DO SPEAKER CABLES AFFECT PERFORMANCE?

6

audioXPRESS

IT JUST SOUNDS BETTER

MARCH 2001
US $7.00 Canada $10.00

Continuing *Audio Electronics, Glass Audio* & *Speaker Builder*

equipment reviews:

SPEAKER CITY MTM-18
D'APPOLITO & FLORIAN

ENIGMA ACOUSTIQUE OREMUS
D'APPOLITO & COLIN

MARANTZ PMD 340 PRO CD
GARY GALO

BUILD THIS DIGITAL CLASS-D SUB AMP
Tom O'Brien

DC HEATERS FOR DIRECT COUPLING

WILLIAMSON PHASE SPLITTERS

CONSTRUCT YOUR NEXT COMPUTER SPEAKER PAIR
G. R. Koonce

REMAKING T DYNACO MARK
Kara Chaffee

A SIMPLE, CHEAP CONTROL UNIT
Joseph Still

www.audioXpress.com

REGULARS:
TEST TRACKS
NEW CHIPS
BOOK SHORTS

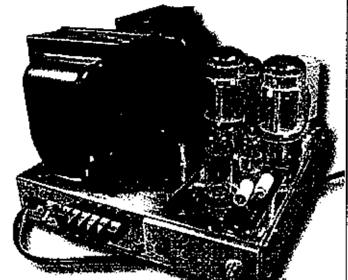
INSIDE SCOOP ON SPEAKER DRIVER DESIGN

audioXPRESS

CONTENTS

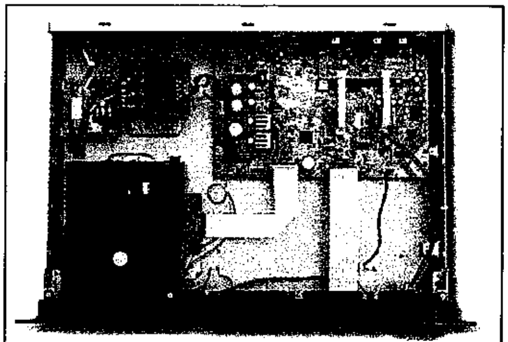
VOLUME 32 NUMBER 3 MARCH 2001

FEATURES



page 18

REVIEWS



page 54

page 44

audioXpress (US ISSN 0004-7546) is published monthly, at $29.95 per year, $53.95 for two years. Canada add $12 per year; overseas rates $59.95 per year, $110 for two years; by Audio Amateur Inc., Edward T. Dell, Jr., President, at 305 Union St., PO Box 876, Peterborough, NH 03458-0876. Periodicals postage paid at Peterborough, NH, and additional mailing offices.

POSTMASTER: Send address changes to: *audioXpress*, PO Box 876, Peterborough, NH 03458-0876.

Digital Class-D Subwoofer Amp, Part 1

Here's a close-up look at the pros and cons of true digital amplifiers. **By Thomas O'Brien**

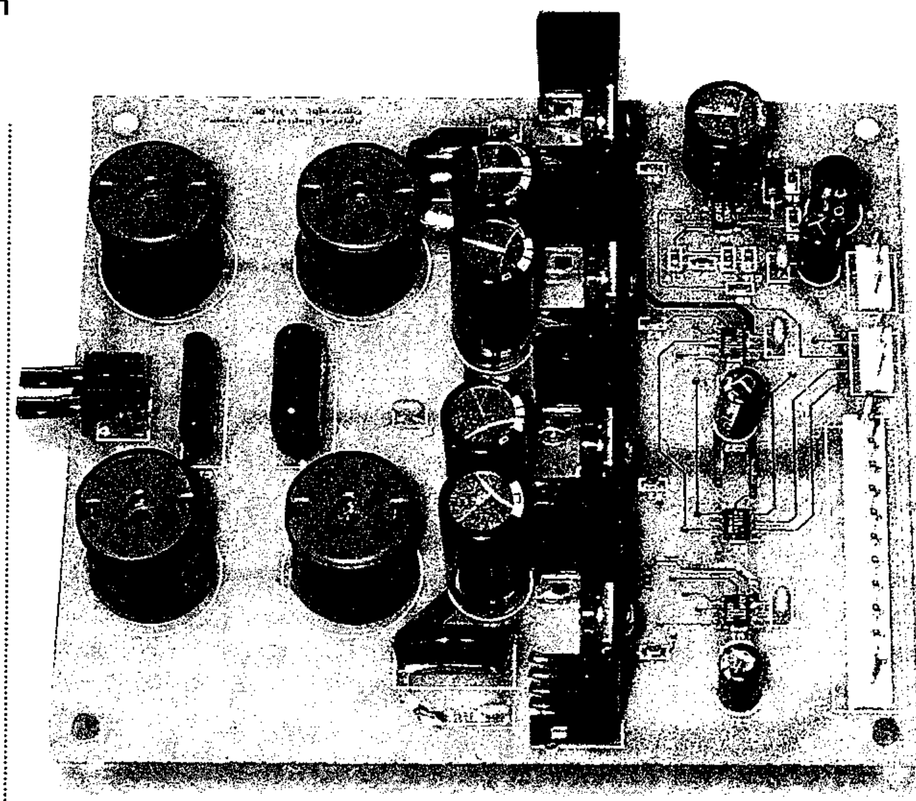


PHOTO 1: Modulator board.

