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In late 2013, I was asked to build an extremely small room for a well-known musician with a house in Marbella, in southern Spain. He wanted to use it for the recording of voices and electric instruments, as well as doing some mixing. The problem was that it was only 2.5m x 2m, and just less than 3m high. Initially I refused, as I doubted that the room could live up to his expectations, but pressure was brought upon me because various people were depending on it. Eventually, I agreed to do it, but not without some reservations.

To do the job meant two trips by air to Marbella, about 14 nights in hotels by the beaches (just to design, and then oversee the job) plus a week of hotels for two carpenters and 2,000km of travelling by road with their tools and materials. Add to that a local electrician and the costs became outrageous for a room so small. Nevertheless, the musician had some serious work to do, he badly needed the room, and he was very pleased with the final results. Of course, when you can apply this amount of resources to such problems, the desired results can often be achieved, but it usually defeats the purpose of working in small rooms in the first place. The most usual motivation for people working in small rooms when dealing with sound production and postproduction is because their economic restraints preclude paying for anything bigger. Spending 25,000€ on a room of 15 cubic metres is usually a non-starter!

Some months later, while in Lisbon for a meeting about designing a new, small dubbing theatre for mixing cinema trailers, I was taken to hospital with a serious infection. During my stay, and the subsequent months of convalescence and recuperation, I had plenty of time on my hands to deal with some enquiries from people wanting better acoustics for their small studios, but who had neither the space nor budgets for more normal techniques to be applied. Well, getting good results in 2.5m x 2m is still stretching your luck without a reasonably decent amount of money to spend, but in 4m x 4m, as long as no significant extra sound isolation is required, some remarkably good acoustics can be achieved for a very reasonable amount of money.

If high degrees of sound isolation are required there are no cheap solutions. Normal systems of high isolation often tend to be space-consuming and heavy, and function by preventing the sound from passing into or out of the rooms. As a result, if the sound cannot leave a room, the highly reverberant internal space which results will require a great deal of acoustic treatment to absorb the sound energy. However, in very small rooms, the space to do this across a wide spectrum of frequencies may simply not be available, although some degree of low-frequency absorption and lighter-weight isolation can be achieved by means of covering the entire surface of the room with open-cell synthetic foams with a mass layer on top. The foams are bonded to the walls and ceilings.
with adhesives, and then the mass layers (such as plasterboard) are stuck to the foam with yet more adhesive. On top of these, the acoustic control materials for the higher frequencies need to be mounted, with suitable depth and spacing. The whole thing can usually be done in less than 20cm, but treatments like this can easily cost 80€ per square metre just for the materials (i.e. no labour), and every single metre of the room must be covered if the isolation is to be effective. On the other hand, if a room is already in use, and/or isolation has proved not to be an issue, much simpler acoustic control measures can be applied, although still with a ‘loss’ of space of around 15 to 20cm. Figure 1 explains the general idea.

A serious drawback with physically small rooms is that the reflected sound returns to the ears from short distances. This means that, for any given wall surface, the reflections return more quickly and with more energy than they would do in a larger room of similar construction. The corresponding reflections of the reflections also build up more rapidly and energetically, so the reflection density can be much more audibly colouring than in larger rooms. Essentially, if we must treat a small room with the goal of achieving good, neutral acoustics, we do need to treat all of the surfaces except the floor. There is no magic device that we can put in a corner to suck the sound from somewhere else. If sound can bounce back to the ears, it will!

Some idea of what can be done to get reasonable acoustics in a tiny room can be seen in Figures 2, 3 and 4. In this case, the stone-walled room was lined with 5cm metal studs, to which plasterboard was fixed. On to this surface was glued 6cm of open-cell, polyurethane foam, which in turn had Celenit glued to it. The various layers can be seen in Figure 2. Sonically, the procedure removed almost any trace of room sound. Aesthetically, however, the owner wanted a ‘cleaner’ appearance. Figure 3 shows the surfaces covered with open-frame panels, cut from sheets of 10mm plywood, which also had the mounting holes for the lights, power points and ventilation grilles, etc. Figure 4 shows the effect after an acoustically transparent fabric had been stretched over the frames. The camera flash reveals the wooden frames, but they are actually much less visible in normal lighting. The result is a very clean room, both acoustically and visually, in which the sound is very natural.

Contrary to popular belief, there is no such thing as electronic ‘room equalisation’. The things that take place in a room that modify the direct sound are all position dependent, so no correction applied via the loudspeakers can...
remedy them. It may be possible to correct certain acoustic characteristics at certain positions for certain sources, but an improvement at one location usually means a worsening of the situation somewhere else. Equalisers tend to move problems rather than fix them. Furthermore, there are only some of the room characteristics that can legitimately be equalised in terms of true correction, and you cannot tell one type from the other by viewing the display on a spectrum analyser. The spectrum analyser lumps together the power of the direct sounds, the reflections, the resonances and the ‘reverberation’ (the energy decay tails). It may well be possible to use equalisation to make it look as though a response in a room has been flattened, but our ears may not agree with our eyes. As Floyd Toole pointed out in a communication to the SMPTE (4 August 2010), ‘Unlike a human, the microphone does not take any note of the angle of incidence of the direct and reflected sounds, nor does it make any allowance for the time of arrival of those sounds, nor does it acknowledge spectral variations among any of the sounds. The microphone simply adds them together. …..It is well known that two ears and a brain are vastly more analytical than a microphone and an analyser. Humans respond differently to sounds arriving from different directions at different times’.

A resonance occurs around a set frequency, and while its amplitude may vary at different positions in the room, its frequency does not. Perhaps equalisation can deal with resonances to some degree; at least to mitigate their worst effects. On the other hand, a reflection gives rise to peaks and dips in different positions at different frequencies. These cannot be equalised, but even if such equalisation could be attempted, it will be noticed that the amount of equalisation applied electrically will not correspond with the change seen on the analyser. This is a sure sign that this is an unequalisable characteristic, and the ear will tend to hear the applied equalisation rather than the effect shown on the analyser. This is a recipe for colouration of the sound; and it is not likely to make an overall improvement. Most of the acoustic problems in small rooms are of a non-equalisable nature, and so the only real solution is to deal with them at source.

