Ari Varla

The man behind much of the research and development at active monitor manufacturer Genelec discusses the domestic reference, low end, the pitfalls of digital filters and control, and everything being there for a good reason. **ZENON SCHOEPE**

A **SELF-DECLARED AUDIO** hobbyist from the age of 14, Ari Varla studied for a masters degree at Tampere Technical University before joining Genelec in 1980. He has worked on every Genelec monitor apart from the first version of the S30 from 1978. Now chief R&D engineer, acoustics, his research has contributed to the application of waveguides in combination with direct radiators. He points out that in the early 80s they simply called them short or semi horns instead of waveguides and adds that waveguides have since been adopted by several competitors but, in many cases, still with 'a lack of deeper understanding of their desirable operation'.

He has also investigated the impedance behaviour of loudspeakers (starting in the 1980s with Finnish amp guru Matti Otala) and their interaction with the feeding amplifier — the 8040A and 8050A models use this type of related technology for distortion reduction. With Genelec's latest models, the LSE-subs and the 8000-series 2-ways, the main research has gone into 'minimising systematically all kinds of nonlinearities and aberrations that we know will eventually limit the perceived sound quality'.

He holds two international patents and numbers guitar playing, vintage tube amp repair, amateur telescope making, fishing, vintage fishing gear, motorcycling, and playing ice hockey among his interests.

What's special about Genelec's approach to monitor manufacture?

We take it seriously. If you look at our designs over the two and a half decades there's clearly a systematic way of improving different sound quality parameters in every model generation. Our first product, the S30 in 1978, already had the on-axis frequency response uniformity of today's accepted standards, which was quite exceptional for its time. Then we moved further on to minimise a monitor's secondary sound source generation and at the same time to improve its directional characteristics closer towards a constant directivity system in the legendary 1022A model (1983) where for the first time our pioneered waveguided mid- and high-frequency systems were integrated into the cabinet mold. Then came the time to maximise the undistorted maximum SPL capacity of a direct radiator large soffit-mounted monitor and our 1035A in 1988 really set the mark about 10dB higher than was achievable with conventional soft dome designs. This technology is still valid today.

In the 1990s we concentrated on redefining close-field monitoring, applying all we had learned and putting it into the smallest possible package. It was the 1031A that had a flat on-axis response down to 47Hz, a waveguided directivity optimisation for the HF and was sufficiently loud for rock and roll monitoring.

Now with our new 8000-series monitors we continued this development process by paying a lot of attention to improving all sound quality parameters at the same time to make the progress more audible.

Are we restricted in monitor design by the industry's conservatism?

The music industry is not necessarily conservative; there are a lot of creative people across the whole human spectrum there. Of course we as a monitor manufacturer are supposed to adapt to the current needs of our customers but we have never really taken commands from the established market trends. If we had, we would have probably never made an active monitor in the first place nor have attempted to sell them in the US or UK, they were so much against the trends back then.

With our new 8000-series we took a long and deep breath before we launched the products, but in the past we had been successful by doing things the way we thought was right, so why not now? Motorcar bodies used to be quite rectangular and were made out of wood some 80 years ago but then somebody started rounding them up and making them out of metal — and for very good reason. The same applies to the 8000-series: everything is there for a very good reason. You can argue an opinion but not the facts.

Cabinet, crossover, driver or amplifier — where are the limitations?

I think today's amplifiers (if designed properly) are the least limiting factor — viewed against their performance. Their power and distortion figures can be considered sufficient. Passive crossovers can many times perform quite acceptably, but their design has so many limitations and tradeoffs that luck presents itself as a big factor in the final result. The passive electronic crossover can be considered as the next evolution from a purely mechanical one.

With an active crossover there is almost total freedom to design the responses, equalisations and transition bands, within good engineering practice of course.

The driver and the cabinet form an inseparable combination (together with the crossover) and that's how we always approach the design. The limitations here are many: driver excursion, stroke volume, cabinet volume and front baffle size. The three first ones affect the LF-limitations, the last one the directivity. To summarise this: a small box has more LF and directivity limitations than a bigger one. To compensate this, a small speaker should be listened to at a shorter distance so that it sounds louder and delivers more direct sound to the listener's ears.

The market is full of highly affordable monitors, how are those price points hit and where are the compromises?

Genelec has never got into the lowest price point battle because that would have meant sacrificing our engineering philosophy. Our only goal has been to be the technology leader because there are always customers for the best. Low-level engineering is sometimes called 'compromising' but the term is equally applicable to a certain human state of mind.

Where do you stand on the use of 'domestic reference' loudspeakers in many studios?

'Domestic reference' as a definition is as vague as it can be. First one should define the 'domestic' i.e. whose house or lodgings we are talking about? Then apparently the 'reference' here means speaker systems possessing flaws or imperfections — isn't that what people's home systems are supposed to have? Now we only need some wise guy to decide whether these 'domestic references' are dull sounding or overly bright, are they boom-boxes or dry in bass and if they have a nasal over-emphasised midrange or a valley response. The number of variables soon become endless, hardly a desirable feature for something called a 'reference'. It's much more useful to call a reference something that lacks all these imperfections to the least and has its performance accurately specified and is widely available.

