The Class-D Amplifier

(From the book *Introduction to Electroacoustics and Audio Amplifier Design, Second Edition - Revised Printing*, by W. Marshall Leach, Jr., published by Kendall/Hunt, © 2001.)

A class-D amplifier is one in which the output transistors are operated as switches. When a transistor is off, the current through it is zero. When it is on, the voltage across it is small, ideally zero. In each case, the power dissipation is very low. This increases the efficiency, thus requiring less power from the power supply and smaller heat sinks for the amplifier. These are important advantages in portable battery-powered equipment. The "D" in class-D is sometimes said to stand for "digital." This is not correct because the operation of the class-D amplifier is based on analog principles. There is no digital coding of the signal. Before the advent of the class-D amplifier, the standard classes were class-A, class-AB, class-B, and class-C. The "D" is simply the next letter in the alphabet after "C." Indeed, the earliest work on class-D amplifiers involved vacuum tubes and can be traced to the early 1950s.

Fig. 1 shows the basic simplified circuit of a class-D amplifier. We assume a bipolar power supply so that $V^- = -V^+$. The amplifier consists of a comparator driving two MOSFET transistors which operate as switches. The comparator has two inputs. One is a triangle wave, the other is the audio signal. The frequency of the triangle wave must be much higher than that of the audio input. The voltage output of the comparator can be written

$$v_C = -V_1 \quad \text{for} \quad v_S > v_T \qquad v_C = +V_1 \quad \text{for} \quad v_S < v_T \tag{1}$$

This voltage is applied to the input of a complementary common-source MOSFET output stage. Each transistor operates as a switch. For $v_C = -V_1$, M_1 is on and M_2 is off. If the voltage drop across M_1 is negligible, then $v'_O = V^+$. Similarly, for $v_C = +V_1$, M_2 is on, M_1 is off, and $v'_O = V^-$. In practice, there is a small voltage drop across the on MOSFET switch so that the peak output voltage is less than the power supply voltage. For the case $v_S = 0$, v'_O is a symmetrical square wave. The low-pass filter consisting of L_1 and C_1 passes the average value of the square wave to the loudspeaker, which is zero. Thus $v_O = 0$ for $v_S = 0$. The network consisting of R_1 and C_2 compensates for the inductive impedance of the loudspeaker voice coil so that the filter sees a resistive load at high frequencies.

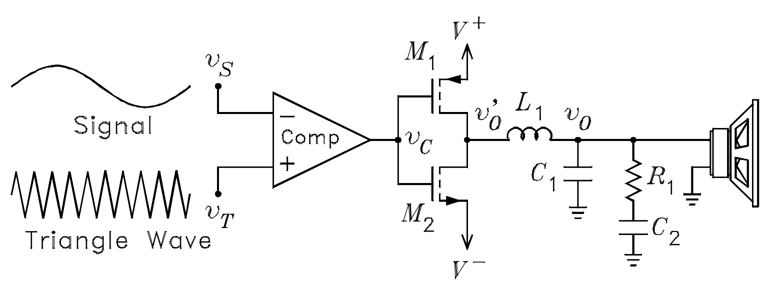

Figure 1: Basic class-D amplifier.

Fig. 2 shows the circuit waveforms for the case where v_S is a sine wave. For purposes of illustration, the sine wave frequency is $f_S = 1$ kHz and the triangle wave frequency is $f_T = 20$ kHz. The sine wave amplitude is $0.75V_{TP}$. For $v_S > 0$, the duty cycle of the square wave changes so that v'_O spends more time at its positive level than at its negative level. This causes v'_O to have a positive average value. Similarly, for $v_S < 0$, v'_O has a negative average value. The waveform for v'_O

is said to be pulse-width-modulated. The passive filter consisting of L_1 and C_1 passes the average or low-frequency value of v'_O to the loudspeaker load and rejects the higher-frequency harmonics of the switching waveform.

