Check out amplifier headroom & music power

Tone burst source for amplifier testing

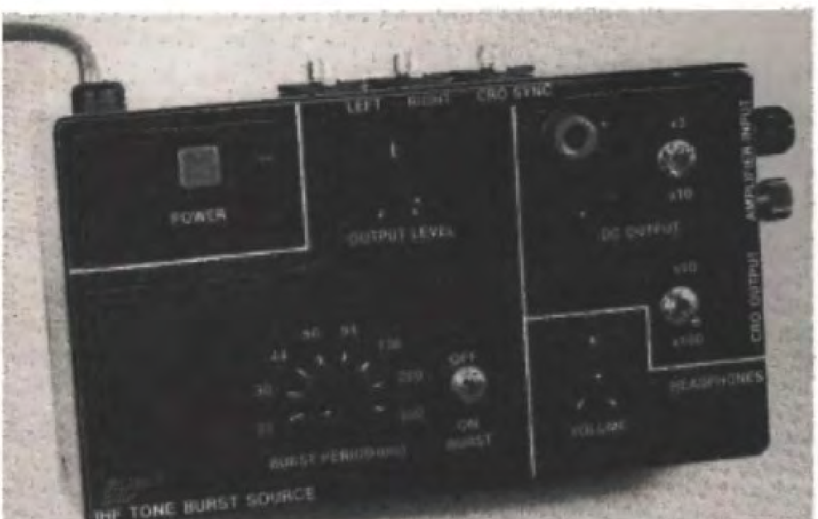
This self-contained test instrument generates the required signals to measure amplifier headroom and music power. It only needs a multimeter to read out the result.

By LEO SIMPSON & JOHN CLARKE

Whether or not you regard Music Power and Dynamic Headroom as a joke, hifi equipment manufacturers are quoting music power and headroom figures these days. This means means that these parameters should be confirmed when appraising a hifi amplifier. At SILICON CHIP we regularly test commercial amplifiers and our own designs so we wanted to have our own test set-up.

Trouble is, we know of no commercial equipment that will do the tests.

We also wanted to be able to go further than the standard IHF Music Power tests. We wanted to do longer pulse testing as advocated by manufacturers such as NAD. So what do you when there's no instrument available? You design and make your own — which is just what we did.



This strange-looking instrument provides all the facilities needed to perform measurements of music power, dynamic headroom and amplifier overload recovery time.

We are presenting the resulting design here, not because we think the design will be produced in large numbers (although there are many organisations which could use such an instrument), but because it features a number of interesting circuit techniques.

Defining terms

Alright, what are the required test signals for the IHF Dynamic Headroom and Music Power? Well, the Dynamic Headroom of a power amplifier is the ratio of the Music Power to the Continuous Power expressed in decibels. So to measure headroom you first have to measure music power. The required signal conditions are set out exactly in the specification IHF-A-202 1978, as published by The Institute of High Fidelity, Inc, USA.

While it is common to refer to the signal as being a tone burst, in reality it is a continuous 1kHz sinewave which increases in level by 20dB for a duration of 20 milliseconds, twice a second. Or to put it another way, the amplitude of the 1kHz sinewave is modulated by a pulse waveform with an "on" period of 20 milliseconds and an "off" period of 480 milliseconds. Further, the transition in level is required to occur at the zero voltage crossing of the sinewave signal.

The problem with such a pulse signal is that it is difficult display on most oscilloscopes (because of its very short pulse duty cycle). Second, once the signal is displayed, it is quite difficult to judge when the onset of clipping occurs (which is the level at which the music power is measured). Then, once the exact onset of clipping is established you

