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Owning the Music We Long For

If it's true that music content is increasingly available in downloadable audio file formats and the industry is in transition to streaming services, there is also a significant market open to new high-quality physical format options—as long as they contain relevant new high-quality content.

And that seems to be the caveat.

For instance, I just received a press release announcing a new edition of an album available as a Limited Edition "hybrid" Super Audio CD (SACD) and deluxe 180g high-quality vinyl pressing. As the release states, "This will be the first time that the celebrated album has been made available on the SACD format."

Of course, the industry wants us to buy everything over and over again, in physical formats or digital files. While streaming is limited to the minimum quality supported by the majority of Internet services, high-quality original releases or properly remastered works can be attractive to music and high-end enthusiasts. But, otherwise, why buy the same music we already own on vinyl or CD, especially if there is no guarantee of a new rewarding experience?

Those physical releases will always enable us to extract the corresponding files, as we always did in iTunes… Right?

Interestingly, the debate has recently re-emerged. In 2014, the United Kingdom Intellectual Property Office announced plans to legalize the practice of "ripping" content from CDs and DVDs.

In most countries around the world, it is illegal to copy content that we bought on a CD/DVD or to convert those tracks into files for our own private use. That is why Blu-ray discs and some DVDs are copy-protected in the first place.

The UK regulations were subject to much debate and, in October 2014, the British government approved what most people believed to already be legal. Now, several music industry organizations in the UK have won a High Court judicial review, which considers copying for private use to be illegal, overturning the previous decision.

In the US, according to the Recording Industry Association of America (RIAA) website, "just because advances in technology make it possible to copy music doesn't mean it's legal to do so." "It's okay to copy music" as long as it is "not for commercial purposes," but there's no legal "right" to copy the copyrighted music, the reason why the owners of copyrights have the right to use protection technology to allow or prevent copying. Even though everyone—including the RIAA—agrees that this "should be permitted," the truth is, there is no "law" that makes it legal.

This also means that the iTunes file "library" we have on our computers is not legally "recognized." We don't actually own the music stored in those files, in the same way we don't own the music stored on a CD or a SACD. We are just entitled to listen to it.

And that's precisely why I think we are entitled to have the best quality possible when we pay for a "service" such as iTunes. That is, if you paid to download an MP3 file from Amazon, you should have the right to download it in the original uncompressed or high-resolution audio (HRA) format—as long it's the same exact original recording.

That's a legal "Pandora's box" that the content "ownership" debaters want to avoid. I think HRA is also a consumer rights debate and not a subjective discussion regarding the consumers' ability to "hear the difference."

That's why I found the recent statement from Neil Young's digital music distribution venture very interesting. Pono Music's renewed promise specifically discusses "Free Album Resolution Upgrades" and states, "all music purchased at PonoMusic.com will now be upgraded for FREE when a label offers a higher resolution upgrade of the same recording."

That should always be the case and it will be interesting the see the industry's reactions.

João Martins
Editor-in-Chief

The Team

In Memoriam—Hugo Van haecke (1951-2015)

Hugo Van haecke, our company's president and publisher, passed away on August 19, 2015, in his hometown of Denver, CO.

A publishing industry veteran and an exceptional manager, Hugo was instrumental in the transition of our business from its early foundations into a future-ready organization—managing acquisitions, mergers, and restructuring the companies to ensure the continuity and evolution of the titles (*audioXpress*, *Voice Coil*, *Loudspeaker Industry Sourcebook*, and *Circuit Cellar*) as well as the company's book publishing business.

Born May 5, 1951, in Antwerp, Belgium, Hugo was the second of four children born to Henri and Alice Van haecke-Verrycken. He was an eager learner from the start, and through the teachings of his older brother, Alex, knew how to read, write and do basic math before he entered the first grade. During his formative educational years, Hugo was a passionate student with a desire to learn, but struggled with the restrictive nature of the educational environment.

In 1973, Frank de Winter gave Hugo a job at Old Charley, a wine wholesale company, to assist in bookkeeping, accounts, and customer relations. Frank taught Hugo everything there was to know about accounting, bookkeeping, and finances. Hugo excelled in this environment and the experience reinforced his lifelong belief that wisdom and success are fostered through real word experiences.

Hugo moved to the US with his family in 1999, while working for Wolters Kluwer, a publishing group based in the Netherlands. During his career, he managed businesses and companies around the world. In 2005, Hugo started his own venture, working with several companies as a consultant, a financial advisor, and president.

Hugo had semi-retired to devote more time to his family and especially his granddaughter, while continuing to be at the helm of our business. A lover of family, life, and friends, Hugo was a great manager, a proud Million Miler traveler, and someone who inspired us all and will be greatly missed.

Hugo Van haecke is survived by his wife, Erna Van Meerbergen; his children Margo Valaika and husband Chris Valaika; Thomas Van haecke and wife, Emilie Van haecke; his granddaughter, Alyse Valaika; his brother, Alex Van haecke; his sister Lieve Van haecke; and numerous other relatives.

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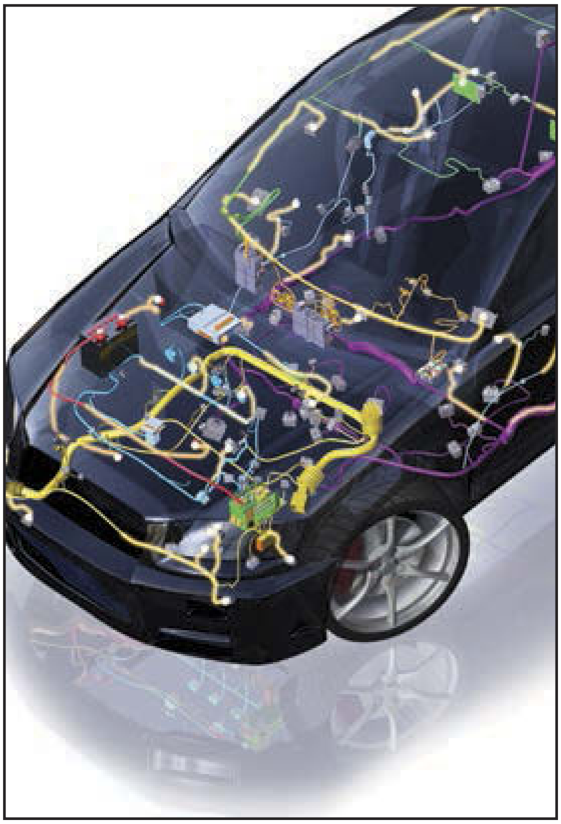

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Car Audio

Industry Standard Set to Change

The Benefits of 48 Volts for Autosound

The possibilities for car audio are going to be almost limitless when the upcoming 48 V industry electrical standard is finally implemented. This article discusses the potential consequences.

By
Mike Klasco and Steve Tatarunis
(United States)

The concept of a higher voltage for vehicle electrical systems is not new, but prior attempts to establish a new standard 10 years ago failed due to component costs and the lack of a driving force for development. However, in today's market where any CO_2 emissions reduction represents a cost savings to the OEM, the industry's motivations have changed. Energy and power demands for future vehicles will continue to increase, so the availability of power at a higher voltage level without major manufacturing or cost penalties means significant benefits for the manufacturers and new opportunities for the consumer.

In 2004, the US auto industry was to begin its move from the 12 V battery (14 V at the generator/alternator) electrical systems toward a 42 V standard, with 25% to 50% of new vehicles incorporating 42 V electrical architectures by 2010, and all new cars by 2020. The higher-voltage system slashes weight, improves fuel economy, permits the replacement of many mechanical parts with electrical ones (electric power steering), powers all sorts of new gizmos (e.g., seat heaters, cup warmers/heaters). It would also improve efficiency for all automotive electrical devices, from opening the door to higher audio amplifier power with far less complexity than the current crop of 12 V, high-power amplifiers. Not only does this translate to lower cost and more compact aftermarket power amps, even factory or aftermarket head units could inexpensively integrate more than 100 WRMS per channel power.

The Old New Standard

The standard US and European automakers and suppliers originally agreed to more than 10 years ago was 36/42 V. Specifically, it triples the current voltage for battery output (from 12 to 36 V) and generator voltage output (from 14 to 42 V). Vehicles with these electrical systems were to have dual batteries (12 V and 36 V) or at least a stepped-down, 12 V circuit on board so that all electrical components did not have to change over all at once.

Since 2011, starting with a joint statement by German automakers, there was a consensus to move to 48 V instead of the previously considered 42 V. The 48 V standard initiative was widely supported by Continental AG and semiconductor suppliers like, Bosch, NXP and Infineon, among many others. Yet today, the push is even more intense, stimulated by the need to increase fuel economy, and by a seemingly insatiable demand for electrical and electronic accessories in vehicles—from heated windshields and seats to dash-mounted navigation systems and rear-seat TVs. Currently, bulky wiring harnesses, heavy power motors, and electronics that are subject to voltage sags and spikes are some of the compromises with 12 V systems (see **Photo 1**).

You can expect the big SUVs, premium luxury cars, and hybrid gas/electrical vehicles to be among the first vehicles to switch to this new electrical system. Moreover, the potential of breakthrough components (e.g., electronically actuated valves, flywheel starter/

beyerdynamic

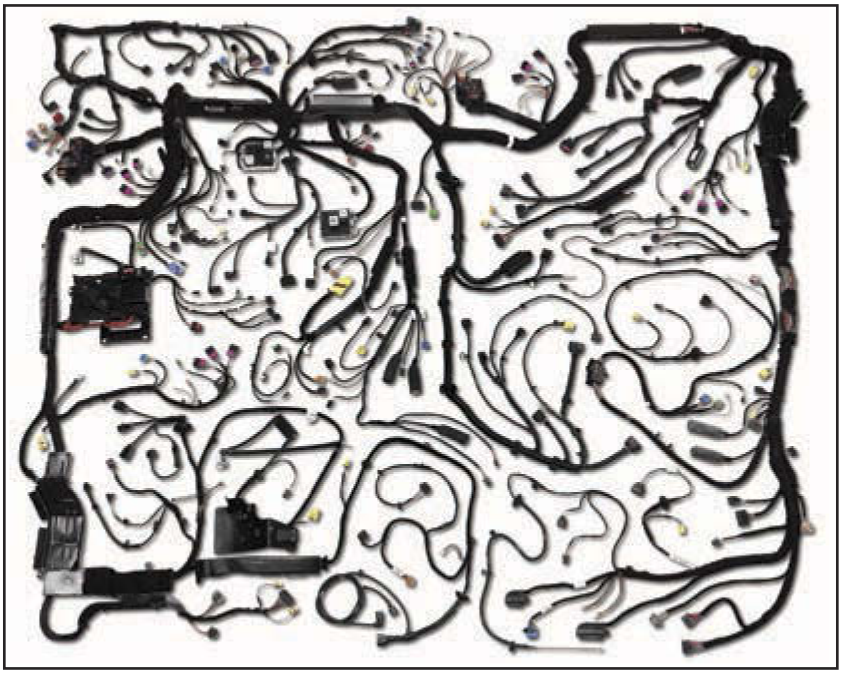

Photo 1: This is an example of typical wiring harness assemblies from a modern 12 V automobile. (Photo courtesy of Delphi Automotive)

alternators, electric brakes, active suspensions, and four-wheel steering systems) continues to go largely unrealized because there just isn't enough electrical power available from a 12 V system. More volts mean rethinking everything from light bulbs to the audio systems (see **Photo 2**).

High Voltage and Its Impact on Autosound

Very low impedance speakers, parallel woofers, and bridged amplifiers are commonplace in today's autosound designs. We've become so accustomed to these expensive kluges that we don't even notice their excessive cost, sound degradation, and awkward complexity. The design and selection of the amplifier circuits in autosound are based on the premise of a 12 V power supply. The amplifier's purpose is to increase signal output. This signal increase can be measured in watts, the multiplication of voltage and current (amperes). Since the power supply is limited to 12 V, the higher power must come from higher current output. To pull the needed current from the amplifier, the speakers must be low impedance.

Think of the power from the amplifier as water going through a pipe. Increasing the pipe's diameter halves the resistance to the signal so more power can flow. The pipe's diameter is analogous to the electrical resistance—the larger the pipe, the less resistance for this power to pass through.

Even the autosound speaker designer has to work around 12 V limitations. The gauge of wire used in the speaker voice coil is heavier than optimum to keep the speaker impedance low. So, there is either a lot of woofers in parallel or a woofer with a very low impedance voice coil. With the 48 V systems, and speaker impedances shifting upward to the range commonly found in professional sound and home theater systems, speaker cables will work better and have less signal loss than with ultra-low impedance speakers (see **Photo 3**).

Consider that for sending power over long distances, even the power companies use high voltage power lines, and step the voltage down at the power sub-stations. In electrical power (and audio amplifier signals), there exists a phenomenon known as I^2R losses, where the power loss (through generation of heat) is worse for high current than for high voltage. So, 12 V amplifier systems send high current (lower voltage) audio signals and have greater loss for the same run to the speakers than a 48 V amplifier system sending the same power, except with lower current and higher voltage. This is true for all the wiring harnesses in the car. The switch to higher voltages means that to carry the same power, the wire will carry less current, and therefore, the wires will get thinner and lighter.

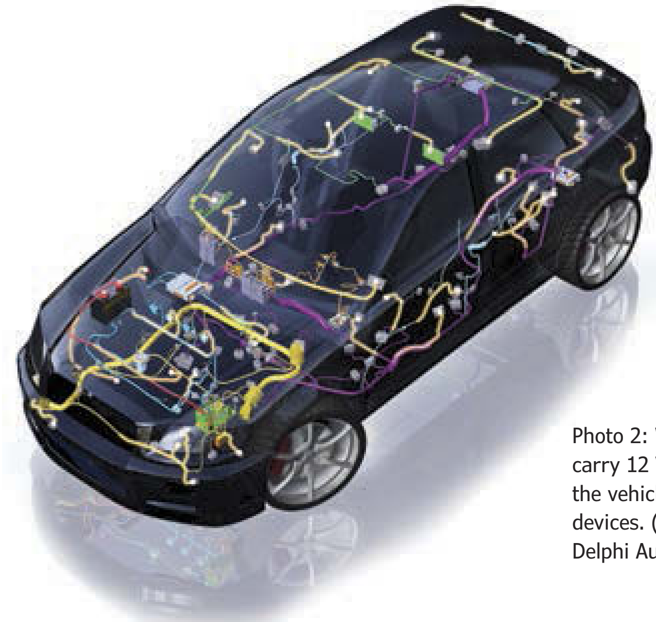

Photo 2: Wiring harnesses carry 12 V throughout the vehicle to a myriad of devices. (Photo courtesy of Delphi Automotive)

The Existing Approach

The existing high-current output approach came out of necessity for autosound aftermarket amplifiers. The technique uses many parallel high-current output transistor devices, which is both a complex and an expensive solution. Bridging is another common way to pull more power from existing amplifiers on 12 V electrical systems. In this case, two amplifier channels work as a single audio channel. In this approach, the audio signal is split down the center, with the upper half of the signal going to one amplifier channel and

Photo 3: A subwoofer's voice coil wire gauge and the voice coil gap (e.g., this one from Earthquake Sound) can be redesigned and optimized for use with 48 V systems. (Photo courtesy of Earthquake Sound)

the lower half going to the second amplifier. On the output of the amplifier, the two halves are "bridged" together to "swing" the higher voltage. This works fine, and is a common solution, but at the cost of double the amplifiers for a given number of channels! For a given power supply voltage, this roughly provides double the wattage. Another consideration with bridging is that the minimum speaker impedance load you can connect to the amplifier cannot be as low as the same amplifier when not bridged. So if your amplifier could normally handle 2 Ω when it is bridged, it can only handle approximately a 4 Ω load.

Beyond medium power 12 V amplifiers, above around 70 WRMS per channel, designers run out of tricks and must use a DC/DC converter within the amplifier. This is a circuit of high complexity that adds significant cost to the amplifier. The 12 V (actually closer to 14 V) power input from the alternator/generator is fed into a DC amplifier that converts the 12 V to a higher voltage.

There are quite a few hidden costs from components such as sophisticated power ICs and other trick semiconductors, and expensive wound inductors. Still another aspect is that DC/DC converter designs require a manufacturer that knows what it is doing, which eliminates much of the Chinese subcontractor autosound amplifier manufacturing industry. The new 48 V standard avoids all this, and enables even the Chinese autosound vendors to jump into the high-power autosound amplifier game, which is great news and will represent cost savings for the consumer.

Since the real electrical system in the car is now about 14 V, and the real electrical supply in the next generation cars will be 48 V, the easily achievable power level will be more than three times what they are now—more than 200 WRMS per channel if all the tricks (e.g., bridging the amplifiers and low impedance speakers) are used. But even 100 WRMS can be attained with costs close to today's amplifiers with half the power (see **Photo 4**).

Remember, the electrical "power" is the product of amperes (current) times voltage, so an amplifier that takes 12 A of current at 12 V requires only 6 A at 24 V or 4 A at 36 V, and so on. So tripling the voltage effectively cuts the current by two-thirds, while still providing the same power capability. This creates opportunities to downsize the amplifier and, perhaps, rethink the amplifier architecture. For example, instead of offering 70 W into a 2 Ω load, the amplifier can be limited to 4 Ω loads, but still reach 70 W (at 4 Ω) and the number of output stage transistors in parallel can be reduced—all with a smaller size and lower cost.

Super Power Head Units

Has anyone ever seriously considered the amplifier in the head unit as a viable option for high-power sound systems? That will soon change and become not only a real challenge for the aftermarket industry but also an opportunity. The typical head unit struggles to deliver 15 WRMS per channel, due to lack of internal real estate for big and hot amplifiers in a 1- or even 2-DIN format radio/MP3/CD unit.

Some History and Background

The 12 V configuration has not always been with us. Up until the mid-1950s, US cars used 6 V systems. Many European brands continued to produce 6 V cars through the 1960s. General Motors made the switch in 1955, and Volkswagen was the last major player to jump to 12 V, back in 1966. The larger trucks moved to 42 V systems long ago. It is generally agreed that safe voltage is limited to 60 V. As voltage levels rise, however, the higher voltage can more readily jump the air gap between electrical conductors and "arc."

Arcing in a 48 V environment is a much bigger headache than 12/14 V systems—the energy in the arc is much higher. The solution is that a lot of subtle redesign of parts has to be done to prevent arcing. The new 48 V standard was driven by the industry wanting the highest possible voltage, with the most safety. But higher voltage beyond a point can have its own costs—a European safety regulation specifies that any voltage above 60 V needs to have more heavily insulated wires and connectors. That would add weight and defeat many of the advantages gained in other areas.

More Benefits

The new electrical system architecture will likely help provide a seamless information link between home, office, and vehicle by combining the Internet, GPS, video, audio, cellular, and satellite info systems.

Other benefits of the extra electrical power afforded by the switch to 48 V will include driving and braking and driverless parking-by-wire systems, which replace a car's bulky hydraulic and mechanical parts with electronics, enabling electronic brakes, steering, valve control, air conditioning compressors, and power steering pumps.

Degrees of hybridization in powertrain enhancement include stop-start technologies, automated manual transmissions (AMTs), engine downsizing and down-speeding, changes in combustion cycles, supercharger electrification, energy recuperation, auxiliary electrification, and chassis and suspension systems. As power and performance demands for vehicles, especially self-driving vehicles, continue to accelerate, the need for computing power, control electronics, mechatronics and real-time software will increase the electrical load on vehicle architectures.

There are dozens of fractional horsepower motors on a car or a truck. Electrically controlled mirrors require multiple motors for changing tilt and angles. Adding car sound systems, plus electronically controlled engine management systems, electric heating, ventilation, and air conditioning systems has resulted in big increases in power needs. Vehicle electrical demands vary considerably by type, size, and model of vehicle as well as by component.

Going to 48 V would enable using a lighter, thinner conductor to handle the equivalent loads. Furthermore, having a high voltage electrical system permits the OEMs to add even more comfort and convenience items. But beyond those considerations, higher voltage permits the car manufacturers to achieve important goals in improving fuel economy, emissions and safety by replacing belt-driven and crankshaft-driven components with electronically controlled electric motors and solenoids. One of the real benefits of jumping to 48 V systems, especially for hybrid vehicles, is the vehicle's ability to offer regular 110 V electrical outlets.

The initiative of a decade ago promised 42 V systems in Jags, Lincolns, and some SUVs, but never saw the light of day. Today, several companies are leading the charge.

Audi has announced it will upgrade part of its vehicle electrical system from 12 to 48 V. The company also has showcased the scope of the 48 V system in the A6 TDI and RS 5 TDI concepts (see **Photo 5**). Both models are fitted with an electrically powered compressor.

LG Chem Power says it is working to develop 48 V batteries for the North American micro-hybrid market. Valeo has developed an electric turbocharger that eliminates turbo lag. The new turbocharger is powered by an electric motor instead of exhaust gases.

Photo 4: High-power amplifiers (e.g., this design from Kenwood) will become less expensive after being redesigned for use with 48 V. (Photo courtesy of Kenwood Electronics)

Johnson Controls has introduced a compact 48 V lithium ion starter battery for micro hybrids, and Continental's "48 Volt Eco Drive" system will go into production with European car manufacturers for the first time in 2016.

The federal government has regulated the fuel economy of cars and light-duty trucks for decades, with the latest 2012 rules dramatically increasing fuel economy and decreasing greenhouse gas emissions.

A 2010 rule raised the average fuel economy of new passenger vehicles to 34.1 miles per gallon (mpg) for model year 2016, a nearly 15% increase from 2011. A second rule, finalized in 2012, will raise average fuel economy to up to 54.5 mpg for model year 2025, for a combined increase of more than 90% over 2011 levels.

The standards also will reduce the carbon intensity of these vehicles by 40% from 2012 to 2025. Vehicle manufacturers will need every trick to reach these standards and changing to 48 V systems is one of them.

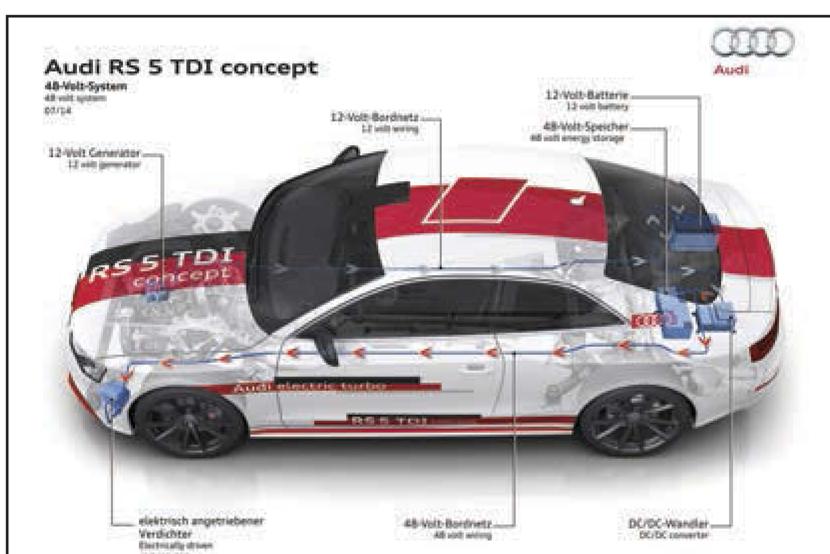

Photo 5: The Audi RS 5 TDI concept uses a hybrid 12/48 V electrical system. (Photo courtesy of Audi AG)

Sound Cards for Data Acquisition in Audio Measurements (Part 5)

The User

In this article series, we are examining ways to create a low-cost system for lab-grade audio electronics measurements. To this point, I've looked at the hardware and software bits of the sound-card-based measurement system. The last remaining piece is the user.

By
Stuart Yaniger
(United States)

One of the hazards of computer-based measurement systems is their uncanny ability to acquire impressive amounts of bad data in a surprisingly short time. The only way to minimize this is by understanding the measurement, choosing the right measurement parameters, and recognizing the artifacts in the data that are due to the parameter choices. Unfortunately, this is the hardest part of the process and unavoidably requires some understanding of the math.

I'll try to make this as painless as possible and make clear what the results of the math mean, rather than going through the derivations with the utmost of rigor. The power of the computer lets us freely use mathematical transforms, which are powerful tools if understood and correctly used. The most basic and frequently used one is the Fourier Transform (FT).

Fourier Transform: Threat or Menace?

Signals can be characterized in two ways: amplitude (usually voltage) vs. time or amplitude vs. frequency. The FT is a mathematical means of converting from one view to the other. For a continuous function of time a(t), the Fourier Transform F(f) is defined as:

$$F(f) = \int_{-\infty}^{\infty} a(t)\exp(-j2\pi ft)dt$$

where a(t) is the signal in the time domain. Note that the integral is over an infinite amount of time and the signals are assumed continuous (we'll come back to this shortly).

The math aside, it is important to understand that the time and the frequency views are exactly equivalent. All the same information is contained in them and they may be interconverted at will. The choice of frequency or time domain is arbitrary and will depend on what's more convenient for the particular phenomenon you're testing.

For example, if I wanted to examine an amplifier's stability, I might run a square wave test and, in the time domain, look for overshoot on leading and trailing edges or even ringing on the top of the waveform. There would be no particular reason to convert the signal to the frequency domain for analysis, even though I could get the same information.

