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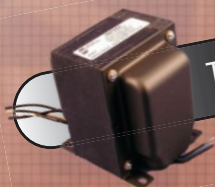


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Spatial Audio Reproduction



In just one week I have attended a sequence of three shows—CanJam SoCal in Los Angeles, CA; the NAB Show 2018, in Las Vegas, NV; and the AXPONA show in Chicago, IL—where, in a short period of time, I listened to some of the best sound reproduction systems that are currently available. And at all those shows—two consumer-focused, and one of the largest professional-oriented conventions in the world—I could always find some very interesting and notable common trends.

There is a clear momentum to explore different forms of sound reproduction in the consumer space, consistently gaining ground in the live entertainment sector, and also seriously discussed for broadcast and digital media distribution at NAB 2018.

Probably, that trend was mostly felt at the show I did not attend that week, which was the Prolight+Sound 2018 show in Frankfurt, Germany, happening concurrently. In Frankfurt, L-Acoustics and d&b audiotechnik made an impact with their respective approaches to spatial audio reproduction, the L-ISA (www.l-isa-immersive.com) and Soundscape solutions (www.dbsoundscape.com). Astro Spatial Audio (ASA) also created an impression with its object-based immersive audio demonstrations, in partnership with TTA Sound (www.tta-sound.com), Alcons Audio and others, exploring the potential of object-based immersive audio for live entertainment and installation.

These companies are exploring state-of-the-art digital processing technologies, object-based mixing, and even room emulation, to create a new approach to live sound mixing, free from the restrictions of conventional stereo imaging. Moving away from the traditional L/R and L/C/R public address systems and adopting multi-array configurations and spatial processing, to generate a three-dimensional audio experience to every member of the audience.

Although not present at Prolight+Sound, Sennheiser was very much present at NAB 2018, where it continues to expand its AMBEO Spatial Audio Partnership with a growing number of manufacturers, and also with strategic acquisitions in leading software and processing solution companies. Understanding that everything should start at the source, from binaural recording to Ambisonics, Sennheiser is investing in a new generation of spatial audio recording tools. And while the company's vision of immersive experiences and 3D sound is very much linked with the promise of Virtual and Augmented Reality platforms—which continue to excite a growing number of companies, as we witnessed at NAB 2018—it is also directly linked with Sennheiser's perception that the market is evolving toward different sound reproduction approaches.

Also, in Las Vegas, the Fraunhofer Institute for Integrated Circuits IIS was demonstrating the MPEG-H (www.mpeg-h.com) 3D audio standard—now part of the new broadcast and media distribution standards officially adopted in the US, Europe, Korea, and China. The 3D audio demos included Fraunhofer's 3D soundbar prototype (www.iis.fraunhofer.de/tv-audio), but there was also a pre-production prototype of an MPEG-H enabled Sennheiser soundbar, as well as MPEG-H enabled TV sets from Samsung and LG that are already in use in South Korea. The MPEG-H TV Audio System is designed to work with today's broadcast and streaming equipment and offers a complete object-based immersive audio solution with interactive features that enable viewers to tailor the playback to sound best on a different range of devices and environments.

At NAB 2018, I also had the chance to meet the creative team at Ambidio (www.ambidio.co), an innovative sound company that has created a new way to generate a spatial sense of depth and dimension through any device with stereo speakers. Ambidio has been working with Skywalker Sound to enhance the way people watch content on laptops, tablets, and smartphones with built-in stereo speakers. And the interesting thing about their approach is that no decoder is necessary on the receiving end, since all spatial cues—generated by object-based audio—are already embedded into the audio files.

These new sound reproduction techniques enable us to fully emulate multichannel and object oriented immersive formats such as MPEG-H, creating engaging experiences in convenient speaker formats or headphones.

As Richard Honeycutt writes in this edition, we still need to master the optimum stereo recording and reproduction techniques, and part of the reason is that it seems we were never totally happy with it. While stereo is undoubtedly the format we are more familiar with, it seems the 21st century is paving the way for a transition from channels- to object-based audio sources and from two-channel to spatial audio reproduction.

J. Martins
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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.

Ron Tipton has degrees in electrical engineering from New Mexico State University and is retired from an engineering position at White Sands Missile Range. In 1957, he started Testronic Development Laboratory, which became TDL Technology, to develop audio electronics. All product sales and services were terminated on December 31, 2015, but the TDL website is still online with a variety of audio information and downloads.

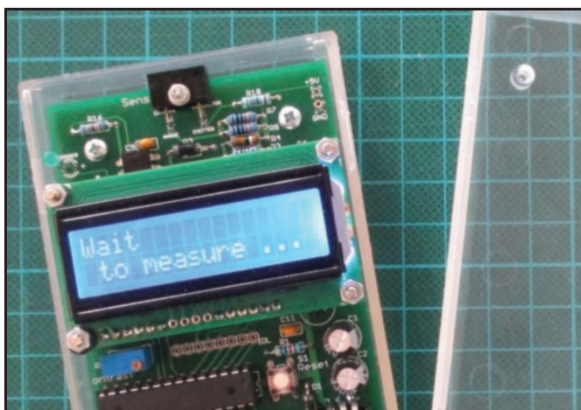
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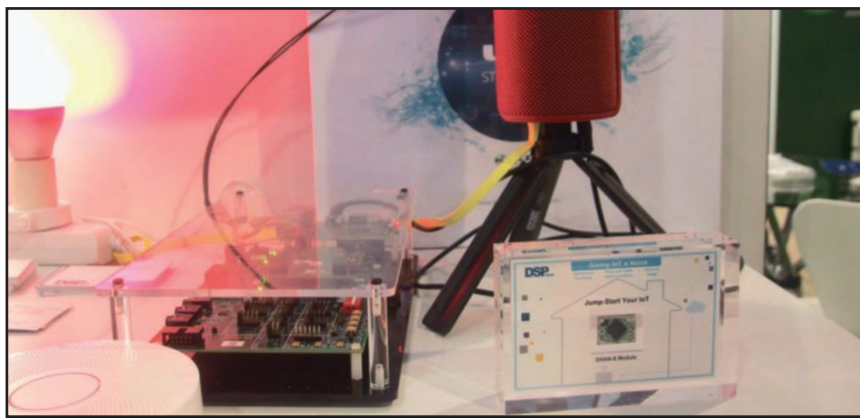
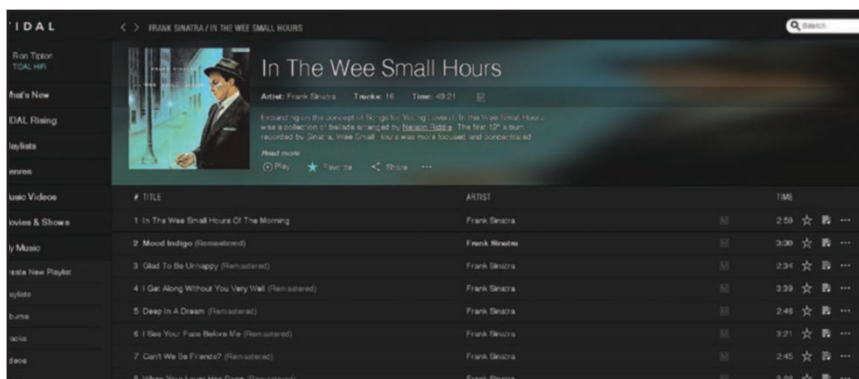
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Exploring Master Quality Authenticated (Part 2)

Measuring Streamed MQA

In this article, I'll continue my exploration of Master Quality Authenticated (MQA) music by looking at a file streamed from TIDAL and at an MQA-encoded CD. I will continue to use the same equipment that I used in the first part of this article series. This includes the Meridian Explorer2 and TEAC UD-301 DACs, the stereo to mono converter, the band-pass filter, the power amplifier, the tweeters, and the M500 ultrasonic microphone.

By
Ron Tipton

(United States)

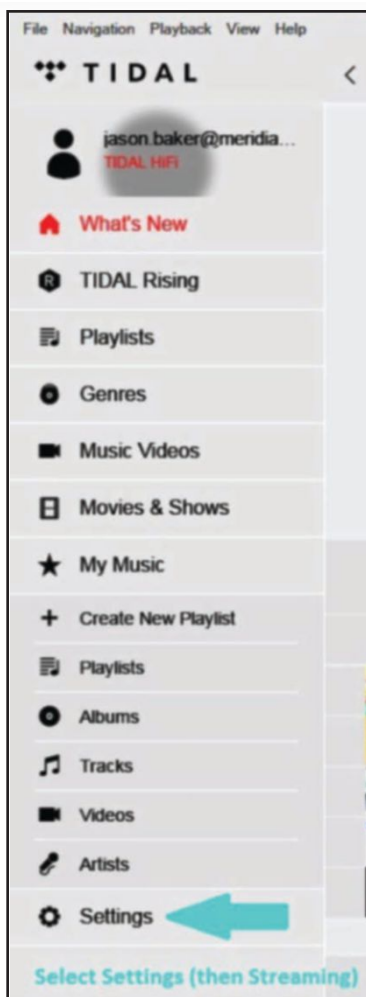


Photo 1: To begin the TIDAL setup for streaming MQA, click Settings (red arrow) in the left-panel of the "Home" screen.

To begin the measurements, you will need a HiFi account and you must install the computer desktop app to make this work. Also, TIDAL has to be correctly set up to stream MQA. In the Windows Control Panel, click on Sounds and make sure that in the Playback Tab either your MQA or non-MQA DAC is the default speaker. In my case, it's the Meridian Explorer2 or the TEAC UD-301. Next, click the default Speakers to highlight it and click Preferences. In the Levels tab, set it to 100%. In the Enhancement tab, all the boxes should be not checked and, last, in the Advanced tab, it doesn't matter what's in the Default Format but both Exclusive Mode boxes should be checked.

Launch TIDAL from the desktop app. On the left-side of the opening screen, click on Settings as shown in **Photo 1**. As shown in **Photo 2**, make sure HiFi/Master and either your MQA or non-MQA DAC is selected, then click the setting icon (blue arrow). Click the two boxes indicated by the blue arrows as shown in **Photo 3**, and you are done. I expect the Mac setup is similar.

When you switch between your MQA and non-MQA DAC, you may have to do this setup for your currently selected DAC. In my Windows 7, I must redo the setup each time, but Windows 10 remembers and does the setup for me. Also, remember that an ASIO driver is needed for your MQA DAC for bit-perfect playback and having the white LED turn blue in the Meridian Explorer2.

Finding MQA music files is easy. In the "Home" Tidal window, click on What's New in the left-side panel. Scroll down to Albums or Play Lists and click on the word MASTERS in the header bar. These are your MQA files and will also be identified with an M in a square box under the Album or Play List icon.

I chose the Frank Sinatra *In the Wee Small Hours* album and the "Mood Indigo" track for my measurement source (see

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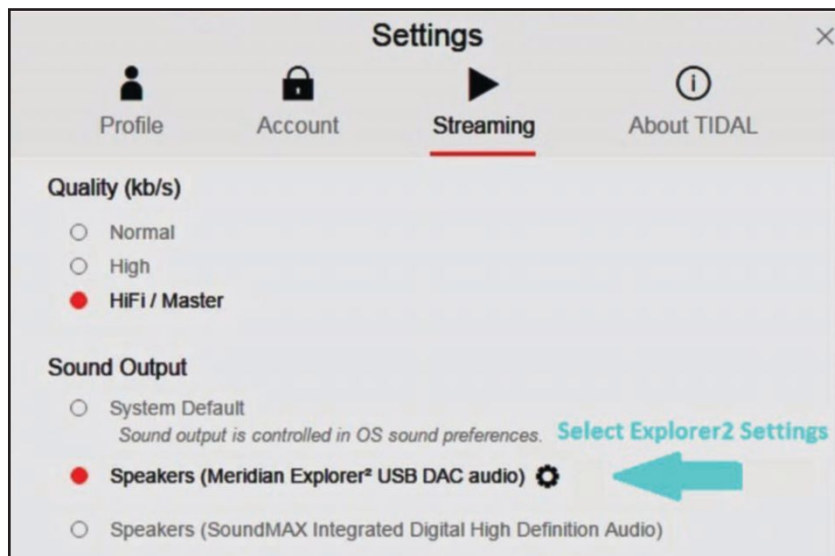


Photo 2: In the TIDAL Settings menu, click HiFi /Master and Speakers (your MQA DAC)—in my setup, this is the Meridian Explorer2. Click on the gear icon (blue arrow).

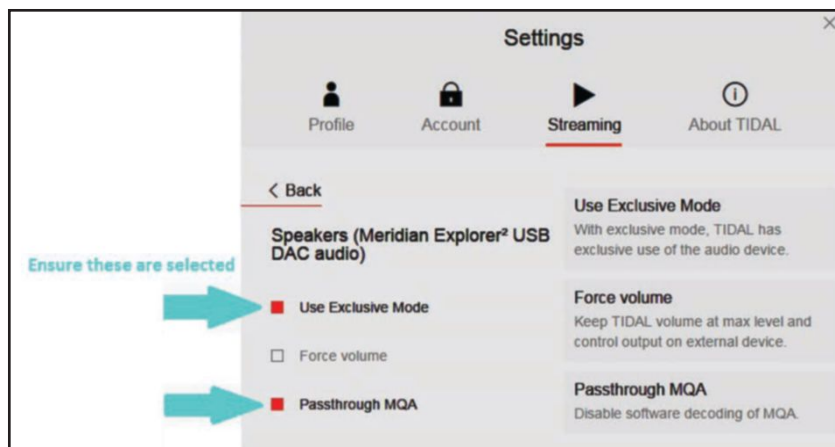


Photo 3: To finish the TIDAL setup, click the boxes marked with the blue arrows.

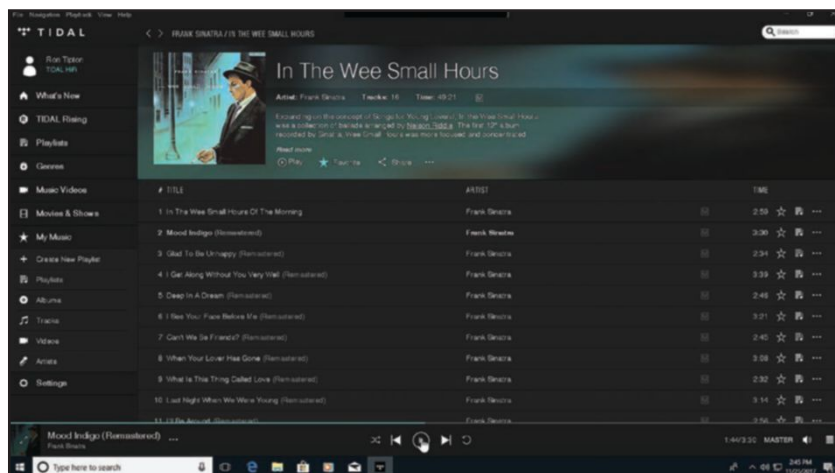


Photo 4: Frank Sinatra's *In the Wee Small Hours* album playlist. Released in 1955, this was his first 12" album with 16 ballads arranged by Nelson Riddle, an award-winning arranger of that era.

Photo 4). I played this file through my Meridian Explorer2 and the TEAC UD-301 DAC to another computer running the TrueRTA spectrum analyzer set to 1/24-octave and a 1000 Hz to 50 kHz display and exported the spectral data to text files for plotting. The results are graphed in **Figure 1** with the green line from the MQA decoding. The two DAC spectrum amplitudes were aligned by matching their peak amplitudes (as best I could) in the 1-kHz-to-8-kHz region. There is definitely more MQA energy in the 22-to-50-kHz region.

Ever mindful of possible clipping in the audio equipment processing chain, I repeated this measurement with a gain increase of 40 dB and got the same results. It turns out that an oscilloscope, in this case, is not a good tool for detecting clipping. The mix of frequencies, all at fairly low amplitude, has nothing on which the oscilloscope can synchronize.

Spectrograms and WAV Files

Using the setup shown in **Figure 2**, I once again found I achieved the best results with the L010 piezo electric tweeter—its frequency response goes to 60 kHz as compared to only 40 kHz for the other two. The BatSound Touch software displays an oscillogram and spectrogram (see **Photo 5**) and can also record the M500 microphone's output as a BatSound .wav file, 16-bits at a 500 kHz sample rate. **Photo 6** shows a screen capture of a portion of the two DAC spectrograms: from the Meridian Explorer2 (a) and from the TEAC (b). As expected, the TEAC's spectrum is very near system noise level.

The Touch program has the ability to play the captured 500 kHz sample rate .wav file in the audible range by down sampling. The audible sound was captured by another computer and recorded as a 24-bit, 96 kHz .wav file. A 30-second segment is included in the Supplementary Material as Mood-Indigo. wav. Remember that the actual frequencies are about 10 times higher. It sounds very "noise-like"—in fact, it sounds like chirping crickets. Just how it enhances the music we hear is probably another example of psychoacoustics in action. That is, our hearing adds the ultrasonic cues to the audible music. To get the full MQA benefit, you need a sound system able to reproduce frequencies higher than 22 kHz.

An MQA CD

As yet, there aren't too many MQA CDs, but I needed only one for my measurements. I found Camille Thurman's *Inside the Moment* and asked the publisher, Chesky Records, about it. It seems the download version is not MQA-enabled but the CD is—with the MQA logo on the disc and album cover (see **Photo 7**).

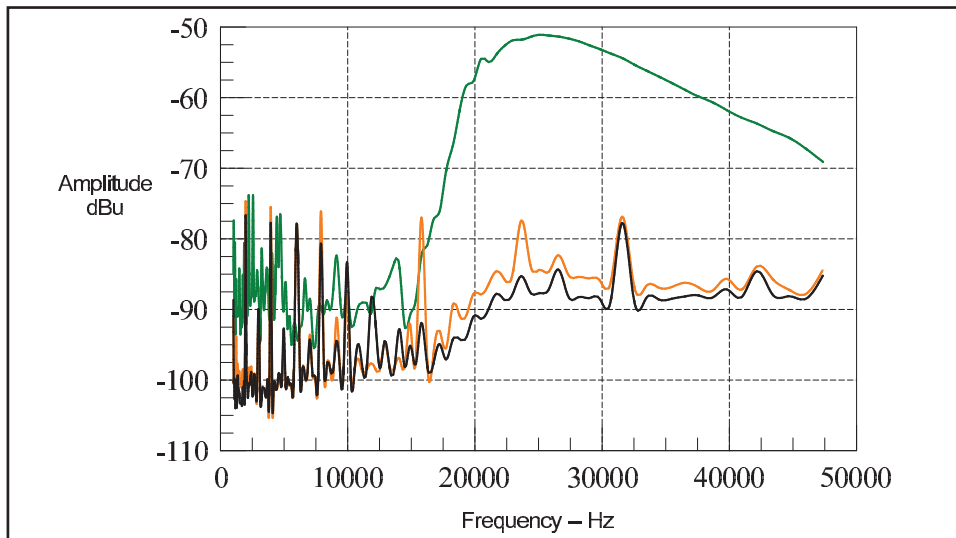


Figure 1: The MQA-encoded “Mood Indigo” track was played through the playback part of the setup (using each DAC in turn) with the output of the variable gain amplifier going to the second computer running TrueRTA to capture the 1/24-octave spectrum. Each display was captured to a text file for plotting. The green line is the MQA-decode, the orange line is from the TEAC DAC and the black line is the system noise. I used a linear frequency scale to expand the 20-to-48-kHz region.

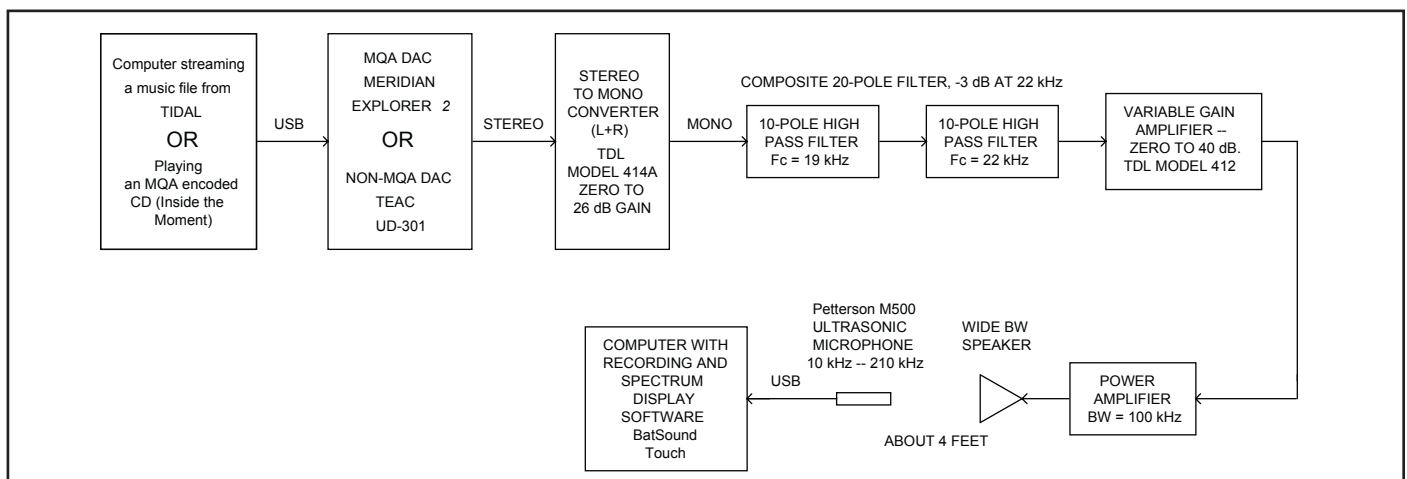


Figure 2: This is my measurement setup block diagram. The MQA-encoded track can be played with TIDAL or foobar2000 running on Windows 10. Both the Explorer2 and UD-301 DACs have a USB input. A Windows 7 computer was running the BatSound Touch program.

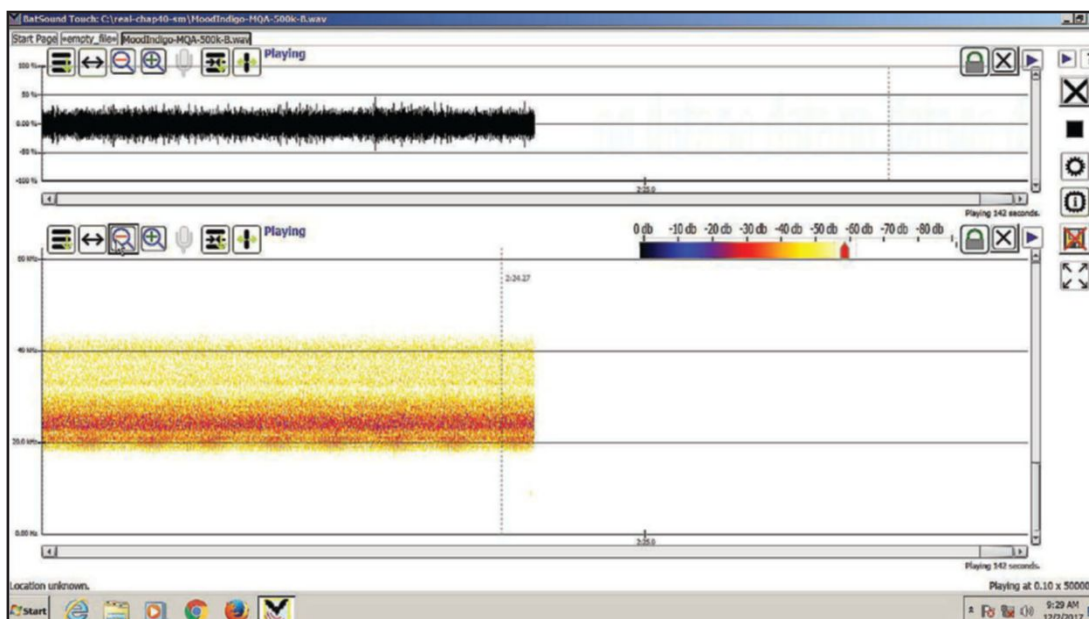


Photo 5: The BatSound Touch main screen has the oscillogram at the top with the spectrogram below. Its many settings and controls make it versatile but a bit difficult to learn to use efficiently. A copy of the User Manual is included in the Supplementary Material.

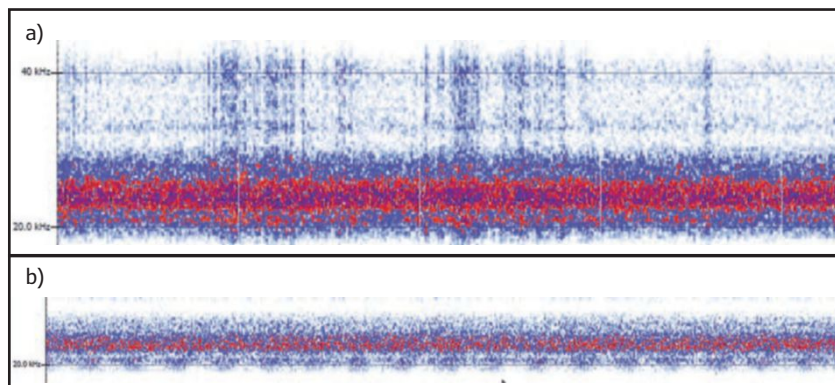


Photo 6: This is a screen capture of a portion of the spectrogram of the MQA-decoded “Mood Indigo” track (a). This is a capture of a portion of the TEAC non-MQA spectrogram showing primarily noise (b). The red spots are the loudest, followed by dark blue and then light blue.

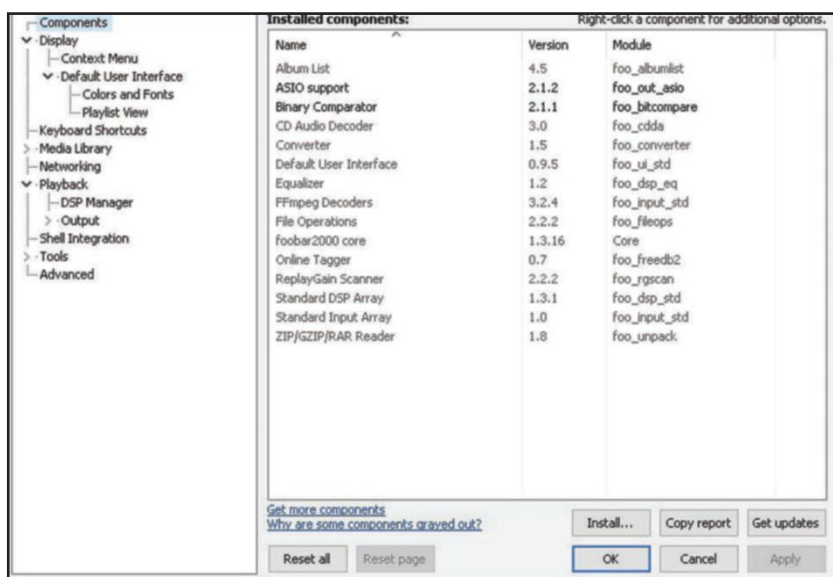


Photo 8: The foobar2000 Preferences menu must have ASIO Support in the Installed Components list to play bit-perfect MQA.

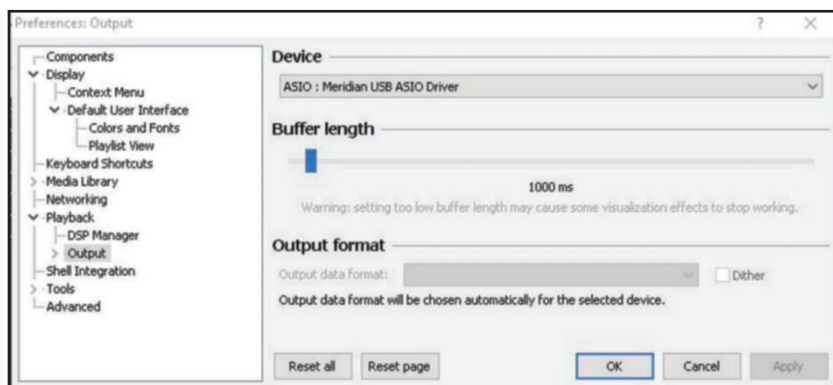


Photo 9: The Output screen (under Playback) must show an ASIO driver for your MQA or non-MQA DAC in the Device box. A Buffer length of 1000 ms is usually a good choice (it worked for me). Click Apply and then OK to complete the setup.



Photo 7: The *Inside the Moment* album cover has a prominent MQA logo. The quartet features Camille Thurman on tenor saxophone and vocals, with Ben Allison on bass, Billy Drummond on drums, and Mark Whitfield on guitar. It was produced by Chesky Records, Inc. and released in 2017.

Of course, I would need an MQA-enabled CD player to go that route but I could play it from my computer’s CD/DVD drive with an appropriate player. The free foobar2000 media player fills the bill, with the proper setup.

The first step is downloading foobar2000 if you don’t already have it and then getting the `foo_out_asio.fb2k`-component from the foobar2000 website and installing it. Start foobar2000 and in the File tab click Preferences. Click Components. Click the Install bar and find the ASIO component, or just drag and drop it on the screen. Press OK and you will be prompted to restart foobar2000 to load it. Your screen should look similar to the one shown in **Photo 8**, with ASIO Support listed under Installed Components.