What is more, using acoustic diffusers to break up reflections only works at distances where the returning energy will be truly diffuse and low enough in level to not detract from the clarity of the source. In small rooms and at short distances these characteristics cannot be achieved. Diffusers at close distances can sound very unnatural, and can even concentrate the returning energy at some frequencies more than a simple, flat, reflective surface. The only effective solution for sonically neutralising small rooms is to absorb the majority of the sound energy, over a wide band of frequencies and in as uniform a way as possible. A hard floor, along with the introduction of a reasonable amount of recording and mixing equipment, will then add all the necessary life required to avoid the sound of the room being too oppressive to work in comfortably, even if all of the other surfaces are highly absorbent over a wide range of frequencies.

Absorbing the high frequencies is a trivial task, but the decisions about precisely how to achieve the necessary degree of lower-frequency absorption is something that will depend on the nature of the outer structure of the room. If a basic room is made with plasterboard walls and/or ceiling, these may already act as useful low-frequency absorbers and simplify the subsequent treatment (as explained in Figure 1). On the other hand, if the room is built from solid stone, with a concrete floor and ceiling, the inner treatment may need an entire lining of a plasterboard/deadsheet sandwich, spaced away from the structure, to reduce the energy in any low-frequency reflections. Without reflections, neither significant reverberation nor resonances can build up, so the room will exhibit a short decay time. With the short decay time comes the neutrality.

In Figure 5, a free-standing framework can be seen, with a plasterboard layer on the outside to deal with the low frequencies. The plasterboard lining will still be reflective to the mid and high frequencies, so a useful way to control this is to use a wood-shaving/cement/air material, such as Celenit, Heraklith, Savolit, or Tectum. If the spaces between the studs are then lined with a fibrous absorbent material of about 40 to 80kg/m², wood/ cement/air boards of 3 to 5mm thickness can subsequently be fixed to the front surface of the studs. The overall result of this combination is usually very effective; an example of which can be seen in Figure 6. The ceiling would normally be of a similar construction, although with a deeper cavity between the plasterboard backing and the inner facing. This is partially due to the need for
the ceiling beams to be deeper to provide sufficient rigidity and also because of the need for more absorption because the ceiling is opposing a highly reflective floor. Figures 1 and 2 show this greater depth.

It should be noted that rooms built in this way are relatively free from any worries about their dimensions. There is no need to angle any walls or avoid square rooms, for example, because the absorption is sufficient that the shortness of the decay time renders the effect of the room dimensions largely innocuous. It should also be noted that trying to calculate modal frequencies in such rooms, using conventional formulae, is futile. The formulae only apply in rooms with rigid walls. Once diaphragmatic and porous walls are introduced, the phase changes which accompany the high absorption make the acoustic dimensions very different from the physical dimensions. First-order reflections will tend to predominate over resonances, so it is important to note that neither can the decay times be calculated from any simple formulae because the absorption coefficients of the surfaces will depend on the angles of incidence of the sound waves. Standard absorption figures are for the materials when placed in a random-incidence, statistically diffuse soundfield. They are usually measured using omnidirectional sound sources in reverberation chambers. However, when the decay times are very low, the reflections are so limited in their ability to bounce repeatedly around that no diffuse field can ever exist, so the standard absorption figures become meaningless.

In the case of a pair of monitor loudspeakers facing across a room, the rear parts of the side walls tend to receive grazing impacts, whereas the rear wall itself receives a plane wave at perpendicular incidence. The absorption characteristics of the walls can therefore be very different, even if their structures are identical. Directly alongside the loudspeakers, the reflections from the side walls are less problematic because the central low-frequency content in the stereo image is actually generated from two, laterally-displaced sources: it is not driven from one place in the centre of the room. On the other hand, the left and right loudspeakers tend to be equidistant from the rear walls, which are therefore often treated with an extra depth of absorber to deal with the direct, simultaneous impact of the sound from both loudspeakers, to reduce the degree of low-frequency reflection. People often fail to understand that identical surfaces on different walls do not necessarily absorb to the same degree, therefore simple absorption calculations cannot apply.

Recently, I have been talking a lot of clients through these types of room treatments on three different continents. They can be very effective, and are unaffected by insects or humidity, so can be used in any climate. As mentioned earlier, though, each room has its own starting conditions, so the precise solutions are also different in each case. Nevertheless, the basic concepts are very adaptable to different circumstances. Some people are happy to work with the wood/cement-board finish, as shown in Figure 7, while others choose to cover the surfaces with acoustically-transparent, fabric panels, as shown in Figure 4. Mounting lights in the ceiling is no problem either, because the wood/cement/air panels are non-inflammable. These treatments are proving to be an inexpensive and practical solution to the small room question, and the results can be very good indeed.

Figure 7. Some owners choose to leave the rooms with the natural finish of the boards, which are available in a range of colours. (Photos for Figs 5, 6 and 7: Nick Smith, Wales)
Small multichannel control rooms for broadcast

DONATO MASCI, in collaboration with Genelec's AKI MÄKIVIRTA, illustrates some of the solutions he has designed and the results obtained from small multichannel control rooms.

In the world of broadcast, postproduction and audio for video is always chasing new technologies yet when I visit these types of facility I come across innovative technological solutions built in to acoustic environments that have a lower quality than the monitoring standards I am used to in music studios. The worst problems stem from the fact that, except for large mix theatres, audio for video is increasingly produced in very small rooms — often with inappropriate dimensional proportions, such as square rooms or in OB vans. These rooms often do not use the same monitor type for all channels — at least for 5.1. In general, it is much more difficult to design a small room than a larger room with traditional dimensions built using a classic design (Non-Environment, LEDE, RFZ, etc. that imply certain minimum dimensions). This is because room simulation methods are not typically used when designing control rooms having standard designs. The traditional acoustic modelling software based on ray-tracing methods does not properly work at low frequencies where the fundamental problems in small spaces obviously occur.

