SOFTWARE FOR AUDIO TESTING

Perform audio testing with these "virtual" instruments, and go easy on the pocketbook!

by Ron Tipton

Traditional audio testing is done with a variety of lab instruments. A signal generator's output is applied to the input of the device under test (DUT) and its output is examined with an oscilloscope, distortion analyzer, spectrum analyzer, etc., depending on what characteristics are wanted about the DUT. The DUT can be any active or passive network, such as an amplifier or filter. It can also be the acoustical properties of a room or the sound attenuation of a wall. These test set-ups are shown graphically in Figures 1a and 1b. Some time ago, I set out to look at how much of this work could be done with a personal computer and a sound card. I've learned that many of the usual lab instruments can be replaced with "virtual instrument" software that ranges in price from free to moderate. However, I found that price is not always a measure of quality. One of the best programs I found is low-cost shareware. In this article, I want to share my evaluation and comparisons with you.

There is one downside to using virtual instruments, and that is the sound card itself. In order to use the card's full dynamic range, the peak input level must be set just below the point where the analog-to-digital converter (ADC) is overdriven. Depending on the gain or loss of the DUT, you may have to add attenuation or gain in the signal path to keep your sound card's input level where it needs to be. This means having to know the characteristics of the added devices so their effect can be calibrated out. In addition, you need to know a lot about your sound card, so we'll look at that next.

SOUND CARDS

Sound cards are not created equal. There is a lot of variation in quality, with the better cards usually being higher priced. However, if you shop around, you can find several good cards for less than $100.00. Fortunately, Mr. Arnold B. Krueger evaluates sound cards and posts the information on his web site [1]. He updates his data frequently to keep up with the introduction of new models. Oddly (to my thinking), the best cards have the lowest score. He rates frequency response, 1 kHz THD, SNR, IM, dynamic range, and jitter in six steps from excellent to bad. (Please see the Glossary for unfamiliar terms.) The best cards have a score of five and rate excellent in all categories. The Turtle Beach Santa Cruz card has a score of seven with three excellents and three very goods. The "street price" is about $80.00. and all the software evaluations in this article were made with the Santa Cruz.

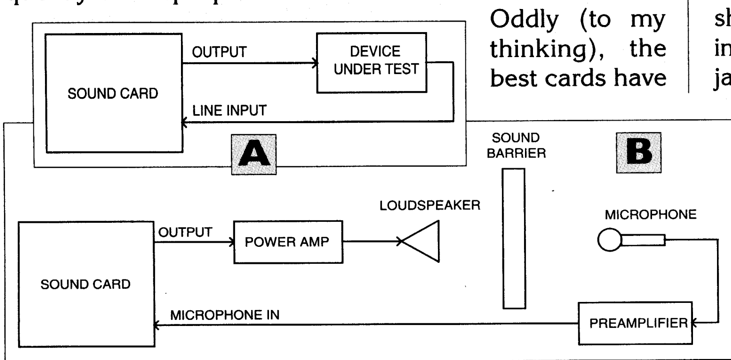

Figure 1. A) Testing an active or passive network. Attenuators or amplifiers may be needed in the signal path to keep the line input level at the optimum value. B) Testing with a loudspeaker and microphone. This setup can also be used to test the loudspeaker or its enclosure.

Although some of the software will perform post processing, that is, analyze a recorded file, I included only programs that do real-time analysis because that's how lab instruments work. So I didn't include programs such as DADiSP [2] or MathCAD [3]. Undoubtedly these are useful, but not for real-time audio measurements. I also didn't include software that costs $1,000.00 or more, such as Clio [4] and SYSid-Labs [5]. As fine as they may be, they are out of my range. If your budget can take it, you may want to check them out.

You also need to calibrate your sound card's line input. That is, you need to know the maximum input level that avoids overdriving the ADC. Some programs contain an integrated way to perform the calibration, but I'll describe a general method that requires only a physical voltmeter, virtual (or physical) signal generator, and a virtual spectrum analyzer. I've summarized my findings in Table 1 to give you an overview and also for quick reference. Let's start our discussion with the third table line (AudioTester) which I rate a Best Buy. It's rich with features and easy to learn. For $28.00 — you shouldn't be without it!

AUDIO TESTER

This package of three integrated programs (wave generator, 'scope, and spectrum analyzer) includes a built-in calibrator to set the optimum input level. You can follow the on-screen instructions, so I won't repeat them here. Instead, I'll cover the general method I mentioned earlier and then go on to a practical example — testing a tunable filter.