Click Playback and set ReplayGain to none in both boxes. (ReplayGain is a metadata value which can be used by some media players to set the playback loudness.) Click Output (under Playback) and make sure the ASIO driver for your current DAC is in the Device box—1000 ms is usually a good setting for Buffer Length (see **Photo 9**).

Measurements

I played the album’s first track: “The Night has a Thousand Eyes” through each DAC in turn to the second computer running TrueRTA and captured the 1/24-octave spectrums to text files for plotting

(see **Figure 3**). Again, the amplitudes were aligned by matching the 1-kHz-to-8-kHz peaks. Clearly, the MQA decoding has greater ultrasonic energy with the TEAC decoding near the noise level.

I used the Figure 2 setup to measure the spectrograms and capture an MQA-decoded batsound.wav file, 16-bits at 500 kHz sample rate. The MQA and non-MQA spectrograms are shown in **Photo 10**.

The Batsound file was played using the down-sampling feature in the Batsound Touch program and recorded on another computer as a 24-bit, 96 kHz .wav file. A segment of it is included in the Supplementary Material as The-Night-has-a-Thousand-Eyes.wav.

An Experiment

I wondered if it would be possible to hear any MQA enhancement using my existing speakers, all of them limited to an upper, reasonably flat limit of 20 to 22 kHz, by augmenting them with the piezo tweeter. The tweeter would be reproducing a mono file, the left and right addition, but I thought it worth a try, although I wasn't expecting too much.

I moved the tweeter board (shown in Photo 2 in the first part of this article, *audioXpress*, May 2018) to a shelf between the main left and right speakers (shown in Photo 5 in the first part of this article). Then, using the "playback" part of Figure 2, I played the same tracks used for the measurements. An on-off comparison was easy, just turn off the tweeter feed. What did I hear?

Wow! Even with this rather crude addition the music sounded "fuller." Well, maybe "richer" would be a better word—certainly an improvement.

Final Thought

I believe I need to build another pair of speakers that incorporate the piezo tweeters to be able to enjoy high-resolution music. 

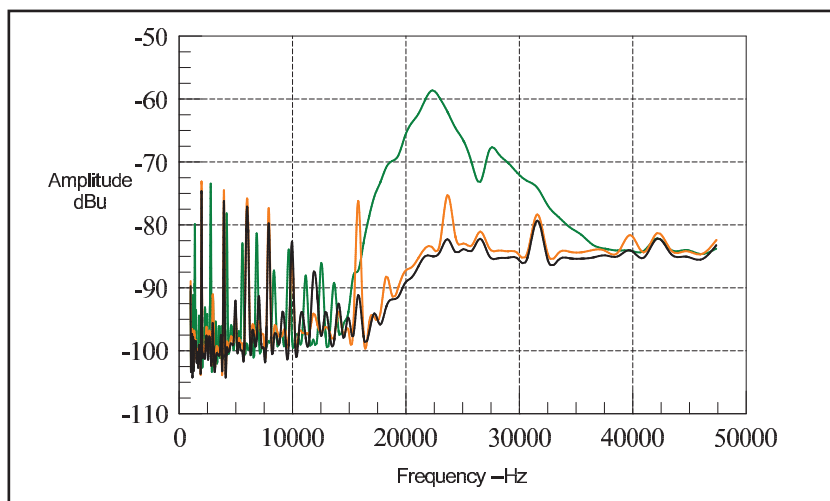


Figure 3: The MQA-encoded "The Night has a Thousand Eyes" track was played through the playback part of Figure 2 (using each DAC in turn) with the output of the variable gain amplifier going to the second computer running TrueRTA to capture the 1/24-octave spectrum. Each display was captured to a text file for plotting. The green line is the MQA-decode, the orange line is from the TEAC DAC and the black line is the system noise. I used a linear frequency scale to expand the 20-to-48-kHz region.

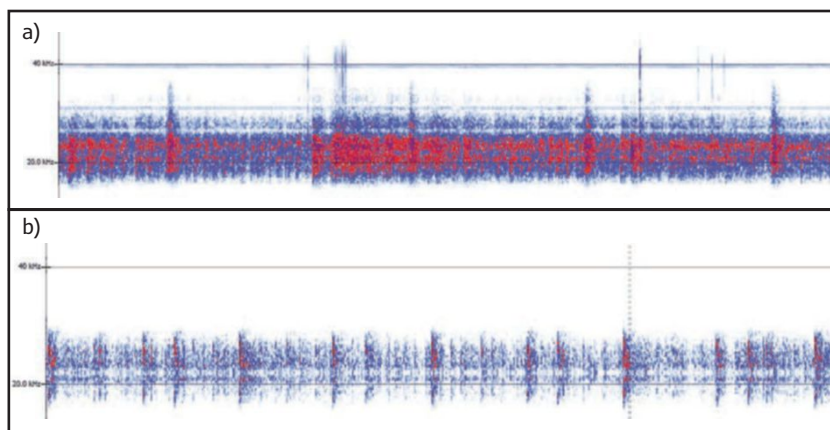


Photo 10: Similar to Photo 6, these screen captures for "The Night has a Thousand Eyes" track has the MQA-decode (a) with the non-MQA decode (b). The red spots are loudest, followed by dark blue and then light blue for the quietest.

Project Files

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

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Sources

BCT L010 tweeter

Kemo Electronic | www.kemo-electronic.de/en/Car/Speaker/L010-Piezo-Loudspeaker

Explorer2 DAC

Meridian Audio | www.meridian-audio.com/en/products/dacs/usb-dacs/explorer

Batsound Touch and M500 mic

Pettersson Elektronik | www.batsound.com

TrueRTA

True Audio | www.trueaudio.com

Back EMF Phase Relationships in Moving-Coil Loudspeakers (Part 2)

Thought Experiments

We have noted that, in a loudspeaker, there are three modes of Back EMF generation—below, at, and above resonance. These three modes result in Back EMF which is inductive, resistive, or capacitive, respectively. We explained how to identify Back EMF in each of these modes, and how to determine its magnitude and phase. Now, we will perform simple “thought experiments,” in order to make qualitative observations about the activity of a moving-coil driver in the region of its resonance.



By
Andy Lewis
(Acme Sound, LLC)

As was observed in the moving-coil loudspeaker, there are three modes of voltage generation, occurring below, above, and at the frequency of resonance. Back EMF, and resulting impedance, can be inductive, capacitive, or purely resistive. Consequently, it will take at least three separate examinations of this AC motor to describe what’s happening.

We will start at very low frequencies, where the delay in time between input current and cone position is easy to visualize. We use the word “slip” to refer to this delay in time. This concept will be explained in greater detail both in this article and in the third part of this article series.

We will proceed to higher frequencies after that. An examination of the generation of EMF at very low, DC and near-DC frequencies can provide a means to an understanding of what’s happening as frequency increases.

Each illustration shows a phase delay of a specific angle. Phase delay, for each illustration, is simply described in terms of phase angle: 5°, 10°, and 14.4° of phase delay in these first illustrations, and by greater angles of delay as we proceed.

It is not necessary to reference a particular driver

or frequency for this exercise, because this is only about qualitative behavior. For a slip of 5°, for example, each driver will have its own frequency. These experiments will be about what happens when a driver is driven at its own frequency for a specified angle of slip. When actual frequencies are specified in these illustrations, they pertain to the Fisher driver we have been using.

A Loudspeaker Exposed to DC

To begin, we will start with a sustained DC voltage. This would be considered an extremely low frequency indeed, and will provide a starting place; a frequency at which angle of slip is zero. We will then make observations at increasing AC frequencies, and examine the behavior of slip, as it affects the creation of Back EMF.

(Others have used DC spikes or pulses as tools for testing moving-coil loudspeakers^[1] and have drawn certain inferences from their observations. We will not use DC in that way. We will view a sustained DC voltage as an infinitely low AC frequency, just as a place to start.)

So, let us consider a woofer being exposed to DC voltage. The cone will accelerate at the moment the

voltage is applied, and the position of the cone will, for a short time, lag behind the resulting DC current (and force), as its mass is accelerated. As it changes position, it comes to its point of equilibrium for a given current, and its position remains constant—meaning velocity is zero. This point of equilibrium is the position at which the motor force on the voice coil is equal to the restoring spring force of the woofer’s suspension.

Current continues to flow at this point of equilibrium, even though the cone is not moving. Because cone velocity at this point of equilibrium is zero, there is no reverse voltage being generated. A motor force, as a result of this DC current still exists, but because it is opposed by an equal but opposite spring force, due to the speaker’s compliance, velocity is zero.

When exposed to DC, the unit’s impedance can be calculated, just as with AC. The impedance is equal to DC resistance plus Back EMF. Since velocity is zero, Back EMF is also zero. Therefore, the impedance at this (zero) frequency is equal to the DC resistance of the voice coil.

Figure 1 shows the lack of separation in time between input current and cone position. The two are precisely in phase with one another, and at every DC current magnitude, the system is in equilibrium, velocity is zero, and Back EMF is zero.

Yes, Figure 1 looks like a sine curve. Trust me. This is in order to provide continuity to further illustrations, in which we will observe the two curves becoming separated in time. This is an image of current vs. position at DC currents. Ignore the shape of the curve. If it’s difficult to make that leap, consider this to be a sine wave of 0.00001 Hz. In the real world, as will be seen, the frequency need not be that low to produce nearly the same result. But at DC, the input voltage, current, and cone position will be precisely in phase with one another.

At a frequency of 0 Hz (DC), the impedance is equal to the DC resistance of the voice coil, by definition. We will see how and why the impedance becomes quite inductive even at low AC frequencies.

Interestingly, with DC, the mass of the cone is irrelevant. When a DC voltage is applied, a large mass will accelerate more slowly, and come to rest at the point of equilibrium a brief time after an identical unit with less moving mass. But the position of equilibrium will be the same. This is a compliance-controlled situation.

At frequencies below resonance, a woofer’s motion is compliance-controlled, meaning that the inertia of the cone is of less importance relative to the effect of the suspension on the motion of the cone. This makes sense—any heavy object is easier to move if you don’t have to move it too quickly. As frequency is increased, the mass becomes more difficult to accelerate. At the

point of resonance, the effects of the mass and the compliance are precisely equal. Above resonance, the difficulty of accelerating the mass overwhelms the influence of the suspension, and the speaker operates in a mass-controlled mode.

Slip and Back EMF at Low AC Frequencies

At very low frequencies, the input voltage and the cone position are nearly in phase with one another. Even so, the minimal Back EMF generated at low frequencies is still enough to result in a substantial inductive component of its impedance. The reason for this will be explained in a subsequent section.

The next four illustrations will show a loudspeaker exposed to sine waves of increasing frequency, below and approaching resonance. The actual frequencies aren’t important, only that each is below the unit’s resonance.

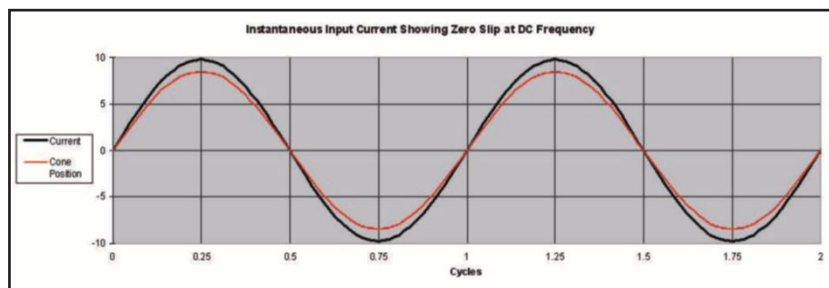


Figure 1: The cone position and the input current are in phase at all DC voltages.

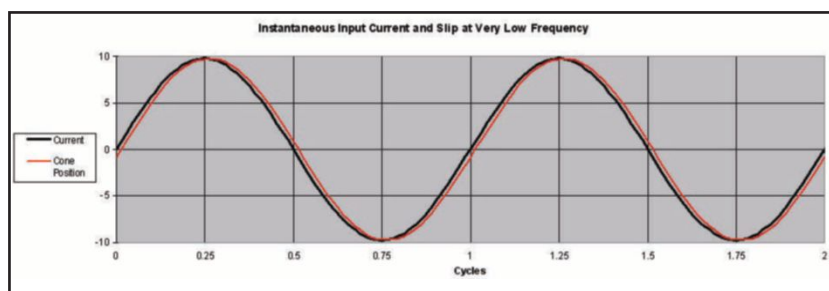


Figure 2: A 5° phase delay is shown between the input current and the cone position at a very low frequency.

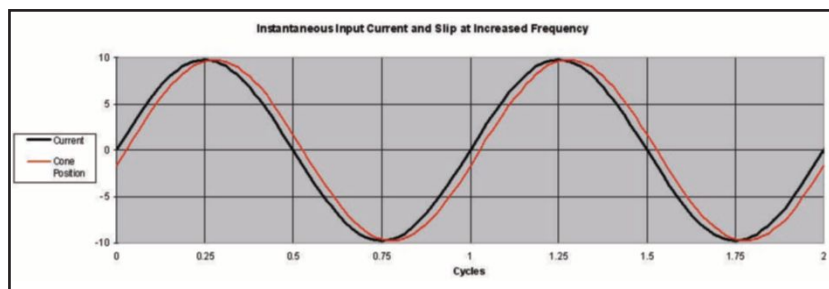


Figure 3: A 10° phase relationship is shown between the input current and the cone position at a slightly higher frequency.

We don't need to specify a specific frequency, or a specific driver. These illustrations are a qualitative representation of the delay in time, or phase angle, between the input current and the resulting position of the cone. As was mentioned, each driver will have its own frequencies at which it exhibits any of these specific angles of slip.

With pure DC, input current and position are in phase with one another, and slip is zero. But as frequency is increased, cone position lags behind input current due to inertia, as slip increases, in turn. Therefore, the two curves begin to exhibit a separation in time.

We will gradually increase the frequency from zero (DC), and examine the position of the cone, relative to input current, below resonance. Even at extremely low frequencies, it's easy to observe the slip graphically.

Which low frequency should we illustrate? Let us arbitrarily use the frequency at which the angle of slip is 5° as shown in **Figure 2**.

The third part of this article series will show how to calculate the angle of slip for a given Frequency. For now, we will graphically illustrate the behavior.

Referring to the Fisher woofer we discussed in the first part of the article series (*audioXpress*, May 2018), it can be calculated that the frequency at which the

phase delay between the input current and the cone position is 5°, is roughly 12 Hz. Again, that's specific to this woofer. Each driver will have its own frequency of 5° slip.

As we increase the frequency slightly more, the phase delay between the input current and the slip increases to 10° (see **Figure 3**). This 10° slip corresponds to an input frequency of roughly 19 Hz using our Fisher woofer. As was said, each moving-coil driver will have its own frequency for a specific amount of slip.

As frequency is increased from near-DC, slip increases, in turn, because the moving mass and its acceleration remain constant as the period decreases. And therefore, the moving mass lags further behind the input current, in Newtonian fashion:

$$F = ma$$

As frequency increases, so does angle of slip. It's intuitively obvious, when viewed this way, and it's important to gaining an understanding of the bigger picture.

Slip At Test Frequency f_1

Figure 2 and Figure 3 illustrate the trend as frequency is increased. We will now increase the amount of slip slightly further, to 14.4°. **Figure 4** shows slip increasing. Cone position falls farther behind input current as frequency is increased. The trend continues as we approach resonance.

Why 14.4°? It might seem arbitrary, but it's not. This is about our Fisher woofer. For this illustration, we specify a frequency of 23.0 Hz. This is the frequency " f_1 " at which we calculated the magnitude and phase angle of inductive Back EMF in the section on the AC Linear Motor (Part 1 of the article series). You can see the greater separation in time between the input current and the cone position (slip), compared to previous illustrations.

Approaching Resonance

Referring to the impedance curve (see **Figure 5**), we can see that as frequency increases below resonance, the impedance rises as well, as with any moving-coil driver. At f_1 , 23 Hz, we can see that the impedance curve is still relatively flat. As the frequency increases, phase of the impedance, and the Back EMF, both approach maximum inductive reactance, associated with the maximum phase angle of impedance, slightly above this frequency of f_1 . At resonance, however, less than an octave above this frequency, the impedance increases by a factor of almost four! Figure 5 shows a rapid rate of change of both magnitude and direction of impedance.

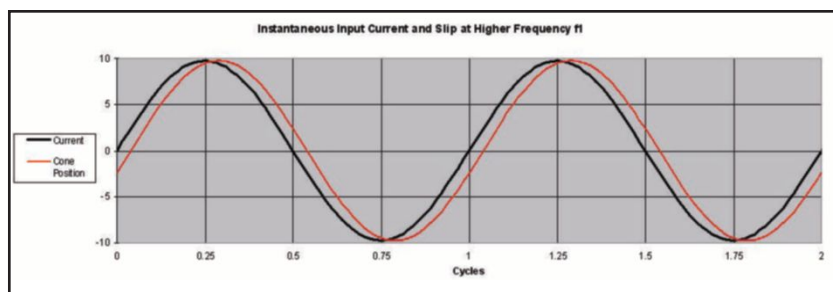


Figure 4: A 14.4° phase relationship is shown between the input current and the cone position at a higher frequency f_1 .

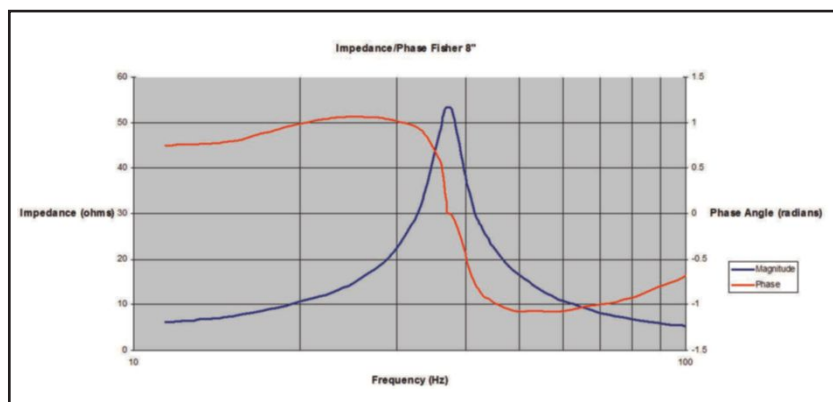


Figure 5: The impedance curve of the Fisher Woofer (detailed in Part 1 of this article series) shows that as frequency increases below resonance, the impedance rises as well, as with any moving-coil driver.

Above 30 Hz (see Figure 5), we can see a steep slope. This would suggest a similar rapid increase in slip, just as we see rapid changes in both magnitude and phase of impedance. **Figure 6** bears this out. The correlation is striking. Slip does increase rapidly as the frequency approaches the driver's resonant frequency.

Above f_1 , approaching resonance, the mass of the moving assembly becomes more of a "drag," as we approach the transition to its mass-controlled mode. Phase angle of slip increases rapidly, as do the magnitudes of both Back EMF and the impedance.

Slip and Phase Delay at Resonance

The previous figures show a rapid increase both in slip and impedance as we near the frequency of resonance. They also show that at the pivotal frequency of resonance, 36.93 Hz, two thresholds are realized: The first is that the impedance has a phase angle of zero. This indicates a purely resistive impedance, which means a purely resistive Back EMF, as calculated in Part 1 of this article series. The second is that the phase angle of the slip is exactly 90° (1.57 radians) behind input current. **Figure 7** illustrates the slip of 90° between input current and cone position at resonant frequency f_s graphically.

Slip and Phase Delay Above Resonance

As the input frequency is increased above resonance, slip increases, just as it did below resonance. Therefore, the phase delay between input current and cone position also increases. **Figure 8** shows an increasing angle of slip as frequency is increased above resonance. The selection of the frequency, 45 Hz, is arbitrary.

Slip and Phase Delay at Test Frequency f_2

At test frequency f_2 , 59.3 Hz, slip is equal to 166° behind input current. **Figure 9** illustrates this delay. Just as impedance and Back EMF are very inductive at f_1 , they are similarly capacitive at f_2 , in typical resonant-circuit fashion. Figure 9 shows the Fisher woofer we've been using, at a frequency of f_2 , 59.3 Hz, showing a phase delay between input current and cone position of -166° .

In the final article in this series, I will delve deeper into the concept of slip, in order to explain how and why this slip causes the impedance to be inductive, resistive, or capacitive, depending on frequency. 

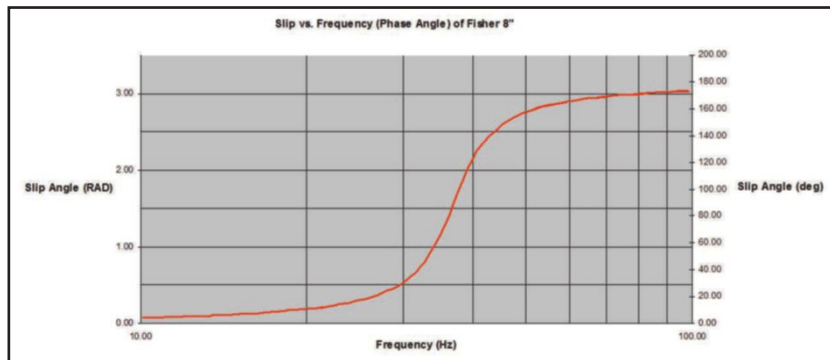


Figure 6: The slip delay is shown between the input current and the cone position with respect to frequency.

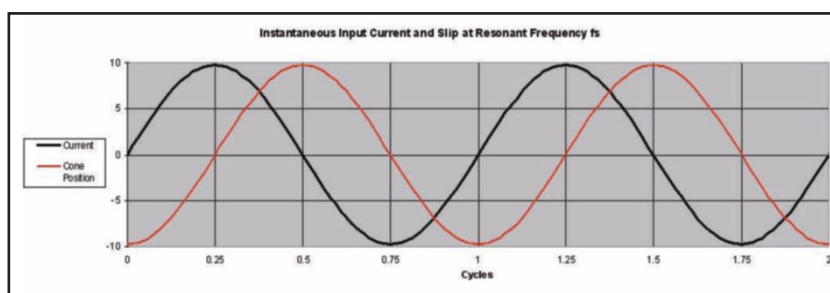


Figure 7: This figure illustrates the slip of 90° (0.25 cycles) between input current and cone position at resonant frequency f_s , 36.93 Hz.

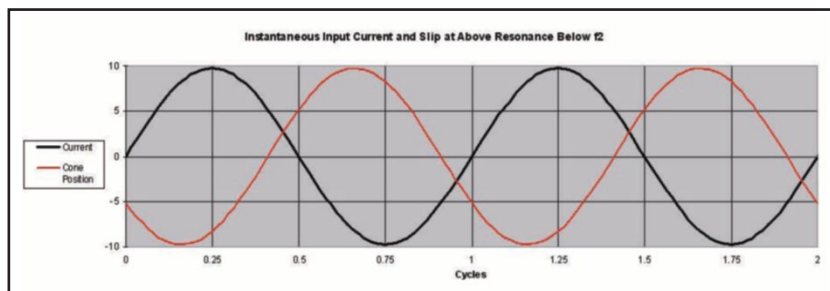


Figure 8: There is an increasing angle of slip (-148°) as frequency is increased above resonance. The selection of the frequency, 45 Hz, is arbitrary.

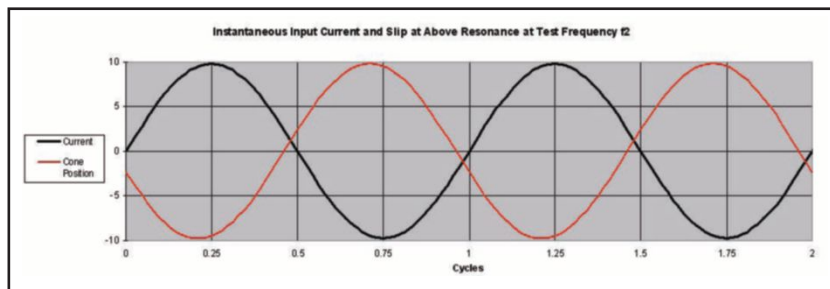


Figure 9: The Fisher woofer we been using, at a frequency of f_2 (59.3 Hz), shows a slip of 166° .

References

- [1] D. Jenkins, "Dynamic Characteristics of Driver Impedance," *Speaker Builder*, One: 1997.

About the Author

Andy Lewis lives in Englewood, CO, plays drums, and produces "Acme Sound Low-B" low-coloration loudspeakers for bass players. He is also involved in Special Olympics and other disability activities with his son, Collin.



Stereo Listening

Acoustics, Electroacoustics, and Psychoacoustics

Our experience of sound is necessarily subjective, but knowing the history of the science of stereophonic sound can help us to make intelligent decisions about our listening spaces and equipment.

By
Richard Honeycutt
(United States)

The first method for recording sound was Léon Scott's phonautograph, developed in 1857. It provided a visible trace of sound waves, but did not provide a way to play back sounds. A way to retrieve sound from a recording was suggested by French inventor Charles Cros, and demonstrated independently by American inventor Thomas Edison in 1877. The gradual move from acousto-mechanical to electrical recording/playback proceeded throughout the early 20th century, but always with only one channel of sound. Even though fidelity improved with these advances, no spatial information was available to the listener.

French inventor Clément Ader demonstrated a two-channel audio system in Paris in 1881, using telephone transmitters connected from the stage of the Paris Opera feeding pairs of telephone receivers through which listeners could experience something very like what we today call binaural reproduction. This process was commercialized in France and England from 1890 to 1932, but it was used for live performances only—sound recording and reproduction were not involved.

British engineer Alan Blumlein invented and patented stereo recording and playback in the early

1930s, with the first stereo discs being cut in 1933. A recording of Mozart's *Jupiter Symphony* was released in 1935, the same year that the first stereophonic motion picture sound was demonstrated.

Meanwhile, in the US, Harvey Fletcher of Bell Telephone Laboratories (BTL) was experimenting with a "maximalist" approach to providing realistic spatial information for electrically transmitted audio. Fletcher used up to 80 microphones to pick up the sound of an orchestra, with the signals being transmitted via equalized telephone lines to the same number of amplifiers and speakers. The speakers were spatially distributed in the listening room to match the microphone positions in the recording room.

After much research, BTL decided to focus on three-channel stereo, and in 1940, it demonstrated its system using three state-of-the-art, all-horn loudspeaker systems to reproduce stereophonic recordings of the Philadelphia Orchestra for listeners in Carnegie Hall. Because of the limited dynamic range of the optical stereo recording/playback system, compression and expansion had to be used, and these were not entirely automatic, so

that operator tastes influenced the outcome of the demonstration.

Also in 1940, the Walt Disney motion picture *Fantasia*—the first multichannel movie—was released. The optical recording system included left, center, and right tracks, plus a track of pilot tones to synchronize the compression in the recording with the expansion upon playback. A variety of examples and technologies for multichannel cinema sound have been used over the years, with significant variations in the methods employed and the results obtained.

Research on Reproduced Sound

All these explorations of means to add realism to reproduced sound involved significant research. The landmark Symposium on Auditory Perspective papers described BTL's demonstration of three-channel stereo on April 7, 1933 in which the Philadelphia Symphony Orchestra performed in the Academy of Science Hall in Philadelphia. The performance was picked up by three microphones and reproduced via three amplifiers and three speaker systems in Constitution Hall in Washington, DC. Harvey Fletcher's lead paper in the series, "Audio Perspective—Basic Requirements," details four requirements for realistic audio reproduction:

- A reasonably flat frequency range covering all the audible components of the sounds to be reproduced
- A dynamic range sufficient that soft sounds can be easily heard above the ambient noise, yet loud sounds do not overdrive the system into nonlinear distortion
- Accurate reproduction of the original reverberation characteristics of the recorded program
- Preservation of the spatial pattern of the original sound

The paper "Audio Perspective—Physical Factors," by J. C. Steinberg and W. B. Snow described experiments comparing two- and three-channel stereo, in which both discrete and "phantom-derived" three-channel systems were tested. Both the discrete and the phantom-derived third channel had advantages in terms of keeping the center sonic image stable for a range of listener positions. (Indeed, Paul Klipsch was a strong advocate of using a phantom-derived third-channel speaker, even though the source recording was two-channel stereo.) Steinberg and Snow considered sound localization to be a function of interaural sound pressure differences. (Today, we understand that both arrival time and interaural phase differences are

also important.) Among their conclusions, Steinberg and Snow stated:

"Because of ... physical factors controlling auditory perspective, point-for-point correlation between pick-up stage and virtual stage positions is not obtained for 2- and 3-channel systems.... The application of acoustic perspective to orchestral reproduction in large auditoriums gives more satisfactory performance than probably would be suggested by the foregoing discussions. The instruments near the front are localized by everyone near their correct positions. In the ordinary orchestral arrangement, the rear instruments will be displaced in the reproduction depending upon the listener's position, but the important aspect is that every auditor hears differing sounds from differing places on the stage and is not particularly critical of the exact apparent positions of the sounds so long as he receives a spatial impression. Consequently 2-channel reproduction of orchestral music gives good satisfaction, and the difference between it and 3-channel reproduction for music probably is less than for speech reproduction or the reproduction of sounds from moving sources."