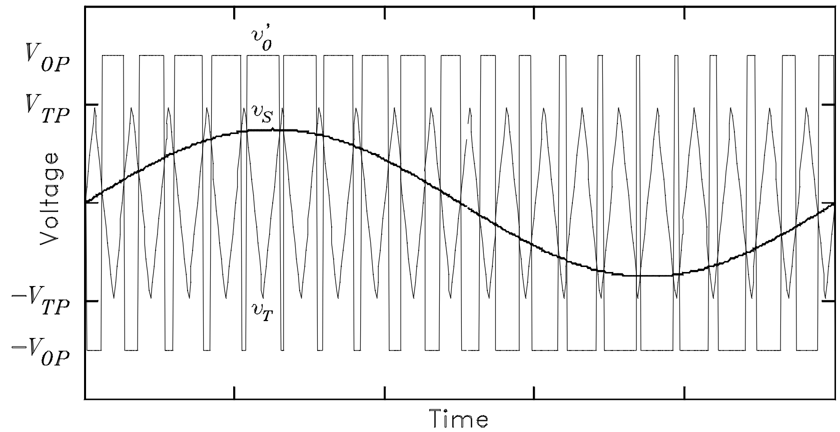

Figure 2: Amplifier voltage waveforms.

The effective gain of the amplifier can be determined by applying a dc voltage at the input and calculating the ratio of $\langle v'_O \rangle$ to v_S, where $\langle v'_O \rangle$ denotes the low-frequency time average of v'_O. If v_S is increased, $\langle v'_O \rangle$ increases linearly until it reaches the level V_{OP}, which corresponds to the positive clipping voltage at the output. This occurs when $v_S = V_{TP}$. It follows that the effective gain k is given by

$$k = \frac{\langle v'_O \rangle}{v_S} = \frac{V_{OP}}{V_{TP}} \tag{2}$$

Fig. 3 shows the waveforms of the output voltage v_O for two values of the cutoff frequency of the LC filter. The transfer function of the filter is

$$\frac{V_o}{V'_o} = \frac{1}{(s/\omega_c)^2 + (1/Q_c)(s/\omega_c) + 1} \tag{3}$$

where $\omega_c = 2\pi f_c = 1/\sqrt{L_1 C_1}$ is the resonance frequency and $Q_c = 1/(\omega_c R_L C_1)$ is the quality factor. The load resistance R_L is the effective high-frequency resistance of the loudspeaker voice coil in parallel with the matching network consisting of R_1 and C_2. The quality factor is $Q_c = 1/\sqrt{2}$ for the waveforms in Fig. 3 so that the gain is down by 3 dB at ω_c. The signal frequency is $f_S = 1$ kHz. The filter resonance frequency for the v_{O1} waveform is $f_c = 1$ kHz. For the v_{O2} waveform, it is $f_c = 8$ kHz. The harmonics of the pulse-width-modulated signal are clearly visible on the v_{O2} waveform.

For minimum distortion, the frequency of the triangle wave should be as high as possible compared to the cutoff frequency of the filter. Because the filter resonance frequency corresponds to the signal frequency for the v_{O1} waveform in Fig. 3, the phase lag is 90°. The phase lag for the v_{O2} waveform is less because the resonance frequency is greater than the signal frequency. A higher-order filter can be used to more effectively remove the high-frequency switching harmonics. For example, a third-order LC filter or a fourth-order filter consisting of the cascade of two second-order LC filters could be used.

Fig. 4 shows the spectrum of the v'_O waveform. It contains a fundamental at f_S. Above f_S, the significant switching harmonics are at f_T, $f_T \pm 2f_S$, $2f_T \pm f_S$, $2f_T \pm 3f_S$, etc. The lowest of these is at the frequency $f_T - 2f_S$. The triangle wave frequency must be chosen high enough so

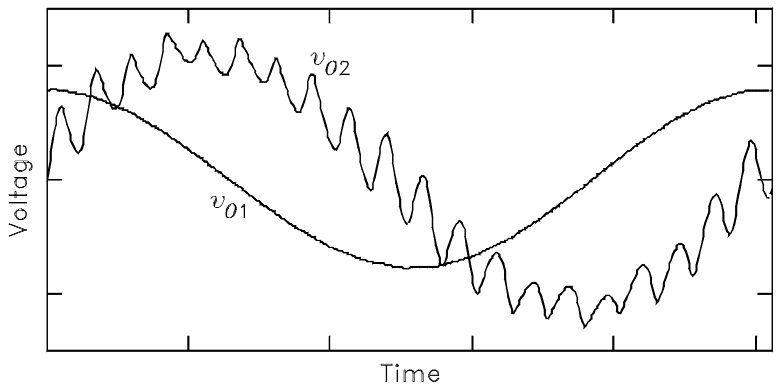

Figure 3: Output voltage waveforms for two different LC filter cutoff frequencies.