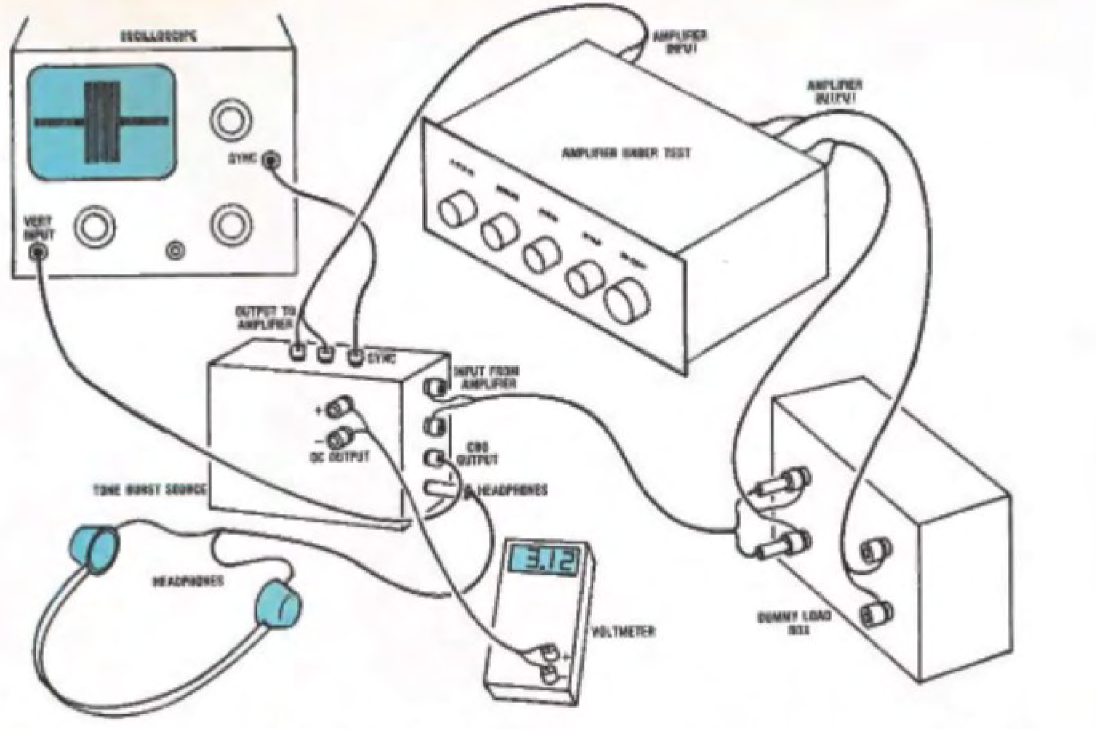


Fig.1: this diagram shows how the Tone Burst Source is connected to a stereo amplifier. The DC reading on the voltmeter is squared and divided by the load resistance to give the amplifier's music power output.

have to measure the peak-to-peak voltage displayed on the oscilloscope screen. This is difficult to do with any real accuracy since, for a high power amplifier, say with 300 watts music power, the waveform may be 140 volts peak-to-peak or more.

Once the peak-to-peak voltage is known, it needs to be divided by a factor of 2.828 to convert it to RMS value, then squared and divided by the load resistance to find the music power output.

Clearly then, measuring music power output of an amplifier is a fiddly business. Which is why most magazines reviewing high fidelity equipment don't bother to do it. (Either that, or they don't know how!)

And when you want to measure with even longer pulse durations, say up to 300 milliseconds, you really need a storage oscilloscope to do the job.

What we wanted was a self-contained instrument which would generate the required modulated sinewave signal, measure the power amplifier's output signal and convert it to a DC voltage which can be measured by any voltmeter, digital or analog. The voltage reading is then squared and divided by the load resistance to get the music power output. The result is our IHF Tone Burst Source.

Features

The IHF Tone Burst Source has a sinewave output which can be switched for continuous or modulated output, in eight burst lengths of 20, 30, 44, 66, 94, 136 and 200 milliseconds. The duty cycle for all burst lengths is fixed at 24:1.

The pulsed or continuous sinewave signal is variable in output level from zero to 2 volts RMS. So that the output waveform of the amplifier under test can be displayed on an oscilloscope, a 15V square wave sync pulse is provided.

Fig.1 shows how an amplifier would be connected to the Tone Burst Source to perform a measurement of Music Power. The oscilloscope is desirable but not absolutely mandatory for the test, as we shall see later.

Besides the oscilloscope and Tone Burst Source, you need a DC voltmeter (which can be a digital or analog multimeter) and a switchable dummy load which can be set to provide the rated load resistances for the amplifier under test. For most amplifiers, this means that the dummy load will have to provide 4 and 8-ohm loads at the very least, as well as 2-ohm loads for more stringently rated amplifiers. Naturally, the resistive loads need power ratings to cope with the amplifier's full output power.

Depending on which channel of the amplifier is being monitored, the left or right channel output is connected to a pair of binding post terminals on the Tone Burst Source, as well as to the dummy load.

