
Chapter 11

Oversampling, noise shaping, and digital filtering

11.1 The CD player as a digital signal processing system

The stream of bits recovered from the disc is processed through a series of stages which reverse the encoding process which occurred when the signals were recorded. Minor errors can be completely corrected using the eight-to-fourteen redundancy and parity checking built into the system. Major errors may result in the unavoidable loss of information, but most CD players then use a pre-programmed algorithm to ‘fill in’ or *interpolate* occasional lost samples. The details of this algorithm for masking information loss will differ from one player to another. The recovered stream of digital values can then be passed to two digital to analog convertors (DACs) for conversion into an output pair of analog audio signals.

In principle, we could simply use the CD player to recover 44,100 pairs of digital samples per second and employ a pair of 16-bit DACs to obtain analog signals. Whilst this approach would have the advantage of simplicity it may produce an output which exhibits the ‘staircase’ distortions mentioned earlier.

Provided the input signal was dithered before sampling, any staircase distortions can — in theory — be removed by passing the output from the digital to analog convertors through low-pass filters which reject frequencies above half the sampling frequency. This is because, in an ideal system, all the unwanted frequencies produced by the staircase effect will be above 22.05 kHz. Some of the earliest CD players did employ this approach, but it soon proved unsatisfactory for a variety of reasons and has largely been superseded by better methods. Generally speaking, simply using analog filters to ‘clean up’ the output waveforms works poorly for two reasons:

Firstly, the CD player (or the information on the disc!) may be imperfect. For example, any production problems in manufacturing the digital to analog convertors will alter the form of the staircase distortion and may

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produce unwanted components inside the analog signal's frequency range.

Secondly, in order to realise the full potential of the CD encoding system we would require low-pass filters which perform amazingly well. Ideally, they should pass any signal frequencies up to almost 22·05 kHz without altering them in any way, but must reject any distortion components above 22·05 kHz by at least 95 dB to prevent them from degrading the potential dynamic range. Analog filters capable of simultaneously meeting both these requirements can be made. However, they are difficult to produce as they must contain a large number of very accurately tolerated components. This makes them large and expensive. It is also inevitable that the values of some components will tend to change with age, temperature, or humidity. This would mean a very expensive CD player whose performance might deteriorate audibly with use.

To avoid these problems, almost all modern CD players process the digital data in some way before presenting it to the convertors. The main objects of this processing are:

- To perform a computation equivalent to low-pass filtering. This is intended to reduce the severity of the staircase distortions, easing the demands imposed upon any analog filters placed after the convertors.
- To help prevent any imperfections in the digital circuits, especially the digital to analog convertors, from producing other signal distortions.

The details of this digital processing vary considerably from one type of player to another. (And, of course, every manufacturer claims to use the 'best' method for their newest models!) Fortunately, all of these processes are aimed at achieving the same end result so we need only consider one example. Here we will look at the original system employed by Philips in their 'first generation' CD players using the SAA7030 and TDA1540 integrated circuits. The following explanation has been simplified to some extent, to make it easier to follow, but contains the essential features of the process.

This system employed a combination of two techniques, *Oversampling*, and *Noise Shaping* to achieve the desired results. Oversampling means that a set of sampled values is used to calculate the values we 'would have obtained' at intermediate moments if the original input had actually been sampled more frequently. Provided the sample values we start with satisfy the

sampling theorem these extra values don't contain any new information. This is because there is only one possible waveshape which can fit the sampled values read from the CD. The first Philips CD players employed $\times 4$ oversampling, converting an input data stream of 44,100 samples/second (per channel) into 176,400 samples/second. We can regard staircase distortion as being an unwanted high-frequency variation which has been added onto the signal we wish to communicate via CD. By $\times 4$ oversampling we produce the effect shown in figure 11.1.