On the other hand, if I wanted to understand an amplifier's distortion characteristics, I could apply sine waves of different frequencies and levels, then squint at the sine waves on the amplifier's output and try to see if they looked different. The problem is that eyeballing the results on an oscilloscope screen does not give much sensitivity: distortion has to be pretty gross to be visible. It's also not very quantitative. And to compound the difficulties, the type of distortion will not be clearly evident (although the symmetry between the top and the bottom of the sine wave will give some indication of whether the predominant distortion is even or odd order). A much better way to quantify and characterize the distortion is by taking the amplifier output and transforming it into the frequency domain, creating a frequency spectrum. The basic Fourier theorem that we learn in freshman math is that any periodic signal, irrespective of its shape, may be broken down into a fundamental sine wave frequency and harmonics (that is, sine waves with frequencies that are whole

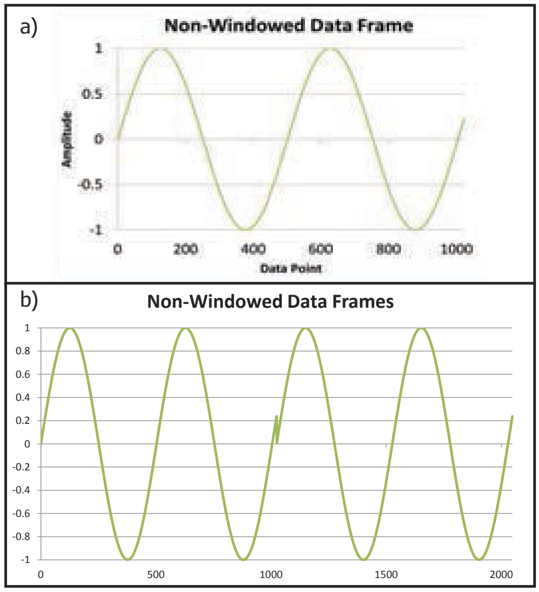

Figure 1: The sine wave is shown in a sampling frame (a). Note that because the sine wave and frame length aren't synchronous, the sine wave values are not zero at the frame edges. Joining frames to force periodicity causes discontinuities (b).

number multiples of the fundamental frequency); the relative magnitudes and phases of the harmonics determine the waveform's shape in the time domain. So the frequency spectrum will be convenient for quantifying the linearity, showing what new frequency components (harmonics) are present in the output signal that weren't present in the input signal and what their relative sizes are, which is the basic definition of distortion. An undistorted sine wave will have only one frequency component, the fundamental frequency. Distortion adds new components at the harmonics.

The old-fashioned way of determining the frequency spectrum was with a wave analyzer. Essentially, a wave analyzer is a narrow tunable bandpass filter attached to an AC voltmeter. A signal is fed to the input of the device under test (DUT). The DUT's output is connected to the tunable filter's input, and the filter's frequency is swept. The AC voltmeter records the output voltage as a function of frequency. As it sweeps through the harmonic frequencies, a spectrum can be recorded. The most popular high-quality wave analyzer was the Hewlett-Packard 3581A, which, although expensive in its day, can be found online for remarkably low prices. It's slow, inconvenient, and not terribly sensitive, but it's my back-up instrument that I use to check my work when readings I get with a sound card look funny.

With the advent of electronic computing power, FT methods became a better solution. In particular, the Fast Fourier Transform (FFT) was an efficient algorithm for computing the FT of a sampled waveform in a finite time window. [It should be noted that we've silently slid from a Fourier transform (continuous function) to a Discrete Fourier Transform (set of sampled points); that's a bit beyond the scope of this article, but I can hear engineers and mathematicians wincing.]

In a basic FT measurement, a waveform is sampled in the time domain (just like the digital recording of music) and the time domain record for the length of the measurement (called a frame) is Fourier-transformed using an FFT to give a frequency spectrum. That's the simple view—the reality starts getting a bit more complicated.

Just as in digital recording, we sample a signal at time intervals t_S. Equivalently, we can say that we sample at a frequency of $f_S = 1/t_S$ (sample rate). The numbers corresponding to each sample (time point and measured amplitude at that time) can be stored in the computer memory or in a file as a data record. The number of data points in the data record is equal to the sample rate multiplied by the measurement time (which is ideally infinite, but is generally limited to a finite interval). For example, if the sample rate is 48 kHz and we sample for 0.5 seconds, the data record (which we will call a "frame") will contain 24,000 points. For most software, one selects the number of data points in each frame, from which you can calculate the measurement time:

Measurement Time per frame = number of data points/sample rate.

The FT of that time record of the samples will contain half the frame's number of time/amplitude data points (remember that the frequency range is Nyquist limited to half the sample rate), so we can calculate the distance between the FT's frequency/amplitude points (and frequency resolution) from:

Frequency Resolution = 2x frequency range/number of data points = sample rate/number of data points.

What are the implications of these basic relationships? Although we draw the frequency spectrum as a continuous line, in reality, each frequency is not actually a point, but rather an interval of frequencies with a width equal to the resolution. It is referred to as a "bin," largely because all amplitude components within that interval are essentially grouped together as a package. You can think of a bin as a narrow band-pass filter. As the number of samples in a frame increase (for a given frame length in time), the width of the bins narrow. In the limit of an infinite number of points in a frame, the bins shrink to points, resolution becomes infinitely fine, and the frequency spectrum is continuous.

If the sampled signal is a pure sine wave, the FT of the data record ideally will be a single spike at the fundamental. If we distort the sine wave a bit, other spikes at whole-number multiples of the

About the Author

Stuart Yaniger has been designing and building audio equipment for nearly half a century, and currently works as a technical director for a large industrial company. His professional research interests have spanned theoretical physics, electronics, chemistry, spectroscopy, aerospace, biology, and sensory science. One day, he will figure out what he would like to be when he grows up.

fundamental frequency appear, as expected. The spikes at frequencies that are even whole-number multiples of the fundamental are called (logically) even-order harmonics, and analogously, the harmonics appearing at odd-number multiples are called odd-order harmonics.

Some rules of thumb for relating frequency and time domain:

- Odd-order distortion components cause symmetrical distortion around the horizontal (time) axis in the time domain. So, for example, if you see a flattening of both the tops and the bottoms of a sine wave, you know that the frequency spectrum will be dominated by odd-order harmonics. The extreme example is a square wave, which contains all possible odd-order harmonics and no even-order ones.
- Even-order distortion components cause non-symmetrical distortion around the time axis. So, for example, if you see flattening of the tops of a sine wave but not the bottom, the frequency spectrum is dominated by even-order harmonics.
- "Sharper" features in a waveform (e.g., the leading and trailing edges of square waves or the small zero-crossing notch from crossover distortion) are indicative of high-order harmonics (i.e., where the whole number multiples are large).

Windows (Not the Microsoft Kind)

A real waveform and its set of samples is not infinite in length. It will have a start and a stop. The longer the sampling window, the lower the frequency that can be measured. This makes intuitive sense if you consider that you want at least one full cycle of the lowest frequency of interest to fit within the sampling time. So for a 10-Hz wave, you want a sampling window that is at least 0.1 seconds long. Let's say that we have a sample rate of 48 kHz. Then, the minimum frame to measure 10 Hz would be 4,800 data points. The number of data points also sets the measurement resolution. In this example, each frequency point will be 10 Hz from its neighbor, so that one will not be able to resolve a frequency more precisely than that.

You'll often hear the objection, "Fourier transforms are only good for periodic or repetitive signals." In a formal sense, since the FT works on periodic signals, the finite length of the frame conceptually represents the period. Then, the periodic signal (in a mathematical sense) is an infinite string of the frames. This brings up another complication—continuity. The waveform's values at the point where the conceptual frames join up have to be equal or the resulting spectrum will be misleading because of a discontinuity where the waveforms meet.

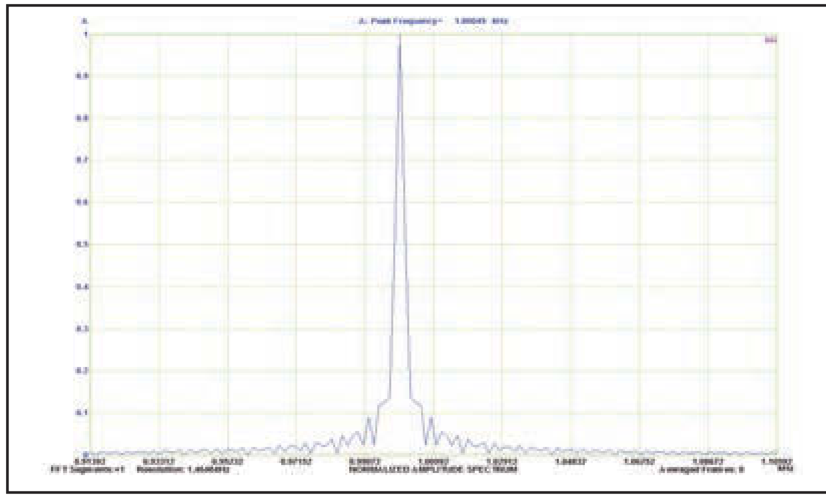

Figure 2: Ideally, the spectrum would just be a single line at the sine wave frequency, but the effect on the frequency spectrum shows discontinuity at the frame edges.

To illustrate, if the measured signal is a sine wave, it's unlikely that its period will be such that its zero-crossings end up perfectly at the start and stop of the frame. It's far more likely that the frame edges will correspond to some other point on the waveform (see **Figure 1a**). When we conceptually join the frames to make a periodic signal (with the period being the time span of the frame), the discontinuities where the frames meet up are evident (see **Figure 1b**). These discontinuities are a function of the finite sampling window and aren't part of the actual waveform we want to analyze.

What's the effect of this discontinuity? In the unlikely event that the sine wave is at a zero crossing at the beginning and end of the frame, the discontinuity disappears. In the far more likely event that the signal will have some non-zero value at the edges of the frame, the frequency domain spectrum obtained from the FT will no longer be a single line at the sine wave frequency. Instead, it will be broadened

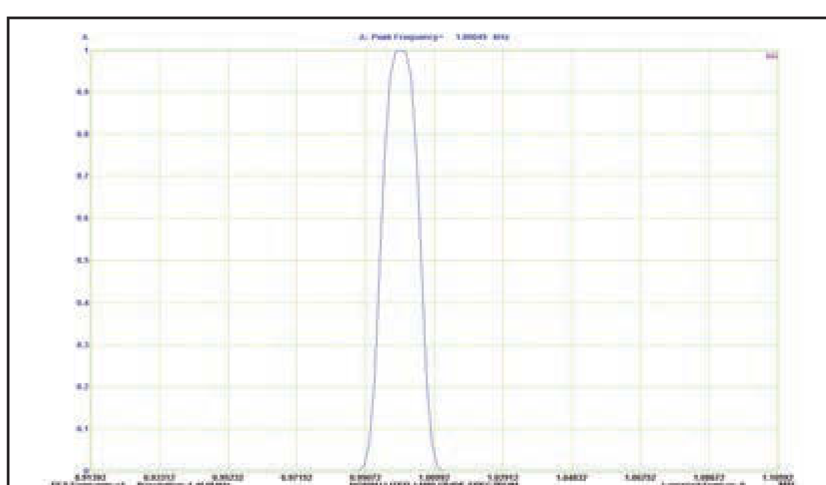

Figure 3: Here is a Fourier Transform (FT) spectrum of a sine wave with a Flat-Top window.

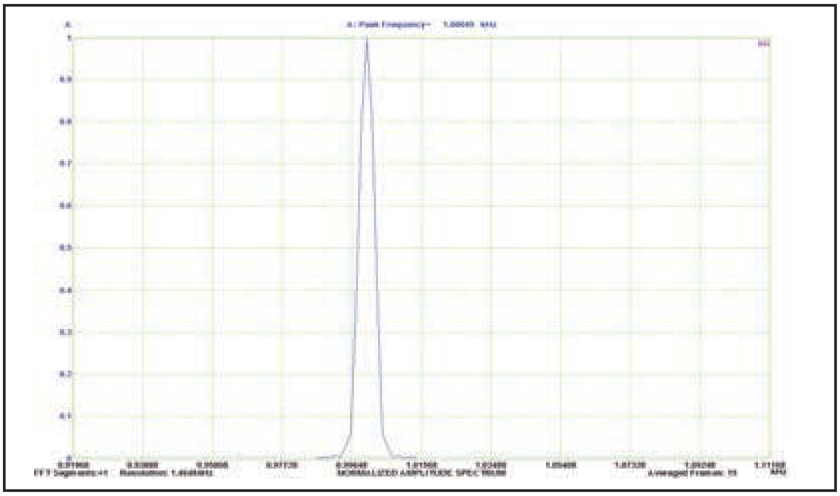

Figure 4: This is a Fourier Transform (FT) spectrum of a sine wave with a Hanning window.

from the spectrum "leaking" into adjacent bins (see **Figure 2**), which smears out the spectrum. Note that the vertical axis in **Figure 2** is linear, so the "wings" on either side of the fundamental frequency are only 20 dB down. The apparent noise floor will also be significantly higher from that same leakage. Clearly, this is not desirable for precision measurement!

To overcome this problem, we use a trick called "windowing" or "apodization." Ideally, the sampled signal would be zero at the frame edge, so we can multiply it by a function that is in fact zero on each edge of the frame, but unity in the center of the frame. Generally, the variation from edge to center is smooth and continuous. Most common window functions look more or less Gaussian to the eye (though they are not at all the same mathematically).

Since we're now putting a new function (the window) into the signal data record, there will be both good and bad effects on the calculated spectrum, and to maximize the good and minimize the bad, there are a variety of possible window functions. The trade-offs are between side-lobe suppression, frequency resolution, and amplitude accuracy. Therefore, the specific choice of window function will depend on what it is you're trying to measure. I have compiled a non-comprehensive survey of some common window functions.

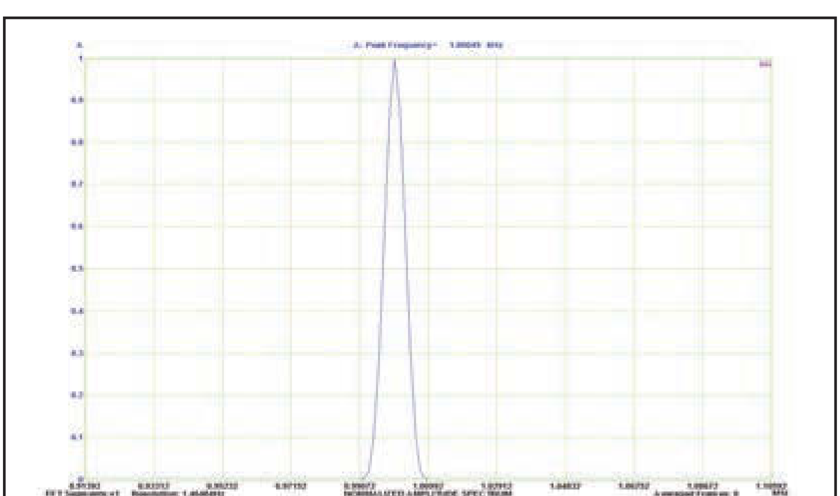

Figure 5: Here is a Fourier Transform (FT) spectrum of a sine wave with a Blackman-Harris window.

Rectangular: This is actually no window at all, and we've seen the effect on a sine wave spectrum. Sometimes a rectangular function does have its uses. It is likely the window of choice if you're looking at a frame containing a single transient event, since we do not want the calculation to vary depending on where in the frame the transient is sampled. Likewise, it's the window of choice for measurement of random noise.

Flat-Top: Despite its name, the flat-top does not have a flat top. Sharp peaks in the frequency domain show flattened tops with this window, thus the name. It is derived from a sum of cosines:

$$W(n) = 1\text{-}1.93\cos\left(\frac{2\pi n}{(N-1)}\right) + 1.29\cos\left(\frac{4\pi n}{(N-1)}\right) - 0.39\left(\frac{6\pi n}{(N-1)}\right) + 0.028\left(\frac{8\pi n}{(N-1)}\right)$$

where N is the number of samples in the frame. An FT of a sine wave using the flat-top window is shown in **Figure 3**. A flat-top window is a good choice if you're trying to accurately measure the amplitude of a frequency component. It is a poor choice if you're trying to resolve two closely spaced (in the frequency domain) signals, since it broadens the spectral lines.

Hanning: The Hanning is a good general purpose window, defined by the rather simple function:

$$W(n) = 0.5\left[1 - \cos\left(\frac{2\pi n}{(N-1)}\right)\right]$$

It does not have the amplitude accuracy of a flat-top, but can resolve closely spaced signals of similar amplitude. **Figure 4 s**hows an FT of a sine wave using a Hanning window. The Hanning should not be confused with the Hamming, which is similar window, but not the same.

Blackman-Harris: The Blackman-Harris window is defined by:

$$W(n) = 0.36 - 0.49\cos\left(\frac{2\pi n}{(N-1)}\right) + 0.14\cos\left(\frac{4\pi n}{(N-1)}\right) - 0.01\cos\left(\frac{6\pi n}{(N-1)}\right)$$

It has better amplitude accuracy than the Hanning, but not as good as the flat-top. It's most useful for resolving two closely spaced signals of different amplitude (e.g., power supply intermodulation sidebands near the fundamental) because of its superior side-lobe suppression. **Figure 5** shows an FT of a sine wave using a Blackman-Harris window.

Kaiser: The Kaiser window is derived from Bessel functions, rather than cosines, and is defined by:

$$W(n) = \frac{I_o\left[\pi\alpha\,sqrt\left(1-\left(\frac{2n}{(N-1)}-1\right)^2\right)\right]}{I_o(\pi\alpha)}$$

where I_O is the zero-order Bessel function of the first kind and α is a constant that determines the overall shape of the window. The Kaiser window is most often used for high dynamic range signals where it offers a good trade-off between amplitude accuracy and peak width to resolve small signals close in frequency to large ones (much like Blackman-Harris). **Figure 6** shows an FT of a sine wave using a Kaiser window.

Figure 7 shows a comparison of various window functions and their effect on a single spectral line. Note that all the window functions are defined as being equal to zero outside of the frame. Most of the software packages I mentioned have selectable window functions (the Virtins enables you to create your own and add them to its already-huge list). It's worth experimenting with them using known input signals to get an intuitive feel for their characteristics.

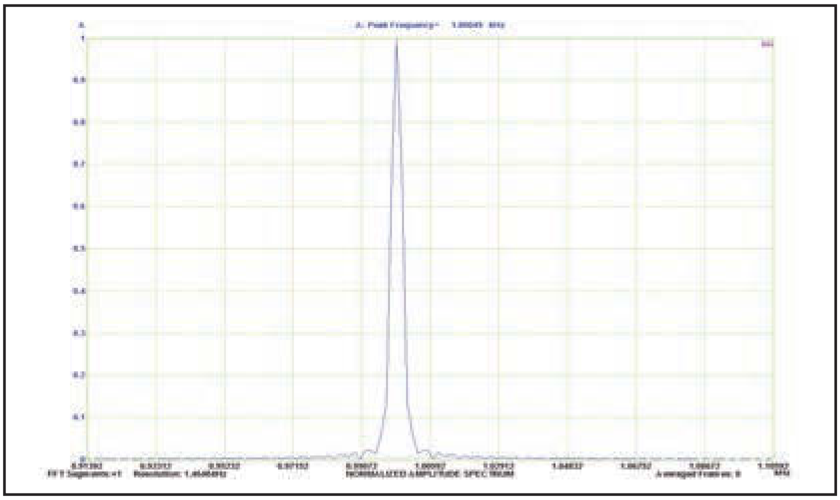

Figure 6: To compare, we have a Fourier Transform (FT) spectrum of a sine wave with a Kaiser window.

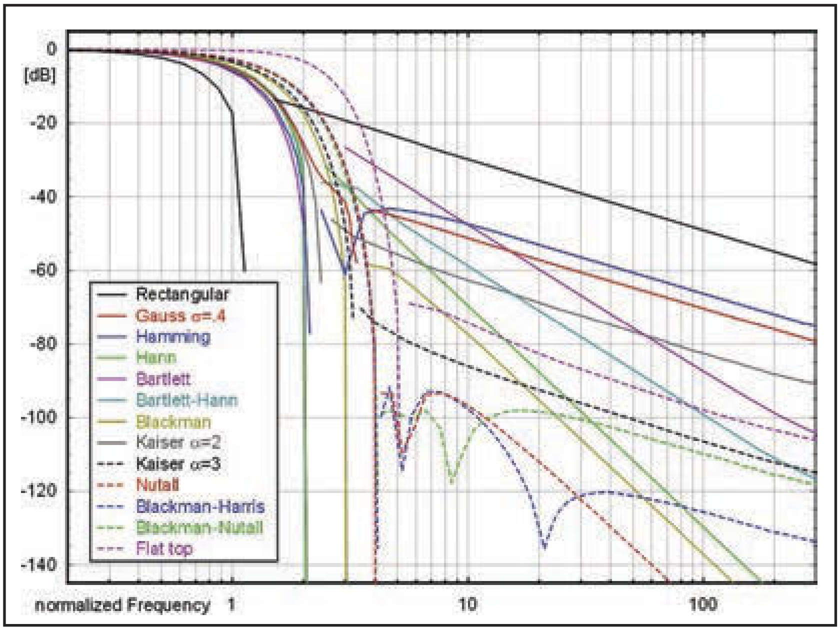

Figure 7: This is an overlay comparison of sine wave spectra with most common types of windows. (Image courtesy of Marcel Mueller, used under Creative Commons license)

Averaging

A way to increase the signal-to-noise ratio (SNR) in an FT measurement is to take the average of multiple frames. The signal increases by 6 dB with each average because it's correlated between frames, but the noise increases by 3 dB because it is uncorrelated (that's why it's noise!). Another way to say this is that the correlated signal adds as voltage but the uncorrelated noise adds as power. So for each doubling in the number of frames in the average, the signal-to-noise improves by 3 dB.

In general, one of the DUT's attributes is its noise, and we want to measure it rather than average it out, but we do want to reduce the fluctuations between frames (i.e., the noise intrinsic to the measurement). In measurement parlance, the distinction between the fluctuations in the measurement and the DUT's noise is referred to as detector noise and source noise, respectively. The way detector noise is handled in the calculations, rather than taking the vector sum of the frames' data, is to average the frames via a root-mean-square sum. That averages out the measurement deviations from the DUT's noise floor (but not the DUT's noise) by treating any deviation as positive and uncorrelated with the deviations in other frames.

In summary, increasing the number of averaged frames reduces the random fluctuation in the measurement compared to non-random signal by $\sqrt{N}$, where N is the number of averaged frames. Measurement time is increased in direct proportion to N, of course.

Wrap Up

In the next installment, I will run some example measurements that bring together the various ideas discussed in this series, and show how they are interpreted. You'll get a sense of how to use these remarkably powerful and accessible tools.

Resource

F. J. Harris, "On the Use of Windows for Harmonic Analysis with Discrete Fourier Transform," Proceedings of The Institute of Electrical and Electronics Engineers (IEEE), January 1978.

Binaural Sound from Two Loudspeakers (Part 1)

Crosstalk Cancellation and Spectral Compensation

Continuing this series on realistic sound reproduction, this month we discuss binaural recording from two loudspakers. To begin, we explain two new concepts: crosstalk cancellation and spectral compensation.

By
Ron Tipton
(United States)

To understand crosstalk cancellation, we need to re-examine the way classical binaural is recorded with a Head Related Transfer Function (HRTF) head. Keep in mind that we locate a sound source in space by comparing the arrival times and loudness as heard by our two ears. An HRTF head can record the overall sound in two channels that retains the spatial information. But when we replay this, it's essential that the left channel sound goes only to our left ear and the right channel only goes to our right ear. To do so, we must use headphones.

For the playback sound to be three-dimensional from a pair of loudspeakers, the "crosstalk" must be canceled (as much as possible). That is, the left channel sound must be blocked (or canceled) from the right ear and similarly the right sound from the left ear (see **Photo 1**).

Figure 1 illustrates this in a simplified manner. The block labeled "Filter" does the canceling and enables the listener to accurately pinpoint the sound's arrival direction. Note that the location can be well outside the loudspeakers physical placement width.

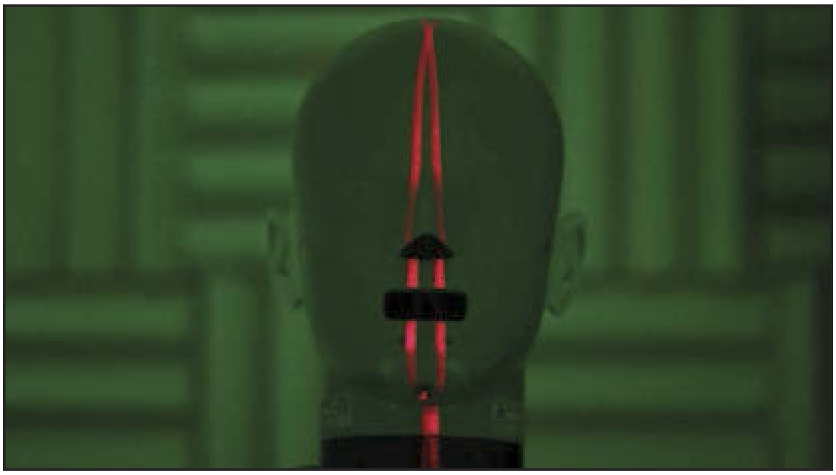

Photo 1: This Head Related Transfer Function (HRTF) head demonstrates how the left-side sound is heard by only the left ear and the right-side sound is heard by only the right ear.

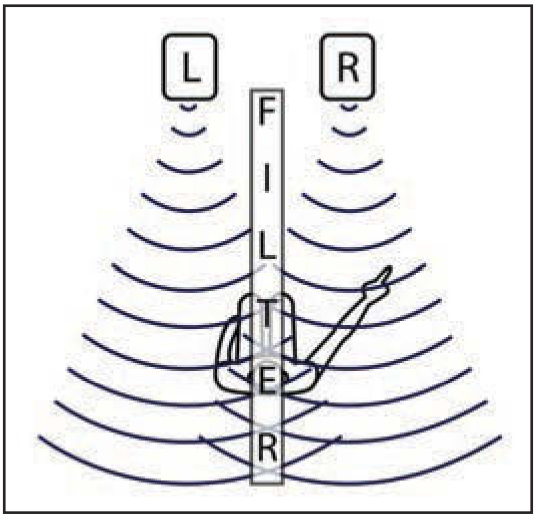

Figure 1: This simplified diagram illustrates the processing that makes binaural listening possible from a pair of loudspeakers.