PARTS LIST

1 PCB, code SC4-1-488, 130 x 103mm
1 Scotchcal front panel, 195 x 110mm
1 plastic utility case, 198 x 113 x 60mm (Altronics Cat No H-0102 or equivalent)
1 15V or 12.6V mains transformer (see text)
1 push-on/push-off mains switch
3 DPDT miniature toggle switches
1 single-pole 8-position rotary switch (Jaycar SR-1210 or equivalent)
1 stereo 6.5mm jack socket
4 banana jack sockets (two red, two black)
4 panel-mount RCA sockets
1 mains cord and plug
1 cordgrip grommet
3 knobs
1 5mm LED and bezel
4 6mm standoffs
1 solder lug
4 rubber feet

Semiconductors
2 555 timers
1 4013 dual D flipflop
1 4066 quad analog switch
1 TL072 dual JFET op amp
1 CA3130 op amp
1 7815 3-terminal regulator
5 1N4002 diodes
1 1N4148, 1N914 diode

Capacitors
1 470μF 25VW PC electrolytic
1 47μF 16VW PC electrolytic
1 15μF 16VW PC electrolytic
3 10μF 16VW PC electrolytic
1 6.8μF 16VW PC electrolytic
1 4.7μF 16VW PC electrolytic
1 3.3μF 16VW PC electrolytic
2 2.2μF 16VW PC electrolytic
1 1.5μF 16VW PC electrolytic
3 1μF 16VW PC electrolytic
4 0.033μF metalised polyester
1 0.01μF metallised polyester
1 0.001μF metalised polyester
1 22pF ceramic

Resistors (0.25W, 5%)
1 x 1.5MΩ, 1 x 620kΩ, 2 x 470kΩ, 3 x 220kΩ, 1 x 180kΩ, 1 x 100kΩ, 1 x 68kΩ, 1 x 47kΩ, 1 x 39kΩ, 1 x 27kΩ, 1 x 24kΩ, 1 x 20kΩ, 2 x 18kΩ 1W, 1 x 15kΩ, 3 x 10kΩ, 4 x 4.7kΩ, 1 x 3.3kΩ 1W, 1 x 1.8kΩ, 1 x 1kΩ, 1 x 22kΩ miniature vertical trimpot, 1 x 5kΩ linear potentiometer, 1 x 1kΩ log potentiometer

Miscellaneous
Hookup wire, rainbow cable, insulating tubing, solder, screws, nuts etc.

Inside the Tone Burst Source, the amplifier's output signal is fed to the internal signal monitoring circuitry and to an RCA socket for oscilloscope monitoring. This signal is divided down from the amplifier signals by a factor of 10 or 100, to enable the oscilloscope to correctly display the signal. We have provided this order of signal division because we are assuming that the signal will be connected directly to the CRO instead of via a 10:1 divider probe and because most CROs have a minimum (calibrated) sensitivity of only 5 volts/div.

In addition, the amplifier output signal is fed to a headphone socket which has an associated volume control. With practice, you can judge the onset of clipping "by ear" to within one or two percent. This means that it is possible to dispense with the oscilloscope although, ideally, you should have it anyway to do the test properly.

The DC output voltage from the Tone Burst Source is calibrated to provide the DC equivalent of the RMS voltage for two ranges. These are the x3 and x10 ranges. To calculate the IHF power the DC voltage is multiplied by the range, then squared and divided by the load resistance for the amplifier:

$$P_{IHF} = (V \times Range)^2/R_L$$

The DC output is also available when the tone burst is disabled so that continuous RMS power can be measured at 1kHz.

Using it

Having connected the Tone Burst Source as shown in Fig.1, the test for Music Power is straightforward. Feed the burst signal to the amplifier and display the output waveform on the oscilloscope screen. Increase the signal amplitude until the waveform just begins to flatten at the peaks (this is the clipping point) and then back off slightly to obtain an undistorted waveform.

Take the meter reading and do the power calculation as outlined above. It's as easy as that.

One other test which this instrument can perform is the overload recovery time of an amplifier. The same signal is used but this time it is set so that the amplifier is overloaded by 10dB during the burst times. Most amplifiers will then take a number of cycles (at 1kHz) to recover their equilibium. The overload recovery time is then measured simply by counting the number of cycles at 1kHz so the time can be quoted in milliseconds.