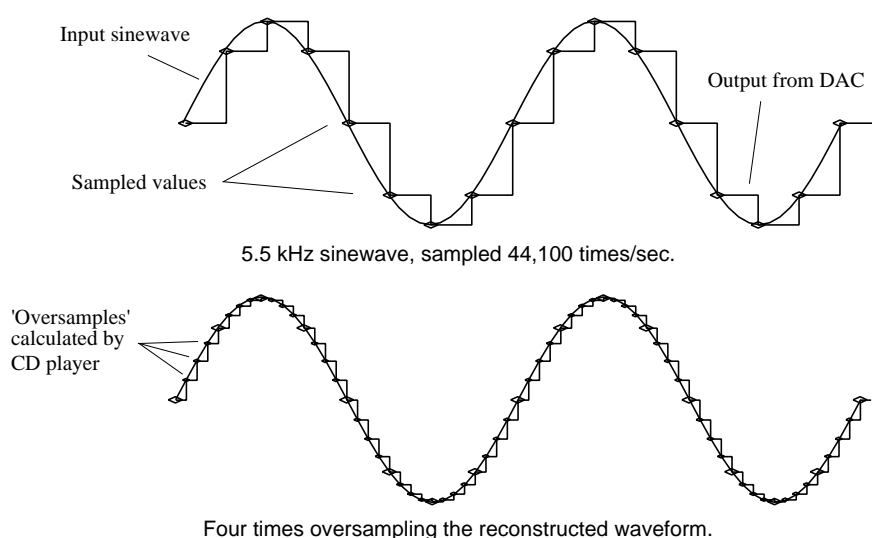


Figure 11.1 Effect of $\times 4$ oversampling.

For the sake of the illustration we can consider an input signal in the form of a 5.5 kHz sinewave which is sampled by the CD recording process at 44,100 samples/second. The simplest way to convert these sampled values back into an analog waveform would be to use a digital to analog convertor (DAC) which produces an output level appropriate for each sample and then 'holds' this level until it is time to output the next sampled level. This kind of output is called *Sample and Hold* and produces the kind of staircase distortion shown.

The sampling theorem says that, provided a series of samples form a complete record of the original information, we can use the measured sample values to calculate the actual signal level at any moment in between the sampled instants. These calculated values can then be given to the DAC in between the 'genuine' samples to produce the

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improvement showed in the lower waveform of figure 11.1. It is important to realise that these calculated samples are not 'guesses', but really do represent the signal level which would have been observed if the input waveform had been measured at these moments. If the CD recording process had actually recorded these extra values on the disc the player would not be provided with any extra information since the original set already contain a complete record of the waveform information. Hence the term '*oversamples*', which indicates that these extra values — calculated or measured — don't actually contain any fresh information.

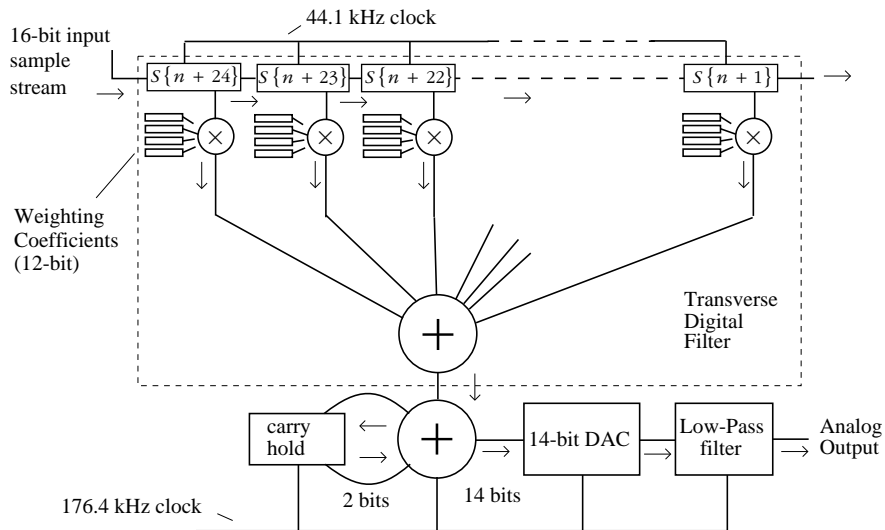


Figure 11.2 Schematic diagram of Philips SAA7030 + TDA1540.

