KEF R&D

THE REFERENCE

Contents

1 Heritage

1.1 KEF Reference – A Brief history

The birth of the Hi-Fi industry in the 1950s was followed by it's adolescence in the 1960s with the 1970s being the decade in which the industry matured to adulthood. KEF's success in the 1960s meant that in the early 1970s the founder of KEF Raymond Cooke, a great believer in applying objective engineering methodology, was able to make a massive investment in digital technology and specialist engineers. By purchasing Hewlett-Packard computers and Fourier analysers the KEF engineers could acquire acoustic measurement data and use it in a pioneering computer aided design process. An added bonus was the improvement in production quality control since this data could be stored and response variations quantified.

A new line of products was conceived to herald the use of this technology in KEF's design and manufacturing processes.

Because KEF wished to balance a number of client needs – the BBC's technical criteria, burgeoning audiophile tendencies, commercial realities and other considerations – a conscious decision was made to improve the performance of domestic loudspeakers with the aid of computers, while respecting various external influences.

The first speaker to benefit from computer-aided design was the Model 104, the début product in 1973's all-new Reference Series, designed by Laurie Fincham, then Research Director of KEF Electronics, and Malcolm Jones. Raymond once stated that 'Finding a new name for this product was a challenge. I wanted to avoid misleading words like "monitor", which had been over-used and discredited. A description was needed to convey the idea that every loudspeaker is subject to test and scrutiny at all significant stages of assembly – culminating in a final test comparison with a laboratory maintained reference standard which in practice is the final approved prototype. Hence the method became the name of the new series: "Reference".'

Among the features that appealed to more sophisticated markets were electronic overload protection, matched stereo pairs and – especially for territories such as the USA, with average room size being far greater than that in British, European or Japanese homes – high power handling. Raymond was not unaware of, nor afraid to discuss the different needs of the various markets, even remarking, 'We [the UK audio community] generate as much discussion and talk about hi-fi as any other country. But the equipment bought by most British people is fairly middle-of-the-road, and it should be good value for money.' Raymond's mild pessimism about the potential sales for costlier models for the UK market proved to be unfounded, as the first Reference Series product was a success, and added immeasurably to KEF's prestige at home.

In addition to exceptional performance, KEF speakers bearing the Reference name exhibited superior construction offering a sense of luxury. This quality has been revered by the team responsible for building Reference products, from the first models through to the present day.

KEF developed the testing protocol in-house to ensure that every production speaker sounded exactly as the the laboratory reference sounded. One cannot overestimate the value of the Reference Series to KEF, for nearly 40 of its 50 years, for the standards it set, from veneer matching to rigorous testing. Today's Reference models still adhere to the self-same methodology and scrutiny before being shipped. Of course the engineering techniques have progressed and as it will be seen are still setting the standard for methodical engineering design refined by and for music lovers.

The Reference 104 from 1973.

2 Philosophy

Loudspeakers are the final stage in the sound reproduction chain. It is ultimately down to the loudspeaker to generate the sound that the listener will hear. While other pieces of audio equipment have quite clearly defined roles, and it is fairly obvious to outline how they would ideally perform, the ideal loudspeaker is more difficult to define. For example, an ideal CD player would recreate the encoded waveform as a voltage without any deviation or added artefact. That is not to say that the design of a good disc player is straightforward, but simply that the ideal function is quite clear.

To define the ideal loudspeaker, it is simplest to first consider what the audio system as a whole is trying to achieve. The ideal audio system should be able to recreate a live sonic event so that it is indistinguishable from the original. The listener should be transported to the original environment of the live event. He should be convinced that he is sitting in the actual concert hall in which the live event occurred. He should experience the acoustic of the space, perceive the locations of the instruments, interact with the space and hear the change in the sound as he turns his head toward the soloist.

Many recordings are available that never existed as live events. For example, a rock band captured in a studio on a multi-track system or music with synthesised instruments. Nevertheless, the same objective applies for these situations: the sonic event that we wish to hear is the one that was envisaged by the musicians and producer.

Can this be achieved? Clearly, there are implications for the fidelity of the replay system: the system must not colour the sound with the introduction of distortion or dynamic range compressions; the system must have a neutral timbral character, without resonance or imbalance; the system should have a good temporal resolution such that it does not "smear" the sonic event. Each of these fidelity requirements provides clear targets for the loudspeaker designer.

However, this ideal audio system has two further implications that are more difficult to handle. Firstly, the spatial information of the original event should be captured and replayed. Secondly, the listener should hear only the acoustic space of the original event and not the acoustic space in which he is actually located.

Technically, neither stereo nor conventional multichannel is sufficient to recreate the exact sound field of an event. However, our perception is not exact: our auditory system builds a scene in our mind's eye (ear) based on cues in the signals arriving at our ears. Cues such as the relative arrival time and level of the sound at each of our ears, such as the loudness and decay rate of the reverberation following a staccato note, such as the relative loudness of instruments in an ensemble. Stereo provides a simple means by which the artist or recordist may communicate these cues to the listener. The listener builds a picture of the sonic event in their mind, perhaps not to the extent that he perceives the sonic event indistinguishably from the original, but sufficient to emotionally connect with the experience of listening to the original.

Loudspeakers must be designed to maximise the communication of these spatial cues. To do this a loudspeaker must have a response that does not change rapidly with direction. An irregular dispersion can result in the situation in which the loudspeaker itself results in spatial cues that conflict with those in the recording.