In the "old days" of audio, amplifiers were purely analog (tubes, then transistors) and inefficient, wasting power by generating heat. The wasted power was a concern, especially for portable and miniaturized audio systems, so Class-D was used to conserve power. Most Class-D amps use pulse-width modulation (PWM) to convert the incoming audio to a pulse train, although there are several variations of how this is done.

Class-D is the same technique used in classic motor drivers, but modern high-speed electronics are required to run at appropriate frequencies for accurately driving a loudspeaker. There are benefits aside from efficiency when using the Class-D approach. Elimination or reduction of some types of distortion is possible, and good transient response leads to a reputation for accurate sound.

In a Class-D amplifier, a modulator converts the input signal to a pulse stream. The modulator output is a low-power signal. High-power circuits (called the "output stage") amplify this signal, and the high-power output drives a speaker through a passive filter. The filter removes high-frequency noise from the signal driving the speaker.

ABOUT THE AUTHOR

Thomas J. O'Brien attained a BSEE at Drexel University in 1992, and is currently pursuing an MSEE at Villanova University. He co-founded Sycom Technologies in 1993, producing personal digital voice recorders. He joined Quadrant International in 1997, designing DVD-player and Internet-TV hardware, and began working for InterDigital, Inc. in 2000, designing ASICS for cell phones. Thomas has pursued audio electronics on his own for 15 years, and has ten years of experience designing analog and digital Class-D audio power amplifiers. He also holds a patent, granted in August 2000, in the area of digital audio amplification. For more information about digital amplifier companies, check out www.digitalamp.com.

DIGITAL CLASS-D

Although it may seem reasonable, the "D" in Class-D doesn't stand for "digital"; it's merely a configuration designator, as in Class-A, Class-B, Class-AB, and so on. Class-D amplifiers came to be known as "digital amplifiers" because they output a two-state signal (although some Class-D amps use more states), as do digital logic circuits. The output of a Class-D amp drives the load through a filter, removing the high frequency "carrier." However, most Class-D amps deal only with analog signals, and the output is not discrete-time (as in a computer); these amps are called analog Class-D amps.

It is difficult to achieve high performance with analog Class-D because analog modulators are inherently noisy. At least one level of analog feedback is usually employed to lower distortion to reasonable levels. Amplifiers requiring large amounts of feedback are sometimes viewed as poor designs, and are susceptible to instability.

The term "digital amplifier" is also used as a marketing tool. "Digital" has come to represent high quality in mass-market audio. For example, the word "digital" has even been applied to headphones. This can be confusing to the typical consumer.

Digital Class-D amplifiers, often called "true digital amplifiers," convert digital audio data (pulse-code modulation (PCM), usually from CD or DVD) to

PWM with logic circuits; this process is discrete-time. Converting the digital signal to analog before amplification is not necessary. The process of converting the signal from analog to digital or vice-versa adds artifacts to the audio. Also, digital Class-D amps do not require analog feedback to reach high-performance levels, as do typical analog Class-D amps.

On the qualitative side, digital Class-D is the most direct connection from a digital audio source to a loudspeaker. This directness provides very transparent sound, and some believe it to be the best way to hear a CD. Folks who swear by tube amps would rather hear the recording on LP instead, anyway.

Digital Class-D amps can achieve outstanding performance levels, including low distortion and high dynamic range. However, in some applications, performance is slightly compromised to lower the price of this technology by reducing signal processing in the modulator or by using cheaper FETs, FET drivers, inductors, and capacitors. The output stage transition time must be very fast and the output impedance must be very low in order to minimize distortion. The modulator must produce very accurate PWM with low noise, as well.

It is difficult, but not impossible, to build a digital Class-D amp with high performance and reasonable cost. The cost of digital Class-D is slightly higher than analog Class-D. Digital Class-D requires a better output stage because analog Class-D can correct for output stage problems with feedback, and digital Class-D doesn't have feedback.

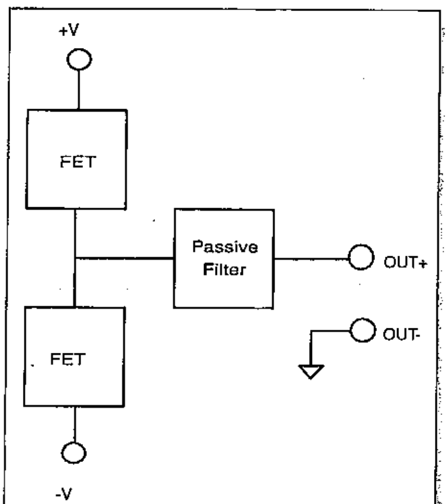


FIGURE 1: Half-bridge configuration.

