

ABOUT THE DEMODULATION OF PWM-SIGNALS WITH APPLICATIONS TO AUDIO AMPLIFIERS

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Abstract—In this paper, the reason for the relatively high distortion level of conventional class-D audio amplifiers is discussed. It is pointed out that the classical approach is insufficient for high performance applications. Therefore the importance of the matching between modulation and demodulation is demonstrated by simulation.

I. THE CLASS-D-AUDIO AMPLIFIER

The concept of class-D amplifiers was invented by Baxandall [1] in 1959 for applications in oscillator circuits. In contrast to these tuned amplifiers first ideas for broadband applications were presented in the early sixties. In the following several papers were published about this subject (for further references see e.g. [2], [3], [4], [5]). Although class-D amplifiers are very attractive because of their high power efficiency several disadvantages occur. Especially the switching behaviour of the power transistors and the distortion performance were responsible that class-D amplifiers were not very successful in audio applications.

With the invention of power MOS transistors the class-D amplifiers became interesting again [6]. For example Sony developed its TA-N88 around 1977 where this amplifier principle is used. More recently class-D amplifiers are discussed in conjunction with digital audio concepts. New commercial realisations of amplifiers were presented by Harris Semiconductor and Peavey as well as research papers of this subject were published by Lau [8] and Smedley [9].

The main idea behind the class-D concept is the conversion of an audio signal into a pulse width modulated signal (pwm signals) with constant frequency. Since pwm signals have only two levels a simple switched mode amplifier can be used to obtain high output power. It is well known that the good power efficiency is due to the switched mode. The pwm signal must be demodulated in order to reconstruct the (amplified) audio signal. Usually a low pass filter is used for this purpose. Clearly this filter has to be lossless. The structure of a class-D amplifier is shown in figure 1.

In the literature several pwm modulators are described. In the next section we consider the

properties of these modulators. On the other hand it is not obvious in which sense low pass filters work as a pwm demodulator. Especially since several pwm modulators are available we discuss the advantages and disadvantages of known demodulation techniques.

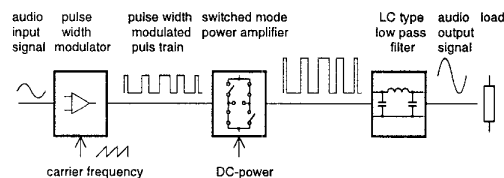


Fig. 1: Structure of a Class-D-Amplifier

II. PWM-MODULATION

Different methods for pwm modulation can be classified if we consider the sampling process (see e.g. Mellor et al. [10]). In the case of natural sampling the input signal is compared with a sawtooth or triangular waveform. In uniform sampling the pulse width is simply proportional to the value of a given sample. In both cases double sided variants are available. In Fig. 2 the four basic principles of pwm modulators are shown.

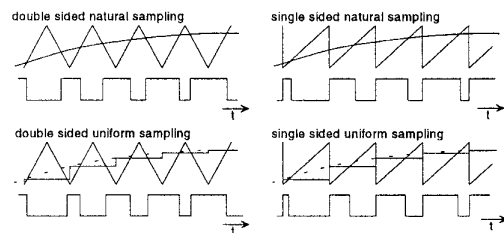


Fig. 2: Different kinds of pwm modulators

If we restrict ourselves to sinusoidal input signals the spectra of the pwm modulated signals can be calculated for these cases by means of the double Fourier series method. In the paper of Mellor et al. [10] these spectra are presented. Unfortunately no analytic results are available for more complex waveforms of the input signal. Therefore we used an fft algorithm for the numerical analysis of pwm signals and implemented this in the MATLAB system. A comparison with the analytical results in

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the sinusoidal cases shows a sufficient correspondence. Fig. 3 shows two selected pwm spectra. In both cases a single input signal with $f_{\text{mod}}=5\text{Hz}$ and a carrier signal with $f_c=49\text{Hz}$ is used. Note that the absolute values of the frequencies are not important for our considerations. Two basic differences can be seen in the spectra. In the case of natural sampling there are no higher harmonics of the 5Hz input signal but there are some high level folded back lines from the 49Hz carrier. One line has a frequency below the input signal itself. Note that the distances of the 5Hz line and the other spectrum lines are not integer multiple of 5Hz, that is these lines are not higher harmonics of the input signal line. In the case of uniform sampling the folded back lines are much lower because of the unsymmetrical side bands of the carrier. But there are many higher harmonics of the input signal.