Yet the market demands that facilities adapt to smaller, usually very urban, working environments that have to look good enough to host client quality checks and sound good enough to pass client's stricter QC departments. In this article I will illustrate some of the solutions I have designed and the results obtained from small multichannel control rooms, explain the project's ideas, the design steps and the problems that I have encountered.

The problems of designing small rooms are the same as with larger rooms, but need more attention because the problems can be larger due to the small room size. To understand how a control room works you observe what happens in the acoustic field generated by the monitors. The good features of rooms do not only depend on their passive acoustic properties, like absorption and diffusion, which determine the reverberant field. Acoustic parameters (such as reverberation time etc.) can easily be optimised by choosing materials with proper absorption or diffusion characteristics. You can do this with the help of the Sabine or, even better, the Eyring law. It is challenging to understand some devices' absorption properties, such as those for a bass-trap. The neutrality of the sound at the listening position depends largely on early reflections and the interaction between the sound source and the room. A studio monitor is usually considered neutral when the frequency response deviation is flat to about ±2dB measured in an anechoic room.

An optimal reverberation time is not enough for a control room to be considered 'professional'. The real target of creating the necessary neutrality for monitoring is obtained by having a flat frequency response and the optimal values for the room's acoustic parameters (reverberation times, early reflections, etc.). That's why, in order to understand the quality of a control room, all paths that sound takes when leaving the monitor, through reflections in the room, to the listening area must be carefully analysed.

One of the key points in small room design is the study of low frequency reproduction and therefore the modal resonance analysis. This is a complex and specialised subject. Books can teach you to calculate the mode distribution showing locations of SPL maxima and minima for a rectangular room and there has been much discussion on how to choose optimal room dimensions to distribute the mode resonances evenly in frequency. Unfortunately in most cases the rooms are neither rectangular nor can their dimensions be easily chosen with an optimal dimensional ratio. This is not necessarily a big problem because the amount of sound absorption needed in these environments is considerable to achieve acoustic performance in accordance with AES, EBU, and ITU guidelines. This makes the choice of the optimal room dimensions less of a problem. It is also very difficult to simulate standing wave distributions in a room because of varying geometry and the real-life acoustic impedance of the wall. Even with strong sound-proofing, the walls can never be assumed rigid, especially when using a drywall construction.

The most practical way to address these problems is through the use of a FEM (finite element method) simulator such as Comsol Multiphysics, which can also perform optimisations. This can provide important information about isolating layers, positioning of the sound-absorbing materials, and the optimal listening position.

Another key aspect of the room design is isolation between adjacent rooms. Usually rooms are placed close together to optimise space but have to operate independently and without sound leakage. Control rooms often share a vocal booth that is generally small with a high level of isolation. Isolation is the simple part of studio acoustics but determines the quality evaluation of the rooms. Choosing the right partition strategy is not enough, you have to pay attention to construction details, such as ceiling suspension with vibration damping hangers and the floating floor dampers, the HVAC system and audio cabling. All of them can transport noise between rooms.

The vertical partitions and ceilings are built with drywalls made with plasterboard, gypsum board, with different types of rubber and absorbent materials. Regarding the floating floors, the target is to build an efficient ‘mass-spring’ system that has a resonance frequency at least half of the lowest vibration frequency you want to damp. To lower this frequency you add mass and this is the reason why instead of multilayered drywall flooring with MDF top and cork, rubber and mineral wool suspension, a concrete slab placed on isopads is preferred.
An important aspect of the acoustics is the interaction with the reflective surfaces that are always present in a control room. When the listening position is very close to the monitors, it may seem simple to obtain a flat frequency response, but nearby hard surfaces can generate dips and comb filter effects. Aside from optimised acoustics, the room also needs good ergonomics for typically two people. We prefer to use projection on an acoustically transparent screen rather than a flat TV screen, which is a reflective surface. In my opinion, the traditional quadratic residue diffusers (QRD or Skyline) are not suitable for small rooms because the distance between the listening position and the diffuser is not large enough to feel the diffusion effect. In small rooms, my choice is usually to use a binary amplitude diffuser (E.C. Payne-Johnson, G.A. Gehring, and J. Angus — Improvements to Binary Amplitude Diffusers, Audio Engineering Society Convention Paper 7145). This is an absorber-resonator device with a panel featuring pseudorandom perforations, placed in front of a thick layer of sound absorbing material. This device provides good sound diffusion even at short distances and the effect extends over a much larger range of frequencies.

The last key aspect is choosing and placing the studio monitors. As discussed in my article Monitors in a room (Resolution V13.3), flush mounting can be really useful, even for small rooms as it eliminates many diffraction effects and improves the monitors’ low frequency output. With flush mounting, it is also easier to insert a projection screen. However, you have to pay attention to the design of the baffle wall because the front wall is not the only location for sound sources as often the surround monitor interaction with the baffle wall in the front is problematic. In these cases, it is preferable to finish the baffle wall with absorbent material. For stereo control rooms, I usually prefer reflective finishes.

The studio monitor choice is challenging. In some applications, 3-way monitors are needed, for example, in postproduction for control of the midrange frequencies in voices. Then you have to pay attention to the distance from the listening position. Let me illustrate some of the factors affecting listening at close range and to help you understand if the setup is OK and why.

The Figures show how the relative listening direction becomes increasingly different for a listener when he approaches the monitor. This will increase problems with flatness when the listener moves off-axis to some drivers (Figure 1); timing error between drivers, particularly at the crossover frequencies; and the definition of being on-axis becomes narrower when you approach the monitor (Figure 2). Driver directivity changes the frequency response in off-axis listening (Figure 3); the actual layout of the drivers on the front baffle can affect being more off-axis with certain drivers (Figure 4); when moving closer, the direct sound to reverberation ratio changes so the influence of reverberation goes down, and if there are diffraction effects in the monitor design, moving closer changes these in frequency, and their relative level compared to direct sound can change. Figure 5 shows the benefit of using a Directivity Control Waveguide. The DCW can maintain the frequency response off-axis and at close range can produce a more stable sound.
It is impossible to determine the minimum listening distance of a monitor by just looking at it as acoustical data and measurements are needed. Genelec 1238CFs were chosen for Fox International Channel UK Studios and these work at a range of 1.5m because the manufacturer has taken care of controlled directivity and minimised cabinet and other diffractions.