The use of $\times 4$ oversampled digital values reduces the staircase effect in two ways. The basic frequency of the unwanted staircase distortion is increased by a factor of 4, and its amplitude is reduced by a factor of 4. As a consequence it becomes much easier to produce analog filters which, placed after the DACs, will suppress this distortion without significantly affecting the wanted signal. Figure 11.2 shows a schematic diagram of (one stereo channel of) the initial Philips processing system. This used two integrated circuits (ICs), the SAA7030 and TDA1540. The 16-bit samples read from the disc are clocked through a serial *Shift Register* which, in this case, can hold 24 successive sample values. The rate at which the system processes the data samples is determined by two *Clock Frequencies*, 44.1 kHz and 176.4 kHz, which are supplied to the ICs. These

two clock signals are *Phase Locked* so that every fourth cycle of the 176.4 kHz starts at the beginning of each 44.1 kHz cycle. In figure 11.2, $S\{n + 24\}$ represents the ‘newest’ sample value and $S\{n + 1\}$ represents the ‘oldest’. (It’s assumed that n samples have already passed through the system and have been discarded.)

The 44.1 kHz clock signal is also used to control the rate at which digital samples are recovered from the CD, hence samples should be presented to the input end of the shift register at the same rate they are read from the disc. At the beginning of each 44.1 kHz clock cycle all the sample values stored in the register locations are shifted along one place. A new sample is entered into the first register location and the ‘oldest’ sample value is thrown away from the last register location. The registers are linked to an array of multiplier circuits. Each of these has a set of four coefficient values connected to it. These coefficient values are usually built into the processing IC when it is manufactured, although some modern CD systems allow the coefficients to be modified by replacing or reprogramming a ROM (memory) chip.

The 176.4 kHz clock controls the data processing carried out by the circuit. Each *Processing Cycle* takes one 44.1 kHz clock period — i.e. four 176.4 kHz clock periods. To see how the system operates we can examine what happens during each of the four 176.4 kHz clock periods of a processing cycle.

At the start of the first period all the samples are shifted along and a fresh sample value is entered at the ‘newest’ end of the line of registers. This event triggers the start of the processing cycle. Immediately after the sampled values have been updated they are all multiplied by the first set of coefficients and the results are added together to produce an output value which is sent forward to the digital/analog convertor system.

During the second 176.4 kHz clock period the register values are multiplied by the second set of coefficients, and the results added, to produce a new output value for the convertor. The third set of coefficients are used during the third 176.4 kHz clock period, and the last set during the fourth and final period of the processing cycle. As a result, each processing cycle produces four distinct output values which are sent to the convertor. These have all been obtained from the same 24 input samples, but used four distinct sets of coefficient values. During the next processing cycle the input data is shifted along, a new sample is injected, and the process is repeated to produce four more output values.

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In Chapter 7 we saw how it is possible to recover the signal value at instants in between samples. Here the action of the IC may be seen as carrying out a similar task. We could therefore use a set of coefficients which correspond to the values of the sinc function indicated by the expressions in Chapter 7. This would serve to ‘smooth out’ some of the output signal distortion effects. In practice, however, it can be an advantage to slightly alter the coefficient values to obtain a flatter frequency response, lower distortion, or whatever we require.

The circuit which carries out this process (an SAA7030 in this case) is called a *Transverse Digital Filter* (TDF). By choosing an appropriate set of $4 \times 24 = 96$ coefficients we can carry out a series of computations which mimics the effect of a ‘96'th order’ analog filter. The frequency response of this filter depends upon the values chosen for the coefficients. In theory we could build an analog filter, using capacitors, inductors, etc, to achieve the same end. This is because, in principle, identical results can be achieved by either analog or digital processing of the information. In reality, of course, the analog equivalent would prove far more difficult to make, and its properties would be relatively unstable. There are, therefore, good practical reasons for carrying out this filtering process in the digital domain.