The next three papers, by E. C. Wente, A. L. Thuras, E. O. Scriven, H. A. Affel, R. W. Chesnut, and R. H. Mills, discussed the microphones and loudspeakers, amplifiers, and transmission lines, respectively, used for the demonstration. The concluding paper, "Auditory Perspective—System Adaptation," by E. H. Bedell and Iden Kerney, examined the acoustical challenges. **Figure 1** shows the floor plan of the

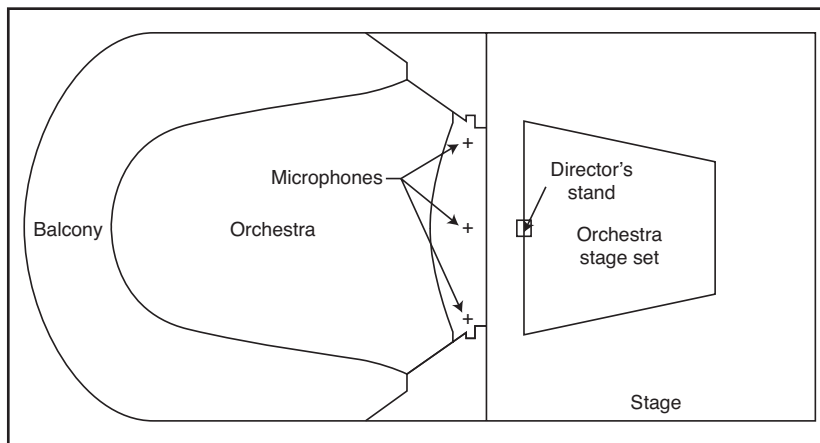


Figure 1: This floor plan of the Academy of Music Hall in Philadelphia shows the positions of the recording microphones used in the BTL demonstration. Here the term orchestra refers to the part of the theater where the orchestra plays, typically in front of the stage and on a lower level than the audience or the seats on the ground floor in a theater. (Original image courtesy of *Electrical Engineering*, January, 1934)

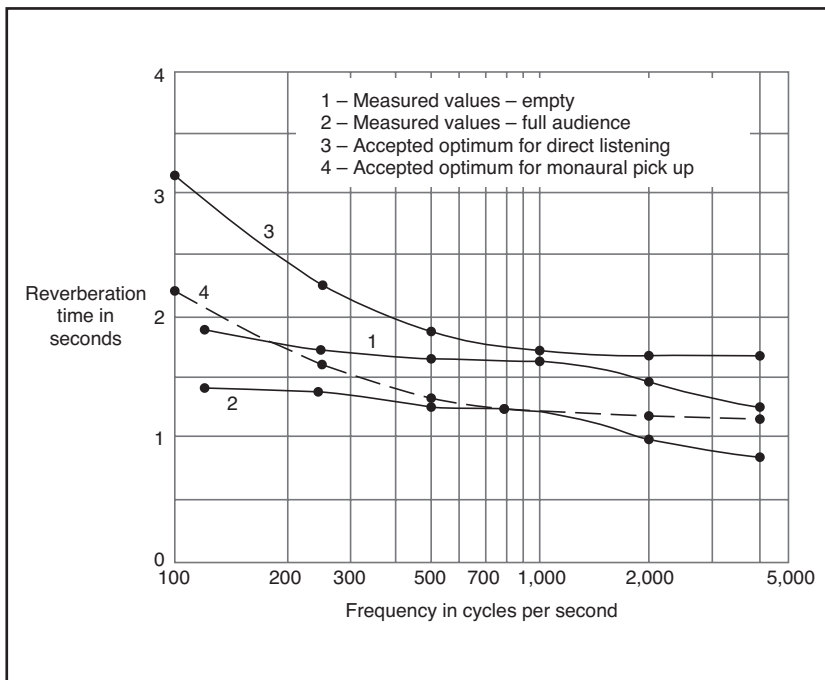


Figure 2: The measured RT for both unoccupied and fully occupied conditions of the Academy of Music Hall are graphed, along with ideal RT characteristics for direct listening and for monaural recording. (Original image courtesy of *Electrical Engineering*, January, 1934)

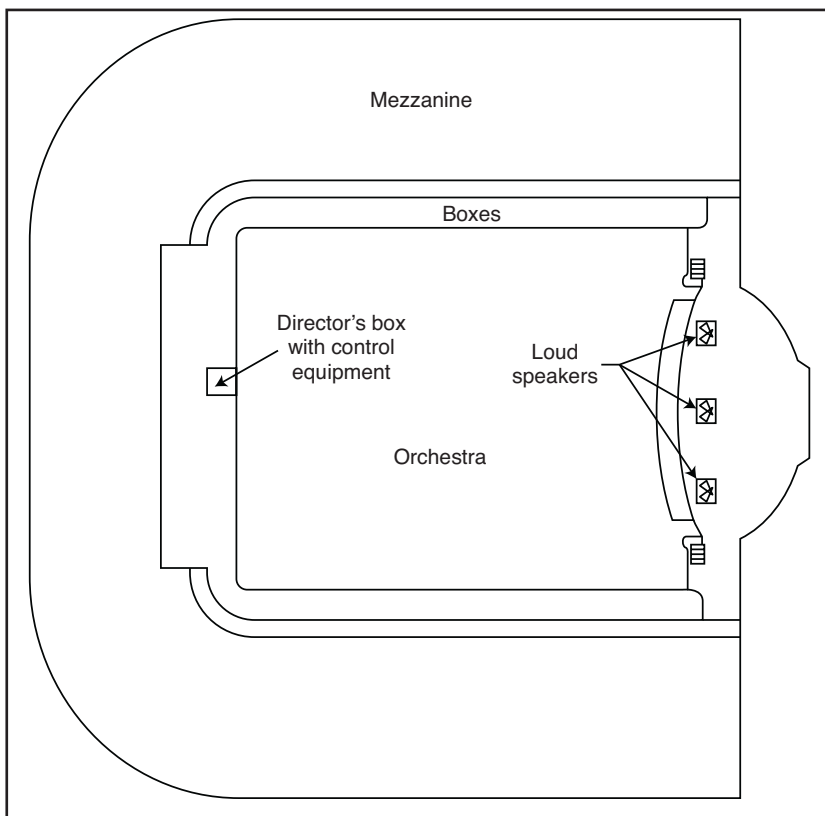


Figure 3: The floor plan of Constitution Hall shows the positions of the three loudspeakers. (Original image courtesy of *Electrical Engineering*, January, 1934)

room in which the orchestra performed. **Figure 2** is a plot of the measured Reverberation Time (RT) in the room. While the Academy Hall is somewhat dry for direct listening, its acoustics did fall near the optimum for monaural recording. **Figure 3** and **Figure 4** show the floor plan and RT characteristics, respectively, of Constitution Hall.

Recordings with Well-Defined Sound

The entire goal of the experiments described in the Symposium was to present as realistic an acoustical experience for the listeners as possible. The loudspeakers in the listening room were hidden behind optically opaque curtains. For many years, serious music listeners have had the same goal: realistic reproduction. It has sometimes been overlooked that this goal does not apply to all—perhaps even to the majority of—home listening systems today. First, there is the original sound source. A recording of a symphony orchestra represents an acoustical event for which the idea of “realistic reproduction” makes sense.

Many recordings heard by today’s listeners are of electrical and/or electronic musical instruments, which have no well-defined sound. In many cases, the recordings result from multi-session efforts in which there was never a time when all the musicians were present in the recording room at the same time.

In these cases, there is no original acoustic event in respect to which a reproduction can be assessed as realistic or not. Fletcher’s first two requirements should be met in order for the listening experience to be optimal, but the third and fourth requirements (realistic reverberation and spatial realism) are undefined. In these cases, the completely subjective criteria for a good music-reproduction system reduce to “pleasant-sounding.” Many people in the 1950s and 1960s considered a boomy jukebox pleasant-sounding.

Two important characteristics of an acoustic performance in a particular hall are the envelopment and the related apparent source width (ASW). These are perceptual factors corresponding mainly to the difference in the details of the acoustical signal from ear to ear, as quantified by the Interaural Correlation Coefficient (IACC). IACC, in turn, is related to:

1. The “lateral fraction,” which is the fraction of the total sound reaching the listener that comes from the sides rather than front or rear.

Experienced listeners prefer an ASW similar to the actual source width in the original acoustic event. Casual listeners may be easily impressed by an unnaturally large ASW.

2. The Interaural Time Delay (ITD) caused by the head (usually located between the ears!)

Multiple Speakers

In the 1960s, *Popular Electronics* magazine published a DIY article on a speaker system using 16 inexpensive radio speakers in a closed box. The author said the design had come to him after he heard a retailer's display of dozens of AM radios stacked together and tuned to the same station. To him, the sound was wonderful. Naturally, the multiple speakers radiating in phase mutually improved low-frequency loading, boosting the mid-low frequencies. (Those 5" speakers could not be said to have any bass response.)

Certainly this provided an improvement over the single 5" speaker in an open-back cabinet common to table radios of that era. Another part of the characteristic sound of multiple speakers mounted in close proximity is interleaved comb filtering. While comb filtering from two or three sources can create unnatural "honky" sound, comb filtering from many sources seems to trick the ear/brain system into thinking it's hearing a sound field reflected from a complex diffusing surface. This increases IACC and creates a perception of greater envelopment and ASW. Many non-critical listeners enjoy the resulting sound, although critical listeners often find that it obscures intimacy and perception of details. At least one major loudspeaker manufacturer made a substantial amount of money from a speaker design using multiple small drivers. This manufacturer also touted the combination of direct and reflected sound their speakers produced—this was supposed to be some sort of advantage.

Sound Reflections

The effect of sound reflections from the walls of a listening room has been much discussed. Certainly the reflections increase the lateral fraction, but the reflected sounds inevitably combine both direct and reflected sounds when considered from the point of view of the recording microphones. It is not clear whether sound reflected from the sidewall of a listening room, with a delay of about 15 ms is perceived in a way that is at all similar to the reflection from the side wall of a concert hall, having a delay of about 87 ms.

In the first place, the work of Helmut Haas instructs us that the sidewall reflection in the listening room will fuse with the direct sound into a single acoustical event. In the second place, the ear/brain approximates the size of a listening space from the initial time delay gap: the "quiet time" between

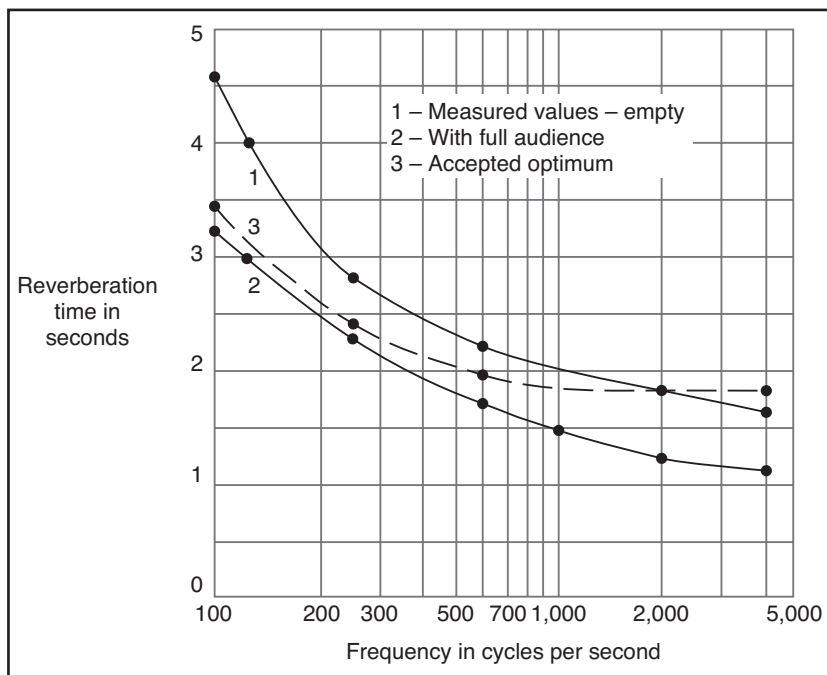


Figure 4: As you can see, Constitution Hall measured as much more reverberant than the Academy of Music Hall. (Original image courtesy of *Electrical Engineering*, January, 1934)

the onset of the first direct sound at the listener's ears and the beginning of the reverberant sound. To a certain extent, the ear/brain accommodates to the room whose characteristics it has approximated, more or less removing its effects from what is consciously heard. This accommodation affects both



Photo 1: The finest home-music speakers of the late 1940s and early 1950s were similar in size to the Klipschorn, shown here as a left-right pair with Klipsch's "small" Heresy speaker (23.8" × 15.5" × 13.25") providing the derived center channel. (Photo courtesy of Klipsch)



Photo 2: The Acoustic Research AR-1 with its “Acoustic Suspension” design was a bit smaller than the Klipsch Heresy. It became more popular in home listening rooms than the older, larger speakers.

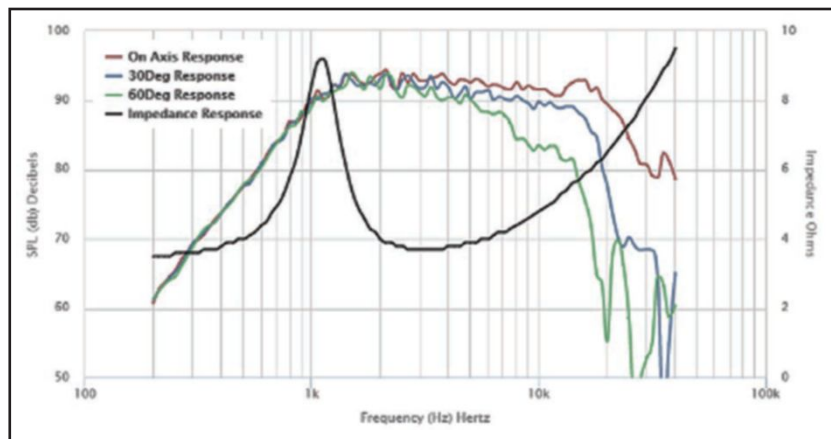


Figure 5: The frequency response curves on axis, 30° off-axis, and 60° off-axis of a 1” dome tweeter illustrate the high-frequency directivity of a typical home speaker today.

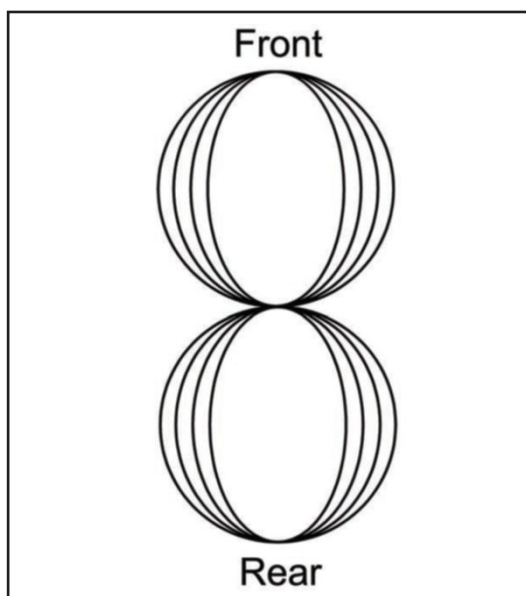


Figure 6: A dipolar speaker radiates from both the front and the rear. The width of the radiation pattern decreases as frequency increases.

frequency response and reverberation perception.

The situation addressed in the Symposium on Auditory Perspective is significantly different from that of a personal or family listening environment. Although the Philadelphia recording room and the Washington, DC listening room had very different reverberation signatures, both were adequate for listening to orchestral music. And by locating the microphones fairly close to the orchestra, the experimenters were able to keep the audio signal fairly dry, so that most of the reverberation heard by the listeners was that of the listening room. Today, someone enjoying a recording of an orchestra concert is usually listening in a room that is much drier than the hall or studio in which the concert was recorded. In a few cases, the listening room’s RT may be so short that it can be ignored: perhaps under one second. More often, the listening room’s RT will be an appreciable fraction of the recording space’s RT—likely one-quarter to one-third. In this case, the listener will hear a combination of the “recording RT” and the “listening RT” combination of these two reverberation signatures is not likely to have a value that is optimal for all recordings, or indeed for all combinations of recording and playback venues.

Loudspeaker Directivity

For many years, audiophiles have disagreed about loudspeaker directivity: Should one’s listening room have directional or non-directional loudspeakers? The speakers used in the Symposium were all horns, and they had directivities at mid and high frequencies of about 120° horizontal by 60° vertical. They were aimed using the same criterion that most pro-sound designers use today: “Put the sound where the people are.” Prior to 1950, most home listening rooms had speakers that were directional at mid and high frequencies (e.g., Klipsch recommended 90° vertical × 40° horizontal coverage angles, as provided by the mid- and high-frequency horns in the Klipschorn shown in **Photo 1**).

Low-frequency directivity required too much size. Then new wide-frequency-range speaker lines were introduced, which permitted much smaller cabinets—too small to provide controlled directivity—and omnidirectionality came to be touted as a desirable characteristic (see **Photo 2**). I personally believe that if it had been possible to build an AR1-sized and -priced speaker with the directional characteristics of a Klipschorn, the race to omnidirectional speakers would have been more like an amble. Actually, virtually all speakers on the market today are directional at high frequencies, as shown by the curves for a 1” dome tweeter (see

Figure 5). But there is another type of speaker that is neither unidirectional nor omnidirectional: the bidirectional dipole speaker that radiates from front and back (see **Figure 6**).

In the early days of surround sound, dipole speakers were recommended in the surround positions for home listening rooms. Some listeners prefer dipoles for stereo speakers as well. In October 1989, Gary Eickmeier presented a listening room design concept using dipole speakers located and aimed so that the reflections from the two sides of the speakers would simulate reflections that would be heard by a concert-goer present when the recording was made. Eickmeier's purpose was to fulfill the elusive fourth requirement for realistic playback: recreating the spatial information in the reproduced sound. For a certain class of recordings, his idea can work. The use of image modeling is scientifically valid, and is certainly an improvement over the haphazard results to be expected when omni or dipolar speakers are placed willy-nilly in the listening room. Yet difficulties remain.

The Search Continues

There is no standardization of microphone techniques for recording orchestras. Right-left or right-center-left mic placement will obtain very different reflected signals than any of the coincident pairs. Stanley P. Lipshitz has argued convincingly—and proved his point well—that the Blumlein pair (see **Figure 7**) is the best way to mic a sound source for stereo recording, but his suggestion has not yet been universally adopted by recording engineers.

Although most listening rooms today have reasonably predictable wall construction (gypsum or wood-paneled stud walls or masonry walls, all of which are highly reflective), there are some rooms having absorptive treatment on at least some of the walls, in order to prevent annoying acoustical artifacts. And even though the walls are reflective, their acoustical absorption is not uniform with frequency, and so neither will the reflected sound exactly retain its original frequency characteristics.

The interaction of the RT in the recording and listening rooms is complex and not subject to standardization, so the spatial reproduction of the recorded reverberant sound remains a wild card.

There is no standardization of recording venues. Some performances are recorded in large studios, and some, in concert halls. The dimensions that control the aural localization of sources vary widely. The sound stage may be wider, narrower, deeper, or shallower than it would sound in the recording venue.

We are left with a situation familiar to all

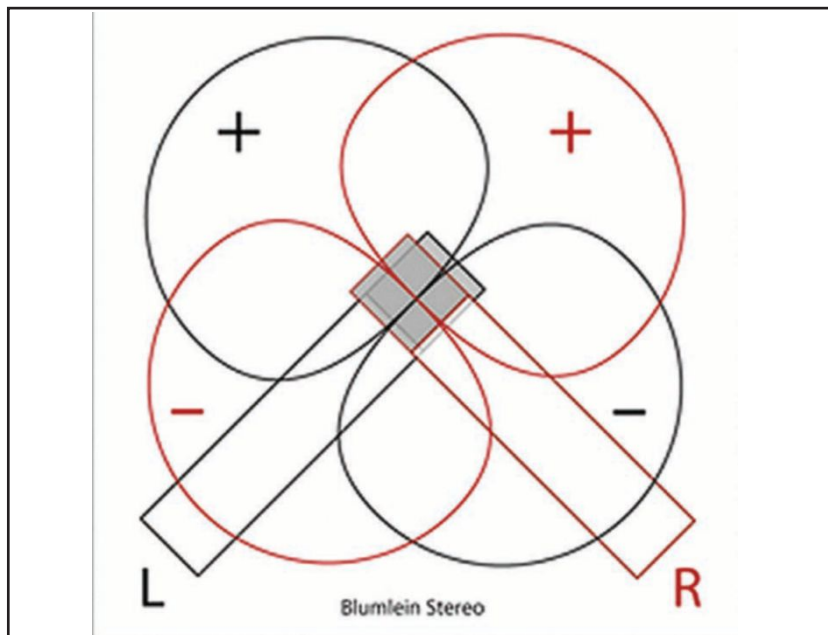


Figure 7: The Blumlein pair stereo miking technique uses crossed bidirectional microphones. It blends well into a mono signal, and Stanley Lipshitz has proven that it outperforms spaced-mic techniques for stereo recording.

audiophiles: There is no perfect way to meet all of Fletcher's requirements. In our own listening rooms, we can choose what sounds best, or most accurate, or most realistic, or even most impressive to us. Pity the acoustician who has to please many clients with varying tastes! But hopefully this survey of the science of listening room design has introduced us to some valuable tools. 

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MPEG-H Audio Brings New Dimensions to TV Sound

Fraunhofer Institute for Integrated Circuits IIS designed a way to significantly improve immersive sound reproduction capabilities on consumer playback devices, based on MPEG-H decoding and rendering and Fraunhofer's upHear soundbar processing. This article is a technology update on MPEG-H and Fraunhofer upHear implementations, now available for licensing to audio video receiver (AVR) and soundbar manufacturers.

Photo 1: MPEG-H offers three dimensions of efficient audio transmission: channels, objects, and ambisonics. Channel-based audio ensures, among other things, that all conventional content can be broadcast using MPEG-H.

By
Stefan Meltzer

(Technology Consultant for Fraunhofer IIS)

Imagine watching a crime drama on TV. The detective interrogates a suspect as suspense-filled music plays in the background. The back-and-forth dialog, however, is hard to understand because of the music's volume. A similar dilemma happens while watching a live sports broadcast, where the commentary drowns out the stadium atmosphere. TV stations have to deal with issues like these all the time. They have to find a compromise in the audio mix that suits a diverse audience. What would change the game is enabling viewers to change the audio playback to their own preferences.

The MPEG-H TV Audio System, substantially developed by Fraunhofer IIS, is designed to offer new interactive options for the viewer as well as deliver immersive 3D sound. The format is integrated in the ATSC 3.0 and DVB television standards. It has been on air in South Korea since May 2017 as part of the world's first regular terrestrial UHDTV service. MPEG-H, however, is not only a new audio codec, but a complete

system—from the microphone to the loudspeaker—and opens up new options for creatives as well as new sound dimensions for viewers.

MPEG-H Audio: Creative Freedom Through Channels, Objects, and Ambisonics

MPEG-H offers three dimensions of efficient audio transmission: channels, objects, and ambisonics. Channel-based audio ensures, among other things, that all conventional content can be broadcast using MPEG-H (see **Photo 1**). Additionally, the capabilities of channel-based audio are enhanced by the introduction of height channels, enabling the playback of true 3D immersive sound. For usage in ATSC 3.0 and DVB, the number of available loudspeaker channels in the MPEG-H TV Audio System is limited to 12. That way, common formats such as 5.1+4 and 7.1+4 are possible. For the Japanese market, the MPEG standard allows for a variant that supports the 22.2 system proposed in Japan.



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About the Author

Stefan Meltzer studied electrical engineering at the Friedrich-Alexander University in Erlangen, Germany. In 1990, he joined the Fraunhofer Institute for Integrated Circuits (IIS) in Erlangen, Germany. After working in the field of IC design for several years, Stefan became the project leader for the development of the WorldSpace Satellite Broadcasting System in 1995 and in 1998 of the XM Satellite Radio broadcasting system. In 2000, he joined Coding Technologies in Nuremberg as Vice President for business development, Germany. After Coding Technologies was acquired by Dolby Labs, Stefan joined Iosono as CTO in April 2008. Since January 2010, Stefan has worked as an independent technology consultant with the main focus on audio and multimedia. In this role, he supports Fraunhofer IIS in the business development and marketing activities within the TV broadcast market.

The second dimension is audio objects, which include interactive elements or dynamic sound effects, such as a helicopter flying “above” the viewers. Objects are not tied to a specific playback setup, but are defined via their positions in the room. Supplementary metadata, which describe the characteristics of each object, are also transmitted inside the bitstream. With the help of such metadata, the object is reproduced over the loudspeakers in the playback setup. Another advantage of audio objects is their ability to be manipulated by the viewer within boundaries defined by the content providers. For example, a viewer can switch between different languages or choose from several commentators for sports broadcasts.

The transmission of music and effects in a channel-based “sound bed” along with dialog and commentary tracks as objects enables TV broadcasters to efficiently provide different commentary options—including in multiple languages—since the sound bed only needs to be transmitted once instead of separately for each language.

Another benefit of using objects for the speech elements in a broadcast is that the volume and, if desired, the position of each object can be manipulated. For the first time, this enables the user to individually optimize the mix of dialog and music/effects based on their specific listening situation. This option, also known as “Dialogue Enhancement,” also makes it easier for viewers with hearing impairments to understand the TV broadcast. When it comes to playback on mobile devices, the mix can be further adapted to surrounding noise conditions.

Ambisonics, the third dimension, is a mathematical reproduction of the sound field. As such, it is independent from the playback setup. This makes Ambisonics an excellent option for all those applications that require a manipulation of the audio contents (e.g., Virtual and Augmented Reality). The

order of the Ambisonic representation—First Order Ambisonics (FOA) or Higher Order Ambisonics (HOA)—defines the spatial resolution of the signal: The higher the order, the more precise the localization in the sound field. The MPEG-H TV Audio System supports Ambisonic representations up to sixth order. The combination of Ambisonics and objects enables working with a low order for Ambisonic representations. That way, Ambisonics is used for the sonic background atmosphere and audio objects for the exactly localizable sound elements.

MPEG-H Used in Production

MPEG-H Audio has been on the air since May 2017 as the sole audio codec of South Korea’s terrestrial UHDTV service and was most recently used to broadcast the 2018 Winter Olympics in Pyeongchang. This means that the complete production and transmission chain was fully realized using MPEG-H. The respective equipment is readily available, including broadcast encoders from various manufacturers, audio monitoring and authoring units for the production of metadata in live broadcasts as well as various post-production plug-ins for Digital Audio Workstations (DAWs). The transmission of metadata in the studio and production ecosystem happens via SDI, meaning that the audio tracks are transmitted in an uncompressed way in the PCM format, with the metadata in the “Control Track” format via the 16th SDI channel.

In the last few years, the team of sound engineers at Fraunhofer IIS gained significant knowledge and expertise in recording and producing content with the new opportunities offered by MPEG-H (see **Photo 2**). This knowledge has been utilized in several TV productions in collaboration with European broadcasters, such as a documentary by one of the major public TV stations in Germany. The know-how in creating immersive and interactive content is also being passed on to other sound engineers at various workshops. In South Korea, Fraunhofer IIS operates a training and demo facility where courses and demonstrations for the Korean TV stations are offered.

MPEG-H Playback

The biggest challenge for MPEG-H playback is making the interactive features user-friendly and easily accessible. Therefore, the MPEG-H TV Audio System comes with two levels of interaction. The first level includes broadcaster-set presets that offer a selection of useful alternatives to the original mix. These may include raised dialog volume, access to a different sports commentator, or the isolated stadium atmosphere. The second level enables users to fine-tune

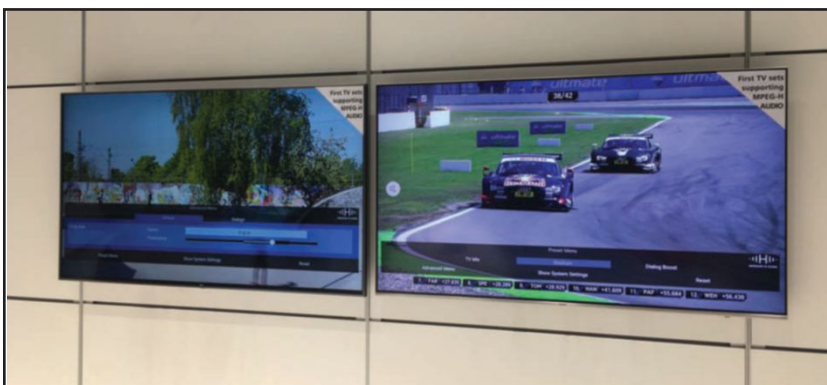


Photo 2: In the last few years, the team of sound engineers at Fraunhofer IIS gained significant knowledge and expertise in recording and producing content with the new opportunities offered by MPEG-H.

the various objects they can individually manipulate. The extent to which the mix can be manipulated by the user is determined by the presets, which are defined via metadata from the broadcaster.