**Case study: Beep! Studios, Rome – Gorilla room**

This was the acoustic design of a 5.1 TV mixing room with a square area that was barely 9sqm with the contractor Proaudio Consulting. The brief was challenging; they wanted top audio quality standards and unique aesthetics to meet the room's name — it was used to host an ingest system called Gorilla and had to look like a jungle.

The fundamental problem was to contain the annoying resonances of a square room. In a square room acoustic modes overlap creating very peaky maxima and minima and to minimise the effects an 'old trick' was used. We turn the room by 45° and the real improvement in this design is related to the front side of the monitoring system. Every corner in the room has its acoustic treatment with sound absorbing materials to give significant absorption at low frequencies. The front corner of the room has been closed with a 45° heavy multilayer structure with differing acoustic impedances. This is not possible with masonry, and MDF, plasterboard, and rubber were used.

This wall structure, as well as dampening typical resonances of the square room, also serves to minimise the effects of phase cancellation of the LCR speaker system so typical with a conventional room corner. I mentioned this effect in last year's article Monitors in a room (Resolution V13.3). Applying the ITU circle with surround monitor angles of 120°, all monitors are now close to walls and the monitor arrangement is perfectly integrated with the room structure.

The small console table is designed to minimise reflection with a circular armrest. Another possible improvement would be to make the table top slightly tilted, to reflect sound away from the listening point, but in this case it would not be ergonomic because the table is also used for notes and scripts. Auto-calibration can help by compensating the remaining interactions that otherwise cannot be removed.

Proaudio's project manager wanted to deliver a unique-looking room and this led me in collaboration with Valentina Cardinali to design palm shaped acoustic diffusers and to use a bamboo cane finish for a panel at the back of the ceiling.
Case study: FOX International Channels
UK, London

This project was for the isolation and acoustic design of two 5.1 TV mixing rooms, sharing an isobooth and a video postproduction room, all to be hosted in a barely 42sqm area. The contractor was again Proaudio Consulting. The brief was close to impossible — top audio quality standards and branded aesthetics and we were provided with guidelines to follow that were many inches thick.

At a first glance this was an almost hopeless assignment, especially considering that the volume needed for soundproofing partitions had to be included in this space. I worked with Alessandro Travaglini, Fox’s resident sound supervisor and engineer in Rome from the very beginning, who was our client’s appointed technical interface.

We created a binocular-shaped shell to host twin 5.1 control rooms sharing an isobooth between (the postproduction room was easier to locate), optimising the use of the available space to the centimetre and considering all the technical requirements (D-Command, projector and projection screen, cabinets, extended table-console, 5.1, 2.0 and soundbar audio systems) as well as the ergonomics.

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We chose a Genelec 5.1 audio system of three 1238CFs for LCR and two 8250As at the rear plus a 7270A subwoofer. This design was a great challenge and needed a careful study to understand if it was possible to insert 3-way monitors flush-mounted in such a small space. After a series of trial designs we noticed that the listening position was located about 1.5m from the monitors as the manufacturer's minimum listening distance requirement and was enough to get a good listening experience. We rotated the DCWs because otherwise the tweeter would have been too high and would have been a problem for the projection screen placed in front of the monitors.

The front baffle wall is not only a technical element, but also a technical area for the cabling of the room. The baffle wall is partially covered and filled by damping material to avoid acoustic reflections of the surround speakers. At the back of the room there are the binary amplitude diffusers and absorbers I mentioned earlier, playing probably the most important role for creating the room acoustics.

The geometry of the side walls has been designed to not directly reflect audio waves to the listening position while creating a good view of the isobooth window. Another specular reflective surface was placed on the other side of the room to create symmetry with the window. This was used for the National Geographic rectangular logo and a Fox poster.

The ceiling has a thick absorbent layer and has reflective surfaces guiding waves coming from the front out towards the diffusers at the back, acting as a sort of geometrical ‘amplifier’ of the diffuser. The console was custom-designed to Travaglini’s concept and contained all the racks and workspace. For this reason the console has a very large surface and instead of tilting it I chose to cover specific areas with absorbing material to reduce first reflections from the L and R monitors. The reflections coming from the centre monitor are partially screened by the displays. This resulted in an extremely useful design. This project has been one of the most complicated I have ever dealt with but I considered it a challenge to overcome.

In conclusion, I’ve shown examples of how it is possible to build small control rooms for professional quality monitoring. Extreme care is needed at low frequencies and with acoustic reflections; a monitor auto-calibration system can reduce the inevitable remaining acoustic problems caused by reflections especially reflections from the console surface. However, I don’t want to be taken for an acoustician who prefers small rooms! Although these rooms sound very good and we are extremely proud of them, it is clear that working with traditionally sized rooms is much easier, the acoustic solutions are more natural and the performance results are higher. If you are not just looking at the acoustic parameters, this difference will be appreciated at the end of the work.

The next stage in the design of small monitoring rooms will surely be the inclusion of 3D immersive audio. Today there is no format that can be used in such small rooms as to-date only large mix theatres have these systems. But with the introduction of 3D audio in broadcast, for TV drama productions and sports events, the ITU recommendation for multichannel monitoring systems must be updated (or will need hybridisation) for systems such as Dolby Atmos and Auro-3D.