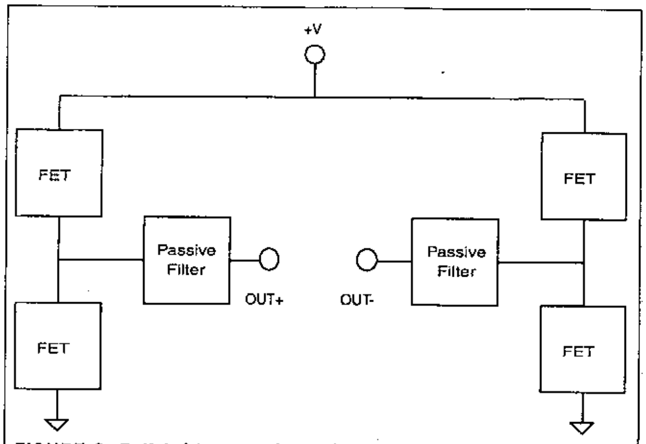


FIGURE 2: Full-bridge configuration.

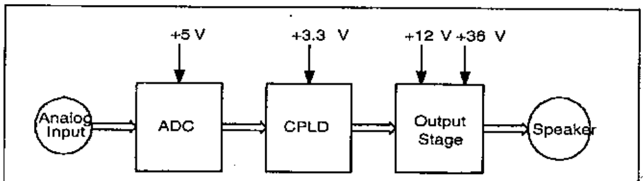


FIGURE 3: SA1 system configuration and required voltages.

ADVANTAGES

Digital Class-D amps do not need "calibration" or adjustment for DC offset because the offset can be digitally fixed near zero. In fact, digital Class-D amps are very repeatable in production because their behavior is almost completely defined in the digital domain, and it doesn't vary as much over age, temperature, and component variations as with analog Class-D amplifiers. Also, digital modulators don't rely on tight component tolerances, super-low noise components, and precise PCB layout, as do analog modulators.

A crystal oscillator digitally clocks a Class-D amp. Crystal oscillators provide the timing for accurate pulse widths and dead-time delays (see section on dead-time delays later in this article). In an analog Class-D amp, the pulse widths and dead times are determined by analog circuits and are more prone to noise than is a digital Class-D amp.

Digital Class-D also benefits from operating in the digital domain because circuits like equalizers, filters, and effects are often implemented digitally with no need to convert to analog.

Amplification does not add a level of complication to systems running DSP on the signal, and sometimes can be performed in the amplifier itself. A simple example is volume control. Many "integrated amps" have simple selection switches and a volume control. In this case, tone controls or equalizers are performed outside the amplifier. Selection switches are loss-less in the digital domain, and volume controls simply scale the audio data and are easily built into a digital Class-D amp.

In addition to eliminating the need for feedback, digital Class-D enjoys the same benefits as analog Class-D. There is no zero-crossing distortion. Efficiency is high, so there's little wasted heat; therefore, heatsinks, power supplies, and drive electronics can be very small without loss of quality or output power.

OVERCOMING DISCRETE-TIME LIMITATIONS

In a continuous PWM system (analog Class-D), the average output voltage is continuously variable because the pulse width is also continuously variable. In a digital Class-D system, the PWM is a clocked output, meaning that the pulse width must be a multiple of clock cycles.

For the pulse width to have 16-bit accuracy, you would need 65536 available pulse widths. At a pulse repetition rate (switching frequency) of 200kHz, this would require 76ps clock cycles (a 13GHz clock!). Because this is impractical (at least today), you must use a slower clock.

Typical Class-D PWM switching frequencies are between 200kHz and 500kHz. Digital PWM clocks range from 10MHz to 100MHz. For example, with a 50MHz clock and 200kHz PWM, you get 250 available pulse widths. This is less than 8-bit resolution!

The trick here is to exploit the fact that the switching frequency is much higher than the intended frequency

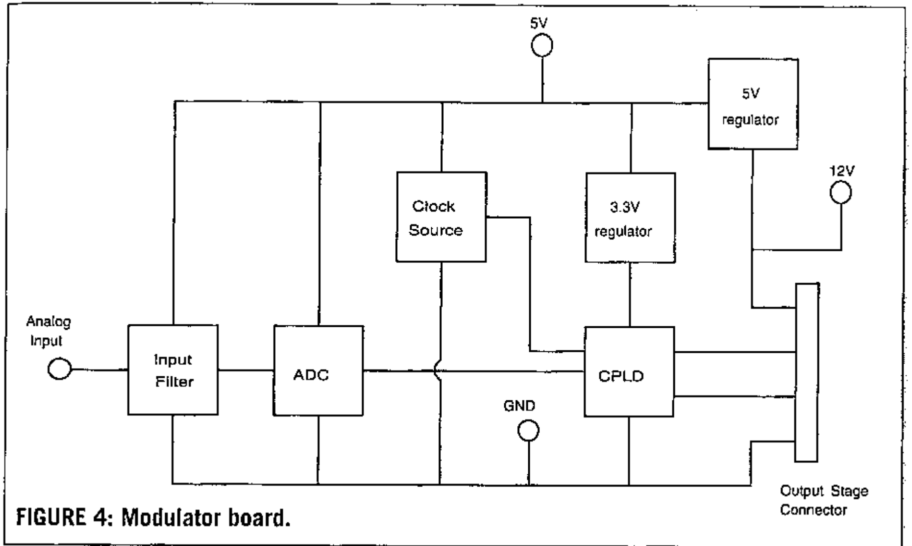


FIGURE 4: Modulator board.

