Digital Power Amplification based on Pulse-Width Modulation and Sigma-Delta Loops. A comparison of current solutions

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A low cost digital audio power-amplifier suitable for direct processing of digital sources has been of great interest to the consumer electronics industry over the last few years. This presentation reviews current solutions for a digital power-amplifier based on Pulse-Width- and Sigma-Delta Modulation.

1 Introduction

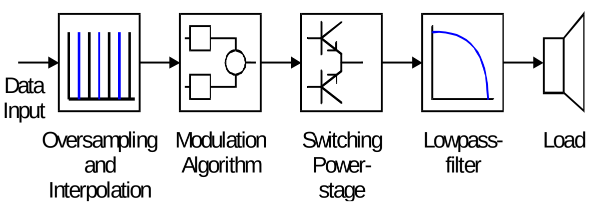
A digital power amplifier provides a direct conversion of a digital signal into the analogue domain where the signal has the capability to drive a high-power load such as a loudspeaker. Figure 1 shows the basic outline of such a system. The first stage performs interpolation of a Nyquist rate PCM (pulse coded modulation) signal to produce an oversampled PCM digital signal. This signal is then fed into a modulation circuit. The modulation is typically sigma delta (Σ-Δ) or pulse width modulation (PWM) and thus produces a final signal that has only two voltage levels. The key aim of the modulation is to produce a signal where the information of the PCM signal value is represented by the mean-value of the bitstream. This allows the original signal information to be extracted by a simple analogue lowpass filter. This filter is typically constructed from passive components like inductors and capacitors. Figure 2 shows how the mean value can be represented by digital pulses.

The two level modulated signal can have a variable width and a fixed cycle-time or the width can be fixed and there is no specific cycle-time. A large width or a high number of pulses represents a greater input value than a small pulse-width or a low number of pulses. Amplifiers for these two level signals belong to the so-called class-D amplifiers, because the power stages have no linear area of operation [1].

1.1 Demands on a high-quality power D/A-conversion

To perform a high quality power conversion the following criteria must be adhered to:

- Good linearity of the modulation in the audio baseband to avoid harmonic distortion.
- High timing accuracy for the switching-points of the pulses to give a precise representation of the input level in the mean value of the pulses.
- Stable power supply for the switching levels. Variations in the switchings levels lead to distortions and further modulation products.
- Steep edges to create a nearly ideal rectangular form of the pulses.
- Feedback from the output to cancel errors produced by by the power circuity.



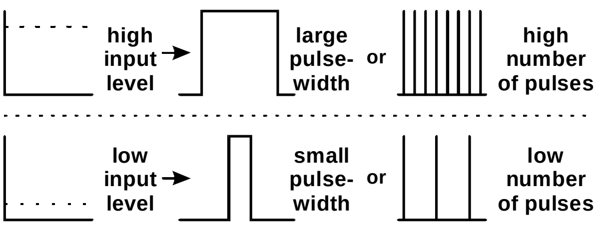


Figure 1: *Basic outline of a digital power amplifier.* **Figure 2:** *Pulse modulation*

2 Pulse-Width Modulation

Pulse-Width Modulation (PWM) is an established technique in industrial applications for motor control and power supplies. PWM is based upon pulses having a fixed cycle-time and a variable pulse-width with the width being proportional to the mean value of the input signal. A fundamental system to create pulse-width modulated signals is shown in Figure 3. The input signal is compared with a linearly ascending reference signal. The period of the reference signal defines the cycle-time of the PWM pulses. If the input signal is higher than the reference, the pulse is switched to the HIGH level, otherwise to the LOW level. Usual forms for the reference signal are ascending, descending and also double-sided ramp [2].

2.1 Analogue PWM Systems

The switching times in all analogue PWM systems are derived from a continous input signal not held constant over a pulse period. This is called natural sampling [2],[3]. Complete PWM based audio amplifiers in integrated form are offered by some semiconductor manufacturers, including Texas Instruments [4] and Philips. These sytems only use analogue components; thus input and output signal are analogue.

