# Analogue Signals

M J Brockway

February 5, 2018

# Digital vs Analogue

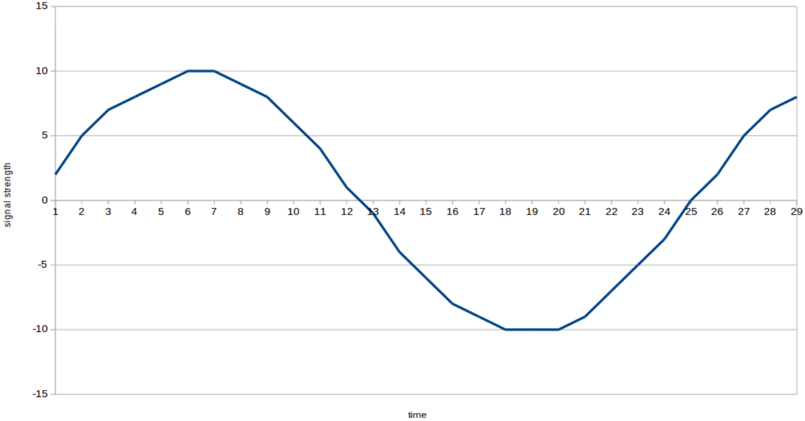
Digital (electrical) inputs to a CPU register as logical values - true or false, 1 or 0, on or off.

- typically arise from switch contacts opening or closing.
- eg button press or release

A digital output is used to switch something on or off.
Analogue signals vary smoothly over a range of values and may momentarily take any value in between.

- Eg an audio signal from a microphone;
- a voltage from a sensor - temperature, light level, acceleration, force (strain gauge), humidity, pressure, ...

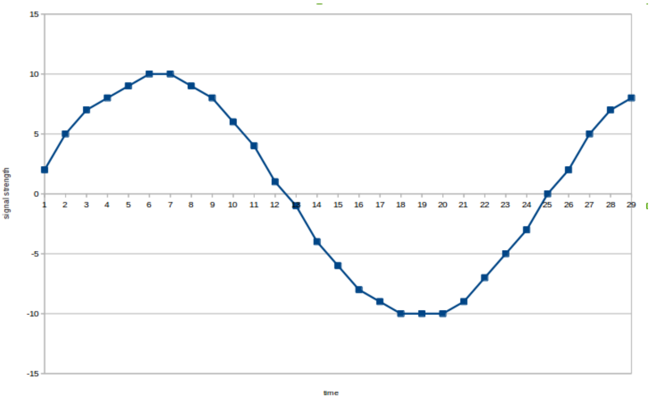
# Analogue signal




To input an an analogue signal it has to be *digitised*

- *sampled* at a regularly spaced series of times;
- to produce a series of numbers.

# Analogue signal




- An *analogue-to-digital converter* (ADC) is a hardware device that does this.
- It is configured with a *sampling rate* and an *output resolution*.
- Output resolution is the number of bits available to store a sample value. With 12 bits you can store values in range 0 to $2^{12} - 1 = 4095$.

# Analogue/digital conversion

The opposite process is *digital-to-analogue conversion* (DAC).

- From a series of digial values, creates a time-varying analogue signal.
- An analogue output may drive an audio speaker or a dimmable lamp. (Theatre lights 'synthesize' colours by combining red, green, blue with variable brightnesses.)

Digital audio

- takes analogue input from a microphone (or a mixed audio signal from several),
- passes it through an ADC;
- resulting stream of bits is saved raw (WAV) or in compressed format (FLAC, OGG, MP3).
- To reproduce the sound, the bit stream is read from the save medium,
- and passed to a DAC to recreate the audio signal,
- which (after amplification etc) drived a speaker.

# ADC techniques

A *successive approximation converter* first

- compares the input with a voltage which is half the input range;
- if input $>$ this level, it compares it with $\frac{3}{4}$ the range;
- and so on: twelve steps => 12-bit resolution.
- During the comparisons, signal is frozen in a *sample and hold circuit*.

A *dual-slope integrating converter*

- lets the input signal charge a capacitor for a fixed period;
- then measures the time for the capacitor to fully discharge at a fixed rate;
- this time is proportional to the 'integrated' (averaged over the sample period) input voltage.
- Slower than successive approximation, but reduces the effects of electrical 'noise'.

There are other types of ADC which refine these ideas.

# ADC techniques

The *resolution* of an ADC is

- $n$ *bits*, where the input range is divided into $2^n$ *steps*.
- Eg a 12-bit ADC will have $2^{12} = 4096$ steps;
- A 0-10 volt input range will then resolve into 2.5 mV steps.

*Linearity* of an ADC ...

- Ideally, with $n$-bits resultion you get $2^n$ steps *of equal size*.
- In practice, the sizes of the steps vary a little – non-linearity.
- *Maximum linearity error* of $n$ percent means the steps vary in size no more than $n$% from the ideal step size, $2^{-n}$ of the range.

A *sample-and-hold* circuit ...

- freezes the analogue input voltage at the moment the sample is required,
- holds it constant while the ADC digitises it.

# ADC techniques

*Thoughput* ...

