TUBES VS TRANSISTORS: A CALL FOR MORE TESTING

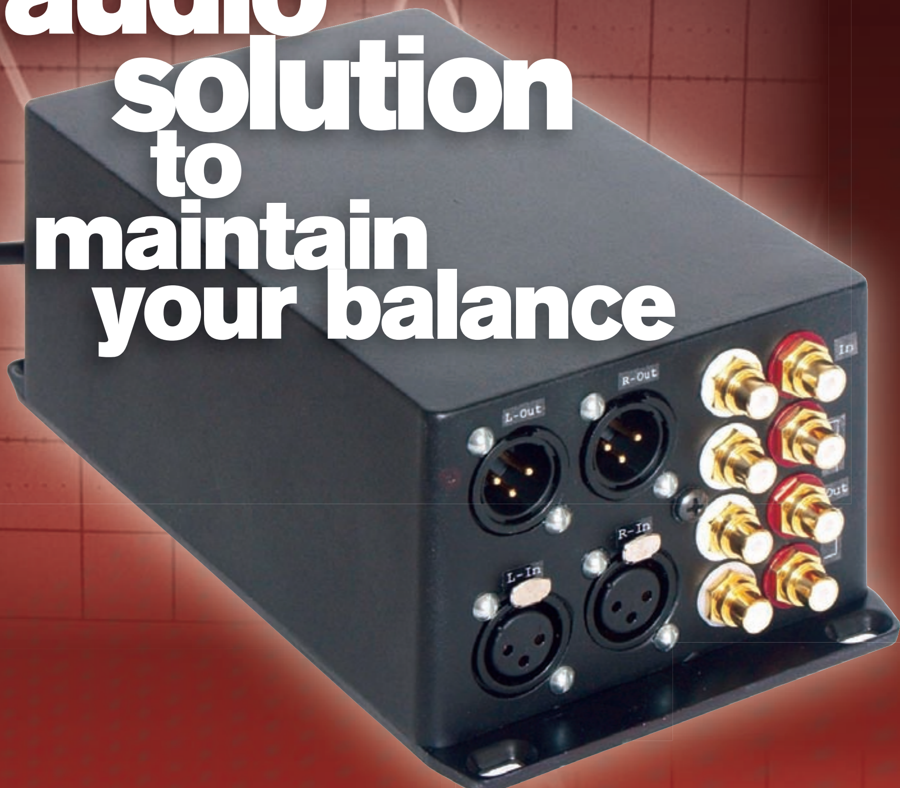
audioXPRESS

July 2009

US $7.00/Canada $10.00

Tube, Solid State, Loudspeaker Technology

audio solution to maintain your balance



TESTING TURNTABLES

Tube Amp Feedback

Building A True RMS Voltmeter

www.audioXpress.com

Amplifier Differences and Lack of Tests

Everyone is aware of the tube versus transistor, feedback, and other debates that have existed for decades. Only recently have I seen papers that actually attempt to test and explain the results with real data, such as the Nelson Pass paper on negative feedback. I believe *audioXpress* printed an article on perceived differences in harmonic structure of amplifiers. That article made a lot of sense to me; I have noticed that 1% of second harmonic is not audible to my ears, yet I seem to notice lower levels of higher order harmonics that seem common in many commercial transistor amps!

Now, I am not an audio engineer; I'm an RF Tech by trade, and maybe the AES has all kinds of data on these things and prefers to keep secrets from the common masses. In RF, we do extensive testing on devices: harmonics, intermod, behavior over varying temps and load impedance, even transient effects such as "ghosting." Transistors can do strange things under the right conditions! But try and find anything past harmonic distortion on the web if you're not an AES member. Kudos to National Semi for giving lots of hard data on their amplifier chips, but what happens when you put a reactive load on it, such as, a speaker?

A few years ago, I decided to try building a tube amp—just out of curiosity—even though there's no way it could be better than an LM3886, right? It was a cute retro project that was fun to build (just 2W), and brought back some memories. I hooked it up and played something. . . WOW! It sounded cleaner, more dynamic even. One engineer said it was just some "Euphonic" artifact. . . how can an artifact be cleaner and more detailed, more like live acoustical sound? When pressed for details, these guys can't explain why, so it's clearly not a rational argument.

Sadly, neither side seems capable of proving and explaining the differences; doesn't that seem odd? I think that performing the following tests would help, and none of these should be unusual or very time consuming:

1. Harmonic and two-tone intermod tests into a real two-way or three-way speaker. I would think the varying phase angles from a reactive load must have some effect in the feedback loop!
2. Waveform test (time domain) of transients, again, with real speaker load. I am convinced that transient peaks come through better in my tube amp. . . the LM chip sounds dull and lifeless in comparison. I can imagine a possible reason for this: the combination of low source Z of the amp with higher reactive load is providing an over-damped condition. Or I might be wrong—but I believe my ears.

One experiment I have tried is listening to clean acoustical music through both amps, then trying processed and compressed studio-produced music. The chip amp does well with compressed music, but not as well with uncompressed quality recordings. In other words, transistor amps are great for the music that kids listen to with their subwoofers. They would see no benefit with anything else. But if you know the sound of acoustic guitar or drums, high-feedback transistor amps seem to consistently fail to sound right. My tube amp does a surprisingly good job.

There are other tests that should be done. Some types of tests have been done on the human ear, but do we know what level of distortion is audible for any given harmonic, or for intermod levels? There are complications to doing this, but a well set up and documented test could be very instructive. Out of my own curiosity, I would love to do these tests, but I don't have the equipment. I thought that someone would have done this long ago.

Perhaps I am looking at this all wrong. Maybe AES knows all this stuff, but needs to cater to the current market of compressed music. (Isn't it funny how overly compressed music "sounds dynamic" to the kids? They really have no idea what the term actually means—ignorance is bliss!) And maybe the audiophile market has worked too hard at being magicians—they believe being scientific about it will destroy the magic.

Yeah, that's it. . . clearly, the audio market is even more insane than politics. I think I will just try to ignore it, and enjoy the music on my good-sounding system.

Kent Smith
Wylie, Tex.

POSTSCRIPT

As a follow-up, I just did a test with my son who plays drums, as a sanity check. Musicians know the sounds of their instruments and will notice things the rest of us don't. I used music from Sheffield Labs that is compression free. He immediately noticed that the drums sounded correct with the triode amp (no feedback).

With the LM3886 amp, the "overtone" (he described it as a sustained tone from the drum cavity) of the bass and toms was overemphasized (as in an over-damped system response). He also noted that the high-hat was muted, especially the effect you hear as the cymbals open and close. He said that both of these effects are common problems in recorded music, in spite of efforts to damp the drum overtones with absorptive materials.

I just don't believe now that all of the problem is in the recording. Further, I am sure that if I can hear this effect, it can be measured, if someone is willing to specifically test for this. aX

By Thomas M. Perazella

Maintaining Your Balance, Part 1

Building a better converter box to handle balanced/unbalanced devices.

When studying geography in grammar school, you learned that the Great Divide was the North American mountain range separating the watersheds that drain into the Pacific Ocean from those that drain into the Atlantic. Sometimes it feels as though we have a similar divide in the audio world, with the choice between balanced and unbalanced equipment interconnections.

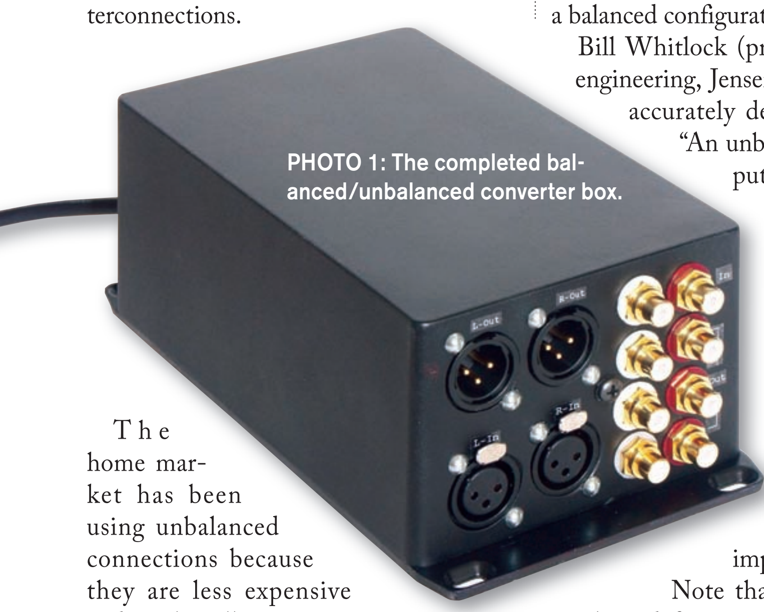


PHOTO 1: The completed balanced/unbalanced converter box.

The home market has been using unbalanced connections because they are less expensive and work well in most situations. You see an increasing number of pro-audio products with balanced connections that are reasonably priced and attractive alternates for home audio use. I use Behringer equalizer and speaker control units along with several amplifiers that all have balanced connections. They need to be integrated with the unbalanced devices that make up the rest of my system. The key is to use the proper method for those connections.

My interest in the balanced/unbalanced scenario was piqued by Gary Galo's excellent article[1] about noise in audio circuits. It was based on information from a seminar that he attended given by Bill Whitlock. This article, a concise description of noise problems and their solutions, mentioned devices from IC-manufacturer THAT Corporation (Milford, Mass.) that were based on Whitlock's work. It also included an analysis of passive devices that can be employed in the quest for lower noise.

DEFINITIONS

There seems to be confusion regarding what constitutes a balanced circuit. People talk about two signal-carrying conductors instead of one, opposite signal polarities in a balanced configuration, and so on.

Bill Whitlock (president and chief of engineering, Jensen Transformers)[2] has accurately defined "unbalanced": "An unbalanced input or output connects one of its signal conductors to ground and has a non-zero impedance at the other signal conductor."

His correct definition of "balanced" is: "A balanced input or output uses two signal conductors which have equal impedances to ground."

Note that the key difference in these definitions is the impedance between each signal line and ground. The unbalanced circuit does have two signal lines and current does flow in opposite directions in the two lines, but one of them connects to ground. Therefore, the impedances are neither the same nor balanced. The balanced circuit has equal impedances to ground from each line.

Also note that there is no mention of equal voltages, because it is the impedance balance that is critical. Failure to maintain impedance balance on the signal lines is the root cause of common-mode failures with some circuit configurations.

An understanding of this is important because a true balanced circuit has a greater ability to reject common-mode signals, which are of the same polarity and magnitude on both signal lines. These signals represent unwanted noise interference that will degrade signal quality. Reduce those and you will have less noise to interfere with your signal.

A great introduction to the subject of noise, grounding, and safety in audio connections is the *Journal of the Audio Engineering Society* v43n6 (June 1995), which includes papers and reports by Neil Muncy, Bill Whitlock, Charles Atkinson, Philip Giddings, Stephen Macatee, John Windt, Cal Perkins, and Kenneth Fause that are a gold mine of important information. There are also citations to additional references. Copies of the individual papers in PDF form are available from the AES website: www.AES.org/journal/search.cfm. You can search for any articles by the volume and number, title, or author.

Probably the least effective method to connect balanced and unbalanced devices is to make adapter cables. This is generally not a good solution, but there are adapter cable designs that take into account the type of connectors and the direction of the connection (balanced-to-unbalanced or unbalanced-to-balanced). One of the best explanations of interconnects, with a table of different connections, is in a paper from Rane™ Corporation[3]. I have built cables using their tables and have obtained effective results in situations where noise is not a major factor.

SOLVING THE NOISE PROBLEM

With noise, reality sets in when two problems are introduced: improper connections of an XLR's pin-1 to ground [AES14, SMPTE RP134, EBU R50-1988, and IEC 268 part 12 all standardized that an XLR connector's pin-1 is ground, pin-2 is the positive signal connection, and pin-3 is the negative signal connection—**David J. Weinberg**], and differences in the impedances of the two signal lines in a balanced circuit. This article does not address the pin-1 problem, which is covered in several articles in the previously-mentioned AES issue.

To reduce susceptibility to noise, high-quality transformers can retain common-mode rejection even with relatively high levels of impedance mismatch, due to their high common-mode input impedances. In addition, they can provide a high degree of voltage isolation where offsets exist between devices. Unfortunately, they are not inexpensive, and less-expensive versions often have severe performance limitations. As a result, many attempts have been made to develop active circuits that can be used as balance inputs and outputs.

At first blush, some of these active devices seem to offer ideal solutions. The Analog Devices AMP03 precision unity-gain differential amplifier has a common-mode rejection ratio (CMRR) of 100dB at 0Hz (= DC), and a respectable 70dB at 20kHz. The amp has internal resistors balanced to within 0.002% to make this performance possible and free the equipment manufacturer from having to select precisely matched external resistors.

