GENELEC®

Acoustical Insights Behind The Development of Genelec AutoCal™

Introduction

The way loudspeakers interact with a room defines the quality and precision of reproduction in that room and, as a consequence, determines the resulting quality of the production process. Loudspeaker placement in relation to boundaries, furniture design and layout, calibration of sound sources, room shape and acoustic treatment, all are critical parameters which affect the sound field. These issues should all be carefully considered before a monitoring system is set up, equalised and calibrated. The crucial question is; which kind of anomalies should be corrected and which should not, as the answers have an important role in the final perceived sound quality in the room.
Finally, if the technology is to offer real benefits to the user, integration of DSP into monitoring requires a number of key decisions to be made in the design and implementation stages. The most important benefit of such a DSP-based system is clearly the possibility for an automated calibration system to fully align a set of loudspeakers. This is what Genelec's AutoCal™ is doing.
Genelec's Christophe Anet and Ilpo Martikainen explain the company's take on these various subjects.

1.Loudspeaker-Room Interaction

Many multichannel control rooms are not originally designed for multichannel production. Room dimensions, equipment layout and installation vary from one room to the next. These factors have direct consequences on the quality of the reproduction and hence they are more or less audible in the final production. However, our ear/brain combination is very capable in filtering and adapting to different conditions, and hence the final results are actually much better than what could be expected from simple room response measurements. Successful recordings are made in very different spaces.

Depending on the room size there is a certain transition frequency which divides the acoustical behaviour of the room. Above the transition frequency the sound field starts to be diffuse while below it the modal behaviour dominates. In very large spaces like concert halls the sound field is diffuse at all frequencies, while another extreme might be a car. Traditionally room acoustics calculations relate to large spaces and their validity and applicability to describe small room phenomena is limited.

Room Symmetry
The general rule that the control room layout should be symmetrical is applicable in all cases, be it for stereo or multichannel audio reproduction. As the largest and heaviest piece of equipment in the control room is the mixing console, its location along the middle axis of the room is important.

Listening Position

It is generally recommended that the listening position should be in the front half of the room, so that the engineer has better direct-to-reverberant signal ratio. This has been augmented with several control room construction principles with selected acoustic treatment in different parts of the room. However, the multichannel loudspeaker configuration makes this definition more complicated, as the distance to all loudspeakers shall be the same. If the room has, for example, hard and reflective front wall surfaces, direct sound from the rear left loudspeaker might bounce on the front right loudspeaker front baffle and nearby boundaries. This situation should be avoided as these strong first reflections may alter the front loudspeaker's direct sound and sound stage imaging. This calls for some planning in room geometry and location of absorptive surfaces. It is obvious that we need more research to understand what is important and what is not in order to make better control rooms for multichannel use. Not all reflections are bad or destructive, just the contrary. And, no one would like to live or work in an anechoic chamber.

Acoustic Loading

When a loudspeaker with a flat anechoic (4π) response is placed against one solid boundary - which is large compared to the wavelength - the radiation space halves to 2π, and the theoretical amplitude gain is +6 dB for frequencies below a few hundred Hz. (At higher frequencies the loudspeaker baffle itself defines the radiation space). This applies to flush-mounted loudspeakers or loudspeakers placed with their back against a wall. In these cases the amplitude gain has to be compensated to retrieve a flat and neutral frequency balance.

A subwoofer's typical location is on the floor and against a wall. Here are two large boundaries (π radiation space), which causes a +12 dB amplitude gain compared to free field conditions. In this case the gain is beneficial, as it provides additional headroom and/or less distortion. Corner placement adds another +6 dB to the situation. However, at subwoofer frequencies the source location has a strong influence in the modal excitation of the room, and the response will depend on both source and listening locations.