you are reproducing. By selecting several different pulse widths in a repeating pattern, you can produce average pulse width values between the actual ones. For example, if the pulse width had 100 allowed positions (from 0% to 100%), and an average of 50.5% pulse width is desired, the output pulse width would alternate between 50% and 51%. The algorithms that produce the pulse widths, effectively increasing the resolution, vary in complexity and performance.

RFI CONCERNS

Like any switching amplifier, Class-D generates high voltage and current skews during output state transitions. This in turn produces RFI (radio-frequency-interference) that must be contained. In order to keep the Class-D amp from radiating this interference, it is important to enclose the output stage in a shield.

Typically, this shield is the amplifier enclosure, which contains not only the output stage(s), but the entire amplifier system. Non-metallic enclosures (for example, speaker cabinets) require the output stage to be in a shielded "box" with ventilation to allow heat to escape. The simplest type of enclosure for a Class-D amp is an aluminum (or other metal) box with evenly spaced vent holes.

Typical hole spacing is ½", and typical diameter is ⅛". The enclosure needs vent holes to provide some convection airflow while containing emissions. Good PCB layout is critical, not only to minimize transmitted RF, but also to ensure well-controlled power output and low output impedance.

Analog audio is still prominent. In some digital audio systems, there is no access to the digital data, and "legacy" audio equipment outputs only analog audio. Existing analog equipment can drive a digital Class-D amp if an analog-to-digital converter (ADC) is used to derive digital data from the input.

BRIDGED OUTPUT

By "bridging" the output of an amplifier, you can obtain nearly twice the voltage across the load, resulting in roughly four times the output power. A dual supply (positive and negative voltages) is not necessary to run in this configuration. Typical Class-AB amps are half-bridge, dual supply type.

Another consideration with Class-D amps that makes bridging popular is that non-bridged output stages suffer from the power-supply "pump" effect. During output state transitions, the output filter recoils from being driven at a high voltage when the drive is released (FET is turned off). After you turn the power supply switch (FET) off, flyback current from the inductor pumps the capacitor of the opposite side through the body-diodes of the FETs. This moves the power supplies up and down, distorting the output.

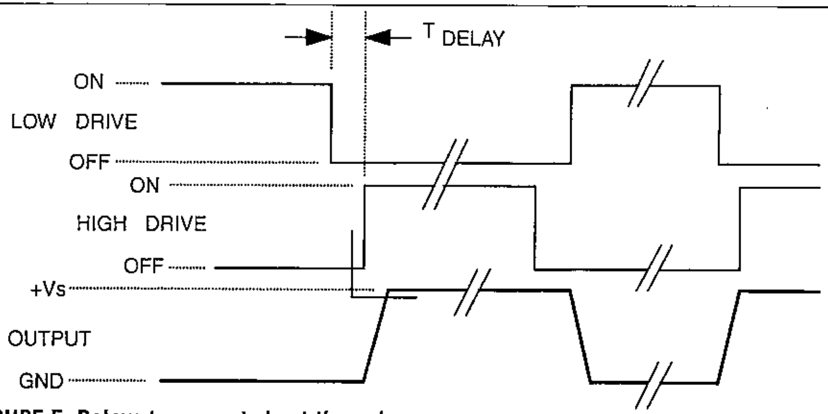


FIGURE 5: Delays to prevent shoot-through.

In a bridged configuration (*Fig. 1*), the current pump from the output filter can be applied to only one supply reservoir (capacitor), so the pump effect doesn't occur. The full-bridge configuration (*Fig. 2*) has two sides, the A-side and the B-side. The A-side switching output inverts with respect to the B-side switching output.

In a Class-D amp, the power-supply voltage determines the gain of the amplifier. It is possible to vary the power-supply voltage to control "volume." This works only if the system is open-loop (no feedback) because feedback corrects most of the gain difference.