Begin by building the circuit shown in Figure 2 in a shielded box. It's useful to have left and right outputs and inputs for testing and it avoids wear on your sound card's jacks. Connect the left output to both the left and right inputs, as well as to an AC voltmeter. The optimum input level is typically 200 to 600 millivolts (mV) RMS so your voltmeter should have adequate resolution in that range. Launch your sound card control panel (the little speaker in the lower right corner). Select line-in and mute the other inputs. Set the line-in gain to maximum and the master volume slider to about midrange.

Start AudioTester — you'll see the spectrum analyzer and the analyzer's control panel as separate screens. Arrange them one above the other and then click the Signal Generator tab. Pull the generator control panel off to one side and then return to the spectrum analyzer by clicking its tab. Select 1,000 Hz and sinewave. Set the "L" and "R" sliders to 0 or -1 dB and then click Start. On the analyzer panel, choose a Blackman-Harris window, 4K FFT length, 44.1 kHz sample rate, 16-bits, and then click Start. You should see a trace at one kHz on the analyzer screen, and you should have a reading on the voltmeter. Your goal is to set the master volume level (generator panel slider labeled Ana) to minimize the THD+N. If the level is too high, you will also get an OVER displayed in the Input area on the analyzer control panel.

You can accomplish this and get the same results by using NCH Tone as the signal generator and Spectrogram as the analyzer. Spectrogram doesn't have a built-in THD measurement, so you have to adjust the line-in level to minimize the heights of the harmonics compared to the one kHz amplitude. (You can calculate the THD with program thd.exe [6].) I have compared all the software analyzers in Table 1 to a physical analyzer (a Hewlett-Packard model 3581A) and they all agree to within a few dB. This convinces me that I can rely on the accuracy of the virtual instruments. Now on to a practical example.

Program Name	**Version**	**Type**	**Signal Gen**	**'Scope**	**Spectrum Analyzer**	**Post Processor**	**Comments**
At Spec Pro **www.taquis.com**	2.2	Shareware $95.00	Yes	Yes	Yes	Yes	Very nice. Does not have sweep freq., in signal generator, but pink noise does a good job displaying response curves. Shareware version somewhat limited.
Audio Meter Pinquin **www.masterpinguin.de**	PG-S 2.2	Commercial $245.00	No	Yes	Yes	No	Has digital peak meter, stereo meter, and correlation meter. Apparently no choice of FFT length.
AudioTester **www.audiotester.de**	1.4h	Shareware $28.00	Yes	Yes	Yes	Yes	Selectable input filter, up to 140 dB dynamic range, windowing, FFT size to 16K points, THD+N, lots more. A best buy. (Updates are free.)
NCH Tone **www.nch.com.au**	1.03	Shareware $36.00	Yes	No	No	No	1 Hz to 22 KHz sine, square, triangle, saw, impulse, white noise. Log and linear sweep. Sound card out and wav file for settable time duration.
Sample Champion Pro **www.purebits.com**	2.8	Commercial $325.00 plus plugins	Yes	Yes	Yes	Yes	Very complete package. I found it a bit hard to learn so I printed the 90-page User Manual (pdf file) and followed the examples. This one grows on you the more you use it; I have a licensed copy.
SpectraPlus **www.spectraplus.com**	2.3.2	Commercial $299.00 plus plugins	Yes	No	Yes	Yes	Choice of window, FFT size to 32K, THD, THD+N (with plugin), 100 dB range, octave, 1/3 octave, spectrogram, 3D surface. Excellent program.
Spectrogram **www.visualizationsoftware.com/gram.html**	7.2	Shareware $45.00 ($25.00 if student)	No	No	Yes	Yes	No window choice, FFT size to 16K, 90 dB range. Also 1/2 octave. Very easy to use.
TrueRTA **www.trueaudio.com**	2.0.1	$99.95 for all features	Yes	Yes	Yes	No	Octave-band analyzer only version is free. Very useful program, see write up in the article. I have licensed the full-featured version.
Wavetools **www.mda-vst.com**		Freeware	Yes*	Yes*	Yes*	Yes	* Separate programs. Analyzer and scope screens small and not resizable, but can put data on clipboard. No sweep freq., but has pink and white noises — and it's free!

TABLE 1. Comparison of virtual audio instrument software evaluated for this article.