Crosstalk Cancellation

Conceptually, crosstalk cancellation is straightforward, but practical problems abound. Various people have worked on it for several years. A concise overview of the effort between 1961 and 1996 is included in William Gardner's doctorial dissertation (see Resources). The cancellation is done with filters constructed with digital signal processors (DSPs) and the problems include:

- A "ragged" frequency response that must be corrected, as much as possible, with other digital filters—the spectral compensation.
- A relatively small "sweet-spot" that depends on loudspeaker placement and the music's frequency content. For example, closely angled loudspeakers tend to make the sweet-spot larger but don't provide good cancellation at low frequencies. Conversely, a wider loudspeaker separation has better low-frequency performance but narrows the sweet spot.
- Filters with so much gain or attenuation that the overall output dynamic range is limited.
- Designing a filter with the necessary specifications that can't be built, even digitally, in the real world.

Spectral Compensation

Perhaps the most comprehensive solution is described in Dr. Edgar Choueiri's paper: "Optimal Crosstalk Cancellation for Binaural Audio with Two Loudspeakers," (see Supplementary Material). This 24-page treatise is highly mathematical so I will summarize the key point.

Most of the previous crosstalk cancellation filters operated on the whole audio frequency range and introduced an uneven response. That is, the amplitude at some frequencies was increased enough to adversely affect the reproduction. Choueiri filters multiple adjacent frequency bands and combines their outputs for a smoother response but it's still not perfect. This brings in the second new concept: spectral compensation.

Basically, filters are added to filters to smooth the frequency response. It's not too difficult to see that a "box" designed to accomplish this can get rather complex. Choueiri graphs the cancellation vs. frequency in Figure 7 of his paper. Some reviewers have been critical of the performance, but I think it's impressive—40 dB of cancellation in the important 1-to-12-kHz band. But this quality comes at a price, as we'll see in the section on Bacch-SP. Less expensive solutions come at a lower cost but even 10 to 15 dB of cancellation can sound amazing.

Speaker placement is still critical, as is listening in the sweet spot, which may be rather small even with the recommended close loudspeaker spacing. Choueiri gives some examples on page 18 of his paper.

As an aside, I'll mention that others working in this field (e.g., Takashi Takeuchi and Philip A. Nelson) have created crosstalk filters that avoid, or at least minimize, the spectral coloration problem but take at least six loudspeakers for playback – which defeats the two-speaker goal.

Photo 2: The BACCH-SP processor's front panel hides the filter selection buttons and the controls to program a custom filter matching the listener's head and room description.

BACCH-SP

The "SP" in BACCH-SP stands for Spectral Purifier and is Choueiri's name for his processor box, shown in **Photo 2**, which provides crosstalk cancellation and spectral compensation. Photos and specifications are online. It apparently costs $55,000 and is available

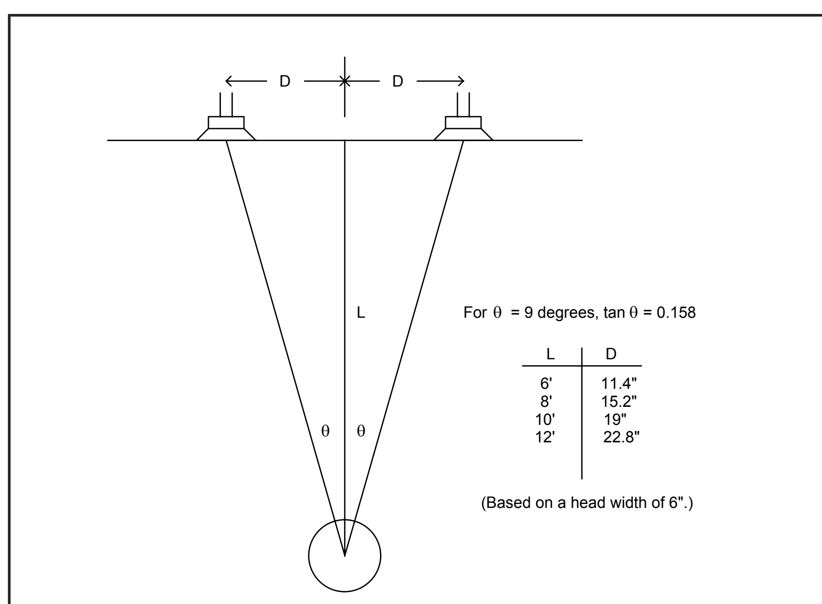

Figure 2: This figure shows the optimal sweet spot listening distance, L, for different speaker separations, D, based on a theta angle of 9°. This angle is the result of a practical assumption of a maximum spectral "coloration" of 7 dB and a crosstalk correction cut-off frequency of 6 kHz. (Image courtesy of Dr. Edgar Choueiri, "Optimal Crosstalk Cancellation")

Photo 3: This is a David Chesky recording session being held at the Hirsch Center, Greenpoint, in Brooklyn, NY. The rear of the Head Related Transfer Function (HRTF) head is visible at center left. A quote from the Chesky website says: "The 130-year-old building is run down, but the High Victorian Gothic cathedral's plastered walls and ornate wood trimmings yielded ravishing acoustics."

from a few high-end audio retailers. It has an analog stereo input, A/D converters, and digital processing and D/A converters for an analog stereo output. For best listening, you need to calibrate it. This includes setting up your loudspeakers in exactly the right position. **Figure 2** is based on the solution for a larger sweet spot, that is, for a speaker span of 18°. The table in **Figure 2** shows a rather close speaker spacing for normal listening distances, certainly closer than is usual for ordinary stereo which is usually about 60°.

There are several buttons, hidden by the front cover, that can be used to select predefined filters and enable the user to program a custom filter that matches his head and room environment. It also features a motion sensor that tracks the listener and keeps his (or her) head in the sweet spot, which implies this volume is not too large.

Head tracking is not a new idea. The Gardner dissertation I mentioned earlier primarily focuses on its use for 3-D audio. It apparently works well but is complex and expensive because the processor must continually update, in real time, the angles and the distance between the loudspeakers and the listener's ears. It would appear the Bacch-SP might not be too useful for a home theater because most seats would be outside the sweet spot—and what would it track?

I have not had the opportunity of listening to a demonstration of this equipment although I expect it would be impressive. It is also priced outside my budget so I looked for, and found, some less costly alternatives. But first, we'll look at a technique that eliminates the need for your having to have crosstalk cancellation equipment.

David Chesky

Dr. David Chesky's *Sensational, Fantastic, and Simply Amazing Binaural Sound Show* CD is available from several sources including an HDTracks.com download. It was recorded at 192 ksps, 24-bits using a Krall HRTF head and BACCH-SP processing. This is interesting because the recording retains the processing and can be played in binaural on an ordinary stereo setup.

The recording was made with a generic (universal) spectral compensation filter so it's not optimized for any particular environment (see **Photo 3**). It also retains the fairly small sweet spot, which I have verified by listening to most of the 26 tracks. But you can more-or-less maximize the sweet spot volume by using the guidelines in the previous section for speaker placement.

Regardless of its short comings, this is a viable method to market binaural recordings without the expense of a BACCH-SP or other processor. The only downside is the current lack of music with this processing.

About the Author

Ron Tipton has degrees in electrical engineering from New Mexico State University and is retired from an engineering position at the White Sands Missile Range. In 1957, he started Testronic Development Laboratory (now TDL Technology) to develop audio electronics. He is still the TDL president and principal designer.

Project Files

To download additional material and files, visit http://audioxpress.com/page/audioXpress-Supplementary-Material.html.

Resources

D. Chesky, *Sensational, Fantastic, and Simply Amazing Binaural Sound Show* CD, www.chesky.com.

E. Choueiri, "BACCH 3D Sound," www.princeton.edu/3D3A/PureStereo/Pure_Stereo.html.

W. Gardner, "3-D Audio Using Loudspeakers," doctorial dissertation, Massachusetts Institute of Technology, September 1997.

HD Tracks, www.hdtracks.com.

T. Takeuchi and P. A. Nelson, "Optimal Source Distribution for Binaural Synthesis over Loudspeakers," *Journal of the Audio Engineering Society* (*JAES*), 2002.

Source

BACCH-SP
MASI Audio Services, Inc. |
www.masiaudio.com/bacch/thesolutions/bacch-sp

Precision Performance

The Pursuit of a Perfect Measurement

Researchers at Brüel & Kjær are using simulation to achieve new levels of precision and accuracy for their industrial and measurement-grade microphones and transducers. This article details how Brüel & Kjær used a combination of COMSOL's Multiphysics base package, together with the Acoustics Module, the Structural Mechanics Module, and the AC/DC Module in their R&D process.

Photo 1: This is the 4134 microphone including the protective grid mounted above the diaphragm.

By
Valerio Marra
(United States)

There will never be a perfect measurement taken or an infallible instrument created. While we may implicitly trust the measurements we take, no measurement will ever be flawless, as our instruments do not define what they measure. Instead, they react to surrounding phenomena and interpret this data against an imperfect representation of an absolute standard.

Therefore, all instruments have a degree of acceptable error—an allowable amount that measurements can differ without negating their usability. The challenge is to design instruments with an error range that is known and consistent, even over extended periods of time.

Brüel & Kjær A/S has been a leader in the field of sound and vibration measurement and analysis for more than 40 years. Its customers include Airbus, Boeing, Bosch, Caterpillar, Ferrari, Ford, Honeywell, Lockheed Martin, NASA, Rolls-Royce, Toyota, and Volvo, just to name a few.

Because industry sound and vibration challenges are diverse—from traffic and airport noise to car engine vibration, wind turbine noise, and production quality control—Brüel & Kjær must design microphones and accelerometers that meet a variety of measurement standards. To meet these requirements, the company's R&D process includes simulation as a way to verify the precision and accuracy of its devices and test new and innovative designs.

Designing and Manufacturing Accurate Microphones

Brüel & Kjær develops and produces condenser microphones covering frequencies from infrasound to ultrasound, and levels from below the hearing threshold to the highest sound pressure in normal atmospheric conditions. The range includes working standard and laboratory standard microphones, as well as dedicated microphones for special applications. Consistency and reliability is a key parameter in the development of all Brüel & Kjær microphones.

"We use simulation to develop condenser microphones and to ensure that they meet relevant International Electrical Commission (ICE) and International Organization for Standardization (ISO) standards," says Erling Olsen, development engineer in Brüel & Kjær's Microphone Research and Development department. "Simulation is used as part of our R&D process, together with other tools, all so that we know that our microphones will perform reliably under a wide range of conditions. For example, we know precisely the influence of

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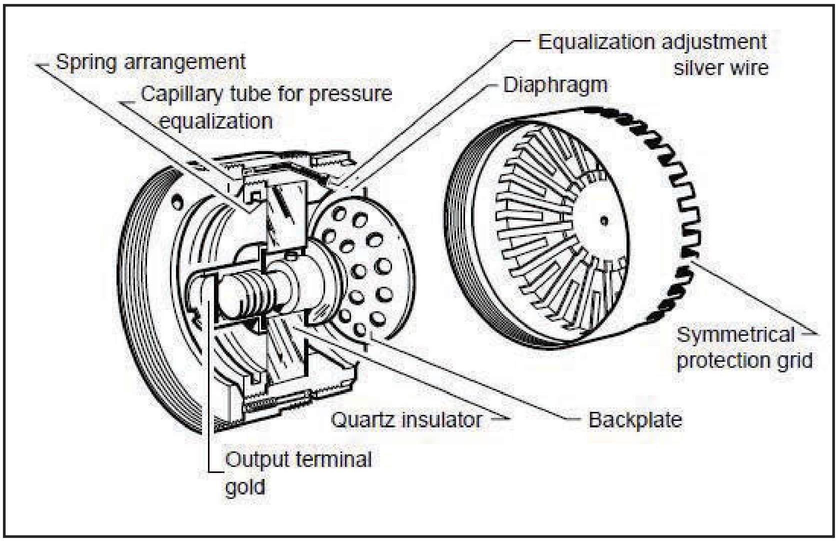

Figure 1: Here is a sectional view of a typical microphone cartridge showing its main components.

static pressure, temperature and humidity, and the effect of other factors for all of our microphones—parameters that would have been very difficult to measure were it not for our use of simulation."

The Brüel & Kjær Type 4134 condenser microphone shown in **Photo 1** is an old microphone that has been subject to many theoretical and practical investigations over time. Therefore, the 4134 microphone has been used as a prototype for developing multiphysics models of Brüel & Kjær condenser microphones (see **Figure 1**). To analyze the microphone's performance, Olsen's simulations include the diaphragm's movement, the electromechanical interactions of the membrane deformations with electrical signal generation, the resonance frequency, and the viscous and the thermal acoustic losses occurring in the microphone's internal cavities.

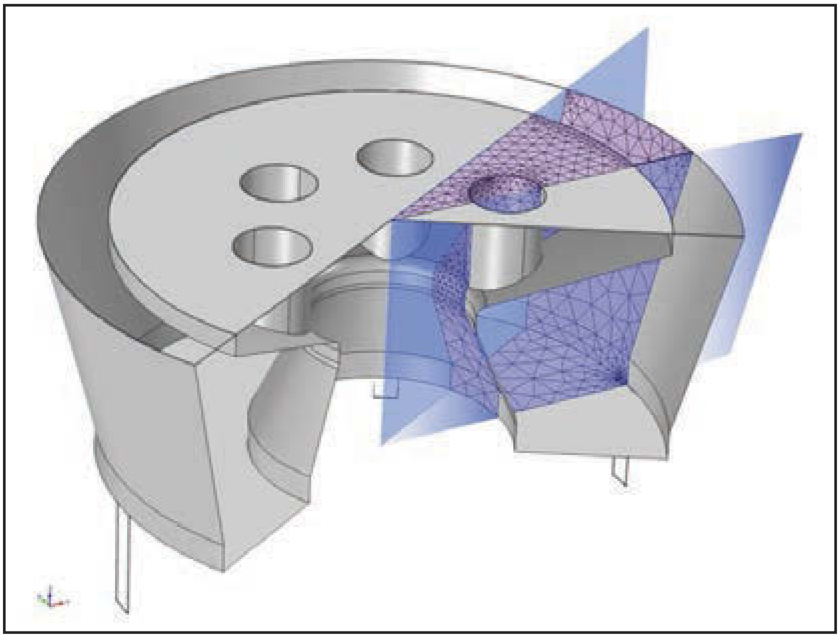

Figure 2: The 4134 condenser microphone's geometry plot shows the mesh used in the reduced sector geometry, representing 1/12 of the total geometry.

Microphone Modeling

When sound enters a microphone, sound pressure waves induce deformations in the diaphragm, which are measured as electrical signals. These electrical signals are then converted into sound decibels. "Modeling a microphone involves solving a moving mesh and tightly coupled mechanical, electrical, and acoustic problems—something that could not be done without multiphysics," says Olsen. "The models need to be very detailed because in most cases, large aspect ratios (due to the shape of the microphone cartridges) and small dimensions cause thermal and viscous losses to play an important role in the microphone's performance."

The model can also be used to predict the interactions that occur between the back plate and diaphragm. Among other things, this influences the directional characteristics of the microphone. "We used the simulation to analyze the bending pattern of the diaphragm," says Olsen.

For simulations (e.g., thermal stress and resonance frequency), model symmetry was used to reduce calculation time (see **Figure 2**). The reduced model was also used to analyze the sound pressure level (SPL) in the microphone for sounds that are at a normal incidence to the microphone diaphragm (see **Figure 3**). However, when sound enters the microphone with non-normal incidence, the membrane is subjected to a nonsymmetrical boundary condition. This requires a simulation that considers the entire geometry to accurately capture the bending of the membrane (see **Figure 4**).

Simulation was also used to determine the air vent's influence in the microphone for measuring low-frequency sounds. "We modeled the microphone with the vent either exposed to the external sound field, outside the field (unexposed), or without a vent," says Olsen. "While the latter would not be done in practice, it allowed us to determine the interaction between the vent configuration and the input resistance results for different low-frequency behaviors. This is one of the most important things about simulation: We can make changes to the parameters of a model that move away from already manufactured devices, allowing us to test

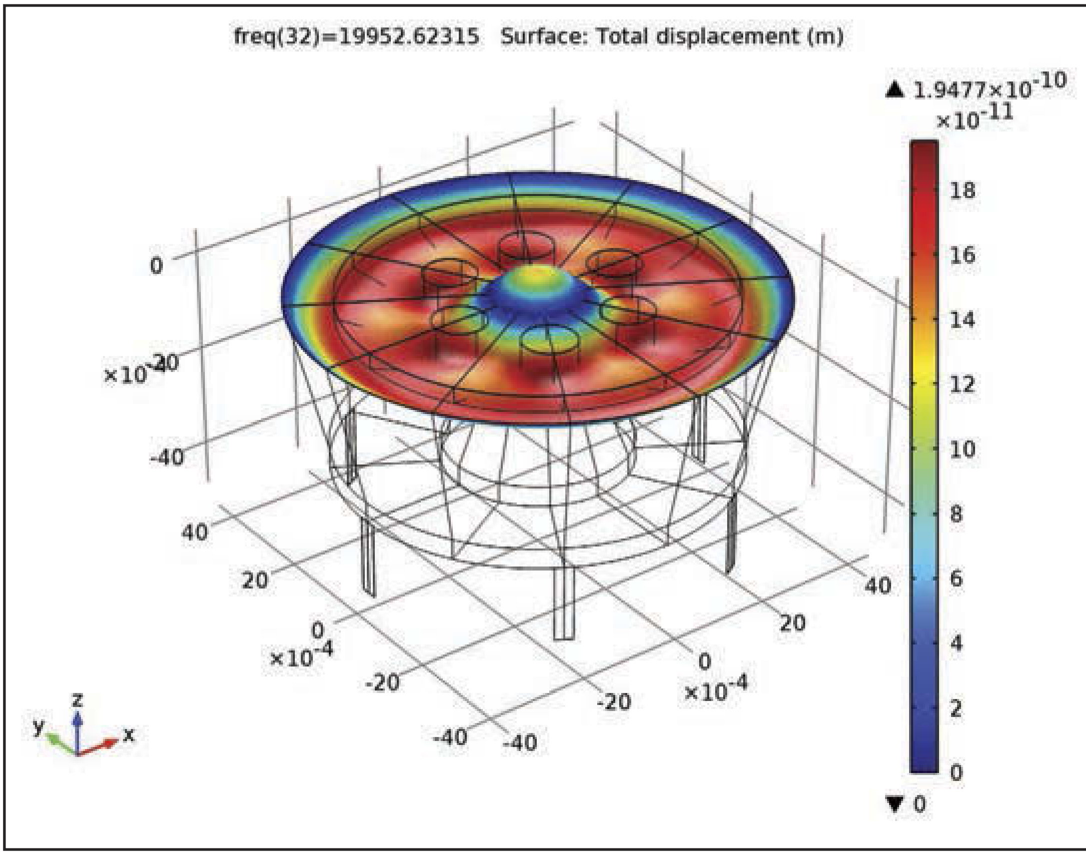

Figure 3: Representation of the sound pressure level below the diaphragm for normal incidence is calculated using the sector geometry. The membrane deformation is evaluated at f = 20 kHz.

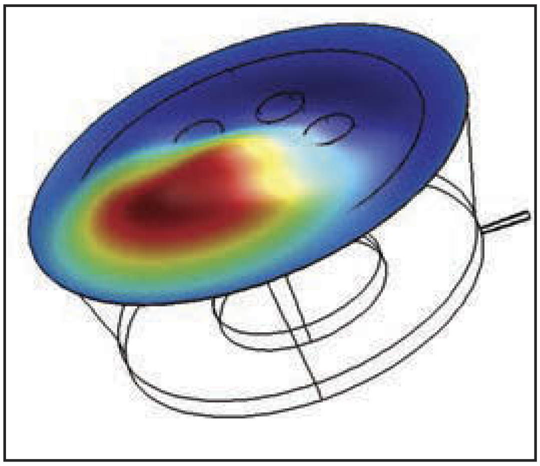

Figure 4: Simulation results show the membrane deformation calculated for non-normal incidence at 25 kHz. Since the deformation is asymmetrical, this is calculated using the full 3-D model.

other designs and explore the limits of a device (see **Figure 5**)."

With simulation as part of the R&D process, Olsen and his colleagues are not only able to design and test some of Brüel & Kjær's core products, but they can also create devices based on a specific customer's requirements.

"With simulation, we can pin-point approaches for making specific improvements based on a customer's needs. Although microphone acoustics are very hard to measure through testing alone, after validating our simulations against a physical model for a certain configuration, we are able to use the simulation to analyze other configurations and environments on a case-by-case basis," Olsen states.

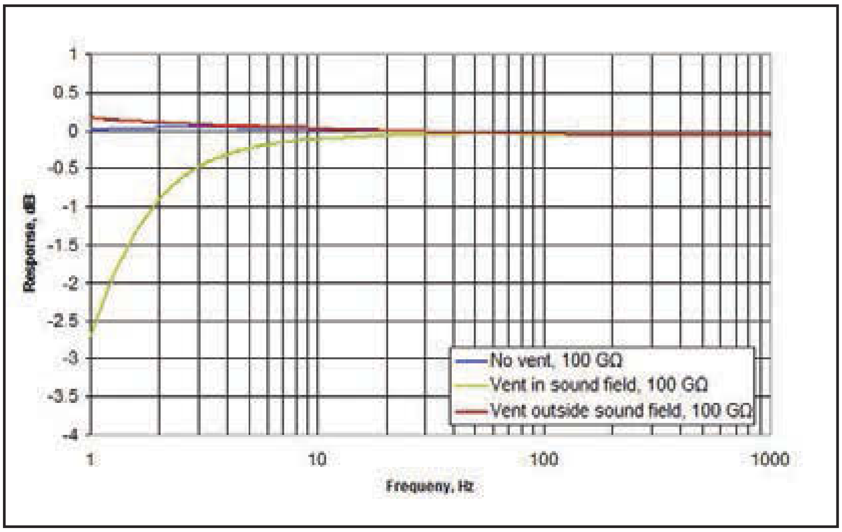

Figure 5: In the no-vent configuration, the sensitivity increase is due to the fact that the sound field becomes purely isothermal inside the microphone at low frequencies. In the vent outside the sound field configuration, the curve initially follows the no-vent curve, but sensitivity increases as the vent becomes a pressure release on the back of the diaphragm.

Vibration Transducer Modeling

Søren Andresen, a development engineer with Brüel & Kjær, also uses simulation to design and test vibration transducer designs.

"One of the complications with designing transducers for vibration analysis is the harsh environments that these devices need to be able to withstand," says Andresen. "Our goal was to design a device that has so much built-in resistance that it can withstand extremely harsh environments."

Most mechanical systems tend to have their resonance frequencies confined to within a relatively narrow range, typically between 10 and 1,000 Hz. One of the most important aspects of transducer design is that the device does not resonate at the same frequency as the vibrations to be measured, as this would interfere with the measured results. **Figure 6** shows the mechanical displacement of a suspended vibration transducer, as well as a plot of the resonance frequency for the device.

"We want the transducer to have a flat response and no resonance frequency for the desired vibration range being measured," says Andresen. "We used COMSOL to experiment with different designs in order to determine the combination of materials and geometry that produces a flat profile (no resonance) for a certain design. This is the region in which the transducer will be used."

When designing the transducer, a low-pass filter or a mechanical filter can be used to cut away the undesired signal caused by the transducer resonance, if any. These filters consist of a medium, typically rubber, bonded between two mounting discs, which is then fixed between the transducer and the mounting surface.

"As a rule of thumb, we set the upper frequency limit to one-third of the transducer's resonance frequency, so that we know that vibration components measured at the upper frequency limit will be in error by no more than 10 to 12%," says Andresen.

As Accurate and Precise as Possible

While it may not be possible to design a perfect transducer or take an infallible measurement, simulation brings research and design teams closer than ever before by enabling them to quickly

About the Author

Valerio Marra holds a BSc and MSc in Nuclear Engineering and received a PhD in Energetic Systems and Fluid Machines Engineering from the University of Bologna, Italy. He joined COMSOL in 2008 and has worked in applications, sales, and marketing. He is currently the Technical Marketing Manager and works out of the Burlington, MA, office.