Reflections and Cancellations

Reflections form what we perceive as room acoustics. As said earlier, nobody wants to work in an anechoic chamber, and therefore reflections are a good thing. Reflections in the control room will occur from all boundaries and directions (side walls, floor, ceiling, console top, equipment racks, etc). There is a certain amplitude, time and direction range in relation to direct sound when reflections actually enhance the perception. Outside the range the situation deteriorates and hence the reflections should be controlled in such a manner that their effect on the monitoring work remains positive. The disturbing side is perceived as image shift, smeared image or changes of timbre.

The resulting irregularities depend on the distances between loudspeakers and reflective surfaces. Positioning loudspeakers as far as possible from reflective surfaces or minimising them will reduce the level of the reflections and move reflection-induced irregularities, perceived by the ear as smearing of the direct sound, to as low a frequency as possible, which is beneficial for imaging. This kind of early reflections are generated by reflective surfaces (19" outboard racks, keyboard racks, computer tables, etc) placed between the loudspeakers and the listening position. Occasionally a large DAW screen is placed so that it is on the direct sound path between the loudspeaker and engineer. This shall be avoided as it shadows the direct sound from the loudspeaker. Strong reflections, which cause cancellations below 1 kHz and thus affect the frequency response balance and perceived timbre, should be eliminated.

In general, the further away from the listening position we place the loudspeakers, the more reflected signal we will listen to. If reflection levels in the room are too high compared to the direct sound, the sound field and the dynamic sound object panning become smeared. By increasing the direct-to-reverberant sound ratio in the room, the localization and the uniformity of the sound field will be improved. However, as the ear perceives both direct and reverberant field, both should have an equal frequency balance. As the reverberant field is excited mainly by the off-axis radiation of the loudspeaker, such off-axis performance is in this respect very important. Off-axis response, or to be more precise, power response, should be flat and uniform. To achieve this, loudspeakers should have controlled directivity. Furthermore, the room surface characteristics (reflection, absorption, diffusion) should not depend on frequency. In typical control rooms these characteristics should be uniform down to about 200 Hz. In the time domain the reflections should be substantially identical from both the left and right sides of the room.

A special boundary interference to consider at lower frequencies is the 'back wall' cancellation effect, generated by the single reflection from the wall behind the loudspeaker. When two identical signals are out of phase, they cancel each other. This happens when there is half wavelength phase difference, i.e. when the loudspeaker is quarter wavelength from the wall. The importance of the cancellation depends on the distance (affecting frequency) and the reflection coefficient of the wall (affecting cancellation notch depth), but is usually well audible. A 100 Hz notch appears when the distance is 85 cm. Often a wide well is seen in frequency responses associated with perception of missing bass. The cure is to move the loudspeaker either very close to or further away from the wall. Flush-mounting is beneficial.

Reverberation Time

Reverberation time is defined for diffuse sound fields and therefore actually not applicable in small acoustic spaces. Also, it should be measured using omnidirectional sound sources, which is seldom the case in control rooms, where the sound source used is usually the monitoring loudspeaker. Reverberation time should also be measured at several frequencies. In most cases it would be better to talk about modal decay times, as most of the issues of interest are in the frequency range where individual room modes dominate.

Low Frequency Damping

The very low frequency damping is an issue all of its own. Its implementation does not vary much from standard stereo room to multichannel room design. However, if rear loudspeakers are also flush mounted in a sufficiently large wall, the space available for conventional absorbers may become limited. In general, lack of bass trapping becomes more problematic in multichannel control rooms, as there are multiple low frequency sources in different locations. It may sound a bit strange but this fact may also be part of the solution, as separate subwoofers can be placed in such locations where they do not excite the dominant room modes. All this should highlight to designers the importance of carefully controlling the low frequency energy decay and modal behaviour of multichannel control rooms.

2. Today's Control Room Environment

After many years of collecting in-situ measurements of our loudspeakers during calibration sessions, a statistical analysis of 372 three-way Genelec loudspeakers in 164 professional control rooms was done. 75% of the impulse responses had been measured at the engineer's position. In most cases the physical aiming was correct, as 90% of the measurements matched also with the acoustical axis of the loudspeaker. Some of the findings of that analysis are highlighted here.