A TUNABLE FILTER

Some time ago, I rebuilt an SKL model 302 Tunable Filter [7]. I replaced the original vacuum tube circuits with op-amps, decreasing the weight, power consumption, and output noise level. I thought this filter would make a good DUT. I could look at a frequency response, THD+N, and output with no input signal.

The sweep frequency generator in AudioTester has an especially useful feature. If the right output line is looped back to the left line-in, the frequency response of the sound card is compensated out. Referring to Figure 2, I connected the right output line to the filter's input and to left line-in. I connected the filter's output to the right line-in, a voltmeter, and a physical 'scope. I tuned the lowpass filter cutoff frequency to two kHz and the highpass cutoff frequency to 500 Hz. I set the start frequency on the sweep generator to 100 Hz, the stop frequency to 10 kHz, and the number of frequency steps to 50. I clicked Start and got the bandpass response shown in Figure 3. AudioTester will print the screen or save the data in a text file. I chose the text file and then imported it into a graphing program. Using the wave generator set for a one-kHz sinewave, I got the spectrum shown in Figure 4 and a THD+N reading of -71.5 dB, which is 0.0266 percent. With the signal generator off, the one Hz was gone, but the spectrum was otherwise unchanged, so the output noise level is -71.5 dB. Now, let's take a look at making these measurements in a couple of the other programs.

SPECTRAPLUS

SpectraPlus does not feature automatic compensation of the sound card's frequency response, but it does let you perform the compensation. Loop the output back into the line-in with a jumper cable and connect a voltmeter to the line-in to make sure you aren't overloading it. Select the Utilities menu and set up the sweep frequency generator. Set the start and stop frequencies to 20 Hz and 22 kHz and the sweep time to 10 seconds or longer. Select Realtime on the Mode menu. Select 16-bits, mono, 44100 sample rate, a Blackman window, and 4096 FFT length.

Clicking the Spectrum Analyzer Run button will start it and the sweep generator. You should start seeing the response of your sound card on the analyzer screen. It may be flat or it may show a decrease in amplitude at higher frequencies. If it's flat, you don't need a compensation file. But to make a compensation file, stop the analyzer and then click Edit (menu bar) and then Copy. This puts an ASCII table of frequency and response values on the clipboard.

Minimize SpectraPlus and start an editor, such as Notepad, and then paste the clipboard table and save it as a text file. There will be lots of data in this file — more than you need in your compensation file. Edit out most of the lines (or make a new file) and keep only primary points: 20 Hz, 40 Hz, 60 Hz, 80 Hz, 100 Hz, 200 Hz, etc. Put this file in the MICCOMP subdirectory with a file extension of .mic because it's treated as a microphone compensation file. Load it for use by going to Options | Scaling and click the Select button under Microphone Compensation controls. Select the file you just created using the file selection dialog box. Now we're ready to reconnect the SKL Tunable Filter between the output and line-in. I followed the procedure just described to measure the sound card's response. I got the bandpass response on the screen and saved the data with a copy and paste. The plot was so close to the one shown in Figure 3 that I didn't see any need to include it.

Next, I selected a single-tone at one kHz instead of the sweep generator. The spectrum display looked about like Figure 4. I clicked THD and THD+N on the Utilities menu and two small boxes appeared on the right side of the screen displaying the distortion numbers. THD read 0.007 percent and THD+N was 0.062 percent. (The HP 3581A Wave Analyzer read 0.04 percent, about half way between AudioTester and SpectraPlus.) This illustrates the difficulty in making audio measurements on low distortion equipment. It's also the reason I like to have more than one measurement program so I can compare results. Now I'll go through the same measurement with Sample Champion Pro.

SAMPLE CHAMPION PRO

As I mentioned in the Comments in Table 1, Sample Champion Pro (SCP) is different, but the more I use it, the better I like it. AudioTester and SpectraPlus both have a real-time mode; turn on the signal generator and immediately see the frequency spectrum. SCP doesn't work quite the same way. First, you capture a sample of the line-in signal to a 'scope display. Then you select all or a portion of the 'scope waveform, and finally, you perform an FFT to see the spectrum in its own window.

