission is bi-directional in time the link is said to be *full duplex.*

Hamming window - A window function used in the design of FIR windowed filters. The Hamming window function is given by

$$W_h(n) = \begin{cases} \alpha + (1-\alpha)\cos(2\pi n/(L-1)) & |n| \le (L-1)/2 \\ 0 & otherwise \end{cases}$$

where L is the filter length and α = .54. This function is identical to the von hann window function except for the value of α

Hanning window - Another name for the von hann window function.

Harmonic distortion – Alteration of a signal by adding or subtracting to its harmonic frequencies. This typically occurs when a circuit introduces nonlinear effects that are frequency dependent. Also see Total Harmonic Distortion.

High pass filter - A filter which attenuates low frequencies and allows high frequencies to pass.

Hum – A low frequency signal that is sinusoidal in nature and is regarded as noise to a signal carrying information. Hum usually originates in a power supply driven by a 60Hz line.

IIR - (Infinite Impulse Response) - A category of digital filters that have feedback allowing a single input pulse to produce a potentially infinite output stream. IIR filters are usually computationally more efficient than their FIR counterparts.

Impedance - The proportionality factor between voltage an current expressed as a complex number. When the impedance becomes a real number it is referred to as resistance.

Impulse - A momentary change in signal amplitude. For discrete systems a unit impulse, $\delta(nT)$ has a value of 1 at nT = 0 an is zero at all other values of nT. For continuous time systems the unit impulse $\delta(t)$ has a value of zero except at t = 0. At t = 0 the amplitude goes to infinity but the time width goes to zero in such a way that the area goes to 1.

Integrating A to D converter – An analog to digital conversion technique in which the analog signal is integrated over time. The digital output is the average of the signal over the time interval. In one type of integrating A to D converter a counter is used to measure the time it takes to charge a capacitor (integration) to a prescribed level. Integrating A to D converters are used in such applications as digital volt meters.

Integral nonlinearity error – A static error parameter for A to D (or D to A) converters that measures the maximum deviation of the output from the ideal over the range of output values.

Jitter - variations in the sample time which add noise to a signal.

Kaiser window - An window function used to design FIR filters which can be adjusted to meet specifications of the desired filter. The Kaiser window function is a based on Bessel functions and is given by

$$W_K(n) = \frac{I_0(\beta)}{I_0(\alpha)}$$

where $I_0(x)$ is a Bessel function and α and β are parameters related to the ripple in the pass and stop bands as well as the filter length.

LaPlace transform – A generalization of the Fourier transform in which the complex number $j\omega$ is replaced by the general complex variable $s = \sigma+j\omega$ where σ is real. The LaPlace transform has the advantage that it exists for many more functions than does the Fourier transform.

Linear phase - The phase is said to be linear if there is a straight line relationship between the phase shift and the frequency.

Linear Phase filter - A class of filters (usually FIR) which exhibit linear phase in the pass band. A linear phase filter provides constant delay to an incoming signal and thus does not distort the shape of the phase curve.

Low Bit Conversion - A digital-to-analog conversion technique in which converts a short portion of a whole digital word to its analog counterpart. Low bit converters operate at frequencies considerably higher than the sample frequency and may be regarded as a serial conversion technique. Low-bit converters generally produce a pulse-width modulated signal. (See also one-bit conversion).

Low pass filter - A filter which passes a band of low frequencies beginning at 0Hz and attenuates a band of high frequencies.

MAC - An acronym for multiply and accumulate. Since difference equations are written as the sum of products many specialized DSP processor chips have a single assembly language instruction which does multiply and accumulate. Hence the term MAC.

Monotonic - continues in one direction without wavering. For example, the stop band of a Butterworth filter is said to be monotonic because it proceeds from a relatively high gain value to zero with a constant negative slope.

Moving average difference equation - A difference equation in which the output variable y(k) is a function only of the input variable x(k) and its past values. The general form of the equation is given by $y(n) = \sum_{k=0}^{M} a_k x(n-k)$.

Multiplexer - Any device which allows multiple signals to share a common channel. For example, many different telephone channels can be assigned different carrier frequencies and share the same fiber optic fiber. This is referred to as frequency multiplexing. Multiplexing can also be done in time or space.

