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A SMALL SIGNAL MODEL OF THE DESIGN OF A CLASS D POWER SWITCHING AMPLIFIER

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Abstract: The class D amplifier is not well known in audio applications. Its excellent power ratio (greater than 90%) is the most important advantage. The MOS transistors switching power stage is able to drive a useful power up to 100W to the loud speaker. However, designing such an amplifier is more difficult than designing a classical class A or AB power amplifier. The distortion level is also slightly higher than in AB structures. Despite the abundant knowledge on class D, there was no electronic model available till now. Then, a small signal model is presented to make the design as simple as possible. The experimental results are given to illustrate the design method.

1. INTRODUCTION

The Class D amplification [1] uses MOS power transistors in switching mode. It is able to obtain a power ratio close to 95%. It is then possible to reduce the size and thereby to miniaturize the audio amplifier. The principle is given in figure 1. The audio signal is compared to a high frequency triangular signal. We then obtain a Pulse Width Modulated intermediate signal. This two level signal drives a full bridge MOS power transistor which works in switching mode. Free running diodes are not shown in figure 1. The power supply over 30V, enables it to carry a high current in the load. A low pass output filter between the MOS stage and the speaker restores the audio signal. And the feedback active network set the voltage gain of the circuit.

\[ \text{Figure 1: class D amplifier principle} \]

2. DESIGN METHODOLOGY

2.1 Principle of Pulse width modulation

An analog signal is compared to a high frequency triangular waveform. The output is a two level pulsed signal as shown in figure 2.

\[ \text{Figure 2: PWM modulation} \]

2.2 Spectrum analysis of the output PWM signal

Let “m” be the modulation index. It is given by the ratio between the peak level Vbfe of the audio signal and the peak level Vtri of the triangular waveform:

\[ m = \frac{V_{bfe}}{V_{tri}} \quad \text{with} \quad 0 < m < 1 \]

Let \( \omega_{bf} \) be the pulsation of the audio signal and \( T_{in} \) the period of the triangular signal. The modulated pulse duration \( \tau \) can be written:

\[ \tau = T_{in} \left( 1 + m \sin (\omega_{bf} t) \right) \quad \text{with} \quad \omega_{bf} = \frac{2\pi}{T_{in}} \]

The "equivalent gain" at low frequencies (ratio between the peak value of the output Low Frequency contribution of the modulated signal and the input peak value Vbfe) is constant. It is given by:

\[ G_{PWM} = \frac{V_d}{2V_{tri}} \]

The frequency spectrum of the PWM signal can obviously be approximated with Bessel functions. But, it is easily obtained by numerical simulation. An example of the normalized spectrum is given in figure 3 for two values of m, 5% and 99%.

The spectrum of the output signal contains the useful audio signal and also frequencies around \( nF_{in} \). The peak value of these contributions depends on the value of m. For its maximum value (m=1 ie Vbfe = Vtri), the peak value of the audio frequency Vmax will be obtained from equation (1), thus:

\[ V_{max} = \frac{0.5V_d}{2} \]

Taking \( F_{in} \) around 200kHz, (i.e greater than ten times \( F_{bf} \)), the reproduction of the audio signal will
be quite easy with a simple output second order lowpass filter.

Figure 3: PWM signal spectrum

2.3 Starting the design

Let Va to be the bridge power supply and Pu the maximum power transmitted to the load. Assuming that the load is fully resistive in audio bandwidth, (loud speaker 8 or 4 Ω), we first calculate the peak value of the useful low frequency audio lines of the modulated signal by 

\[ Pu = \frac{V_{bfs}^2}{R} \]

We initially consider a perfect output filter without losses and we assume that the MOS "on state" drop voltage is negligible. As the "equivalent voltage gain" of the full bridge is: \( 2Va/Vd \), and taking \( m=1 \) (best case of modulation), we can determine the minimum supply voltage Va of the bridge; from (3), it yields:

\[ V_{bfs} \text{ max} = 0.5.Vd \times (2Va/Vd) \]

and:

\[ V_{bfs} \text{ max} = Va \]

Knowing the normalized input voltage level, the value of the required voltage gain is easily calculated. The power ratio of the amplifier obviously depends on the switching and the conduction losses of the MOS transistors and also of the output filter losses. Thus, a small "Ron" MOS transistor should be used.

3. OPEN LOOP DESCRIPTION

3.1 Driver stage

The driver circuit (in our example HIP4080 [3]), located between the PWM modulator and the full bridge, allows the MOS transistors (here IRF 530) to switch under the best conditions by:

- delivering a 12V pulse and a peak current of 3A to improve the switching times.
- generating a dead time to avoid the simultaneous conduction of two transistors at the switching times.
- inhibiting the gates command signal in case of current overload.

3.2 Overload protection circuit

The current overload protection consists of one "shunt" resistor, a low pass filter, a threshold comparator, and a MOS transistor driver inhibition circuit (inside the driver circuit).