First observation was that in many surround setups the actual loudspeaker positioning was significantly different from the recommendations. Also most free standing loudspeakers were closer than the generally recommended 1 m minimum from the neighboring walls. It is clear that flush-mounting, even for small loudspeakers, would bring audible benefits.

Low Frequency Response Irregularities
Most of the frequency response irregularities found below 1 kHz were produced by destructive interferences due to low order reflections. Considering the notch distribution and spread, notches around 100 Hz were much more common than at other frequencies. This matches well with the reflection from the wall behind the loudspeaker. A distance of 85 cm corresponds to a 100 Hz notch. Also, the median depth of the 10 deepest notches in each frequency response measurement was -14.2 dB! This shows clearly that room boundary reflections can be very strong. The interfering signal needs to be almost as strong as the direct signal to produce notch down to -30 dB.

Resonances associated with room modes begin to dominate listening location characteristics below 200 Hz where the wavelength becomes large compared to room dimensions and size of objects in the room. At higher frequencies notches were created because of low-order reflections in the room. Typical reasons for notches were first order floor reflections, interaction of the console with the first order floor reflection, incorrect front wall design (e.g. discontinuities in the front wall, such as non-aligned windows, large TV screens or cavities). Also, in the case of soft front wall constructions, insufficient bass trapping will cause notches due to reflections from the hard wall behind the loudspeaker, just like with free standing loudspeakers.

Magnitude Response
All loudspeakers included in the analysis fulfilled the requirement of ±2.5 dB passband flatness in anechoic conditions. On the other hand, frequency responses measured at the engineer's position using third-octave smoothing should be flat within ±3 dB from 50 Hz to 16 kHz, with some level reduction allowed towards higher frequencies. We found that in multichannel rooms the sound pressure level 50% distribution limits indicate a good frequency response control above 1 kHz but an increasing spread below this frequency. Opposite to this, the 50% distribution limits in stereo rooms remain consistently within ±3 dB bounds also at low frequencies. The longer experience in building stereo control rooms showed up in these results.
Still frequent failures exist in low frequency design and in the management of low order (early) reflections. Many, but not all, large modern control rooms achieve adequate low frequency damping by having a properly designed acoustic treatment. Small control rooms with free-standing loudspeakers and minimal acoustical treatment often exhibit large variations in low frequencies. Unfortunately, the trend leads to use acoustically non-treated rooms as control rooms with the addition of tighter budgets for room acoustic design.

Reverberation
In the analyzed large control rooms the mean reverberation time was 380 ms from 200 Hz to 4 kHz. Considering that most of these rooms were quite large, as control rooms, this is reasonably short and controlled. The reverberation time was usually longer when diffusers were used to control the reverberant decay field. On the other hand, some rooms exhibited very constant reverberation time over the whole frequency band, including low frequencies. This indicates that there is still work to be done to increase the quality of monitoring room design, especially at low frequencies.

3. Equalization

The purpose of electronic equalization is to improve the perceived sound quality. Such techniques have been widespread for at least 40 years. Equalization is applied either to improve the loudspeaker’s inherent response or to correct phenomena caused by the room. Equalization is often implemented using third-octave or parametric equalizers, which are normally set with the help of real time analyzers. Room response correcting equalizers are now also increasingly built into active loudspeakers, but these equalizers have different approach to address acoustic problems: the idea is to gently shape the acoustic response. Conventionally, the target response is a flat magnitude response and linear phase response within the pass-band of the loudspeaker. Most room related issues are of minimum phase and hence the equalization can improve the perceived response.