In addition, SCP's signal generator is based on maxi-

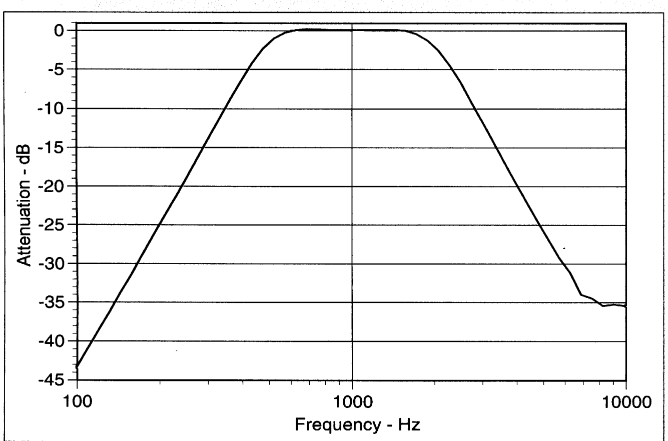

Figure 3. Response of rebuilt SKL-302 Tunable Filter. Lowpass cutoff frequency set to 2 kHz. Highpass frequency set to 500 Hz. Data saved as text file in *Audio Tester* software.

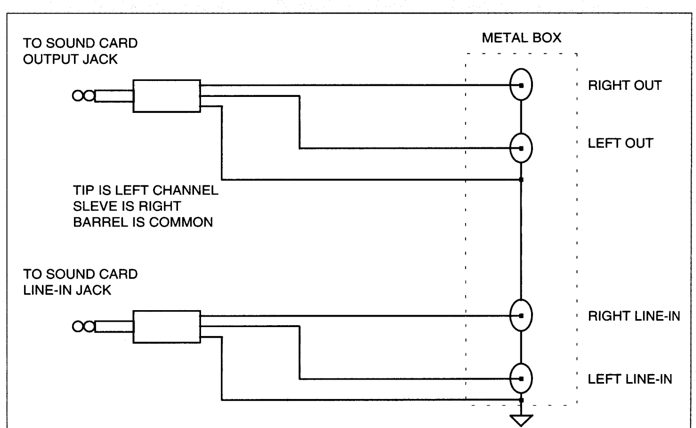

Figure 2. A sound card plug box makes audio testing easier. Use whatever connectors you prefer: audio jacks, BNC, pin jacks, etc.

mal length pseudo-random sequences (MLS), which is a good way to generate pseudo-random noise. (The signal is wide band, but it repeats at a rate dependent on the sequence length and clock rate. Hence the name pseudo-random noise.) This may seem strange to you, but it provides some measurement choices not found in the other programs. For instance, instead of using a sweep generator to measure the tunable filter's response, you can directly measure its impulse response and then FFT it to get the frequency response. The result is a nearly instantaneous bandpass plot that matches Figure 3. (The trial version of SCP does not let you save data to a file.)

A one-kHz tone is available for measuring THD and THD+N and I set this up as follows. I clicked Options | Settings and then selected 16-bits, mono, and a 44100 sample rate in both the Input and Output boxes. I chose a Blackman window and 4096 FFT length. Under "MLS type" I selected Amplitude calibration signal (one-kHz sinewave).

I opened a 'scope window by clicking the Scope Window icon and then I clicked Y Auto Scale. I clicked Measure in the menu bar and then Start/Stop sync Rec/Play. This captures about 100 mS of the one-kHz sinewave in the 'scope window (this time can't be changed in the trial version). I clicked Select All in the 'scope window to use the whole sample. I clicked FFT on the menu bar and then Selection in the Time Window under Spectrum. This displays the one-kHz tone much like Figure 4. I clicked FFT in the menu bar again and then Audio Quality plugin, and finally, the THD/THD+N tab. The THD reading was 0.006 percent and THD+N was 0.14 percent. The THD+N is high, probably because of the short duration of the one-kHz signal which tends to over-emphasize the noise. The level of each harmonic, up to the tenth, is also displayed on a small screen in the THD box.

Although I used the SKL Tunable Filter in all these examples, remember that you can measure any active or passive device that you can connect between your sound card's output and line-in. Let's look at another example.

ACOUSTIC LOSS THROUGH AN INTERIOR DOOR

It's easy to measure the loss through an interior (or exterior) door using the set-up in Figure 5. Just to show you how you can sometimes do a lot for a small price, I used the freeware WaveTools and my "Best Buy" rated AudioTester for this measurement. The procedure is just about the same for both programs. With the door open, start the signal generator using pink noise and start the spectrum analyzer. Adjust the power amplifier input level and microphone preamp gain for a non-overloading voltage on line-in. AudioTester has a red Overload indicator on the analyzer screen and WaveTools displays CLIP if the input level is too high. (You can load a microphone compensation file in AudioTester if you know your mic's frequency response. But WaveTools doesn't have this feature, so I didn't use it in AudioTester either. This demonstrates better agreement between the two programs.)