However, any output impedance differences from the driving equipment can destroy the performance improvement achieved by this precise on-board resistor matching. It does not take much of an imbalance to produce a large drop in CMRR[4]. A balanced system acts like a Wheatstone bridge, with the driver output impedances and receiver input impedances forming the legs of the bridge. If any of the impedances become out of balance, rejection drops. For example, for an input impedance of 5kΩ, a 1Ω imbalance in the output of the driving circuit can cause a 60dB drop in CMRR.

The severity of the problem is minimized if the impedances of the bridge's output and input sections are very different. The impedances in each line of the driving side, and of the receiving side, must still be closely matched for the rejection to be maximized. Low output impedances are common in driver stages, but high common-mode input impedances in receiving stages are not, generally due to noise concerns. Transformers are an exception, routinely having common-mode input impedances in the tens of megohms.

The products from THAT Corporation include newly developed input line receivers based on Whitlock's work. Balanced-output drivers were also mentioned. There are two device families:

• The "InGenius"™ family of line receivers uses a bootstrapping technique developed by Whitlock to feed common-mode signals back to the input to increase the common-mode input impedance while maintaining lower DC impedances[5]. These devices can provide many of the benefits of transformers without the cost penalty or extreme low-frequency losses.

• The "OutSmarts"™ family of line drivers uses a dual-feedback-loop design described by Gary K. Hebert[6] that prevents excessive ground currents when clipping into single-ended loads. In cross-coupled output stages, that can lead to difficulties in the power supply and additional distortion[7].

PROJECT IDEA

These devices seemed to be the perfect solution for a project I was formulating for copying many of my LPs to CDs. Part of the setup included the use of an equalizer to correct recordings that were either not tonally balanced to my taste, or used equalization curves other than RIAA. You can find a reference to EQ curves of old records at www.

Geocities.com/midimagic@sbcglobal.net/mixcurve.htm. You can find the settings needed when using an RIAA phono section to correct for the non-RIAA curves at: www.Geocities.com/midimagic@sbcglobal.net/mixcompn.htm.

First look up the code for the equalization curve of your recording (such as 500C-16 for Columbia LPs). Then look up that code in the second resource. In this example you need these corrections:
20Hz -8.0dB
40Hz -5.0dB
62Hz -4.5dB
100Hz -2.0dB

The plan was to run the output from my custom stand-alone RIAA phono preamp through my Behringer DEQ2496 equalizer into my computer's sound card. In the analog domain, the Behringer only has balanced inputs and outputs. It would be fed from an unbalanced preamplifier and would feed an unbalanced computer audio card.

Therefore, the signal would undergo a transformation from unbalanced to balanced and then back to unbalanced in its journey from the phono cartridge to the computer sound card. Instead of using adapter cables to convert between unbalanced and balanced, as I had done in the past, it seemed time to "kick it up a notch."

This application was a good way to test the circuits from THAT Corporation. The concept was to build a box that could take the unbalanced output of the phono preamp, convert it to balanced (using two THAT 1646 differential line-drivers) for the equalizer inputs, and then convert the equalizer output back to unbalanced (using two THAT 1206 balanced line-input receivers) to feed monitoring headphones and the computer.

STARTING THE PROJECT

Any electronics project I have built started with an idea, moved to a schematic, then layout, construction, and testing. The concept part was made easy with these new ICs and their supporting datasheets.

Usually I draw the schematic by hand and then do the layout and build the project using perfboard and terminals. This time I tried another approach—find an easy-to-use PC software package that would enable me to produce a printed circuit board onscreen.

A web search led me to www.ExpressPCB.com, an Oregon-based company offering free tools that allow you to produce a schematic, check it for errors, link the schematic to a board-layout program, complete the layout, and electronically ship the results to them for circuit-board production.

Their schematic program (ExpressSCH) comes with a library of device components. You can also produce your own components either by modifying one of their existing components or by producing your own and storing them in a custom component library.

Although reasonably intuitive, it took a while for me to understand the production of a device from scratch, especially the placement of leads, connections, and the numbering scheme. Their manual will guide you through the process of producing a schematic. One subset of that manual is a guide to help you when making schematic components to use in the program (that part is available from them in PDF format at www.ExpressPCB.com/ExpressPCBBin/ExpressSCHComponentGuide.pdf).

In addition to the library of components, the program also comes with a library of symbols. As with components, you can make custom symbols and store them in a custom symbol library. I especially liked the use of ports, which are symbols indicating a connection that can be for signals, power, grounds, and so on. Using them eliminated the need to show a lot of repetitive connecting lines on the schematic.

For example, you could use a symbol on the schematic for a +15V DC supply connection wherever you needed one, without running a line back to a single supply point. Because the schematic will later be linked to a PCB layout program, the ports will indicate connections that need to be made on the board without actually showing the connecting links as lines. This resulted in a very clean schematic that looked more like a group of functional blocks than my freehand drawings. It also allows you to save your frequently used components and symbols in a library called Favorites that minimizes search effort when drawing the schematic. Once learned, the program was quite effective. An additional bonus is a tool that checks for netlist errors such as pins that were not connected.

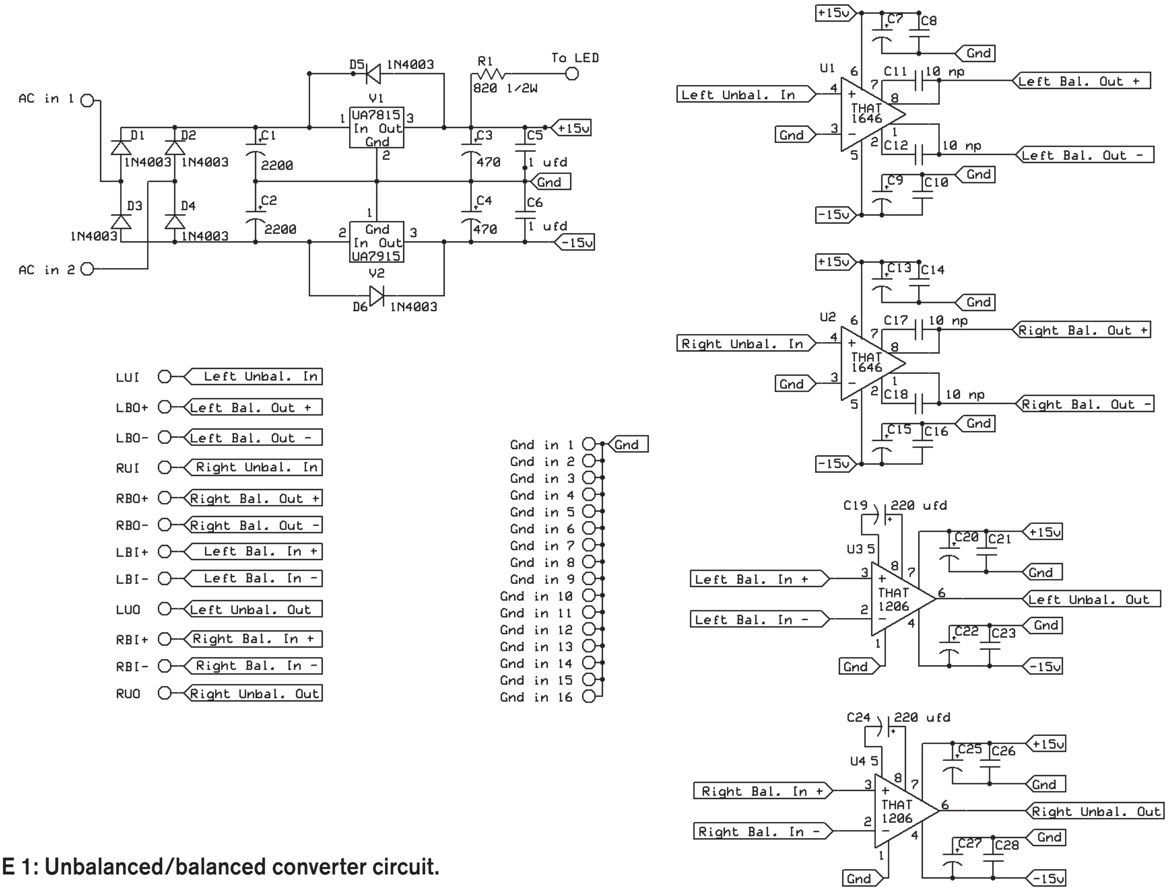


FIGURE 1: Unbalanced/balanced converter circuit.

CIRCUIT DESCRIPTION

Figure 1 shows my completed schematic, including the power supply but less the power transformer (which is not mounted on the board). The power supply section takes the secondary output of a 34V AC center-tapped transformer, uses four diodes (D1 – D4) in a full-wave bridge configuration to rectify it, and filters it using capacitors C1 and C2. That DC voltage then passes through two IC regulators (V1 for the positive bus; V2 for the negative bus), and is further filtered using capacitors C3 – C6.

Diodes D5 and D6 provide protection for the regulators against reverse bias. Resistor R1 feeds power to a chassis-mounted power indicator LED. The transformer secondary's center tap is connected to the circuit board's ground.

For this project, where high levels of RF interference would not be a problem, I used a basic configuration for the unbalanced-to-balanced and the balanced-to-unbalanced sections. If RFI is a problem, the ICs' datasheets detail methods to reduce RFI without materially affecting common-mode rejection.

The active sections of the circuit are simple, as shown in the schematic. The unbalanced-to-balanced section of the left channel consists of U1 (the balanced-line driver), power supply bypass capacitors C7 – C10 mounted right at the IC leads, and two capacitors C11 and C12 that serve to reduce the output DC common-mode voltage to zero. It is repeated for the right channel using U2.

The balanced-to-unbalanced section of the left channel consists of U3 (the balanced input line receiver), power supply bypass capacitors C20 – C23 mounted right at the IC leads, and capacitor C19 that provides the bootstrapping current to generate a very high AC common-mode impedance.

Again, these are simple circuits. If the circuits will be subjected to very high RFI or the possibility of electrostatic discharges, refer to the datasheets for suggestions on how to reduce those risks with minimal impact on common-mode rejection.

Also note the ports for power, signal, and ground leads. The signal and some of the ground ports are connected to stand-alone holes with descriptions indicating the functions of each port. The power supply positive and negative ports are directly connected to components, because no external input or output connections are needed.

Next month, we'll tackle construction and testing of this converter box. *aX*

REFERENCES

1. Gary Galo, "Grounding and System Interfacing," January 2007 *audioXpress*, p. 26.

2. Bill Whitlock, "Interconnection of Balanced and Unbalanced Equipment," Jensen Application Note AN-003, http://Jensen-Transformers.com/an/an003.pdf.

3. Stephen Macatee, "Grounding and Shielding Audio Devices," Rane Note 151, www.Rane.com/pdf/ranenotes/Grounding%20and%20Shielding%20Audio%20Devices.pdf.

4. Bill Whitlock, "A New Balanced Audio Input Circuit for Maximum Common-Mode Rejection in Real World Environments," AES preprint, http://Jensen-Transformers.com/an/ingenaes.pdf.

5. THAT Corporation DataSheet, InGenius Balanced Input Line Receivers, www.ThatCorp.com/datashts/1200data.pdf.

6. Gary K. Herbert, "An Improved Balanced, Floating Output Driver IC," 108th AES Convention, February 2000.

7. THAT Corporation DataSheet: "Outsmarts Balanced Line Drivers," www.ThatCorp.com/datashts/1600data.pdf.

Parts List

Schematic Designation	Description
D1-D6	1N4003 diode
C1–C2	2,200µF/35V DC electrolytic capacitor
C3–C4	470µF/16V DC electrolytic capacitor
C5–C6	1µF/50V DC film capacitor
C7, C9, C13, C15, C20, C22, C25, C27	3.3µF/16V DC tantalum capacitor
C8, C10, C14, C16, C21, C23, C26, C28	0.1µF/50V DC ceramic capacitor
C11, C12, C17, C18	10µF/25V DC non-polarized capacitor
C19, C24	220µF/10V DC electrolytic capacitor
V1	UA7815 +15V DC IC regulator
V2	UA7915 -15V DC IC regulator
U1–U2	THAT Corporation 1646 balanced line driver
U3–U4	THAT Corporation 1206 balanced line receiver
Power transformer	Tamura PF24-34 34V AC at 700mA.
Fuse	Buss MDL 3/10A at 250V
Chassis	Hammond Manufacturing 1590 EFL 7.38″ × 4.70″ × 3.07″
RCA connectors	Vampire CM7F
XLR connectors	Neutrik NC3FD-L-1-B and NC3MD-L-1-B

Suppliers

Digi-Key
701 Brooks Avenue South
Thief River Falls, MN 56701
800-344-4539
www.Digikey.com

THAT Corporation
45 Sumner Street
Milford, MA 01757
508-478-9200
www.ThatCorp.com

Vampire Wire
203 Flagship Drive
Lutz, FL 33549
813-948-2707
www.VampireWire.com

Markertek Video Supply
1 Tower Drive
Saugerties, NY 12477
800-522-2025
www.Markertek.com

A True RMS Audio AC Voltmeter

This voltmeter design gives you the opportunity to add a useful tool to your workbench.