SA1 SUBWOOFER AMP

The SA1 amp consists of a modulator board and an output stage board. The separation of these two sections allows future upgrades, including full-range operation or more power, without replacing the entire system. For example, a stereo system would consist of a stereo modulator board and two output stage boards. Separation of the modula-

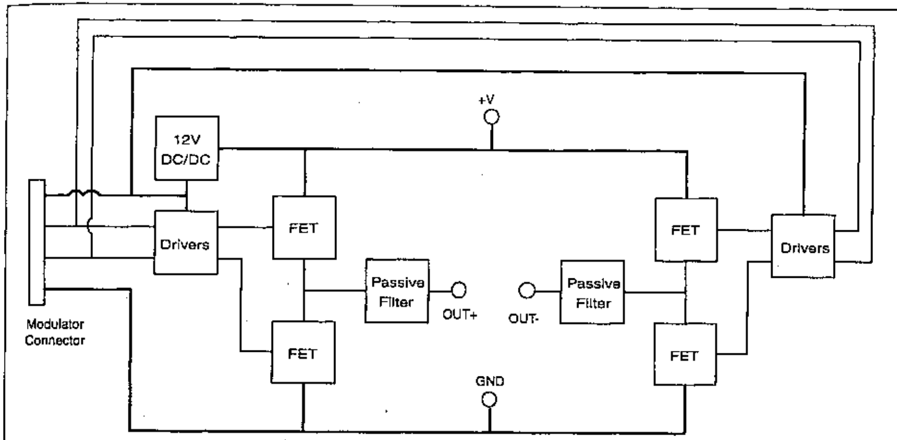


FIGURE 6: Output stage board.

tor and output stage makes it easy to shield the output stage by itself.

The SA1 runs from an analog input, but is a digital Class-D amplifier. An ADC converts the analog input to digital; this configuration is compatible with current surround-sound systems, which require a powered subwoofer and have a line-level analog output for that purpose. The system consists of two boards and a power supply. The ADC and complex programmable logic device (CPLD) are on the modulator board, and the output stage is on its own board (*Fig. 3*).

MODULATOR BOARD

The modulator board (*Photo 1*) converts the analog input to control signals used by the power circuits (*Fig. 4*). The heart of the system is a CPLD, which functions as a signal processor and controller. The controller portion runs the ADC and is used to shut down the output stage in the absence of audio (see next paragraph). The audio input to the CPLD is I^2S type (common interface for ADCs and DACs), and the output of the CPLD is PWM.

AUTO SHUTDOWN

Second, in digital Class-D, the audio level of the input is measured digitally, eliminating the need for a level detector (comparator). Third, the timer used to measure how long audio has been below the threshold is simply a counter in the digital domain. These advantages eliminate circuitry and increase reliability in the case of a digital Class-D amp.

DEAD-TIME DELAYS

PWM, generated internally by the CPLD, drives the output stage with two signals, high-drive and low-drive. This is done to prevent shoot-through, which is the condition where current flows through both FETs in series, effectively shorting out the power supply (*Fig. 5*). Shoot-through is a concern because the FETs and FET drivers have inherent propagation delays and limited slew rate. Delays in the drive allow one FET to turn fully off before another in series is turned on.

OUTPUT BOARD

The output board (*Photo 2*) converts the low-level drive signals from the CPLD to an output powerful enough to drive a speaker. It drives the load through a passive filter used to remove the switching frequency. High-current inductors are used in the filter to reduce series resistance. FETs function as switches between the power supply and the filter (*Fig. 6*).

The circuit accepts 3.3V-level signals and converts them to 12V-level signals using a FET driver. The 12V-level signals drive a "half bridge FET driver," which in turn drives the FET gates through small resistors. The FETs drive the passive output filter in a full-bridge configuration.

Digital Class-D amplifiers can be made full range. In order to reproduce

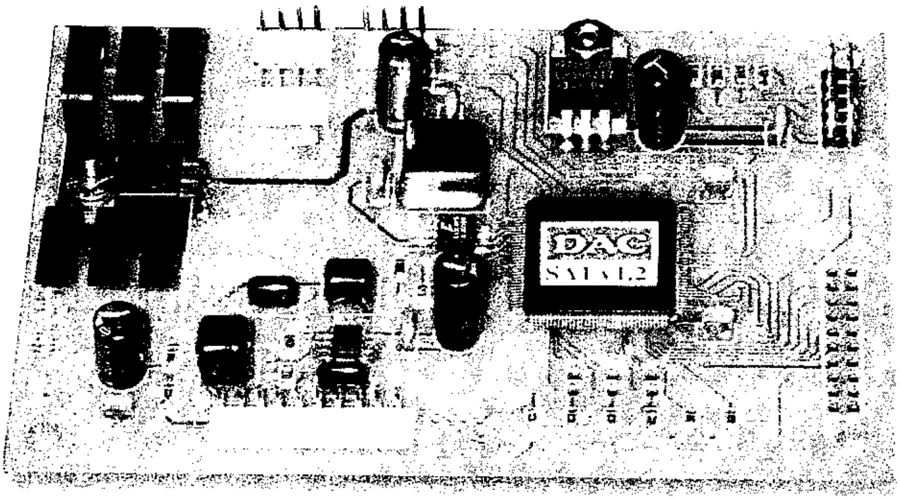


PHOTO 2: Output stage board.