3.3 Triangle generator

The triangle generator is a commercial circuit MAX 038. The output level is ±1V. A voltage level shift is added to center the signal around 6V. Its frequency can be adjusted between 100kHz and 1Mhz.

3.4 Feed back network

The diagram of this active lowpass filter is given in figure 4.

The first order transfer function \( H(p) \) is:

\[
\frac{V_s}{V_{o}} = \frac{R_3}{(R_1+R_2) + \frac{1}{p}}
\]

The gain in the audio bandwidth is:

\[
GBP = \frac{R_3}{(R_1+R_2)}
\]

Figure 4: feedback network

4. CLOSED LOOP MODEL

4.1 Small signal model

From a low frequencies point of view, The PWM modulator and the bridge can be considered as a simple gain \( K \), proportionnal to Gpwm, Va and 1/Vd (see equation (3) and (4)).

\[ K = Gpwm \cdot 2Va/Vd \text{ soit }: K = Va/V_{tri} \]

Moreover, at the audio frequency, the feedback network may be considered as a real gain. The circuit can be simplified as indicated in figure 5. A classical small signal analysis is then possible.

Figure 5: amplifier small signal model

Simplifying the circuit, it comes:
The closed loop function $B(p) = \frac{V_1 - V_2}{V_e}$ is:

$$B(p) = \frac{R_4 R_1 + R_2}{R_5 R_3} \frac{1}{1 + \frac{C R_4}{K R_3} (R_1 + R_2)p}$$

For a given voltage gain $G_v$, an audio bandwidth $B_p$ and a chosen gain $K$, it comes:

$$G_v = \frac{R_4 R_1 + R_2}{R_5 R_3} \text{ et } B_p = \frac{1}{2p} \frac{C R_4}{K R_3} (R_1 + R_2)$$

From these formulas, we can calculate the required capacitor and resistor values.

4.2 Stability study

The feed back loop should be taken just before the loud speaker for an optimum effect. Including this filter into the loop would generate a too great phase rotation. And it would be impossible to compensate the system. So, the feed back is done just before the output filter.

5. OUTPUT FILTER

It is a second order low pass filter with a cut-off frequency of 12KHz. The order choice comes from a compromise between complexity and filtering efficiency. The $R,C$ serial network value depends on the speaker characteristics.

6. EXPERIMENTAL RESULTS

For safety reasons, the test was performed on the amplifier limited to an output power of 15W.

6.1 Frequency response

The experimental open loop transfer function is given in figure 8 and figure 9.

The gain and phase margin are large enough to make the amplifier unconditionally stable. The output filter transfer function $\frac{V_{HP}}{V_1 - V_2}$, including the speaker load is given in figure 10.

The loss of 1dB in the audio bandwidth, comes from the parasitic serial resistor of the coils which are not negligible compared to the speaker resistor value.

6.2 Time response

Figure 11 shows the PWM signal modulated at 90%.
6.3 Spectrum measurements

Figure 12: speaker output signal spectrum

Figure 12 shows the output spectrum for a 500 Hz sinewave input signal. The second harmonic attenuation is better than 40db. Thus, the distortion rate is smaller than 1% as indicated before.

7. DISCUSSION

7.1 Sample frequency variation effects

The power ratio obviously depends on the output level. In our example it reaches 83% for an output power of 15W and a "sample" frequency F_{tri} around 200 kHz. Increasing the frequency F_{tri} does not improve the distortion level. Moreover, it increases the switching losses and lowers the power ratio. Thus F_{tri} should not exceed ten times the maximum audio frequency.

7.2 Bridge power supply variations effects

Assuming that nothing else changes:

- if the supply voltage increases, then the gain K also increases. The PWM modulator input signal decreases making the index m smaller. The HF spectrum lines increase. The distortion and the bandwidth in closed loop increase. The noise to signal ratio is then worse and the power ratio is smaller.

- if the supply voltage decreases, m increases. In the worst case, the LF signal is always greater than the triangular signal. We obtain then, the modulation indicated in figure 13. The amplifier saturates and the output signal is then a squared waveform with a fundamental frequency of F_{bf}. The harmonic contents 3, 5, 7 can appear in the bandwidth and give high harmonic distortion (figure 14).

Figure 13: overmodulation

Figure 14: power supply setup effect

Theses effects must be taken into account to choose carefully the power supply voltage.

8. CONCLUSION

In this study, we presented a method to design a power switching class D amplifier as simply as possible. Modelling the closed loop circuit at the audio frequency for small signals was the most difficult step in this method. The experimental results show a good matching with our theoretical approach. However, this method only enables us to find the most important parameters of the amplifier. The power ratio must be taken into account when the electronic design is started. In our example, it could be easily improved by reducing the power consumption of the triangle generator and the active feedback filter.

The optimisation of the power ratio will allow the reduction of the size of the heatsinks and probably their elimination for a medium power audio amplifier (i.e 15W). Thus, the full integration of our amplifier will be possible.

REFERENCES