When you have the "open door" display on the screen, stop the analyzer and copy the data to the clipboard. Open Notepad (or other ASCII text editor), paste, and save the file as "open.txt." Without making any gain changes, close the door and restart the analyzer. Then stop the analyzer and capture the data to "closed.txt" and you are almost

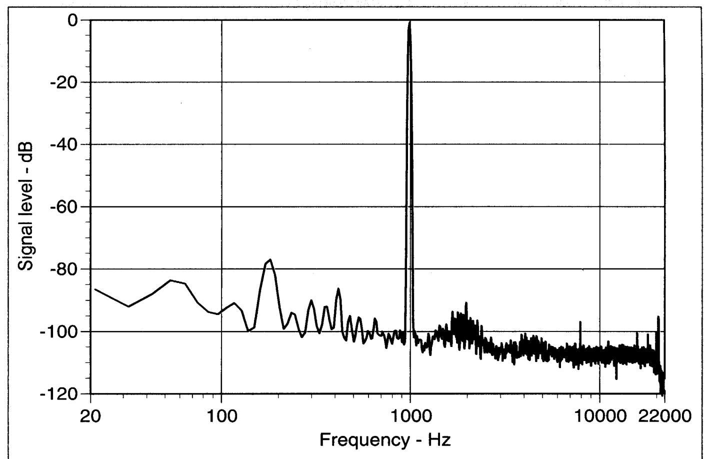

Figure 4. Output spectrum of rebuilt SKL-302 Tunable Filter with 1 kHz sinewave input. Spurious responses are more than 20 dB lower than original vacuum tube circuit.

done. You can import the data into a scientific graphing program or spreadsheet for plotting. My results are shown in Figures 6 and 7, and these are typical. Hollow core interior doors have only a small loss at low frequencies — they are best between about 1 and 10 kHz. Both programs let you average spectra for a smoother display, so set it to at least 20. (You can watch the display change as you vary the averaging number.) As a final example, let's look at one that's easy in theory, but somewhat difficult in practice.

ROOM REVERBERATION TIME

Reverberation Time (RT) is a measure of a room's "liveness" and different times are needed depending on the room's primary use, as shown in Table 2. Reverberation is the persistence of sound in a room after the sound source has been turned off. The time it takes for the sound level to decrease 60 dB is a common measurement and is abbreviated RT60.

To make this measurement, you fill a room with sound

GLOSSARY

Dynamic Range
The difference in amplitude level between the maximum and minimum useful signals. The maximum level is limited by the overload point of the DUT (or the sound card). The minimum level depends on the residual system noise or "noise floor."

FFT — Fast Fourier Transform
The Fourier transform of a time series of amplitude values is a frequency response. The FFT is a method of digitally performing the transformation.

IM — Intermodulation Distortion
The mixing of two or more pure tones in the DUT to produce spurious tones at other frequencies. IM is primarily a test for nonlinear distortion at the highest frequency used in the test.

Jitter
Frequency modulation distortion. That is, short term frequency changes of the test tone by the DUT.

THD — Total Harmonic Distortion
Non-linearity in the DUT produces harmonics of the test signal. That is, a one-kHz tone (the fundamental) would have harmonics at 2 kHz, 3 kHz, 4 kHz, etc. THD is the square root of the sum of the squares of all measurable harmonic voltage levels. THD can be expressed as in dB with respect to the fundamental level or as a percentage of the fundamental level.

THD+N — Total Harmonic Distortion plus Noise
THD as defined above added to the non-harmonic noise (broadband noise) from the DUT.

SNR — Signal-to-Noise Ratio
A dimension less number equal to signal voltage divided by noise voltage or signal power divided by noise power. It can also be expressed in dB by taking 20 * log(SNR) for a voltage ratio or 10 * log(SNR) for a power ratio.