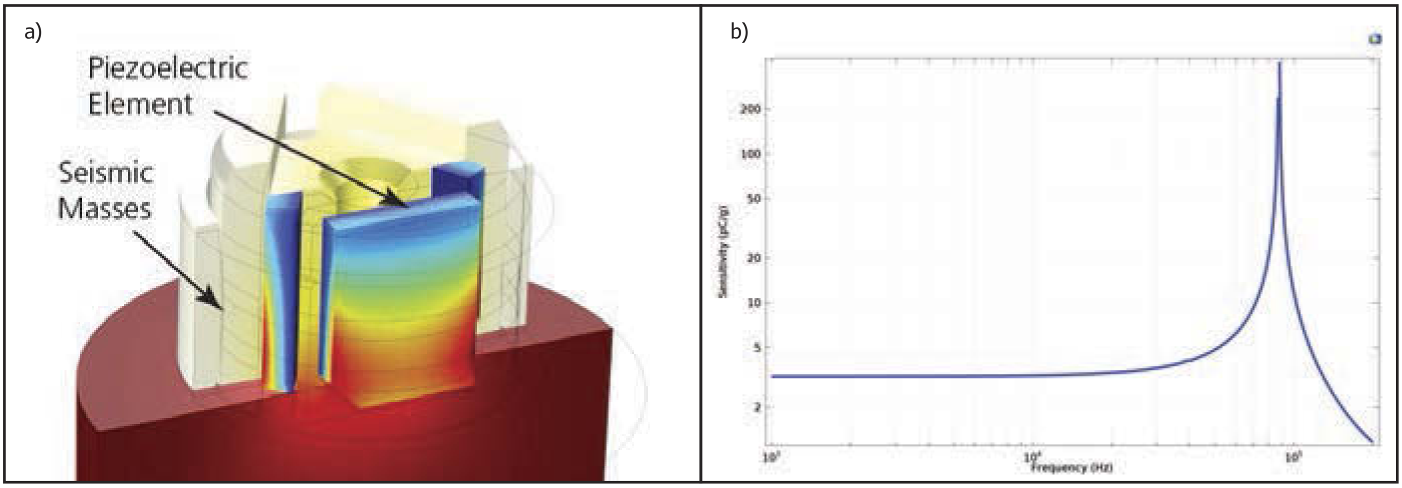

Figure 6: These are the simulation results of a suspended piezoelectric vibration transducer. a: This is the mechanical deformation and electrical field in the piezoelectric sensing element and seismic masses. b: The frequency-response plot shows the transducer's first resonance at around 90 kHz. This device should only be used to measure objects at frequencies well below 90 kHz.

and efficiently test new design solutions for many different operating scenarios.

"In order to stay ahead of the competition, we need knowledge that is unique," says Andresen. "Simulation provides us with this, as we can make adjustments and take virtual measurements that we couldn't otherwise determine experimentally, allowing us to test out and optimize innovative new designs."

Resource

COMSOL, Multiphysics Software Product Suite, www.comsol.com/products.

IsoAcoustics Aperta Speaker Stands

Effective Floating Isolation

Photo 1: The Aperta is IsoAcoustics' latest offering in its line of speaker isolation stands. The top and the bottom platforms are made of cast aluminum. The stands are available in black or natural aluminum finish. (Photo courtesy of IsoAcoustics)

IsoAcoustics recently introduced Aperta, a new aesthetically pleasing and cost-effective line of aluminum acoustic isolation stands built with a unique, patented isolation technology that enables speakers to float in free space, resulting in clear uncolored sound.

By
Gary Galo
(United States)

Aperta Speaker Stand
IsoAcoustics, Inc.
4981 Highway 7 East
Unit 12A, Suite 100
Markham, ON
Canada L3R 1N1
905-294-4672
info@isoacoustics.com
www.isoacoustics.com
List Price: $240/per pair

IsoAcoustics is a Canadian firm specializing in loudspeaker isolation stands for both audiophiles and audio professionals. Company founder and President Dave Morrison worked for the Canadian Broadcasting Corp. (CBC) for nearly 20 years. While there, he was closely involved in the design and the construction of radio and television studios. Perhaps his foremost technical challenge was as a member of the design team for the world's largest multi-media center. Located in Toronto, with a size of more than 1.72 million ft^2, the facility includes recording studios for drama, musical ensembles, special effects, plus radio and television production. The design of the IsoAcoustics speaker stands was an outgrowth of the experience he gained while working for the CBC.

Patented Design

Morrison's "floating design" concept is described in detail in US Patent No. 7,640,868, co-authored by Canadian inventor and designer Robert G. Dickie (available from Google Patents). In email correspondence, Morrison noted that his design concepts work "in a different direction than the common paradigms and status quo."

As we learned from Isaac Newton, for every action, there's an equal but opposite reaction. And when a loudspeaker cone moves forward, the enclosure wants to move back and vice-versa. In an ideal world, the enclosure would not move or vibrate at all. But that could only be possible with an enclosure of infinite mass, which is impossible in this universe. A loudspeaker enclosure must normally rest on something, whether a floor or a desktop. Unless the supporting surface also has infinite mass, it will vibrate in sympathy with the enclosure. Sympathetic vibrations from the supporting surface will color the sound produced by the loudspeaker. Vibrations reflected back to the loudspeaker enclosure will only exacerbate the problem.

The conventional solution to this problem is to isolate the loudspeaker from the supporting surface with metal spikes or cones, which are made with a variety of materials including steel, brass and aluminum. But, spikes and cones are only partially effective. As Morrison explains, spikes create secondary reflections similar to throwing a rock in


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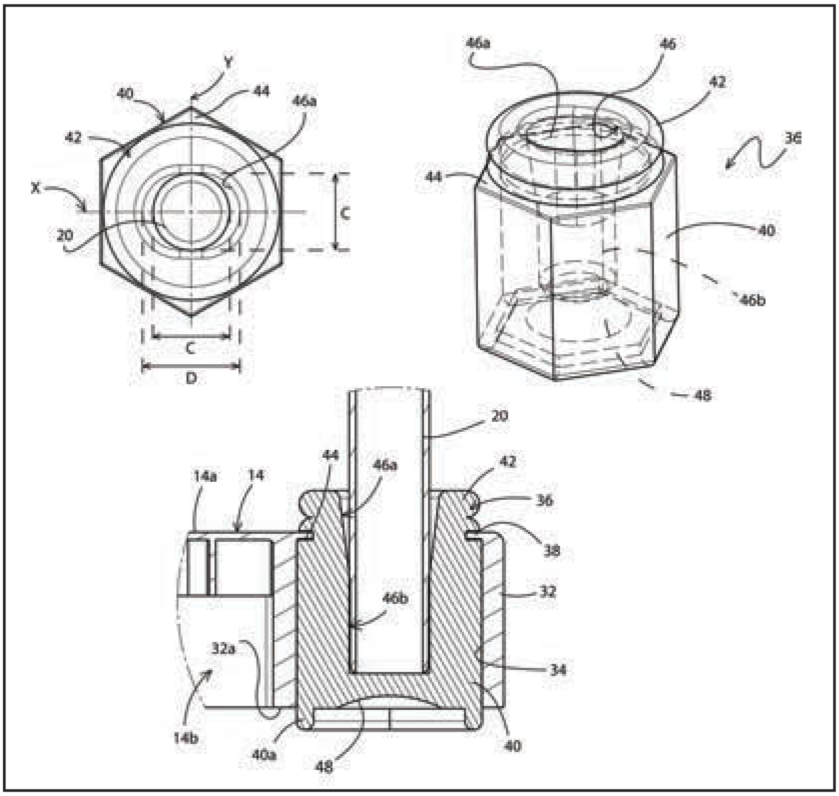


Figure 1: An illustration from the David Morrison and Robert G. Dickie patent shows the rubber bushings and the connecting rods. The upper left and right drawings show the oblong shape of the bushings, which enable front-to-back movement of the connecting rods, while minimizing side-to-side motion. The bottom drawing shows a cutaway view of a connecting rod inserted into a rubber bushing. (Image courtesy of IsoAcoustics)

Photo 2: At the Munich High End Show 2015, Dave Morrison demonstrated IsoAcoustics' unique speaker stand approach, with its Modular series supporting a 3.5-way Focal Maestro Utopia loudspeaker weighing 256 lb (116 kg). (*audioXpress* photo)

a swimming pool. The ripples hit the hard edges and converge again as we are about to throw in the next boulder. The only way to eliminate those reflections would be to make the pool infinitely large.

The IsoAcoustics stand consists of two platforms—one on which the loudspeaker sits and a second which rests on the supporting surface. Each platform has four injection-molded, rubber bushings in each of four corners, which function as isolators (see **Photo 1**). The two platforms are connected by four aluminum support rods inserted into the bushings. The openings in the bushings are oblong, which "polarizes" them so they enable front-to-back movement of the top, speaker support platform, while minimizing side-to-side movement (see **Figure 1**). They are also tapered vertically, which enables the support rods to be firmly seated.

The system is designed to allow front-to-back movement of the speaker and the top platform, while preventing that motion from being transmitted to the lower platform. If the lower platform remains stationary, the supporting surface is isolated from speaker vibration.

Morrison notes that even when used on a high-mass supporting surface (e.g., a solid granite pedestal), "the speaker is allowed to breathe and we mitigate the bass smear. A buildup of smear and secondary reflection leads to a collapse of the sound-stage, as any replication of these elements in the left and right channels is perceived to be in the center."

Morrison explains that "The isolators are tuned by adjusting the shape, thickness, and durometer of the isolators to manage the response and on-axis alignment." The two aluminum platforms are not identical. According to Morrison, "We have to keep the mass of the upper frame down so it reacts with the speaker, and so the articulation happens in the body of the isolators. The upper and lower frames act as a parallelogram as the isolators maintain alignment. The shape of the frame provides the torsional strength so the isolators can provide a consistent response."

Each platform also has four concave rubber flanges that contact the bottom of the speaker and the supporting surface. The concave flanges act as suction cups, improving the integrity of the mechanical connection between the upper platform and the loudspeaker, and between the lower platform and the supporting surface. The concave shape also contributes to the compliance of the isolators. The flanges and isolators are actually one piece, inserted through holes in the corners of each platform. There's also a secondary benefit of using these stands—by raising the speaker at least 3" off the supporting surface, airborne reflections off that surface are reduced.

IsoAcoustics manufactures five different stands in its ISO-8LR series, ranging in sizes to accommodate the smallest mini monitors to large studio monitors and subwoofers. At 17″ × 9″ (43.2 cm × 22.9 cm) the ISO-L8R$_{430}$ is the largest in the series, and is big enough to handle musical instrument amplifiers. Most models come with two sets of vertical supports, allowing heights of either 3″ (7.62 cm) or 8″ (20.32 cm). They also make a Modular series that can be custom-sized for a variety of speakers from small to very large. The Modular stands can also be used in performance and recording situations, including isolation of drum-set platforms.

Morrison mentioned that the Modular stands work well with Focal's 573-lb (260 kg) Grande Utopia loudspeakers (see **Photo 2**). IsoAcoustics products are beginning to be embraced by the loudspeaker industry. The respected Danish manufacturer Dynaudio bundles its stands with the BM mkIII series of studio monitors, and IsoAcoustics is currently working to create similar arrangements with several other loudspeaker manufacturers. On its website Dynaudio states: "When we started looking for the

Photo 3: Threaded inserts in two of the aluminum support rods enable a tilt adjustment of up to 6.5°. By rotating the stand 180°, the speaker can be tilted backward or forward. A red mark on the threaded inserts appears when the maximum tilt angle has been exceeded. (Photo courtesy of IsoAcoustics)

Photo 4: A loudspeaker is shown mounted on an IsoAcoustics isolator, which was tested with a laser vibrometer in an anechoic chamber at the Canadian National Research Council in Ottawa, Canada. (Photo courtesy of IsoAcoustics)

best solution for isolating the BM mkIII monitors from the surface, we found that IsoAcoustics provide the best and most flexible product on the market."

Aperta Specifics

The Aperta (shown in **Photo 1**) is IsoAcoustics' newest model, and is the same overall size as the lower-cost ISO-L8R$_{155}$. The ISO-L8R$_{155}$ is manufactured from high-strength Acrylonitrile Butadiene Styrene (ABS, a thermoplastic polymer), whereas the Aperta is cast, sculpted aluminum.

The Aperta's overall dimensions are 6.1" × 7.5" (15.5 cm × 19.05 cm), for medium-sized studio monitors weighing up to 35 lb (15.9 kg). The Aperta comes with a single set of vertical supports, configured for a height of 3". Morrison notes that IsoAcoustics "continued to tune and refine the isolators for the Aperta to the extent our evaluation group unanimously selected the Aperta over the ISO-L8R$_{155}$ in final testing."

The Aperta stands cost $240/pair, and are available with a black or natural aluminum finish. They're sold by several pro audio and Internet dealers, including Amazon, and the "street" price is usually around $200/pair. A complete list of retail and Internet dealers, plus information on international sales and distribution, can be found on the IsoAcoustics web site. All IsoAcoustics products are warrantied for two years.

The Aperta stands are equipped with a tilt adjustment that enables users to dial in the optimum tilt angle for their listening positions (see **Photo 3**). Two of the aluminum support rods have threaded inserts, which enable height adjustment on one end of the stand. The inserts are calibrated in degrees, with fine gradation marks, and a red mark appears when the maximum angle of 6.5° is exceeded. By rotating the stands 180°, the loudspeaker can either be tilted backward or forward.

The owner's manual is bare-bones—a small, five-page foldout with illustrations that are largely self-explanatory. There are no verbal explanations, which avoids the need for multilingual versions. More detailed information can be found on the IsoAcoustics website.

Performance

IsoAcoustics isolators have been subject to testing at the National Research Council in Ottawa, Canada, using an anechoic chamber and laser vibrometer. As Morrison explains, "The anechoic chamber was used to verify that the stands did not change or colorize the speaker's enclosure and the response of the stands to the speaker's movement. The laser vibrometer measures the acceleration of the speaker enclosure and the response of the stands to the speaker's movement."

Photo 4 shows a loudspeaker mounted on an IsoAcoustics isolator, inside the anechoic chamber. **Photo 5** shows the test results for three different height stands. These graphics illustrate 75-Hz tests, with an amplitude of up to 314 µM/s on the front plane of the speaker enclosure. The red and green

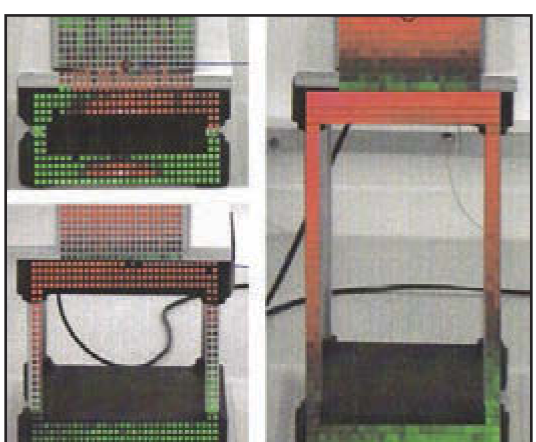


Photo 5: These are the laser vibrometer test results for IsoAcoustics speaker stands of three different heights. The change in color from red to green shows excellent vibration attenuation through the stands. (Photo courtesy of IsoAcoustics)

About the Author

Gary Galo retired in June 2014 after 38 years as an Audio Engineer at the Crane School of Music, SUNY, in Potsdam, NY. He now works as a volunteer, transferring vintage analog recordings in the Crane archive to digital format. He is the author of more than 280 articles and reviews on musical and technical subjects. He is an active member of the Association for Recorded Sound Collections (ARSC) and has been a regular presenter at ARSC conferences. Several of his conference papers have been published in the *ARSC Journal*.

colors indicate maximum and minimum vibration (i.e., attenuation through the stands). The results show excellent attenuation of vibration from the loudspeaker and the top of the stand to the bottom of the stand.

Subjectively, I found the IsoAcoustics Aperta speaker isolators to deliver what the manufacturer claims. I evaluated the Aperta speaker stands in my studio at The Crane School of Music, SUNY Potsdam, using a pair of Monitor Audio Bronze BX2 loudspeakers driven by Monarchy Audio SM-70 power amps (see **Photo 6**). The Monarchy amplifiers were operated as fully-balanced monoblocks fed by a Benchmark DAC1 USB D/A converter. The loudspeakers were bi-wired with DH Labs T-14 speaker cable. Interconnects between the D/A converter and power amps were made with DH Labs Pro Studio cable and Neutrik XLR connectors with gold-plated pins.

I've experimented with metal cones, but never found them particularly effective in desktop applications, and they tend to slide around on smooth surfaces, which is an annoyance. The Apertas stay put on the desktop and definitely improve the sound.

Photo 6: For the listening tests, I mounted a Monitor Audio Bronze BX2 loudspeaker on an IsoAcoustics Aperta speaker stand. Monarchy Audio's SM-70 power amplifier, operating as a fully-balanced monoblock, is seen on the left.

Both the bass and midrange show improved clarity and definition.

I prefer the Monitor Audio BX2 loudspeakers with the foam plugs inserted into the vents, which essentially converts the woofer loading from vented to aperiodic. Even so, the bass is still a bit boomy due to excitation of desktop resonances. The Apertas minimized this problem, resulting in a smoother, more natural low end. I also found that the sound-staging improved, taking on a more holographic quality. The placement of individual sections of an orchestra are rendered with greater precision.

One installation issue not discussed by the manufacturer concerns the positioning of speaker cables. It's important that speaker cables don't interfere with the movement of the speaker and top platform. My short runs of DH Labs T-14, which is relatively stiff, are dressed to form an inverted "U"-shaped loop between the amplifier and the speaker, and don't allow the cables to simply hang off the backs of the speakers, vertically pulling down on them. Free movement of the speaker and the top platform, relative to the bottom platform, are essential to achieve proper isolation.

The Aperta speaker isolators are fairly priced for the quality of design and construction, and the performance benefits they deliver. Anyone using medium-size monitors should consider the Aperta stands a cost-effective way to improve loudspeaker performance.

Resource

D. Morrison and R. Dickie, "Stereo Speaker Stand," US Patent No. 7,640,868, www.google.com/patents/US7640868.

Sources

DAC1 USB D/A converter
Benchmark Media Systems, Inc. | www.benchmarkmedia.com

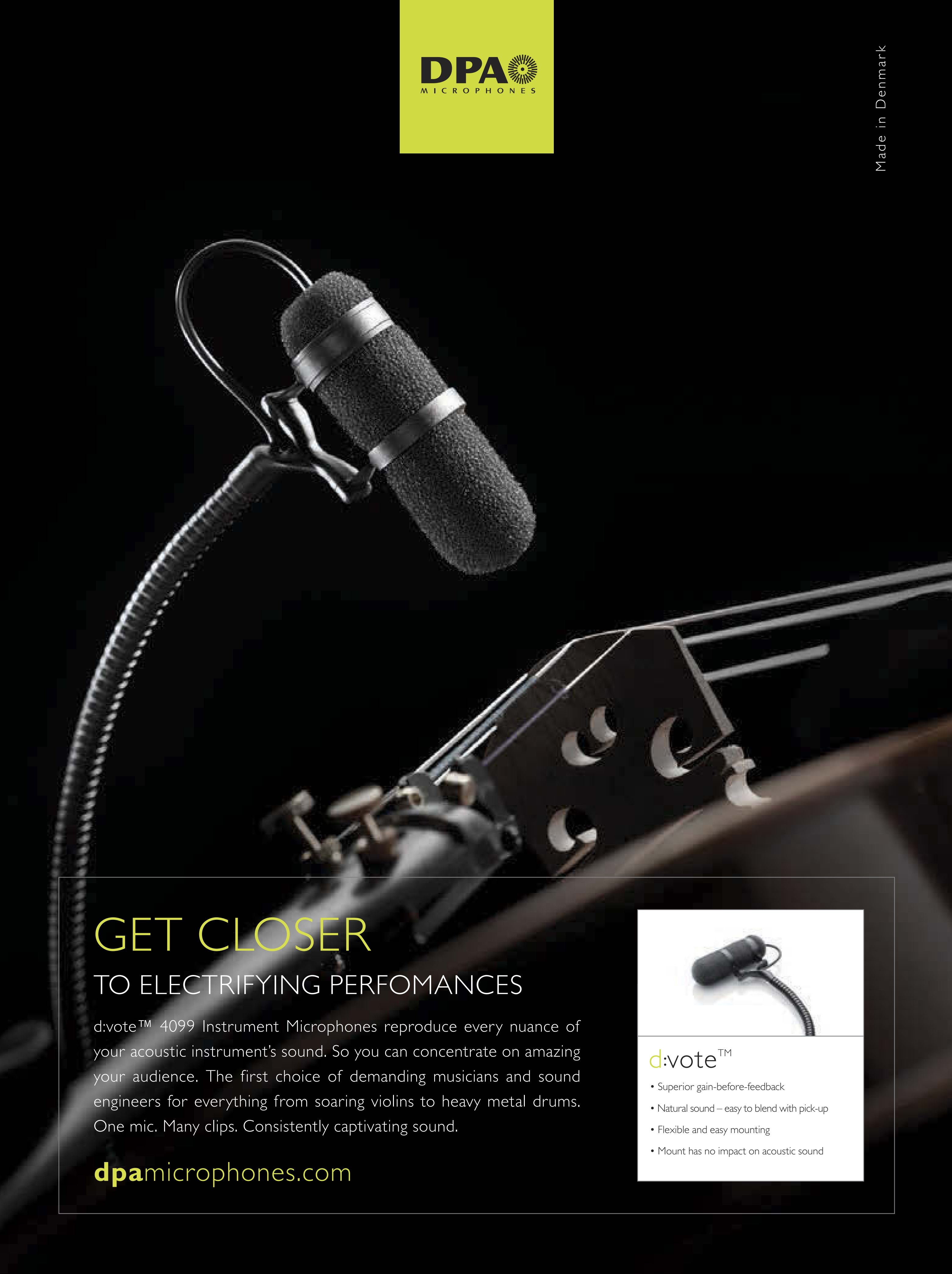
T-14 Speaker cables
DH Labs | www.silversonic.com

BM mkIII Series monitors
Dynaudio Corp. | www.dynaudioprofessional.com/en/bm-series

Monarchy Audio SM-70 power amplifiers
Monarchy Audio | www.monarchyaudio.com

Monitor Audio Bronze BX2 loudspeakers
Monitor Audio, Ltd. | www.monitoraudiousa.com

Neutrik XLR connectors
Neutrik USA, Inc. | www.neutrik.us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Audiomatica's CLIO Pocket Electro-Acoustical Portable Measurement System

Photo 1: The CLIO pocket really is small enough to fit in your pocket.

I have been using CLIO electro-acoustic measurement systems for about 20 years, going all the way back to the CLIO 4.0—a DOS-based system using the HR–2000 signal generator/audio analyzer PCI board. So I was especially excited when the folks at Audiomatica asked me to beta test their latest creation, the CLIO Pocket.

By
Joe D'Appolito
(United States)

Photo 2: The CLIO Pocket is a powerful, lightweight portable measurement system.

The CLIO Pocket is a lower cost electro-acoustic measurement system intended for the experienced amateur speaker builder, but it also represents a highly portable measurement system for the professional who is doing a lot of onsite work. And, as its name implies the CLIO Pocket audio interface literally fits in your pocket (see **Photo 1**).

The CLIO Pocket arrives nicely packaged in a padded carrying case (see **Photo 2**). The case contains the CP–01 audio interface, a USB cable, an impedance test cable, the MIC–02 microphone with a 2.7-m cable, and the installation software. Basically, it contains everything you need to start taking measurements.

The CLIO Pocket system comprises the CP–01 Audio Interface (hardware) and software designed to run on Windows XP, Vista, 7, 8, and 8.1. It will also run on Apple Mac OSX. The CP–01 audio interface measures 125 mm × 90 mm × 24 mm.

It consists of two systems—a signal generator and an audio analyzer. On the CP–01's front face you can see a blue LED, an RCA input jack to the audio analyzer, and an RCA output jack from the signal generator. Connection to the computer is via a USB 2.0 port, which powers the CP–01 and interfaces to the software.

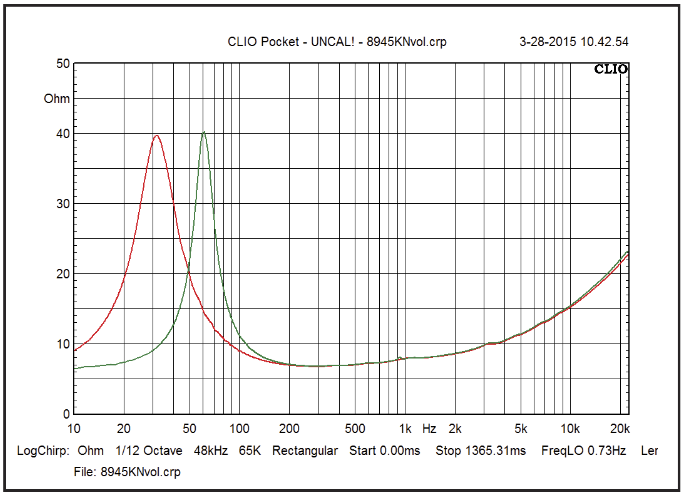


Figure 1: Impedance data for Thiele-Small parameter estimates

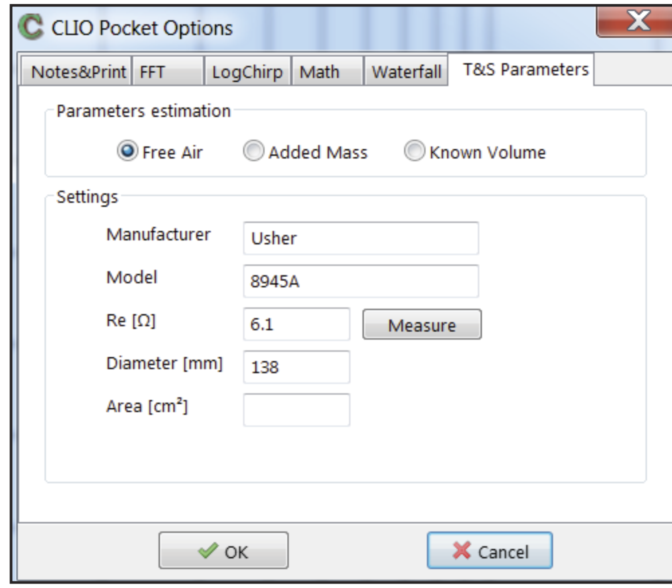


Figure 2: Driver data for Thiele-Small analysis

Installation and Calibration

Hardware installation is fairly straightforward. After the CLIO Pocket is connected to a USB port and recognized by your computer, you must manually install two additional USB drivers available on the distribution disk. Then, the software can be installed.