SA1 SPECIFICATIONS

Rated power output: 115W into 4Ω (1% THD+N @ 100Hz)

Instantaneous peak power: 230W into 4Ω

THD+N: <0.1% up to 50W @ 100Hz

Signal-to-noise ratio: 90dB

Dynamic range: 90dB

Intended load: 4Ω

Frequency response: 20Hz to 1kHz ±1dB

Auto standby: 1 minute, 25 seconds after input is <30mV RMS

PWM switching frequency: 384kHz

Master clock: 49.152MHz

REFERENCES

1. Alan Bayko, "Build a 200-watt Digital Power Amplifier," *Electronics Now* 4/97, pp. 33-44.

2. Michael Fremer, "Sharp SM-SX100 Digital Integrated Amplifier," *Stereophile*, 7/00, pp. 73-79.

3. Bascom H. King, "Tact Audio Millennium Digital Amplifier," *Audio*, 6/99, pp. 52-57.

higher frequencies than the subwoofer range (>1kHz) with low distortion, additional digital signal filtering is needed. This additional filtering is not included in the SA1, which can be used full range if the analog input filter is taken out of the circuit, but the amp may sound harsh at high frequencies.

In Part 2 we'll continue with a look at digital amps in general, and the SA1, in particular. ❖

Digital Class-D Subwoofer Amp, Part 2

Part 1 introduced readers to the specifications of the SA1 digital amplifier kit. We continue here with a look at the theory behind the company's design. **By Thomas O'Brien**

Efficiency is power in (from the power supply) divided by power out (into the load). Because of switching losses, which heat up the FETs, the output stage wastes some power to heat. As power-supply voltage increases, so does heat dissipation. The equations that calculate efficiency are simplified because there are many parasitic losses and complex component interactions that are ultimately negligible. Efficiency is a number from 0 to 1 (0% to 100%).

Efficiency of the output stage is:

$$n_{out} = \frac{1}{\left(\frac{V_S I_{SW}}{P_O} + \frac{R_T}{R_L}\right)}$$

EFFICIENCY OF THE SYSTEM

Powering the modulator circuits and FET drivers reduces efficiency. In the SA1, +12V is used for these circuits, and requires 200mA, or 2.4W. The 2.4W comes from a DC/DC converter that is 75% efficient, so the DC/DC converter requires 3.2W. Even if the output stage were 100% efficient, at 3.2W output into the load, the amplifier as a system would be only 50% efficient. In this example, half the power is wasted in the modulator.

Fortunately, as the output power rises, so does efficiency. At 100W output and 90% output-stage efficiency, the output stage (not including the FET drivers) dissipates 11W (111W × 90% efficiency = 100W out). If the other circuits in the amp dissipate 3.2W, total dissipation is 100W + 11W + 3.2W = 114.2W (87% efficiency).

Efficiency improves as output power approaches maximum. As the amplifier goes into clipping, the average output spends more time in the maximum and minimum pulse-width areas, where the output stage is most efficient.

Consider the difference in the size of the heatsink for a 90%-efficient Class-D amplifier versus a 45%-efficient Class-AB amplifier. The heat dissipated in the 90%-efficient amp is 10% of the output power, and 55% in the 45%-efficient amp. The Class-AB amp requires 5.5 times the heatsinking!

Efficiency of the complete amp system is:

$$n_{SYS} = \frac{1}{\left(\frac{V_S I_{IDLE}}{P_O} + \frac{R_T}{R_L}\right)}$$

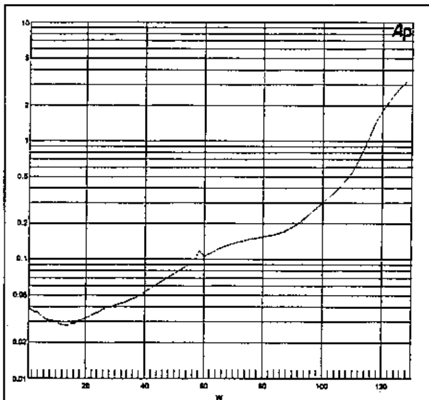


FIGURE 7: THD versus power (input = 100Hz). All figures use load = 4Ω, with Astrodyne power supply.

Idle current of the amp is:

$$I_{IDLE} = I_{12V} + I_{SW}$$

I_{12V} is the current used by the 12V DC/DC converter. DC/DC conversion from the main supply voltage (36V in the SA1) to 12V is about 75% efficient. For example, if the 12V supply provides 200mA to the FET drivers and modulator circuits, the DC/DC converter would require 89mA from 36V. The calculation used to figure the supply current for the system can also be applied to DC/DC converters (I_S, later in this article).

The 5V supply is linearly regulated from the 12V supply. This is simpler, but less efficient, than using an additional DC/DC converter, and is done to drive the ADC with a low-noise power supply. The 3.3V supply is linearly regulated from the 5V supply, but the voltage drop (only 1.7V) is small enough that efficiency of this conversion is not an issue.