Multi-bit converter - A digital-to-analog converter that operates on an n-bit digital word and converts the whole word (in parallel) to its analog counterpart. Multi-bit converters produce one analog sample each sample period. The ladder converter is typical of this type of traditional D to A converter.

NaN - (Not a Number) - a representation in floating point arithmetic that indicates data that is not a representable number.

Normalized - When used with reference to floating point numbers, normalized means adjusting the exponent and the mantissa such that the most significant bit of the mantissa is 1. This is done to preserve significant bits. When used in reference to a frequency response function, the term normalized means that the gain constant is adjusted so that the gain has unity value at some prescribed frequency (usually 0Hz for lowpass filters).

Nyquist rate - A sampling frequency that is twice as high as the highest frequency contained in the signal being sampled.

Nyquist Theorem - see Sampling Theorem.

Offset Error - For A to D (or D toA) converters the offset error is a static error which measures the dc value by which the output deviates from the ideal.

One-Bit conversion - A form of low-bit conversion for digital to analog conversion which operates on one bit at a time and produces a pulse-density modulated signal. One-bit conversion may be regarded as a serial conversion technique. It produces output bits at frequencies much higher than the sample frequency. One-bit conversion is sometimes called *bitstream conversion.*

Overflow - A condition that occurs (especially in fixed point arithmetic whereby a result can not be represented in the number of bits available in a register. Also see saturation and twos complement overflow.

Over Sampling- sampling faster than the required Nyquist rate. Oversampling also refers to the process of creating more samples than what appears in the original sampled signal. In audio systems this is done to provide a more efficient way to convert the sampled signal to analog.

Overshoot - The amount by which an output exceeds some steady state value when a standard signal such as a unit step is applied to the input. This is usually measured as a percentage.

Pass band - That band of frequencies which a filter passes with a gain above some preset value. The pass band is often taken to be the band of frequencies in which the gain is above the 3 db point or where the ripple stays within the pass band ripple specification.

Pass band ripple - The amount of ripple in the pass band (see *ripple*).

PCM – See Pulse Code Modulation.

Phase shift - A measurement of the time delay in a sinusoidal signal measured in degrees or radians.

Phase - A measurement in degrees for a sinusoid which expresses the difference in time between the sinusoid and some reference point.

Phase delay - The phase delay is formally defined as the negative of the frequency normalized phase curve. In equation form

$$\tau_p = -\frac{\theta(\omega)}{\omega} \equiv PhaseDelay$$

Phase modulation - A modulation technique in which the phase of a sine wave carrier signal is altered to carry the information.

Piecewise linear phase - the phase is said to be piecewise linear if there is a straight line relationship between the phase and the frequency except for a finite number of discontinuities. These discontinuities are the result of zeros on the unit circle in the z-plane.

Pipeline – In computer architecture, a technique whereby an instruction is broken down into a sequence of simpler operations and operations are executed in assembly line fashion. Pipelining allows sequential operations to be executed in parallel.

Post-filter - see anti-imaging filter.

Pre-filter - see anti-aliasing filter.

Pulse - A signal which has a sharp positive change followed momentarily by a sharp negative change.

Pulse Amplitude Modulation (PAM) - an modulation scheme in which the amplitude of a signal at a particular time is represented by the amplitude of a short pulse at the sample time. PAM is a basic type of analog pulse modulation. Note that the pulse height is not discrete but has an analog value. Other basic types include *pulse width modulation* and *pulse position modulation.*

Pulse Code Modulation (PCM) - an modulation/encoding scheme in which the amplitude of a signal at a particular time is represented by a binary number in a pulse sequence. PCM is similar to PAM except that the pulses are discrete and may be encoded. PCM can be derived from PAM by quantizing each samples amplitude and encoding it as a binary number.

Pulse Density Modulation (PDM) - an encoding scheme that is similar to pulse width modulation except that the signal amplitude is represented by the density of pulses within a given sample window instead of the width of a single pulse. The PDM technique typically uses both positive and negative going pulses. Thus, the high time in PWM would be represented by a sequence of positive pulses in PDM and the low time in PWM would be represented by a sequence of negative pulses. This encoding scheme is often called *one-bit conversion*.