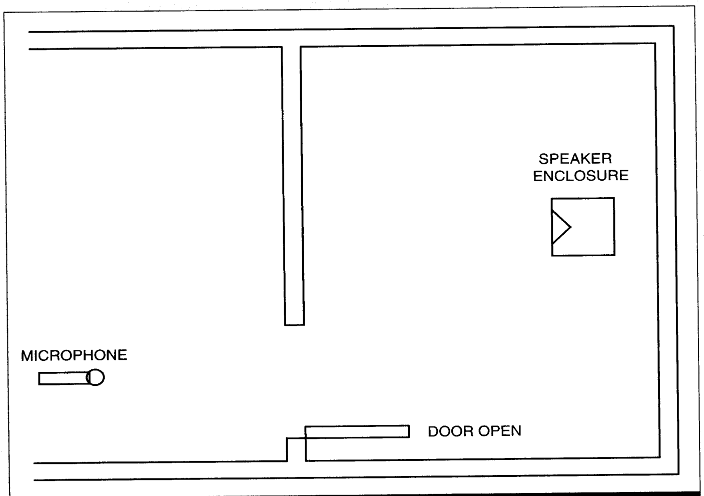

Figure 5. Floor plan of speaker and microphone placement for measuring the increase in sound loss from closing the door. When measuring an exterior wall or door, it's better to place the speaker outside to minimize microphone noise pickup.

from a loudspeaker and pick up the sound with a microphone. Start recording the microphone's output, turn off the sound, and then stop recording. By looking at the recording, you can find the time it took for the level to go down 60 dB, and that's RT60. Ideally, you should use an omnidirectional speaker and omnidirectional microphone to excite all reflecting surfaces, and to pick up all those reflections (called a diffuse sound field). But you can make-do with an ordinary speaker and microphone and your results will be useful, especially if you can move the speaker around and average several measurements. Start with the speaker in a corner, facing the corner about five or six feet away and tilted up about 20°. Place the microphone stand near the center of the room with the microphone oriented so the pick-up is as omnidirectional as possible — this would be vertical, open end up for most mics. You may need a power amplifier to drive the speaker and you will need a mic preamplifier because it's better to use your sound card's line-in, as it usually has a better SNR than the mic input. The problem using the above procedure is getting the needed 60 dB range. The limiting factors are your mic's preamp noise level and the residual room noise. You can lose up to 10 dB of your noise floor from a running refrigerator or fan, or from the wind blowing around the building. Also, you need a very low noise preamp, a high speaker sound level, or both. SCP provides a simple way to make the measurement, but first let's work through doing it the "hard" way.

I couldn't get a 60 dB range, as I didn't want to risk losing a speaker. With SpectraPlus in recording mode, I started the pink noise generator and adjusted the speaker volume for as high of a screen trace as I was comfortable with. Then I stopped the noise generator, and a couple of seconds later I stopped recording. (SpectraPlus has an application note on RT60, so I'm not going to repeat all of the measurement details here.) An unregistered SpectraPlus will make this measurement because it is fully functional. The restriction is a 30-day use limit.

The screen trace clearly showed the level decay from generator turn-off to the noise floor. If this change were 60 dB or more, I'd just use the cursor to measure time and level (in dB) and I'd be finished. But with less than 60 dB, I needed to do some more work. And I found I had some problems. First, the measurement signal is noise, so the screen trace is "noisy" and there isn't any way to average in the recording mode. Second, capturing data to the clipboard in recording mode gave me a WAV file and not level vs. time. So I used a utility program I wrote some time ago — wav2asc [8] — to convert the WAV file to ASCII text. Then I wrote a new program to make all data positive, perform a 100 point moving average, add time values (based on a 44.1 kHz sampling rate), and finally, write all this to a new file. I imported this file into DeltaGraph Pro and fit an exponential curve to the data. The resulting plot, shown in Figure 8, shows an RT60 of about 410 mS. This is a rather "live" room in spite of being carpeted and having an acoustical tile ceiling. One wall is covered with bookcases and storage cabinets with smooth wooden doors. And there is a large cable-weight exercise machine with lots of bare steel reflective surfaces. I didn't know the room was this live, but I'm not surprised by the measurement result.

As I've mentioned, SCP makes this and many other room measurements easy. There is a reverberation time application note in the User Guide, so I won't repeat it all here. But I will address a couple of details. First, you need a registered version with the Room Acoustics plug-in. The trial version won't make this measurement. Second, you start the test by measuring the room's impulse response.