Before the CLIO Pocket can be used, you must go through a calibration process. Start the program and allow the CP–01 to warm up for 15 to 20 minutes. (The blue LED should be lit.) Then, an initial sine wave analysis is performed using the Fast Fourier Transform (FFT) option. If this is successful, the calibration process can proceed. Just select "calibration" from the main menu. The process is fully automatic and takes about a minute to complete. As described in the manual, several tests can then be run to validate the calibration.

CLIO Pocket Functions

At the heart of the CLIO Pocket setup and control are three windows: the options window, the generator control window, and the measurements settings window. In the options window, you can set the sample rate (48 or 96 kHz), measurement units (e.g., microphone sensitivity), graphics options, and notes on saved measurements. The 96-kHz sample rate sets a maximum analysis bandwidth of 40 kHz.

In the signal generator window, you can select sine, two sine, all wave, chirp, white noise, and pink noise signals. You can also select the new CEA tone burst used for subwoofer testing. Upon selecting

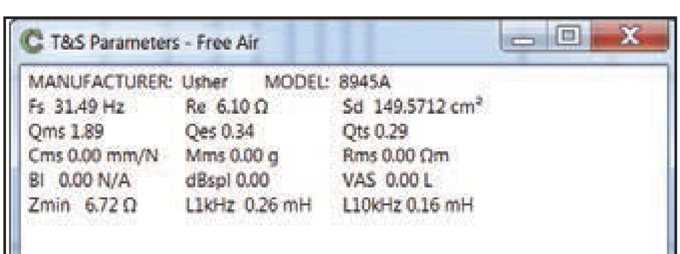


Figure 3: Thiele-Small free-air parameters

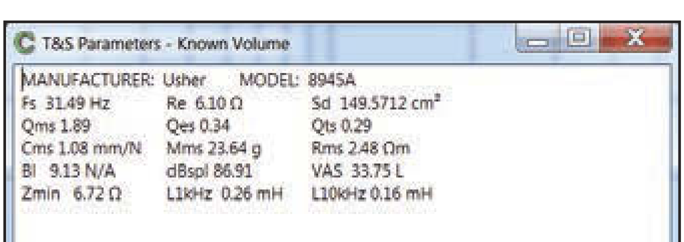


Figure 5: Final results of Thiele-Small parameter estimates

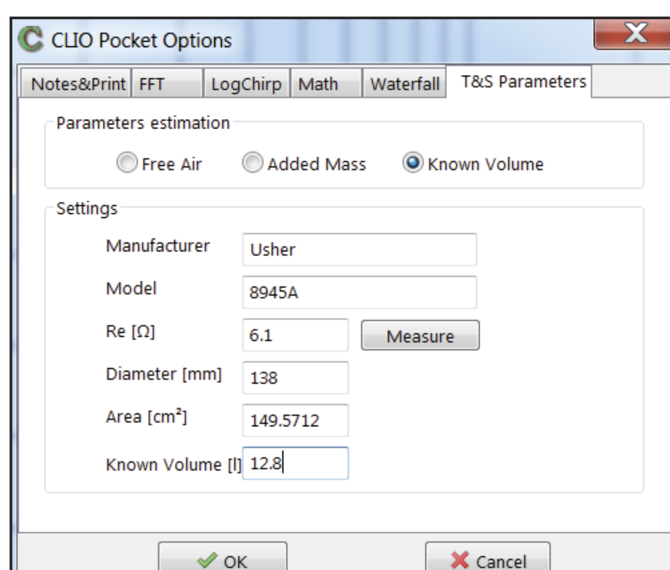


Figure 4: Data input for known-volume Thiele-Small analysis

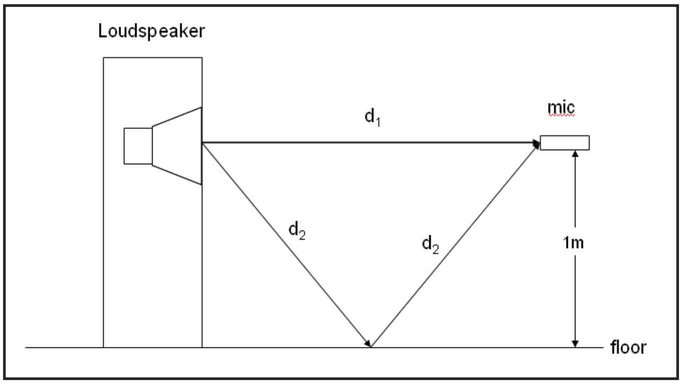


Figure 6: Test setup

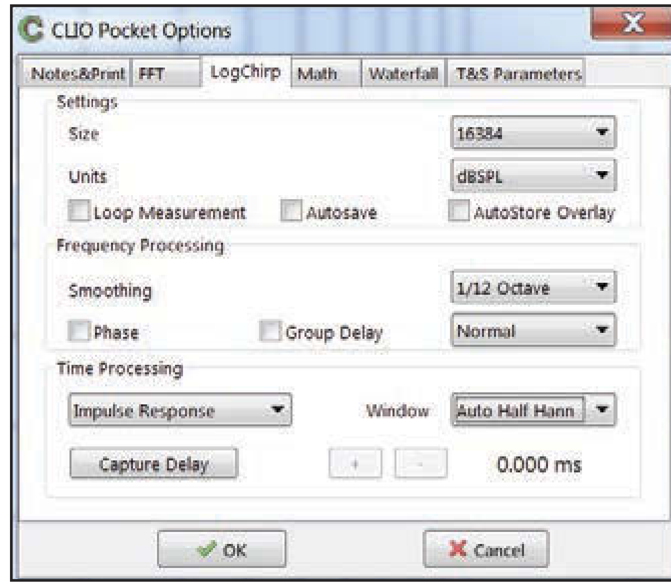


Figure 7: Parameter selection for Loudspeaker Response test

sine, it can play continuously or you can generate a tone burst by setting time on and time off values. Selecting two sine, you can set the frequency of each sine wave and the relative amplitude. This signal set is useful for intermodulation distortion (IMD) tests. All-wave and pink noise signals are available in lengths of 4K, 16K, and 64K. (This becomes important in the FFT measurement option discussed later.) Chirp lengths of 4K to 256K are available. You can also select compatible .wav files from this window.

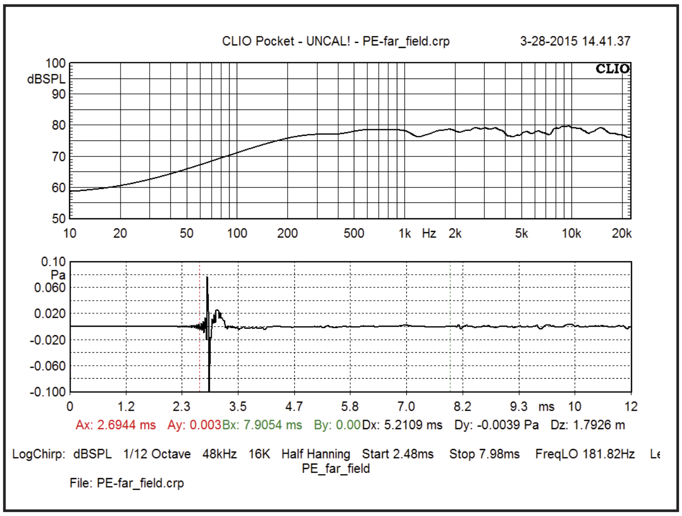


Figure 8: Time and frequency response for our two-way example

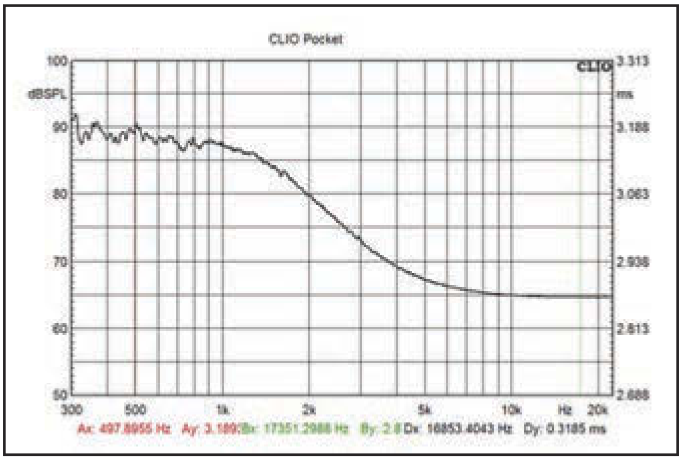


Figure 9: Excess group delay for our two-way loudspeaker example

However, the measurement settings panel is where you really get down to the business of the CLIO Pocket. Clicking on the measurement setting icon opens the CLIO Pocket Options window where you find measurement and processing options for FFT, log chirp, waterfall, Thiele-Small (T-S) parameters, math post-processing options, and a notepad to attach descriptions to saved measurements.

Selecting the FFT tab you can set the FFT size at 4K, 16K, or 64K and set measurement units of dBV, dBu, dBRel, or dBSPL. Available windows include Rectangular, Hanning, Hamming, Blackman, Bartlett, and Flattop.

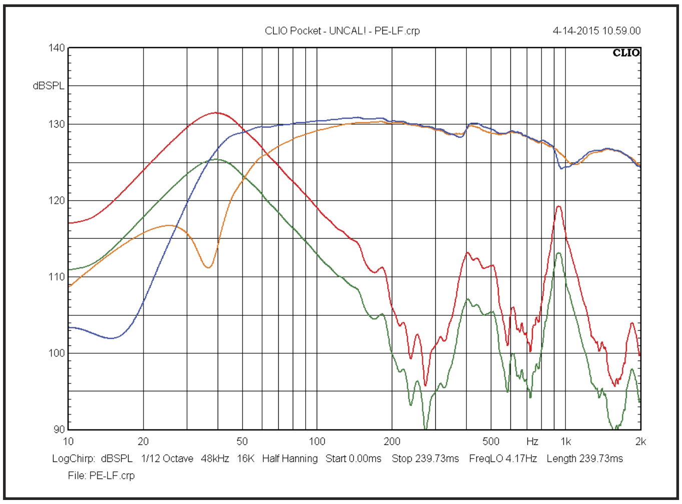


Figure 10: Near-field measurements and final low-frequency response

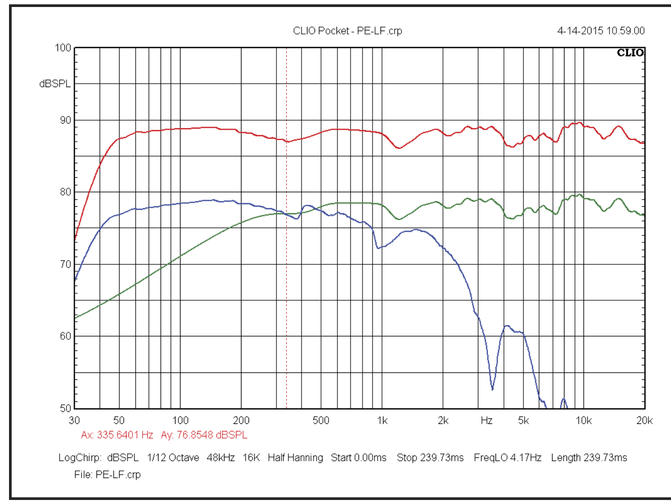


Figure 11: Far-field, low-frequency, and full-range responses

Averaging can be exponential or linear. Hold options include no hold, maximum hold, and minimum hold. Frequency processing is by FFT with smoothing options from 1/48 octave to 1 octave or real-time analysis (RTA) plots. There is also an event trigger option.

Turning to the Log Chirp window, you can select chirp lengths of 16K and 64K. Measurements units of dBV, dBu, dBRel, dBSPL, or ohms are available. Time data processing options include impulse response, step response, energy-time curve, and Schroeder decay with rectangular and auto half-Hann windows.

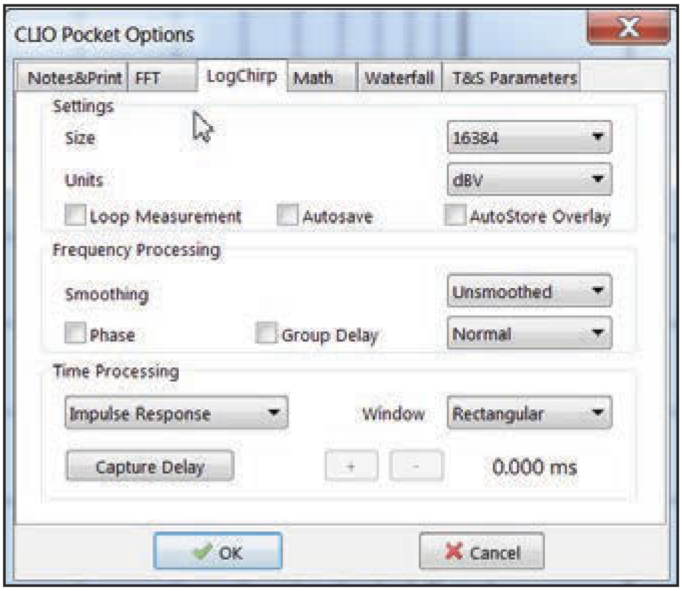


Figure 12: Log Chirp parameters for subwoofer equalizer test

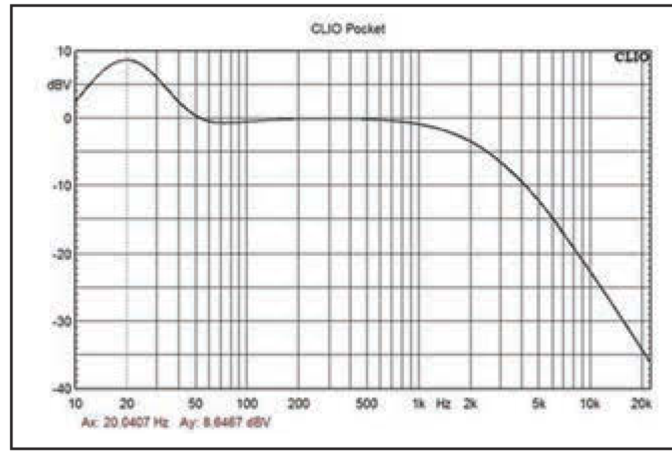


Figure 13: Equalizer frequency response

A capture delay can also be set. Frequency processing options include smoothing from 1/48 octave to 1 octave and display of normal, minimum, and excess phase and group delay.

The Thiele-Small window sets up estimation of the T-S parameters using two impedance measurements of the driver under test. The impedance data is obtained from log chirp measurements. Both the added mass and the known volume techniques are implemented.

The waterfall window sets up the parameters for computing the cumulative spectral decay (CSD) of an impulse response measured in the log chirp mode. In this window, you can set the start and stop frequencies, smoothing, decibel range, number of spectra, and time shift between spectra.

Clicking on the math tab reveals several post-processing

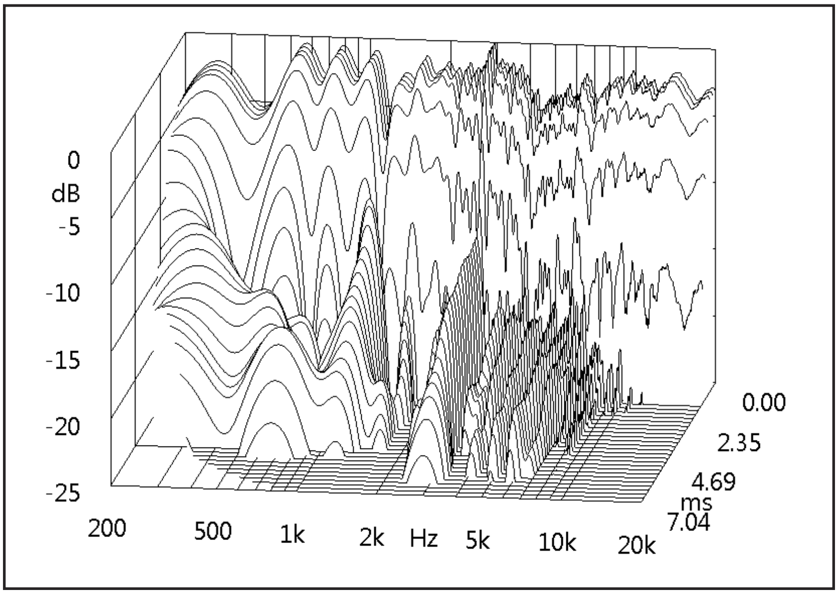


Figure 14: Two-way speaker cumulative spectral decay

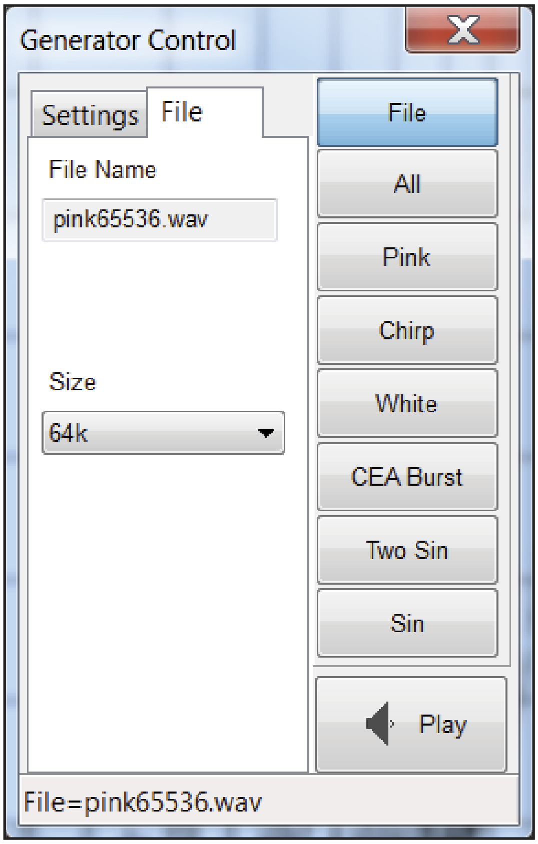


Figure 15: Pink noise setup on a generator control panel

operations that can be applied to the log chirp data. The options include adding two files together, dividing one file by another, merging a low-frequency file with a high-frequency file, level shifting a file by a specified decibel amount, and finally, implementing the microphone-in-a box low-frequency measurement technique.

Examples

Thiele-Small (T-S) Parameters: In this example, we will calculate the T-S parameters of a 7" woofer-midrange driver using the "known volume" method. Estimating the T-S parameters in CLIO Pocket is a multi-step process. Prior to estimating the parameters, two impedance measurements are needed.

Shown in **Figure 1**, they are a free-air measurement (red) and a measurement with the driver mounted in a known volume of 12.8 ltr (green). The impedance data is taken in the log chirp processing option. A 64K chirp length was chosen for improved low-frequency accuracy. Each measurement is stored for later recall during the T-S processing.

Next, load the free-air impedance measurement, open the CLIO Pocket Options window, and click the T&S Parameters tab opening the data entry menu for T-S measurement shown in **Figure 2**. Here you enter information describing the driver including manufacturer, model number, voice coil resistance (RE), and diaphragm diameter or area. If you do not know the RE, click on the "measure button" and CLIO Pocket will measure it. Then click on the Free Air button as this is the impedance data to be processed first.

Close the measurement settings menu and click on the T&S Parameters icon in the upper tool bar. As shown in **Figure 3**, a window appears showing parameters that can be calculated from the free-air impedance data. Notice that the CLIO Pocket has calculated driver cone area (SD) from the input diameter.

Load the known volume impedance data, open the T-S measurement window again, click the "Known Volume" button and enter the test box volume, 12.8l in this case (see **Figure 4**). Close the window and again click on the T&S Parameters icon

About the Author

Joe D'Appolito has been an independent consultant in audio and acoustics for 22 years. He is a long time contributor to *audioXpress* and its predecessor, *Speaker Builder.* He heads his own firm, D'Appolito Laboratories, Ltd., specializing in the design, test, and evaluation of loudspeaker systems for two-channel and home theater applications. He also served as Chief Engineer for Snell Acoustics from 2003 to 2010. In that position, he designed or led the design of some 80 loudspeaker systems for retail and custom home installation. He is the author of *Testing Loudspeakers,* the acknowledged bible on the subject, which has been translated into four languages, including Chinese. Prior to his work in audio, he led a group developing advanced nonlinear signal processing algorithms for passive sonar under government contract.

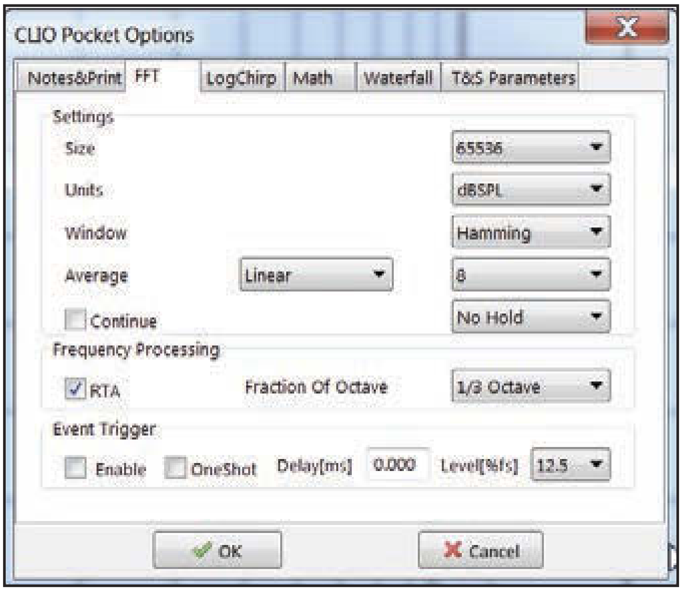


Figure 16: Fast Fourier Transform (FFT) measurement setup

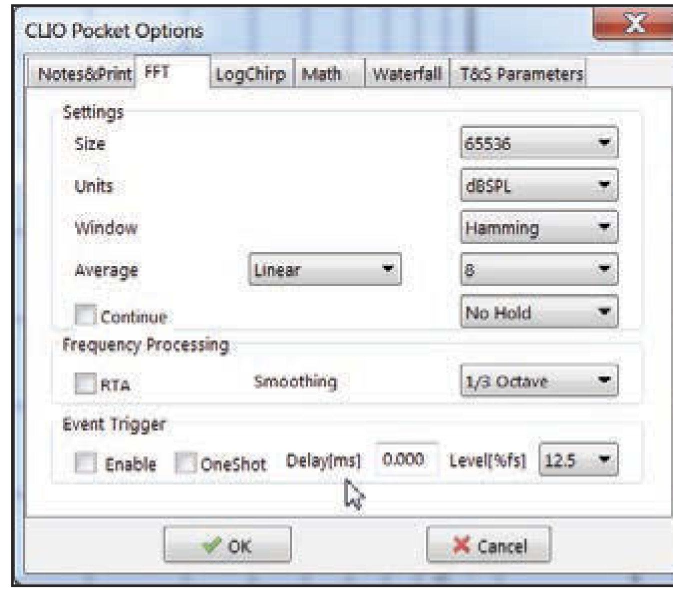


Figure 17: All-wave measurement setup

in the upper tool bar. **Figure 5** shows a window that appears with the full set of T-S parameters. If desired, the T-S data can be exported as a text file.

Loudspeaker Frequency Response: Loudspeaker frequency response is measured using the Log Chirp option. This example will examine the response of a small two-way vented loudspeaker. In the absence of an anechoic chamber, loudspeaker response measurements will invariably be corrupted by later arriving reflections from nearby surfaces. **Figure 6** shows a typical setup a hobbyist designer might encounter. In this example, the MIC–02 microphone is placed on the tweeter axis at a distance of 1 m. The first corrupting reflection will come from the floor bounce.

Parameters for the test are selected in the Log Chirp window (see **Figure 7**). Chirp length is set at 16K. The sound pressure level in decibels (dBSPL) is selected for the Y-axis frequency response display. Time processing will produce the impulse response. A Half-Hann window will be applied to the selected time interval and the computed frequency response will be displayed with 1/12th octave smoothing. **Figure 8** shows the resulting measurement.

Frequency and time-domain plots can be individually shown, but the CLIO Pocket has a useful split-screen display where both time and frequency plots are shown together. Examining the time display first and ignoring the transient build-up of the FIR anti-aliasing filter, the direct signal from the loudspeaker arrives at the microphone at about 2.7 ms.

The floor reflection appears at about 8 ms. So we have 5.3 ms of reflection-free response, which corresponds to a low-frequency limit of Fmin = 1/0.0053 = 188 Hz, so the computed frequency response is valid only above 188 Hz. Plotted response below 188 Hz is simply an artifact of the

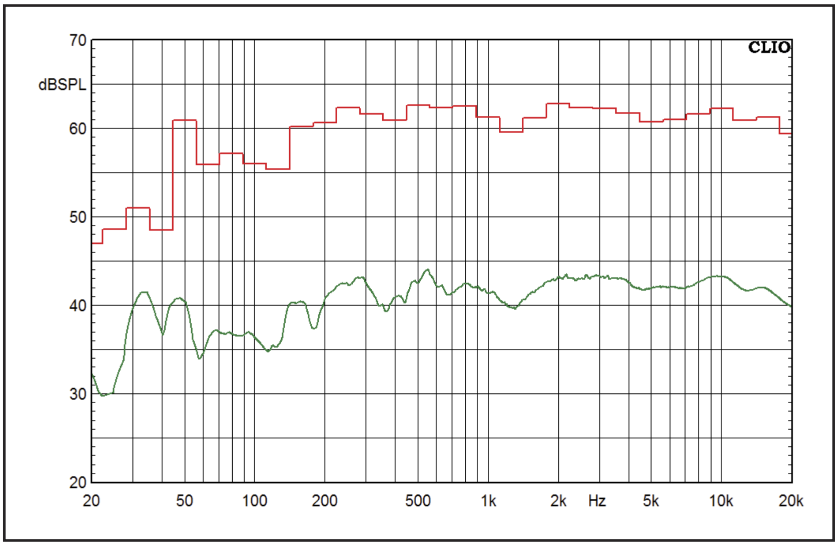


Figure 18: Real-time analysis (RTA) and all-wave response of a two-way loudspeaker

FFT. The 188-Hz number is also the resolution of the response data. The FFT algorithm least-squares fits the points between 188-Hz intervals. Above 200 Hz, the response fits within a 3.5-dB window.