- The *acquisition time* is the time for the ADC to capture the input voltage during a read; the *conversion time* is time to determine from this the output value (eg by timing a capacitor discharge).
- *Throughput* = 1/(acquisition time + conversion time).
- A *pipelined* ADC improves thoughput.

An *integrating* ADC, such as the dual-slope ADC

- times the charge or discharge of a capacity to get an average of the voltage over the sampling cycle.
- The time to do this is the *integration time*. Convdersion time of a dual-slope converter is a function of this.

# Digital to analogue conversion - DAC

This is an electronic circuit which accepts at regular intervals a (digital) *number* at its input and produces a corresponding analogue signal, usually a voltage at its output.

- Over time, a series of analogue signals are output.

These might be voltage or current *control signals* ...

- Frequency (number of output per second) is low;
- Outputs determine a motor speed or light intensity or current in a heater or ...

They might be to generate a *waveform* ...

- an audio or video signal for example;
- frequency can be hundreds to millions of times per second.

# DAC applications

- digital audio, video;
- high-end instrumentation: waveform genertors, medical imaging;
- wireless communication systems: mobile phones, satellite terminals, point-to-point and multi-point communication links.
- radar systems

# DAC output range

- the maximum and minimum voltage or current that can be generated:

- bipolar - eg -5 V to +5 V; or
- unipolar - eg 0 to 20V.
- There is often a choice of ranges; choose smallest that will do the job.

# DAC resolution

- the number of steps into which the output range has been divided.

- *n*-bit resolution => $2^n - 1$ steps ($2^n$ values).
- For instance DAC with 12-bit resolution divides its output range into $2^{12} = 4096$ steps.
- If the output range is 0-10 V, it is resolved to 2.5 mV steps.
- Thus, output is not truly analogue: it is stepped!

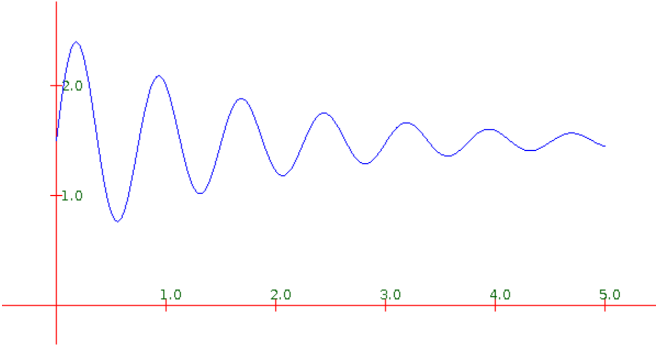
# DAC slew rate and settling time

Slew Rate

- the maximum rate of change of the output signal:
- measured by the rise in voltage divided by time
- Eg volts per microsecond.

# DAC settling time

- When the DAC changes from its minimum output level to its maximum, the output signal swings through its *full scale*.
- The settling time indicates how long it will take the output to settle to its final voltage
- time to settle to a percentage of the full-scale voltage or current range, following a full-scale change.
- actual output wobbles about for a few microseconds before setting ...



# The Nyquist criterion

How do you decide the sampling rate of an ADC?

- You want to know that you will get an accurate copy of the signal when you feed the data to a DAC!
- The *Nyquist criterion* says: sample at *twice the bandwidth* of the original signal: twice the highest frequency present in the original signal.
- This guarantees you enough data to rebuild a fair copy of the signal with a DAC, provided ...
- you feed the rebuilt signal through a *filter* - an electronic circuit which reject frequencies outside the band you are interested in.
- This is based on *Fourier theory*, a mathematical theory that shows how any waveform with a maximum frquency $f$ can be built of 'sine waves' of frequencies up to $f$. An ADC datum for each half-cycle at the maximum frequency will do the trick, according to Nyquist.

# The Nyquist criterion - examples

Use a sampling rate of $2.2 \times f$ max to allow for practical filters.
Landline telephony supports audio for speech conversation in the range 300 to 3400 Hz.

- sampled at 8 kHz

'CD quality' audio is based on the idea that we hear sounds up to 20 kHz.

- CD quality sampling rate is 44.1 kHz
- CD is recorded in stereo and each channel uses a 16-bit ADC....
- Combined ADC output is 1411200 bits/sec: 10.582 Mb/min.
- a nominally 700 Mb compact disk will support around 66 minutes of playing time.

# The Nyquist criterion - aliasing

If sampling is at a *lower* frequency that demanded by the Nyquist criterion, i.e. at less than twice the maximum frequency in the input waveform, then

- the sum and difference components associated with each *harmonic* of the input waveform overlap with those of adjacent harmonics and
- the sampled waveform can no longer be separated out by filtering.

This is a bit technical (mathematical Fourier theory again) but the effect is that the waveform reconstituted by the DAC (in your CD player for instance) will not make a faithful copy of the original waveform.

A slightly mathematical discussion of the Nyquist criterion is to be found at https://en.wikipedia.org/wiki/Nyquist_ISI_criterion and in the same spirit, the article on aliasing is also worth a read: https://en.wikipedia.org/wiki/Aliasing.