# Flat Out: The World of Digital Signal Processing

Why do we use it? W.G.Marshall

## The Real world is Analogue!

So why is every machine or instrument around us in the home or at work increasingly described as 'digital'? To answer this question we had better establish what we mean by the word: 'Analogue'.

All measurable things in life vary continuously in amplitude (size) with time: the outside temperature, the speed of a car or even daylight. We can convert a varying temperature to a varying electrical voltage using a *transducer*. We now have an electrical *analogue* of the original effect. As the temperature varies so does the voltage. Now we have this *analogue signal*, we can process it using other electronic components and display a temperature reading on a simple pointer instrument.

The thing to remember is that all natural parameters vary continuously, not in discrete steps; even devices that appear to operate in a discrete or digital manner can be deceptive.

**Digital or Analogue signal?**

When you switch on a filament lamp, it may look as if the light comes to a steady brightness instantly, but in fact it takes time to build up and the bulb current varies wildly before it settles down. It happens too fast for you to see.

A human heart-beat signal has a distinctive ‘shape’: it is not just a pulse.

## Measuring the World

If we want to measure a length we could use a ruler or tape measure. But what if we want a machine to take the measurement? Perhaps we would like the measurement displayed in different units according to a switch position, or even have the machine use the information itself and perform some appropriate action.

We need an electrical or electronic system that converts whatever we are trying to measure to an electrical signal, maybe performing some *signal processing* before *displaying* or *outputting* the result.

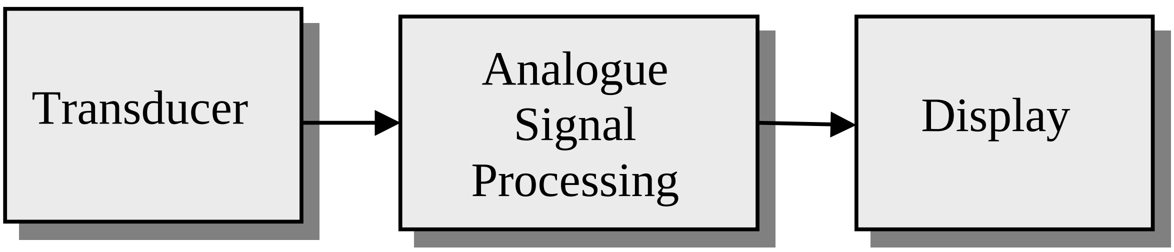


**Fig. 1 A simple electronic analogue signal processing system**

So in summary, the component parts we need are (see Fig.1):

1. Transducer(s), to convert the measured parameter to an electrical signal,
2. A signal processing system, Analogue or Digital,
3. Output Devices, to provide a Visual/Audio Interface to humans,

and perhaps

4. Feedback for ‘Closed-loop’ or Automatic Control.

## Simple Analogue Systems

The *Single-Unit Measure-Display* System requires no processing. Examples of these are available in any domestic situation. Some are listed in the box opposite. Notice that they are all ‘Analogue’ Systems, converting a parameter directly to a visible display.

**Single Unit Measure-Display**

Mercury thermometer
Mercury Barometer
Moving-coil current meter

The *Two-Part Measure-Display System* usually requires little or no processing. Once again you should be able to spot these in everyday use. The examples shown can be found in almost any car, although more sophisticated systems employing digital techniques are now found in many up-market vehicles. In both cases we have a separate transducer converting the parameter to an electrical signal, in these cases current, and a display instrument. The latter uses a hot-wire method to convert the signal to a pointer position.

**Two-Part Measure-Display**

Car coolant temperature sender and gauge
Car fuel level sender and gauge.

*Measure-Process-Display Systems* are more complex electronically than the examples above because some sort of signal conditioning circuits are included between the transducer and the display/output device. Traditionally these would be 'analogue' circuits composed of transistors, resistors, capacitors and more recently integrated circuits or 'chips'. Note that not all chips are digital. A typical piece of 'signal processing' might be to remove high-frequency electrical *noise* from a nearby electric motor. The circuit would probably be a *Low-Pass Filter* in this case.

**Measure-Process-Display**

A Hi-fi system using vinyl LPs.
The magnetic cartridge converts the stylus vibration to an electrical current. Input circuits of the amplifier boost the signal and perform frequency equalization A power amplifier drives audio transducers which convert the electrical signals to sound (Loudspeakers).

A very common example of this type of system is described in the box above: make sure you can recognize the input transducer, signal processing elements and the output device.

## What's wrong with tradition?

The systems discussed above a described as 'traditional' because the represent an era of measurement and electrical/electronic techniques that goes back centuries. Even the car instruments were devised over 50 years ago, and have changed little. They have the advantage of being reliable and cheap to make. Analogue signal processing was kept to a minimum because electronic components were expensive, unreliable and required skilled design engineers to make it work. Let's look at this in more detail.