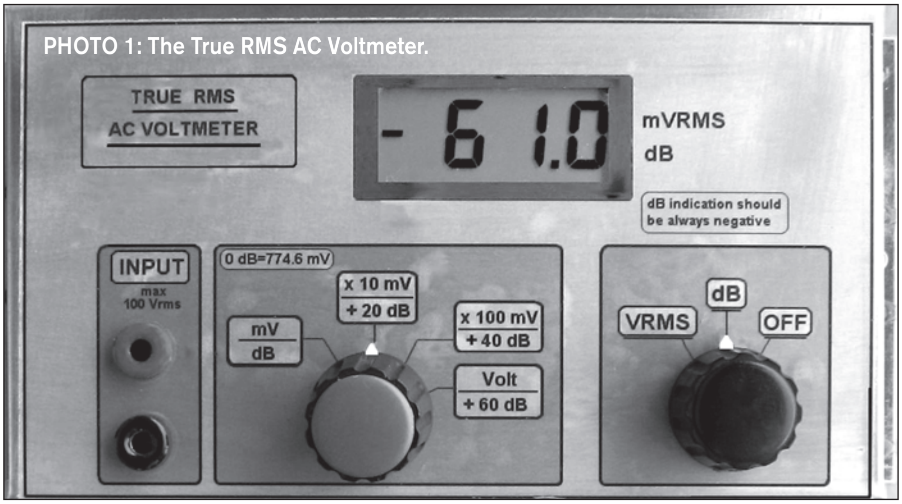


PHOTO 1: The True RMS AC Voltmeter.

The True RMS Audio AC Voltmeter is a portable unit based on the Analog Devices AD636JH true RMS to DC converter and on the ICL7106 A/D converter and LCD driver. It measures the true RMS value of an audio signal up to a level of 100V RMS in four scales: ×1mV RMS, ×10mV RMS, ×100mV RMS, and ×1000mV RMS.

A rotary switch in the front of the meter lets you display either volts or dBs. For the dB measurement, the 0dB level is set at 774.6mV RMS (0dBm = 1mW at 600Ω). The voltmeter is powered by a 9V battery. **Photo 1** shows the complete meter.

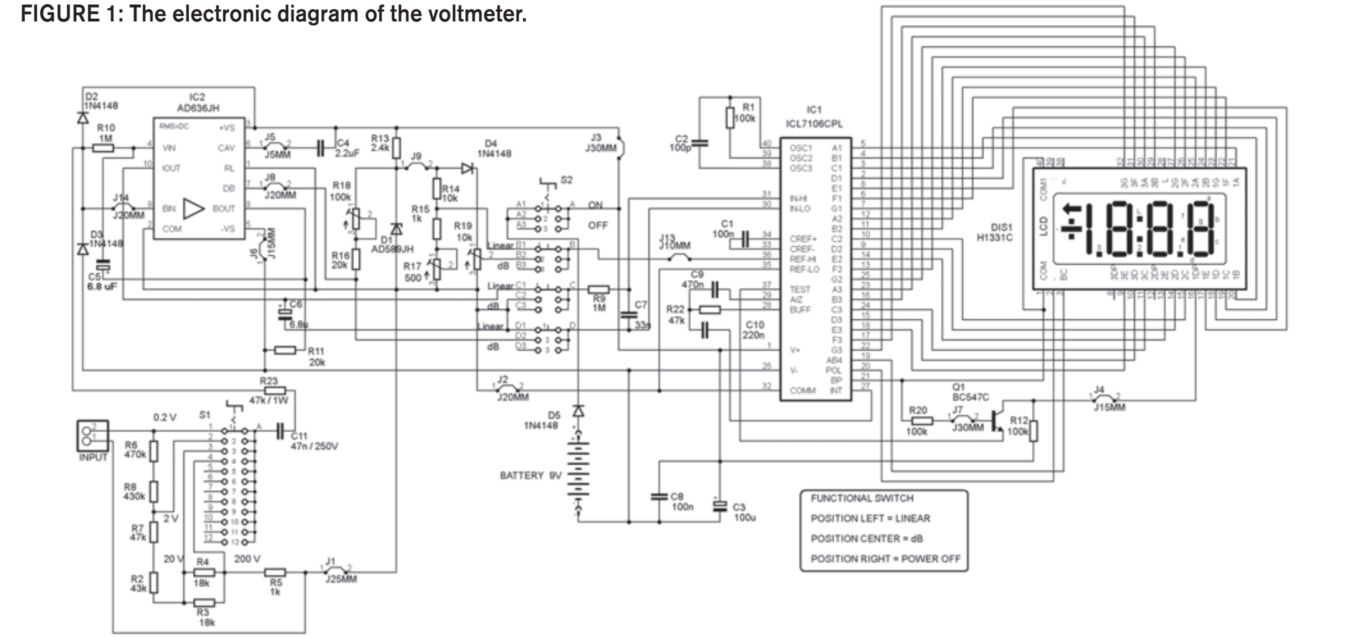
THE AD636JH TRUE RMS TO DC CONVERTER

The Analog Devices AD636JH, on which the voltmeter is based, is a low-power monolithic IC that performs true RMS to DC conversion on low-level signals from 0mV to 200mV RMS. The AD636 includes an auxiliary dB output derived from an internal circuit point that represents the logarithm of the RMS output. An externally supplied current sets the 0dB reference level and corresponds to an input level of 774.6mV RMS. According to the manufacturer, the frequency response ranges from 1.2MHz at 0dBm to greater than 10kHz at –50dBm.

The AD636 computes the true root-mean-square of a complex AC (or AC plus DC) input signal and gives an equivalent DC output level. The true RMS value of a waveform is a more useful quantity than the average rectified value because it measures the power in the signal.

An averaging capacitor is the only external component required to perform measurements to the fully specified accuracy. Its value optimizes the trade-off

FIGURE 1: The electronic diagram of the voltmeter.



between low-frequency accuracy, ripple, and settling time.

The 200mV full-scale range of the AD636 is compatible with many popular display-oriented ADCs. Also, the low-power supply current requirement—typically 800μA—permits use in battery-powered hand-held instruments.

THE INTERSIL ICL7106 AND THE LCD

The Intersil ICL7106 is a low-power, 3½ digit A/D converter with seven segment decoders, display drivers, a reference voltage, and a clock. The ICL7106 is designed to interface with a liquid crystal display (LCD), and for this reason includes a multiplexed back-plane drive. It also features auto zero to less than 10μV, zero drift of less than 1μV/°C, input bias current of 10pA (maximum), and rollover error of less than one count.

The display that I used for the meter is a typical 3½ digit LCD with ½″ character height. I don't have the exact part number of the LCD that I used because I had it in my stock for a long time, but I believe that the LCD available from Digi-Key with the part number 67-1789-ND is suitable.

THE ELECTRONIC DIAGRAM

The voltmeter is based on the block diagram of Figure 17 (*Portable, High-Z Input, RMS DPM and dB Meter Circuit*) of the Analog Devices AD636 datasheet. The complete electronic diagram of the voltmeter is shown in **Fig. 1**.

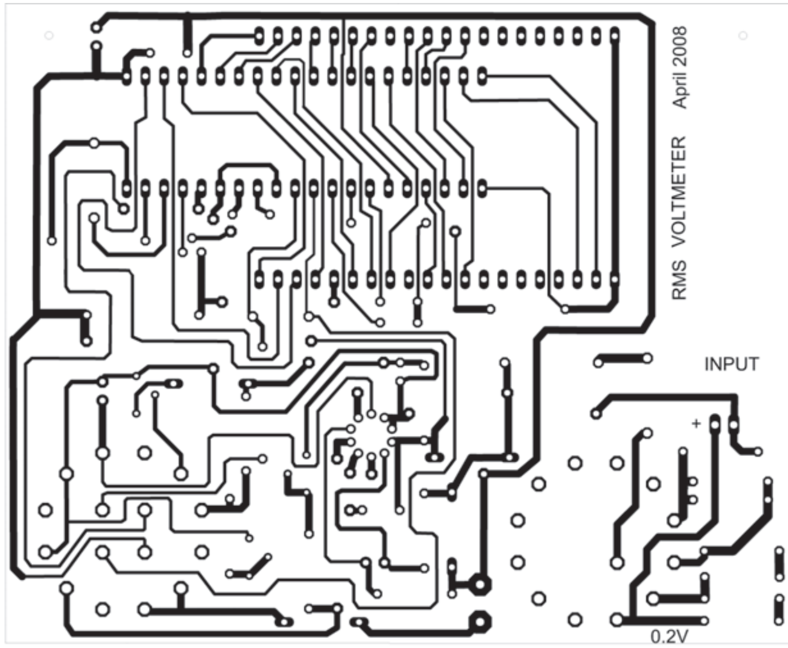


FIGURE 2: PCB.

A brief description of the operation of the meter is as follows:

The 1MΩ input attenuator allows full-scale range for signals up to a maximum of 100V RMS in four positions: ×1mV RMS, ×10mV RMS, ×100mV RMS, and ×1000mV RMS. After the attenuator, the signal is coupled through a 47nF capacitor and a 47kΩ protection resistor to the AD636 buffer amplifier, which is connected in an AC bootstrapped configuration to minimize the loading of the input attenuator by increasing the input impedance of the buffer.

Then the buffer amplifier drives the 6.7kΩ input impedance of the AD636. The LINEAR or the dB outputs of the AD636 are driven through the S2 switch to the input of the ICL7106. The COM terminal of the ICL7106 provides the false ground required by the AD636 for single-supply operation.

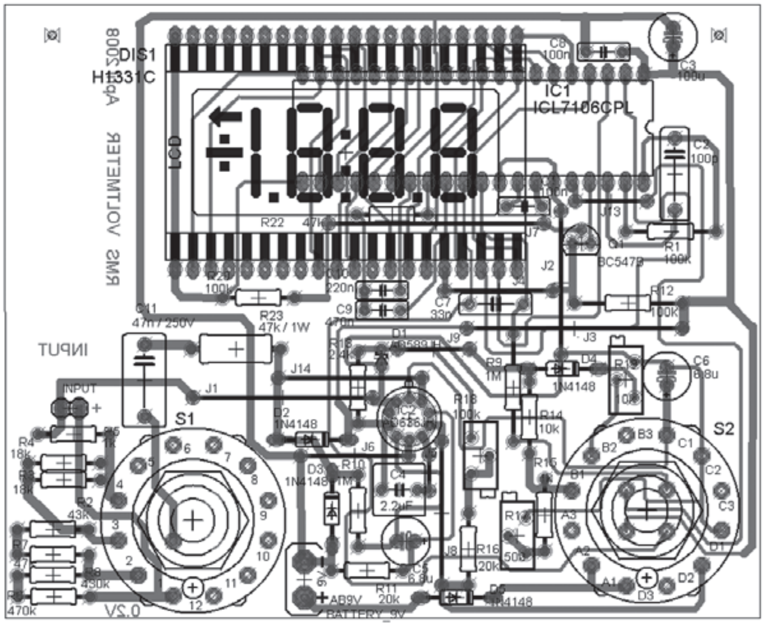
An AD589 1.2V reference diode is used to provide a stable 100mV reference for the ADC in the linear RMS mode by using the trimmer R17 (500Ω); in the dB mode, the trimmer R18 (100k) is used to set the 0dB indication. A diode 1N4148 is inserted in series to provide correction for the temperature coefficient of the dB scale factor. The trimmer R19 (10k) is used to adjust the scale factor of the dB scale.

THE PCB

The construction of the voltmeter is complicated, so I designed a PCB using the demo version of the Eagle Layout editor. You can download this demo version of the program free of charge from the Cadsoft website (www.cadsoftusa.com). The demo version is fully operational, except for a limitation on the maximum dimensions of the PCB, which was not a problem for this project.

The PCB is shown in **Fig. 2**, while the placement of the components on the PCB is shown in **Fig. 3** and **Photo 2**. The two rotary switches (S1 and S2) are placed directly on the PCB to facilitate the construction of the meter. The ICL7106, the resistor R22, the capacitor C1, and the jumper J7 are placed below the LCD. The LCD is not soldered directly to the PCB; two 20-pin sockets are used. This also permits some extra space between the ICL7106 and the LCD.

For the correct operation of the LCD, use two cables to connect pin 2 of the LCD with pin 20 of the ICL7106 and pin 3 of the LCD with pin 19 of the ICL7106 as shown in **Photo 3**. **Figure 4** shows the front plate that I used for the meter. I printed this diagram to a self-adhesive transparent paper, which I then put on the front face of the metallic



NOTICE: Two cables should be used to connect: a) the pin 2 of the LCD with the pin 20 of the ICL7106
b) the pin 3 of the LCD with the pin 19 of the ICL7106 as shown in photo 3.

