

Lyngdorf Engineering

History and Overview

Lyngdorf Audio products are best understood when experienced in the context of our company's vast experience and the history of our innovative engineering. Lyngdorf develops, designs, and creates products for high-performance audio enthusiasts seeking the highest quality and most exciting components on the market. Lyngdorf Audio products do not re-produce music; they re-play music with all original passion and musicality intact. The brand is known throughout the world as the leader in high-performance digital audio.

HISTORY:

In 1993, when Snell Acoustics brought out the world's first full range digital Room Correction System (RCS), it was owned by Peter Lyngdorf (later the founder of Lyngdorf Audio). Sigtech and others had worked with RCS from 1991, but none of the systems were effective below 200 Hz. The 1993 Snell RCS had a much better correction resolution, making it effective at all audible frequencies.

In 1997 Peter Lyngdorf entered a joint venture with TacT Audio, in order to merge room correction technologies with this new design ideas of a fully digital amplifier.

TacT's correction system did everything an audiophile could ask for. It provided extremely precise correction against any target curve, and it allowed users to select any measurement window, measurement resolution, smoothing, and averaging. Plus, the results of the correction could be verified for accuracy. The system was "perfect" as far as users could see. This RCS was reviewed by *Stereophile* and many other magazines with universally great results, and sales were robust. The basic approach is identical to what the industry is using for room correction to present day.

However, there were early warning signs:

- A cottage industry developed to improve the sound of the RCS units. Everybody thought the correction was perfect, but the sound was often fatiguing. As a result, people started to experiment with components in the power supply, often changing op-amps and adding many other modifications.
- We knew that the actual *perceived* results of correction were not consistent. The same target curve would *sound* quite different in different rooms and could be totally different with different dispersion characteristics of the speakers. Furthermore, the better a frequency response measurement *looked*, the poorer it *sounded*.
- We experimented with in-room phase correction, but again, the better a measurement *looked*, the poorer it *sounded*. (In one experiment, we did a five-second complete inverse phase filter. It nailed the impulse response perfectly—and it sounded like a hilariously bad reverberation system.)

Shortly after Lyngdorf Audio was formed in 2005, we finally decided that we had exhausted all options with conventional RCS technology. We were getting results, but they were not consistently *better* results.

A new team of 30 engineers was employed with a challenge to start from scratch with room correction. These engineers were told not to use or investigate any existing RCS units—the goal was to generate a truly new approach.

Some things had already dawned on Lyngdorf's engineers. For instance, the engineers recognized that in signal theory it is assumed that a flat frequency response will result in a good impulse response. Previous correction systems were based on that assumption. Basically, we had treated the problem in the same way you would treat frequency deviations in amplifiers, in near-field monitors, or in an anechoic chamber, where the issue is two-dimensional.

In a room however, the measured frequency response is a plethora of sound arrivals coming at different times at different angles from the same event, which in this case is the loudspeaker. A normal system simply measures accumulated energy versus frequency, *but the tonality we hear is very different*. For instance, a null in the measurement may not sound like a null to the human ear. In other words, it's a frequency and time problem.

In addition, the measurements we were using had errors, particularly at low frequencies. We had been measuring with sweeps and pulses of varying lengths, but the signal-to-noise ratio was, in many cases, too poor to get valid data for correction.

Before the project commenced, we identified some indispensable requirements:

1. Measurement technology had to be "real-world" practical, not something that would work only in a totally silent room.
2. Correction had to be based on the listener's perception of sound, not on the way correction would look on a graph.
3. The character of the speaker should be unchanged. We would compensate for the unwanted coloration from a listening environment, and we would allow the system to take out box resonances.
4. We would not allow any tendency to aggressiveness in the sound.
5. We would not allow negative effects of room correction outside of the listening area.
6. We would make the setup so easy that no expert help would be needed for calibration.
7. If possible, all functionality should be "on-board" without the need for a PC.

A short while into the project, our chief acoustical scientist Jan Abildgaard Pedersen presented a radically new approach which would later be named RoomPerfect™. At first, the team thought Jan's ideas were totally counter-intuitive. Jan wanted to make measurements completely unrelated to the listening position, then use mathematics to derive the best correction filters. It took some time for everyone to get on board with Jan's new ideas, but once they did, things progressed quickly.

The result was a completely new measurement technology which allowed accurate measurements even in noisy environments. This technology also tested for signal-to-noise ratio of measurements at all frequencies and ascertained that every measurement was valid at all frequencies, even where there were severe dips in the response. Given enough time, RoomPerfect™ can make accurate measurements in situations where background noise exceeds the measurement signal. Initially, all the math was done on a PC, and the compilation for a single room correction would take hours to compute. In the final product, everything was done in our own DSP code within minutes.

When we first listened to the final RoomPerfect™ we were honestly disappointed to find there was not a huge, dramatic sound difference. But then we realized the annoying coloration from the listening room had *disappeared*, and we could listen to music forever without fatigue. The proof is in the pudding—we have never heard of a Lyngdorf RoomPerfect™ customer who wanted to change components in the power supply.

The RoomPerfect™ patent was applied for and later granted.

RETROSPECTIVE:

Many people in the audio industry have tried to use graphic or parametric equalizers to achieve a good frequency response from a speaker setup in a normal living environment. When you ask about the success, most will admit that after hours of work and countless measurements to achieve flat response, the sound is worse! This is simply because a single response measurement is meaningless when compared to what we actually hear.

EQ's have been used successfully (though in acoustically dead environments) with very short reverberation time, because if you have short reverb time, the room behaves almost anechoically, and then the problem is close to two-dimensional.

This may also explain the propensity for home theater acoustical consultants to design extremely dead acoustic environments where traditional RCS's and EQ's work quite well.