*Component tolerances* are a major headache for the analogue hardware designer. Very specific values of resistors or capacitors might be needed to realize a particular specification, but only certain *preferred* values are manufactured. This might mean resorting to variable components at greatly increased cost plus the need for setting up adjustments after production.

*Component ageing* is less of a problem nowadays thanks to new materials, but it can still be significant. For example, a resistor might have had a certain resistance value when it left the factory, but years later it may have changed enough to take the circuit outside its original specification or even to cause complete failure.

*Electrical noise* or *Interference* induced in the analogue circuit can sometimes be removed by additional circuitry if it can be distinguished from the wanted signal. More often than not, the electronics cannot tell the difference between noise and signal. Consider the old hi-fi system: it is impossible to remove the needle 'scratch', turntable bearing 'rumble', clicks, pops and hiss without removing chunks of the music as well. Your brain can sort it all out, but even the most sophisticated analogue processing system cannot. The best you can hope for is to reduce the noise to an acceptable level.

*Complex hardware design* is needed for even simple processing tasks. Even if all you want to do is implement a low-pass filter, that is, remove all frequency components above a certain value from the signal, you will find it no easy task. Given a precise performance specification there are a large number possible techniques, each of which has an even larger number of possible circuit implementations. The tolerance problems come in to play, and if that wasn't enough, the layout and design of the printed circuit board (PCB) it's all built on may add 'stray' capacitance effects leading to instability in a high-frequency design. Design compromises are inevitable.

*Difficulties in debugging, modifying,* or *updating* an analogue hardware design cause the product to be expensive at the outset, with much wastage of effort later. Mistakes in the circuit design lead to the physical replacement of components and remaking of PCBs. Updates later on will often involve similar physical changes, so much so that usually it is not worth bothering and the whole system is designed again from scratch.

## The Digital Cavalry to the rescue…

By now you could be forgiven for thinking that designing and making any new electronic system is fraught with such difficulty that it's a miracle 'high-tech' products are manufactured at all. Fortunately salvation is at hand with the invention of the computer and *Discrete-Time* or *Digital Signal Processing*. In the 1920's work by a telegraph engineer called Nyquist formed the basis of what we now call digital signal processing, although even he based his ideas on much earlier work by others. In order to realize the benefits of DSP we must move from the *Continuous-Time* processing that we have been using up to now, to *Discrete-Time* processing.

What is meant by 'Discrete-Time'? Nyquist and others were able to show mathematically that you could work on *samples* of a signal taken at regular intervals and still get a satisfactory output. It seems bizarre, but it's true: you can sample a continuous waveform or signal, and then reconstruct the original continuous signal *exactly* from those samples. It gets better. The rule that governs this sampling, known as Nyquist's Sampling Theorem, is very simple but without it there would be no digital signal processing. So for example, if you have an audio signal with a maximum frequency limit of 15000 Hz, then you will need to sample it at 30000 samples/sec or more.

**Nyquist's Sampling Theorem**

If the sampling rate is at least twice the highest frequency component present in the analogue signal, then the original signal can be reproduced exactly.

All we need is a device called an *Analogue-to-Digital Converter* (ADC) which takes a 'snapshot' of the signal voltage at regular intervals, converts this voltage level to a digital binary number and passes it to a digital computer which of course just loves binary numbers….

A simpler device known as a *Digital-to-Analogue Converter* (DAC) takes numbers from the computer and turns them back into voltage samples. The implications of all this are enormous….

*Component tolerances and other hardware design problems almost disappear*. This is because digital signals are generally binary in form. That is, they work with only two voltage levels corresponding to the two binary number states of 0 and 1. For example, as long as the digital signal is less than about 0.4 volts a binary 0 will be recognised. If the voltage is between 4 and 5 volts a binary 1 is assumed. It is usually not difficult to ensure that induced electrical interference is kept well below the threshold for a 0.

*Ageing is now only a very long-term problem*, and it too has a digital characteristic: it either works or it doesn't.

*Noise problems are reduced*. If the transducer produces a 'clean' signal, and any electrical noise in the processing circuit is kept below the logic threshold mentioned above, then the output signal can be just as free from noise as when it went in. That is why audio CDs have none of the hiss and scratch of analogue tape and vinyl LPs.

*Hardware exchanged for software*. In other words, (almost) all the signal processing is performed by the computer (usually a DSP chip) crunching the numerical samples according to a *program* implementing a mathematical *algorithm*.
This makes debugging and updating potentially much simpler and cheaper, because DSP systems are far easier to reprogram than analogue ones are to rebuild. Of course you need a very fast computer for this *real-time* processing; hence the DSP chip.