A STUDY ON INDIRECT APPROACH IN DIGITAL CLASS D AMPLIFIER DESIGN EMBODYING INTERPOLATION

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Abstract

In general, a digital Class D amplifier comprises a Pulse Width Modulator (PWM) and an output stage. The PWM output can be generated by either direct approach or indirect approach. In this paper, we investigate the use of indirect approach in Pulse Width Modulator for the design of low-power low-distortion digital Class D amplifier, employing interpolation and delta-sigma modulation. By means of spectrum analysis, we show that the interpolation does not affect THD directly, but instead improving the sampling process in time domain significantly. We verify our analysis by means of MATLAB simulation.

1 Introduction

Recently, Class D audio amplifier has increasingly become more prevalent compared to their linear audio amplifier counterpart, particularly in portable electro-acoustical instruments such as hearing aid. The primary reason is because it offers greater power efficiency, low distortion (comparable to Class A) [1] and well suited for low-voltage (1.1V) low-power (<1mA) application. Recent developments make use of fully digital Class D amplifier in the sense that the digital input is not converted to analog anymore but directly interface to the digital processor, thus eliminating the use of DAC conversion circuit. This offers greater programmability, easier interface to digital processors and a higher immunity to noise.

Typical Class D amplifier mainly comprises of three functional blocks: (i) PWM circuit, (ii) output stage, and (iii) output filter. In many cases, output filter is realized as part of the load, while the PWM circuit and output stage can be integrated on-chip

The Pulse Width Modulator (PWM) block consist of two processes: (i) sampling process – to determine the instantaneous amplitude of the input modulating signal sampled with a carrier (ii) PWM signal generation process – to generated PWM pulses correspond to the digital value provided by the sampling process. This is the direct approach.

The PWM block could also be realized using indirect approach, which is involving oversampling by interpolation and delta-sigma modulation [6]

There are three kind of sampling process used in direct approach, namely: (i) natural sampling, (ii) uniform sampling, (iii) algorithmic-based sampling.

Fig. 1 Digital Class D amplifier

Fig. 2 Natural and Uniform Sampling

Fig. 3 Algorithm-based sampling: Linear Interpolation

Although theoretically there is no signal harmonic in the natural sampling process (zero THD) [2], natural sampling directly emulates its analog counterpart by sampling in very high clock frequency. For example, to get 12-bit resolution of signal sampled at 48 kHz require $2^{12} \times 48 \text{kHz} \approx 200 \text{MHz}$ clock frequency, which definitely yield undesirable amount of power dissipation. Hence, it is not suitable for portable low power application, such as hearing aid.
As depicted in the figure 2 above, uniform sampling does not require any digital PWM sampling, thus much simpler design but suffer from intolerably high THD [8].

Another method is algorithmic-based sampling. There are some reported algorithms proposed, i.e. Linear Interpolation algorithm [3], Delta-Compensation algorithm [3,4,5], and Direct Interpolation algorithm

All the three algorithms are basically an improvement of Uniform Sampling process by approximating the pulse duration $t_p$ through an interpolation between sampled points.

Some reported methodologies in the design of PWM pulse generator are: (i) fast-clock-counter-based, (ii) tapped-delay-line-based, (iii) combination of counter-delay-line-based, and (iv) 12-bit hybrid PWM pulse generator.

One major shortcoming of the direct approach is the high clock frequency. Obviously, natural sampling and uniform sampling process could not be adopted as the former uses very high clock frequency and would dissipated too much power and the later, though use low clock frequency, exhibits highly undesirable THD. Any other methods reported before are also still facing the same problem: considerably high clock frequency, which is impractical for portable low-power instrument such as hearing aid.

To circumvent the problem, it is suggested to use another approach, which is the indirect approach. This indirect approach is involving interpolation and delta-sigma modulation [6]. The usefulness of the interpolation is then investigated.

### 2 The effect of interpolation on THD

To illustrate the practical situation, digital audio input was simulated using sinusoidal signal at 1 kHz sampled at 48 kHz (far above the Nyquist criterion) and take Fourier transform to see its spectrum.

We interpolated by ‘zero-padding’ M times, means we padded (M-1) zeroes between the samples, and then we observed the spectrum.