FIGURE 3: PCB assembly of the voltmeter.

box of the meter. This gave a very nice look for the meter as shown in **Photo 1**. I used the same diagram as a guide for the opening of the two holes for the rotary switches and the hole for the LCD.

CALIBRATION OF THE VOLTMETER

For the calibration of the voltmeter, I used a Hewlett-Packard HP 3478A multimeter. This digital multimeter offers 3½ to 5½ digit resolution for measuring DC volts, true RMS AC volts up to 300kHz, two-wire and four-wire ohms, and DC and true RMS AC current. The input impedance of the HP3478A is 1MΩ shunted by a capacitance less than 60pF. The accuracy of the HP3478A multimeter for the measurement of the AC voltage as given by the

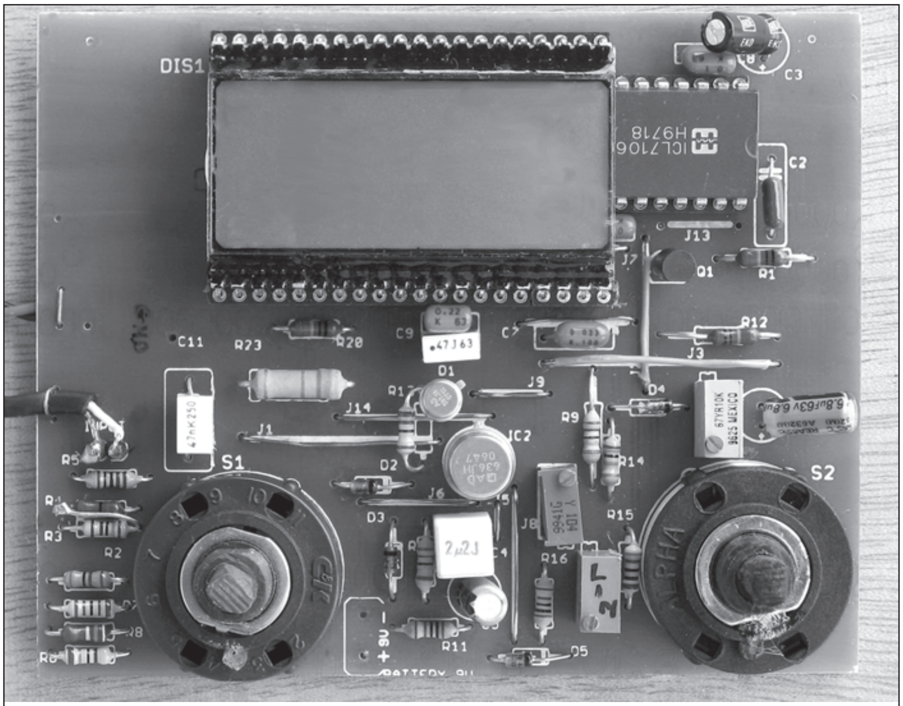


PHOTO 2: The PCB assembly of the voltmeter.

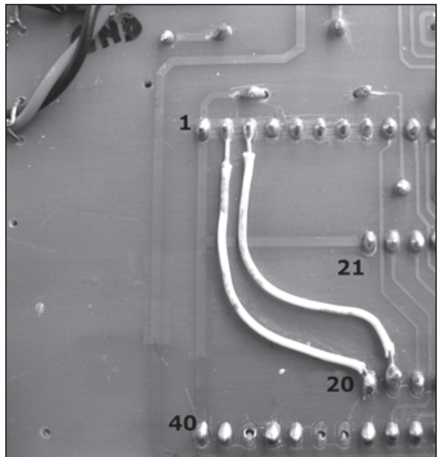


PHOTO 3: The additional connections on the PCB of the voltmeter (see text).

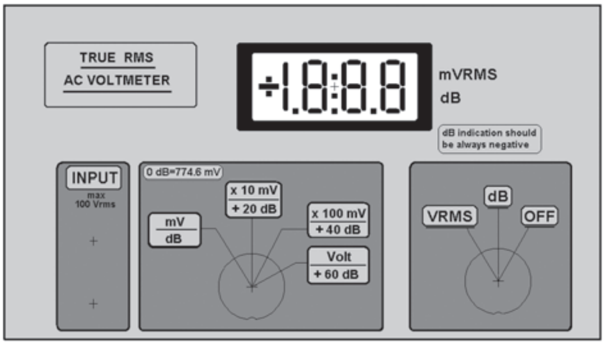


FIGURE 4: The front face of the voltmeter.

manufacturer is shown in **Table 1**.

I performed the calibration of the voltmeter as follows: I connected a sine-wave generator at the input of the RMS voltmeter with the frequency set at 1kHz and the level at 180mV RMS. Then I adjusted the trimmer R17 (500Ω) until the voltmeter display was exactly the same as the HP3478A multimeter display.

For the calibration in the dB range, I set the level of the generator at 774.6mV RMS and adjusted the trimmer R18 (100k) until the meter indicated exactly 0.0dB. Then I reduced the level of the generator to 7.74mV RMS and adjusted the trimmer R19 (10k) until the display indicated -40.0dB. The total battery current required for the operation of the voltmeter is about 2.8mA.

Hewlett Packard HP3478A
AC Volts accuracy (True RMS)

Measurement Accuracy:

±(% of reading + number of counts)
Auto-zero ON. 5½ digit display. Accuracy is specified for sine-wave inputs only, >10% of full scale.
1 Year, Cal. Temp. ±5°C

Input Frequency	300mV	Ranges 3V, 30V	300V
20Hz-50Hz	1.14 + 163	1.14 + 102	1.18 + 102
50Hz – 100Hz	0.46 + 163	0.46 + 103	0.50 + 102
100Hz – 20kHz	0.20 + 120	0.20 + 70	0.24 + 70
20kHz – 50kHz	0.38 + 205	0.26 + 140	0.42 + 140
50kHz – 100kHz	1.20 + 840	0.87 + 780	0.98 + 780
100kHz – 300kHz		10.1 + 3720 (30V Range Only)	

TABLE 1: Accuracy of HP multimeter.

ACCURACY MEASUREMENTS

After the calibration of the meter, I performed some measurements in order to compare the indications of the meter against the HP 3478 multimeter. The accuracy of the HP multimeter as given in its operating manual is shown in **Table 1**. The accuracy is specified only for a sine-wave input with a level greater than 10% of the full scale.

For the accuracy measurements, I used sine-wave and square-wave inputs in the range from 5mV RMS up to about 25V RMS. This was the maximum level that I could have from the power amplifier I used for the tests.

Note that when the meter is set in the dB range, the LCD display should always be kept less than 0dB; otherwise, the accuracy of the meter is not good. When the indication goes above 0dB, the higher voltage range should be used.

I connected both voltmeters to the same signal and checked the voltmeter displays. The results are shown in **Table 2** for the sine-wave and in **Table 3** for the square-wave.

The first column gives the results of the HP multimeter, and the second column shows the AC voltmeter results.

Input ---> SINUS 20 Hz

HP3478 multimeter indication (mV)	VRMS meter indication (mV)	mVrms Relative error with HP (%)	Indication in db by VRMS meter	dB of the HP in reference to 774.6 mV	Error difference (in dB) between HP and Vrms meter
4,997	5,0	0,06	-44,1	-43,81	0,29
10,054	10,0	-0,54	-37,6	-37,73	-0,13
19,920	20,0	0,40	-31,6	-31,80	-0,20
49,455	50,0	1,10	-23,7	-23,90	-0,20
178,350	180,0	0,93	-12,6	-12,76	-0,16
218,750	220	0,57	-10,8	-10,98	-0,18
771,510	775	0,40	0,0	-0,03	-0,03
994,450	1000	0,56	2,3	2,17	-0,13
4947,5	5000	1,06	16,2	16,11	-0,09
24780,0	25000	0,89	30,2	30,10	-0,10

Input ---> SINUS 1 kHz

HP3478 multimeter indication (mV)	VRMS meter indication (mV)	mVrms relative error with HP (%)	Indication in db by VRMS meter	dB of the HP in reference to 774.6 mV	Error difference (in dB) between HP and Vrms meter
5,155	5,0	-3,01	-43,5	-43,54	-0,04
10,185	10,0	-1,82	-37,6	-37,62	-0,02
20,142	20,0	-0,70	-31,6	-31,70	-0,10
50,075	50,0	-0,15	-23,7	-23,79	-0,09
180,040	180,0	-0,02	-12,6	-12,67	-0,07
220,235	220	-0,11	-10,9	-10,92	-0,02
775,700	775	-0,14	0,0	0,01	0,01
999,850	1000	0,02	2,3	2,22	-0,08
4995,0	5000	0,10	16,2	16,19	-0,01
25014,0	25000	-0,06	30,2	30,18	-0,02

Input ---> SINUS 20kHz

HP3478 multimeter indication (mV)	VRMS meter indication (mV)	mVrms relative error with HP (%)	Indication in db by VRMS meter	dB of the HP in reference to 774.6 mV	Error difference (in dB) between HP and Vrms meter
4,921	5,0	1,61	-46,8	-43,94	2,86
10,022	10,0	-0,22	-38,7	-37,76	0,94
20,140	20,0	-0,70	-31,9	-31,70	0,20
50,230	50,0	-0,46	-23,8	-23,76	0,04
180,409	180,0	-0,23	-12,6	-12,66	-0,06
224,301	220	-1,92	-10,7	-10,76	-0,06
788,990	775	-1,82	0,0	0,16	0,16
1016,65	1000	-1,64	2,3	2,36	0,06
4907,2	5000	1,89	16,0	16,04	0,04
24040,0	25000	3,99	30,1	29,84	-0,26

TABLE 2: Sine-wave accuracy measurements.

In the third column, the relative error between the two displays is computed. The fourth column shows the results of the voltmeter when it is set in the dB scale with the same signal as before. In the fifth column I computed in dB the indication of the HP3478 in reference to 774.6mV RMS. The sixth column shows the difference between the fifth and the fourth column.

By examining the results for the voltage range of the meter, you can see that for signals in the frequency range from 20Hz to 10kHz the differences are less than 2%, except in the measurement at a level of 5mV RMS at 1kHz, which is 3%. With the voltmeter set in the dB scale, the differences are very small and below ±0.3dB.

When the frequency is set to 20kHz, the differences are also below 2%, except at the level of 25V, where the difference is about 4%. When the measurements are performed in the dB scale, the results are good except at the 5mV RMS range.

CONCLUSION

This is a very useful piece of test equipment. It measures the true RMS value of an audio signal both in volts and dBs with very good accuracy. It is also portable and very reliable. If you don't have such equipment in your lab, this is a good opportunity to build one. *aX*

Input ---> SQUARE 1kHz

HP3478 multimeter indication (mV)	VRMS meter indication (mV)	mVrms relative error with HP (%)	Indication in db by VRMS meter	dB of the HP indication in reference to 774.6 mV	Error difference (in dB) between HP and Vrms meter
5,113	5,0	-2,21	-43,9	-43,61	0,29
10,210	10,0	-2,06	-37,6	-37,60	0,00
20,133	20,0	-0,66	-31,6	-31,70	-0,10
49,991	50,0	0,02	-23,7	-23,80	-0,10
179,930	180,0	0,04	-12,6	-12,68	-0,08
220,346	220	-0,16	-10,9	-10,92	-0,02
776,481	775	-0,24	0,0	0,02	0,02
1002,20	1000	-0,22	2,3	2,24	-0,06
4960,0	5000	0,81	16,2	16,13	-0,07
24970,0	25000	0,12	30,3	30,17	-0,13

Input ---> SQUARE 10kHz

HP3478 multimeter indication (mV)	VRMS meter indication (mV)	mVrms relative error with HP (%)	Indication in db by VRMS meter	dB of the HP indication in reference to 774.6 mV	Error difference (in dB) between HP and Vrms meter
5,129	5,0	-2,52	-45,4	-43,58	1,82
10,305	10,0	-2,96	-38,4	-37,52	0,88
20,175	20,0	-0,87	-31,7	-31,69	0,01
50,165	50,0	-0,33	-23,7	-23,77	-0,07
180,750	180,0	-0,41	-12,6	-12,64	-0,04
222,850	220	-1,28	-10,8	-10,82	-0,02
783,700	775	-1,16	0,0	0,10	0,10
1011,25	1000	-1,11	-2,3	2,32	4,62
4925,0	5000	1,52	16,0	16,07	0,07
24490,0	25000	2,08	30,2	30,00	-0,20

TABLE 3: Square-wave accuracy measurements.