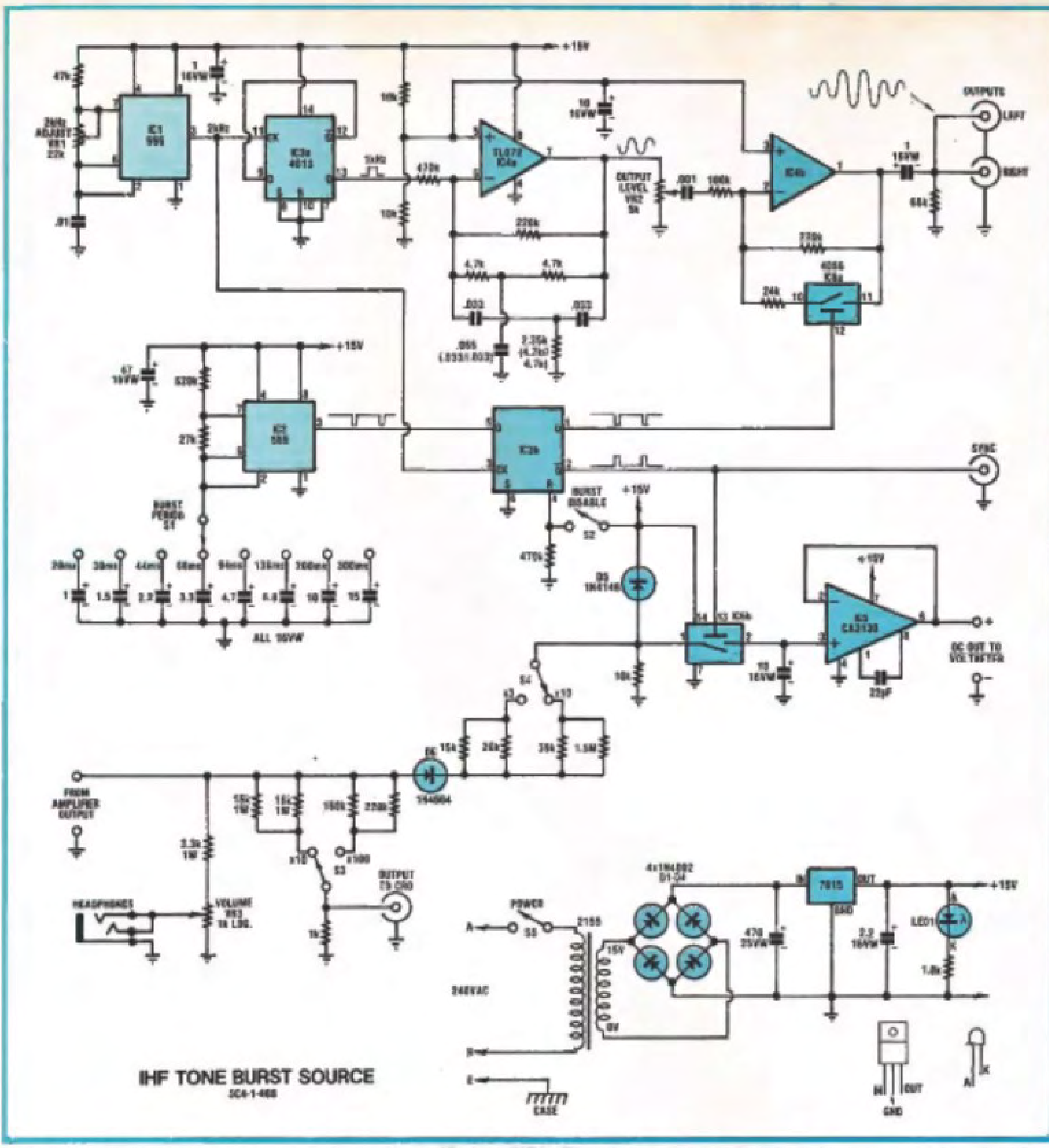
Circuit features

The circuitry for the Tone Burst Source consists of two 555 timers, one 4013 dual D flipflop, one single and one dual op amp package, one 4066 analog gate package plus assorted diodes, passive components and the power supply components.

First of all, the circuit is required to generate a clean stable sinewave at precisely 1kHz. We do this in an unconventional way, using 555 timer IC1, D-type flipflop IC3a, and twin-T filter stage IC4a.

IC1 is set to produce a 2kHz square wave. This is fed to D-type flipflop IC3a to produce a 1kHz square wave with a duty cycle of exactly 50%. The square wave is then fed to the twin-T filter stage which then produces a clean sinewave with harmonic distortion of less then 2%. That might not sound like a particularly low distortion value but it is low enough for this purpose.