POWER-SUPPLY CONFIGURATION OPTIONS

36V was chosen to drive 4Ω with more than 100W. The recommended supply is regulated, current-limited, switching-power supply. You can use

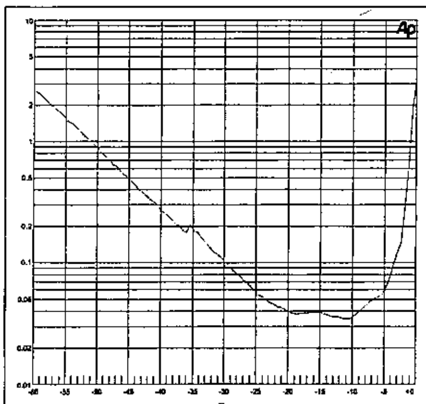


FIGURE 8: THD versus input amplitude (input = 100Hz).

RECOMMENDED POWER SUPPLY

The recommended power supply for SA1 is from Astrodyne.

ASTRODYNE LPP-150-36 (+36V @ 150W)

a linear supply, but it should be regulated to maintain good audio performance. A linear supply may be less expensive, but takes more space, is heavier, and requires custom circuitry to regulate.

If you use a separate 12V supply, the 12V DC/DC converter circuit must be bypassed. See the SA1 kit for details on this modification.

If you use a main supply of more than 40V, the 12V DC/DC converter should not be used because it has a maximum input voltage of 40V.

If lower power output is required, and no external 12V supply exists, you can use a main power supply of as little as 18V. Another option is to use a 12V-only supply, connect the main supply to the 12V, and bypass the DC/DC converter, which has a minimum input voltage of 18V.

POWER-SUPPLY CURRENT LIMITERS

Although switching power supplies are more expensive than linear ones, their outputs are likely to be regulated and current-limited. Regulation ensures that the voltage won't "dip" with increasing load current, and that the 120Hz ripple found in linear supplies is not present. The current limiter in these supplies causes the output voltage either to disconnect momentarily or to dip, keeping the output current below a set value. This protects the power supply and amplifier from damage due to short circuiting the outputs or overdriving the audio input.

You can build over-current protection into the amplifier, but this adds to the series resistance of the output and increases the complexity of the output stage. It was omitted from the SA1 because the recommended power supply is current limited.

It is important that the power supply doesn't use a current-limit circuit that shuts down the supply until AC power is cycled. Suppose that you're watching a DVD that includes an explosive scene. Along comes a loud sound, and the subwoofer turns off. Not only that, but there is no power switch, so you must unplug the unit and then plug it in again to get it working!

POWER-SUPPLY CIRCUIT

The power-supply current required to run the SA1 is not calculated in the same way as for a Class-AB amp. Usually, the maximum current from the power supply is the same, for example, as the peak current through the load at the top (or bottom) of a sine wave driving the amplifier at 1kHz. However, take into account the efficiency of the amplifier, so the power in is nearly the same as the power out. Since the efficiency is high and the voltage driving the amplifier is higher than the output voltage, the required supply current is lower than the output current.

The peak current into the load is:

$$I_P = \sqrt{\frac{2P_O}{R_L}}$$

and the power-supply current at maximum output power is:

$$I_S = \frac{P_O}{nV_S}$$

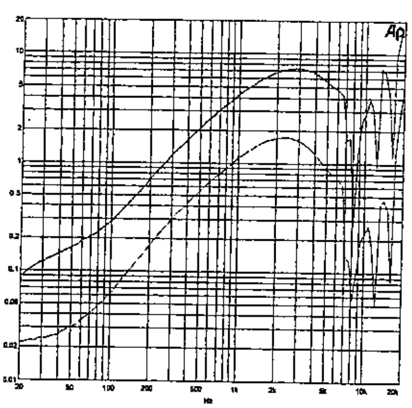


FIGURE 9: THD versus frequency (output = 50W, 100W).

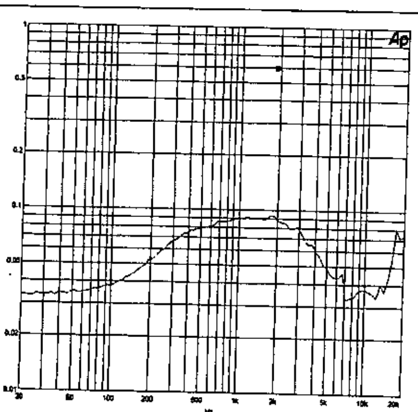


FIGURE 10: THD versus frequency (input = –20dB).

MAXIMUM OUTPUT POWER

The system is designed to run from 36V, but a supply as low as 18V is permissible. The maximum power output before clipping in a bridged configuration Class-D amplifier is:

$$P_O = \frac{PW_{RANGE}^2 V_S^2 R_L}{2R_T^2}.$$

PW_{RANGE} is the range of pulse widths from the highest output to the lowest. In the case of SA1, the pulse-width range is ⅞ (limited digitally to exclude the top and bottom 16th). During peak output voltage, the maximum pulse width is 15⁄16, and the A-side of the bridge operates at 15⁄16 pulse width. Meanwhile, the B-side operates at 1⁄16 pulse width, so 14⁄16 of the supply voltage is across the load. The pulse width is limited to ensure controlled operation of the output stage during peak output voltage.