The problem with dead acoustics is that...they are dead. You will need much more power or will have much higher distortion for the same SPL. In our opinion, you should not aim for less than RT 60 of 0.4. We have seen dedicated home theaters with less than 0.1.

REVIEWING ROOMPERFECT™:

In general, reviewers have vast experience in achieving good frequency response from speakers in their listening environments. Good frequency response is achieved by carefully manipulating resonances and cancellations due to the speaker-room interaction. Often, the best result is achieved by placing speakers far from the front wall and placing seating far from the rear wall. In other words, sitting close to the speakers provides more direct sound and less sound from the room.

If you already have "perfect" speaker placement, then RoomPerfect™ will probably do very little. But users should try to put speakers and seating where they would realistically be placed by someone who does not do this for a living—and then apply RoomPerfect™.

Think about this for a moment: If you get too much bass because you placed your speakers near the rear wall, it is because you have *less cancellation of energy*, or—in other words—*less energy coming to you out of phase*. It will sound "boomy" because you have too much bass, but after RoomPerfect™, you will end up with less energy coming to you out of phase. RoomPerfect™ will perfectly scale back the level, and you will have higher headroom/less distortion from your speakers.

VERIFYING ROOMPERFECT™:

The inquisitive mind would want to verify by measurement that RoomPerfect™ is indeed giving you the best possible response in the listening position. However, throughout our long history, we have found that a single measurement is virtually meaningless in terms of depicting the perceived balance of a system. If you want a "perfect" response in your listening position, please revert to the old TacT or even NAD units.

Or try a simple experiment: Use the voicing tool in the 3400 or MP60 to make the response "better" for your measurement. You can do that quite easily with the parametric EQ's. You will find that it *looks* better and *sounds* worse. This is why we do not show the user the result of the correction—because what you see and what you hear are entirely different.

THE FULLY DIGITAL AMPLIFIER:

In 1996, Lars Risbo and Peter Lyngdorf met for the first time. Lars had an audacious idea—to design a fully digital amplifier which would convert digital PCM information to PWM and would drive the

speaker directly from what was essentially the DA converter. Peter listened to the first early prototype, and although it was noisy, there was something inherently right about the sound.

Peter agreed to finance the full development and patents for the technology, and in 1998, the first product using these inventions was brought to market. The Millennium Amplifier was a 2 x 150 W fully digital amplifier which did not actually do any “amplification.” The signal was generated at the output based on a calculated conversion from PCM to PWM. The only components in the signal path were one coil and one capacitor. Voltage was supplied from a very quiet and steady power supply which could be regulated.

The Millennium was an incredible breakthrough. As “What Hi Fi” wrote in their test in 1999: “So you want to know how many stars out of 5? About thirty-eight.”

In 2000, Lars Risbo sold the Millennium technology to Texas Instruments. Peter Lyngdorf received money back for the patents and entered an agreement with TI to continue to share knowledge. Lars became head of development at a new TI facility in Copenhagen, and the friendship between Peter and Lars continued. (In 2014, Lars and Peter joined forces again in a new entity, PuRiFi.)

At Lyngdorf Audio, we have used the core technology invented by Lars since our founding. Interestingly, we seem to be the only company combining this technology with a fully regulated power supply. The reason is probably that this design relies on a super quiet, DC-regulated voltage. In a typical amplifier with feedback, you have a huge amount of noise rejection due to the feedback loop. In our designs, we have 0 dB! Any noise on the PSU is also present on the speaker terminals.

The beauty of the fully regulated DC voltage for the switching is that we effectively change the “size” of the amp depending on the loudness. At low levels, the TDAI 3400 is a 3.6-watt amp, and at the highest level it’s a 400-watt amp into 4 ohms.

The PSU design is so immune to noise from the power grid that we guarantee no measurable difference in performance due to noise from the grid. This means that at normal playback levels there is no attenuation (digital or analog). There is no potentiometer to interfere with the sound. In fact, there is no way the system can change the sound dependent on playback level. There is also no way that component deterioration over time will affect the sound. Even without feedback, Lyngdorf amps have low noise and low distortion.

Even without feedback, Lyngdorf amps have low noise and low distortion. You can achieve lower THD+Noise with feedback, but then you would have to design a more conventional Class-D amplifier which relies on a DAC and a Pre-Amplifier for optimum performance.

Think about this: Do you really want a 400-Watt amplifier if you drive a high efficiency speaker at low levels? In this scenario, you would be using your pre-amp at a very low output and the potentiometer would probably not be so accurate. Would it not be better to change the size of the amplifier according to your need?

To be fair, a conventional approach can offer great results, but we love the relaxed clarity of fully digital technology and the added dynamic range achieved with the variable power supply.

Lyngdorf amplifiers incorporate our Inter-Sample Clipping Correction (ICC) technology. This technology will substantially improve the sound of typical modern digital recordings with low PLR (Peak to Loudness Ratio)—in other words, with dynamically compressed music.

ICC will not have any effect on good typical audiophile recordings, but it will have a dramatic effect on compressed modern pop music.

To understand the problem, we recommend this video, which features our good friend Thomas Lund at AES explaining the issue:

<https://www.youtube.com/watch?v=BhA7Vy3OPbc>

Listen in at 2:45, 5:50, and 21:20 for highlights, if you are in a hurry.

Even the highest quality DACs will exhibit extreme distortion when converting compressed pop music due to True Peaks above 0 dB FS.

DIGITAL OUTPUT:

The digital output is coming directly from our DSP. It is intended to be paired with equipment using proper re-clocking. If measured directly, it will exhibit some jitter, which is NOT representative of the internal jitter of our systems.

When measuring small signals, remember that we are controlling volume by the PSU—so reduce levels appropriately.