Parts list for the True RMS Voltmeter

All resistors ¼W

Part	Value
C1, C8	100n
C2	100p
C3	100µ, 16V, electrolytic
C4	2.2µ, MKT
C5-C6	6.8µ, 16V, electrolytic
C7	10n
C9	470n
C10	220n
C11	22n
D1	AD589JH
D2-D5	1N4148
DIS1	3½ digit LCD with ½″ character height (see text)
IC1	ICL 7106CPL
IC2	AD636JH
J1	Jumper 25mm
J2, J8, J14	Jumper 20mm
J3, J7	Jumper 30mm
J4, J6	Jumper 15mm
J5	Jumper 5mm
J9, J13	Jumper 10mm
Q1	BC547C
R1, R12, R20	100k
R2	430k
R3, R4	180k
R5, R14	10k
R6	4.7M
R7	470k
R8	4.3M
R9, R10	1M
R11, R16	20k
R13	2.4k
R15	1k
R17	Trimmer 500Ω, multi-turn
R18	Trimmer 100k, multi-turn
R19	Trimmer 10k, multi-turn
R22	47k
R23	47k/1W
S1	Rotary switch 1 pole × 12 positions
S2	Rotary switch 4 poles × 3 positions
	2 × 20 pin IC socket for the LCD
	2 × 20 pin IC socket for the ICL7106 9V battery with clip

Setting Feedback On the **Mullard 20**

Demystifying feedback loop gain and compensation networks optimization.

This article shows how you can use network analysis programs to optimize feedback loop gain and associated compensation networks to achieve good damping and stability margins without peaking, when feedback is applied on an amplifier. The amplifier used to illustrate this is the Mullard 20W tube amp[1]. The circuit diagram used to set up the amplifier model is shown in **Fig. 1**.

AMPLIFIER MODEL

The amplifier model has been set up using the ESACAP network analysis program[2]. You can use other network analysis programs, but a copy of the model as developed in ESACAP is available at www.audioXpress.com. The ESACAP model includes the following:

section $$DES—amplifier description

sub-section $FUN: limit (x, min, max);—defines the limit function

sub-section $FUN: pwrs(x,y)—defines the power function

sub-section $CON:—provides input signal parameters, tube parameters, core geometry, OPT magnetic parameters, and topology

sub-section $NET:—describes the complete amplifier network.

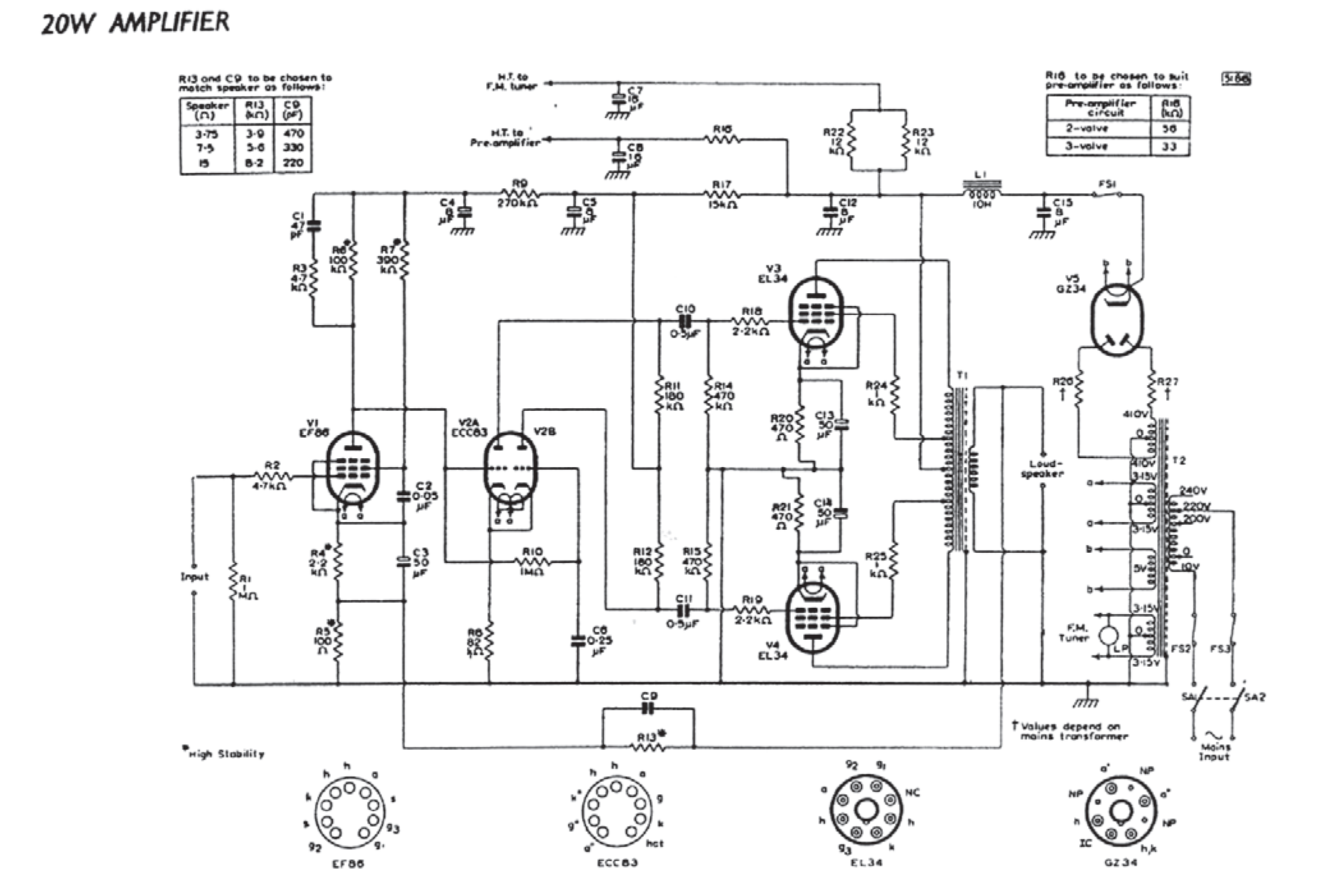


FIGURE 1: Circuit diagram of 20W amplifier.

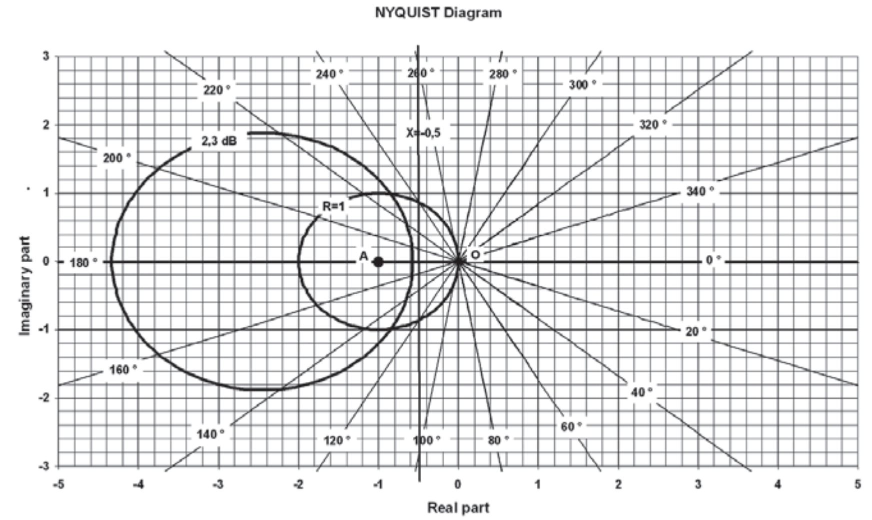


FIGURE 2: Nyquist diagram.

NYQUIST DIAGRAM

The feedback loop transfer function is graphically represented in the Nyquist plane (**Fig. 2**) with the following additions:

The critical point A (-1, 0)

The circle R=1, centered on point A, which defines the area of positive feedback

The circle Q=2.3dB, having limit points A and O, which defines the minimum stability margins

The vertical line x=-0.5, which defines the peaking conditions

FEEDBACK LOOP TRANSFER FUNCTION OPTIMIZATION

To be acceptable, the feedback loop transfer function must be warped in such a way that it stays:

—Outside the circle Q=2.3dB, to achieve good stability and damping

—On the right-hand side of the vertical line x=-0.5, to avoid peaking at both ends of the frequency bandwidth.

These conditions define the strategy for the optimization of the feedback loop transfer function. It is generally achieved using an optimum feedback loop gain and additional compensation networks.

GAIN OPTIMIZATION

The feedback voltage is defined using a voltage divider across the output load R_L (**Fig. 3**). Its transfer function (assuming that $R_t + R_k \gg R_L$) is

where: $H(p) = \dfrac{R_k}{R_t + R_k}$

R_k resistance set to $.1k$ in the Mullard 20W amplifier

R_t resistance to be determined to define the feedback loop gain

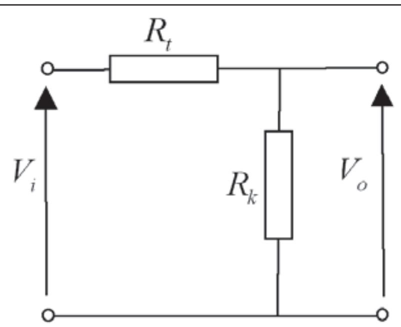


FIGURE 3: Voltage divider.

With the value of R_t set to 9k, the Routh's criterion for the Nyquist diagram of the feedback loop (**Fig. 4**) is not fulfilled. As a result, the amplifier is unstable. However, this choice is maintained because it provides an important feedback loop gain, and I will show that the present amplifier instability can be overcome properly using dedicated compensation networks.

DIFFERENTIAL COMPENSATION

The first compensation network is obtained using a bypass capacitor across the resistor R_t of the voltage divider (**Fig. 5**). The transfer function of the compensation network is:

$$H(p) = \frac{R_k}{R_t + R_k}\frac{\left[1 + R_t C_t p\right]}{\left[1 + \dfrac{R_t R_k}{R_t + R_k} C_t p\right]} = \frac{1}{a}\frac{1 + a\tau_d p}{1 + \tau_d p}$$

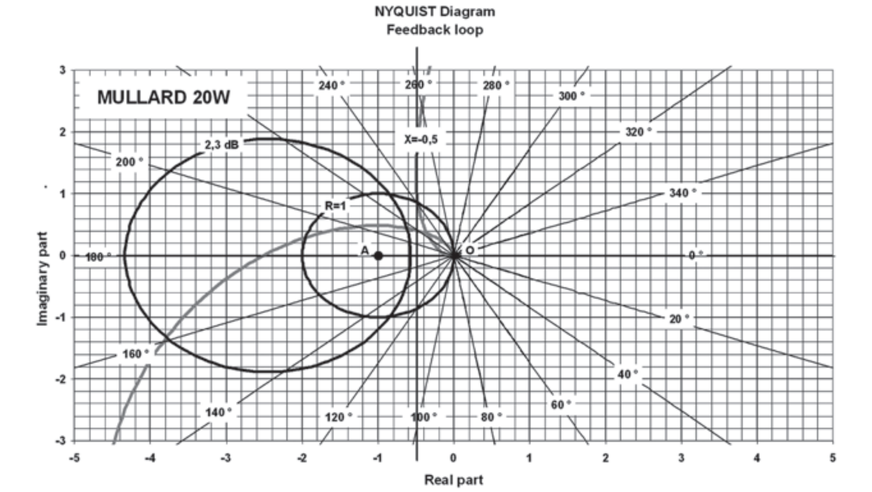


FIGURE 4: Feedback loop Nyquist diagram with a voltage divider.

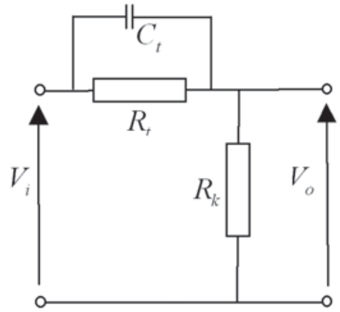


FIGURE 5: Phase lead compensation network.

with:

$$a = 1 + \frac{R_t}{R_k} \text{ and } \tau_d = \frac{R_t R_k}{R_t + R_k} C_t$$

You can recognize[3] the transfer function of a differential or phase lead compensation giving the maximum phase lead

$\varphi_m = Arc\sin\dfrac{a-1}{a+1}$ for the angular frequency

$\omega_m = \dfrac{1}{\tau_d \sqrt{a}}$. Applying these results

to the Mullard 20W tube amplifier, and using the value defined for R_t in section 5.1, you get

$$a = 91 \qquad \varphi_m = 78\,\mathrm{deg}$$

C_t must be adjusted to have $\omega_m \approx \omega_r$, where ω_r is the angular frequency resonance of the feedback loop after the phase lead compensation effect. A value of $C_t = 250p$ fulfills this requirement as it is shown on the Nyquist diagram of the compensated feedback loop given in **Fig. 5**.