that the lowest significant harmonic is well above the highest signal frequency of interest. Thus we have the requirement $f_T - 2f_S \gg f_S$ or $f_T \gg 3f_S$. To minimize ripple on the output, the cutoff frequency of the LC filter should be much lower than f_T. For example, in a wideband amplifier with a maximum signal frequency of 20 kHz, the switching frequency should ideally be 600 kHz or greater. Because of limitations imposed by a high switching frequency, a more practical value might be 300 kHz. The -3 dB frequency of the LC filter should be much lower than the switching frequency. For example, it might be 30 kHz for a 300 kHz switching frequency. Note that the amplitude of the harmonic at f_T is larger than that of the signal. At the signal clipping level, the signal harmonic becomes 1.5 times as large as the harmonic at f_T.

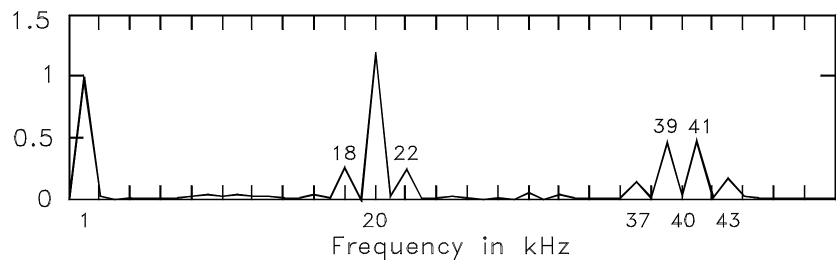

Figure 4: Unfiltered spectrum of the output voltage.

Negative feedback can be used around the basic amplifier circuit to improve its performance. Fig. 5 shows such a circuit. The input op amp acts as an integrator to set the bandwidth. For a sinusoidal input signal with a frequency much lower than the switching frequency, the effective transfer function for the circuit and its pole frequency are given by

$$\frac{V_o'}{V_s} = -\frac{R_F}{R_2}\frac{1}{1+s/\omega_0} \qquad \omega_0 = 2\pi f_0 = \frac{k}{R_F C_F} \tag{4}$$

The pole frequency must be greater than the highest frequency to amplified but lower than the switching frequency. Because the integrator has a very high gain at dc, it acts to minimize dc offsets at the output. For a wide band amplifier, a typical value of f_0 might be 60 kHz.

Class-D amplifiers are often operated in a bridged configuration to increase the output power without increasing the power supply voltages. A bridged output stage is shown in Fig. 6. A typical input circuit is shown in Fig. 7. The feedback voltage v_F is proportional to the difference voltage

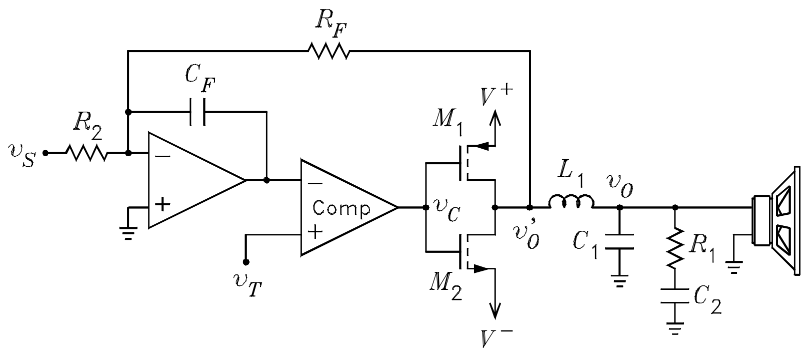

Figure 5: Amplifier with negative feedback.

$v_{O1}' - v_{O2}'$. The bridge circuit is often designed with $V^- = 0$ and a dc offset at each output of $V^+/2$ V, thus eliminating the need for a bipolar power supply.

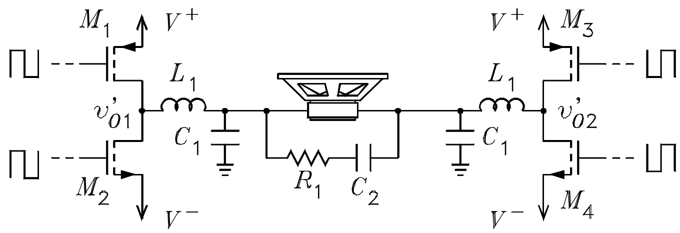

Figure 6: Bridged output stage.