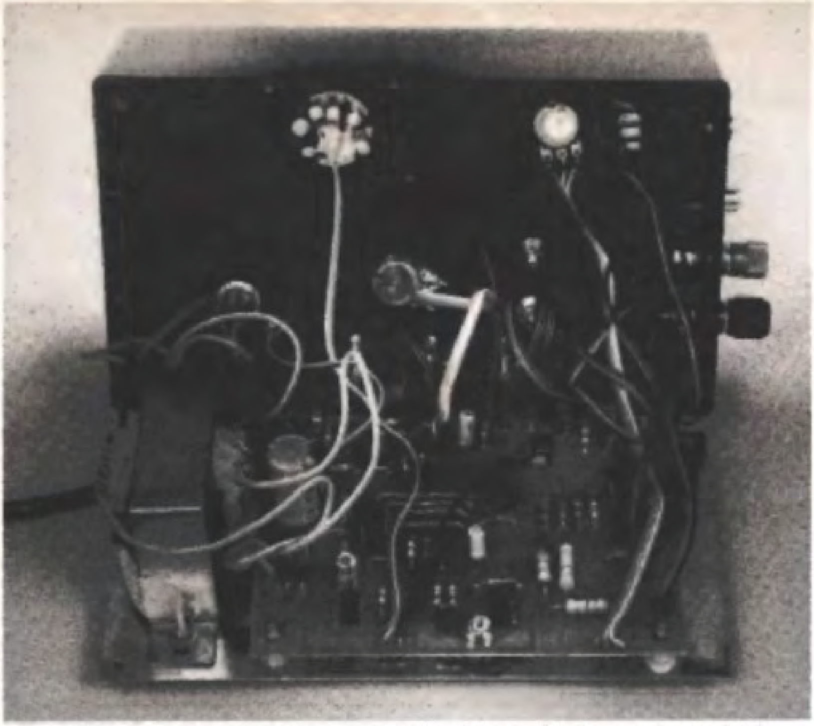
The output signal from IC4a is then fed to output level control VR2 and then to output amplifier IC4b. The amplifier either feeds the signal straight through with fixed gain, as for the continuous signal mode, or with gain switched between two levels, for the burst mode.



The circuit of Tone Burst Source is essentially a pulse modulated oscillator with selectable pulse times, combined with an AC rectifier (D6) and a sample-and-hold circuit (IC6b, IC5) to produce the DC output signal for the external voltmeter. Amplifier monitoring is via headphones or oscilloscope.

In the continuous mode, the gain of IC4b is fixed at 2.2 by the associated 100kΩ and 220kΩ resistors. In the burst mode, the gain is switched between 2.2 and 0.22 by analog switch IC6a. For 480 milliseconds out of each 0.5 second, IC6a shunts the 220kΩ feedback resistor with a 24kΩ resistor and thereby reduces the gain of IC4b to 0.22. Thus the required step up in gain of ten times (+20dB) is obtained whenever analog switch IC6a closes, in reponse to a pulse signal.

The required pulse signal is generated by IC2, another 555 timer which generates a train of pulses with the required 24:1 duty cycle. This is determined by the 620kΩ and 27kΩ resistors which charge and discharge the capacitor connected to pin 6 of the 555. The 24:1 duty cycle is obtained by dividing the sum of the two



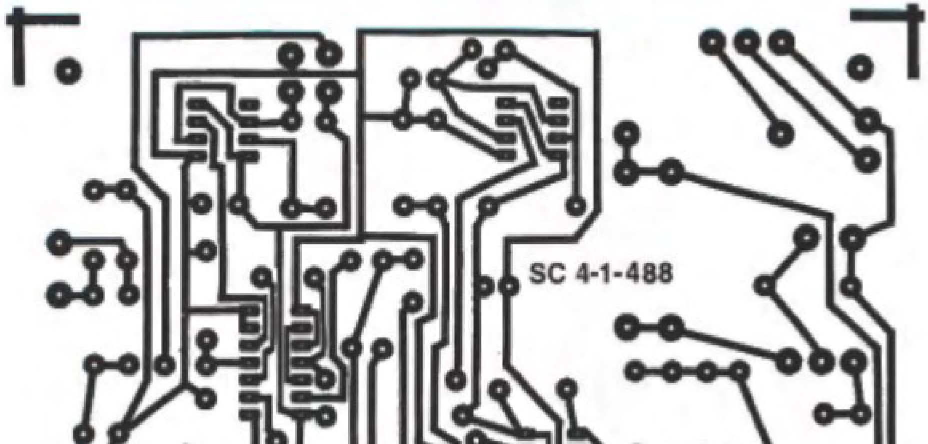
The wiring inside the Tone Burst Source is a bit of a bird's nest but is fairly straightforward if rainbow cable is used. Don't bind the wires together as this could create crosstalk problems.