The main effect of the phase lead

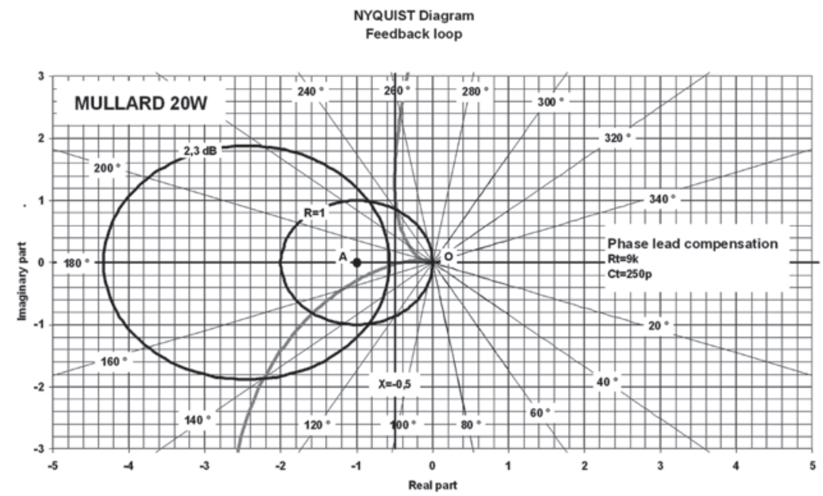


FIGURE 6: Feedback loop Nyquist diagram with a phase lead compensation.

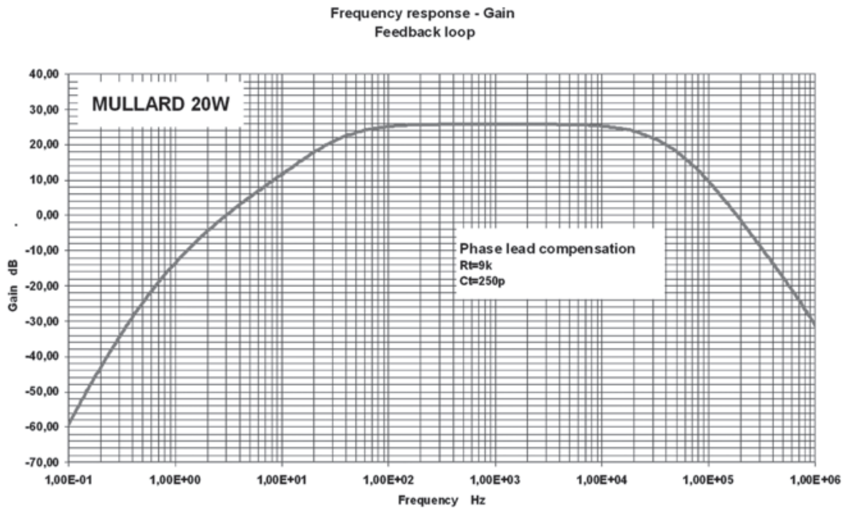


FIGURE 7: Feedback loop gain with a phase lead comparison.

compensation on the feedback loop transfer function is that you achieve stability for the amplifier. However, if the amplifier is now stable, the damping is not sufficient because a part of the feedback loop transfer function is entering into the circle Q=2.3dB. And if no peaking appears at the low end of the frequency bandwidth—because the transfer function stays on the right-hand side of the vertical line x=-0.5—it is unacceptable at the high end of the frequency bandwidth, because a part of the transfer function is on the left-hand side of the vertical line x=-0.5[4,5]. This situation is better shown on the diagrams in **Figs. 7** and **8** giving the feedback loop gain and phase shift versus the frequency and in **Figs. 9** and **10** giving the closed loop gain and phase shift versus the frequency.

As a result, to achieve a better optimization of the feedback loop, it is necessary to improve the present situation by using an additional compensation network.

INTEGRAL COMPENSATION

A second compensation network is obtained using a high-frequency step circuit across the loading plate resistance of the input stage of the amplifier, as shown in **Fig. 11**.

where:

ρ is the internal resistance of the input tube, and

R_p is loading plate resistance

The transfer function of this compensation network is:

$$H(p)=\frac{1+R_u C_u p}{1+\left(\mathrm{R}_{eq}+R_u\right)C_u p}=\frac{1+\tau_i p}{1+b\tau_i p}$$

with:

$$R_{eq}=\frac{\rho R_p}{\rho+R_p} \qquad \tau_i=R_u C_u \qquad b=1+\frac{R_{eq}}{R_u}$$

You can recognize[3] the transfer function of a gain reduction or integral compensation. To match this compensation network, it is necessary to have

$$\tau_i \gg \frac{1}{\omega_R}$$

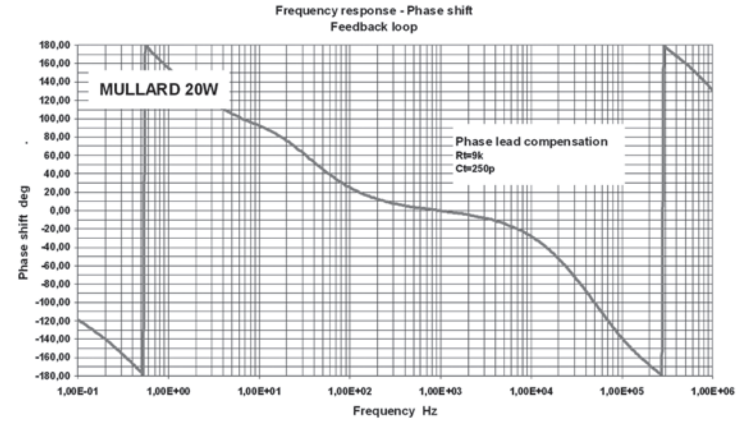


FIGURE 8: Feedback loop phase shift with a phase lead compensation.

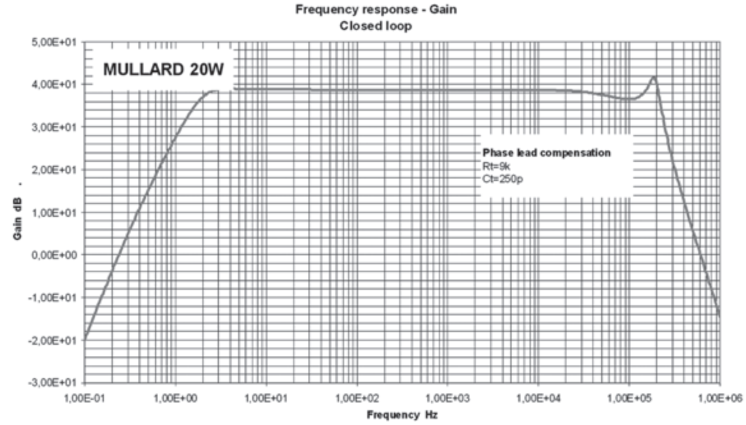


FIGURE 9: Closed loop gain with a phase lead compensation.

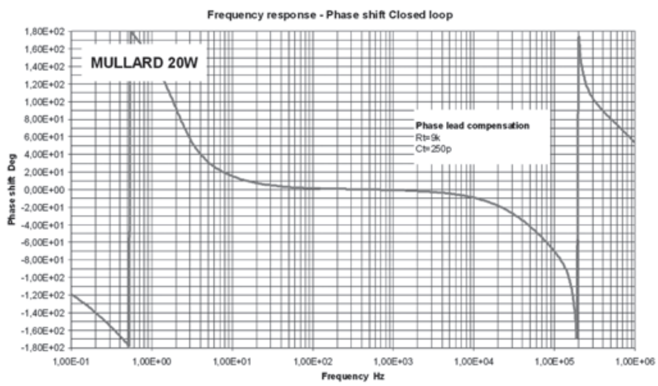


FIGURE 10: Closed loop phase shift with a phase lead compensation.

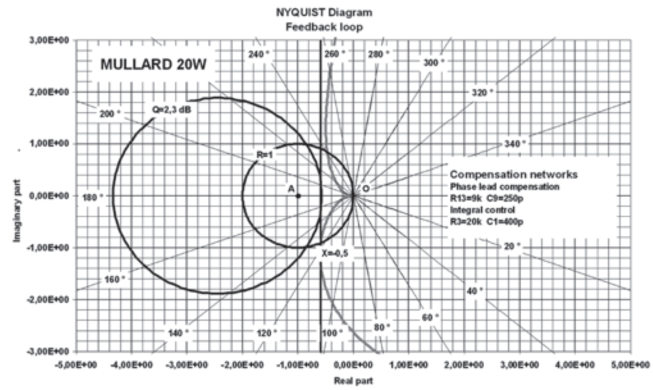


FIGURE 12: Nyquist diagram with a phase lead compensation and an integral control.

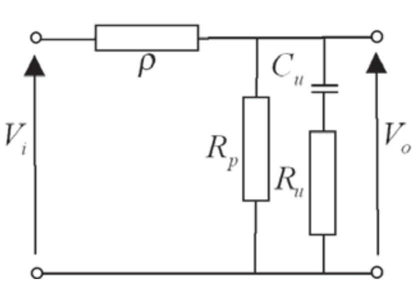


FIGURE 11: Integral compensation network.

For the Mullard 20W tube amplifier, you have:

$$\rho = 2.5M \quad R_p = 0.1k \quad R_{eq} = 96k$$

C_u and R_u must be adjusted so that

$$\tau_i \gg \frac{1}{\omega_r},$$

where ω_r is the angular frequency resonance of the feedback loop after the integral compensation effect. Values of $C_u = 400p$ and $R_u = 20k$ fulfill this requirement, as it is shown on the Nyquist diagram of the compensated feedback loop given in **Fig. 12**.

In conjunction with the differential compensation, the additional effect of the integral compensation on the feedback loop transfer function is that it allows achieving the required damping and stability margins with practically no peaking[4,5]. The transfer function has been warped enough to avoid entering into the circle Q=2.3dB and the left-hand side of the vertical line x=-0.5. This situation is again better shown in **Figs. 13** and **14** giving the feedback loop gain and phase shift versus the frequency and in **Figs. 15** and **16** giving the closed loop gain and phase shift versus the frequency.

COMMENTS

With the defined feedback loop gain and associated compensation networks, the objective was to warp the feedback loop transfer function so that it stays outside the circle Q=2.3dB and on the right-hand side of the vertical line x=-0.5. This has been achieved. As a result, a feedback loop gain of at least 20dB from 30Hz to 6kHz and 15dB from 15Hz to 15kHz is available. These results are interesting if you consider that the amplifier shows no peaking at both ends of the frequency bandwidth, with excellent damping and stability margins.

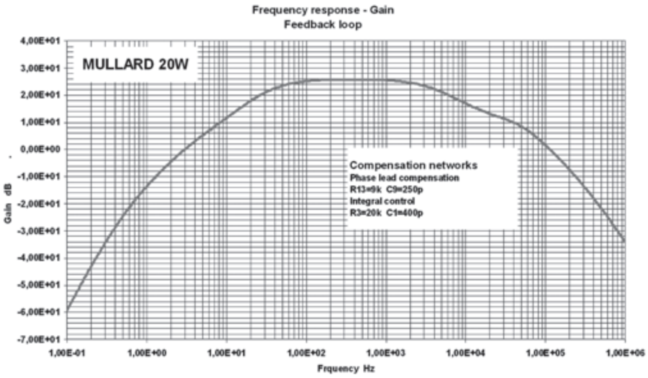


FIGURE 13: Feedback loop gain with a phase lead compensation and an integral control.

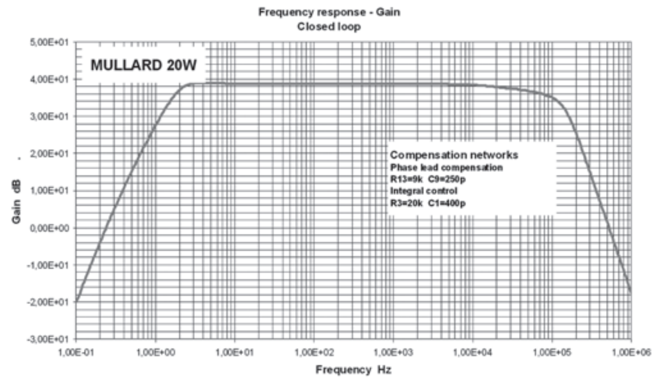


FIGURE 15: Closed loop gain with a phase lead compensation and an integral control.

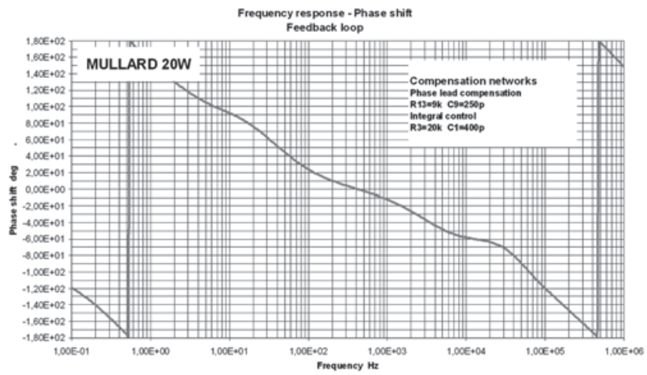


FIGURE 14: Feedback loop phase shift with a phase lead compensation and an integral control.