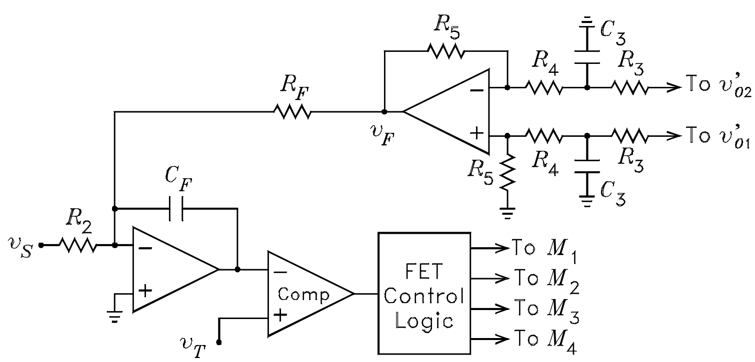

Figure 7: Input stage to the bridged amplifier with feedback.

In the circuit of Fig. 7, a diff amp is used to subtract v_{O2}' from v_{O1}'. The capacitors labeled C_3 act as low-pass filters to limit the rise time of the signals applied to the op amp to prevent slewing. The transfer function for the diff amp and its pole frequency are given by

$$\frac{V_f}{V_{o1}' - V_{o2}'} = \frac{R_5}{R_3 + R_4} \frac{1}{1 + s/\omega_1} \qquad \omega_1 = 2\pi f_1 = \frac{1}{(R_3 \| R_4)\, C_3} \tag{5}$$

The overall transfer function for the amplifier is given by

$$\begin{aligned}\frac{V'_{o1}-V'_{o2}}{V_s} &= -\frac{R_F}{R_2}\frac{R_3+R_4}{R_5}\frac{1+s/\omega_1}{s^2/\omega_1\omega_2+s/\omega_2+1} \\ &= -\frac{R_F}{R_2}\frac{R_3+R_4}{R_5}\frac{1+s/\omega_1}{(s/\omega_0)^2+(1/Q_0)(s/\omega_0)+1}\end{aligned} \tag{6}$$

where ω_2, ω_0, and Q_0 are given by

$$\omega_2 = 2\pi f_2 = \frac{R_5}{R_3+R_4}\frac{k}{R_FC_F} \qquad \omega_0 = 2\pi f_0 = \sqrt{\omega_1\omega_2} \qquad Q_0 = \sqrt{\frac{\omega_2}{\omega_1}} \tag{7}$$

The amplifier should exhibit good stability for $Q_0 \leq 1$. Thus we have the condition $\omega_1 \geq \omega_2$. However, the simple model used here neglects higher order poles. Because the existence of such poles degrades stability, a conservative approach might be to design for $Q_0 \leq 0.5$ so that the poles in Eq. (6) are real. This requires $\omega_1 \geq 4\omega_2$. For example, in a wideband amplifier utilizing a switching frequency of 300 kHz, f_1 might be chosen to be equal to the switching frequency and f_2 might be chosen to be 75 kHz.

An example triangle wave generator circuit is shown in Fig. 8. The circuit consists of an integrator driving a comparator that is connected as a Schmitt trigger. The output of the comparator drives the input to the integrator. Let the comparator output voltages be $+V_1$ and $-V_1$. When the voltage is at the $-V_1$ level, the triangle wave output rises with the slope $m = V_1/R_6C_4$. Let the peak values of the triangle wave be $+V_{TP}$ and $-V_{TP}$. It follows that $2V_{TP} = mT/2 = V_1/2f_TR_6C_4$, where $T = 1/f_T$ is the period of the triangle wave. The comparator switches states when the voltage at its non-inverting input goes through zero. This occurs when $V_1/R_8 = V_{TP}/R_7$. Solution for f_T and V_{TP} yields

$$f_T = \frac{R_8}{4R_6R_7C_4} \qquad V_{TP} = \frac{R_7}{R_8}V_1 \tag{8}$$

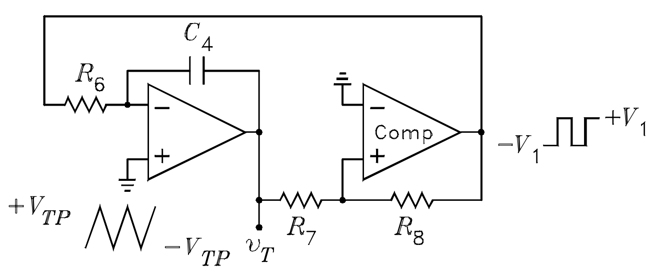

Figure 8: Triangle wave generator.