resistors (ie, 620kΩ + 27kΩ) ÷ 27kΩ); this gives a result of 23.963 which is within a gnat's whisker of 24:1.

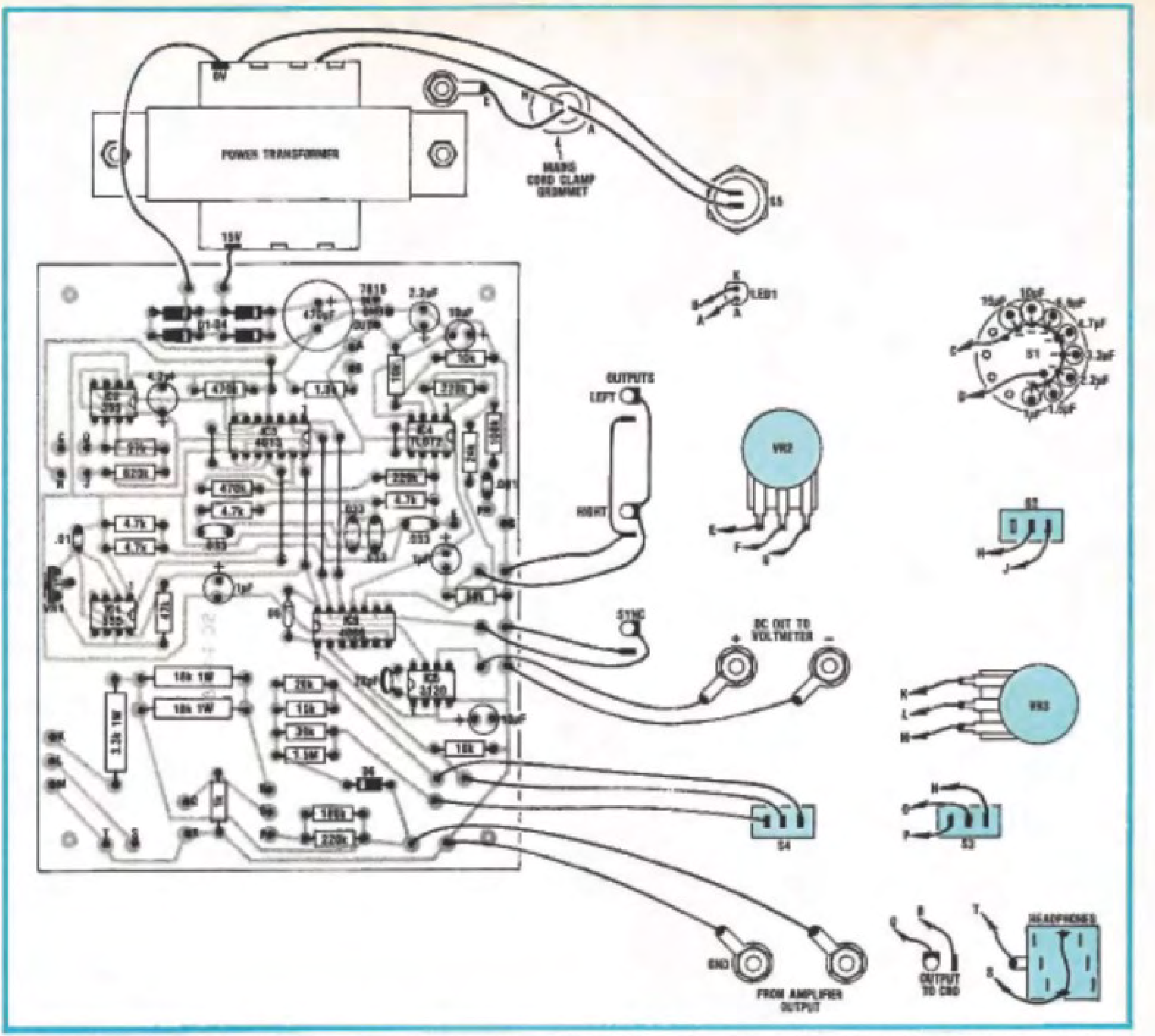
Switch S1 changes the capacitor connected to pins 6 and 2 of the 555, to change the pulse duration from a minimum of 20 milliseconds to a maximum of 300 milliseconds. Note that the duty cycle remains constant no matter what pulse length is selected. This means that for a pulse length of 300ms, the time between the end of one pulse and the beginning of the next is 24 x 300ms = 7.2 seconds.

