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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 feed back active network set the voltage gain of the circuit

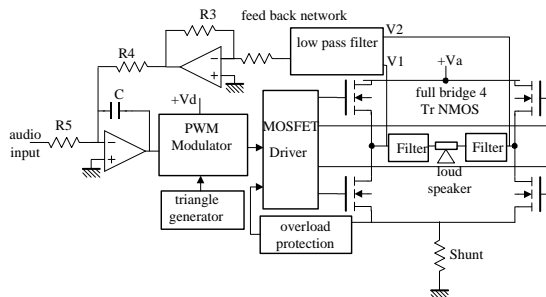


Figure 1 : class D amplifier principle

## 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.

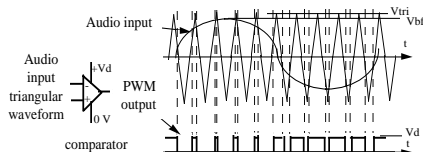


Figure 2 : PWM modulation

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 :

$$m = V_{bfe}/V_{tri} \quad \text{with } 0 < m < 1$$

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

$$\tau = \tau_0 (1 + m \sin(\omega_{bf} t)) \quad \text{with } \tau_0 = T_{tri}/2$$

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 :

(1)  $G_{pwm} = V_d/(2 \cdot V_{tri})$ , where  $V_d$  is the peak value of PWM signal.

### 2.2 Spectrum analysis of the output PWM signal

Let  $F_{tri}$  be the frequency of the triangular signal and  $F_{bf}$  the frequency of the audio input signal. As a complete mathematical approach is very difficult, we can simplify it, by assuming that the pwm signal is periodic. Then, we can write : (2)  $pwm(t) =$

$$\frac{V_d \tau}{T_{tri}} + 2V_d \sum_{n=1}^{\infty} \left( \frac{1}{\pi n} \sin[(1 + m \sin F_{bf} \cdot t) \frac{n 2\pi F_{tri} \cdot \tau_0}{2}] \cdot \cos(n 2\pi F_{tri} \cdot t) \right)$$

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%.

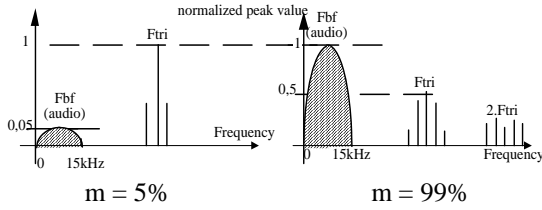
The spectrum of the output signal contains the useful audio signal and also frequencies around  $n \cdot F_{tri} \pm k \cdot F_{bf}$ . The peak value of these contributions depends on the value of m. For its maximum value ( $m=1$  ie  $V_{bfe} = V_{tri}$ ), the peak value of the audio frequency  $V_{max}$  will be obtained from equation (1), thus :

$$(3) V_{max} = 0.5 \cdot V_d$$

where  $V_d$  is the peak value of the PWM signal.

Taking  $F_{tri}$  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  $V_a$  to be the bridge power supply and  $P_u$  the maximum power transmitted to the load. Assuming that the load is fully resistive in audio bandwidth, (loud speaker 8 or 4  $\Omega$ ), we first calculate the peak value of the useful low frequency audio lines of the modulated signal by  $P_u = V_{bfs}^2/R$ . 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 :  $2V_a/V_d$ , and taking  $m=1$  (best case of modulation), we can determine the minimum supply voltage  $V_a$  of the bridge ; from (3), it yields :

$$V_{bfs} \max = 0,5 \cdot V_d \times (2V_a/V_d)$$

and :

$$V_{bfs} \max = V_a$$

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  $\pm 1V$ . 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

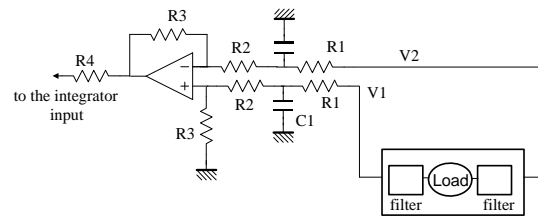
The diagram of this active lowpass filter is given in figure 4.