At this point, we can open the Log Chirp window and request plots of phase and group delay. Perhaps the most interesting plot is the excess group delay plot. The concept of excess group delay is discussed in detail in my *Testing Loudspeakers* book. For our purposes, it is a measure of signal arrival time vs. frequency. **Figure 9** shows a plot of excess group delay for our two-way example.

High frequencies from the tweeter will arrive first. The green cursor set at 17.3 kHz shows a tweeter arrival time of 2.87 ms. The red cursor set at 500 Hz shows a woofer arrival time of 3.19 ms. Thus, the woofer is delayed relative to the tweeter by 0.32 ms or 320 µs. This delay is caused by the woofer frequency response and the low-pass filter used in the crossover network.

We have the frequency response above 188 Hz. Without an anechoic chamber, we can get a good approximation to the low-frequency response of our two-way example using the Keele near-field method. This technique is described in detail in my article "Measuring Loudspeaker Low-Frequency Response" and Don B. Keele's "Low-Frequency Loudspeaker Assessment by Near-Field Sound Pressure Measurement" (see Resources).

Here is where the power of CLIO Pocket's post-processing math functions comes in to play. The procedure involves measuring the woofer and the port near-field responses, scaling the port response by the port/woofer diameter ratio, and then adding the two responses accounting for both magnitude and phase. This low-frequency response is then merged with the high-frequency response to get the full-range response of our example. The port/woofer diameter ratio is:

R = 68 mm/138 mm = −6.14 dB

Figure 10 shows the measured near-field port response (red), the port response reduced by 6.14 dB using CLIO Pocket's "dB shift" math option (green), and the near-field woofer response (orange). Now using the "add file" math post-processing option, the near-field woofer response, and the scaled port response are added to get the final low-frequency response estimate (blue).

We need one more step to get the full-range response of our two-way example, merging the far-field and the near-field responses to get the full-range response. First use the "dB level shift" math function to level match the near and far-field responses.

Figure 11 shows the far-field response (green) and the low-frequency response (blue) level shifted to match the far-field sensitivity. The two plots meet around 350 Hz. Using the "Merge LF file" math option, the low-frequency response is merged with the far-field response at 350 Hz to get the final full range response (red), which has been level shifted up 10 dB for clarity.

Besides loudspeaker frequency response, the log chirp option is also useful for measuring the frequency response of electronics. This next example measures the response of a low-frequency equalizer circuit for a closed-box subwoofer. To do this, the CLIO Pocket output is connected to the equalizer input and the equalizer output is sent to the CLIO Pocket input.

Resources

J. A. D'Appolito, "Measuring Loudspeaker Low-Frequency Response," www.audiomatica.com.

——, *Testing Loudspeakers*, Chapter 6, Audio Amateur Press, 1998, Item # AA-BKAA045 available at www.cc-webshop.com.

——, *Testing Loudspeakers*, Chapter 7, Audio Amateur Press, 1998, Item # AA-BKAA045 available at www.cc-webshop.com.

D. B. Keele, "Low-Frequency Loudspeaker Assessment by Near-Field Sound Pressure Measurement," *Journal of the Audio Engineering Society* (*JAES*), April 1974.

R. H. Small, "Simplified Loudspeaker Measurements at low Frequencies," *Journal of the Audio Engineering Society* (*JAES*), January/February 1972.

Figure 12 shows the Log Chirp test parameters. Notice the response will be measured in decibels of voltage (dBV). **Figure 13** shows the resulting equalizer response. You can see the equalizer produces a 8.65-dB boost at 20 Hz. High-frequency equalizer response is deliberately rolled off.

Waterfall Option: The waterfall option is used to compute the cumulative spectral decay (CSD). The CSD measures the frequency content of a loudspeaker's decay response following an impulsive input. Ideally, a loudspeaker response should instantly fade away.

Real world speakers have inertia and stored energy that take a finite time to dissipate. For a full discussion of the CSD read my book *Testing Loudspeakers*. The CSD computation involves a series of FFT calculations. In CLIO Pocket, these calculations are performed in the Waterfall Option and displayed with a 3-D waterfall plot with time running along the x-axis, frequency along the y-axis and amplitude along the z-axis.

The CSD for our two-way example shown in **Figure 14** is computed from the impulse response shown in **Figure 8**. The initial decay is somewhat disorganized, but after about 2 ms low-level decay modes appear in the 2-to-7-kHz region.

The CSD is commonly used to examine resonant decay modes of a loudspeaker. However, in the CLIO Pocket manual, Audiomatica shows a clever application of the CSD to analyze room acoustics and to find any resonant modes in a listening room.

To do this, you must first get the room response to an impulse, an acoustic impulse! A spark gap is often used for this purpose, but Audiomatica does it by bursting a balloon! Check the manual available on the Audiomatica website (www.audiomatica.com) for details.

FFT Analysis: As discussed earlier, CLIO Pocket has several signals available in the FFT analysis option. We will look at two that can be used in the frequency response assessment of our two-way example. They are pink noise and the all-wave signal. Pink noise is the signal used to get the 1/3 octave response or RTA. The all-wave signal has tones corresponding to all FFT bins. In CLIO Pocket, it can be used to produce an RTA plot or a continuous frequency response plot.

As shown in **Figure 15**, we open the generator control panel and select a pink noise signal of 64K length. **Figure 16** shows the measurement setup for the RTA analysis using pink noise.

Analysis length is set at 64K to match the pink noise length. This puts more FFT samples in the lowest 1/3 octave (20–25 Hz) to give us a better estimate of low-end response. We are using a Hamming window, averaging eight measurements and plotting the result in 1/3 octave bands. Notice the RTA button is selected to give the traditional RTA plot.

The all-wave signal is also set at 64K length. **Figure 17** shows the measurement setup for the all-wave analysis. The settings are similar to the RTA analysis, but the RTA button is not checked so the resulting response plot will be a continuous curve.

The measured RTA (red) and the all-wave (green) responses are plotted together in **Figure 18**. They are offset by about 15 dB for clarity. These responses are similar. Both fit in a 5-dB window above 200 Hz. This agrees well with the chirp analysis shown earlier in Figure 8. Response below 200 Hz differs from Figure 8 because the FFT measurements cover the full frequency range and thus include room effects.

Summary

CLIO Pocket—presented at the 2014 Audio Engineering Society (AES) show in Los Angeles, CA—is now a finished hardware product sold since March 2015. Audiomatica confirmed to me that the software (now at release 1.21) is going to be developed further and that all users receive free upgrades for all the life of the actual 1.*xx* major release.

This review shows that it already provides all the measurement and analysis capabilities needed to design and analyze loudspeaker systems and electronic circuits. Considering the rich set of stimulus signals, the data acquisition modes, and the post-processing functions, it is clear that much more capability will be added to the CLIO Pocket as the software develops.

Build a Powered Subwoofer with a Unique Design

This DIY subwoofer design adds dimension to a 3.1-channel loudspeaker system.

By
Bill Christie
(United States)

Some months ago, I designed and built a retro-style preamp. It included the standard volume, balance, and tone controls. But I also decided to add several features to the basic design, including a phono preamp, a derived center-channel output by summing the right and left channels and finally an active crossover to provide a subwoofer output.

In order to put it to use, I then built a small power amplifier, using the LM4782 three-channel amplifier chip from Texas instruments (TI) and three small vented speaker systems using Dayton RS100-8 speakers.

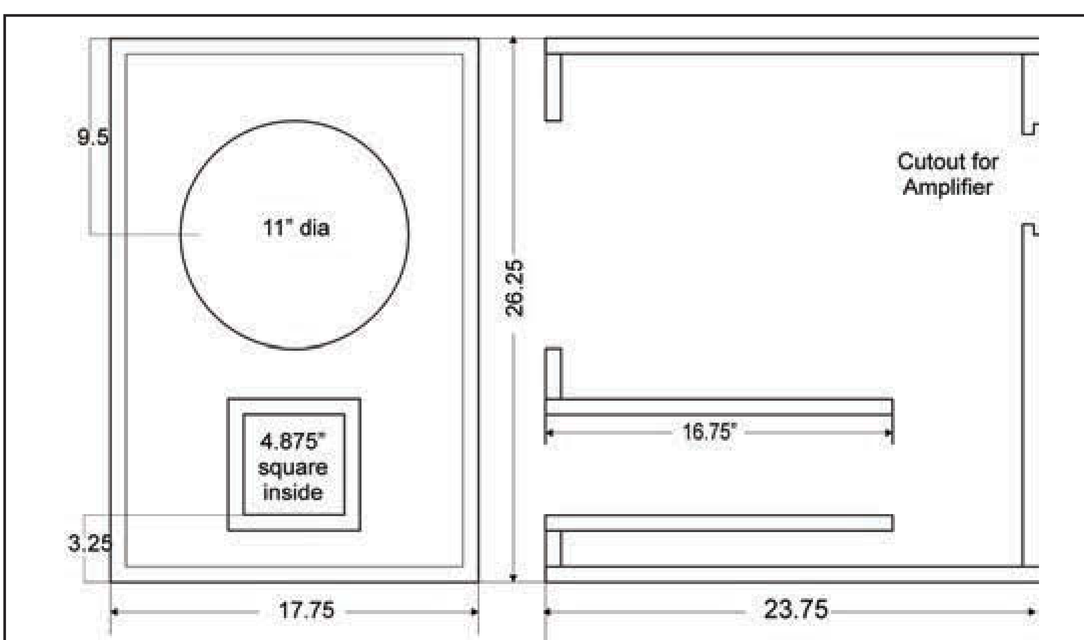


Figure 1: Here are the dimensions for the subwoofer enclosure that I constructed from MDF and covered with oak veneer (see Photo 1).

While I was happy with the sound, the system lacked deep base. So I decided it was time to build a subwoofer to complement the system.

To complete my 3.1 channel system, I made a rather unusual speaker choice—the Pioneer model TS-W303R 12" component subwoofer with 1,200 W nominal power. This speaker was intended for car audio in a rather small enclosure that restricted the bass response (down 20 dB at 20 Hz). But in a larger enclosure, it reproduces exceptional bass.

I built the enclosure using 0.75" MDF and covered it with oak veneer (see **Photo 1**). I added some internal bracing and filled about one-third of the enclosure with acoustically absorbent material. **Figure 1** shows the dimensions of the enclosure I constructed for this speaker. **Figure 2** shows the simulated response.

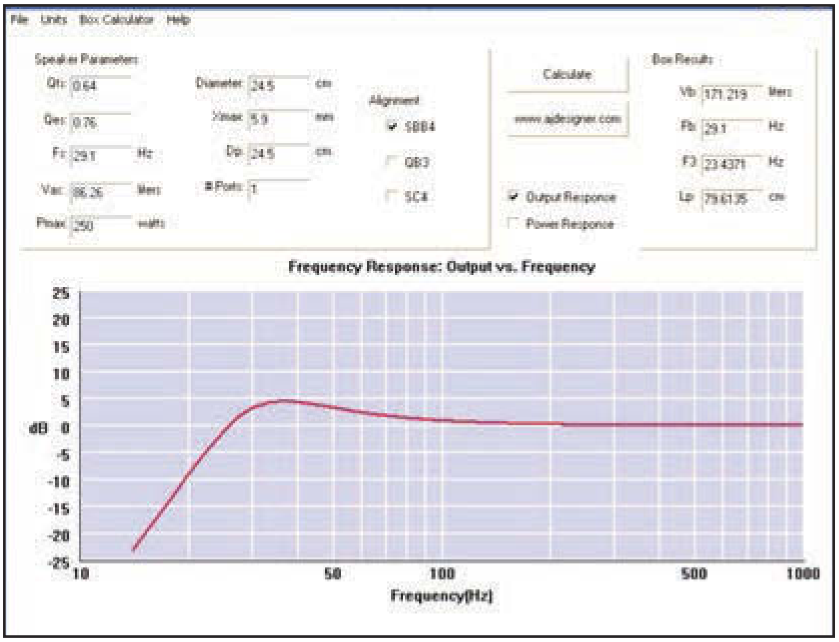


Figure 2: These are the results of the simulation test I ran for my new subwoofer.

Adding an Amplifier

Next, I selected an amplifier to power the subwoofer. The amplifier is based on the Texas Instruments LM49830 integrated circuit (IC) driving a MOSFET output stage. **Figure 3** shows the various blocks of electronics and how they are tied together. With some minor variations, I basically implemented the reference amplifier shown in Texas Instruments' datasheet.

Figure 4 shows the schematic that I used to

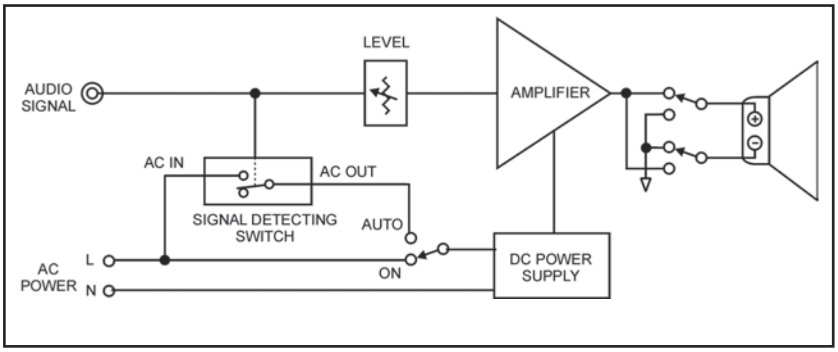


Figure 3: The block diagram details the various electronics and how they are tied together.

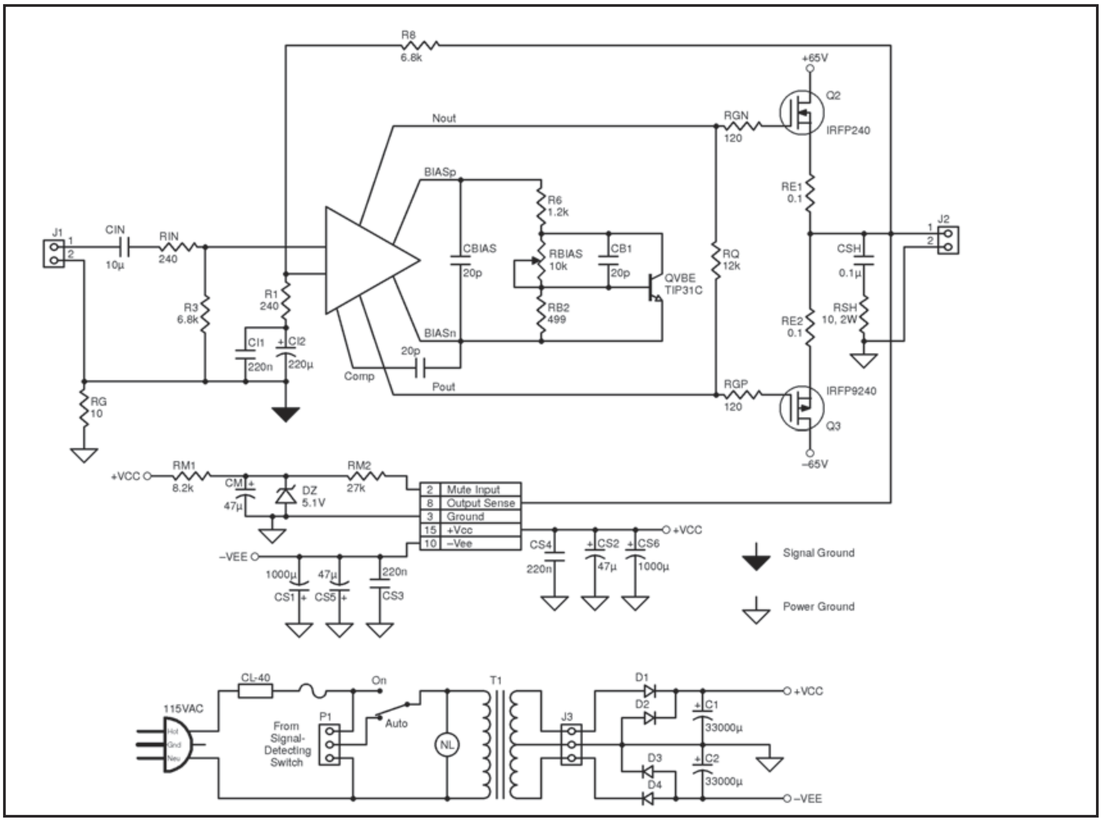
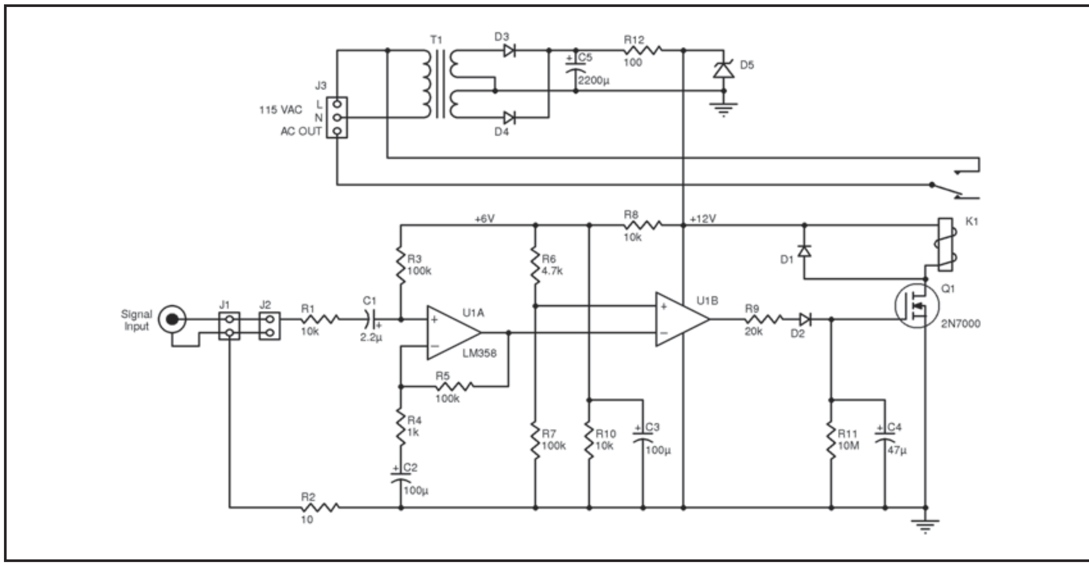


Figure 4: The schematics used to build the subwoofer is based on the Texas Instruments LM49830 IC driving a MOSFET output stage.

Figure 5: You can either manually turn on the amplifier or use a signal detecting relay circuit.



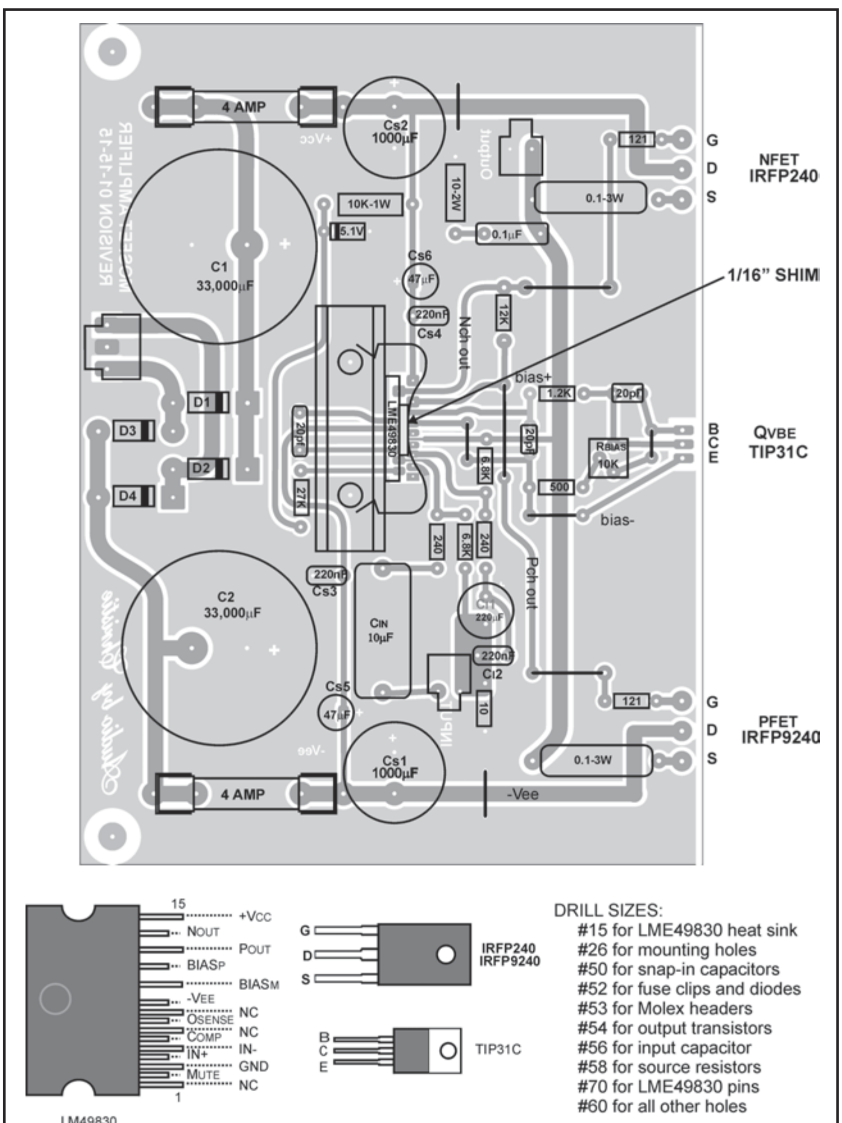


Figure 6: This is the PCB layout for the amplifier.

build the amplifier. Since I included an 80-Hz low-pass filter in my preamplifier, I didn't include one here. However, if you build this subwoofer to be used with another preamplifier or a processor that does not include a low-pass filter, you should use an electronic crossover or a passive low-pass filter.

To switch power to the amplifier, I provided two options—you can either manually turn on the amplifier or use a signal detecting relay circuit to turn it on (see **Figure 5**).

Rod Elliott of Elliott Sound Products has published this circuit as "Project 38: Signal Detecting Auto Power-On Unit." I've made a few adjustments based on suggestions that were in his article. For a complete description of the circuit and how it functions, visit his website, http://sound.westhost.com.

Printed circuit boards (PCBs) are available for many of these and can be purchased from him at reasonable costs. As for this circuit, I can only take credit for the printed circuit board layout, and

Resources

R. Elliott, Project 38: Signal Detecting Auto Power-On Unit, Elliott Sound Products, http://sound.westhost.com.

HeatsinkUSA, http://heatsinkusa.com.

Sources

TS-W303R subwoofer
Pioneer | www.pioneerelectronics.com

LM49830 Integrated circuit (IC)
Texas Instruments, Inc. | www.ti.com

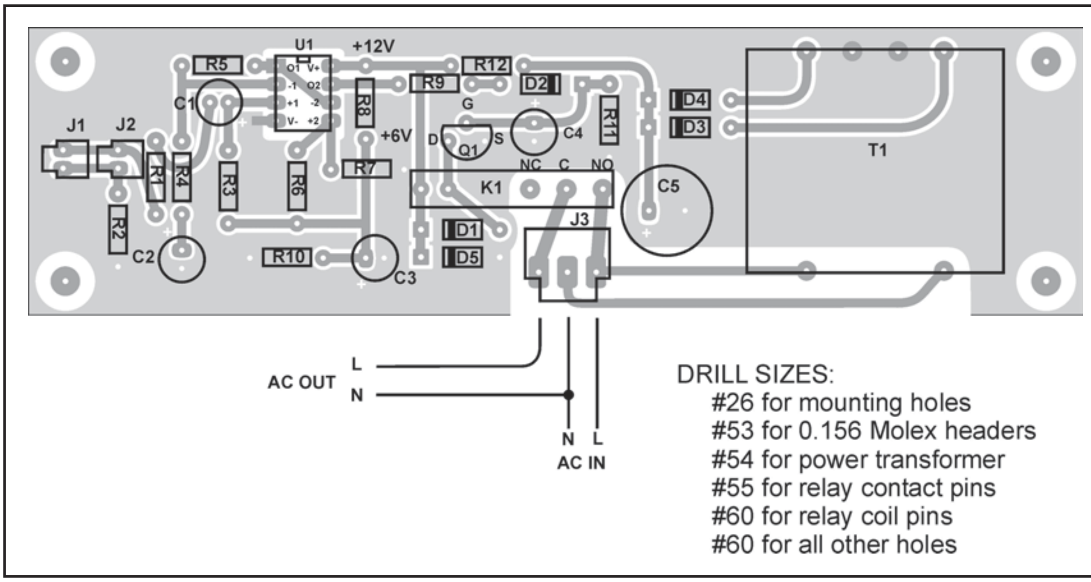


Figure 7: This is the PCB layout for the signal detecting switch.