The result shown in the figure below for the case of four times interpolation:

![Fig. 5 Zero-padding input samples four times and its spectrum](image)

The spectrum was ‘packed’ four times in the frequency domain as expected. Next, we made use of digital filter to get rid of unwanted images. Just for design simulation purpose, we made use of ideal low-pass filter with cut-off frequency as follows:

$$f_c = \frac{f_s}{2M}$$  \hspace{1cm} (1)

We all are aware that the practical digital filters are far from ideal. If the interpolation is going to be used very high such that the spectrum is too packed in the frequency domain, the difficulty would lie on the design of the filter, i.e. to design a low-pass filter with pass band narrow and sharp enough to pass through only the frequency wanted. One way to circumvent the situation is by splitting the process into cascading stages, e.g. 3 stages of 2 times interpolation is equal to 8 times interpolation, thus ease the requirement on the filter design.

![Figure 6. After four times interpolation and filtering process](image)

Observing the spectrum in frequency domain, we notice the “frequency leakage” called the side lobes [7] occurs on frequencies of multiple integers. Since there are no harmonics in the signal simulated before, we made use of these side lobes for the role of harmonics to calculate the THD, just for the sake of simulation only.

The THDs calculated for M times interpolation were tabulated below:
As seen from the result above, the THD was not been affected so much; in fact, it can be said that the interpolation do nothing on the THD.

The result can be explained as follows:

As the interpolation factor increase, the spectrum outcome becomes more and more ‘packed’ in the frequency axis. The spectrum becomes denser as the magnitude decrease as well.

If both the magnitude of the signal frequency and the magnitude of the harmonics decrease with the same scaling factor, certainly we expect the THD would approximately the same.

### 3 The effect of interpolation on sampling

According to the sampling theorem, although the sampling process generates high frequency component, we will see that every frequency component of the original signal is periodically replicated over the entire frequency axis, with the period given by the sampling rate[7].

Thus, if we increase the sampling frequency, the periodically replicated spectrum would be further apart. If we keep on increasing the sampling frequency, up to the point of infinity, we will see only one spectrum lies along the whole frequency axis. This goes well with the theory that a truly analog signal would have a non-periodic spectrum (periodic with period of infinity).

Hence, if we observe carefully, the spectrum after interpolation and filtering process would have the periodic spectrum further apart from each other in frequency domain. This is equal with increasing the sampling frequency in the time domain. Taking the inverse fourier transform, we will be able to see clearly that the interpolation actually have been doing “oversampling” in time domain.

<table>
<thead>
<tr>
<th>M</th>
<th>THD</th>
</tr>
</thead>
<tbody>
<tr>
<td>No interpolation</td>
<td>14.351481698077</td>
</tr>
<tr>
<td>2 x</td>
<td>14.37752334988937</td>
</tr>
<tr>
<td>4 x</td>
<td>14.37651993345477</td>
</tr>
<tr>
<td>8 x</td>
<td>14.40817921592625</td>
</tr>
<tr>
<td>16 x</td>
<td>14.38859204791383</td>
</tr>
<tr>
<td>32 x</td>
<td>14.37157093564880</td>
</tr>
<tr>
<td>64 x</td>
<td>15.00213379826626</td>
</tr>
<tr>
<td>128 x</td>
<td>19.65489840106442</td>
</tr>
</tbody>
</table>

Table 1: Calculated THD after interpolation

Note that the amplitude of the signal is half of the original. The reason for this is simply because half the energy content of the original signal is being filtered away.

In digital signal case, this is totally acceptable since the information is still well preserved.

The in-between samples lie exactly on the original unsampled waveform. If they did not, the interpolated waveform would not have the same spectrum as the original. But we know for sure that the spectrum after interpolation is just ‘packed replication’ of the original spectrum, thus, the new samples exactly fall on the original unsampled waveform.

It is clearly understood that any other interpolation scheme, no matter how sophisticated, will generate points which lie slightly off of the original unsampled waveform, which means the spectrum of the interpolated waveform would be slightly different. But this zero-padding interpolation yields a better job with less computation.

### 4 Conclusion

The indirect approach actually is a elegant way of emulating the natural sampling. Simply by zero-padding the samples in the time domain, it actually resolves the in-between points exactly the same as the original unsampled waveform.
We have investigated the effect of ‘zero-padding’ interpolation on Total Harmonic Distortion for the design of Class D amplifier. We have shown that the interpolation does not directly improve the THD but instead it greatly increases the sampling rate in time domain, emulates the natural sampling process without even make use of impractical high clock frequency.

Due to time constraint, the simulations have not been able to cover the whole objective of the project, which is including simulating the overall process using delta-sigma modulator and output stages.

Suggestion is made for future project to actually use this approach and find out how far the interpolation can improve the THD indirectly.

References