Pulse Number Modulation (PNM) - an encoding scheme in which the amplitude of a signal at a particular time is represented by the number of short pulses appearing within a sample time window. All pulses are of uniform height. For high resolution a large number of pulses is needed and this method of encoding is usually unacceptable.

Pulse Position Modulation (PPM) - an encoding scheme in which the amplitude of a signal at a particular time is represented by the position of a short pulse within the sample time window. All pulses have the same height.

Pulse Width Modulation (PWM) - an encoding scheme in which the amplitude of a signal at a particular point in time is represented by the width of a pulse of fixed height. This is also referred to as *Pulse Duration Modulation.*

Quantization - The process of representing a signal with a finite set of values as opposed to an infinite set of values as in typical continuous time signals. Signals may be quantized in time, amplitude, or frequency. If a 10-bit A to D converter is applied to a continuous time signal, the signal is said to be quantized to one part in 2^{10}. Such a signal has only 2^{10} allowable amplitude values.

Quantization Error - Quantization necessarily produces some error since a continuous spectrum of values are not permitted. An 8-bit A to D converter operating on a continuous time signal with an amplitude of 0 to 10 volts would quantize the signal to one part in $2^8 = 10/2^8 = .03906$ volts. Since values between quanta cannot be represented, the maximum quantization error is $.03906/2 = .01953$ volts.

Quantized Signal to Noise Ratio - QSNR. The signal to noise ratio for quantized signals. This term is sometimes used to refer exclusively to quantization error only. See Signal to Noise Ratio.

Ramp function - A signal which rises linearly in time.

Random sequence -A sequence is random if amount of information needed to describe the smallest algorithm used to generate the sequence is longer than the number of bits in the sequence.

Real Time system - a system which is synchronized to events in the outside world. This is typically done by polling, by interrupts, or by software delay loops. The input and output of real time systems is not typically stored in a memory device.

Reed-Solomon - A digital error correction technique commonly used in CD players.

Reverberation - echoes which reach a listener as a result of sound reflection in a closed space such as a concert hall. Thus reverberation refers to multiple echoes.

Reverberation time - Reverberation time is generally defined as the time it takes the reverberation to fall 60 db from it's original value. This is typically a function of frequency.

Ringing - An oscillation in the output of a signal that dies out in time to some steady state value. This is usually in response to some standard input such as a unit step.

Ripple - A measure of the variation in gain within a frequency band of a filter. This variation is often expressed in decibels. Ripple does not necessarily imply that the signal varies in an undulating motion but is an expression of the total variation between a maximum and a minimum.

Rise time - The amount of time it takes a signal to move from 10% to 90% of its final value. This time is usually measured in response to a step input.

Sample-and-Hold - An analog circuit which samples a continuous time signal and holds its sampled value until the next sample period. The sample and hold circuit typically precedes the analog to digital converter so that the converter can act on a fixed value.

Sample Frequency -The frequency, f_s at which samples are taken of a continuous time signal to convert it to a discrete time signal. The sample frequency is often specified in terms of its period T, where $f_s = 1/T$. The sample frequency for discrete time systems is typically measured in Hertz (Hz) but may be given in radians per second and designated ω_s.

Sampling Period - The inverse of the sample frequency. The symbol for the sampling period is T and T is measured in seconds.

Sampling Theorem - A theorem which gives the minimum sampling rate at which a continuous time signal must be sampled in order that it's information content not be lost. The theorem is stated as: *A signal which has no frequencies higher than f can be reconstructed from samples taken at a rate of 2f or greater without loss of information.* The Sampling Theorem is sometimes referred to as Nyquist's Theorem (after Harry Nyquist) or Shannon's Theorem (after Claude Shannon).

Saturation overflow - A system of managing overflow in which numbers whose magnitude cannot be represented because they require more bits than are available are set equal to some predetermined value - usually the maximum value allowed.

Settling time - The amount of time it takes a signal to settle to within some prescribed percentage of a steady state value.

Shannon's Theorem - See Sampling Theorem.

Sigma delta converter - A conversion process in which a summer (Sigma) and a delta modulator are connected in a feedback loop. The delta modulator produces +1 or –1 in response to the changes in the slope of the signal. The summer accumulates these differences to produce a binary value proportional to the original signal. See also *Delta modulation*.