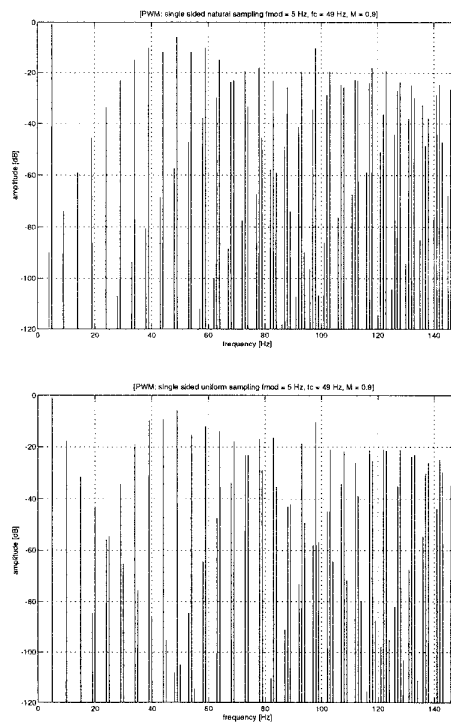


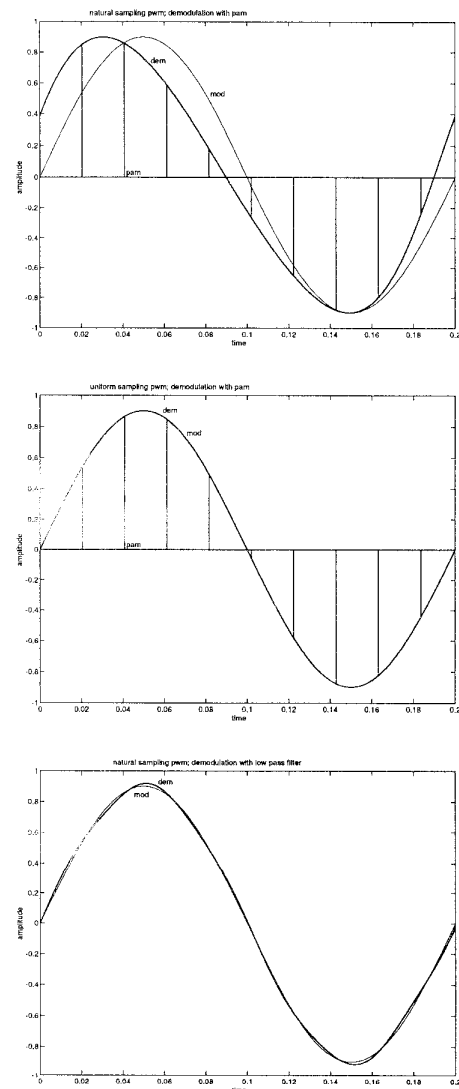
Fig. 3: spectra of a natural and a uniform sampled pwm signal

The differences between the two spectra result in different waveforms of the output signals in time domain. This fact will be considered in next section.

III. PWM-DEMULATION

For the demodulation of the pwm signal the classical concept of class-D audio amplifiers employs a simple low pass filter (e.g. [5]). Another possibility is described in Black [13]. There the demodulation is made in two steps. First the pwm signal is transformed into a pulse amplitude signal (pam). As a result we have an equidistant pulse train

where the amplitude of the pulses contains the information. In the second step the pam signal is processed with a low pass filter. This reconstructs the original signal waveform. These two step demodulation is exactly the reverse way to the uniform sampling process. The typical output waveforms of the four different combinations of modulation and demodulation are shown in fig. 4. The plots are calculated also with the above mentioned MATLAB program.



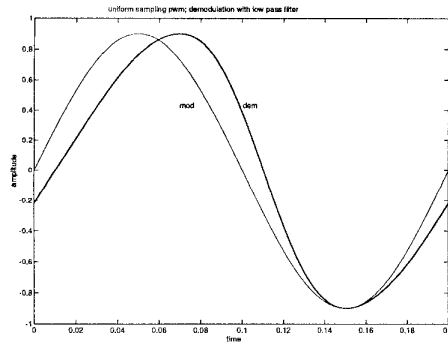


Fig. 4: input and output signals at different pwm modulation/demodulation types

The third plot shows the usually realised class-D amplifier. The pwm signal generated by natural sampling is demodulated with a low pass filter. In the case of the 5 to 49 ratio of frequencies a significant distortion of the output signal can be observed. Obviously an improvement is possible with a higher carrier frequency but the practical realisation becomes more difficult. The second plot demonstrates that the combination of uniform sampling and pam demodulator leads to an exact reconstruction of the input signal waveform. The first and the last plots show other possible combinations where we found even worse results. Tab. 1 shows the summary of the different combinations. As mentioned before the combinations natural/pam and uniform/low pass filter lead to output signals of a sinusoidal input signal with none acceptable amplitude and phase distortions. A correct reconstruction of sinusoidal input signals is possible in the case of uniform/pam and the combination natural/low pass filter is limited by distortions. These results do not depend on single and double sided modulation, respectively.

pwm modulation type		demodulation	
sampling	modulation	with pam	low pass filter
natural	single sided	none	limited
	double sided	none	limited
uniform	single sided	correct	none
	double sided	correct	none

Table 1: correspondence of pwm types and demodulation

IV. PRACTICAL CONSIDERATIONS

Both the analytical calculations and the numerical analysis of the four modulation/demodulation processes is based on several simplifications. Therefore we compared these results with

corresponding measurement data in the cases of natural sampling/low pass filter as well as uniform sampling/pam. We found that the tendencies of calculated results were confirmed essentially. The measured results are in good qualitative agreement with the calculations and simulations given above.

There are two possibilities to get a better matching between modulator and demodulator. The first is to search a demodulation circuit which operates with an intermediate pam signal. But the problem is that the high power pwm signal has to be transformed into a pam signal without significant loss. Until now this approach seems to be not very successful. Another possibility, which was already discussed in literature, is to study a variation of the modulation process in order to get a better adaptation to the „low pass filter“ demodulator. Essential results are given in Mellor et al. [10] as well as in Hawksford [11] and Goldberg / Sandler [12].

In our presentation we will show some measurement results of a practically realised Class-D Audio Amplifier with a digital pwm modulator. The core is a DSP implemented linearisation algorithm which achieves real time performance. Furthermore we will discuss the possibilities of feedback when coupling real loads (loudspeaker) to the amplifier.

V. CONCLUSIONS

In this paper we discussed the principles of an ideal class-D amplifier for audio applications. By means of analytical calculations and numerical analysis with the program MATLAB the time and the frequency behaviour of this amplifier was studied. As a result we showed that the usual realisations of demodulators for pwm signals with low pass filters are not well adapted to the modulation process. Therefore we suggest to use an advanced modulation principle to improve the distortion behaviour of class-D audio amplifiers. A realised circuit and some measurement results are presented. In the future we will study the behaviour of the amplifier with respect to multitone input signals.

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