When setting up the signal generator, the peak input level is shown on the right-hand side of the screen. I got the best results with this level at 12 to 15 percent, which

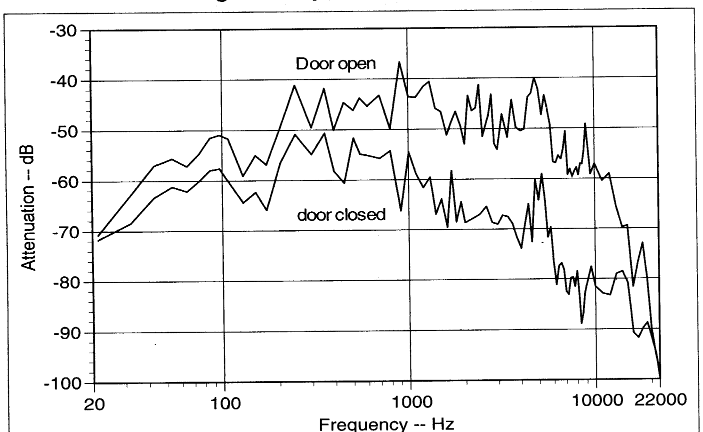

Figure 6. Sound attenuation with open and closed interior door. *Audio Tester* with pink noise. Note that this hollow core door loss is maximum from about 1 kHz to 10 kHz.

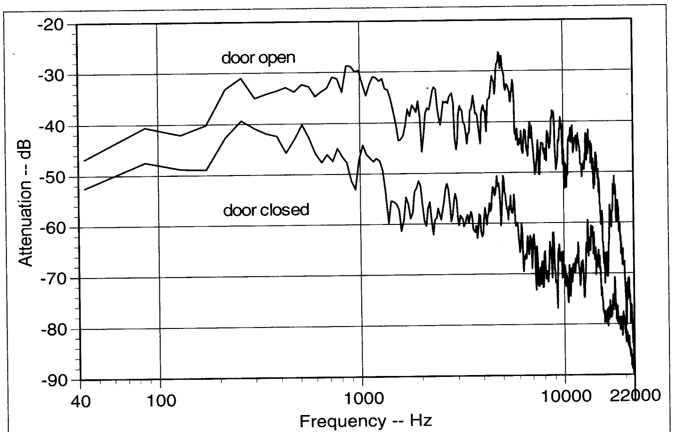

Figure 7. Open and closed interior door response using *Wave Tools* Signal Gen and Spectrum Analyzer. As expected, this graph is very similar to Figure 6.

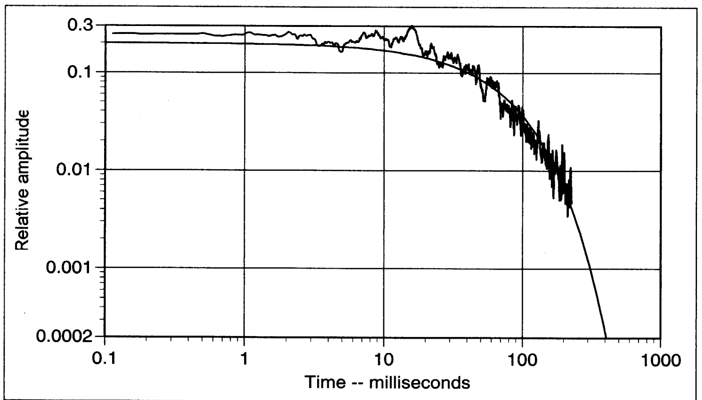

Figure 8. Measured RT60 using SpectraPlus, wav2asc.exe, rt60prep.exe, and DeltaGraph Pro with exponential curve fitting. Room acoustic measurements can be made quicker and more accurately with Sample Champion Pro. (An amplitude change of 1000 is equal to 60 dB.)

is only a little bit of sound out of the speakers. Then the Room Acoustics plug-in makes the measurement by reverse time integrating the impulse response, a method first introduced by M.R. Schroeder [9]. SCP gave me an RT60 of 330 mS, which is very consistent with the 410 mS, considering the amount of data manipulation I had to do to get any number at all.

Rooms with the same RT60 can sound very different, so RT60 isn't the whole story. Sample Champion Pro Room Acoustics plug-in also measures D50 (Definition), which is the ratio of early-arriving to total sound energy and C50 (Clarity), which is the ratio of early-arriving to late-arriving sound energy. These are automatically computer along with RT60 and can be displayed for the whole audio bandwidth or for octave or third-octave bands. In his Case Study #2: Critical Listening Room — Home Theater [10], Sam Berkow says that RT60 "has little correlation with subjective preference." What all this means is, you take all the measurements you can to help point you in the right direction, but then you have to make the room sound good to you (or to your client)!