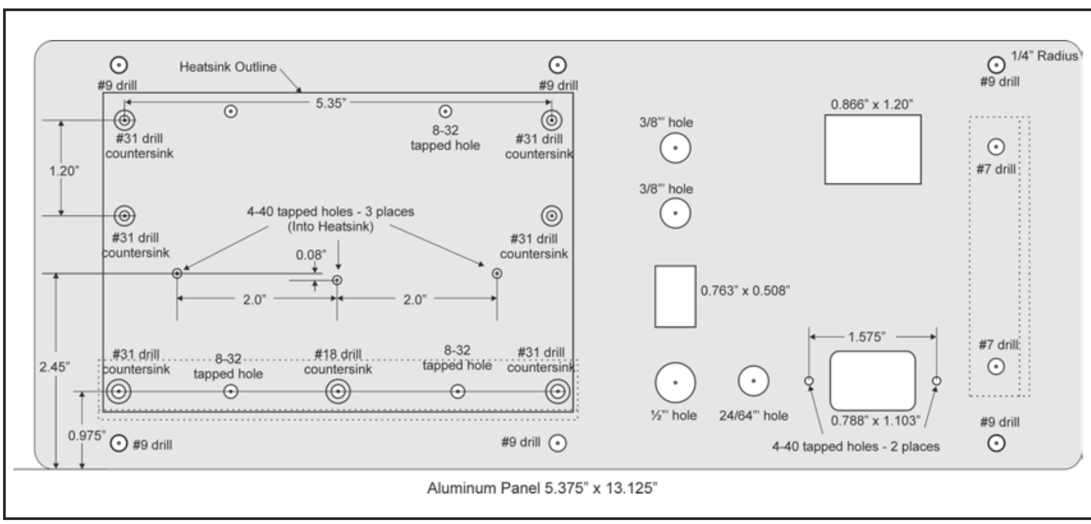


Figure 8: The drawing details the main panel.

even that was improved by suggestions from Elliot.

The circuitry is laid out on two separate printed circuit boards (see **Figure 6** and **Figure 7**). Note that on the amplifier board, I used a shim between the LME49830 and the heatsink retaining clip. This is required to hold the chip tightly to the heatsink. Also, an insulator is required between the chip and the heatsink. **Figure 6** shows the PCB layout for the amplifier. **Figure 7** shows the PCB layout for the signal detecting switch.

The entire amplifier is mounted on a 0.125″ aluminum plate that measures 13.125″ by 5.375″. Fasten a sub plate to mount the power transformer to the main plate with 0.75″ angle and 10-32 screws. Another sub plate is used to support the back of the amplifier's PCB. I printed the main panel drilling drawing full scale and used it as a drilling template (see **Figure 8**).

Figure 9 shows the mounting plates for the PCB and transformer. All the screws that are between the heatsink and the main panel are countersunk so the panel and the heatsink are in direct contact. In fact, I used heatsink compound between the two. This is because the output devices are mounted to the panel and the heat from them has to be conducted through the panel to the heatsink. Note that I used a heatsink that I had on hand.

About the Author

Bill Christie developed an interest in electronics at the age of 6 after his father helped him build his first crystal set. Since then he has built numerous kits. Bill started building amplifiers from scratch while he was still in high school, starting with a mono tube amp designed by Mullard. He graduated from the University of Akron in Ohio with a BS in Physics. Bill is now a retired electrical engineer from Goodyear Tire & Rubber Co., where he spent 35 years designing controls and instrumentation for manufacturing.

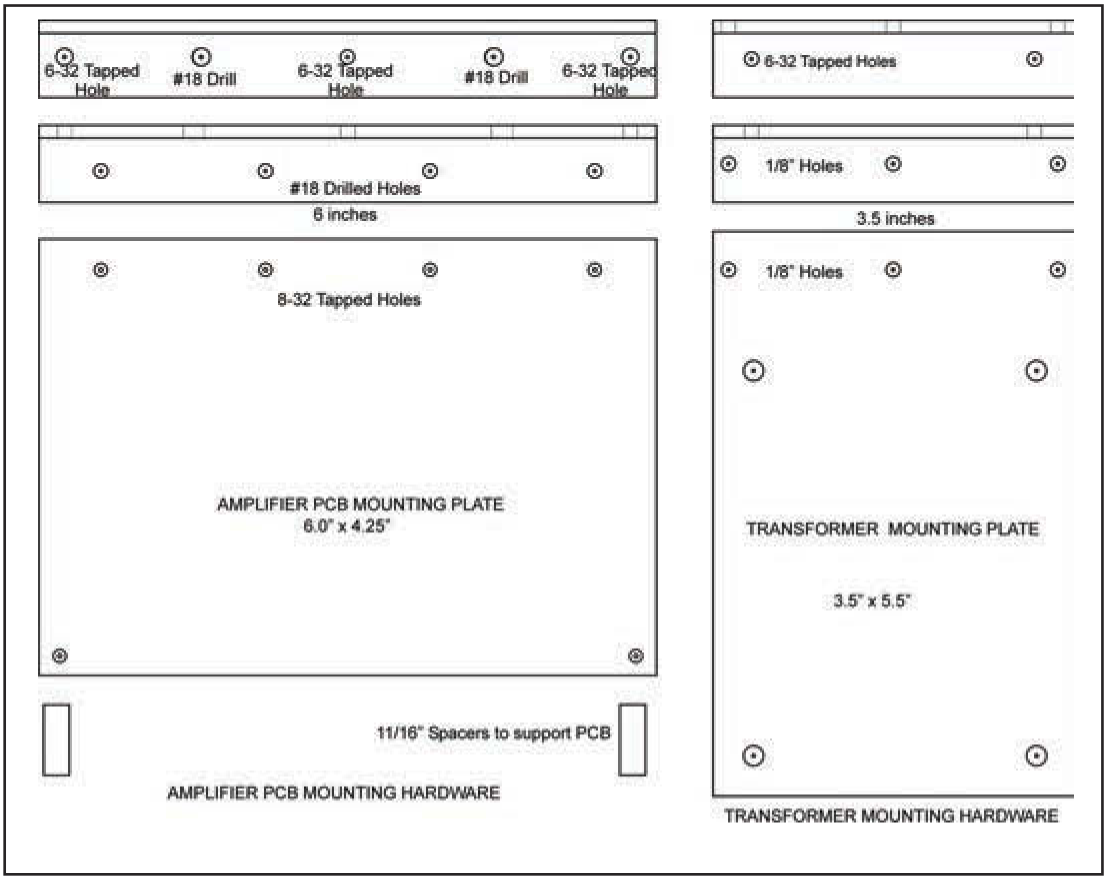


Figure 9: The mounting hardware for both the PCB and the transformer are detailed.

The heatsink shown that can be purchased from HeatsinkUSA and is more ideal for this application.

The assembly sequence is critical with respect to the amplifier board. The following sequence will insure that the transistors line up with the holes in the main panel.

Figure 10 shows a top view of the panel assembly and **Figure 11** shows a side view. The correct sequence is as follows:

- Assemble all the mechanical parts to the main panel. Save the heatsink till last.
- Make sure that the countersunk screws are flush or slightly below the panel's surface. Then fasten the heatsink to the panel using a generous amount of heatsink compound to the back of the heatsink.
- Populate the amplifier board soldering all but the output and bias transistors.

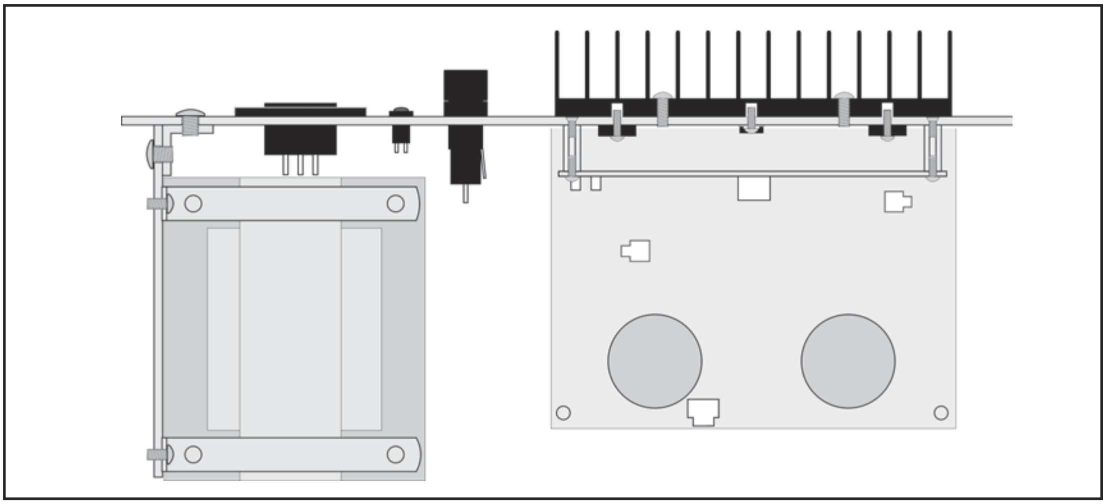


Figure 10: This is the top view of the panel assembly.

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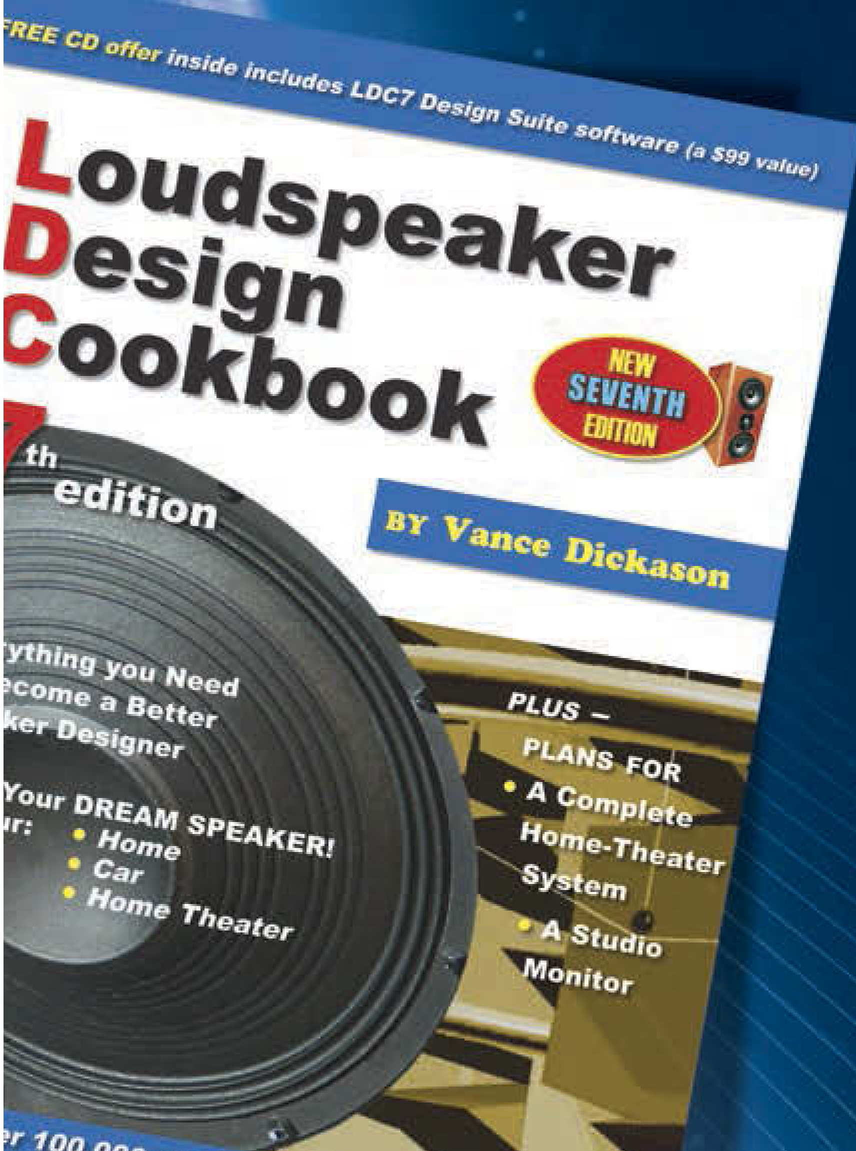
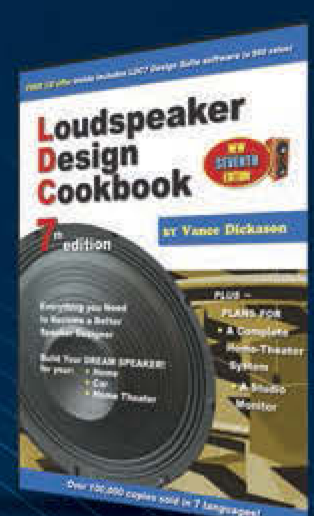




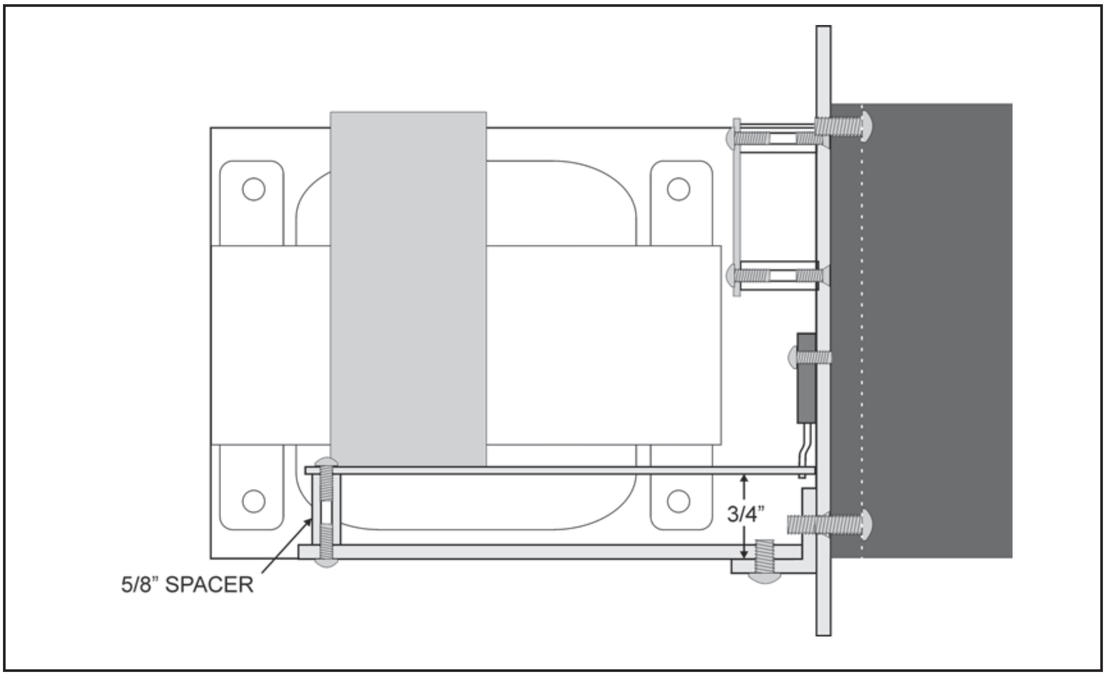


Figure 11: This is a side view of the panel assembly.

- Then with the unsoldered transistors installed in the amplifier board, mount the board to its mounting plate.
- Next, with the board parallel to the plate, fasten the transistor tightly to the panel with appropriate insulator and heatsink compound.
- Temporarily remove the mounting plate and solder the transistors to the board, keeping the

AMPLIFIER PARTS LIST

Component	Value	Source
Capacitors		
C_{IN}	10 µF	Digi-Key # 399-11495
C_{I1}	220 µF	Digi-Key # P5183
C_{I2}	220 nF	Digi-Key # 399-9687
C_{COMP}	20 pF	Digi-Key # 1265PH
C_{BIAS}	20 pF	Digi-Key # 1265PH
C_{SH}	100 nF	Digi-Key # 399-4151
C_{S1}	1,000 µF	Digi-Key # 493-1129
C_{S2}	1,000 µF	Digi-Key # 493-1129
C_{S3}	220 nF	Digi-Key # 399-9687
C_{S4}	220 nF	Digi-Key # 399-9687
C_{S5}	47 µF	Digi-Key # 399-6588
C_{S6}	47 µF	Digi-Key # 399-6588
C_M	47 µF	Digi-Key # 399-6588
C_{B1}	20 pF	Digi-Key # 1265PH
C1	30,000 µF	Digi-Key # 338-3814
C2	30,000 µF	Digi-Key # 338-3814
Diodes		
Dz	1N4733	
D1	6A4-G	Digi-Key # 641-1817-1
D2	6A4-G	Digi-Key # 641-1817-1
D3	6A4-G	Digi-Key # 641-1817-1
D4	6A4-G	Digi-Key # 641-1817-1
U1	LME49830	
Q_{VBE}	TIP31C	
Q_{NFET}	IRFP240	
Q_{PFET}	IRFP9240	

Component	Value	Source
Resistors		
R_{IN}	240 Ω	
R_S	6.8 Ω	
R_I	240 Ω	
R_{B1}	1.2 kΩ	
R_{B2}	499 Ω	
R_{BIAS}	10 kΩ POT	
R_Q	12 kΩ	
R_{E1}	0.1 Ω	
R_{E2}	0.1 Ω	
R_{SH}	10 to 2 W	
R_{M1}	8.2 kΩ	
R_{M2}	27 kΩ	
R_{GP}	120 Ω	
R_{GN}	120 Ω	
Miscellaneous		
Heatsink		HeatsinkUSA SKU: B004
LME49830 Heatsink Aavid Thermalloy		Digi-Key # HS335
Insulators		Digi-Key # WM5225
Headers		Digi-Key # 926-1492
Receptacle		Digi-Key # Q335
S1	SPDT Switch	Digi-Key # 679-1187
Fuse Clips		Digi-Key # F4187
Fuse Holder		Digi-Key # F006
Transformer		
T1	80 VCT at 6 A	

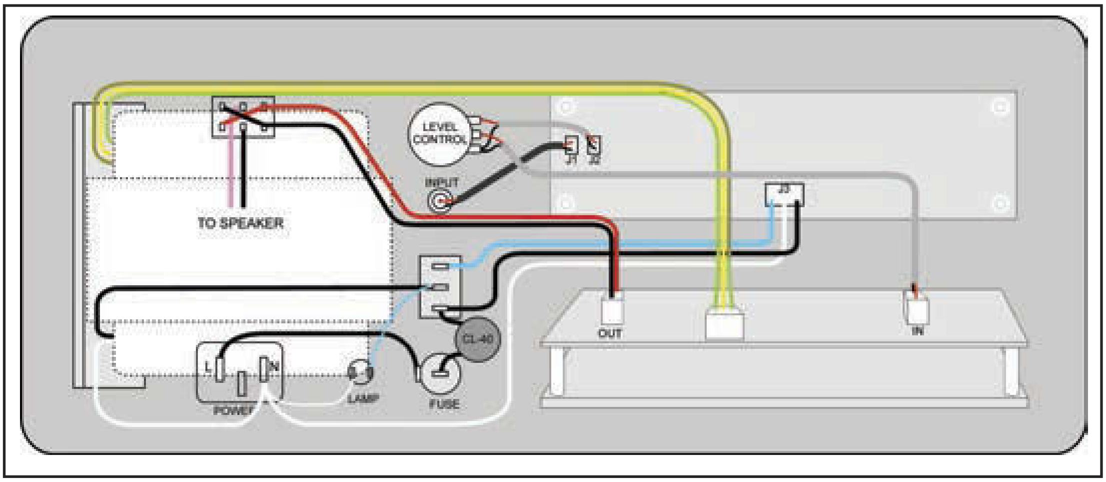


Figure 12: This diagram shows how I wired the panel.

board 0.75" above the angle. Trim transistor leads away from the angle. Once finished, the mounting plate can now be reinstalled.

- Now install the signal detecting board, input jack, power connector, fuse holder, and so forth on the main panel and the panel is ready to be wired as shown in **Figure 12**.

Before applying power to the amplifier, turn the bias control trimpot fully clockwise. Then, with no input signal, apply power with the power switch in the on position.

Finally, adjust the trimpot so that you get 15 mV across the source resistors. Let it warm up for about 20 minutes and readjust the trimpot, if necessary. **Photo 2** shows the completed amplifier ready to install in the speaker cabinet.

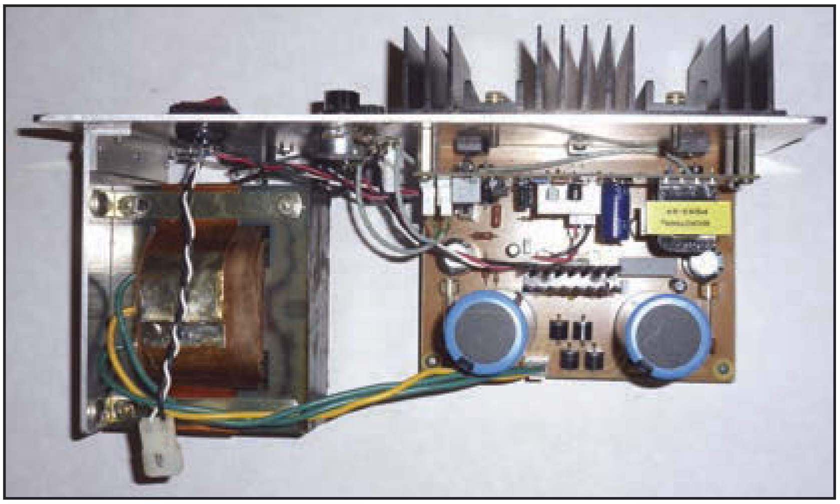


Photo 2: This is a view of the completed amplifier.

Signal Detecting Switch Parts List

Component	Value	Source
Capacitors		
C1	2.2 µF	Digi-Key # 493-10467-1
C2	100 µF	Digi-Key # 493-1548
C3	100 µF	Digi-Key # 493-1548
C4	47 µF	Digi-Key # 483-11789-1
C5	2,200 µF	Digi-Key # P5157
Diodes		
D1	1N4007	
D2	1N4007	
D3	1N4007	
D4	1N4007	
D5	1N4742	
Miscellaneous		
K1	12-V Relay	Digi-Key # PB1019
T1	Transformer	Digi-Key # MT2218
SW1	Single-pole, double-throw (SPDT)	Digi-Key # 679-1187
Q1	2N7000	
U1	LM358	

Component	Value	Source
Resistors		
R1	10 kΩ	
R2	10 Ω	
R3	100 kΩ	
R4	1 kΩ	
R5	100 kΩ	
R6	4.7 kΩ	
R7	100 kΩ	
R8	10 kΩ	
R9	50 kΩ	
R10	10 kΩ	
R11	10 kΩ	
R12	75 to 0.5 W	
Socket		
SKT	8-pin dual-in-line package (DIP)	Digi-Key # ED3038

Entrepreneurs Focus on Sound Quality and Philanthropy

Westin Bye

An Interview with Westin Bye and Cooper Buss, Founders of LIFE Acoustics

Cooper Buss

By
Shannon Becker
(United States)

SHANNON BECKER: Tell us a little bit about your background.

WESTIN BYE: I grew up as the scapegoat and make-up model for two sisters. Our parents are serial entrepreneurs and business consultants. I received two educations. The more normal one was studying mechanical engineering at the University of Minnesota. However, starting much earlier, my parents encouraged me to look at business critically and taught me important skills such as negotiation, market analysis, and networking.

COOPER BUSS: We both grew up in Orono, MN, a semi-rural suburb of Minneapolis. I went to Carleton College where I received a BA in Economics. I worked as an executive leader in Target stores for two years before leaving to start this company with Westin! I have one sister and five little brothers who are all adopted. I spend a ton of time with them and love them dearly. Seeing how much care and attention children need has really motivated me to aim LIFE toward serving children that struggle to achieve their full potentials due to physical disabilities or lack of resources.

SHANNON: How did you two meet? And what made you decide to go into business together?

COOPER: Westin moved in down the road from me in fourth grade. We rode the bus home together every day and quickly became good friends. We have wanted to go into business together for as long as we can remember. We always dreamed of taking one of our crazy ideas and running with it. LIFE was one of those crazy ideas. We ultimately decided to start LIFE Acoustics because we realized we were really good at working together to brainstorm and

This is an early prototype for LIFE Acoustics first earphones.

problem solve, both of which we see as essential for an entrepreneurial team.

SHANNON: How did you become interested in audio?

WESTIN: I have a problem. I get obsessed with a new topic a few times a year. My college roommate was a certified audiophile and a music snob, I did not like that he knew so much more about music and audio than I did so I went on a research rampage. I learned as much as I could and tried out as many headphones as possible to compare. Needless to say he is now a big fan of LIFE headphones!

COOPER: I have been a music enthusiast ever since I started playing piano as a child. I was always in every sort of band I could find in and out of school. After seeing the Blue Man Group (BMG), a few friends and I built one of the BMG's instruments (called a drumbone) and performed a BMG song on it in front of our entire high school! We had a friend, who had a bunch of equipment and lots of experience, help us with the audio. He really opened my eyes to what a difference a better sound system could make. From there, I started buying and combining speakers and eventually moved into headphones when my dorm room couldn't fit my speaker collection. I am an amateur DJ and an avid discoverer of new music. I have also worked part time as a sound engineer for years at various churches and events. All of this has shaped me into the audiophile I am now and informs the decisions I make to guide LIFE headphones to being of the utmost highest quality.

SHANNON: Where did you get the idea for LIFE Acoustics (www.lifeacoustics.com)?

WESTIN: We talked and brainstormed for days in a conference room with nothing but a whiteboard and a pair of early LIFE prototypes. We knew we wanted to do more than sell headphones, because we are both deeply committed to giving. We realized that we had a perfect connection to an organization that we trusted and cared about (Ayúdame a Escuchar) and it all fit together beautifully. After a long phone conversation with Ayúdame's director, we decided to set up our company as a social business to sell high-end headphones for the active audiophile and use our proceeds to donate hearing aids to children living in poverty with profound hearing impairments. We chose the name LIFE Acoustics because our headphones give life to our music, bring music to our lives in a way that is comfortable and ergonomic, and because our headphones enable us to give new life with hearing to some of the most disadvantaged people on earth.