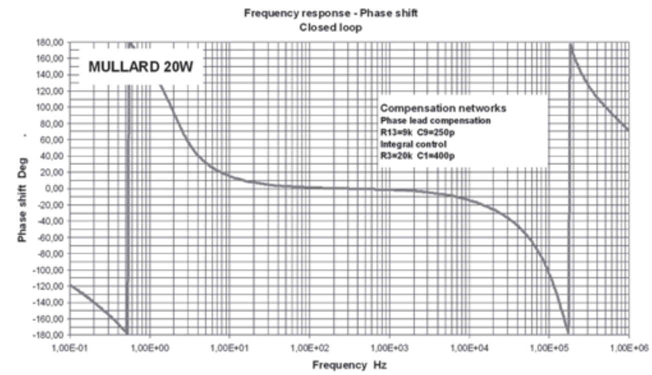


FIGURE 16: Closed loop phase shift with a phase lead compensation and an integral control.

CONCLUSION

Applying a certain amount of feedback on an amplifier is a difficult exercise according to the predefined optimization requirements. From that point of view, it is clear that using a network analysis program is helpful, as I have shown, because it allows you to define simply and surely feedback loop gain and associated compensation networks. *aX*

REFERENCES

1. *Mullard Tube Circuits for Audio Amplifiers*, second reprint edition, Old Colony Sound Lab, www.audioXpress.com.

2. Pierre Touzelet, "Accurate non linear models of valve amplifiers including output transformers," AES preprint 6830, 120th AES convention Paris, France.

3. J.-Ch.Gille, P.Decaulne, M. Pelegrin, "Théorie et calcul des asservissements linéaires," DUNOD Paris 1967.

4. Norman H. Crowhurst *Understanding Hi-Fi Circuits*, first reprint edition, Old Colony Sound Lab, www.audioXpress.com.

5. R. Brault "Basse Fréquence et Haute Fidélité," second reprint edition, Librairie de la Radio. Paris

MEASURING TURNTABLES

By Gary Galo, Regular Contributor

KAB Speed Strobe
Turntable Speed Tester
KAB Electroacoustics
PO Box 2922
Plainfield, NJ 07062
908-754-1479 (Voice)
908-222-3442 (Fax)
www.kabusa.com
info@kabusa.com
List price: $99.95

HIFI4Music DIGIStrobo
Ultra Systems
127 Union Square
New Hope, PA 18938
800-724-3305 or
215-862-6570 (Voice)
215-862-4871 (Fax)
www.ultrasystem.com
ultra@ultrasystem.com
List price: $159.95

DIGISTROBO

HIFI4Music is a small audio company based in Italy whose products are distributed in the US by Ultra Systems. DIGIStrobo is a hand-held device with a digital readout for measuring turntable speed (**Photo 1**). The DIGIStrobo measures turntable speed by reading the reflection off a piece of reflective tape once per revolution. The device is supplied with a 3½″ disc with one piece of tape attached. You can also attach a short length of the tape to your own turntable platter—a 6″ piece that you can cut to the desired length is also supplied.

To operate the DIGIStrobo, simply press the side button and point the red light beam at the revolving reflective tape. After reading multiple rotations, the memory switch on the side of the device will toggle the last speed measured, followed by the maximum and minimum speeds. The manufacturer claims that "This should give you a sense of the wow and flutter you are experiencing."

PHOTO 1: HIFI4Music's DIGIStrobo, distributed in the US by Ultra Systems. The device measures turntable speed by reading the reflection off a piece of reflective tape once per revolution. Unfortunately, the readout has only 0.1-rpm resolution, making precise reading of common speeds like 33.33 and 78.26-rpm impossible.

I beg to disagree! Measuring wow and flutter is a far more complex process than measuring once-around variations in speed, as any turntable designer will tell you. The DIGIStrobo will give you an idea of once-around changes in speed over time, but it *will not* measure wow and flutter. Wow and flutter measurements require sophisticated test equipment costing far more than the DIGIStrobo.

In theory, digital readout would seem to be the best way to measure turntable speed. Unfortunately, the DIGIStrobo's resolution is only 0.1-rpm, which prevents it from accurately measuring two of the most common turntable speeds: 33.33 and 78.26-rpm. When I try to measure 33.33, the DIGIStrobo has a habit of wandering back and forth between 33.3 and 33.4. You also must hold the device absolutely still in order to obtain the most accurate readout. If you aren't steady, you'll have difficulty obtaining repeatable measurements. I disagree with the manufacturer's claim that "HIFI4Music's DIGIStrobo measures the actual speed of the platter at any moment in all its decimal variations."

I also find that a digital readout that makes only one measurement per revolution renders adjustment of turntable speed difficult and time consuming. That, combined with the fact that you never know exactly where you are between 33.3 and 33.4, or 78.2 and 78.3, means that you can never be sure of the exact speed of your turntable. The manual notes that "a variation of around 0.1%-0.5% is acceptable therefore a measurement fluctuating from 33.1 and 33.5 rpm (for example) would not indicate any problem" (sic). The manual is describing these variations as "wow and flutter," but whether they really mean wow and flutter or steady-state errors, I again disagree.

A turntable operating at *steady-state speed* of 33.5-rpm is actually running 0.6% high. Ask any musically astute collector whether an error of 0.6% is of no consequence! A 78.26-rpm record being played 0.6% high is actually being played at nearly 79-rpm, which most pitch-sensitive collectors will find unacceptable. A once-around wow of 0.6% could make for less than pleasant listening.

KAB SPEEDSTROBE

KAB is a familiar company to *audioXpress* readers—I reviewed their excellent EQS MK12 Disc Remastering Preamplifier in the October 2002 issue. The EQS MK12 contains many interesting and unique design concepts, all the work of company owner and designer Kevin Barrett. The KAB SpeedStrobe is another unique product (**Photo 2**). Superficially, it looks like another strobe disc and AC

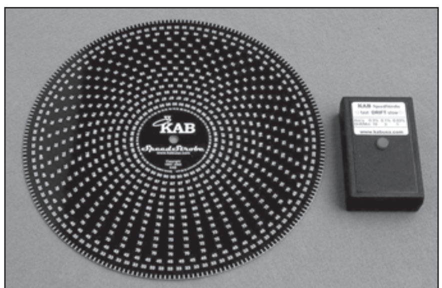


PHOTO 2: KAB's Speed Strobe takes old-fashioned strobe technology a step further by digitally generating a precise 60Hz light source independent of your power line frequency. The SpeedStrobe makes turntable adjustment extremely easy and accurate, and 78 collectors will find the 12 common "78-rpm" speeds extremely useful.

light source. But, Barrett has taken the concept a step beyond the typical turntable strobe.

The SpeedStrobe comes with a 10″ PVC disc with circular markings for three standard turntable speeds—16.66, 33.33, and 45-rpm—plus a dozen for the most common "78-rpm" speeds. Most so-called "78s" were made at some speed in the ballpark of 78-rpm, and many discs made near the turn of the last century weren't even close to the "nominal" speed. The twelve speeds chosen by KAB are 70.59, 72.00, 73.47, 75.00, 76.59, 78.26, 80.00, 81.82, 83.72, 85.71, 87.80, and 90.00-rpm. 76.60 and 75.00-rpm are the most common speeds for Victor acoustically recorded discs; Edison Diamond Discs were recorded at 80.00, as were many Columbia acoustics (Edison was much more conscientious about regulating recording speeds than Columbia).

Instead of the unlabeled circumferential markers found on most strobes, KAB prints the actual speeds (rounded off to whole numbers) on the strobe, in concentric circular rings, which makes identification of the speed you're looking for extremely easy. The only unmarked lines on the disc are for 16.66-rpm, but because they occupy the outer ring, they are easy to separate from the rest.

If KAB wanted to sell just another strobe, they could have stopped right there and let the user view the disc under any 60Hz AC light source. What makes the SpeedStrobe unique is its hand-held light source, a digitally generated, quartz-locked 60Hz oscillator accurate to 99.99% driving a bright red LED, powered by a standard 9V battery. The quartz-locked light source isn't affected by variations in power line frequency, and will work anywhere, regardless of the turntable power source. There's no need for separate 50Hz and 60Hz versions—the SpeedStrobe is a truly universal device.

As KAB points out, "Many Hi End turntables today still use hysteresis synchronous motors which rely on the line frequency to achieve accurate rotation. Using a conventional strobe to measure these turntables reveals a flaw: If the line frequency is low, the turntable will run slow, but so will the fluorescent light used to illuminate the strobe disc. Result: the strobe may show the correct speed, even though it is wrong!" KAB is absolutely correct about this.

Operating the SpeedStrobe couldn't be easier. Simply put the disc on the platter, power up your turntable, depress the red button on the hand-held light source, and point it toward the band that matches the desired speed. The large numbers on the disc and the bright LED make the SpeedStrobe extremely easy to read. In seconds you'll know whether your turntable is running too fast (numbers moving slowly clockwise), too slow (numbers moving slowly counterclockwise), or dead on (numbers remaining stationary). You can tell how accurate your turntable speed is by timing the drift of the strobe display. If 10 digits pass a given point in 1 minute, the accuracy is 0.3%; 1 digit per minute is 0.03%. KAB correctly avoids calling this a "wow and flutter" measurement (in e-mail correspondence I had with Kevin Barrett, it was obvious that he has a well-grounded understanding of wow and flutter).

WHICH IS BETTER?

The digital readout on the DIGIStrobo

may be useful if you wish to know the exact speed of your turntable. Unfortunately, many nominal speeds require accuracy to 0.01-rpm, but the resolution of the DIGIStrobo is only 0.1-rpm. A serious 78-rpm collector must be able to measure speeds such as 78.26 and 76.59-rpm, which the DIGIStrobo can't do, and every LP collector needs to measure 33.33. In order to get a stable reading, you must hold the DIGIStrobo absolutely still. The SpeedStrobe isn't affected by movement of the light source.

I find it difficult to adjust turntable speed using the DIGIStrobo—the once-around measurement is too infrequent to allow quick adjustment of turntable speed, and the 0.1-rpm resolution is not sufficiently accurate. With the SpeedStrobe's continuous and easy-to-read display, you can adjust your turntable's speed very accurately in a matter of seconds. The DIGIStrobo won't measure wow and flutter, regardless of what the manufacturer tells you.

A digital readout of turntable speed may seem like the best approach. But sometimes the "old-fashioned way" still has its place. KAB has taken the "old-fashioned method" and laid it on a new ground by providing SpeedStrobe with an accurate, stable light source unaffected by changes in power line frequency. The KAB SpeedStrobe is my first choice—every collector of disc records should own one.

MANUFACTURER'S RESPONSE:

I'd like to thank Gary Galo and *aX* for reviewing the KAB SpeedStrobe turntable speed measuring product. All the main points were covered nicely. I would only add that one test that is easy to do and meaningful, if you have adjustable speed, is dialing in the effect of static stylus drag.

If the drive system lacks servo feedback, then any added drag will slow the table down. Some tables lose nearly 2% of free spinning speed the moment the needle touches the groove, only to recover full speed at the end of the record. So it behooves these owners to average the speed at the center of the disc. This can be done by placing a 45 rpm 7″ disc on top of the KAB SpeedStrobe disc. Start the table and cue the needle into the start of the record.

Now set your speed. This will average the speed in the center so it will not be too slow at the beginning nor too fast at the end.

One correction in the text about the supplied strobe lamp. Strobe discs that rely on a fluorescent lamp are designed for 120Hz. That is because a gas lamp actually flashes on each half cycle. The KAB SpeedStrobe disc will not work under a fluorescent. I chose 60Hz illumination because it allowed for only half as many marks around the disc, and that freed up the space to use speed numbers instead of tick marks. Because of this we had to develop the dedicated illuminator.

The KAB SpeedStrobe is not new. It was introduced in 1994 and is very well known by audiophiles worldwide.

Kevin Barrett
President
KAB Electro-Acoustics

aX

OPTICAL DISK RECORDERS

Optical disk recorders can have short lives regardless of brand or type (CD or DVD). A warning sign is the player's choking on a blank disk, and it's close to the end when recordings abruptly stop, accompanied by the dreaded bad-disk message on the front panel. Purchase these units with extended warranties; out-of-warranty repairs are expensive because they basically replace the guts of the unit.

Much has been made about the quality of third-party warranties (manufacturers do not sell extended warranties), and you may achieve better satisfaction if you purchase the unit in a store. Often, you can order units online with the extended warranty from a brick-and-mortar store. This gives you the flexibility to return the unit to the store for repairs (assuming the recession has not downsized store locations or the store disappears completely). Amazon offered extended warranties, but stopped the practice after a few months. Most other online-only sites, including those that sell the semi-professional products, do not offer extended warranties.