However, real-time analyzers measure the steady state signal and cannot distinguish different arrival times nor different arrival directions. A notch caused by a destructive cancellation looks just like another notch, but an attempt to equalize it, in this case, will not work as the root cause still remains. Aberrations above one or two kHz are likely a result of either non-uniform off-axis radiation of the loudspeaker or frequency dependent absorption of the room walls. Attempt to compensate for these will affect the direct sound balance and will change the perceived timbre. Very high Q room modes are often blamed but a long decay time means also slow excitation. Room modes are inherently minimum phase and can be addressed with narrow band equalization.

Obviously equalization has to be applied with proper understanding of the relevant phenomena and only within a frequency range where it can improve the perception. Correcting fine details is unnecessary because the ear/brain is more sensitive at detecting wideband imbalances than narrow band deviations in the magnitude response. This underlines the importance of primarily solving acoustical problems by correct loudspeaker design and placement as well as room acoustic treatment before using equalizers.

4. Automated Room Response Correction

Background

In the 70's, the writers of the original N12 specification for monitoring conditions in Nordic broadcast control rooms were actually very modern thinkers. Probably the most advanced requirement was that the monitoring loudspeaker frequency response shall be measured at the engineer's position in the listening room. The response was defined with acceptable tolerances in real life conditions, not in a test chamber.

This led to the question how to guarantee meeting the specification in varying room conditions. The solution was to include some sort of frequency response adjustments in the loudspeaker. In the mid 90's, as mentioned before, we started collecting data from studios worldwide to analyze what the spectrum of acoustic conditions really is and how the products are set up. Examples of the results are the compensating switch to eliminate the 160 Hz boost caused by mixing console when near-field loudspeakers are placed on or near the meter bridge, included now in the larger Genelec 8000 series 2-way loudspeakers, and the DIPtimizer software, which calculates optimum DIP-switch settings based on the measured response, that works in conjunction with the WinMLS measurement software.

Applying DSP

DSP crossover filters have existed for more than 15 years in sound reinforcement and in the first active loudspeaker systems with DSP crossovers that appeared in the consumer world. Typically the frequency response was ruler flat but perceived subjective performance was not as good as one could expect. At the same time understanding of the criteria for excellent subjective performance was somewhat limited. Later it has become common knowledge that excellent on-axis performance is not enough; the off-axis and power responses are equally important. Equalizing even complicated errors is possible, but as explained earlier, the problem of the listening position remains: optimizing response in one point in space often means less desirable response somewhere else. The old wisdom is true also here: it is better to prevent the errors from happening in the first place than to have to correct them afterwards. Therefore the starting point in all respects should be pretty much as good as it can be.

So what are the benefits of using elaborate technologies? Steeper crossover filters can improve directional and off-axis performance. More exact equalization of driver unit magnitude and phase responses is also possible but, if the starting point is excellent, the audible improvement may be small. Correcting the room response is a very important feature, which can clearly improve the perceived response.

Developing The Right Products

The idea to build loudspeakers for monitoring digital audio is not new but the full potential of digital signal processing has not been entirely exploited until now. After years of painstaking research the new Genelec 8200 series DSP monitors and 7200 series DSP subwoofers feature sophisticated digital signal processing to achieve the next level of resolution as accurate reference monitor devices. We were not interested in developing a product to ‘emulate’ older monitors, or provide choices of ‘typical’ responses’ but to design an accurate tool serving our customers precisely and effectively. Over the years we have come to recognize that most of our customers do not pay much attention to their loudspeakers, but expect them to perform effortlessly and reliably, so our development effort has been directed towards optimal and quick adaptation to the room acoustic environment, ease of use and reliability.

Today the monitoring chain is turning digital. As consoles become truly “digital” -when the consoles start passing digital audio from the monitoring volume potentiometers - fully digital loudspeaker systems have a pioneering chance. Currently the overwhelming majority of consoles still have analogue monitoring sections. Considering this reality, our DSP loudspeaker systems were designed in such a way that they can be used in both environments: with analogue or digital input signals.