Now the pulse signal could have been fed directly to pin 12 of IC6a to control the gain switching of IC4b but this would mean that the transitions in level would not take place at the normal "zero axis points" of the 1kHz sine wave.

To ensure that this requirement is met, D-type flipflop IC3b is used. The pulse signal from IC2 is fed to IC3b's D (data) input while the CK input is clocked with the 2kHz square wave signal from IC1. This ensures that the pulse signal fed to



This is the full-size artwork for the printed circuit board.



The wiring diagram of the Tone Burst Source uses letter coding to show the various interconnections. Note that eight capacitors are wired around switch S1. Take care with the mains wiring and use plastic sleeving to cover exposed mains connections.

pin 12 of IC6a is exactly synchronised with each positive edge of the 2kHz square wave and therefore with the zero axis points of the sinewave from IC4a.

Switch S2 disables flipflop IC3b so that its Q output is permanently low. This maintains IC6a in the open-circuit condition so that the gain of IC4b is fixed at 2.2.

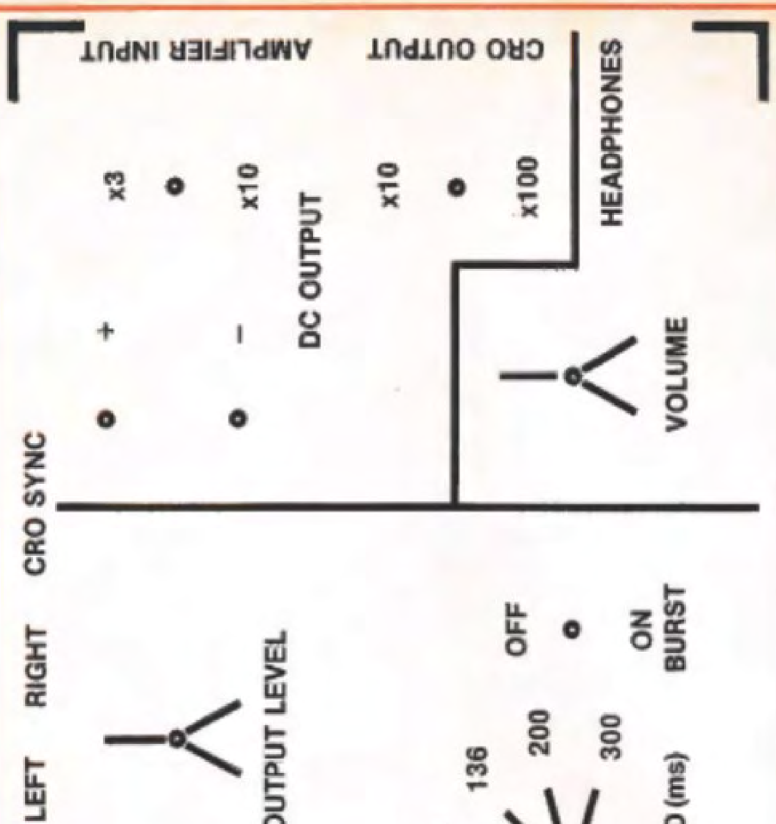
The output of IC4b is fed via a 1μF capacitor to a pair of RCA sockets, for the left and right inputs of a stereo amplifier. IC4a and IC4b are biased to half the supply voltage by a voltage divider consisting of two 10kΩ resistors bypassed with a 10μF capacitor. The half-supply output of this divider is fed to the non-inverting inputs, pins 5 and 3, of IC4a and IC4b, respectively.

The Q-bar output of flipflop IC3b provides the sync signal for an oscilloscope.

Well, the description so far has detailed how the sinewave is generated and how it is switched between two levels with a duty cycle of 24:1. Now let's have a look at how the output signal from the amplifier under test is processed.