There is also an argument that a small room doesn't have an RT60 [11]. This may be true if the room is too small to sustain a dense field of decaying reflections. But making these measurements is not an academic exercise — we do it to get some help in making the room sound better: where to use some acoustical tile or change the carpet, or maybe use some wall hangings. So, whether a room has a true RT60 is beside the point if our measurements are useful. There are many texts on architectural acoustics; two that I like are listed in Resources 12 and 13.

TABLE 2
Optimum reverberation times (500-1,000 Hz). For music performance, time should be longer at lower frequencies.

Recording Studio	0.5 second
Broadcasting Studio	0.5
Lecture Hall	1
Movie Theater	1
Home Listening Room	1
Multi-purpose Auditorium	1.5
Opera Hall	1.5
Symphony Orchestra Hall	2
Pipe Organ	3

There are several really neat, but somewhat pricy software programs that are specifically for sound control in buildings, that is, architectural acoustics. But I'm not going to get into that here, as it's off the topic of general audio testing. That would have to be the subject of another article.

TrueRTA, Version 2.0.1

I didn't use TrueRTA [14] in any of my measurements because it wasn't available when I was making them. But I have since learned that it's a very useful program. It has a sinewave generator covering 5 Hz to 22 kHz, as well as pink and white noise. It also has a digital level meter, a crest factor meter, and a dual trace 'scope. It performs spectral analysis in octave, third-octave, sixth-octave, 1/12-octave, and 1/24-octave bands. The version with the octave-band analysis only is free. And there are three other levels with increased performance at increased price. Level-4 with all the features is $99.95, and (I think) well worth it. Upgrades are free for existing customers.

Some Final Words

The User Guides and printed help files occupy hundreds of total pages for these programs, so I have only scratched the surface in covering their many features. However, I think I've made a good case for using a computer with a sound card and software for most, if not all of your sound testing needs. And with a good sound card the accuracy can be as good as you could get from some pretty high-priced lab instruments. Be sure to check out three more general sound-related web sites listed in Resources 15, 16, and 17. **NV**

RESOURCES

1. Sound card evaluations by Arnold B. Krueger can be found at: **www.pcavtech/soundcards/compare/index.htm**.

2. DSP Development Corp., 1 Kendall Sq., Cambridge, MA 02139. 617-577-1133, **www.dadisp.com**.

3. MathSoft, Inc., 101 Main St., Cambridge, MA 02142. 800-628-4223, **www.mathsoft.com**.

4. Audiomatica. Available from Orca Designs, 1531 Lookout Dr., Agoura, CA. 818-707-1629. **www.orcadesigns.com**. Clio Lite is available from Old Colony Sound Labs, **www.audioxpress.com**.

5. SYSid Labs, 1563 Solano Ave, # 211, Berkeley, CA 94707. 510-559-9075, **www.sysid-labs.com**.

6. MS-DOS program written by the author. You can download it from the *N&V* web site or from **www.zianet.com/tdl/magarts.htm**. "C" source code is included in the audiotst.zip file.

7. "New Life for a Vintage Audio Filter," *Nuts & Volts*, July 1999.

8. You can find this program along with the new program rt60prep and thd in audiotst.zip. All are MS-DOS programs written in "C" and source code is included.

9. "New Method of Measuring Reverberation Time," *Journal of the Acoustical Society of America*, 1965.

10. Case Study #2, SIA Software Company, Inc., 1 Main St., Whitinsville, MA 01588, **www.siasoft.com**.

11. TechTalk: "SMAART and Reverberation Time," *Live Sound International*, Feb. 2003, p.86. (**www.livesoundint.com**).

12. Leland K. Irvine and Roy L. Richards, *Acoustics and Noise Control Handbook for Architects and Builders*, Krieger Publishing Co., Malabar, FL. 1998.

13. F. Alton Everest, *The Masters Handbook of Acoustics, 2nd Edition*, TAB Books, Inc., Blue Ridge Summit, PA. 1989.

14. TrueRTA, True Audio, 387 Duncan Lane, Andersonville, TN, **www.trueaudio.com**.

15. **www.tracertek.com** sells quality sound cards in the less-than-$100.00 to $200.00 price range. They also sell some interesting and useful audio software.

16. **www.tune-town.com** is the Car Stereo-Parts Store site. Click on downloads and find a variety of interesting audio related software.

17. **www.hitsquad.com** is the Musician's Web Center site. Download hundreds of audio and music programs including trial versions, shareware, and freeware in dozens of categories. Also music charts, books, reviews, and discussions.