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

$$(4) \frac{V_s}{V_2 - V_1} = \frac{R_3}{(R_1 + R_2)} \frac{1}{(1 + \frac{C_1 \cdot R_1 \cdot R_2}{R_1 + R_2} p)}$$

The gain in the audio bandwidth is:

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



**Figure 4 : feedback network**

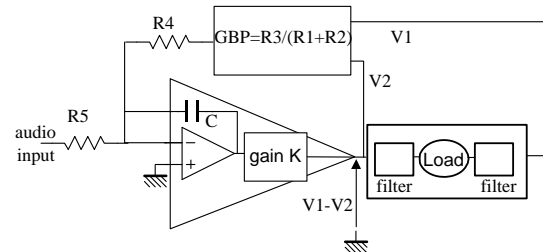
## 4. CLOSED LOOP MODEL

### 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  $G_{pwm}$ ,  $V_a$  and  $1/V_d$  (see equation (3) and (4)).

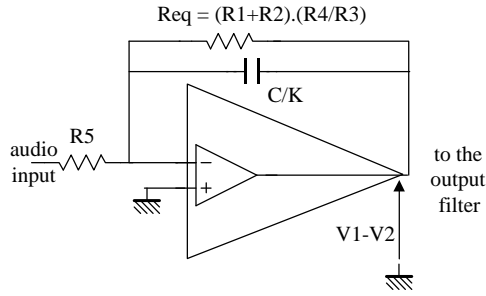
$$K = G_{pwm} \cdot 2V_a/V_d \text{ soit : } K = V_a/V_{tri}$$

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 :



**Figure 6** : final LF small signal equivalent diagram

The closed loop function  $B(p) = (V1-V2)/V_e$  is :

$$(5) B(p) = \frac{R4}{R5} \frac{R1+R2}{R3} \frac{1}{1 + \frac{C}{K} \frac{R4}{R3} (R1+R2)p}$$

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

$$(6) G_v = \frac{R4}{R5} \frac{R1+R2}{R3} \text{ et } BP = \frac{1}{2p} \frac{C}{K} \frac{R4}{R3} (R1+R2)$$

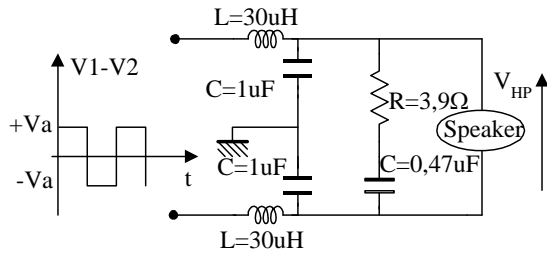
From these formulas, we can calculate the required capacitor and resistor values.

## 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.



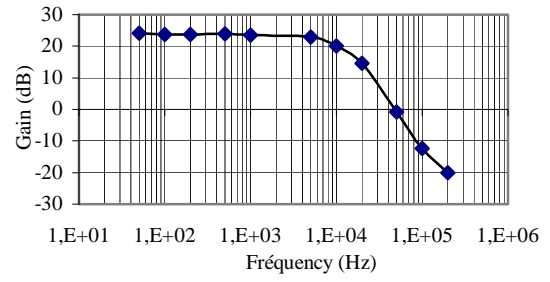
**Figure 7** : output filter

## 6. EXPERIMENTAL RESULTS

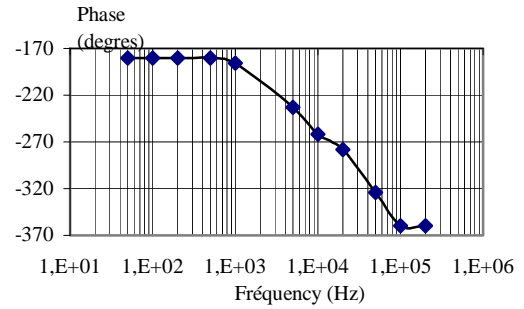
For safety reasons, the test was performed on the amplifier limited to an output power of 15W.

### 6.1 Frequency response

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

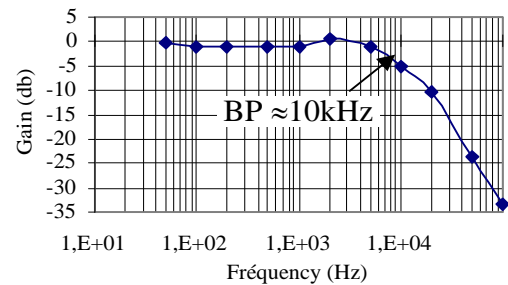


**Figure 8** : voltage gain response



**Figure 9** : phase response

The gain and phase margin are large enough to make the amplifier unconditionally stable. The output filter transfer function  $V_{HP}/(V1-V2)$ , including the speaker load is given in figure 10.

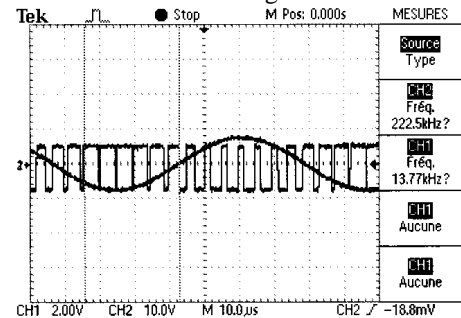


**Figure 10** : output filter gain response

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%.



**Figure 11** : PWM signal modulated at 90%

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<http://www.ti.com/sc/docs/msp/pran/tutorial.html>
- [5] Philips TDA8920 2x50W SOI Class D amplifier  
data sheet 1998