SHANNON: What is Ayúdame a Escuchar (Help Me Hear) and how did you become involved with the organization?

COOPER: Ayúdame a Escuchar (Help Me Hear or HMH) is a nonprofit run through the Mesa Rotary in Arizona. Westin's father, along with several close family friends, helped set up and build the organization to what it is now. We have been listening to Westin's dad's stories from his trips with the organization around the dinner table since we first met. We were always moved by how much they changed the lives of the

Westin Bye and Cooper Buss wanted to design a headphone that would appeal to people with an active lifestyle.

children they served. When we decided to pursue a partnership with them, we contacted Bob Jensen, HMH's director, and he was excited to get the extra support we hope our partnership will offer.

After a few more phone calls with Bob, we were hooked. Unlike with other organizations that give away hearing aids, we learned that HMH bases the care they give on long-term relationships.

LIFE Acoustics's first completed product is the LIFE earphone.

They return to the same communities every year to check in and up on the children they have helped in the past. We were also excited for the opportunity to partner with HMH to donate hearing aids powered by solar-rechargeable batteries. We knew that families struggled to afford batteries for their children's hearing aids, so we saw this as an amazing way to help push HMH's service to the next level of sustainability.

As our company grows and can fund more than just what HMH needs, we will partner with other organizations that provide long term, sustainable care to the impoverished hard of hearing and help them transition into solar-powered devices.

SHANNON: Tell us more about your first product, LIFE Headphones.

WESTIN: We worked with a team of audiologists and engineers to develop the LIFE earphone. It takes the ergonomics and the technology found in hearing aids and adapts them to the ear of the common listener, creating a headphone that fits in your ear like an earplug. It sounds amazing.

Early on, we started steering the headphone design toward the active listener. We had so many bad experiences with other headphones while running, exercising, or just walking around the house, that we wanted our headphones to really solve the problems we experienced with other similar products. We were tired of our headphones falling out or falling off whenever we moved too fast, shook our heads, or snagged the cord on something. We wanted LIFE to do away with all of these problems and it does! The innovative C shape and unique cord placement means our headphones will comfortably stay in your ears no matter what you do, including pulling on the cord.

SHANNON: Can you share some of the challenges involved with the design?

COOPER: It is hard to fit so many components into such a shape! We and our engineers worked hard to get the two microdrivers that each earpiece contains to sit just right. Every little change affects the sound, so you have to be super careful with the physics of everything. We decided to tune LIFE flat with a little emphasis on the bass and the treble, which are the most important aspects of your music when you are exercising. Plus, this tuning diminishes what is called listener fatigue, which many runners suffer from. Essentially, after long term use, your ears start to adapt to sounds in the middle of the sound spectrum so you would have to turn your music up to hear it the same way you did. Our headphones are tuned to minimize this fatigue so you can listen worry free for hours.

SHANNON: You chose to fund your first product through Kickstarter. Why go the crowdfunding route? How has that experience been for you?

WESTIN: Crowdfunding allows us to stay focused on our mission (without having to answer to outside investors) and at the same time promote awareness and gain supporters for our cause. The experience has been great so far, but much more work than we had expected! Running a campaign is challenging because it is still quite an uncharted medium for marketers. We are getting amazing responses from people all over the world who care about what we do and how we are doing it, which is very encouraging to us.

SHANNON: What's next for your company?

COOPER: Really everything is next! The tumultuous path of moving from a start up to a small company and beyond. As the market sees fit, we have dreams of branching out into all sorts of other amazing products that can be paired with social missions. We will start by focusing on personal audio products and then ideally move into products that parallel other global social needs. The possibilities are endless! There is so much need in this world and so many cool ways to help!

SHANNON: Are you working on other new products?

WESTIN: We are working on both single and triple driver systems. After that we will focus on a wireless option, perhaps Bluetooth, but more ideally an alternative that allows for higher quality sound transfer. We are also starting to work on integrating soft rubber elements to the earpiece that will make it even more comfortable for more ear sizes and shapes.

Troubleshooting the Tough Ones

In past articles, we have discussed troubleshooting hollow-state amplifiers and power supplies. The techniques we discussed fall into the "Sherlock Holmes" and "Dragnet" investigative categories: apply logical deductions and then round up all the usual suspects. When these methods work, they often lead to the fastest solution to the malfunction. But sometimes, more drastic measures are necessary. I call these the "divide and conquer" methods.

By
Richard Honeycutt
(United States)

The divide and conquer approaches to troubleshooting include signal injection and signal tracing. In the former method, a signal is injected into the circuit at the input. The output is monitored to see if it is noisy, too low in amplitude, distorted, or otherwise exhibits whatever defective behavior initially brought the equipment to the troubleshooting bench. Presumably, with the signal injected at the input, the output will exhibit the problem under investigation. So the signal-injection point is moved toward the output one stage at a time, until a correct output is observed. The problem has then been isolated to the last stage in which the problem was observed.

Signal injection worked well for troubleshooting the radio frequency (RF) and mixer sections of hollow-state radio receivers, in which the signal could be applied from an AM or FM signal generator feeding its output to a single-turn loop inductor ("gimmick") held near the control grid terminal of the tubes. For audio amplifiers, signal injection becomes more complex. First, there is the need to couple the injected signal through a capacitor to avoid disturbing the DC conditions of the circuit under test. Second, there is the need to know what level the signal should be, so as not to overdrive the amplifier. Third, a hollow-state signal generator should be used, because the charging spike that will appear at the generator output when a signal is injected into a high-DC-voltage point can be high enough to cause primary breakdown (a momentary internal short)—and perhaps secondary breakdown (destruction)—of a solid-state signal generator's output stage.

If the problem is noise or hum, something similar to signal injection can be used on hollow-state amplifiers without running into those three complications. In chasing a noise problem, no signal is fed to the amplifier input. The noise will be heard from the speaker or observed on an oscilloscope. You can start with the input tube and remove one tube at a time. When the noise goes away, you know that the problem is in either the stage whose tube you have just removed, or in the preceding stage. In using this method with the speakers connected, it is safest to use a woofer, as the signal spike produced

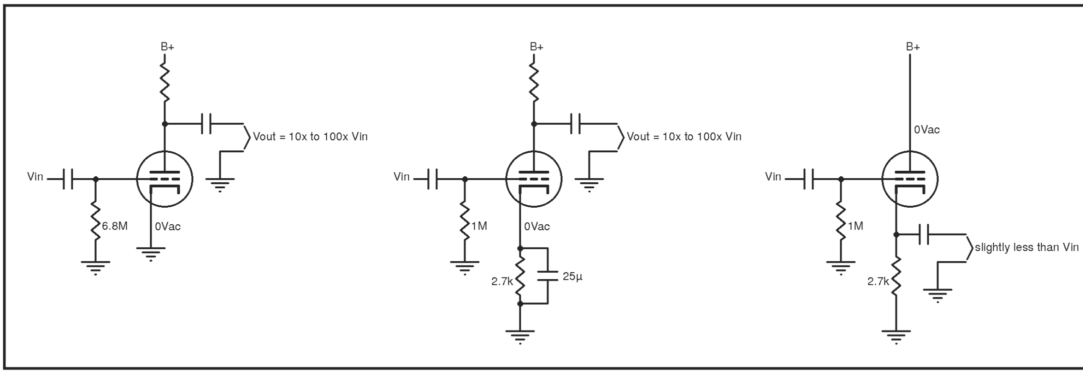


Figure 1: In signal tracing, you need to know the relative signal voltages to expect.

when you pull out a tube can damage tweeters or midrange speakers.

Signal Tracing

Signal tracing is more-or-less the inverse of signal injection. You apply a normal-level signal to the input and trace it through the circuit one stage at a time, using an oscilloscope or a voltmeter. An oscilloscope has the advantage of showing you the waveform as well as the signal voltage; whereas, the voltmeter only shows the signal voltage. To avoid loading the circuit under test by connecting too low an impedance, you should use a 10X probe on the oscilloscope or an electronic voltmeter. Either of these has an input impedance greater than or equal to 10 MΩ, which is high enough not to disrupt the operation of an amplifier circuit anywhere except in a grid-leak-biased control-grid circuit.

Standard practice in years past—when electronic voltmeters (i.e., Vacuum Tube Volt Meters or VTVMs) were more expensive—was to specify an analog voltmeter having a sensitivity greater than or equal to 20 kΩ/V. On a 500-V scale such a meter would have a 10-MΩ input resistance, but the input resistance would be lower on lower scales.

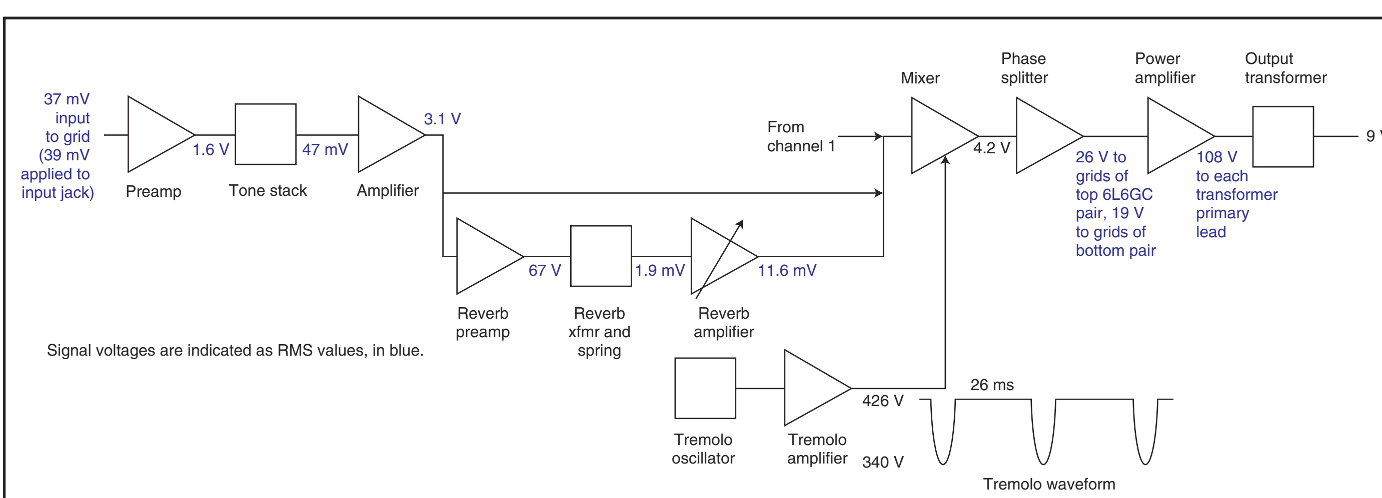
Signal tracing requires you to know what levels to expect at the tube terminals of common amplifier stages. **Figure 1** illustrates expected levels for common-cathode and cathode follower stages, which are the most common stages in audio amplifiers. Although triode stages are shown, the same expected signal levels apply to pentode stages, except that the signal at the plate may be many times larger than for a triode, due to the higher stage gain that can be obtained in a pentode stage. Screen grids are often—but not always— bypassed to ground with a capacitor, so in these cases there will be no signal voltage on the screen grid.

In some cases you will be fortunate enough to have a manufacturer's manual that specifies the signal voltages that you should expect. These may be indicated on the schematic itself, on a separate table, or as a signal-level graph appearing below the schematic. (The latter method is found in service literature for broadcast and recording equipment more often than for consumer hollow-state gear.)

Fender Twin Example

Figure 2 shows a block diagram of a Fender Twin reverb guitar amplifier's vibrato channel, with

Figure 2: This block diagram of a Fender Twin reverb guitar amplifier shows signal voltage levels.



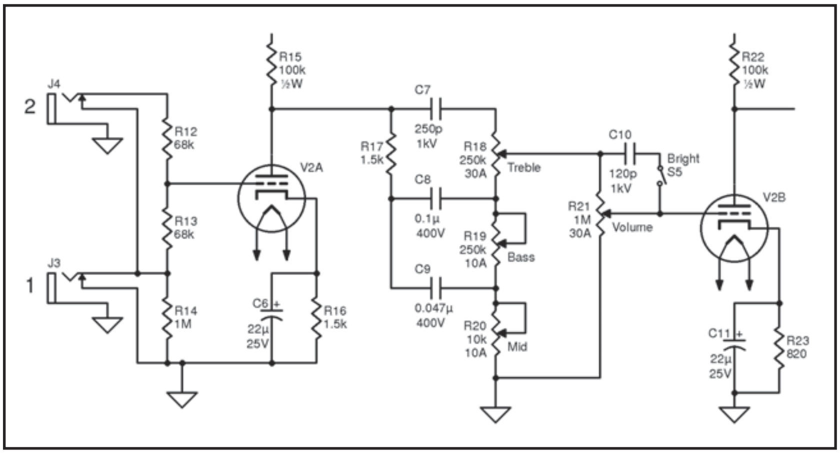


Figure 3: The first two stages of the Twin reverb amplifier's vibrato channel have this schematic.

signal levels for the input and output of each stage. These levels are specified based on the assumption of a 1-kHz input signal, and with the bright switch off, reverb and tremolo off, and volume and tone controls at the 50% position.

The input level and control settings specified in **Figure 2** are not high enough to drive the amp to full output: with these levels and settings, the output would be more like 10 W. Thus, your dummy load resistor does not need to be able to handle the amplifier's full output power for this kind of signal tracing. Incidentally, when you're testing, it's always a good idea to terminate a tube power amp with a dummy load equal to the normal speaker impedance (usually 8 Ω) to prevent damage to the output stage.

The vulnerability to such damage comes from the fact that the output stage's voltage gain depends upon the plate load impedance, which is the output transformer's primary impedance. If the output transformer is not loaded, its primary impedance will be quite high, resulting in a very high voltage gain. This, in turn, can lead to clipping, which can generate inductive spikes large enough to cause flashover, with consequent damage to the output tubes or output transformer.

You may notice that, with the controls at 50%, the gain of the first stage is only about twice enough to overcome the loss introduced by the tone stack. This factor varies from one amplifier to another. Home music amplifiers often use higher gains in the first stage, especially for the phono or tape head inputs, since the signal levels at those inputs are very low—much lower than the level of an electric guitar. With such low input levels, keeping the signal level reasonably high in the control stages helps to minimize noise in the amp's output.

Although the levels shown in **Figure 2** are typical of those found in many guitar amplifiers, they do not really match what you will find in integrated home music amplifiers. In general, with these amplifiers, the input level is either just a few millivolts (phono and tape head inputs) or roughly a volt (line-level inputs for tuners, etc.). If the amplifier contains line outputs or record outputs to feed tape decks, these also usually operate at line level. So you can find where these outputs come from in the circuit, and you'll know that the signal level from any input should be about a volt there.

So what do we know if the signal level at the control grid of a tube is normal, but the plate signal level is low but not distorted? Obviously, we know that the stage gain is low. Looking at the schematic of the Fender Twin reverb guitar amplifier's first two stages (shown in **Figure 3)**, let's see what it could tell us if the signal level at the plate of V2A is low. The signal level at the plate depends on the input signal level and the stage gain. The stage gain depends on the mu of the triode, the load appearing at the plate, and the presence or absence of negative feedback at the cathode. The mu depends upon the triode's condition, and slightly upon the control-grid bias.

Possible Failure Causes

Of the possibilities that emerge, the most likely is an open cathode-bypass capacitor, which would allow negative feedback to appear at the cathode. A defective tube is another possibility. A shorted cathode bypass capacitor would certainly change the control-grid bias, but would also cause a lot of clipping in the process. If the cathode resistor

Figure 4: This signal waveform at the plate can result from too low a control-grid voltage or too high a cathode voltage.

Figure 5: This signal waveform at the plate can result from too high a control-grid voltage or too low a cathode voltage.

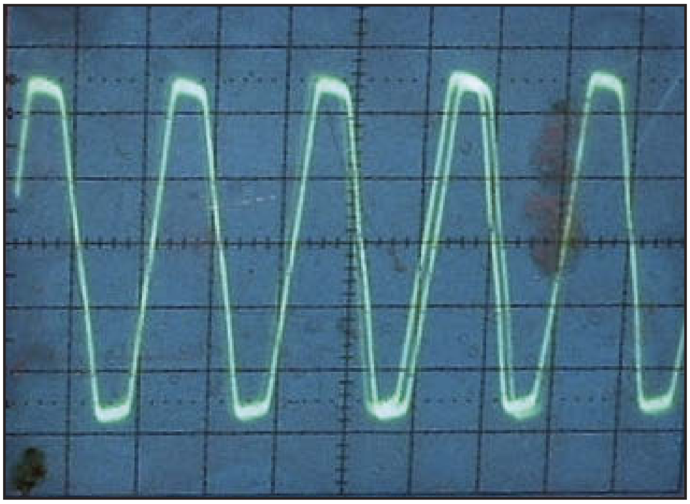


Figure 6: A pentode stage can produce symmetrical clipping.

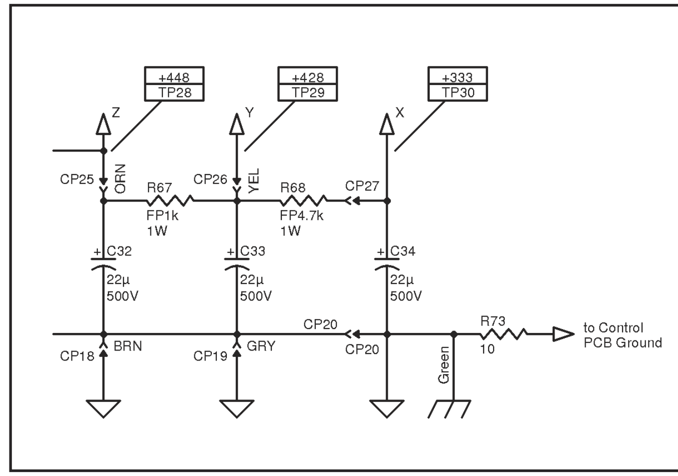


Figure 7: The B+ voltages in a Twin Reverb amplifier are derived by this circuit from the 440-V supply.

had increased in value—a possibility for carbon-composition resistors—again the bias would be changed, but the waveform at the plate would be more distorted. Other possible component failure modes are not as likely, so we should concentrate on these possibilities first.

In addition to using signal tracing to track the signal levels through an amplifier, we can also observe the waveforms appearing at the outputs of each stage. Control-grid bias problems will cause asymmetrical distortion (see **Figure 4** and **Figure 5**). The most common cause of control-grid bias problems is a leaky coupling capacitor from the preceding plate to the control grid. This will cause saturation on positive input peaks (negative output peaks), as shown in **Figure 5**.

Figure 6 shows symmetrical clipping occurring at a pentode stage's output. This clipping can result from overdriving the stage, but if you find it when you are signal-tracing an amplifier using the correct input signal voltage, the cause may be low B+. If a hollow-state rectifier is being used, a weak rectifier tube can cause low B+. The B+ feed to preamp, voltage-amp, and phase-splitter stages is usually derived from the high B+ that supplies the output stage.

Figure 7 shows the schematic for this part of a Twin Reverb power supply. If too much current is drawn from the 448-, 428-, or 333-V points, all the B+ voltages after the 440-V one will be low. A leaky decoupling capacitor (C32, C34, or C38) will also cause low B+ voltages, as will dropping resistors R67 or R68, if their value increases due to heat and age.

When applied along with a thorough understanding of the operating principles of hollow-state amplifiers, signal tracing (or signal injection, in some cases) can lead to a solution for many of the tougher repair issues.

COLUMNISTS

Vance Dickason has been working as a professional in the loudspeaker industry since 1974. He is the author of *Loudspeaker Design Cookbook*—which is now in its seventh edition and published in English, French, German, Dutch, Italian, Spanish, and Portuguese—and *The Loudspeaker Recipes*. Vance is the editor of *Voice Coil: The Periodical for the Loudspeaker Industry*, a monthly publication. Although he has been involved with publishing throughout his career, he still works as an engineering consultant for a number of loudspeaker manufacturers.

Dr. Richard Honeycutt fell in love with acoustics when his father brought home a copy of Leo Beranek's landmark text on the subject while Richard was in the ninth grade. Richard is a member of the North Carolina chapter of the Acoustical Society of America. Richard has his own business involving musical instruments and sound systems. He has been an active acoustics consultant since he received his PhD in electroacoustics from the Union Institute in 2004. Richard's work includes architectural acoustics, sound system design, and community noise analysis.

Mike Klasco is the president of Menlo Scientific, a consulting firm for the loudspeaker industry, located in Richmond, CA. He is the organizer of the Loudspeaker University seminars for speaker engineers. Mike specializes in materials and fabrication techniques to enhance speaker performance.

Steve Tatarunis has been active in the loudspeaker industry since the late 1970s. His areas of interest include product development and test engineering. He is currently a support engineer at Listen, in Boston, MA, where he provides front-line technical support to the SoundCheck test system's global user base.

Industry Calendar

Here are a few places where you might find a copy of *audioXpress* or possibly meet one of our authors and staff members:

October 2–4, 2015 — **Rocky Mountain Audio Fest (RMAF)**
Marriott Denver, Denver CO
www.audiofest.net
www.audiofest.info

This will be the 12th edition of the annual Rocky Mountain Audio Fest (RMAF), the largest high-end consumer audio show in the US, featuring approximately 200 exhibitors (more than 400 companies). From affordable audio systems to ultra-expensive high-end gear and CanJam at RMAF—a show within a show organized by Head-Fi.org—featuring the latest in headphone technology, this is a great show for product introductions.

October 4–6, 2015 — **PLASA Show London**
ExCel London, London, UK
www.plasashow.com

Owned and organized by the Professional Lighting and Sound Association (PLASA), the international trade association for entertainment technologies, PLASA Show—now 38 years old—connects the international live entertainment technology industry and ranks as one of the premier shows of its kind. It presents ground breaking technology and exciting new launches by the world's greatest designers and engineers to an international audience of pro audio, lighting, broadcast, AV and stage technology experts. PLASA has also been promoting focused hubs for networking and product demos on the show floor for audio. There are a series of audio specific sessions for the pro audio community covering live sound training and theory, manufacturer-led panel sessions discussing current technical issues and a look at sound design for global, large-scale events.

October 9–11, 2015 — **The Loudspeaker Sourcing Show**
Nansha Grand Hotel, Guangzhou, China
www.loudspeakersourcingshow.com

The first Loudspeaker Sourcing Show is a first of its kind—an all inclusive solution for loudspeaker brand buyers, engineers, and their decision makers to meet with qualified loudspeaker factories under one roof during this three-day event. The show is for all markets associated with loudspeakers including consumer, contractor AV, mobile, audiophile, musical instrument, and professional. Only actual manufacturers will be allowed to attend. Legal services, packaging companies, and translators will also be available. Anyone in a decision-making role for loudspeakers should attend the show. The show will end one day before the international Hong Kong Electronics Fair (HKES 2015—Fall Edition) trade show.

October 13–16, 2015 — **HKTDC Hong Kong Electronics Fair (Autumn Edition) and ElectronicAsia**
Hong Kong Convention and Exhibition Center
1 Expo Drive, Wan Chai, Hong Kong
www.hktdc.com/hkelectronicsfairae
electronicAsia: http://electronicasia.com

With 4,100 exhibitors and nearly 95,000 buyers from more than 150 countries and regions attending the HKTDC Hong Kong Electronics Fair (Autumn Edition) and the concurrent electronicAsia, the events form the world's largest electronics marketplace and continue to pack a serious punch in the technology world. In 2014, the two fairs attracted a total of around 95,000 buyers from more than 150 countries and regions.

October 14–17, 2015 — **CEDIA Expo 2015**
Kay Bailey Hutchison Convention Center, Dallas, TX
http://expo.cedia.net

This year's Custom Electronic Design & Installation Association (CEDIA) Expo event is being held from October 14–17, 2015, with the actual exhibit sessions taking place from October 15–17. CEDIA Expo has unquestionably become one of the most important shows for home theater and distributed audio loudspeaker manufacturers with an impressive list of loudspeaker manufacturers and other loudspeaker related exhibitors. As has been the tradition since it began, CEDIA University is providing a wide selection of workshops, classes, and seminars related to the audiovisual (AV) installation industry. 2015 CEDIA Expo includes special focus areas for security, two-channel audio, and high performance audio rooms.

October 14–17, 2015 — **Prolight+Sound Shanghai 2015**
Shanghai New International Expo Centre (SNIEC), Shanghai, China
International Exhibition for Event and Communication Technology, AV-Production and Entertainment.
www.prolightsound-shanghai.com

Since 2003, the Prolight+Sound Shanghai and Music China (not Musikmesse) has benefited from a strategic alliance between Messe Frankfurt, Intex Shanghai Co, Ltd., and the Shanghai Municipal Administration of Culture Radio Film & TV, together with many other Chinese associations. If you are interested in audio and the Chinese market, this is the show to attend. The Prolight+Sound side has more than 500 exhibitors, while the Music China show attracts 1,600 manufacturers and suppliers worldwide. 24,000 visitors attended last year's event.

Highlights for 2015 include the first New Advancement in 3D Sound System Seminar and 3D Sound System Showcase, organized in cooperation with the leading music engineering expert, Siwei Music Engineering Design Group, the International Audio Training Courses, and the Acoustics Technology Forum.

October 29–November 1, 2015 — **139th AES Convention/AES New York 2015**
Jacob Javits Center, New York, NY
www.aes.org/events/139

This year's Audio Engineering Society's 139th AES International Convention will include Special Events and Exhibition Floor presentations offered by the free Exhibits-Plus badge, to the four days of in-depth Tech Programs available to holders of the premium All Access badge, with a range topics and technologies for every aspect of the audio industry. Unique topics and programming for all AES attendees include the return of the Live Sound Expo and Project Studio Expo.