Because units can often recover from their own mistakes, bad recordings may not be readily apparent unless you try the disk in a second machine. It is best to get into the habit of finalizing the disk and then playing it on another machine. *Check all tracks*. The higher-numbered tracks have the highest fault frequency. If you do not diligently follow this routine, black vinyl thought to be successfully archived on another medium may be thrown out when, in reality, it has not been copied.

Some recorders are more prone to disk errors than others, but I have found no player that is completely reliable. Semi-professional equipment is not immune to these problems. Make sure you have a money-back guarantee on your units, and avoid stores that do not offer this option. We all know to scan for return-policy differences across stores, but exchange policies can also vary. For some policies, the window of opportunity may be no longer than seven days, which may be insufficient for identifying premature optical drive failure.

Some machine brands are unreliable and, in the case of a DVD-R, often make recordings that will not play on other machines or will play only part of the DVD-R. The title list format is often the bottleneck.

Sometimes, it is best to avoid exchanges of a unit that appears problematic out of the box, and simply buy a unit from another manufacturer. Price does not correlate with reliability or the ability to play a recording on another machine. Internet reviews and blogs may give you clues about reliability, but there are few that have touched on the issue of transportability of disks.

The quality control of the CD–R recordable media is very important. If the media is defective, a bad disk error can occur in the middle of a recording. Once the error occurs, the machine stops. This means you are out of luck unless a backup machine is running as well. Semi-professional recorders accept standard CD-R disks, but I have found CD-Rs can have non-trivial 5%+ defect rates. Maxell CD-R music disks almost never generate a bad disk error (less than 1%) in my experience.

You should check any optical media before you use it. Look for small holes or other defects on the surface. Check to make sure the disk is clean with no fingerprints or scratches, which happen most often with a disk that you did not fill up when you did the first recording, placed the CD-R aside, and then used it sometime later to make an additional recording. It is important to remember that disk cleanliness is much more important during the record process than when a finalized disk is being played back. For DVD-R disks, I have not been able to determine a brand that differentiates itself with regard to minimizing bad disk errors from occurring.

David Rich
darichpublic@mindspring.com

HELP

I want to build very tall line source ribbon loudspeakers. Despite reading *Ribbon Loudspeakers* by Justus V. Verhagen and other *Speaker Builder* articles on ribbon loudspeakers, I am at a loss as to how to properly orient the magnets (in the case of bar magnets). I really need descriptive drawings that are specific to orientation.

Just how are bar magnets oriented in relation to the gap the single strip of aluminum foil sits in? How are they stacked on top of one another in relation to the magnet below and above? How about in relation to the opposing stack of magnets?

I would love to hear back from experienced ribbon loudspeaker builders. Note: I am *not* referring to the pseudo-ribbon, planar types, but the straightforward single strip of aluminum foil running between two stacks of bar magnets.

Angel Rivera
speakerman@excite.com

JUNE ISSUE IS A HOME RUN!

I never write to say how much I enjoy an issue of anything, but the June issue was outstanding. I received it yesterday at my post office box. I came home, and after dinner I read it cover to cover. The only thing I didn't read was the dipole microphone article. Not because it wasn't good, but I'm just not interested in that subject.

Twice a year, I send out a music compilation to my friends. Every now and then, I get a bunch of e-mails back saying, "Thanks. This was the best one yet." I'm sure they enjoyed that particular disc and are just being nice, but there's a little voice in my head that says, does that mean all the rest of them were sh*t and they were just not saying anything?

So, I don't want to insult you by this. All of the issues are good, and I usually find the majority of the articles to be interesting. That's a great achievement when you're trying to appeal to three diverse groups at the same time. But somehow, this issue just really hit it. All of the articles were interesting to me: solid state, glass, and speaker building.

Also, the reason I've never sent you a music compilation is that I don't send them to audiophiles because they generally tend to not enjoy hard rock, which these compilations are mostly composed of, despite the fact that I'm way in my 50s! My tastes run from rock and roll to classical. The only audiophile who I send one to is Noel Keywood from Hi-World in the UK. As a 20-something person, he actually followed the Sex Pistols around in a van with his friends. Hard to believe, until you see what music he uses today to test and review equipment.

To conclude this long-winded epistle, thanks for a *great* June issue. Keep up the good work and don't get insulted!

Clayton Mitchum
aX

audioXpress online

Rebuilding a Cheap Chinese EL84 Amp

By Merlin Blencowe

A cheap and cheerful valve amp gets a lobotomy.

Any hi-fi enthusiasts who, like me, spend too much time on a certain popular Internet auction site can hardly have failed to notice the enormous influx of very cheap Chinese valve amplifiers in the last couple of years. Many of them boast suspiciously remarkable performance at mouth-watering prices. Against my better judgment this was enough to pique my curiosity, so I decided to buy the cheapest one I could find and see whether it would live up to my (understandably low) expectations. I found a tiny little "Mengyue Mini" push-pull EL84 stereo for £100 including shipping (around $200 US).

About eight weeks later it arrived, astoundingly undamaged, and the box read "Aria Mini"—they seem to re-brand these little amps every few weeks. The power valves supplied were actually 6P14s, which are Russian EL84 equivalents. The outward finish was very good—much nicer than I was expecting, and I decided to go right ahead and plug it in; the tasteless blue LED uplights came on and it actually played music as promised! To its credit it actually sounded quite pleasant—worth the money at least—and managed to drive a pair of Leema Xero bookshelf speakers to a satisfactory volume. I also made power, distortion, and noise tests.

Furthermore, after about five hours of continuous playing the transformer cover became very hot, and removing it revealed the power transformer sweating beneath—clearly being overrun. On further investigation I found the EL84s were biased to a scorching 15W each (3W above their rated maximum), the chassis had no safety earth and one of the screen-grid resistors also caught fire after being shorted by some glue near the PCB. Clearly it needed some internal reworking, by which I mean a complete overhaul!

REDESIGNING THE CIRCUIT

I decided to keep the same valve complement of EL84s and 6N3Ps (similar to a 2C51 or 5670), and to reuse some of the PSU smoothing capacitors (I could find no replacements that were so small) but to redesign the circuit. Tracing the PCB indicated that the original circuit consisted of an input triode feeding one power valve and also a potential divider. The divider then fed a second, identical triode, which inverted the signal and fed this to the other power valve. Global negative feedback was applied to the cathode of the input triode from the speaker terminal, though the gain of the amp rose slightly with frequency, implying poor HF stability.

to continue reading

***Visit* www.audioXpress.com**

What's New on the *aX* website?

Web-exclusive content

An Excel application for calculation of Vg1 from Rudy Godmaire's "Vg1 Correlations" (*aX* 3/09).

Jan Didden's show reports from *aX* 1/09: "The Munich High-End Show 2008," "The Rocky Mountain Audio Fest," and "Burning Amplifier 2008."

Tri-Way PC boards and parts list from Paul Stamler's "Tri-Way Low Voltage Supply, Pt. 1," *aX* 1/09.

The *audioXpress* 2008 index.

Articles from Past Issues:

"ezDAC," by Claudio Negro (*aX* 5/09).

"DIY Microphone Calibration," by Ron Tipton (*aX* 4/09).

SchmartBoards, reviewed by Ed Simon (*aX* 4/09).

The Restoration Preamp, reviewed by Don Walizer (*aX* 4/09).

"Modding the Music Hall CD25.2 CD Player," by Kit Ryan (*aX* 4/09).

i2i Transmitter/Receiver, reviewed by Chuck Hansen (*aX* 3/09).

"Noise Measurements of the LSK389B Dual JFET," by Dennis Colin (*aX* 2/09).

"A De-Emphasis Test CD," by Gary Galo (*aX* 2/09).

Benchmark DAC1 USB, reviewed by Gary Galo (*aX* 1/09).

Measurements for the Benchmark DAC1 USB, by Chuck Hansen (*aX* 1/09).

Tascam CD-RW900SL CD Recorder, reviewed by Gary Galo and Chuck Hansen (*aX* 12/08).

Dayton Audio WT3 Woofer Tester, reviewed by G.R Koonce and R.O. Wright (*aX* 11/08).

For information on these and others, visit www.audioXpress.com. And don't forget to check out the links to our other magazines, *Voice Coil* and *Multi Media Manufacturer!*

CONTRIBUTORS

Kent Smith (Guest Editorial: Amplifier Differences and Lack of Tests, p. 5) resides in Wylie, Tex.

Thomas Perazella ("Maintaining Your Balance, Part 1," p. 6) is a retired IT Director who has been actively involved in the audio hobby since building his first EICO FM tuner. He designs and builds custom speakers that minimize room effects and provide wide bandwidth, high dynamic range output. Protection of valuable components has also been a subject of interest and he has built devices to provide soft-start and voltage-monitoring functions for large groups of electronic devices as well as circuits to protect speakers against excess power and DC offsets. When not involved with audio, he enjoys cooking and precision pistol and rifle shooting.

George Danavaras ("A True RMS AC Voltmeter," p. 12) graduated from National Technical University of Athens, Greece in 1986 with a degree in Electronic Engineering. He currently works in the R & D division for a Greek Telecommunication company. His hobbies include design and manufacturing of audio crossovers, amplifiers, and loudspeakers.

Pierre Touzelet ("Setting Feedback on the Mullard 20," p. 18) resides in France.

Gary Galo (Review: Measuring Turntables, p. 24) is Audio Engineer at The Crane School of Music, SUNY Potsdam, where he also teaches courses in music literature. A contributor to AAC since 1982, he has authored over 230 articles and reviews on audio technology, music, and recordings. He has been the Sound Recording Reviews Editor of the ARSC Journal (Association for Recorded Sound Collections) since 1995, was co-chair of the ARSC Technical Committee from 1996 to 2004, and has given numerous presentations at ARSC conferences (www.arsc-audio.org). Mr. Galo is also a frequent book reviewer for Notes: Quarterly Journal of the Music Library Association, has written for the Newsletter of the Wilhelm Furtwängler Society of America, and is the author of the "Loudspeaker" entry in The Encyclopedia of Recorded Sound in the United States, 1st edition.

WE WANT YOUR FEEDBACK!

audioXpress encourages reader feedback in the form of letters, queries, and comments.

Send correspondance to:

audioXPRESS

Editorial Dept.
PO Box 876
Peterborough, NH 03458-0876

Phone: (603) 924-9464
Fax: (603) 924-9467

Website: **www.audioXpress.com**
E-mail:
editorial@audioXpress.com

audioXpress reserves the right to edit letters for length and clarity. Letters should be brief and to the point.

VENDORS

ALL ELECTRONICS CORPORATION

Electronic and Electro-mechanical Devices, Parts and Supplies. Many unique items.

www.allelectronics.com

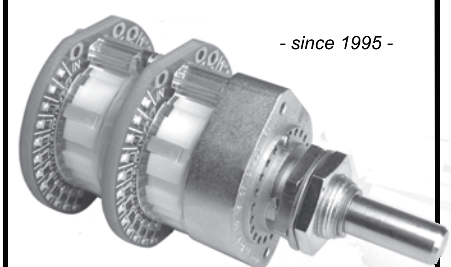
Free 96 page catalog
1-800-826-5432

BILLINGTON EXPORT LIMITED

All types of audio tubes. 300B 6DJ8 ECC81 ECC83 KT88 Mullard GEC
Minimum order £100.00
Discount for large quantity.
Billington UK. Tel (0)1403 784961.
Fax (0)1403 783519.
Email/website www.bel-tubes.co.uk

goldpoint™ LEVEL CONTROLS

- since 1995 -



For best quality and performance
*- when it **makes** a difference -*

- **Precision Volume Controls**
- **Selector Switches**
- **Headphone Amplifiers**
- **Passive Preamps (RCA & XLR)**

visit us online today!

www.goldpt.com

High Performance kits, Audiophile components
Custom designs, Custom Assembly
www.borbelyaudio.com
In North America: LBAudio, Les Bordelon,
lbordelon@roadrunner.com
In Taiwan: TS audiolab, Tai-Shen Lee,
tsaudio.lab@msa.hinet.net

Vista-Audio, Radii, Audio Limits, Trafomatic.
Tubeamplifiers, kits, custom transformers.
www.engineeringvista.com

NEW PRODUCT Tuneable stereo rumble filter, model 603A. 5Hz/step from 15 to 65 Hz.
www.tdl-tech.com/data603a.htm

Yard Sale

Wanted

I need nine more KLH 911B two-way speakers for a project. Brand new on sale they cost $10 each at Best Buy a few years back. Please contact me if you come across any of these.
Angel Rivera angelluisrivera@excite.com

"Yard Sale" is published in each issue of *aX*. For guidelines on how subscribers can publish their free ad, see our website.

SUBSCRIBE AND SAVE!

SAVE NEARLY $50 OFF THE COVER PRICE!

Call 888-924-9465
or subscribe on-line at
www.audioXpress.com.

Ad Index