One interesting consequence of using a set of samples to compute output values is that the results have more bits per value than the input samples! The SAA7030 stores its internal coefficients as 12-bit numbers and the output values obtained from the TDF therefore emerge as $16 + 12 = 28$ -bit numbers. Note that these additional bits don't contain any ‘new’ information. They are a consequence of the way information is ‘redistributed’ by the TDF process. In effect, each bit of real data influences more than one bit of the oversampled results. The oversampled bits aren't all ‘independent’ of one another.

In an ideal world we might choose to employ a pair of 28-bit DACs after the filter. Alas, at the time the first CD players were launched Philips were doubtful that they could mass produce even 16-bit convertors of the required precision at a commercial price! They could, however, make good 14-bit convertors able to run at a clock speed of 176.4 kHz. They therefore decided to use 14-bit DACs in the first generation of CD players. At first glance it seems that the use of a 14-bit convertor will unavoidably cause some audio information to be lost. Fortunately, it is possible to process the data before conversion in a way which can prevent any information loss by using a *noise shaping* technique. As with oversampling

this process may be carried out in various ways. The original Philips design employed a method which we can understand by looking at figure 11.2.

The 28-bit output values from the TDF are passed through an adder into another register. The most significant 14-bits are then sent on to the DAC. The unused, least significant, bits are treated as a 'remainder' which is held for one 176.4 kHz clock period and then returned to the adder to be combined with the next value. This process is repeated with each successive value. This has the effect of 'carrying forward' any error between the converted and presented values. To see how this works we can forget, for a while, the potential ability of the TDF to generate 28-bit numbers and consider what would happen if we just present a series of 16-bit values to the noise shaping system. For simplicity, imagine that four successive values are the same and let F represent the most significant 13 bits of their value. (We'll also assume that we start with a carry of zero from the last cycle.) The 'carry forward' process continued over four clock cycles then looks like:

16-bit input	+Carry	14-bits to DAC	Remainder
			00
F001	F001	F0	01
F001	F010	F0	10
F001	F011	F0	11
F001	F100	F1	00

Note that if we add together the four successive 14-bit output values sent to the DAC we obtain F001 once again. A low-pass filter placed after the digital to analog convertor will have the effect of suppressing any short-term fluctuations in the output level. If this filter attenuates frequencies above half the basic sampling rate (i.e. 1/8th the oversampled rate) it will tend to produce an output which is much the same as if we had averaged the four values, producing an output equivalent to that which would have been produced by a 16-bit convertor.

It is perhaps unfortunate that this process has come to be called noise shaping as the name implies that the process is somehow 'random'. In reality the process operates by attempting to average away the *Truncation* effects produced by the finite number of bits per digital value. It does this by storing any truncation errors and using them to adjust later output to produce a more accurate overall output.

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For the sake of the above explanation we ignored the fact that, using 12-bit coefficients, the TDF is capable of providing 28-bit output values. Some manufacturers of CD players have taken advantage of this by employing DACs which convert 18, 20, or even more bits per sample in an attempt to produce more ‘accurate’ analog output signals. It is important to realise that, although this process can provide a ‘smoother’ output waveform it doesn't magically produce any extra information which wasn't in the original set of 16-bit samples. In principle, an ‘ideal’ 16-bit DAC and analog filter would produce the same results as any other ‘ideal’ noise shaped and oversampled system. Any differences stem from how well the system is designed and built, not from any inherent theoretical differences.

Summary

You should now know what is meant by the terms *Oversampling* and *Noise Shaping*. That these are digital signal processing techniques which can be used to perform functions similar to filtering an analog signal. You should also now understand how a *Transverse Digital Filter* works. It should also be clear that — in theory — the same results can be achieved using systems which produce anything from one to umpteen bits per value presented to the output DACs provided that the digital process is performed correctly.