MPEG-H-enabled TV sets have been on the market since March 2017 from manufacturers such as Samsung and LG. While the interactive features can be enjoyed on every device with MPEG-H support—from large UHD TVs to small smartphones—the reproduction of true 3D sound requires additional hardware. Most TV viewers will, however, be reluctant to install a 3D loudspeaker setup in their living rooms, since it requires numerous loudspeakers and cables along with specific technical knowledge for setting them up. A more consumer-friendly solution to 3D sound reproduction at home comes in the form of a 3D soundbar. The first 3D soundbars capable of playing back MPEG-H content are expected to enter the market later this year, including one from the German company Sennheiser, which debuted its MPEG-H enabled soundbar prototype at CES 2018 to rave reviews (see **Photo 3**).

3D Audio Rendering

The most recent Fraunhofer development in this field takes care of rendering 3D sound in a tailored way: The upHear Immersive Audio Virtualizer is a codec-agnostic audio post-processing technology that enables the reproduction of high-quality 3D sound from any input source in soundbars or TV sets.

Fraunhofer upHear consists of a sophisticated signal analysis and optimal use of the device hardware for signal reproduction (see **Figure 1**). The upHear algorithm is individually tuned by experienced Fraunhofer sound engineers for the various soundbar product categories. This approach has, among other achievements, led to the development of an effective soundbar design that reproduces immersive audio without the need for additional satellite surround loudspeakers and subwoofers.

Thanks to an intelligent sound distribution in the room, an upHear-enabled soundbar can achieve an immersive audio quality that's almost indistinguishable from discrete 3D loudspeaker setups. upHear can process immersive audio formats as well as legacy surround and stereo content sources which are spatially enhanced with the help of the integrated upmix. The result is a 3D audio consumer experience that is convenient and easy to set up, without a compromise in high-quality sound reproduction.

MPEG-H Audio is the perfect partner for upHear, as it enables an efficient transmission of high-quality immersive audio through TV broadcasts or streaming services, at bit rates which are typically used just for 5.1 surround transmissions today.



Photo 3: The first 3D soundbars capable of playing back MPEG-H content are expected to enter the market later this year, including one from the German company Sennheiser, which debuted its MPEG-H enabled soundbar prototype at CES 2018 to rave reviews.

With MPEG-H and associated technologies, Fraunhofer continues the drive to give consumers greater freedom in the way they interact with audio content along with the most convenient way to enjoy immersive sound. Such advancements in audio technology break through the limitations that made enjoying a favorite broadcast a somewhat passive experience. As consumers realize the advancements now available at their fingertips, it is reasonable to expect even more tools and capabilities to be delivered soon by innovators such as Fraunhofer.

More information on the MPEG-H Audio System is available at www.mpegh.com. 

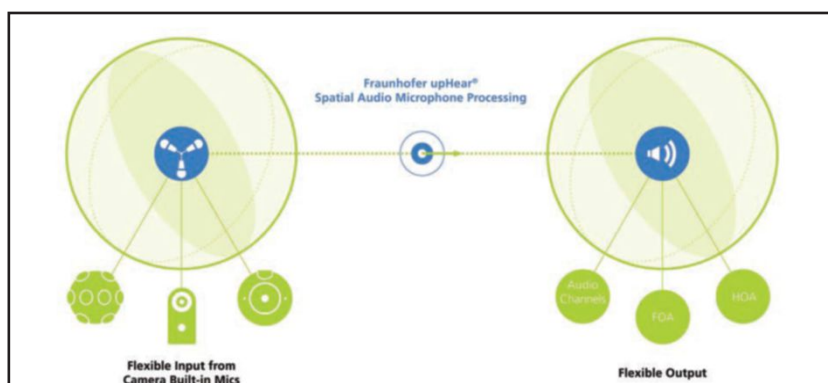


Figure 1: Fraunhofer upHear consists of a sophisticated signal analysis and optimal use of the device hardware for signal reproduction.

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upHear, www.uphear.com

Mixing and Mastering for Today's Music Streaming Services



In 2015, the world changed as music streaming went from being the biggest delivery vehicle to the highest revenue generator in the music industry. Spotify, Apple Music, TIDAL, and others have become the dominant way music is consumed and purchased. While mixing and mastering for CD delivery is now a minority interest, there is still very little understanding and only a few tools available to enable those in the industry to take an alternative approach.

By
Jon Schorah

(Creative Director, NUGEN Audio)

"Streamed audio" means data compression. To reduce the vast amounts of data music streaming services deliver (more than 1 billion streams a day), most of them employ data compression techniques (audio codecs). For example, instead of delivering a large, high-quality .wav file, an AAC or Ogg Vorbis file will be used instead. Unfortunately, these codecs are not very good at handling "hot" audio. What may sound great in the Digital Audio Workstation (DAW) can be easily distorted when converted to a data-reduced audio stream.

Digital audio reduces the analog waveform to a series of discrete measurements (samples). In between these samples, the analog waveform continues smoothly, so in some cases, the reconstructed waveform can be higher than the

sample measurements. This is known as an "inter-sample peak." The problem, well known in the TV broadcast industry, is that codecs can't handle inter-sample peaks, producing distortion as the waveform is incorrectly reconstructed.

Until recently, the only way for the engineer to monitor this behavior was to export the audio from the DAW, apply a compression codec, and listen to the results. While many music streaming services readily publish the necessary information, there are many codecs to consider, and several use different bit rates in different settings (mobile, desktop, premium, etc.), resulting in an extremely arduous and time-consuming set of forensic listening tests for the engineer who is willing to take up the challenge.

Fortunately, new tools are becoming available that

allow real-time codec auditioning and metering within the DAW. Using a suitable plug-in, it is now possible to monitor several codecs simultaneously and check that inter-sample peaks are not causing downstream codec conversion issues. If codec distortion is a problem, this can then be controlled using an inter-sample (true-peak) limiter to transparently rein in transients that the codec finds difficult to render.

However, the situation is a little more complex than this. As previously mentioned, different bit rates are employed in differing situations. A lower bit rate will involve a higher level of data compression, and low bit rate algorithms tend to be much more susceptible to overload distortion than their higher bit rate counterparts. Optimizing for low bit rates can lead to unacceptable compromises to mix quality in higher bandwidth situations. During listening tests, it soon becomes clear that low bit rate streams suffer many other additional issues over and above overloads due to inter-sample peaks. Bearing this in mind—and the fact that in general, low bit rates are employed in mobile listening environments (meaning ear-buds on the subway, etc.)—it soon becomes apparent that optimizing for high-quality streams is the way forward, where the difference can be appreciated and genuine quality can be preserved.

Codec Listening Tests

Codecs employ several tricks in order to reduce the bandwidth required for music streaming. The first is to remove “perceptually irrelevant” signal components. This technique uses psycho-acoustic models to determine audio, which is difficult (or impossible) to perceive and discards this information in the encoded audio. Strategies include high-frequency limiting, absolute threshold of hearing, temporal masking (sudden loud sounds making it difficult to perceive quiet sounds immediately following) and simultaneous masking (one sound masking another simultaneous sound).

A second level of compression is known as Spectral Band Replication (SBS). When SBS is employed, high-frequency content is discarded, and this information is reconstructed using the remaining low- and mid-frequencies, with guidance from an information stream transmitted alongside.

Finally, in highly data-reduced situations, a technique known as Parametric Stereo (PS) can be employed. Analogous to SBS, in this case, audio is down-mixed to mono and reconstructed to stereo at playback. Yet again, this technique clearly reminds us that mono-compatibility is still a very important aspect of audio production.

Flicking between different bit rate codecs of the same audio is a simple procedure to learn to

identify the “sound” of these different processes. A particularly revealing test is to only listen to the “Side” component of a mid/side signal, A/B switching between the original source and encoded signal. Many of the encoding artifacts, which are quite cleverly masked in the full signal, suddenly become extremely obvious as “musical noise” and can then be clearly heard in the full mix once the ear is trained (see **Figure 1**).

Streaming services have also changed the game in another very important way. Because streaming playlists often have many songs from different artists, genres, and periods in time, the services match the loudness of each song from one to the next, to avoid having the listener constantly reach for the volume control. This is achieved by a process called loudness normalization. Loudness normalization works by analyzing the entire song for its average loudness and then turning it up or down to a set level for the system. As a result, all streaming audio comes out on average, at more-or-less the same loudness.

The implications of this simple change in music consumption are widespread. Previously, it was possible to make a song “stand out” from the competition by using heavy audio compression and limiting. Commonly referred to as “The Loudness War,” this practice led to a situation where making a song as loud as possible became a primary objective.



Figure 1: Real-time codec listening tests can be conducted with NUGEN Audio MasterCheck Pro.

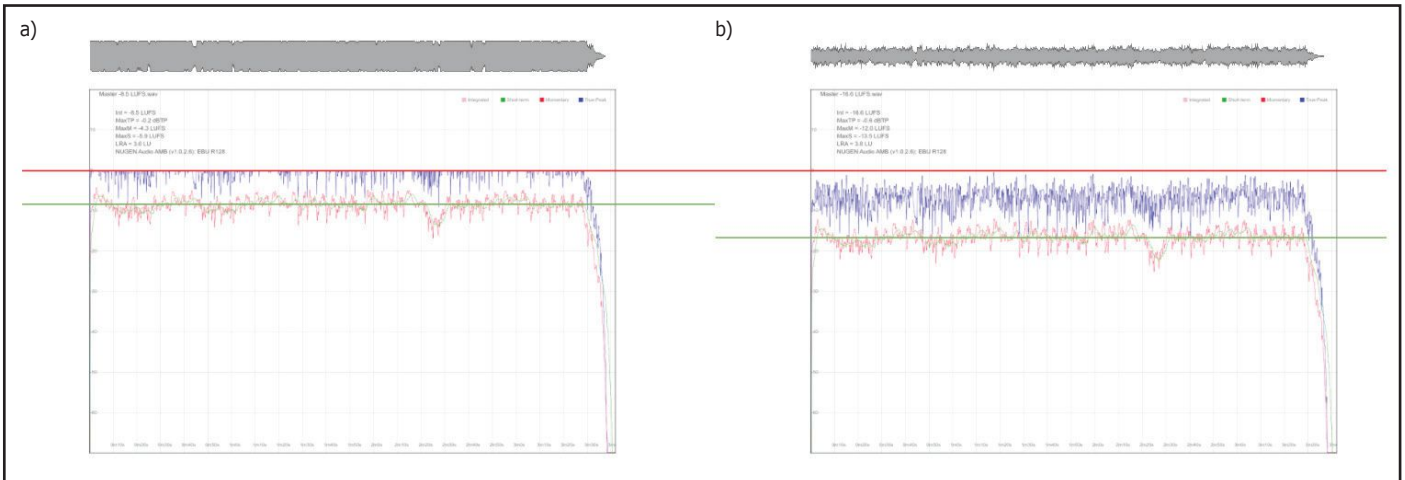


Figure 2: A Peak Normalized system shows the heavily limited master (a) is several decibels louder than the more dynamic master (b).

Loudness normalized audio streams remove this “advantage”—it is not possible to make a song louder than the level determined by the playout system (see **Figure 2**). This means that audio compression and limiting returns to the role of aesthetic enhancement. In fact, in a loudness normalized environment, very heavily limited masters sound flat, lacking presence and detail. Once the master becomes louder than the playout level, the system turns down the level, resulting in audio peaking well below any true-peak limitations, where a less heavily limited version makes use of this available space, capturing more transient detail as a result (see **Figure 3**).

A less well understood implication is that of macrodynamics. If limiting and codec distortion concerns are primarily to do with retaining microdynamic transient detail, then average song

loudness invites us to consider the macrodynamic content—the wider variations between loud and soft within a song.

In a peak normalized environment, producers became used to the concept of a maximum level (0 dBFS) relative to which all audio is measured, a ceiling that cannot be exceeded. Loudness normalization changes this concept. The average loudness level is not a ceiling, but rather, the average level across the entire track. It is perfectly possible to have sections well above the average loudness level if they are balanced by sections of relative quiet. A track that takes creative advantage of this situation will have sections that are comparatively very loud when played back-to-back with a heavily limited master that sits on the system playback loudness level, with very little deviation (see **Figure 4**).

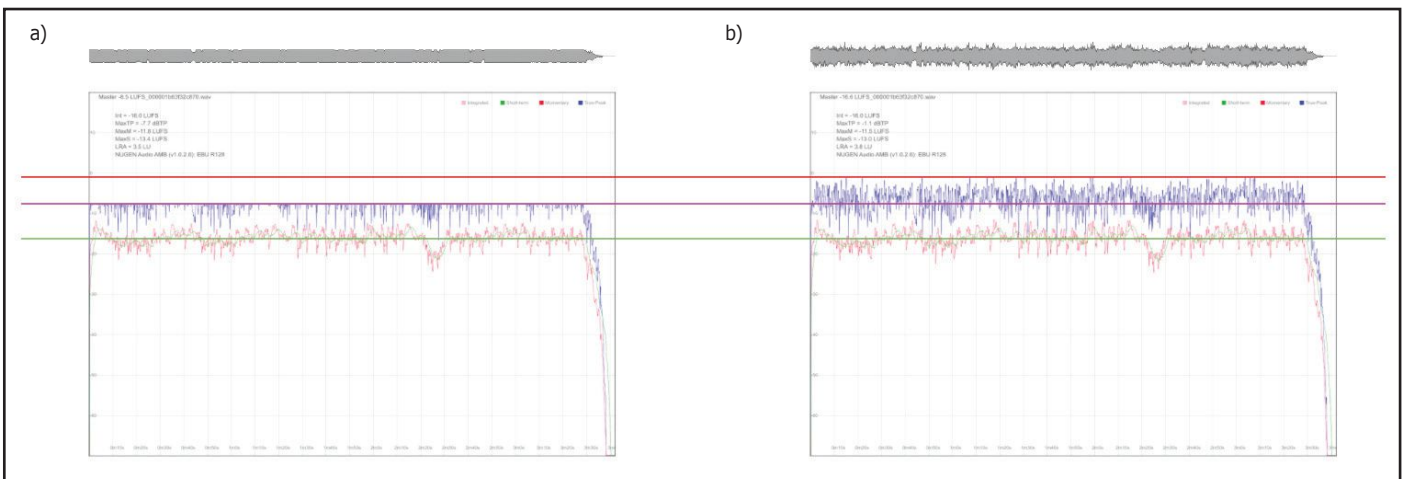


Figure 3: A Loudness Normalized system shows the heavily limited master now lacks transient detail, exhibiting “flat” peaks (a). The dynamic master now shows far more micro-dynamic detail (b). The loudness of both masters is the same.

Setting Industry Standards

As audio engineers grapple with these new consumption realities, a new audio descriptor has emerged—Peak to Loudness Ratio (PLR). This concept combines the True-Peak level and average (Integrated) Loudness level into a single parameter. This can be measured for an audio track and can also be used to describe the properties of a playback system. A streaming service can be thought of as having a “letterbox,” defined by a maximum True-Peak level that avoids unacceptable codec distortion, and an average loudness level. With an understanding of system PLR, an audio engineer can now ensure that their audio fits through this “letterbox” without modification.

A related descriptor is Peak to Short-term Loudness Ratio (PSR). This parameter is currently a subject of great interest, as it promises to become a useful, objective, real-time measure of audio dynamics. The Audio Engineering Society (AES) has already published a discussion paper on the subject, with the hope that an industry standard definition can be developed that will help engineers to consistently compare values from different meter manufacturers.

While industry standard metering will certainly help, there is much about today’s playback services that is far from standard. Unlike television broadcasting, there is no universally agreed level of music playback for streaming services. Fortunately, there has been a recent shift toward harmonization, with most of the major players now using fairly similar targets. There is also a great variety of codecs in use, with more coming on stream as the technology develops. A recent trend is that of high-resolution streaming, which promises to remove many of the issues associated with the lossy codecs generally employed today.

There is also work to be done in the area of music aggregation, publishing, and distribution. Even if one were to go to the trouble of producing multiple masters suited to hi-res streams, medium-quality codecs, and different loudness targets, it is difficult (if not impossible) to submit more than one master, which requires pragmatic compromises to be made.

Although there is still much to be resolved, researched, and harmonized, music streaming is here to stay, and has for the first time in years brought a year-on-year increase in revenue for the music industry. While it is a concern that much of today’s music is now heard through a lossy codec, options to listen to a hi-res or non-lossy version are appearing more frequently, and as bandwidth improves, we will see low-quality encoding fade into obscurity. Undoubtedly, the brave new world

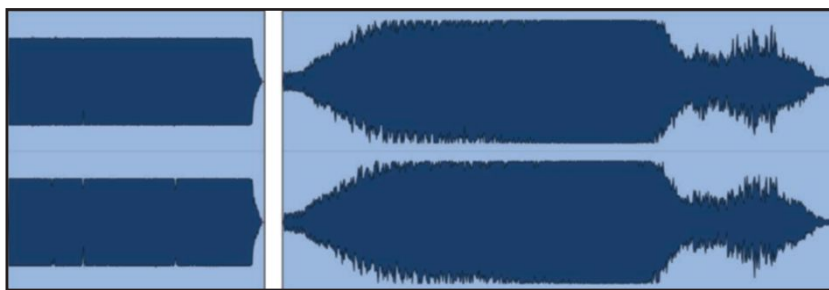


Figure 4: Two tracks, both with the same average loudness, are shown. The track on the right, however, has sections much louder than the heavily limiter master on the left.

of Loudness Normalized playback is good news for creative choice, free from the commercial demand to be as loud as possible. There are of course still many minor skirmishes (it takes a long time for old habits to die, especially in Artists and Repertoire circles!), but the Loudness War is over for those who wish to concentrate on making the music sound as good as possible, without having to worry about what everyone else is doing. 

About the Author

Jon Schorah is the creative director and co-founder of NUGEN Audio, one of the world’s leading manufacturers of loudness products. Jon has a background in mastering and engineering and has considerable experience in wider aspects of the industry. A 1992 Leeds University (UK) graduate, in recent years Jon has focused on product design with a particular interest in the usability and workflow aspects of audio software.

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Engineering Voice-Enabled Smart Gadgets

The first step companies face in enabling a voice user interface (VUI) experience is to decide whether to partner with a current market offering or to build a proprietary voice solution, which requires a significant amount of resources. This article offers an overview of DSP Group's experience in enabling voice in a variety of new consumer electronics devices, using its open audio/voice DSP platform.

By
Ofer Elyakim

(Chief Executive Officer, DSP Group)

DSP Group is recognized in the industry for its high-performance, ultra-low power audio/voice programmable DSP Core, available in a very small footprint device.

Voice is quickly becoming the preferred method of communication between consumers and their smart devices, and this rise in voice user interfaces (VUI) is changing the way people interact with machines. At the beginning of 2017, 49% of US broadband

households were using a personal assistant through an application or dedicated device. Products from Amazon, Google, and Apple have moved voice control into the consumer consciousness, and user expectations are quickly expanding to demand voice as part of all smart device interactions, including wearables.

The US wearable tech market is set to almost double by 2022 as "critical mass adoption" takes place, according to analyst firm Forrester. Wearables will grow from a \$4.8 billion industry in 2017 to just over \$9 billion in 2022, representing a compound annual growth rate (CAGR) of 9% from 2017 to 2022! With wearables growing in popularity, and consumers preferring to control all smart devices via voice, technology must be designed with communication range, reliability, and energy efficiency in mind.

During the first quarter of 2017, 27% of consumers with a smart speaker and personal assistant reported that they regularly control their smart home devices with this device, and adoption of smart speakers with personal assistants among US broadband households increased 3% from the first to the second quarter of 2017. This increase comes as no surprise as the current generation of tech-savvy consumers often rely on personal assistants and VUI to stay organized when they're on the go. Consumers want an easy and clear interaction with their devices, making the desire to speak with devices the natural next step.



A demonstration of a voice-enabled solution was given using DSP Group's development platform with the company's audio/voice processor.

A Chipset Solution

Crisp, natural communication and consistent accuracy has been hard to achieve, but technological developments, such as DSP Group's DBMDxx family audio system-on-a-chip (SoC) solution, have proven that VUI can attain a comprehensive communication range and an interference-free spectrum band that enables devices to listen at scale.

DSP Group's DBMDxx audio SoC solution delivers superior voice recognition and processing capabilities, regardless of device distance or ambient noise, by leveraging voice enhancement algorithms, including acoustic echo cancellation, noise suppression, beam-forming, and far-field support.

The voice enhancement processing integrated in the DBMD5 chip allows devices to actively listen for the right voice commands and process incoming triggers, enabling clearer communication for the end user and accuracy in high-noise environments. An additional benefit of the technology includes seamless cloud operation enabled by voice-buffer technology, which operates with automatic speech recognition (ASR) applications. Also, its dual-core digital signal processor supports digital and analog microphones that combine a variety of application processor interfaces (e.g., SPI, I²C, UART, GPIO and PCM/ I²S).

The solution is available in a BGA package (1.8 mm × 2.1 mm) featuring high-speed I²C, UART, TDM, SLIMbus, and SPI interfaces, as well as up to 20 GPIOs. Other features include 32-bit timers/ watchdog, interfaces for digital microphone inputs, integrated low power ADC for analog microphone, and JTAG-based debug port. The chipset is complemented with a comprehensive software framework and real-time operating system, with drivers for all processor peripherals, stand-alone master or slave with external master mode option, and an "Internal host" for management and communication with external master processor. The solution provides an easy connection with an external processor running Linux OS with Linux/Android drivers, with Dynamic Power Management.



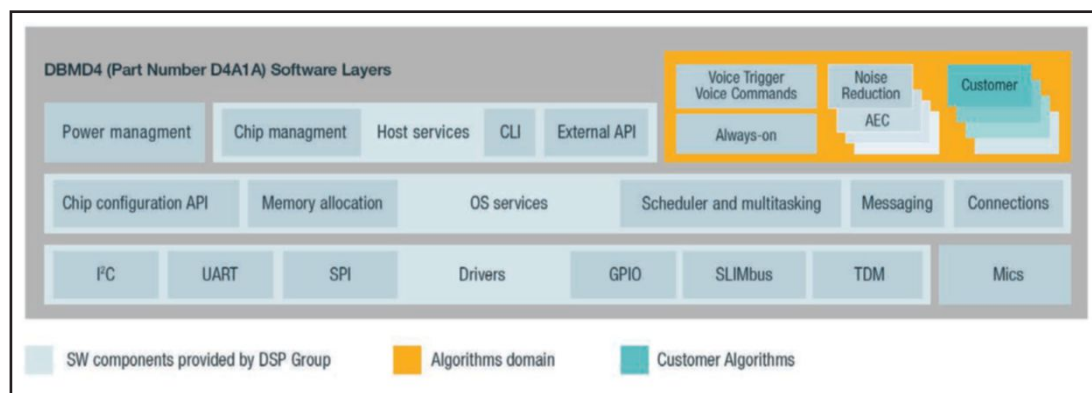
The DBMD4 (Part Number D4A1A) solution enables device activation and operation via voice commands while maintaining extremely low power consumption and high performance using pre-process algorithms such as noise reduction, beam-forming, and acoustic echo cancelling (AEC).

Smart Home and More

In addition to VUI being the preferred mode of interaction for personal assistants on portable devices, over one-third of US broadband households find it appealing to use voice in order to control smart home devices or entertainment devices. Furthermore, 53% of US broadband households that own a smart home device also have a smart speaker with a personal assistant, with only 16% owning a smart home device without a voice-based device in the home. This speaks to the importance of having technology that can enable voice activation to any device, whether it be a smartphone or a speaker, to enable seamless interactions between consumers and their smart home technologies.

Philio-Tech, a company that develops Internet of Things (IoT) products (e.g., network home gateways, temperature control management, intelligent sockets and lighting, and other intelligent home product solutions) is committed to Z-Wave, ZigBee, Ultra-Low Energy (ULE), Wi-Fi, LoRa, and other related technologies that will meet the future needs of the intelligent home. The Philio-Tech voice assistant, which is powered by DSP Group's ULE and SmartVoice technologies, is a tiny, battery-operated voice-controlled personal assistant that can perform a huge variety of tasks such as checking the weather at a destination you are planning to visit or turning on the lights at home as you arrive.

DSP Group's SmartVoice offering provides the far-field multi-mic noise suppression, barge-in, beam forming and "always-on" voice functionality previously mentioned.



The DSP Group DBMDxx family audio SoC processor is complemented with a comprehensive software framework that includes a real-time operating system and drivers for all processor peripherals.

DSP Group's HDClear family of SoCs offers enhanced voice enhancement, voice activation and sensor hub functionality, enabling ultra-low power, and always-on capabilities in all sorts of devices.

	DBMD2	DBMD4/4P	DBMD5/5P
Function			
Far-end noise cancellation	Enhanced	2nd generation	✓
Near-end noise reduction	Enhanced	2nd generation	✓
Maximized ASR	✓	✓	✓
AEC	✓	✓	✓
Speaker improvements	✓	✓	Optional
Supported voice bandwidth	NB, WB	NB, WB, Super WB, Full band	NB, WB
Voice activation and voice commands	✓	✓	✓
Sensor hub		✓	✓
Segment			
Mobile phone	✓	✓	
BT headset	✓	✓	
Cordless headset/Cordless	✓	✓	
Wired headset	✓	✓	
Tablets	✓	✓	✓
Wearable devices	✓	✓	
Smart speakers	✓	✓	✓
DSP			
Processor	DSP 32 bits	DSP 32 bits	Dual DSP 32 bits
Program memory	RAM	RAM	RAM
I/O			
GPID	13	13	✓
UART	✓	✓	✓
PCM/I2S	4	2	✓
I2C	✓	✓	✓
SPI	✓	✓	✓
SLIMbus	✓	✓	
AFE			
Audio In	4 Digital, ADC	3 Digital, ADC	7 Digital, 2 ADCs
Audio Out	Digital	Digital	Digital
Package			
Size	3 mm x 3 mm	2.1 mm x 1.8mm	10 mm x 10 mm



The Philio-Tech voice assistant is powered by DSP Group's ULE and SmartVoice technologies. Philio's Voice Assistant unique battery-operated device supports voice queries and two-way voice to power home automation services, real-time communications, and network connectivity.

One example of cutting-edge VUI comes from Libre Wireless Technologies. The LibreSync platform incorporates unique media streaming technology, scalable mic-to-cloud voice processing modules and broad mobile platform ecosystem support. In this specific case, Libre is using DSP Group's SmartVoice technology for its fully integrated LS5BV LibreSync voice and media streaming module. DSP Group's DBMD5 audio SoC solution helps Libre deliver superior voice recognition and processing capabilities, regardless of device distance or ambient noise, by leveraging voice enhancement algorithms, including acoustic echo cancellation, noise suppression, beam-forming, and far-field support. The voice enhancement processing integrated in the DBMD5 chip enables devices to actively listen for the right voice commands and process incoming triggers, enabling clearer communication for

the end user and accuracy in high-noise environments.

iFLYTEK is another company at the forefront of VUI innovation—a leading voice-recognition cloud service provider in China dedicated to the research and development of intelligent speech and language technologies, artificial intelligence, and hardware and software for the government, education, and financial sectors, among other fields. At the core of iFLYTEK's On-Device Speech Recognition Software is DSP Group's SmartVoice SoC, which enables easy integration for developers with a substantially shorter development cycle, transforming virtually any device into a voice-activated solution for hands-free control.

The ULE Alternative

DSP Group's ULE offering provides several additional benefits, including operation in an interference-free band. ULE operates alone at 1900 MHz, on a dedicated, licensed band, eliminating the chance of crossed signals. The solution's outstanding range allows covering more than 600 m outdoors—compared to less than 150 m for ZigBee, Z-Wave, and Wi-Fi—with superior indoor coverage moving through walls, ceilings, and floors, when compared to ZigBee.

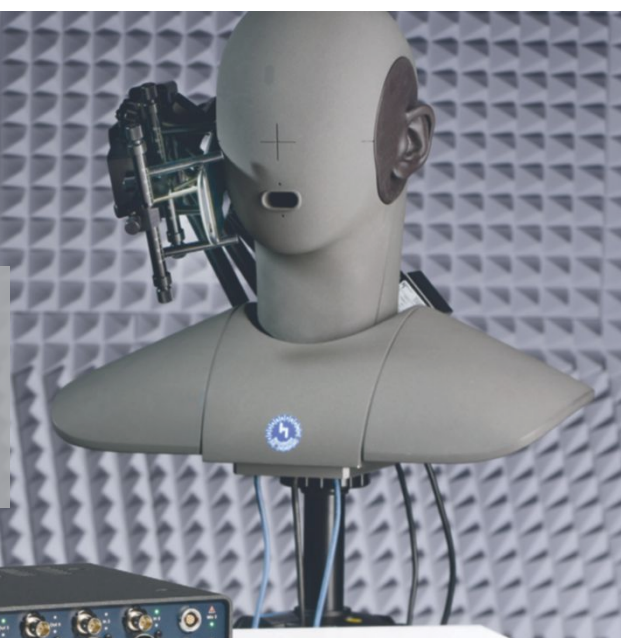


Libre Wireless Technologies' LS5BV LibreSync module features fully integrated and optimized far-field voice interface capabilities as well as extensive music streaming and multiroom audio features. This small, low power mic-to-cloud module uses DSP Group's SmartVoice technology and HDClear SoC with sophisticated algorithms for detecting and processing far field voice, including advanced noise reduction, echo cancellation, music barge-in and audio processing features. The compact LS5BV module is designed to support from 1 to 3 mics, offering flexibility for various applications that can be optimized for cost, front facing 180° or circular 360° applications.