**FOOTNOTE**

Donato would like to thank his business partner Francesca Bianco, Proaudio Consulting’s project manager; Christophe Anet and Aki Mäkivirta (Genelec); Valentina Cardinali and Roberto Magalotti (B&C Speakers); Matt Ward (Spark + Rumble Ltd) and his friend Christopher Martinuzzi for reviewing the text.
Introducing nAxial

Exigy director MATT DOBSON explains the thinking behind the technology in his new monitoring solution that is eminently suitable for smaller multichannel and immersive environments.

Exigy is a UK company that makes direct radiator monitor systems for film, commercials, TV and music mixing rooms and we’re mostly associated with rooms at the larger end of those possible with non-compression driver systems. Since 2005 we’ve been making non-ported main monitors and in 2012 we stopped making ported systems altogether; much of our work is customised in some way, sometimes entire systems are custom.

In 2011 I had an idea for a new loudspeaker format and construction technique but from early 2012 till late 2015 we were flat out so patents weren’t applied for and detail development didn’t start till 2014, although many sketches, computer simulations and much thinking took place in the two intervening years. To test the theory and verify the concept, we decided to make a small self-powered design. With so much untried, plenty to prove but many interdependent aspects, we decided to make detailed prototypes rather than test rigs for each part, as custom system suppliers, that’s in the Exigy DNA anyway.

The concept relies on multiple identical drivers summing to a point-source equivalent when all are radiating from an area whose containing diameter is significantly smaller than the wavelength of the highest frequency they are to reproduce. You often see multiple LF drivers, 4 x 10-inch, 4 x 12-inch and 2 x 15-inch are common in big music monitors, they rely on this principal, it works but sometimes too high a frequency is attempted with a resultant ‘beaming’ and HF cancellations. Ideally, the group diameter will be less than 1/6th of the wavelength at the highest frequency to be reproduced, so a 4 x 12-inch group (contained in a minimum 750mm diameter circle) should not be used at higher than 76Hz (wavelength = 4.54m) if they are to sum perfectly and radiate without beaming.

As can be seen from the diagrams, while radiation begins to reduce from about 75Hz, at 350Hz it is only 6dB down at 90 degrees off-axis and the first full cancellation, at about 60 degrees off-axis, is around 600Hz. Considering these drivers will be crossed over to another (most often single) smaller driver at some point (with once again wide dispersion at its LF cut-off) and there will be some overlap and addition between the four 12s and this higher frequency driver, a more realistic crossover point might be 350 to 450Hz (contained diameter between 1 and 1.35 times the wavelength). Indeed, using linear phase LR crossovers, the summed response may be almost perfect across such a crossover point, at all angles and in terms of both magnitude and phase, provided the smaller driver is concentric with and has an acoustic centre at (or near to) the virtual acoustic centre of the four 12s — in effect the crossover point does not exist; it is blameless. Unfortunately, it is not often possible to get all drivers in a system to be concentric, or to align the acoustic centres. The above may be freely scaled up and down the frequency range applying to different sized drivers and crossover frequencies.

The Exigy idea is to have rotationally symmetric concentric groups, each group using smaller drivers, thereby simulating multi-channel performance. Neodymium magnet technology has given rise to small high performance drivers, the idea came at the right time. The goal for this prototype is a DSP-based, active, 4-way, fully coherent loudspeaker, less than 300mm in height, width and depth.

My preference is for wide dispersion designs; I’m not a fan of horn-loaded, highly-directive loudspeakers in small rooms (they make complete sense in large venues), I followed that path early in my loudspeaker design career but didn’t like the results. For a free standing loudspeaker, my ideal would be an infinitely small radiator capable of high output at LF, the mythical point source, obviously a physical impossibility. I don’t know the difference between ‘horn’ loading and ‘waveguide’ loading except that waveguides tend to mean ‘shallow angle’ horns. In fact a flat baffle is a ‘waveguide’ in that it directs waves forward from it, at least down to a frequency where it remains ‘big’ compared to the wavelengths being produced. Indeed a convex or spherical shape is also a ‘waveguide’ in that it guides waves emanating from its surface if they are small compared to the overall size of the convex shape. If you put a speaker on a shelf, the shelf becomes a waveguide (well, it’s sure going to guide some of the waves!) and at LF frequencies, the room itself is also a waveguide. So, we can’t get away from waveguides. For this prototype design we decided to take the simplest path and use a flat baffle, it is possibly the best compromise in small rooms anyway, that’s arguable and I’m sure many will wish to do so, as I say, personally I prefer the sound of wide dispersion or low directivity designs.

Further goals were:
• LF -3dB point at 45Hz in free space.
• 105dB SPL @1m per cabinet from 50Hz up, without exceeding driver xmax (linear excursion).
• Inertia cancelling LF driver arrangement.
• Individual driver enclosures of maximum dimension significantly smaller than the shortest half wavelength of the highest frequency in each band, so eliminating in-band internal reflections.
• All enclosure wall thicknesses to maximise the internal volume compared to external volume.
• Aluminium construction for every individual enclosure to promote heat dissipation. A low temperature gradient between the amplifiers and external skin so a very large surface area is available and temperatures are kept low.
• Four 250W amplifiers with a PSU peak to mean capability of 3:1 or better.
• Built in DSP processing.

Several driver layouts were looked at but groups of three worked out best in terms of packing density, cost, capability and performance versus cost, at least for this design. So, a threefold symmetrical layout for drivers and cabinet was chosen, the LF section folded back through 90 degrees to keep all 10 voice coils in close proximity and provide the sought after force cancelling arrangement.
I’d like to make some observations on these measurements. The calculated performance assumes an infinite baffle. The measured are with the loudspeaker on a stand, in a room and include some reflections. The measured LF performance looks better than calculated; this is just advantageous room loading. The increase in LF output at high angles is down to a short measuring distance of 0.5m, as the cabinet is rotated, the relative distance between the nearest driver and mic drops significantly; the calculated outputs assume a 1m distance. The prototype uses a 48kHz sample rate A-D convertor, hence the very fast fall in output above 20kHz.