Are end-user expectations of low-end unrealistic for the small sizes of cabinet that most are actually listening on?

Sometimes, but there is no general rule here. End-user expectations vary hugely and a good example is about opinions of what is considered loud. Here the range can be something like 30dB.

The use of subwoofers through dedicated bass-management has enabled people to get more used to small speaker setups delivering powerful full-band performance. Despite this, we have always tried to design our systems so that no matter how small they are, they always deliver a very low useful LF-cutoff for their size. The new 8000-series is an example of this, the 8050A 2-way 8-inch woofer reaching clean down to 35Hz -3dB with good SPL.

Genelec has always majored on the monitor/room interaction issue but how do you reconcile this with the fact that acoustic treatment is often not addressed in small rooms using your small monitors?

For that particular reason we established a technical customer support department some ten years ago whose job is to help our customers with their control room issues and to provide them with speaker in-room measuring services. This on the other hand also benefits us in collecting measurements and experience from real life situations. An example of this, just recently, was our AES-preprint presented in San Francisco about the mixing console loading effects upon meter bridge mounted nearfields. Without a large measurement databank the clear acoustic mechanism behind this phenomenon could not have been so clearly evidenced.

The other factors that we are using in making our speakers more adaptable into difficult room acoustics are a controlled and uniform directivity, offering more direct sound to the listening position while simultaneously ensuring that the indirect radiation towards the room boundaries has practically the same spectral content as the direct sound. This is very important in achieving a good consistency between near field and far field listening.

Third, we have for a long time used the active speaker characteristic to easily include the necessary electronic tone controls for room response tailoring. No matter how well a speaker behaves in an anechoic lab, the result will always be different when it's been brought inside reflective boundaries. The key here is to have speaker specific function of the controls and to have them operated in a predictable way that will most probably fix the problems associated with different speaker positioning. Again, wide experience and plenty of data will help in this matter, too.

The digital monitor is as much of a misnomer as the digital microphone, but what are the real advantages and where could it lead?

There are some potential pitfalls when designing a loudspeaker with digital filters and controls. If we first think about the equalisation of a single drive unit — normalisation of its transfer function — we have to remember that the impulse response (and its resulting frequency domain responses) depends upon the measuring angle because of the interaction of the cabinet edge reflections. We may only obtain one valid equalisation result at each measuring angle because we are just trying to compensate a three-dimensional process in one dimension. We cannot subtract the diffraction artefacts in anything other than one measuring point in space, elsewhere the subtraction turns into adding. So, making things better in one point makes them worse in others. This becomes even more the case when attempting to compensate systems that have poor temporal performance.

The paradox here is that you need an already excellent acoustical system to unleash some of the potential of digital equalising. But compared to what is achievable with the best analogue filtering techniques today the improvements are likely to remain marginal. Many times the flattest on-axis responses do not necessarily result in the best sounding systems because then you would have ignored the total radiated power response — this often looks quite different but is very important, although it is much more troublesome to measure and that's why it is usually neglected.

I think that many acoustic imperfections in a speaker system remain audible regardless of the complexity of the DSP-program used to compensate them.

In an ideal world, should monitors that are used in a multichannel configuration be similar to from those used for stereo?

Stereo is a multichannel format, only a special case of a 5-channel one (or any number of channels). If we think about backward compatibility and the fact that 99% of music is being sold and produced in the stereo format the answer is yes. There's no good practical reason why the speakers couldn't be similar. With a subwoofer, though, you have to take care of proper crossover filtering between the main speakers and the subwoofer to maintain compatibility from 5.1 channel system all the way down to mono. The concept of a subwoofer as a separate unit must be forgotten and instead thought of as an integrated bass extension system for the main speakers making them all fulfill the 'full bandwidth' criteria.

What are your thoughts on the use of the '.1 channel' outside sound for picture work?

It's so amazing how this low frequency effects (.1) channel that takes care of less than 1% of the channel capacity has created most of the reproduction problems in a 5.1-channel system. I think it's a total misconception in musical recording. The principal idea of an LFE-channel was to allow a higher modulation of the main channels without the need for two hours of the movie to be prepared to handle the last minute low frequency blasts and by doing so wasting the precious headroom and bandwidth of the main channels. The musical quality issues of such a LF-reproduction mechanism came second.

Whenever we have multiple sound sources — acoustic or electronic — and we want them to work coherently it requires that these sources have identical frequency and phase responses. Unfortunately neither of these is the case with the LFE-channel in relation to the main channels. A typical production method is feeding part of the main channels' bass information into the LFE in the hope that it will somehow sound 'deeper' or 'more fundamental' than it would through the main channels. It will not. Unfortunately the steep low-pass filtering of the LFE makes it produce comb-filtering interference with the main channel signal around 100Hz, which will make the system sound sloppy and have a lack of punch. There's really no reason why the main channels can't carry all low frequency information in musical recordings — and without artefacts. That's why I keep the LFE-cable disconnected in my DVD audio player. ■