2.2 Digital PWM Systems

A digital Pulse-Width Modulator can process digital input data directly. The basic PWM scheme is implemented as an algorithm in a DSP or as logical functions in a FPGA to convert from Nyquist rate PCM to a suitable PWM signal.

2.3 Non-linear Distortion

Fundamentally PWM is a non-linear process. The spectrum of the output pulse-series contains the pulse repetition frequency and its harmonics plus the spectral components caused by the modulation. A complete derivation of the PWM-spectrum can be found in [2],[3]. In a digital modulator the PWM pulse does not represent the mean-value of the continous input signal, but one sample value. This causes further distortions [2],[3], which reach into the base-band of the audio-signal. The spectrum of the audio base-band from a PWM-simulation is shown in Figure 4. The leftmost line represents the original input tone, all further ones result from the non-linear process. One method of linearization is to get nearer to the naturally sampled PWM. This can be done by polynominal interpolation investigated in [5] and [3] and thus allows an estimation of the sampling time of natural PWM. Another approach developed in [6] is based on the spectral properties of the pulses. A target pulse width with a specific frequency-spectrum is chosen. The aim is to achieve the best equality in the range of the audio baseband in the spectrum of all pulses. This is done by adaptive FIR filters and a recursive coefficients estimation algorithm. In a third method used for a commercial amplifier solution a non-linear model of the PCM to PWM conversion is derived [7]. This model is used directly to form FIR filters for different powers of the input-signal, which cancel the spectral properties caused by the non-linear behaviour. This method is used in the available digital power-amplifier system 'EQUIBIT' which was developed by the Danish company TOCCATA [8].

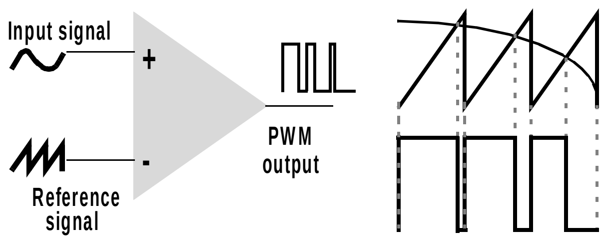


Figure 3: *Generation of PWM*

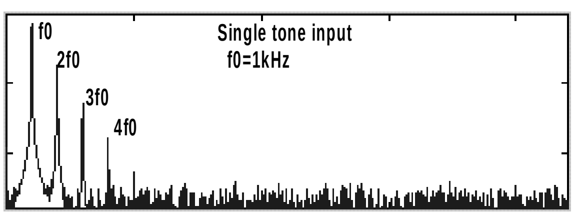


Figure 4: *Spectrum of uniformly sampled PWM*

2.4 Limited resolution of digital systems

The maximum clock-frequency of the digital processing limits the resolution in creating the PWM pulses. To sample a signal quantized to 16 bits, 65536 different pulse-widths would be necessary. With a sampling-frequency of 44.1 kHz and trailing edge modulation the clock-frequency has to be 44100*65536=2.89 GHz. For this reason the quantization is limited to a lower number of bits and noise-shaping is applied [9].

3 Sigma-Delta Modulation (Σ-Δ)

A basic first order sigma delta (Σ-Δ) modulator (or loop) is shown in Figure 5. Similar to PWM the original Nyquist rate PCM signal can be recovered by low pass filtering the bitstream. A sigma delta loop necessarily oversamples the input signal, and the spectra of the bitstream output indicates that the original baseband information has only a low level of quantisation noise, where at freqencies above half of the Nyquist sampling rate, the quantisation noise has been increased. This is refered to as noise shaping. Therefore unlike PWM the Σ-Δ modulator already contains a quantisation noise-shaping . To achieve improved noise-shaping higher order loops can be used, although due to the non-linear nature of the loop, this higher order design is not trivial [10].

3.1 Solutions of Digital Power Amplifiers using Σ-Δ

In theory the Σ-Δ modulator can be used directly to convert oversampled Nyquist rate PCM data into a power bit stream, which can be filtered by analogue lowpass filters. The key modification is to replace the quantizer by a power-switch consisting of transistors. However the main problems of the implementation are the high switching frequencies. Consider a 64 times oversampled Σ-Δ modulator. With a sampling-frequency of 44100 Hz, a clock frequency of 2.8224 MHz can be calculated. The highest possible pulse repetition frequency that can occur then is 1.4112 MHz. The losses in the power-stage are too high at such a frequency and therefore strategies to reduce the pulse repetition frequency have been investigated.