Exigy is a small company so we decided to limit the design to use only ‘off the shelf’ main components, as such there are some aspects of these prototypes that are not fully optimised, we have shown that the theory works very well in practice and all who’ve heard them agree. We have solutions for the shortcomings discovered, we’ve not yet tried linear phase crossovers; development is ongoing.

We will be putting two pairs of these prototypes into London postproduction mix rooms in the near future to assess them in a professional environment. We are making a custom 5.1 system using larger cabinets; however, our first main application will be for use as Dolby Atmos surround loudspeakers where certain properties make them ideal.

Our new nAxial technology is scalable; we have preliminary designs for cabinets from a 3-way of about half the prototype size, right up to a Music Studio 5-way main monitor. We see the limits being approximately the same as for any direct radiator system, i.e. use in small to medium size rooms up to about 500 cubic meters with a mix position to monitors distance of no more than about 8m. Below a certain size, there is little to be gained over a single full range driver or small coaxial driver.

The primary advantages are:
• Very uniform output in all directions.
• Low distortion as drivers only have to work over their optimum frequency range.
• Almost total elimination of crossover artefacts, crossovers are almost ‘text book’ without any added delay.
• Scalable over a wide range of cabinet sizes.
• High output with extended LF from a relatively small size.
• Flexibility — many different variations are possible just by selecting different raw drivers and arranging them as required. Relatively high efficiencies are possible when using multiple drivers. Construction techniques do not require high tooling costs but may take advantage of them if high quantity production is sought.

The main disadvantages are the complexity, although this is probably not a problem in high quantity production, and cost — because of the construction techniques used, the complete loudspeaker weighs about 12kg. Ultimately, in high quantity production, cost is related to the quantity of raw materials used.
Development and scaling

JBL Professional senior systems engineer CHARLES SPRINKLE explains how the ‘space programme’ of the M2 monitor development has been trickled down to lower price points and also been applied to smaller immersive environments.

In 2013, JBL Professional introduced the M2 Master Reference Monitor, a new flagship level speaker that put large-format accurate monitoring within the grasp of a variety of production spaces. As a premium monitor, cost would not be the primary driver but it was still not an easy task. We wanted to be able to provide very high output, extended but neutral frequency response, with a compact footprint and tunability to optimise the interface of the system to the room. While the main attribute of a reference monitor should be accuracy and neutrality, the application of some new technologies would also allow us to deliver a most engaging listening experience.

The final M2 design leverages seven JBL patented technologies with two pending. At the heart of the M2 is the patented D2 Dual-Diaphragm Dual-Voice-Coil Driver. The D2 overcomes the limitations of conventional compression-driver technology: limited high-frequency extension due to mass of the diaphragm and voice coil, and distortion characteristics that arise due to dome breakup modes. The result is a device that delivers dramatically superior high-frequency response. Replacing the conventional dome design with annular diaphragms, the D2 dual driver design allows the M2 to meet seemingly opposing objectives: extended high frequency, very low distortion and very high output and eliminates the need for separate mid- and high-frequency drivers.

The M2’s 2216ND 15-inch Differential Drive Woofer offers very high output and very low distortion, with low frequency extension to 20Hz. The 2216D features five patented technologies to reduce the power compression that is detrimental to a system’s low frequency performance. Echoing the ‘dual driver’ design of the D2, the 2216Nd uses dual neodymium magnets and two voice coils and a special wire formulation that allows the woofer to maintain constant impedance regardless of output.

The D2 and 2216 gave the M2 very high output and extended frequency response in a relatively compact, 2-way design. But acoustic integration of the two is the key to system accuracy and sound stage. The M2’s patented Image Control Waveguide enables neutral frequency response, on-axis and off-axis, in the vertical, horizontal, and oblique planes, all the way down to the M2’s 800Hz crossover point and an ‘imperceptible’ transition between the two drivers.

Finite Element Analysis and rapid-prototyping equipment at JBL allow us to evaluate many iterations of a waveguide design, allowing our engineers to perfect the design. The geometry of this waveguide enables remarkable high frequency detail, soundstage, and natural balance at nearly any listening position in a broad range of acoustic environments. Since design of this complexity and level of detail is an iterative process, JBL performs Finite Element Analysis, creating virtual simulations of off-axis response, pattern control, and the blend of the high frequency driver and woofer.

For sound arriving at the mix position to be smooth and neutral, it is not enough for a speaker to measure flat on-axis and it is essential the speaker has excellent off-axis performance. JBL measures monitor systems over a sphere that encompasses all power radiated into the listening room — in every direction. This data reflects 1296 times the information of a single on-axis response curve. Seventy-two measurements of the direct soundfield, the reflected soundfield, and the reverberant field, the entire soundfield heard by the listener, is correlated to optimise response at the listening position.

Very positive response to M2 performance from music and film post clients indicated we’d achieved our objectives and affirmed the technologies would be useful in future studio monitor lines.

Following the introduction of the M2, we set out with the goal of delivering a new level of performance and accuracy at price points lower than those of any existing JBL studio monitors. We wanted to fill the needs of a very broad segment of the market that includes music recording hobbyists, professional video-post and broadcast facilities in need of an affordable reference monitor.

At the same time, we knew that the entry-level price segment of the studio monitor market was dense with powered models of varying quality from many manufacturers. We wanted to make a real difference at this price and we wanted the benefits to be clearly audible. Factoring in manufacturing costs, overhead and margin, the budget for the woofer, tweeter, electronics and enclosure was very tight. The key to finding the balance between professional-quality sound and affordability was the ability to incorporate our Image Control Waveguide in a cost-effective way. Using experience gained in the M2 programme, we completed development of the new affordable 3 Series studio monitors line.

As with the M2, the curvature of the Image Control Waveguide allows the directivity of the sound in the vertical, horizontal and oblique planes to be uniform. We designed the M2 and 3 Series waveguides based on the size and position of the drivers in each monitor and their crossover frequencies. The result is a seamless transition between high and low-frequency drivers and uniform directivity throughout the bandwidth. A benefit for studio customers is a wide sweet spot for neutrality across a broad listening area in front of the console or workstation.
The graph shows an LSR308 monitor and a competing, identically priced monitor. The graph shows measurements made at 10-degree increments from a point directly on-axis to a point 90 degrees off-axis. The LSR308’s response is free of dramatic peaks and dips while the other monitor has a non-neutral on-axis response, while its off-axis frequency response deteriorates progressively with each measurement.