Signal-to-Error Ratio - The ratio of the squared rms signal value to the squared rms error value. For quantized signals each bit added reduces the signal to error ratio by approximately 6db.

Signal-to-Noise Ratio - The ratio of the signal power to the noise power measured in decibels.

s plane - If s is taken to be a complex number $\sigma + j\omega$, the s plane is a two dimensional plane in which σ is mapped on the horizontal axis and ω is mapped on the vertical axis. The s plane is generally used to map poles and zeros of transfer functions in s.

Step function - A standard signal which has a value of 0 for all time less than 0 and has a fixed value for all time greater than zero. If the fixed value is unity, this function is referred to at the unit step function.

Stop band - A band of frequencies in which the gain is attenuated.

Successive Approximation converter - An A to D conversion technique in which a digital approximation to the incoming analog signal is converted to an analog signal and compared to the original. The result of the comparison is used to adjust the original digital approximation. In a scheme similar to a binary search, an n-bit conversion can be produced in n steps. Successive approximation converters are relatively simple to construct and produce a moderately fast conversion. Eight to 14-bit conversions are practical.

Total Harmonic Distortion – The rms sum of the individual harmonic distortions that are introduced by frequency dependent nonlinearities in a circuit. The Total Harmonic Distortion (THD) is often used as a numeric expression of the distortion present in the output of a circuit.

Transient response - For a stable filter the transient response is that response which dies out in time. This response is usually taken to be the response to an impulse or a step function.

Triangular window - a window function whose time domain shape is that of a triangle. The triangular window typically has a gain of 1 at time 0. This window function is often referred to as a Bartlett window function. The triangular window function may be described by the following equation:

$$W_T(n) = \begin{cases} \dfrac{-2|n|}{L-1} + 1 & |n| \le (L-1)/2 \\ 0 & otherwise \end{cases}$$

Twos complement - A number system in which negative numbers are obtained by taking the positive number, inverting it, and adding one. For this number system, the most significant bit is the sign bit with 0 indicating positive numbers and 1 indicating negative numbers.

Twos complement overflow - In the twos complement number system, for addition and subtraction, numbers which are too large (in magnitude) wrap around and get represented by smaller numbers.

μ **-Law -** a logarithmic companding scheme which is used with PCM. μ -Law encoding follows the equation

$$F_\mu(x) = \frac{\ln(1+\mu|x|)}{\ln(1+\mu)}$$

$\mu = 255$

μ - Law encoding is primarily used in North America and Japan. It is sometimes referred to as mu-255. Also see A-Law encoding.

Voice band - The usable bandwidth of a voice signal. For a telephone system this is often taken to be 300Hx to 3,400Hz.

Von hann window - A window function used in the design of FIR windowed filters. The von hann window function is given by

$$W_h(n) = \begin{cases} \alpha + (1-\alpha)\cos(2\pi n/(L-1)) & |n| \le (L-1)/2 \\ 0 & otherwise \end{cases}$$

where L is the filter length and α = .50. This window function is identical to the hamming window function except for the value of α.

White Noise - Noise which is equally probable over all frequencies. For sampled signals, white noise has a frequency spectrum which is flat over the band $-f_s/2$ to $f_s/2$.

Window function - a function defined in the time domain that is used to produce a "windowed" FIR filter. The purpose of the window function is to provide a smooth truncation of the impulse response derived from an idealized filter. Thus the impulse response of a windowed FIR filter is the product of the window function and the idealized impulse response function.

Wow - Low frequency noise produced by irregularities in the recording or play back mechanism of a sound signal. Wow can be caused by irregularities in tape drive speed or as the result of worn parts. The term flutter is used to high frequency noise of this sort.

z plane - If z is a complex number, the z plane is a two dimensional plane with the real values of z on the horizontal axis and the imaginary values of z on the vertical axis. This is generally used to map poles and zeros of transfer functions in z.

z-transform – A generalization of the discrete time Fourier transform in which the complex number $e^{j\omega}$ is replaced by the general complex variable z which has both a real and an imaginary part and whose magnitude is not necessarily equal to one. The z-transform has the advantage that it exists for many more functions than does the discrete time Fourier transform.