A problem called "shoot through" can reduce the efficiency of class-D amplifiers and lead to potential failure of the output devices. This occurs during the transition when one device is being cut off and another is being cut on. During the transition, both devices are on and a large current pulse can flow through the two. This can be eliminated by driving the gates of the MOSFETs with asymmetrical square waves such that one device is cut off before the other is cut on. One way of achieving this in the circuit of Figs. 1 and 5 is to use two comparators, one for each MOSFET. A positive dc offset is added to the triangle wave input to the comparator which drives M_1 and a negative dc offset is added to the triangle wave input to the comparator which drives M_2. This effectively adds crossover distortion to the v'_O output waveform, but the frequency components are above the switching frequency, thus outside the audio band.

The high switching frequency used in class-D amplifiers is a potential source of rf interference with other electronic equipment. The amplifiers must be properly shielded and grounded to prevent radiation of the switching harmonics. In addition, low-pass filters must be used on all input and output leads, including the power supply leads.

A variation of the class-D amplifier is called a filterless class-D amplifier. In the absence of an input signal, its output signal is zero rather than a symmetrical square wave. This eliminates the need of a low-pass filter to prevent application of the square wave to the loudspeaker. When the input voltage goes positive, the output voltage is a train of pulse width modulated pulses which switch between 0 and $+V_{OP}$. When the input voltage goes negative, the output voltage is a train of pulse width modulated pulses which switch between 0 and $-V_{OP}$. This is illustrated in Fig. 9. The loudspeaker responds to the average value of the signal, which is the audio signal. Because there is no filter, rf interference problems require the amplifier to be mounted as close to the loudspeaker as possible. The spectrum of the output signal is shown in Fig. 10. One problem with the filterless class-D amplifier is crossover distortion. To eliminate this, the FET control logic must be designed so that the amplifier puts out very narrow alternating positive and negative pulses in the absence of an input signal. In effect, this biases the amplifier in the class-AB mode.

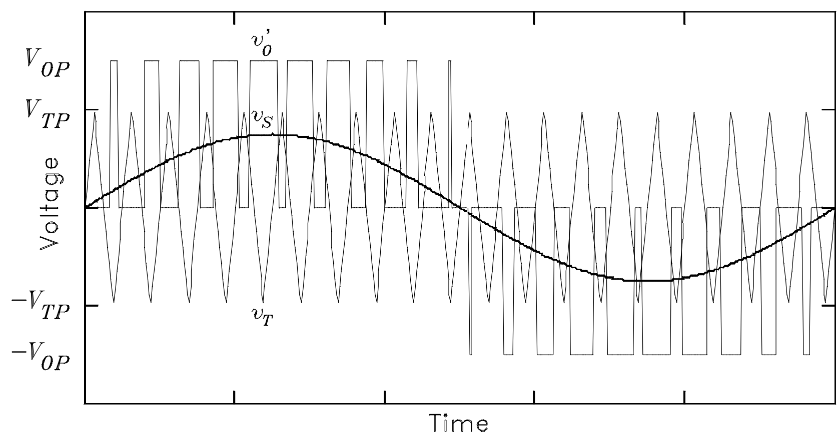

Figure 9: Amplifier voltage waveforms.

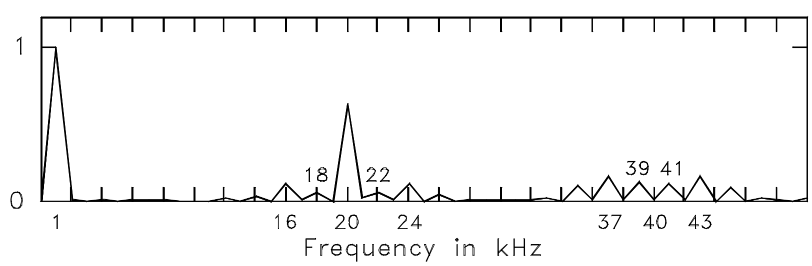

Figure 10: Spectrum of the output voltage v_O'.