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DSP Group's DBM family of SoCs is an innovative audio/voice solution for devices, enabling far-field device activation and operation via voice commands while maintaining low power consumption and high performance using pre-process algorithms. Thanks to DSP Group technology, the latest GoPro Hero 6 camera supports simple voice commands to enable users to start recording video, capture a single photo, begin a time lapse, or create a tag while recording video by saying: "That Was Sick!"



Providing advanced chipset solutions, DSP Group's innovative voice/audio technologies and advanced algorithms, including noise-reduction, acoustic echo cancellation and voice-enhancement allow OEM/ODM manufacturers to maximize the effectiveness of automatic speech recognition (ASR) engines, loudness equalization, and general voice enhancement. SmartVoice chip's small footprint enables it to be easily embedded in a wide range of devices, including tablets, smart speakers, and other microphone-equipped IoT devices.

All communications in ULE are encrypted using AES-CCM, with robust and secure RF channels. There is no un-encrypted mode and no option to switch off encryption as there is in legacy DECT or other technologies. Using a simple star network topology, means there's no need for repeaters to extend range and reliability. All gadgets and appliances can run from one central hub.

The ULE solution also offers natural support for two-way voice applications. It is ideal for carrying voice wirelessly, and together with the SmartVoice solution, it transforms any device into a voice-enabled product.

DSP Group's ULE and SmartVoice offerings are also both energy efficient, taking up minute amounts of power for always-on functionalities, lasting for years. The energy efficiency of voice-enabled smart devices is incredibly important because, whether battery or AC-powered, the staggering number of devices and data generated by IoT devices dictate that the electrical power they require be kept at a minimum.

The Low-Power Challenge

The total volume of data generated by IoT devices will reach 600 ZB per year by 2020, and developers require always-on, small form factor, low-energy technologies to tame and channel that data into useful applications.

SmartVoice's energy-saving capabilities include voice commands that can wake up one portion of a device while keeping other components asleep. These tactics alleviate battery strain and improve device usability, all while maintaining optimal device performance. Technological innovations in the communication range, reliability and energy efficiency of smart devices is why voice-enabled smart gadgets had their biggest year yet. For more information, visit www.dspg.com. 

Applications

The DBMDxx chipset from DSP Group is currently enabling many smart devices to be controlled by voice, and applications are evolving quickly for all kinds of consumer devices. For example, DSP Group's DBMDxx chipset is embedded in the latest GoPro Hero 6 camera, which can now respond to its user's verbal commands. DSP Group's SmartVoice SoC at the core of GoPro's latest smart device enables easy integration for developers with a substantially shorter development cycle, transforming virtually any device into a voice-activated solution for hands-free control.

In the past, the quality of smart devices suffered because manufacturers were forced to choose between incorporating either noise isolation capabilities or voice intelligibility, but never both. When deployed on microphone-enabled devices,

DSP Group's SmartVoice solution allows for both capabilities on a single machine, providing a stronger overall user experience for those speaking to their smart devices. DSP Group's low-power, always-on solution for voice control proves especially important for ensuring that this new functionality doesn't inhibit GoPro's biggest selling points, which includes the ability to record hands-free and the ability to keep recording for hours on end.

In addition, the SmartVoice chip's small footprint enables it to be easily embedded in the small GoPro device, as well as a wide range of devices, including tablets, smart speakers, and other microphone-equipped IoT devices.

DSP Group's SmartVoice technology offers innovative audio and voice processors for many consumer electronics that are used daily by consumers, including smartphones and wearables, in addition to IoT devices.

DSP Group and ULE

The Ultra Low Energy (ULE) Alliance (www.ulealliance.org) was formed in January 2013 by the DECT Forum, with the support of promoter companies Dialog Semiconductor, DSP Group, Gigaset, Lantiq and VTech. ULE is based on Digital Enhanced Cordless Telecommunications (DECT), which is the de-facto standard for residential and business cordless phone communications worldwide. Its primary goal is to establish ULE Technology as the world's leading technology for the wireless home and building, promoting the advantages of DECT, such as interference-free connectivity and extended range. The ULE Alliance develops the ULE Certification program, targeted to assure perfect interoperability of different ULE devices, which may be developed by different manufacturers.

ULE addresses ultra-low energy application requirements by introducing optimized communication methods. Identified with low power consumption, low latency, long range, moderate data rate and value-added complementary voice capabilities, ULE Alliance membership has grown quickly reflecting both the technical benefits of the technology and the market interest.

DSP Group meets the growing demand in the Home Area Network (HAN) market for efficient low-power, wireless solutions, supporting the ULE standard. Leveraging its long-term technological and market leadership in DECT applications, DSP Group designed the architecture for ULE and is driving its adoption, enabling development of multi-year, battery-run HAN applications.

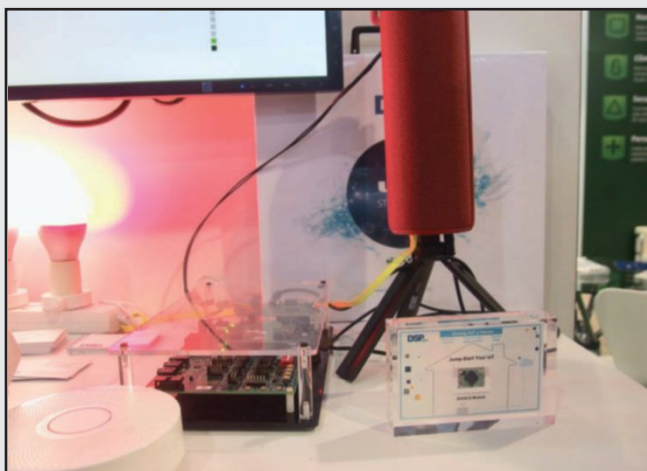
DSP Group offers an advanced ULE family of chips, as well as a comprehensive ULE and HAN FUN (Transport and Application layer) protocol, including a low-power chipset solution for home



automation and security, equipped with audio capabilities and a powerful ARM processor. DSP Group's ULE solution is based on its DHAN-S module, which uses its DHX91 chipset and is designed to work easily with even the smallest smart home nodes. The module serves as either a wireless connectivity channel for applications running on an external microcontroller unit, or as a standalone solution with the DHX91 internal ARM926 32-bit processor for smart device hibernation features and voice control support.

This System-on-a-Chip (SoC) solution has been integrated into many other types of battery-operated nodes, such as smoke and motion detectors, as well as power devices such as AC sockets. The module can serve as a wireless connectivity channel for an application running on an external microcontroller unit (MCU) or can act as a standalone solution using the DHX91's internal ARM926 processor. DSP Group's DHAN-S module offers application developers a turn-key platform for ULE applications, whether battery or AC-powered. Built around the company's DHX91 chipset, DSP Group's DHAN-S module is easily integrated into the smallest of Smart Home nodes.

The module can serve as a wireless connectivity channel for an application running on an external MCU or can act as a standalone solution using the DHX91's internal ARM926 processor. For more information, visit www.ulealliance.org.



About the Author

Ofer Elyakim brings more than 15 years of managerial experience to DSP Group. Prior to being appointed to his current position in 2009, Ofer served as DSP Group's Senior Vice President and SEA President. Before that, Ofer served as Vice President of Business Development and Director of Investor Relations and Business Development. Prior to joining DSP Group in 2006, Ofer was a research analyst covering media and broadcasting companies at CIBC World Markets in New York. Previously, he held managerial positions at Radvision, Tundo Communications, and Kost Forer Gabbay & Kasierer (a member of Ernst & Young Global). A certified public accountant, Ofer holds an MBA with honors from Columbia Business School and a BA in Computer Science and Accounting from Tel Aviv University.

From Cochlear Implants to a Personal Audio Breakthrough



In 1990, Robert Oliveira, Ph.D. (Dr. Bob) spun the Comply brand of products out of 3M.

An Interview with Robert Oliveira of Comply Foam

A serial inventor and biochemist, Robert Oliveira worked at 3M for 18 years, developing products ranging from biochemical diagnostics to surgical instruments. An ear canal expert, he also led a 3M team that developed the world's first FDA-approved cochlear implant. Oliveira's research demonstrates that the shape and opening of the ear canal changes as one's jaw opens and closes. This discovery of the ear canal's dynamic nature led to the development of numerous unique and patented memory foam solutions to protect and enhance hearing, including the world's first memory foam earphone tip.

By
Shannon Becker
(United States)



Comply Foam tips use unique patented viscoelastic memory foam technology that conforms to the shape of the ear canal at body temperature.

SHANNON BECKER: What sparked your interest in the audio field? How did it all begin?

ROBERT OLIVEIRA: A mentor at 3M asked me to turn a fledgling cochlear implant research effort into a full-fledged business. It seemed unbelievable to me that something like that was possible, but I looked into the data and talked with a deaf friend of mine—this combination told me to go forward with the opportunity—and my trek into the audio area.

SHANNON: Tell us about your work at 3M?

ROBERT: Well, I worked at 3M for 18 years starting in their Central Research Labs, their longer-range

research center and the indoctrination source for future 3M leaders. There, I learned I was a natural inventor and ended up patenting a number of biochemical diagnostic products. Next I joined 3M Surgical Products where I contributed tech insight into a business team doing healthcare acquisitions. While in Surgical, I joined management by leading a group of mechanical engineers who developed staplers to speed up surgery.

Then came a life-changing opportunity/challenge to create a global Cochlear Implant business to restore hearing in totally deaf individuals. We built a world-class team of scientists, engineers, manufacturers, and a global salesforce that secured the FDA's first approval of an implant to restore one of the five basic

senses... hearing. Though we were successful with the FDA, we were not as a business and ended up selling our technology, know-how, and patient list to our competitor to be certain our technology and implanted patients had follow-up care.

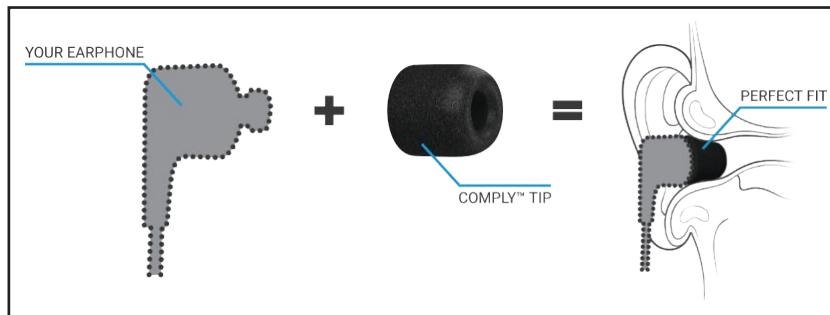
After my stint with the Cochlear Business Creation, I became one of the two Venture Directors (an intrapreneur) at 3M, each given access to \$1,000,000 (1985 dollars) and asked to create a business...of my choice. Because working with hearing implants I had learned of big needs in hearing I began the development of novel chemistry to help bring new material technology to the hearing aid industry. This led to the invention and development of the Comply Foam Tip to improve the connection between a sound delivery device and the ears with the original intent of coming out with an improved and affordable instant fitting hearing aid.

Unfortunately, this was also unsuccessful and 3M wished to shut down the business, which I found unacceptable. After an extensive negotiation, I resigned from my position and left 3M for my home basement with patent and manufacturing rights to our inventions to bring this into reality. Well again, I failed, but was successful bringing our Comply Foam Tips to the military at Ft. Rucker that found a huge need over silicon rubber tips using our Foam Tips for flight crews to improve communication intelligibility to connect to the ear in the extreme 115 dB noise background of a helicopter.

This was a success we had but didn't fully appreciate until we saw a need based on Steve Jobs' creation of bringing personal audio to millions of people, the iPod, with a need for improved physical connections to ears. In the early days of personal audio, most solutions relied on inexpensive, but poor performing rubber acoustic connectors between the ears and earphones. This turned out to be a wonderful opportunity for our team, at Comply Hearing Components, to bring our special memory foam tips into Consumer Electronics.

SHANNON: How did a Ph.D. biochemist like you get to be an ear canal expert?

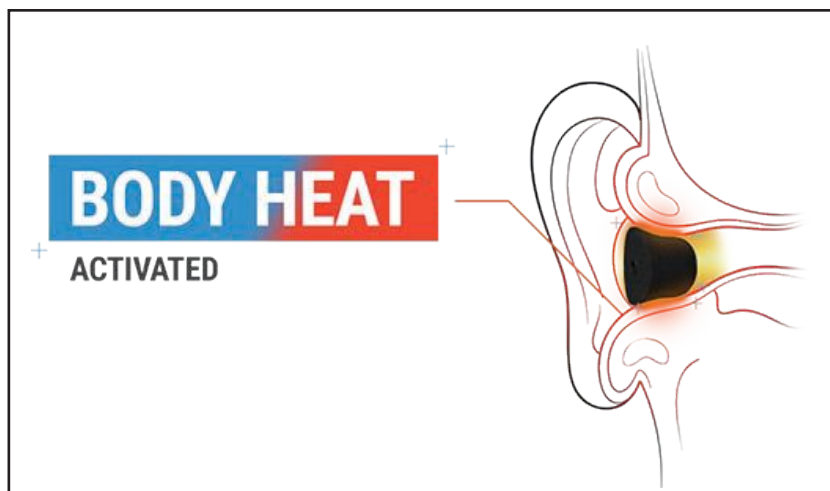
ROBERT: I've always had an inquiring mind... putting your finger in your ear and moving your jaw, suggests that there are some interesting changes occurring and demonstrates that your ear canal likely changes size and shape with common jaw motion. Thanks to our first Small Business Innovative Research grant that we got from the National Institute of Health's Deafness and other Communicative Disorders Institute, we started in my basement office wrapping string around ear



Comply Foam Tips come in different sizes and are designed to expand to fit your ear.

canal impressions to gain an idea of the amount of change in your canal between open and closed jaw positions. It happened that the first ear...mine, had a huge change in volume.

Later, we acquired laser scanners and a couple of really clever software guys who allowed us to get precise measurements of the geometric changes in the canal over a wide population of people. We not only learned that change was more common than uncommon; but that even, within one person, one ear canal volume change could be quite different from the other. This led to our work on the first magnetic resonance imaging of the ear canal changes, which really explained the biomechanics of the changes. It's neat the way it all tied together.



Rooted in Comply Foam's unique viscoelastic memory foam technology, the tips conform to the shape of the ear canal at body temperature for a dynamic custom fit. Your ear canal shape and size change with movement and temperature. And now the earplugs can change with you.

Comply Foam has developed several different replacement earbud tips.



SHANNON: What makes Comply Foam Tips (www.complyfoam.com) so special and why do earphone tips matter?

ROBERT: Our skill in chemistry allowed us to appreciate and bring other chemists into our years of continual improvement of our unique viscoelastic foam materials. This let us achieve a wonderful balance of the right ratio of hydrophilicity and hydrophobicity to give a memory foam material that breathes and becomes “invisible to feel” in your canal, because it feels like a part of your canal and allows long term wearing...many times better than commonly used, inexpensive rubber tips. Good sound isolating tips like Comply Foam Tips gives you the same sound perception at lower volume and, if desired, allows a desirable boost of low frequency sound.

SHANNON: How many different foam tips are available?

ROBERT: We have several different platforms:

- Isolation: our original tip, featuring a tapered shape. This was a result of looking at many hundreds of ear scans to determine the ideal shape for the dynamic human ear canal.
- Isolation Plus: features a WaxGuard to prevent ear wax build-up on your earphone. Extends life and reduces maintenance.
- Comfort: a spherical shape, this tip provides versatility in terms of placement in the ear – shallow or deep, up or down.... Less surface

area in contact with the ear for people who may be more sensitive to wearing earphones.

- Comfort Plus: features a WaxGuard to prevent ear wax build-up on your earphone... extends life and reduces maintenance of delicate balanced armature mini loudspeakers.
- Sport Pro: a tapered tip, features a SweatGuard that helps prevent migration of sweat and moisture into the earphone, potentially obscuring sound and ruining your device....a sport earphone life extender.
- P-Series: designed for musicians and sound professionals, this tip provides the very greatest in isolation for specific professional IEMs (e.g., Shure, Westone, Klipsch, and Etymotic).
- Truly Wireless: for the newest wave of technology. Provides a needed seal and retention for typically active wear devices. Sized to fit into earphone charging station.
- Truly Wireless Pro: features a SweatGuard

SHANNON: How many brand partnerships do you have and how are your partners selected?

ROBERT: To date, more than 80 brands have included Comply tips with their earphone platforms! These include smaller audiophile brands to many of the best-selling sport, general consumer, and truly wireless platforms. You can see most of these brands featured on our website. Ideal partners are brands who have a need for enhanced audio, comfort, and in-ear retention. Of course, most brands have this need.



To provide the perfect fit, Comply Foam earbud tips come in three different sizes to better adapt to the ear canal. The true fit seals in low frequency energy, and tips stay put.

SHANNON: Any advice for *audioXpress* readers who have not used any earphone tips but think maybe they should.

ROBERT: You gain a huge acoustic advantage with Comply Foam Tips. Not only are they barely noticeable in the ear canal, they fill about half the volume of the ear canal giving you a smaller space to send sound. This gives you the benefit of the same perception at lower acoustic gain. This half of ear canal space is occupied by a sound isolating material that dramatically improves the resulting signal to noise ratio. The net result is a tremendous benefit in terms of audio quality, external noise isolation, earphone retention and overall wearing comfort.

SHANNON: Are you currently working on or planning any new projects?

ROBERT: Research and Development are foundational to the Comply Foam product lines. We are always improving our products and developing new ways to enhance the audio experience that each earphone delivers. For example, our proprietary foams create a repeatable acoustic seal from person to person better than any other material on the market.

As for new products, we can't make any specific announcements until products are ready for launch; however, I can tell you that we watch the earphone and wearables markets closely and see wireless, true wireless, and sports/fitness trends where our Comply technology can add a significant performance advantage. 

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The TSM-100

A Turntable Speed Measurement Device

Learn how to make your own microprocessor-controlled device that measures the speed of a turntable's platter. All the device's components are housed in a small transparent plastic box and an external power adapter is used for the power supply.

Photo 1: The completed TSM-100 device is shown encased in a transparent plastic box.

By
George Ntanavaras
(Greece)

The TSM-100 was developed with the Arduino Uno hardware board and the Arduino software IDE. The device indicates the result of the measurement on a LCD with a resolution of three decimal digits. The accuracy is guaranteed by using as internal time base a crystal oscillator accurate to ± 50 PPM ($\pm 0.005\%$) with less than ± 5 PPM aging per year. **Photo 1** shows the completed device.

How It Operates

Figure 1 shows a block diagram of the device. To begin, I used a compact photo micro sensor manufactured by Omron to sense the platter rotation. It incorporates an emitter of an optical beam and a detector as shown in Figure 1. When an object passes in front of its sensing area, it reflects the optical beam of the emitter, thus changing the amount of optical energy reaching the detector. The sensor incorporates a visible light-intercepting filter, which allows objects to be sensed without being greatly influenced by the light radiated from fluorescent lamps.

For the detection of the platter rotation, a small reflective area is attached to the side of the platter, and as it spins, it reflects back the light emitted by the LED every time the reflective area is in front of the sensor. The photo transistor is triggered, turns on, and produces a short duration pulse, which is driven to a Schmitt trigger circuit to increase the noise immunity of the input signal and remove other disturbances.

The digital output of this circuit is directly driven to the microprocessor for processing. It also triggers another circuit, which outputs a low level (0 V) signal indicating to the microprocessor that a valid input was detected. Additionally it triggers a circuit that flashes a green LED, indicating to the user that a valid pulse has triggered the device, making the set up for the operation of the unit easier.

There is a large variety of platters manufactured with different shapes and from different materials. **Photo 2** shows two of the most common types of platter and how I set up the device to perform the measurement. Photo 2a shows a turntable made



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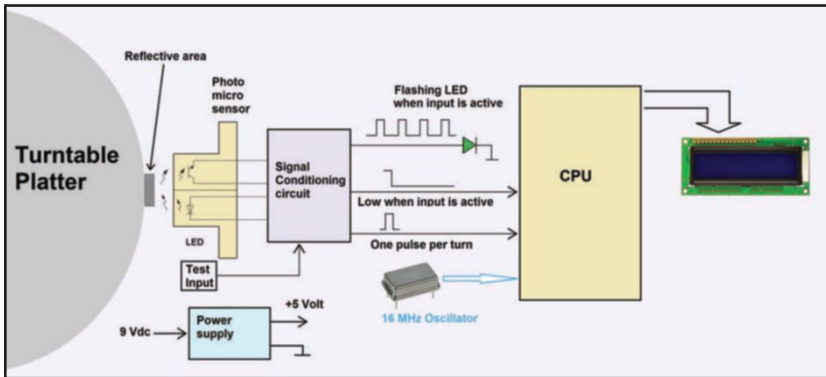


Figure 1: The block diagram of the device

from non-reflective material. In this example, I attached a 10 mm × 15 mm piece of self-adhesive highly reflective aluminum foil tape directly to the side of the turntable's platter. Then I placed the device with the photo sensor at a distance of about 5 mm to 6 mm and perpendicular to the reflective area.

Photo 2b shows a turntable's platter that is made from a more reflective material similar to aluminum, so I attached a small piece of self-adhesive black foam with dimensions of about 10 mm × 30 mm and a thickness of 10 mm to the side of the platter. I then stuck a piece of the highly reflective aluminum foil tape to the top of this foam and placed the device 5 mm from the reflective piece. With this setup, the device's sensor was 15 mm from the platter and could not be triggered until the reflective aluminum foil tape was in front of the sensor as the platter was turning around.

It is very important, in both cases, that the device remains stationary without any movement during the measurement. The rest of the processing is performed by the software running on the CPU, which then drives the LCD module to show the result of the measurement.

I wanted to increase the device's mobility by using a 9 V battery for the supply, but the

consumption was a little high and the battery life would be relatively small. So I chose an external power adapter with an output voltage of 9 VDC.

The Electronic Diagram

Figure 2 shows the completed electronic diagram for the device. The operation is as follows. The 9 VDC of the power adapter output are connected through the connector J1 to the circuit. A slow-blow fuse, mounted on the PCB, protects the external AC/DC power supply from any short circuit that could happen within the device. Diode D1 protects the circuit from an accidentally reverse polarity and capacitors C1 and C2 filter the voltage. The IC3 regulator produce a stable output voltage of +5 VDC, while capacitors C3 and C9 improve the stability of the regulator. A small heatsink for TO-220 devices is used on the IC3 regulator with a thermal resistance of 25.9 °C/W and dimensions 9.52 mm × 13.21 mm × 19.05 mm.

The four pins of the photo sensor are connected as follows: the LED emitter is connected to the pins ANODE and CATHODE and the photo transistor is connected to the pins COLLECTOR and EMITTER. The LED current is set by the parallel combination of the resistors R7 and R8 at about 23 mA. The output of the photo detector drives resistor R10 and goes to the JP2 connector, which is used to select the input between the output of the photo detector and the external signal, which is connected to pin TEST_IN and can be used to check the device's accuracy. The selected input goes to the signal conditioning circuit comprising of the components C6, R3, R4, R5, D2, D3, and IC2F. Capacitor C6 should have a large value to pass correctly the pulse from the sensor and due to its DC biasing should be a bipolar type. After the removal of any disturbances by the signal conditioning circuit, a clear digital pulse is produced each time the detector passes in front of the small

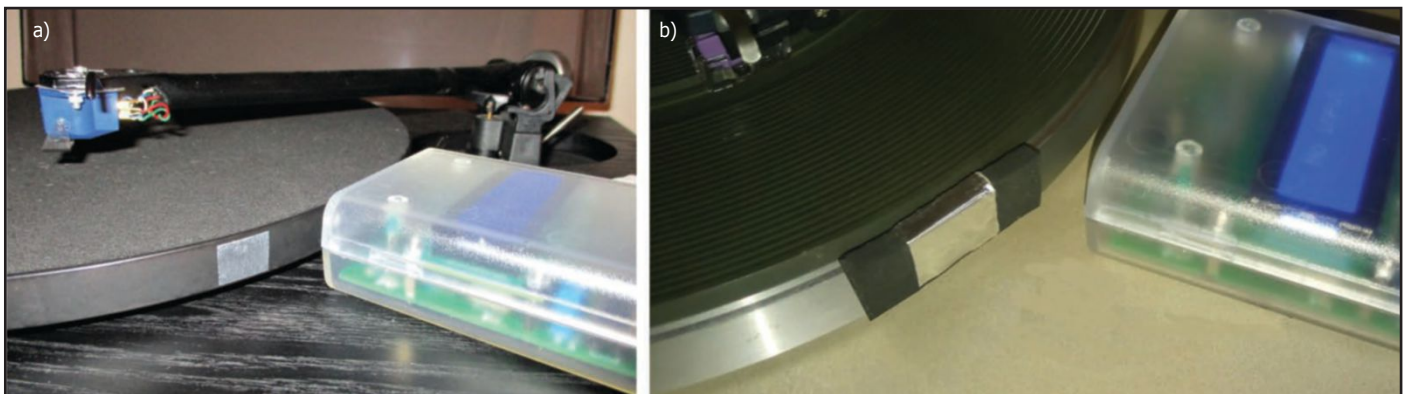


Photo 2a: This example shows the device being used with a platter made from a non-reflective material. b: I had to make a slight alteration because this platter is made from more reflective material.

reflective area, which is placed on the platter of the turntable. This pulse is inverted by IC2B and goes directly to the microprocessor Digital input pin 2.

The components IC2C, D4, R14, R6, C8, and IC2D produce a low pulse with a very long duration of about 2.4 seconds each time that the circuit is triggered by a positive output pulse by the photo detector. This signal goes to the digital input pin 7 of the microprocessor to indicate that a valid pulse has been detected. If this input pin is high, the microprocessor understands that no valid measurement exists and indicates on the display the message "No input."

In addition, the circuit around components D5, R12, C12, IC2A, and R16 oscillates with a frequency of about 6 Hz and shows to the user by blinking the green LED that the device is triggered correctly.

I used an external 16 MHz HCMOS clock oscillator (QG1) with 50 ppm accuracy to drive the CPU in order to have an accurate and stable time base for the measurement.

The microprocessor unit is the ATMEL ATMEGA328-PU. It is a low power CMOS 8 bit

microcontroller based on the AVR enhanced RISC architecture.

All the processing of the measurement is performed by the software running in the CPU. The result of the measurement is shown to the Alphanumeric LCD module, which is connected to the JP1 connector. I used a small LCD module with two lines of 16 characters and dimensions 65.5 mm × 36.7 mm × 13.5 mm. The characters are white on a blue background and white LED backlight. Trimmer R13 adjusts the contrast of the LCD.

A tactile switch S1 is also included on the device to reset it, if necessary. A pull-up resistor R1 keeps the reset input of the CPU high. Connectors IOH, AD, and IOL are not used in this device.

Assembling the Device

I designed the PCB using the Eagle Layout editor Demo version, which can be downloaded for free. It is a fully operational version, with the only limitation being the maximum dimensions of the PCB, which was not a problem on this project.