The JBL 3 Series consists of the LSR305 and LSR308 internally bi-amplified monitors with 5-inch and 8-inch low frequency drivers respectively and 1-inch dome high-frequency drivers. While the 3 Series is very successful, it has always been our goal to expand our premium offering and deliver M2’s level of performance to a broader range of customers. We saw an opportunity: broadcast and film production environments have been gearing up for productions in immersive audio formats, such as Dolby Atmos, Barco Auro-3D, Isoson, and DTS MDA, which are surround sound formats that include overhead loudspeakers. As these formats gain traction in movie theatres, broadcasters and film studios are seeking to deliver the immersive audio experience to the home. While the large film mix dubbing stages are adding overhead loudspeakers to their JBL cinema systems, small and medium size production rooms also need systems to monitor their work in immersive audio formats.

We faced a number of challenges. Broadcast control rooms and mobile trucks are cluttered with video displays and finding a place to mount additional overhead speakers, surrounds and subwoofers is problematic. In addition, traditional studio monitors are powered and running power to all the loudspeakers is costly — as much as US$700 per outlet. The cost of outlets can exceed the cost of the loudspeakers.

In editorial rooms, a costly surround-sound controller is required and even when the same model speaker is installed in each room, varying room acoustics prevent room-to-room consistency. There is no assurance that product created in an editorial room will translate to the dubbing stage.

Last, we knew that traditional nearfield monitors don’t have the high dynamic range required for film production. Traditional nearfield monitors are not ideal for surround production and going forward will not meet the challenges of production in immersive formats.

So, we knew we needed to do three things: first, we had to make the speakers more compact; second, we had to make them more affordable; last, we had to simplify the product across the board.

This simplification of design objectives meant we needed to eliminate the need for electric power at each speaker, provide centralised control of the system; include room tuning filters for room-to-room consistency; make the system scalable to accommodate additional surrounds and subs; and provide digital and analogue connectivity.

In late 2014, we introduced the 7 Series Master Reference Monitors, specifically developed as part of a scalable installed system for broadcast and postproduction facilities engaged in immersive audio production. At the core of 7 Series performance is the driver technology of the M2 and the Image Control Waveguide.

The 7 Series models are designed around new JBL transducers specifically developed for these models: the 8-inch 728G and 5-inch 725G woofers and the 2409H annular diaphragm ultra-low distortion compression driver. These drivers deliver greater output than traditional dome high frequency transducers for greater working distances and dynamic range. Removing the electronics from the speakers allows very compact enclosures to minimise interference with sight lines. Birch plywood enclosures are front-ported, and include rear and bottom mounted speaker points compatible with a range of mounting hardware options. Although compact, the systems deliver extended low frequency response along with high frequency output to 36kHz.

The 7 Series speakers are centrally processed using BSS Soundweb London signal processors, which allow perfectly matched tunings for the LSR708i and LSR70Si models. User-accessible room tuning filters provide a more consistent listening experience in a range of postproduction and broadcast rooms. Session functions such as level, individual speaker solo, fold-down and more can be centrally controlled via BSS hardware, Harman HiQnet Audio Architect software, or tablet via a wireless router. Audio connections can be analogue or digital.

The system uses 8-channel Crown DCi8|300N amps to reduce speaker system size and weight, eliminate the need to provide electric power at each speaker location and reduce system complexity. BSS Blu-link networked audio in the DCi ‘N’ version amplifiers simplifies installation and reduces system noise. The LSR70Si and LSR708i Dividing Network can be bypassed for bi-amplified operation.

The system is cost-effective and as the channel count increases, the cost per channel decreases. In addition, the system can be scaled from 5.1 to large immersive sound systems.

While the M2 project was our ‘space programme’, yielding new technologies, it enabled us to apply these technologies to a most affordable 3 Series studio monitor line available for a starting price of US$150 each. With the launch of the new 7 Series, world-class technologies and listening experience are now available from JBL at a wide range of sizes and price points.
Neumann Monitors

Neumann monitors are made in Ireland in a high tech factory and in a high tech way. ZENON SCHÖPE

Neumann monitors are carefully assembled in Ireland by a skilled and dedicated workforce within a factory that is responsible for a broad range of headphones for the Sennheiser brand. It’s a surprising twist but not an illogical one when you consider that both are effectively output transducers; it’s really only about a matter of size. The headphone models cover a lot of hi-fi types but in pro terms you are likely to be interested by the fact that the HD 25 family is built here too. While the aforementioned difference in size is distinct, so too is the matter of scale.

The headphone manufacturing is heavily automated with extremely sophisticated machines running unaccompanied through three shifts for true 24/7 production and involves a level of intricacy and speed that is as mesmerising as it is breath-taking. It’s delicate work that requires extreme precision in such things as coiling superfine wire on miniature drivers and applying miniscule amounts of glue at precisely the right point and time. There are parts going up via bridges and going over to the next machine for the next stage of a process, like some massive sushi bar conveyor belt of headphone parts.

Apparently there’s a headphone transducer dropping off a line somewhere in the factory, which employs 150, every five seconds. Automated testing of components is built-in to the manufacturing and every pair of headphones has to pass a full suite of tests before it goes off for packing.

While the quality ideology is the same — embedded by more than a little of that Neumann kudos — the manufacture of Neumann monitors at the plant is less frenetic but just as thorough but we’ll get to that in a while.

Sennheiser first moved manufacturing to Tullamore in Ireland in 1990 for the production of aviation headsets as a way of remaining competitive on price compared to manufacturing in Germany. In any company you invest in people, facilities and support and that doesn’t necessarily have to be where you are based or originate from. The experiment went well and other headphone models were added to the point where they outgrew that first facility and moved to the bigger present plot in 2002. They created a very flexible manufacturing footprint with the view to being able to adapt to different product types in the future.