Power amplifier output

As noted above, the (left or right channel) output from the amplifier under test is connected via attenuators to provide monitoring signals for headphones and an oscilloscope, as well as the signal to be measured and processed. VR3 provides the headphone signal, with quite a lot of attenuation in consideration of the high power levels likely to be involved, while the oscilloscope signal comes from the voltage divider associated with switch S3. This gives two division ratios, ÷10 and ÷100, so the



This is the full-size artwork for the front panel of the Tone Burst Source.

resulting display on the CRO needs to be multiplied by 10 or 100 to give the true value.

Diode D6 rectifies the signal from the amplifier and feeds it via switch S4 and its associated resistors to analog switch IC6b which, together with IC5, forms a "sample and hold" circuit. Each time the sinewave signal from IC4b increases in value by 20dB, analog switch IC6b closes and feeds the rectifier output from D6 to the 10µF capacitor at pin 3 of IC5. The 10µF capacitor stores and holds the charge until the next time IC6b closes, when it takes a new voltage sample.

IC5 is a CA3130 Mosfet input op amp arranged as a voltage follower. This op amp has an extremely high input impedance which means that the 10µF capacitor experiences virtually no loading at all from the op amp input. IC5 feeds the voltage from the 10µF capacitor to the output for the DC voltmeter.

Diode D5 is a voltage clamp which prevents input voltages to IC6b and IC5 from exceeding +15.6 volts. This means that for output voltages of more than about 12 volts or so, the attenuator switch S4 should be switched from the x3 range to the x10.

The resistor values associated with DC output switch S4 are arranged so that the DC voltage at the output terminals is equal to the RMS value of the AC voltage from the amplifier's speaker terminals, when in the continuous mode.

For example, in the continuous mode (ie, switch S2 set to disable the burst mode), if 20 volts RMS is fed into the AC input terminals and the DC output switch is set to x10, the voltage across the DC output terminals will be 2.00 volts. Similarly, for the same input conditions, if the DC output switch is set to x3, the DC output will be 6.67 volts.

In the burst mode, the DC output voltage is equivalent to the music power output voltage from the amplifier.

The circuit is powered from a small 15VAC transformer which feeds a bridge rectifier, D1 to D4, and a 470µF filter capacitor. The smoothed DC is regulated to +15 volts with a 7815 3-terminal regulator.

Construction

We housed our Tone Burst Source test set in a standard black plastic case measuring 198 x 113 x 60mm, with most of the circuit components mounted on a printed circuit board measuring 130 x 103mm (code SC4-1-488). The power transformer and the printed board is mounted on the metal lid of the plastic case. The lid then becomes the baseplate of the instrument. All the controls are then mounted on the top of the case while most of the input and output terminals are mounted on the sides.

We used a 15V 1A transformer but since the current drain of the whole circuit is low, you could save a few dollars by going to a 12.6V 150mA transformer (such as the Altronics Cat. No. MM-2006).

Assembly of the PCB is a straightforward job. Install the links, resistors and diodes first, followed by the capacitors and integrated circuits. All the connections to the board are made via lengths of ribbon cable. The capacitors for the burst period switch are wired around the switch itself.

Note that the wiring diagram shows most of the interconnecting wires with letter codes. For example, point K on the printed circuit board is joined to point K on VR3, the headphone level control.

The case can be drilled for all the controls using the Scotchcal panel as a template. Then it is simply a matter of installing all the hardware and completing the wiring.

Setting up

You will need a frequency meter or oscilloscope to set up the instrument. After applying power and checking the 15V supply rail, adjust trimpot VR1 to obtain a frequency of 2kHz from pin 3 of IC1. This done, check that 1kHz is obtained from pin 13 of IC3, then check the sinewave at pin 7 of IC4. Trimpot VR1 can be tweaked to obtain the cleanest sinewave but make sure the frequency is still close to 1kHz.

Check that the continuous and burst modes can be obtained at the appropriate settings of switch S2. Check that the burst length varies in accordance with the setting of switch S1.

Finally, the DC output at pin 6 of IC5 can be adjusted to equal the continuous RMS AC voltage (with burst disabled) from the amplifier output. This can be done by trimming the resistor values associated with switch S4. To trim the x10 setting, the 1.5MΩ resistor can be changed or shunted with higher values to obtain the correct reading. Similarly, the x3 setting can be trimmed by changing or shunting the 20kΩ resistor with higher values.