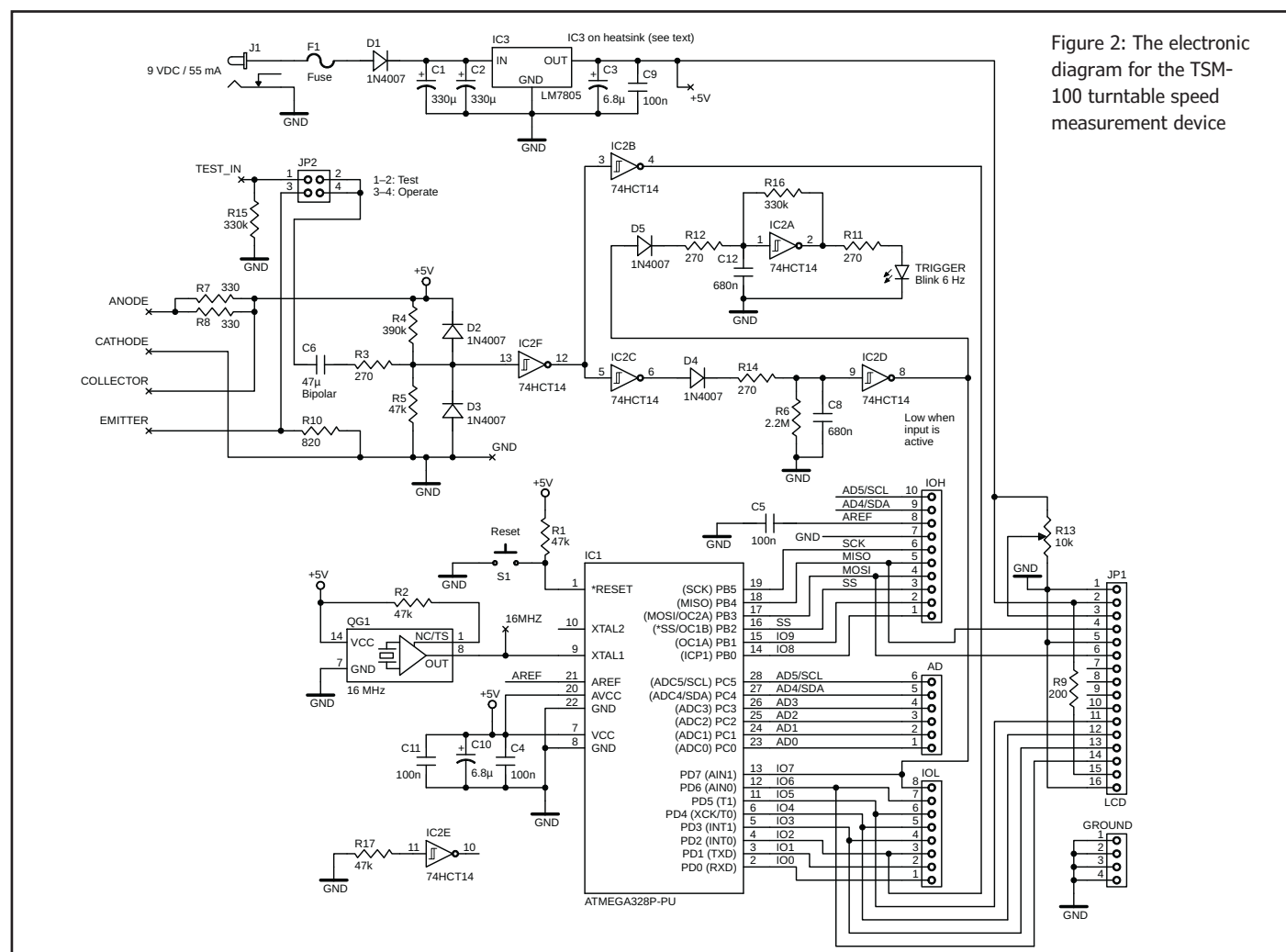


Figure 2: The electronic diagram for the TSM-100 turntable speed measurement device

I ordered a high-quality PCB manufactured from FR4 composite material with a 1.6 mm thickness and 35 µm copper solder on both sides and silkscreen on the top side.

The parts list for the TSM-100 project can be found in the Supplementary Material section of the audioXpress website (see Project Files at the

end of the article). The guide for the assembly of the components is detailed in **Figure 3**. All the components are placed on the PCB, to simplify the construction. **Photo 3** shows some important construction details. When I performed the tests, I found out that the regulator IC3 (7805) was getting a little hotter than I would prefer so I decided to put it on a small heatsink for better cooling. To ensure a safe distance from diode D1, which is very close, I put a small gap between the heatsink and the PCB by inserting an additional nut as shown in Photo 3.1. The heatsink is then placed on the top of this nut and another M3 nut secures the regulator, the heatsink and the PCB together as shown in Photo 3.2. The distance from the PCB offers also better cooling for the heat sink and lowers the temperature.

The photo sensor is placed on the PCB as shown in Photo 3.3. The procedure is as follows: First the four terminals of the sensor are bent and inserted in the holes of the PCB, then a M2x12 screw with a nut secures the sensor on the PCB, and finally the four pins are soldered.

I used a small semi-transparent plastic enclosure with dimensions 124 mm × 72 mm × 30 mm to house the PCB, so that the LCD could be read easily without any opening to the top cover of the plastic box. At the low side of the plastic box, I opened a 12 mm hole so that the connector of the external power adapter could easily enter to the device without opening the plastic box. This opening is also necessary to provide some ventilation for the internal heat that is developed in the device. The PCB is screwed in four points to the bottom of the box. Details are shown in Photo 3.4 and Photo 3.5.

I used nylon washers between the PCB of the LCD and the nuts of the stand-off that hold the LCD as shown in Photo 3.6. This insulator is necessary because otherwise the tightening of the nut may destroy the PCB tracks of the LCD that are very close.

When the device was assembled, I powered it on and measured all the supply voltages to verify that they were right according to the electronic diagram. On the first power on, the LCD is blank and with the trimmer R13, the contrast should be adjusted, until the alphanumeric characters are easily read. The jumper JP2 should be placed on the position 1-2 (Test) for the testing of the device or to the position 3-4 (Operate) for the normal operation. For the testing an accurate low frequency square wave of around 0.5 up to 1 Hz at a level of 5 V should be connected to the TEST_IN and GND pin. The frequency of the signal multiplied by 60 will be indicated on the LCD. For example, connecting an accurate 0.5 Hz square wave gives $0.5 \times 60 =$

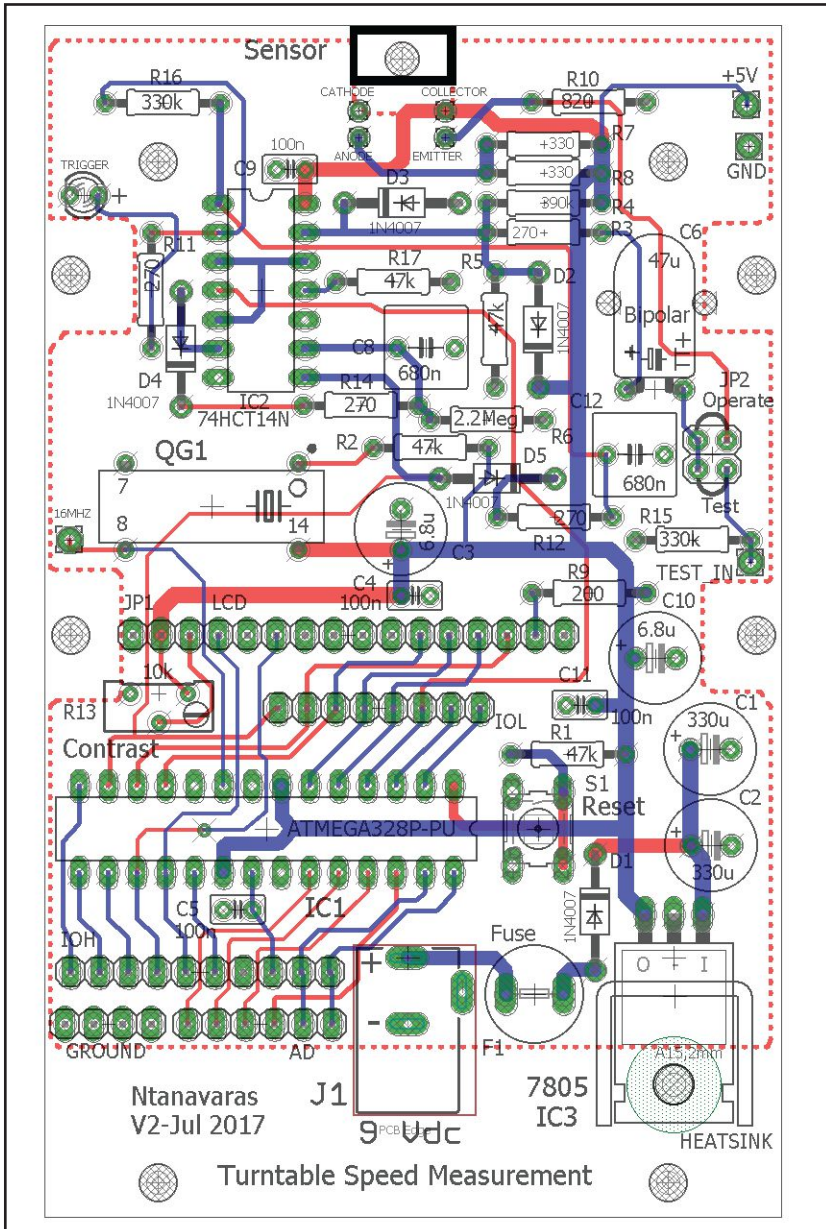


Figure 3: This is the PCB assembly of TSM-100 device.

About the Author

George Ntanavaras graduated from the National Technical University, Athens, Greece, in 1986 with a degree in Electronic Engineering. He currently works in the Development Department for a Greek electronics company. He is interested in the design of preamplifiers, active crossovers, power amplifiers, and most loudspeakers. He also enjoys listening to classical music.

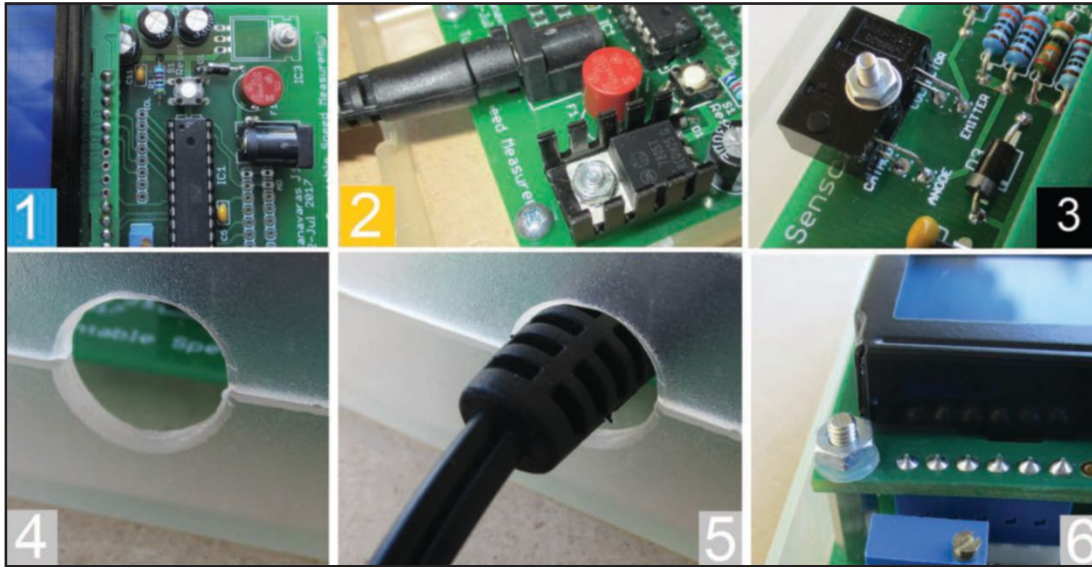


Photo 3: Important construction details are depicted.

30 RPM, and the display will indicate this with three decimal digits as 30.000 RPM.

The Device's Software

I used the Arduino Uno REV 3 hardware board during the development phase of the project. This is a microcontroller board based on the ATMEGA328P. It has 14 digital input/output pins, six analog inputs, a 16 MHz quartz crystal, a USB connector to a PC, a power jack, an ICSP header, and a reset button. It contains everything needed to support the microcontroller. It is connected to the computer with an USB cable, which also will power the board or otherwise can be powered with an AC-to-DC adapter or even a battery.

For the software development, I used the Arduino Integrated Development Environment (or Arduino Software IDE). This supports a text editor for writing the code, a message area, a text console, a toolbar with buttons for common functions, and a series of menus. It is an easy tool, aimed at those without a background in programming.

I wrote a simple software code for the operation

of the device. It calculates the "rpm" for each revolution of the platter and the result is shown to the LCD with a resolution of three decimal digits.

Figure 4 shows how it operates. Each time the reflective area is in front of the sensor, a short pulse is produced. The rising edge of this pulse triggers a CPU interrupt to run a short subroutine that uses the Arduino "micros()" function, which returns the number of microseconds since the CPU start running the program. This function is very convenient but it has some shortcomings. The returning number will overflow after approximately 70 minutes of operation going to zero value and has a resolution of 4 μ s since the returned value is always a multiple of 4. Both of them are not serious problems for this application because the measurement of the "RPM" is performed in every turn of the platter and the result is shown with only three decimal digits.

The calculated time difference T between two successive readings is the period for one revolution and for the most common speeds of 33 1/3 and 45 RPM, the period T is 1.8 s and 1.33 s, respectively. These time periods are very long and

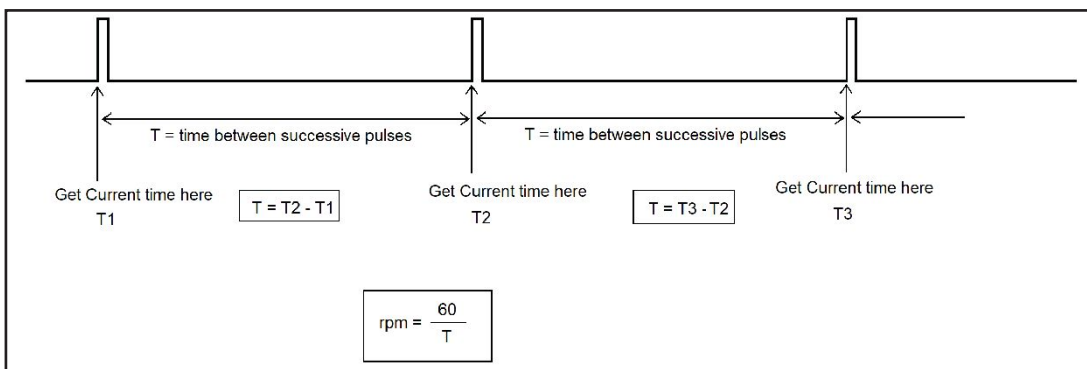


Figure 4: Operation of the software

```
//
//          Turntable Speed Meter .....V1.0
//          by George Ntanavaras
//
#include <LiquidCrystal.h>
LiquidCrystal lcd(12, 11, 5, 4, 3, 6); // LCD pin connections
volatile unsigned long Cur_Time;      // Arduino Current time in micro seconds
volatile long Ttime;                  // Storing total time of a cycle
volatile unsigned long Ttime1;        // Storing start time of a cycle
volatile unsigned long Ttime2;        // Storing stop time of a cycle
volatile int meas ;                   // number of successive measurements
volatile float frequency;             // Storing frequency
volatile float rpm;                   // Storing rpm
volatile boolean trigger;
volatile boolean in_active ;
//-----
void isr ()
{
    Cur_Time = micros();
    meas=meas+1 ;
    if (trigger) Ttime1 = Cur_Time ;    // get end time
    if (!trigger) Ttime2 = Cur_Time ;   // get start time
    trigger = !trigger ;
}
//-----
void setup()
{
    trigger = false ;
    Ttime1 = 0 ;
    Ttime2 = 0 ;
    init();
    meas =0 ;
    pinMode(2, INPUT);
    pinMode(7, INPUT);
    attachInterrupt(0, isr, FALLING);
    lcd.begin(16, 2); // set LCD = 16 char by 2 lines
    lcd.clear();      // clear the whole LCD
    lcd.display();
    delay(1200);
    lcd.clear();      // clear the whole LCD
    lcd.setCursor(0,0); // set cursor to the first character, top line
    lcd.print("TSM-100 V1.0"); // label the LCD top line
    delay(1200);
    detachInterrupt(0);
    if (trigger) Ttime1 = Cur_Time ; // get end time
    if (!trigger) Ttime2 = Cur_Time ; // get start time
    attachInterrupt(0, isr, FALLING);
    lcd.clear();      // clear the whole LCD
    lcd.setCursor(0,0); // set cursor to the first character, top line
    lcd.print("Turntable Speed"); // label the LCD top line
    lcd.setCursor(0,1); // set cursor to the first character, second line
    lcd.print(" Meter"); // label the LCD top line
    delay(1200);
    detachInterrupt(0);
    if (trigger) Ttime1 = Cur_Time ; // get end time
    if (!trigger) Ttime2 = Cur_Time ; // get start time
    attachInterrupt(0, isr, FALLING);
    lcd.clear();      // clear the whole LCD
    lcd.setCursor(0,0); // set cursor to the first character, top line
    lcd.print("by");
    lcd.setCursor(0,1); // set cursor to the first character, second line
    lcd.print("G. Ntanavaras");
    delay(1000);
    detachInterrupt(0);
}
```

(continued)

Figure 5: Here is a listing of the software code that I wrote.

Figure 5: continued

```

    if (trigger) Ttime1 = Cur_Time ; // get end time
    if (!trigger) Ttime2 = Cur_Time ; // get start time
    attachInterrupt(0, isr, FALLING);
    lcd.clear(); // clear the whole LCD
    lcd.print("Wait ");
    lcd.setCursor(0,1); // set cursor to the first character, second line
    lcd.print(" to measure ...");
    delay(2000);
}
//-----
void loop ()
{
    in_active = digitalRead(7); // Check if there is a valid input to measure
    if (in_active == LOW)
    {
        if (meas < 2 ) return ; // If there is not a new measurement, return
        //
        // New measurement
        //
        Ttime = Ttime1-Ttime2 ; // Calculate period in microseconds
        Ttime = abs (Ttime) ;
        frequency=Ttime+88 ; // Compensate to improve accuracy
        frequency=1000000 / frequency; // calculate frequency from Ttime
        rpm=60*frequency; // calculate RPM from frequency
        // Display the readings
        lcd.clear(); // Clear the whole LCD
        lcd.setCursor(0,0); // Set cursor to the first character, first line
        lcd.print (rpm, 3);
        lcd.print (" RPM");
        meas=1; // Prepare for the next measurement
    }
    else
    {
        lcd.clear(); // clear the whole LCD
        lcd.print (" No input ");
        delay(100);
    }
}
// end of loop and program
//-----
//-----

```

the microprocessor unit can easily measure them with high accuracy.

The frequency is calculated as the inverse of the period: $\text{frequency} = 1/T$.

The revolutions per minute (RPM) of the turntable's platter are calculating them as $\text{RPM} = 60 \times \text{frequency}$.

The selection of the software commands within the interrupt routine was very critical and after a lot of experimentation, I managed to have a measurement with high stability and repeatability. A small constant compensation (equal to 88) was added to the computed time for better accuracy.

The software has excellent measurement accuracy for frequencies up to 1.4 Hz (84 RPM).

The software should be downloaded to the ATmega328P CPU using the Arduino Uno board since the TSM100 device does not have such capability. After the programming, the CPU is removed from the Arduino Uno board and placed on the IC1 socket

of the PCB of the device.

A complete list of the software code is shown in **Figure 5**. It has the typical structure of an Arduino program, which is called the "sketch." It starts with the declaration of the variables that are used by the program. The short interrupt routine "void isr ()" then follows with the "micros ()" commands. The main program starts with the "setup ()" which is executed only once when the program starts to initialize the parameters and continues with the "loop ()" which runs continuously, computes the RPM, and indicates the result to the LCD module (see **Photo 4**).

The Reference Oscillator

The accuracy of the measurement depends strongly on the software and during the development phase I was looking for an easy way to check the various versions of the software and adjust them to perform as accurately as possible. For this reason, I

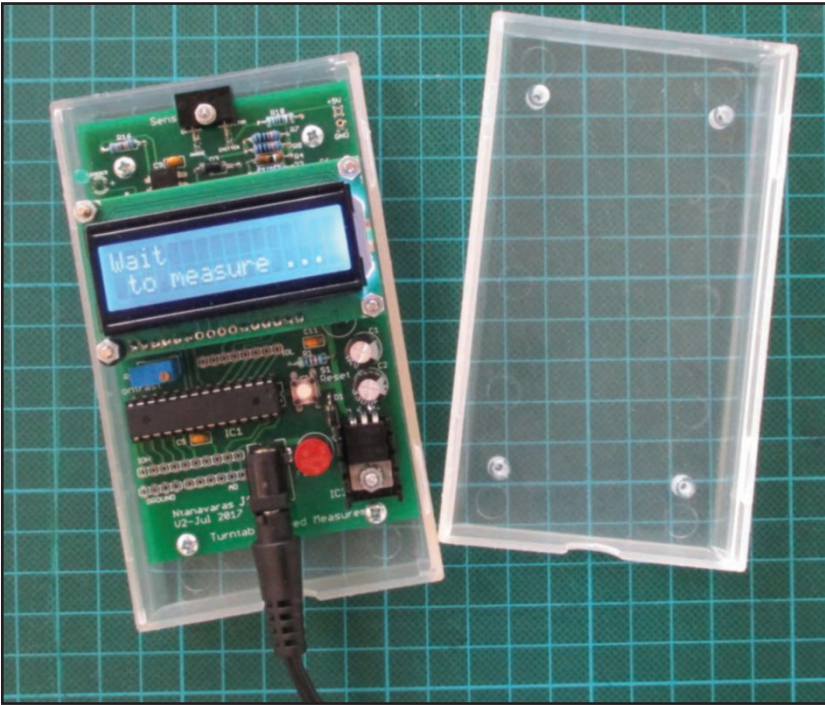


Photo 4: This is an internal view of the TSM-100 device.

designed a special unit with an accurate and stable clock output which I called it "Reference Oscillator."

The unit is based on an Oven-Controlled Crystal Oscillator manufactured by Abracon (AOCJYR-10.000MHZ-M5625LF). It is an expensive SMD component, with a ± 0.5 ppm initial frequency tolerance at 25°C and a stability as low as ± 25 ppb (note that it is parts per billion, not parts per million) over the temperature range from -40 to +85°C.

Figure 6 shows the completed electronic diagram. The output of the 10.000 MHz Oven Controlled Crystal Oscillator is at a level of 3.3 V and the MC74VHC1GT04 CMOS buffers and shifts this level to 5 V so that the rest of the circuit can operate smoothly. The 10 MHz clock is driven to eight stages of dividers. The seven of them divide by 10 while the last one divides by 2. The output of each divider is connected to a 12-position switch, which selects the signal that will be sent to the output connector from one of the following frequencies: 10 MHz, 1 MHz, 100 kHz, 10 kHz, 1 kHz, 100 Hz, 10 Hz, 1 Hz, and 0.5 Hz.

The selected signal is buffered by another MC74VHC1GT04 buffer inverter before it is connected to the output connector.

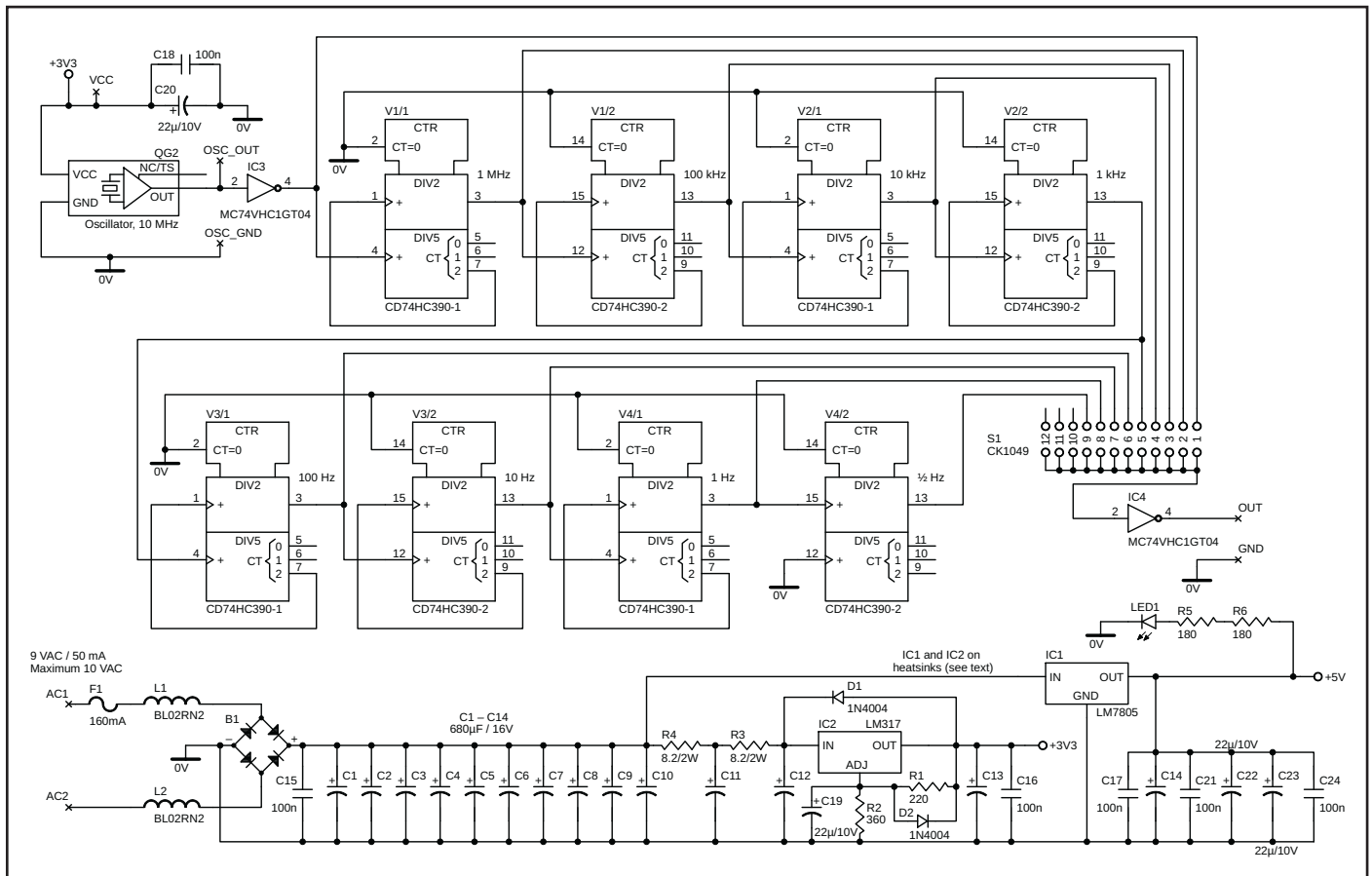


Figure 6: The electronic diagram of the Reference Oscillator

For the supply of the oscillator, I used a Class B, external plug-in AC/AC power supply. Its mains input is protected with a 130°C thermal fuse and has a 9 VAC/10 VA secondary output which through the front panel connector goes to pins AC1 and AC2 of the circuit.

A slow-blow fuse, mounted on the PCB, protects the external AC/AC power supply from any short circuits that could happen within the device. L1 and L2 are ferrite beads inductors that have small impedance at low frequencies, which increases to a high impedance at high frequencies. They offer electromagnetic induction (EMI) protection from several megahertz up to gigahertz by suppressing any noise coming from the power supply.

The AC voltage is rectified by the bridge B1 and filtered by the capacitors C1 to C10. Resistors R3, R4 and capacitors C11, C12 form low-pass filters to smooth the ripple of the input voltage before it enters to the 3.3 VDC regulator. Two voltage regulators are used, a 7805 for the 5 VDC and a LM317 for the 3.3 VDC. The regulators IC1 and IC2 required some cooling and I used two small heatsinks to keep their temperature low.

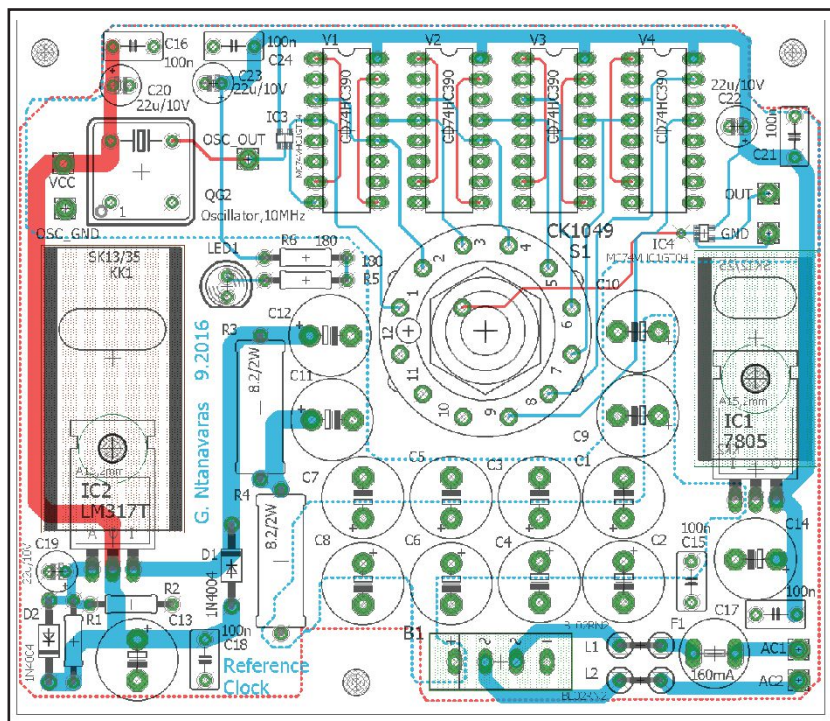


Figure 7: Here is the PCB assembly of the Reference Oscillator.