R_L is the load resistance, which is 4Ω. R_T is the total resistance through the FETs, output filter inductors, and load. R_{FET} is R_{ON} (series resistance) of the FET when it is fully turned on. The IRFZ44 FET's R_{ON} is approximately 0.03Ω, and the inductors are about 0.02Ω each (R_{DC}).

The total resistance around the output loop is:

$$R_T = R_L + 2R_{FET} + 4R_{IND} = 4.14.$$

This translates to 0.14Ω output resistance, and the maximum output power of SA1 into 4Ω is:

$$P_O = 0.09V_{SUPPLY}^2.$$

With a 36V supply, the maximum output power into 4Ω is more than 115W.

The LT1776 DC/DC converter has a maximum input voltage of 40V. To provide a higher supply voltage for the output section, you must use a separate +12V supply, and the LT1776 circuit must not be populated. With a 48V supply, the SA1 can output 207W into 4Ω.

POWER OUTPUT AND CLIPPING

The amplifier can output more than the so-called "maximum output power" because this is defined by where the amplifier is at 1% distortion (THD+N). This is where the amplifier starts being overdriven, and the output waveform looks as though the top (and bottom) of the signal is flattened out. If the input signal (sine wave) is large enough, the output will look more like a square wave than a sine wave. In this case, the distortion is high, and the limit of the power to the load is based on the maximum current output of the power supply.

The output devices are stressed during clipping, resulting in extra heat. The audio quality is poor when the amplifier is clipping, but if the power supply is current-limited, clipping should not damage the amp. Power into the load during severe clipping approaches instantaneous peak power.

MORE POWER

IRFZ44 FETs can handle a maximum of 60V, and the FET drivers can handle 125V. Why, then, can't you drive the SA1 with 60V, for example, to deliver 324W into 4Ω? Aside from the expense of the required power supply, the three basic reasons are overshoot, current requirements, and heat dissipation.

You can minimize overshoot on the output by slowing the output transitions (or by using additional circuitry), but you pay a penalty in distortion performance if the transitions aren't fast enough. Either way, a rule of thumb is to stay within 10% of the FET's rated voltage to allow for some overshoot.

A power-supply voltage greater than 48V is not recommended for the SA1. Suppose the efficiency of the output section is 90% into a 4Ω load. It is safe to surmise that most of the heat dissipated will be from the FETs, not the inductors. For example, also supposing that the output power is 300W using a 58V supply, then about 10% of that power (30W) would be dissipated through the FETs. Since the dissipation is distributed evenly among the four FETs, this is 7.5W per FET.

To be safe, your design should handle twice the required dissipation (or close to it). This results in additional heatsinking requirements as the power-supply voltage increases. At 207W using a 48V supply, each FET dissipates about 5W, and a 10W heatsink can be quite large (one for each FET, too). At 115W using a 36V supply, each FET is below 3W dissipation, easily provided by a slide-on "5W" heatsink.

The output inductors are current-limited as well, and can overheat and fail if they are pushed too hard. The inductors used in SA1 are rated at 7.2A RMS and 21A DC. The DC maximum current specification (A DC) of the inductors should be greater than I_P, and the AC maximum (in A RMS) should be greater than 70% of I_P.

INSTANTANEOUS PEAK POWER

A common marketing trick of amplifier makers is to claim huge "peak power" numbers. Actually, instantaneous peak power is twice the output power, unless the output is not a sine wave, or is clipped. At the top and bottom of a sine wave output, the "instantaneous" power is the output voltage divided by the load resistance. The large capacitors in an amplifier's power supply provide the momentarily higher current.

DRIVING 8Ω

Driving 4Ω is more difficult than driving 8Ω because of the series resistance of the FETs and inductors. However, 4Ω was chosen as the ideal load for the SA1 because you can deliver almost twice the power into 4Ω versus 8Ω (with the same power-supply voltage). 4Ω is a common subwoofer impedance, and maximum output power is usually first priority.

The output filter is a tuned circuit designed to drive a 4Ω load. To drive an 8Ω load, the filter capacitors must be half the value. This is described in the SA1 kit.

The maximum output power of SA1 into 8Ω is:

$$P_{OUT} = 0.046V_{SUPPLY}^2$$

SA1 KITS

The SA1 documentation kit (available from Digital Amplifier Company, 1 Turret Drive, Limerick, PA 19468) includes schematics, layout information, full specifications, and hardware description. The full SA1 kit includes the documentation kit, one modulator board, and one output-stage board. The configuration of the hardware drives 4Ω at 115W, and is for use with a single 36V supply.

CONCLUSION

The SA1 subwoofer amplifier delivers high power with low distortion using digital Class-D technology. The modular architecture allows future configuration changes, and flexible mounting in (or on) a subwoofer. The power-supply type and voltage can be varied to produce a range of power levels. High efficiency means compact size and almost no heatsinking. ❖

If you are interested in receiving schematics and tables to build Thomas O'Brien's amp, please send a large, self-addressed envelope with a loose stamp or postal coupon to Audio Amateur Corporation, PO Box 876, Peterborough, NH 03458, or visit our website at www.audioXpress.com