Tullamore is near the middle of Ireland and is well connected to the ports and airports of Dublin and has a skilled workforce pool as there is other manufacturing in the area. There’s also a local technical university and many of the first technicians came from there.

The decision to move the manufacture of monitors to Ireland happened before the Klein + Hummel brand that Sennheiser bought was placed under the Neumann brand.
That manufacturing started in 2006 and is indicative of a corporate attitude that really does believe in developing internal skills and keeping them in-house. Neumann took over the monitoring brand in 2010 together with an intention to develop it. K+H was a specialised German monitoring brand with a Made in Germany image that sold mostly to German speaking markets. The decision was made to look at how they were built, to improve the ease of manufacture and to work towards making more of them.

If you think the Neumann brand is only about mics then you betray your relative youth; the 85-year-old company built its reputation on a broad range of products that included mics but also such things as consoles and disc cutting lathes. While it never made loudspeakers it probably could have done, so monitors are a good ‘fit’ for the firm. They took on the engineers formerly with K+H who were then with Sennheiser and in Neumann got them to change a lot of things.

The old and original production line in Tullamore was taken out, refashioned and rebuilt with feedback from the manufacturing staff and with a view to making lives easier and making consistently better products, as Neumann president, Wolfgang Fraissinet, explains. ‘We were looking for ways in which we could increase the consistency and the quality level. Once you have a Neumann logo on a product the expectation gets high and we set the bar high,’ he says. ‘We looked at ways to increase the efficiency of production but without diminishing the quality level. The KH 120 was the first as the successor to the K+H 0110 and we have increased the volume we sell every year by a substantial amount. Our capacity in production is high; we could make even more.’

It’s important to accept that Neumann monitoring products are re-engineered variants of former K+H products and there was a loyalty and pride attached to the manufacture of the K+H products at the Tullamore factory. Wolfgang says he was acutely aware of this when he visited the factory that would now make Neumann monitors. ‘I said that changes had to be made in manufacturing and that with K+H we had allowed ourselves to have rejection rates in production in two-digit percentages,’ he remembers. ‘I said we would like to reduce that to one digit and measure it in permil not percent. I know they thought I was a bit crazy but the rejection rate now is 0.3%.

Changes were made to everything from the

**CONNECTED COMPANY** — In a move that has been largely transparent to end users Sennheiser and Neumann have been doing quite a bit of internal reorganisation to move themselves closer to their customers. The Connected Company initiative is about dividing the business into different sectors. So instead of just dealing with microphones for all markets, for example, they are concentrating instead on the buyer and the application of that buyer, as Wolfgang explains. ‘It’s a different thing selling a mic to a broadcaster or to a theatre or to live or to a studio,’ he says. ‘When you focus on those customers specifically you think differently when you develop products for them. You also think differently about platform development.

‘If you fine tailor from a conceptual point of view more towards a target group of users, by the time we design it there may be derivatives that are more broadcast, live and studio related,’ he says. ‘They are not three different developments but they are satisfying exact demands of three different types of customer. This has implications not only for the technology of the product but also how we communicate it to a customer.

What this initiative aims to create is smaller and more agile units dealing with and serving defined market segments. This will also manifest itself in the Neumann monitor range which will have to apply its development to the needs and requirements of the market sectors that it will operate in.

‘It a change and it’s good to have a change,’ adds Wolfgang. ‘It’s not just something for the youngest people coming from university who are used to change because they haven’t really go used to anything stable yet, it’s also an opportunity for experienced managers like us to change. The point at which you don’t want change is the point at which you need to retire!’

This year is a big one for Neumann as it sees it in new company headquarters in Leipziger Platz in the centre of Berlin complete with anechoic chambers and labs.

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packaging to the actual design which was reworked for more efficient manufacture. They are easier to put together and if they’re easier to build then they’ll be easier to service.

Wolfgang says that the whole project is an example of the strengths and synergies of Neumann being part of Sennheiser since 1991. He gives a very strong hint that, just as in the past, there are areas that Neumann will be looking to provide new products and services in the near future.

Returning to the quieter end of the factory with its compact production lines for monitoring products, this is a final assembly and test operation from parts some of which are delivered preassembled. The amps are produced in Germany, for example, drivers come from the appropriate suppliers and the boxes are made in three different sites in Europe.

The components come in and are all inspected and are then transferred in to ‘kits’ depending on the production plan and are mapped to the production line to be as close to it as possible. The lines are designed for different models and different build times as they know exactly how long the various processes take to complete. Assembly is done in stages with individual workers taking care of mounting the drivers in the baffle, for example, while others fit the amplifier boards into the cabinets. A serial number follows each product through manufacture and is scanned repeatedly and serves as its history and the key to the next stage. For example, when a product reaches security testing — making sure that the unit is safe to use — unless it passes this test it will not be able to progress to any other stage.

Each completed monitor is tested in an anechoic chamber against reference curves for a variety of parameters. It’s also tested for air leaks and a full function test of all operational parameters including whether the Neumann badge lights up on power. The test spread is well within +/-1dB and the whole test routine takes about five minutes and includes adjustment of the trimmers. Subs are tested in the same manner. Final assembly and testing also feeds automatically into the logistics chain.

Flying in the face of those who believe that ‘burning-in’ a monitor is some essential part of the new product ritual, Neumann doesn’t subscribe to the idea. Only pre-series run production units — the units that are produced in the initial stage of testing a new manufacturing process — are run for 24 hours at high level as a means of soak and reliability testing them. Neumann does not soak or ‘run-in’ production run models.

The production is well paced and continuous without massive back logs building up — there is a relaxed efficiency about the way things go around here. There is also a lot of spare capacity for ramping up numbers should it be required — apparently they can comfortably output two-thirds more than they currently do.

‘The good thing here,’ says Wolfgang, ‘is that every speaker you see being built has already been sold!’