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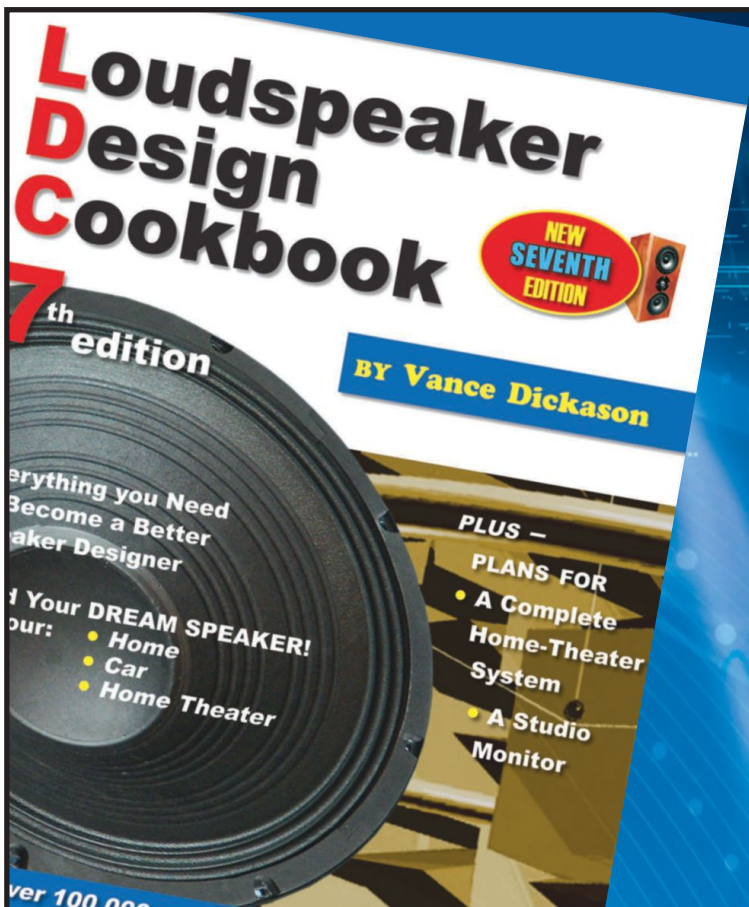
Photo 5: This is a top view of the Reference Oscillator's completed PCB.

Additional supply filtering is offered by capacitors at the output of the regulators and near the Oven-Controlled Crystal Oscillator and the other ICs. An LED illuminates to indicate the operation of the circuit.

I designed another PCB for the oscillator using the Eagle Layout editor Demo version and I ordered a prototype board without any silk screen to assembly the circuit.

The parts list for the Reference Oscillator can be found in the Supplementary Material section of the audioXpress website (see Project Files at the end of the article). The guide assembly is detailed in **Figure 7**, while **Photo 5** shows the final assembled PCB.

The Oven-Controlled Crystal Oscillator that I finally used was different from the initial one I had selected when designing the PCB, and for this reason, the new oscillator was connected to the PCB with short lengths of wire as shown in the left top corner of Photo 5. I used two small heatsinks for each regulator. For the LM317T regulator, I used a heatsink with a temperature constant of 21°C/W and with the dimensions 6.7 mm x 22 mm x 35.6 mm. For the 7805 regulator, I used a heatsink with a temperature constant of 25.9 °C/W and with the dimensions 9.52 mm x 13.21 mm x 19.05 mm.



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Photo 6:
This is the
completed
Reference
Oscillator.

I used an aluminum enclosure to house the PCB and the connectors. Using the "Front Panel Designer" program, I also designed a front panel for the unit. The result was a more professional-looking device that was also easy to operate. I attached the PCB to the front panel using as support the switch, which I placed at the center of the PCB. A BNC connector is used for the output of the oscillator. **Photo 6** shows a view of the completed reference oscillator.

Conclusion

I have found the turntable speed measurement device to be a useful piece of test equipment. It is easy to build it, the cost is reasonable, it measures with excellent accuracy, and it is simple to use. If you don't have this piece of equipment in your lab, here is a good opportunity to build one. 

Author's Note: I have a small quantity of PCBs for the construction of the TSM-100 device. Also I can provide a programmed CPU with the proper software. If you are interested, send me an e-mail at gntanavaras@gmail.com

Project Files

To download the TSM-100 Parts List and the Reference Oscillator Parts List, visit <http://audioxpress.com/page/audioXpress-Supplementary-Material.html>

Sources

Oven-Controlled Crystal Oscillator
Abracon | www.abracon.com

Arduino software
Arduino | www.arduino.cc/en/Main/Software

Eagle PCB design software
Autodesk, Inc. | www.autodesk.com/products/eagle/overview

Front Panel Designer
Schaeffer AG | www.schaeffer-ag.de

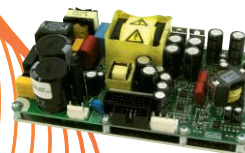
hypex NCxxxMP series

overview

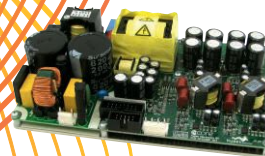
The NCxxxMP amplifier module incorporates a low power standby power supply (meets 2013 ERP Lot 6 0.5W requirements), a highly efficient switch mode power supply and a high-performance Class D amplifier in one compact and easily applicable power brick.



NC122MP



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NC502MP

Add-on Module NC100HF
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Transformers for Tube Amplifiers (Part 2)

Rating and Building the Circuitry

In Part 1 of this article series we discussed output transformers and their fundamental properties. In Part 2, we discuss the ideal core sizes and core types. We also share our thoughts about the electrical performances needed for a good transformer.

By
Gerhard Haas
(Germany)



In the 1950s, the maximum magnetic resilience, also called the saturation limit, often was at around 1.2 T (Tesla), which corresponds to 12,000 Gauß. In old literature, the magnetic load was expressed in Gauß, the new designation is expressed in Tesla, 1 T = 10,000 Gauß. If an iron core is magnetically saturated, there is no use in applying even more current to the windings. When all iron particles are magnetized, an increased coil current intensifies the stray field and the transformer gets hot.

In the past, a load of 0.4 T was recommended for good low-frequency transformers in amplifiers without negative feedback and up to 0.6 T for those with negative feedback. With a transformer, you want to be far from the saturation limit of the iron and, above all, not have it register near the bent part of the characteristic of the magnetic modulation curve shown in **Figure 1**. When reaching the bent part of the characteristic curve with a high modulation of the transformer, odd harmonics are generated, especially the third harmonic.

The induction depends on several factors. It increases with rising field strength H (A/m = ampere per meter). The higher the number of windings

at a given current, the lower the field strength. If the field strength becomes too high, the core goes into saturation. Metaphorically speaking, an iron core made from a certain alloy with a fixed iron cross-section can only handle a finite number of lines of force.

When calculating the induction, the peak value $\hat{B}$ is crucial, not the effective value. It is calculated using the transformer formula:

$$\hat{B} = \frac{U_{eff}}{\frac{2 \times \pi}{\sqrt{2}} \times f \times w \times A_{Fe}}$$

where several factors are considered. The factor:

$$\frac{2 \times \pi}{\sqrt{2}}$$

contains a portion of the angular frequency $\omega = 2 \times \pi \times f$ by means of $2 \times \pi$ and with $\sqrt{2}$ it also contains the relationship between effective value and peak value of the input voltage. When calculating this factor, you get the rounded value of 4.44, which is used in the transformer formula for convenience only. Now, the input voltage as an effective value can

be used for U_{eff} . The frequency f is entered in Hertz, the iron cross-section A_{Fe} in cm^2 (cm = centimeter). With an installed transformer, the number of windings w and the iron cross-section A_{Fe} are fixed. When looking at the maximum output power with the typical reference frequency in low-frequency technology of 1 kHz, it is possible to determine the input voltage U_{eff} , before the amplifier reaches the clipping area. Starting from this operating point, the frequency is lowered.

Since the frequency f in the formula is in the denominator and everything else is constant, the induction increases with decreasing frequency. As the induction increases with decreasing frequency, the core goes into saturation because of the flattening curve as from the point "S" in Figure 1. The sine signal visibly deforms (as shown in Figure 7 from Part 1 of this article series). If you want to avoid these distortions, the input voltage needs to be lowered, which also lowers the induction. This, however, is at the expense of the achievable output power. With the negative feedback of the amplifier, the distortions can be somewhat concealed, but the fundamental problem remains. A different winding, better core qualities, and a bigger iron cross-section will produce improvements, however, there are both physical and financial limits.

Differing Formulas

There are several formulas for determining the core size, which lead to very different results. To transmit the amplifier power, a certain iron cross-section is needed, which is then used to determine the core size. Using a mains transformer, which is operated only with 50 Hz or 60 Hz depending on the layout, you can approach the saturation limit, but it is absolutely essential to take a possible mains overvoltage and peak loads into account. Many cheap or incorrectly dimensioned transformers of all types work so close to the saturation limit that they get hot and can blow if there is a low mains overvoltage and/or additional load. However, they also generate a very strong stray field.

The size of the iron cross-section depends on the power output to be transformed as well as on the desired lower limit frequency. For example, if the lower limit frequency—instead of 50 Hz or 60 Hz as it is with the mains transformer—is halved and the primary winding number remains the same, the magnetic load is doubled. To avoid distortions, the number of windings or the core cross-section need to be doubled or a combination of both approaches need to be applied. Furthermore, core materials featuring a higher magnetic resilience can be used. In order to transmit the lower limit

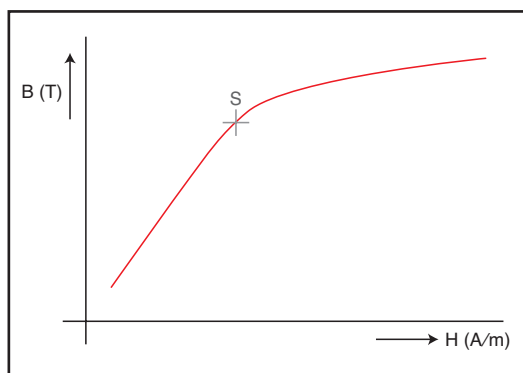


Figure 1: Here is the magnetic modulation curve.

frequency without distortions, the iron (depending on its quality), must be modulated only to a third up to slightly over one half, otherwise you get into the bent part of the hysteresis curve.

Figure 2 describes how a typical hysteresis curve evolves with periodic modulation. The core is magnetically saturated as from the B_s line (B is the magnetic induction, the index "s" stands for the peak induction). With medium modulation, the characteristic curve is relatively linear and becomes more and more bent as the modulation increases. Then, more odd harmonic distortions, predominantly the third harmonic, will emerge, because a transformer is an unbalanced system. Unbalanced systems (e.g., a triode) preferably generate second harmonics. So, if the harmonic distortion of a single-ended power amplifier is measured at high modulation and there is a higher percentage of second harmonics, this harmonic distortion is caused only by the tubes, preferably if they are triodes.

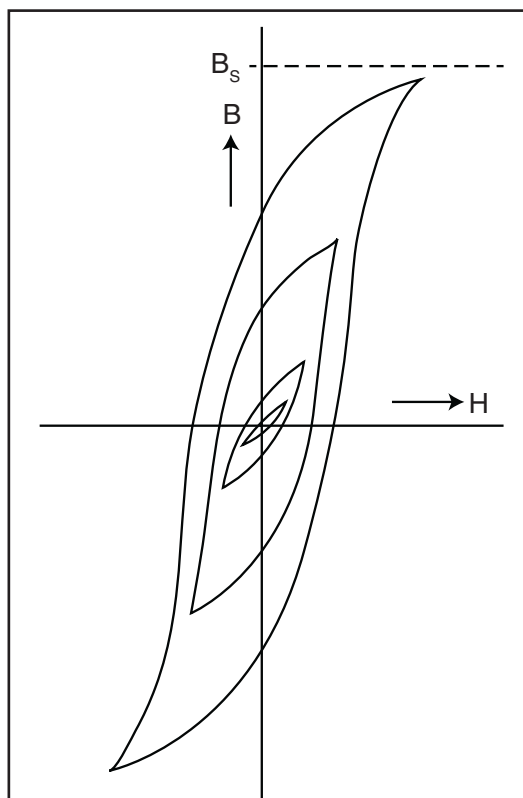


Figure 2: This details how a typical hysteresis curve evolves with periodic modulation.

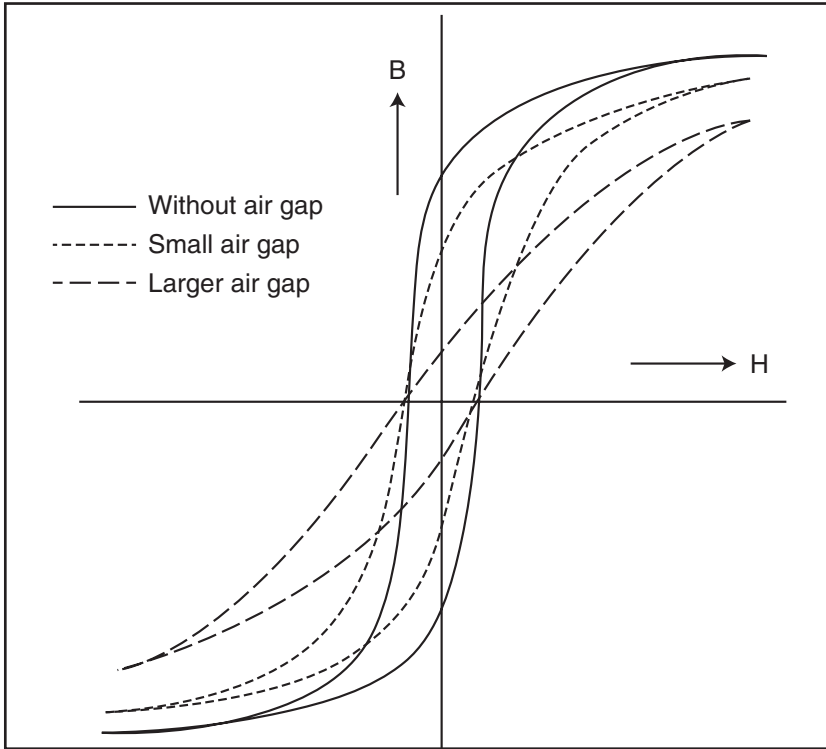


Figure 3: If, in a single-ended type-A power amplifier, an air gap needs to be integrated because of the DC bias and as a magnetic negative feedback, the curves flatten even more.

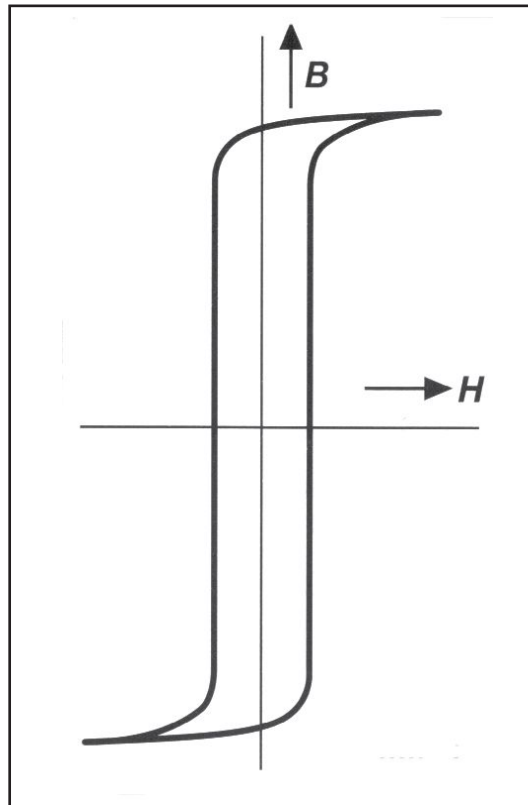


Figure 4: This is a hysteresis loop of another material.

When using triodes in push-pull AB circuits, the second harmonic more or less cancel each other and the third harmonic can become dominant again. This destroys the “triode sound.” If, in single-ended type-A power amplifier, an airgap needs to be integrated because of the DC bias and as a magnetic negative feedback, the curves flatten even more (see **Figure 3**).

Figure 4 shows a hysteresis loop of another material. Here, you have a far bigger dynamic range and you can go substantially closer to the magnetic saturation limit than is possible with the material shown in Figure 2.

Let’s take a look at the Williamson amplifier, which was released in 1947 and is considered the first real hi-fi amplifier. With two KT 66 tubes, which is the English equivalent to the EL 34 or the American 6L6, a power output of only 15 W was achieved. The EI 150 N cut was used as the core. Let’s take a hi-fi push-pull AB power amplifier with 2 × KT 88 and a power output of 50 W. The lower limit frequency f_u shall be 20 Hz.

Now, we use four formulas to determine the required core dimensions. The first formula determines the core size via the power output to be transmitted and via the lower limit frequency according to the weight. The three other formulas refer to the iron cross-section because it is used to calculate the magnetic load for the iron, but not the weight of the iron! The power and the lower limit frequency are considered once, the power to be transmitted is considered only twice.

Minimum weight of the iron core in kilograms (kg):

$$G \geq \frac{7.5 \times P}{f_u} \quad (1)$$

Minimum cross-section A of the iron core in cm²:

$$A_1 \geq 10 \times \sqrt{\frac{2 \times P}{f_u}} \quad (2)$$

$$A_2 \geq 4 \times \sqrt{P} \quad (3)$$

$$A_3 \geq 1.6 \times \sqrt{P} \quad (4)$$

For clarification, we used the following formula and table symbols:

A = iron cross-section in cm²

B = magnetic load in T (Tesla, 1 Tesla = 10,000 Gauß)

G = iron weight in kilograms (kg)

P = amplifier power to be transmitted in watts (W)

f_u = lower limit frequency in Hertz (Hz)

The factors 1.6, 4, 7.5, and 10 are experience values, however, they are used somewhat randomly.

Now, let's calculate the core dimensions using the defined requirements and formulas for our sample amplifier with 50 W.

$$G \geq \frac{7.5 \times P}{fu} = \frac{7.5 \times 50}{20} = 18.75 \text{ kg}$$

$$A_1 \geq 10 \times \sqrt{\frac{2 \times P}{fu}} = 10 \times \sqrt{\frac{2 \times 50}{20}} = 22.4 \text{ cm}^2$$

$$A_2 \geq 4 \times \sqrt{P} = 4 \times \sqrt{50} = 28.3 \text{ cm}^2$$

$$A_3 \geq 1.6 \times \sqrt{P} = 1.6 \times \sqrt{50} = 11.3 \text{ cm}^2$$

Then, we implement the results in some common core dimensions and compare them with the modern hi-fi transformer B-434 S featuring the MD 85 B core. **Table 1** shows the differences. B_{max} = the core's maximum magnetic load measured in Tesla at 50 Hz as mains transformer. Uniformly hot-rolled, low-loss M111-035-type silicon iron is used as the core material (i.e., the sheet is 0.35 mm thick and the iron features hysteresis losses of 1.11 W/kg, which

A	G	Core	Core Dimensions/ Package thickness in millimeters (mm), Dimensions a x b with M, Dimensions a x b + f with EI	P _{max} as mains transformer at 50 Hz	B _{max} T
66.1 cm ²	19.4 kg	EI 192 F*	192 x 160/110	2240 VA	1.60 T
47 cm ²	18.6 kg	EI 231 B	231 x 209/97.5	3320 VA	1.73 T
31 cm ²	8.6 kg	EI 170 C	170 x 140/7.45	1370 VA	1.73 T
22.2 cm ²	5.3 kg	EI 150 C	150 x 120/60	780 VA	1.73 T
22 cm ²	4.6 kg	EI 135 B	135 x 112.5/64	491 VA	1.65 T
16.5 cm ²	3 kg	M 102 B	102 x 102/5.4	271 VA	1.66 T
17.2 cm ²	2.5 kg	EI 96 C*	96 x 80/58	223 VA	1.63 T
8.5 cm ²	2 kg	MD 85 B	85 x 102/45.5	160 VA	1.72 T
69.5 cm ²	16 kg	EI 150 N#	150 x 120/150	1540 VA	1.61 T

Table 1: Here are the differences when we implement the results in some common core dimensions and compare them with the modern hi-fi transformer B-434 S, featuring the MD 85 B core.

is the best sheet quality possible). The core next to the calculated value was chosen and sorted by weight. The EI-type cores labelled * are of normal N-type cut and therefore waste-free, the other EI-type cores are of low-waste cut. In comparison, the core labeled # at the end of the table is that of the Williamson amplifier.

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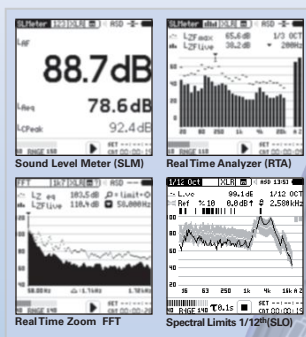
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NTI AUDIO

B-434 S Transformer	
Core	MD 85 B
Frequency response at 60 W	17 Hz (-0.2 dB)...64 kHz (-1 dB)... 105 kHz (-3 dB)
Primary inductance	>189 H at 100 mW
Stray inductance	6 mH
Raa	1.9 kΩ
Secondary impedances	4 Ω, 6 Ω, 8Ω
Screen grid tapplings	25% and 50%
Copper resistance of the primary winding	182 Ω
Dimensions	
Core dimensions	85 mm × 102 mm
Core package thickness	47 mm
Height above coil body	75 mm
Weight	3.1 kg

Table 2: The data of the hi-fi transformer B-434 S demonstrates what can be achieved using the highest-quality core material and applying a good winding technique.

Table 2 shows the data of the hi-fi transformer B-434 S, which demonstrates what can be achieved using the highest-quality core material and applying a good winding technique.

Now the question is why are there formulas that lead to such big cores. As can be seen in Table 1, three formulas lead to transformer cores that are uneconomical, too big, and too heavy. In the past, it was not really necessary to have a lower limit frequency of 20 Hz, which could be transformed without any noteworthy waste. However, in order to obtain useful transformer dimensions from this calculation half a century ago, a lower frequency of 40 Hz (-1 dB) or even 50 Hz (-3 dB) was assumed, immediately resulting in significantly smaller transformers.

In radiograms and radio receivers of that time, one will mostly find relatively small output transformers. In this case, the lower limit frequency has been reached to some extent using a high number of windings and a small wire diameter.

About the Author

Gerhard Haas has written books as well as articles for *Elektor* Germany since 1995. *The Roehren Sonderheft* magazine was his creation, along with some partners. The first issue came out in 2005. Now, there are 10 issues containing a collection of interesting articles. Gerhard has also developed many electronic circuits and equipment. In the last four decades, he primarily worked on tube amplifiers and the circuitry around them, including the audio transformers, filter chokes, power transformers. He sold his company, Experience Electronics, several years ago to focus on designing tube amps and audio transformers, write articles for *Sonderheft* and finish his book *High-End Mit Roehren*. The first copy of his book was printed in 1995. The follow up was published in March 2017.

The high copper resistance was accepted, because with an operating current of a tube measuring 50 mA and a copper resistance of 400 Ω, a voltage drop of 20 V occurs causing a copper loss of only 1 W. To compensate, the operating voltage can be increased accordingly. Furthermore, these transformers often show a drop of 3 dB at 100 Hz. That was perfectly acceptable, since the mains frequency of 50 Hz or 60 Hz as well as the hum frequency of 100 Hz or 120 Hz caused by the bridge or the two-way rectifier would be slightly attenuated.

In many of those devices the single-ended output transformers still have a primary tap to which the preamplifiers are connected. In this case, the transformer inductance and the air gap are utilized; consequently, the output transformer additionally acts like a filter choke. Furthermore, due to the inversely flowing currents inside the primary winding, the DC bias is partially compensated, therefore, the core can stay small. In order to compensate for the audible bass drop, the treble range must also be attenuated.

In this case, the formula 400,000 mentioned in various papers between 1947 and 1970 is applied. When multiplying the ends of the audio range of 20 Hz and 20 kHz, this value is obtained. If there is a drop of 6 dB at 50 Hz, at 400,000: 50 Hz = 8000 Hz, the drop must be similar to achieve a balanced and pleasing sound. If this drop does not occur, at the upper end of the frequency band, the audio playback will sound too sharp. This explains the good sound of old tube radios despite the presence of small output transformers.

With a VHF transmission, which was at that time considered the best audio transmission, the frequency range in the technical specifications of the broadcasting stations had been set at 40 Hz....15 kHz (+0.5 dB/-2 dB), and that also applied to the possible dynamic range of the high-frequency transmitter and its allowed channel width. With a VHF station, the low-frequency band needs to be steeply limited at 15 kHz in order to avoid a collision between the low frequency and the 19 kHz stereo pilot signal. Therefore, it was sufficient if the low-frequency amplifier inside the receiver could just deal with the 15 kHz.

Today, these techniques are unsuitable for high-end playback. A modern single-ended hi-fi transformer with 7 W features a core size several times larger than those in former radiograms and should be capable of dealing with 20 Hz up to about 50 kHz without any problem. Vinyl records, as well as CDs have their upper limit frequencies. With a CD, rigorous filtering occurs at about 22 kHz, otherwise

the entire system would not work. So, there is a requirement that a low-frequency amplifier or transmitter must not show any noteworthy drop at this frequency. If the bandwidth is considerably wider, this ensures that at 20 kHz there still is no noteworthy phase shift. This is especially important with negative feedback amplifiers because otherwise there may be signal deformations and oscillating tendencies.

Fifty years ago, there wasn't the same sheet quality of today, the sophisticated winding techniques using modern computerized winding machines, the isolating plastic films, or synthetic resins for vacuum impregnation. The developers counted on safety consuming a lot of material. EI-type cuts were used preferentially, since they existed and still exist in various versions. It is virtually the historical transformer core cut with the most variants. Furthermore, the common transformer sheets normally could often be modulated only up to 1.2 T. Sophisticated laminations, as well as grain-oriented material with a considerably longer and straight modulation curve, still had to be developed. This is the explanation for the compact design and the high-performance of modern transformers.

By comparing laminations, it is clear how best results can be achieved by applying an intelligent use of material, combined with the best sheet alloys, grain-oriented material, and modern processing technology. Thanks to a sophisticated winding technology with most modern winding machines, the winding space can be used optimally. The winding technique from that time featured significant tolerances because the wire was not optimally guided. Insulation risks were mitigated by applying several layers of enamel paper. Enamel paper has been banned for years because of its flammability. It is replaced by significantly thinner, hardly inflammable or non-flammable plastic foils that provide better insulation. In this way, more space is left for the copper wire.

The Toroidal Core

It is often claimed that only toroidal core transformers can be good for achieving the best sound. The toroidal core, thanks to its closed magnetic circuit without any air gap, undoubtedly gets very close to the ideal transformer. However, there are some

serious disadvantages especially with tube amplifiers with high operating voltages.

The winding technique for the toroidal core requires special winding machines. First, the predicted wire length must be wired onto an opening wheel from which it is then applied to the toroidal core. The wire is in principle wound twice. In addition, it is not possible to wind the wire neatly in layers. Either all windings are adjacent to each other on the inside of the core and on the outside, there is a gap between the winding. This technique is used for toroidal core variable transformer, where the output voltage is tapped using a slider and where there is only one layer. Or the wire is wound one next to the other and there are crossings on the inside. Normally, it will be something between those two techniques. With wire crossings, there will be pressure marks. So, if there is also a high voltage difference at the superimposed wires, this is the weak point leading to punctures and to shorted coils. Between the superimposed windings, an insulating foil is bandaged. Due to the winding technique, vacuum sealing is difficult because the resin can hardly penetrate the wrap.

Imagine this as if you were wrapping a finger with insulating tape and dipping it into paint. If the duct tape is removed from the finger, there is no paint where the insulating tape was. This is the same with attempts of sealing toroidal core transformers. The external grouting does not protect from internal punctures! With mains transformers where the primary winding is located below, and the secondary winding is above, a proof voltage of 4,000 V and more can be reached between the primary and the secondary winding. However, if interleaved windings are required as it is the case with tube output transformers, the insulation effort rises substantially.

When sheet scores and the standard winding technique is used, it is possible to wind in layers and, even without sealing, it is possible to achieve proof voltages over 4,000 V if necessary. The better the winding technique, the more copper increases the performance of the component.

In the third part of the article series, I will discuss long-term stability and how to choose the right transformer. 



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The Evolution of Tone Controls in Tube Amplifiers

One of my father's favorite sayings was, "There are cycles in nature." Whether the topic under discussion was truly nature (e.g., the weather), political cycles, economic cycles, or sociological cycles, the clear message was that the only constant is change, so we shouldn't get too upset about—or attached to—the way anything is today. The same can be said about entertainment electronics. In this article, we'll discuss one example: the evolution of tone controls.

By
Richard Honeycutt
(United States)

A tone control is a switched or variable filter. Filters come in four basic varieties: low-pass, high-pass, bandpass, and band-reject or notch (see **Figure 1**).