One straightforward method is the pulse group modulation [11]. The output of a Σ-Δ modulator is divided into sections of N bits and these bits are ordered according to their sign. This produces a PWM from the Σ-Δ output. As the Σ-Δ modulator bit-stream represents uniformly sampled data, the typical distortions of PWM are introduced as well. A feedback of the erroneous signal over a lowpass-filter is suggested in [11]. Another method investigated controls the number of consecutive changes in the state of the Σ-Δ bitstream. This method is called bit-flipping [12]. When the number of consecutive changes exceed a certain number, the current bit is inverted to save one transition. This method can be varied in the maximum number of transitions and the number of bits to be inverted. Another problem in the implementation of ΣΔ for power-amplification is the dependence of the noise-spectrum on the input signal and the generation of idle-tones [13]. Chaos in the noise-shaping filter and dithering can be used to reject these unwanted frequencies [14].

4 Comparison of both methods and conclusions

Both PWM and Σ-Δ have their advantages and disadvantages in the usage for power amplification. PWM offers the lowest possible switching-frequency, but without proper linearization the audio baseband will be distorted if applied to uniformly sampled data. The

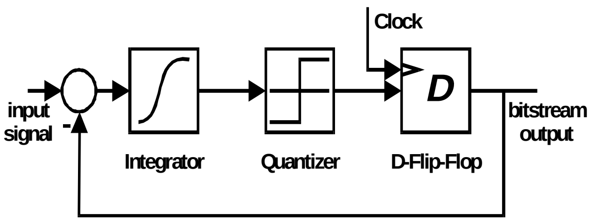


Figure 5: *Basic Sigma-Delta Modulator*

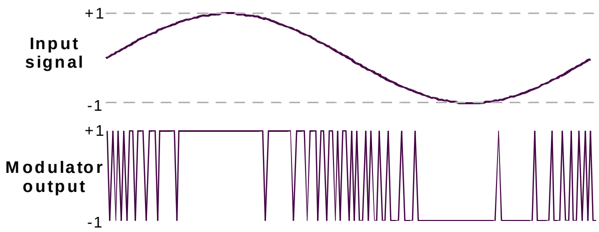


Figure 6: *Modulation of a sinusoidal signal*

linearization methods based on interpolation and non-linear modelling require high computational power. The Σ-Δ modulator is a more linear method, if the presence of idle-patterns is reduced by dithering or by the use of chaotic modulators [13]. Σ-Δ produces a bit-stream, where the pulses are distributed uniformly over the oversampling period. This causes high switching frequencies in the power-stage. The number of bit-transitions can be reduced by controlling or re-organising the current bit-stream.

So far linearized PWM provides good audio quality at moderate switching frequencies and thus was chosen for the implementation in one of the first fully digital implementation by the Danish company TOCATTA [8]. Less consideration has been given to the real behaviour of the power-switches, the reconstruction filter and the loudspeaker as a strong non-linear load. In [8] the design of an approximately ideal output stage as a hybrid element is described, but the effort to produce such a device is very high. The introduction of feedback from the filtered or unfiltered output would provide a possibility to control the distortions caused by these elements. This would ease the demand on the circuit design drastically. The use of feedback implies a conversion from the analogue to the digital domain to be processed by the modulator system. This causes a delay in the feedback path which is harmful for the stability of the system.

Another possibility is to make a model of the output circuitry and to provide a correction of the input-signal in the digital processing. A third method used in [15] is to implement the Σ-Δ modulator containing the power-stage in the analogue domain and to provide an error-feedback. The investigation of the switching-stage, the output circuitry and the influence of a non-ideal power supply is our current work. Other possible ways to reduce the pulse-repetition frequency in order to optimise the effort of signal-processing and output-performance are also being investigated. The introduction of feedback to bring further linearity into the conversion process and to reduce the cost of switching elements should also be of benefit.

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