The AutoCal™ Tool

It is more the rule than exception to read or hear comments of too much or too little bass, added with the variety of colourful terms, in describing the unsatisfactory control room sound reproduction quality. Loudspeakers are usually blamed. Although the perception is true, in most cases there is no problem in the loudspeaker itself. The reasons are either in the room or in the placement of the loudspeaker into the room, or both. Probably this is the area where the most audible results of the power of DSP can be achieved. However, the magnitude of the aberrations a room or wrong placement can cause may exceed any reasonable correction capacity,

and hence everything possible should be done to remove the root causes, as said earlier.

Genelec AutoCal™ is a fully automated acoustical calibration tool for a single room multichannel loudspeaker system. The AutoCal™ system uses loudspeaker-generated log-sweep sine signals recorded by a calibrated high quality microphone to determine the correct acoustical alignment for every loudspeaker and subwoofer on the Genelec Loudspeaker Manager (GLM™) control network. One (SinglePoint™) or more (MultiPoint™) measurement positions can be used. MultiPoint™ is a central feature of AutoCal™, and is especially useful for rooms with multiple operators or with producer listening positions. Once the frequency response of every monitor and subwoofer is calculated, AutoCal™ determines the correct acoustical settings for flat frequency response at the listening position (or an average over the MultiPoint™ area), aligns for equal delay from all monitors to the primary listening position, and aligns output levels and subwoofer/main loudspeaker crossover phase. There are several internal algorithms to match the applied correction to the perception capabilities of the human ear. All calibration information is then saved in a single ‘System Setup’ file that can be recalled in less than a second.

Each 7200 series subwoofer has four notch filters to be set up while the 8200 series loudspeakers have four notch filters, two high frequency shelving filters and two low frequency shelving filters. The automated calibration procedure aligns distances within 1.5 cm (3/4”) and levels to within 1 dB. A typical 5.1 system takes less than five minutes to calibrate.

The technology in Genelec 8200/7200 DSP systems is not there to correct and fix mistakes in the electro-acoustic design or in the room acoustic, but rather to offer improved usability and consistency for often complicated and rapidly changing productions, and to provide an extremely efficient and effective tool for integrating loudspeakers into the acoustics of the listening environment. The monitoring system should adapt to the working needs and production environments as far as possible, not the other way around. Built on the solid acoustical foundation of the 8000 and 7000 series products, these new products are the logical step in the line of development to make the end user’s work easier, more enjoyable and hence more productive.

Christophe Anet & Ilpo Martikainen

References

A Survey Study Of In-Situ Stereo And Multi-Channel Monitoring Conditions
Aki Mäkivirta and Christophe Anet,
Genelec Oy, FIN-74100 Iisalmi, Finland.
AES 111th Convention, 2001 September 21–24, New York, NY, USA.

Compensating the Response of Near-Field Loudspeaker Monitors to Minimize the Effects of Desktop Acoustic Loading
Andrew Goldberg, Aki Mäkivirta and Ari Varla,
Genelec Oy, FIN-74100 Iisalmi, Finland.
Journal of the Audio Engineering Society, Vol. 54, No. 5, 2006 May.

Performance Comparison of Optimised Equalisation Techniques for In-situ Loudspeaker Measurements
Andrew Goldberg and Aki Mäkivirta,
Genelec Oy, FIN-74100 Iisalmi, Finland
Institute of Acoustics, 21st Conference on Reproduced Sound, 2005 Nov, Oxford, UK.

Modal Equalization of Loudspeaker–Room Responses at Low Frequencies
Aki Mäkivirta (1), Poju Antsalo (2), Matti Karjalainen (2) and Vesa Välimäki (2)
(1) Genelec Oy, FIN-74100 Iisalmi, Finland.
(2) Helsinki University of Technology, Laboratory of Acoustics and Audio Signal Processing, FIN-02015 HUT, Espoo, Finland.
Journal of the Audio Engineering Society, Vol. 51, No. 5, 2003 May.