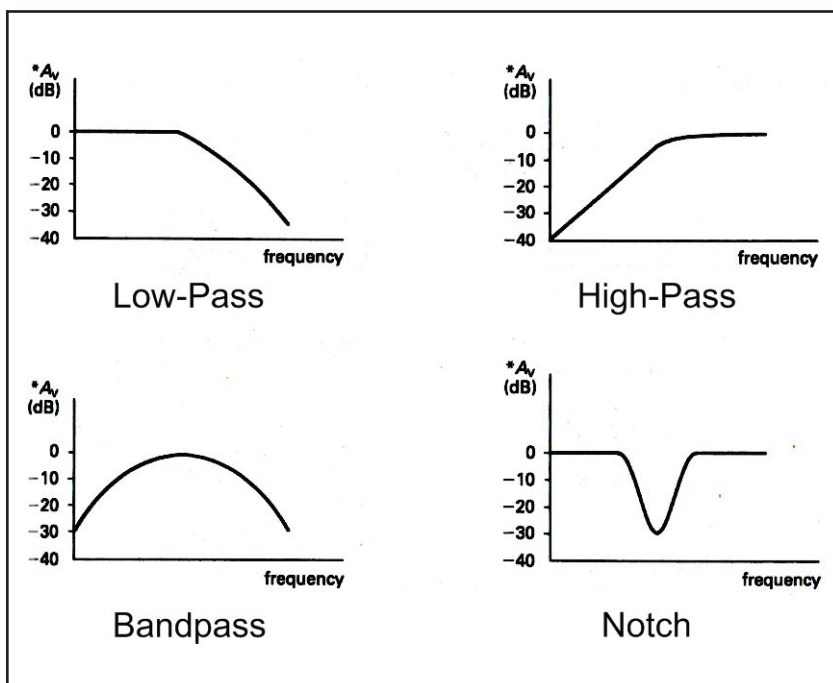


Figure 1: These are the frequency responses of the four basic filter types.

Figure 2 shows basic resistor-capacitor (RC). The "cutoff (or turnover) frequency" of a low-pass or high-pass filter is the frequency at which the gain is 3 dB below that in the "passband"—the frequency range where the filter provides negligible cut. Of course, the gain is fractional (negative decibel) in a passive filter. The turnover frequency of a low-pass or high-pass RC filter is given by:

$$f_c = \frac{1}{2\pi RC}$$

Until the mid-20th century, the only piece of entertainment electronics to be found in many households was a radio receiver. In some cases, the radio receiver had a built-in phonograph; in other households, a "phono oscillator" was used. This was a small, low-wattage (~100 mW) AM radio transmitter into which the output from a phono pickup was plugged. The resulting AM wave could be played on nearby radio receivers.

The first consumer electronic devices were AM radio receivers. These had only one control: volume. In the 1930s, manufacturers introduced simple tone controls. The earliest radios and phonographs had no tone controls. The first tone controls were treble-cut controls and were either switched or varied by a potentiometer (see **Figure 3**). The potentiometer

version was sometimes called a “losser” control.

The reason a treble-cut control could be called a “tone control” is that not only could it give a less treble sound, but if treble cut above ~500 Hz is combined with an increase in the volume control setting, the sound becomes bassier. The central small knob on the Philco model 645B radio receiver pictured in **Photo 1** was a two-position treble-cut switch. The treble-cut positions were probably used mostly when there was a thunderstorm, to reduce the crackly high-frequency energy in the radio signal resulting from lightning strikes.

High-Fidelity

So far, we’ve been discussing treble-cut (low-pass) tone controls. For many years, there was little interest in treble-boost or bass-cut controls. A bass-boost control would probably have been well received, but would also have been expensive because it would have required inductors. In the late 1940s and early 1950s, “high-fidelity” home music amplifiers were introduced by companies such as H. H. Scott and others.

These had additional tone controls for two reasons. First, home music listeners were beginning to invest in better speaker systems, and phonograph defects (e.g., rumble and hiss) were more audible and more annoying. So switched treble cut controls called “hiss switches” or “scratch filters” and switched bass-cut controls called “rumble filters” began to be standard features. Second, the introduction of the Baxandall tone control in 1952 made it practical to include a treble cut/boost and a bass cut/boost control in an amplifier, using only two potentiometers, a handful of relatively inexpensive capacitors and resistors, and no inductors.

The H. H. Scott Type 130 preamplifier was a state-of-the-art preamp in the early days of home stereo, shortly after this time period. **Photo 2** shows the tone controls of the 130 preamp. By the time this preamp was designed, we’d gone from no tone controls to five: bass cut/boost, treble cut/boost, scratch, rumble, and loudness. The “loudness control” was designed to compensate for the human hearing system’s decreased sensitivity to the lowest- and highest-frequency sounds at low listening volumes. The preamp’s output level control could be made to act as a simple volume control or a loudness control by means of a volume/loudness switch.

Figure 4 shows the frequency response of a typical loudness control circuit, for various settings of the potentiometer. Proper results from a loudness control depended upon several assumptions:

- All listeners have normal hearing.

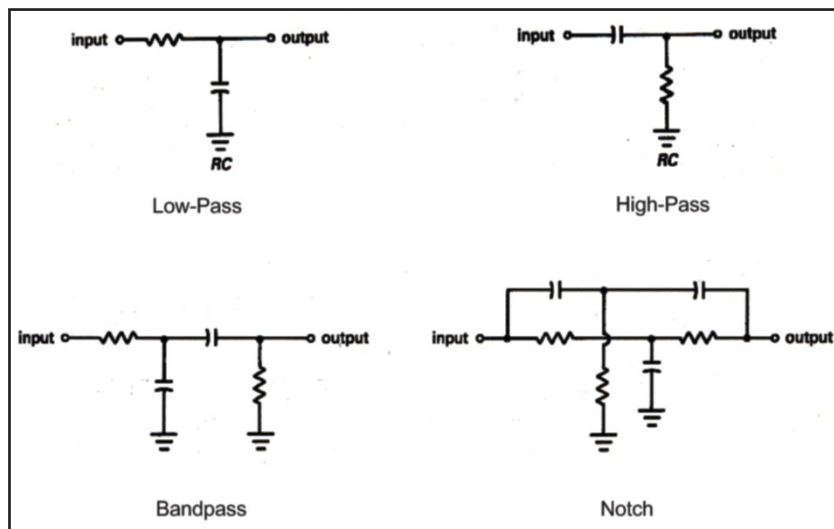


Figure 2: Low-pass, high-pass, bandpass, and notch filters can be made of resistors and capacitors.

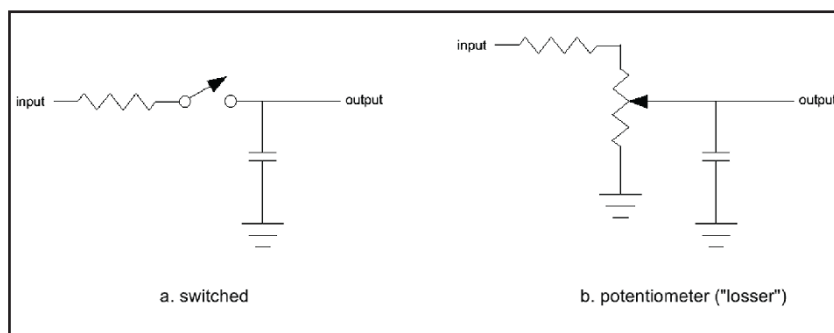


Figure 3: Treble-cut tone controls have been made in both switched and potentiometer configurations.

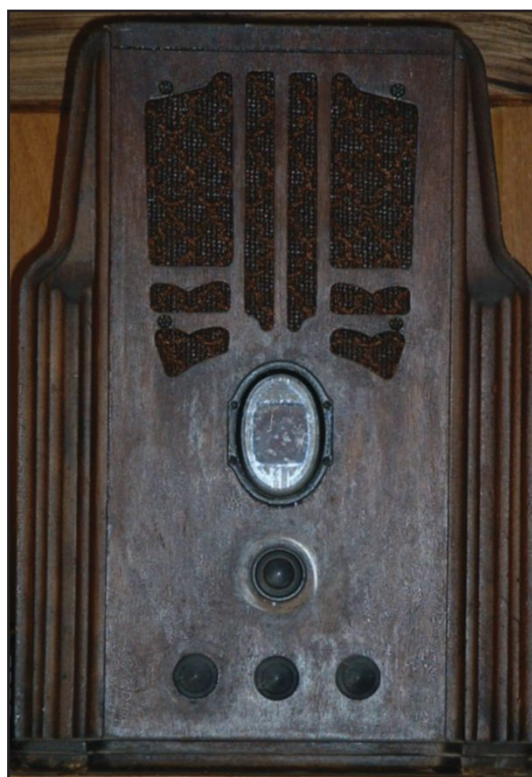


Photo 1: This 1936 Philco table radio had a two-position treble-cut tone switch.



Photo 2: High-fidelity mono and early stereo amplifiers had several tone controls such as these found on the H. H. Scott Type 130 preamplifier.

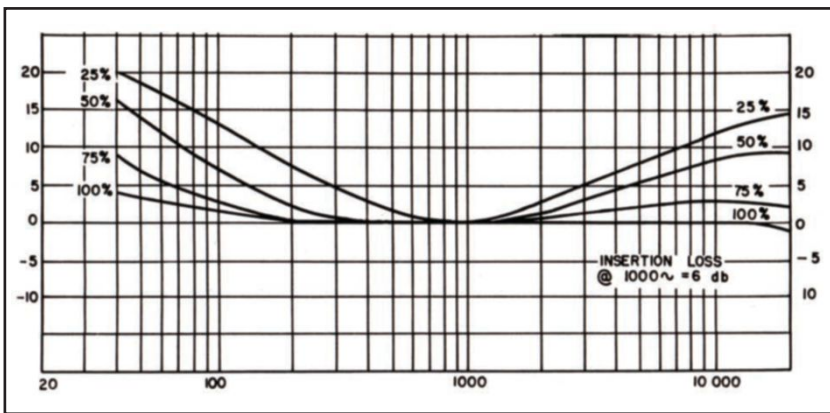


Figure 4: The loudness circuit boosted bass and treble, depending upon the control setting.

- All loudspeakers have identical frequency response and power sensitivity.
- All listening rooms have identical acoustics.
- All recordings were made under identical conditions, including the use of identical studio monitors.

Since none of these assumptions is really valid, loudness controls fell out of favor with most audiophiles after a few years.

The Baxandall Tone Control

The Baxandall tone control has become the standard for home audio preamps and amplifiers. **Figure 5** is a schematic of a preamp similar to the H. H. Scott Model 130. The bass and treble controls in the circuit of Figure 5 are part of a passive Baxandall network. Figure 5 also shows the circuitry of a loudness control, scratch filter, and rumble filter. **Figure 6** shows the frequency response of a specific passive Baxandall tone control network.

A design-oriented calculator for an active Baxandall network can be found at www.petervis.com/record_players_and_turntables/baxandall/baxandall-calculator.html. This calculator enables the user to specify maximum cut and boost values, as well as bass and treble turnover frequencies.

Although the description mentions the use of an op-amp, any amplifier using shunt negative feedback will work. The higher the open-loop gain of the

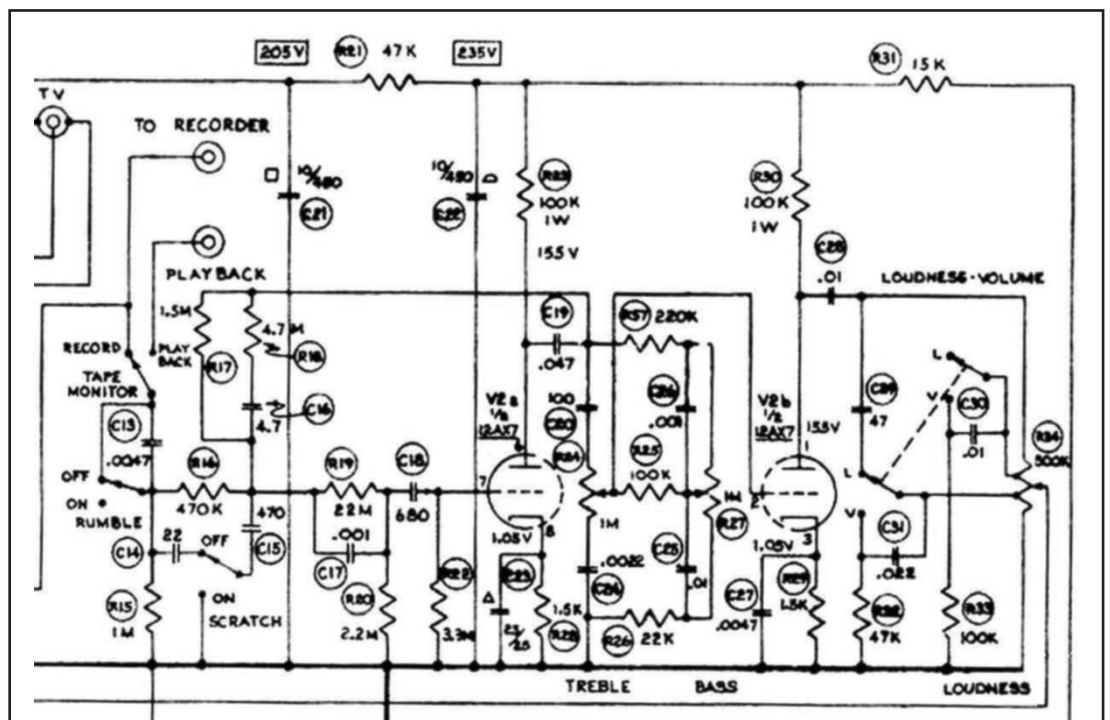


Figure 5: This schematic is similar to that of the H. H. Scott Type 130 preamp, and shows the tone-control circuitry.

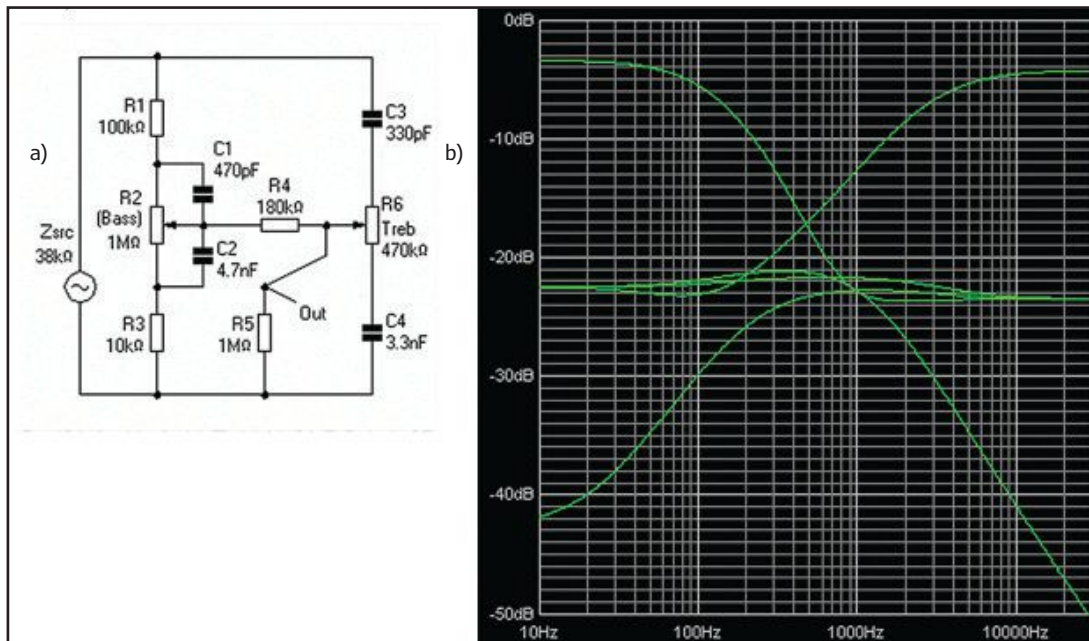


Figure 6: A passive Baxandall tone control network (a) produces the frequency responses illustrated—flat, max cut, and max boost positions (b). (From the Tonestack calculator, www.duncanamps.com/tsc)

amplifier, the more closely the actual results will match the predictions. **Figure 7** shows the active Baxandall circuit with components for a bass turnover at 300 Hz, a treble turnover at 1500 Hz, and a maximum boost/cut of 30 dB. **Figure 8** shows the frequency responses as modeled in LTspice. The upper curve is the response

with bass and treble fully boosted; the lower curve, with both fully cut. Although it seems that the input resistance of the amplifier stage would affect the performance (it does for a passive Baxandall circuit), in fact the resistance at the inverting input of an amplifier using shunt negative feedback approaches

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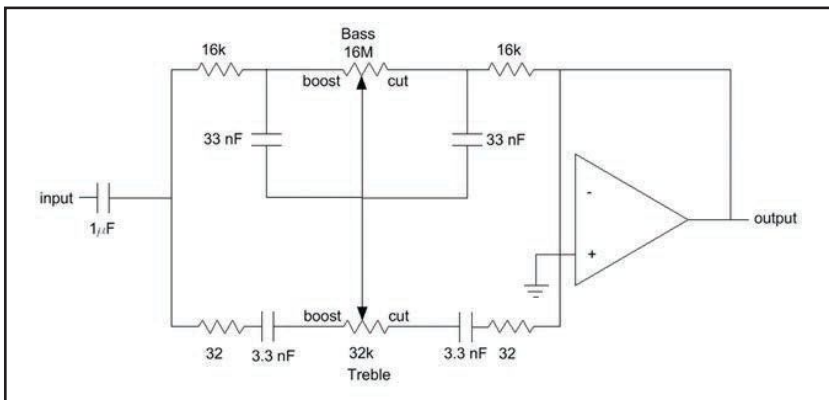


Figure 7: This schematic shows one variant of an active Baxandall tone control.

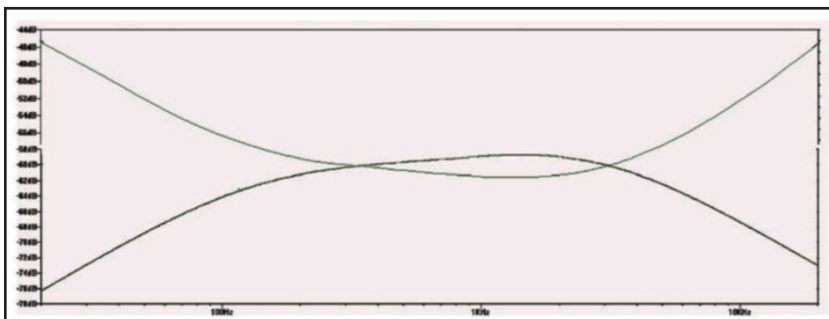


Figure 8: These curves show the result of full boost and full cut of the bass and treble controls of the active Baxandall circuit.

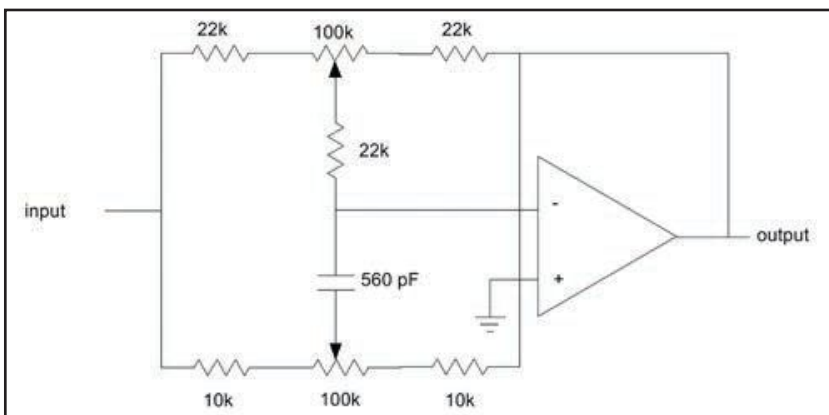


Figure 9: Here is another variant of the Baxandall circuit.

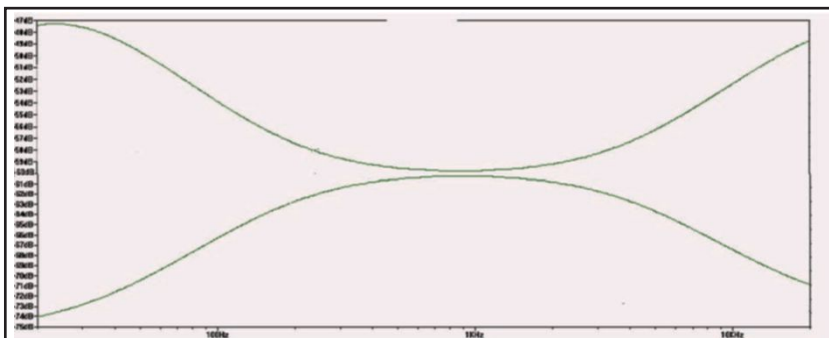


Figure 10: The circuit of Figure 9 produces a similar, but not identical, frequency response.

zero ohms as the open-loop gain is increased.

Figure 9 shows a slightly different active Baxandall circuit, from <http://sound.whsites.net/dwopa2.htm>.

Figure 10 illustrates its frequency response from an LTspice simulation. This circuit has a bass turnover of 150 Hz, and a treble turnover of 2.5 kHz.

The Benefits of Tone Control

So what good are tone controls? Although some purists decry their use entirely, the majority of the market prefers having tone controls for one or more of the following reasons:

- Personal taste—Different strokes for different folks.
- Making up for poor speaker performance—Sure, a purist may just say to go out and buy a better speaker, but depending upon personal financial priorities, using tone controls makes more sense for a lot of people.
- Acoustical wallpaper—Experience has shown that background music with a full frequency range is more likely to be distracting than music with reduced bass and treble.
- Hearing compensation—Hearing-challenged people lose high-frequency sensitivity first, and a treble boost can make music sound more realistic to them.
- Feedback avoidance when listening to vinyl recordings—Depending upon the position and mounting of the turntable with respect to the speakers, bass cut may sometimes be needed to avoid electroacoustic feedback at low frequencies.
- Compensation for poor amateur recordings—Especially older cassette and reel-to-reel recordings may have been made using poor microphones having weak response at low and high frequencies.

Throughout the history of electronically reproduced audio, we have seen a progression from no tone controls, to tone switches, to hiss and rumble filters, to treble-cut “losser” controls, to treble/bass controls, to loudness compensation, to a short-lived attempt to introduce graphic equalizers into home stereo systems, and now we’re back to no tone controls on the highest-end amplifiers. Truly, there are cycles!

Despite the long and storied history of our affair with tone controls, certainly the majority of listeners, including a significant percentage of serious listeners, prefer the combination of a Baxandall bass/treble control set, no loudness compensation, plus perhaps hiss and rumble filters for vinyl enthusiasts.

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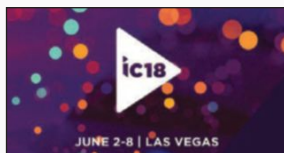
Here are a few places where you might find a copy of *audioXpress* and possibly meet one of our authors and staff members.

June 2-8, 2018

InfoComm 2018

**Las Vegas Convention Center
(North and Central halls), Las Vegas, NV
Show Dates: June 6-8**

www.infocommshow.org



InfoComm organizers, AVIXA (the Audiovisual and Integrated Experience Association, www.avixa.org) have announced new initiatives, content program, and exhibition for InfoComm 2018, which returns to the Las Vegas Convention Center from June 2-8, 2018. The show will kick off on June 5, with the opening reception at OMNIA Nightclub Las Vegas.

The AV Tech Tours will take visitors to view tech marvels, such as the nightclubs at MGM and Aria. There are more than 80 new education sessions, training, and workshops—from the Emerging Trends Day to the new Integrated Life @ InfoComm 2018 (www.parksassociates.com/integratedlife2018), a special track of workshops and sessions, and a dedicated area on the InfoComm 2018 show floor, for spotlighting smart building, smart home, mobile, security, and related audiovisual solutions that create integrated experiences wherever people use technology.

AVIXA also announced that developers of C2 Montréal, a ground-breaking business conference merging commerce and creativity, will bring their innovative brand of immersion and collaboration to the InfoComm show floor.

June 8-10, 2018

Los Angeles Audio Show

**Hosted in collaboration with
The Los Angeles & Orange County
Audio Society**

Hilton Irvine/Orange County Airport, Irvine, CA

www.laaudioshow.com



The West Coast Consumer Show is back to its roots in Irvine, Orange County, CA. In 2017, there was a split between the promoting parties and what was once T.H.E. Show (Newport and Irvine), is now the Los Angeles Audio Show, hosted in collaboration with the Los Angeles & Orange County Audio Society. Other parties attempted to create a new event, designated only as T.H.E. Show, scheduled to happen in Anaheim, in September 2017. That show was canceled due to "limited participation." While the first Los Angeles Audio Show did take place successfully (albeit smaller) in early June, at the Sheraton Gateway LA, close to the Los Angeles International Airport. This location was not ideal so this year, the promoters—the Los Angeles & Orange County Audio Society, The Orion Group, and Marine Presson (former director of T.H.E. Show)—have moved the show back to Orange County—at the Hilton Hotel close to John Wayne's Santa Clara airport—, which should help bring the event back to its former glory and original spirit.

June 13-15, 2018

CES Asia

Shanghai, China

www.cesasia.cn



Owned and produced by International CES (Shanghai) Exhibition Co., Ltd. and co-produced by Shanghai Intex, CES Asia 2018 will run from June 13-15, in Shanghai, China. The premier event for the consumer technology industry in China, CES Asia showcases major brands and innovative companies across several vertical markets in the Asian marketplace. Key global businesses come to this new event to grow and reinforce their brands by showcasing the latest products and technologies to consumer tech industry executives, foreign buyers, international media and a limited number of consumers from China. Attendees have exclusive access to some of the largest brands from China and around the world, while celebrating the innovation that defines the consumer technology sector.

This will be the fourth edition of the show. In 2017, CES Asia shattered expectations and broke records, with 40,000 gross square meters and more

than 460 exhibiting companies from 22 countries. An independent audit confirmed total attendance at CES Asia 2017 to be 38,531, including 1,258 members of the media, from 78 countries, regions and territories; a 13.7% increase from 2016 and a 34% increase from 2015.

The Consumer Technology Association (CTA) has already announced that the United States Department of Commerce (USDOC) has granted Trade Fair Certification for CES Asia 2018. This certification is a formal federal endorsement of CES Asia as a recognized opportunity to showcase US products and services. It is granted by the USDOC, which will provide promotional outreach to expand US exhibitor participation. Trade Fair Certification also benefits buyers and companies that seek to import products and technology from the United States.

June 20-22, 2018

**2018 AES International Conference
on Music Induced Hearing
Disorders**

Columbia College, Chicago, IL, USA

www.aes.org/conferences/2018/hearing

AES Conferences are held periodically in locations around the world. Conferences focus on a specific field of audio engineering. They serve to create a forum amongst professionals in that field and through these gatherings and the publication of their proceedings, contribute to the advance of international science and technology.

The Keynote Address "Music for the Brain" will be delivered by Nina Kraus (Northwestern University, Evanston, IL). Panel discussions include "Safe Sound & IEM Fundamentals," during which legendary monitor engineer Mark Frink (Tony Bennett, k.d. lang, Super Bowl); Matt Engstrom of Shure; Dr. Michael Santucci, Au.D. of Sensaphonics; and Dr. Heather Malyuk, Au.D. of Soundcheck Audiology delve into best practices and demonstrations for creating safer in-ear monitor mixes, and "Many Ounces of Prevention Where There IS No Pound of Cure," which addresses teaching strategies for hearing awareness and conservation, moderated by Benj Kanter of HearTomorrow.Org.



June 28-30, 2018

**2018 AES International Conference
on Audio Archiving, Preservation
& Restoration**

**United States Library of Congress
National Audio-Visual Conservation
Center, Culpeper, VA, USA**

www.aes.org/conferences/2018/archiving

This conference will present an opportunity to bring researchers and practitioners together for three days to discuss critical preservation issues and focus on strategies that will support and encourage collaboration and interoperability between industry and the preservation, restoration and archiving communities.

Some important themes will be addressed including:

- Physical/object based preservation and material science
- Handling and storage of audio carriers
- New trends in material research
- Preservation by digitization
- Mechanical vs. optical transfer
- Magnetic media
- Digitization, and restoration of historical obsolete disc formats
- Preservation issues for emerging high-end audio formats
- Preserving legacy professional formats
- Multi-track tapes and obsolete digital formats
- Storage and access technology
- Current state and future development of digital archives
- Role of archives in preserving commercial audio
- Aesthetic considerations in digital restoration of historical audio





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What if you could wring every last drop of performance from your subwoofer, whether the cabinet is the "optimum" size or not? Who wouldn't want that? With the Digital Signal Processing (DSP) in the new Dayton Audio SPA250DSP, you can. This plate amp allows you to optimize the output of your woofer *in your cabinet*, rather than trying to build the cabinet to suit the driver. You no longer have to sacrifice low frequency for SPL, or box size for low frequency. You can adjust PEQs, HPF, LPF, Phase, Limiting, and Mains Delay, all according to your specific taste and needs. All the power is in your hands. Various parameters can be easily adjusted using the LCD display screen, or (for the deepest level of management over all functions) via USB connection to your computer using the included software package. Don't be a slave to the sub cabinet. Be the master of your own destiny with the SPA250DSP!

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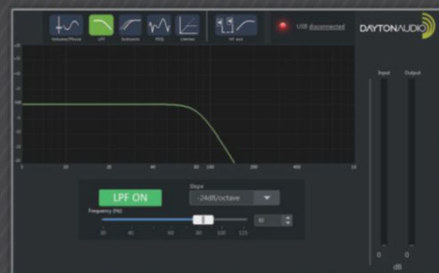
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Volume/Phase



Low Pass Filter



Subsonic Filter



5-Band Parametric Equalizer



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High Frequency Output