Controlling the loudspeakers' directivity is also key to avoiding loss of midrange and treble fidelity, which can happen when loudspeakers are placed in a real listening environment. One of the features of our auditory perception is that we are well used to hearing sounds that include reflections off close surfaces. Our auditory system can easily identify the direct sound and separate out reflections to the extent that we do not perceive the reflections as separate events. Indeed, the listener will attribute any timbral imbalance in the reflections to the original source. This means that loudspeakers must have a frequency response that is good in all directions, not simply in the direct path to the listener. Loudspeakers must have a smooth and flat on-axis frequency response and a smooth and balanced frequency response in other directions. If this is achieved, the listener is able to "hear through" the room in which he is located and perceive the acoustic space captured in the recorded sound.

In summary, loudspeakers must have a smooth and balanced response both in terms of frequency and space. The sound from loudspeakers should emanate from the drivers themselves and not from other components, such as resonating panels or openings. The drivers should operate in a well-controlled manner throughout and beyond their band. Loudspeaker should have low distortion and compression, and a good temporal response.

"Of all art, music is the most indefinable and the most expressive, the most insubstantial and the most immediate, the most transitory and the most imperishable. Transformed to a dance of electrons along a wire, its ghost lives on. When KEF returns music to its rightful habituation, your ears and mind, they aim to do so in the most natural way they can… without drama, without exaggeration, without artifice." (Raymond Cooke OBE, KEF founder).

Model 105 (1977-79) | *Model 105/4 (1980-82)* | *Model 105/2 (1979-87)* | *Model 107 (1986-89)* | *Model 107/2 (1990-96)* | *Model 207 (2001-2006)* | *Model 207 (2006-2014)*

After 30 years of continous innovation and development, the Reference loudspeakers are the perfect embodiment of the KEF philosophy to acoustic engineering.

3 The model range

The perfect reproduction of recorded sound is what KEF's Reference Series has always stood for.

Back in the 1970s KEF was the first manufacturer to use computers to create better loudspeakers. By pioneering the use of these powerful analytics, KEF engineers matched pairs of speakers to within half a decibel - the audio equivalent of identical twins. Exact pair matching delivers perfect stereo reproduction, so these revolutionary speakers won instant acclaim for their superior acoustic precision. The name 'Reference' was born.

Today's Reference is enhanced by new technologies and advanced materials that simply didn't exist before, massively extending their performance envelope to exploit the full potential of modern music and moving image formats. But the spirit is the same: to achieve the purest and most accurate reproduction of recorded sound in a way that perfectly captures the full emotional range, depth and detail of the original performance.

The Reference comprises six designs:

REFERENCE 1: a three-way, stand-mount design featuring the innovative, 125mm (5inch) MF and 25mm (1inch) vented aluminium domed tweeter, Uni-Q point source driver array and single 165mm (6.5inch) aluminium bass driver.

REFERENCE 3: the smaller of two floor standers, the three-way design features twin 165mm (6.5inch) bass drivers, perfectly positioned above and below a 125mm (5inch) MF and 25mm (1inch) vented aluminium domed tweeter, Uni-Q point source driver array, effectively in a D'Appolito configuration.

REFERENCE 5: a formidable three-way floor stand design, it utilises four 165mm (6.5inch) bass drivers, positioned above and below a 125mm (5inch) MF and 25mm (1inch) vented aluminium domed tweeter, Uni-Q point source driver array.

REFERENCE 4c: a full range centre channel using four 165mm (6.5inch) bass drivers, positioned either side of a 125mm (5inch) MF and 25mm (1inch) vented aluminium domed tweeter, Uni-Q point source driver array.

REFERENCE 2c: a compact centre channel speaker that features two 165mm (6.5inch) aluminium bass drivers, positioned either side of a 125mm (5inch) and 25mm (1inch) vented aluminium domed tweeter, Uni-Q point source driver array.

REFERENCE 8b: a compact yet powerful subwoofer, it uses twin 500W Class D amplifiers, each driving a 225mm (9inch) long-throw, ultra-low distortion drive unit, connected back-to-back in a heavily braced, acoustically inert cabinet.

Through this white paper we will explain in much more detail the acoustic fundamentals, which are at the core of the latest Reference loudspeakers.

Reference 5

4 Technology

The technology used in The Reference is introduced in this section. Whilst it is tempting to delve straight in to the component details, it is easier to understand the design by first looking at the outside of the enclosures. Indeed this is the starting point for the engineering development.

4.1 External acoustics

In the philosophy section some aims are described that the perfect loudspeaker should try and achieve. Primary among these was that the loudspeaker should have smooth and balanced dispersion and frequency response. The key to achieving this goal, particularly in terms of the dispersion, is careful design of the acoustics on the outside of the enclosure. The shape of the cabinet, the number and size of the drivers, the positioning of the drivers are all critical to ensuring a high level of performance.

4.1.1 Mid and high frequencies

In a high quality loudspeaker it is necessary to use multiple drivers of different size. This is due to conflicting requirements for drivers designed to reproduce high and low frequencies. To create significant sound at low frequencies it is necessary to use a large diaphragm that can move plenty of air. However, at high frequencies a small diaphragm is needed for good dispersion and to avoid diaphragm resonance. For example, Figure 1 shows the dispersion characteristics of an ideal 160mm bass driver at low, mid and high frequencies. At high frequencies the driver is very directional and there are some particular directions where there are nulls in the output. By contrast, Figure 2 shows the dispersion characteristics at the same frequencies for a 25mm ideal driver. The 25mm driver has very wide dispersion at all three frequencies shown, this is because it is small compared to an acoustic wavelength in all cases. However, to reproduce a 50Hz signal at 90dB at 1m the 25mm diaphragm would need to move 20cm back and forth.

It is important that the output from each of the separate drivers is integrated together into a single coherent sound. As sound is a wave it is possible to get destructive summation, where sound from two sources cancels resulting in lower output than either source individually. For example, Figure 3 and Figure 4 show the dispersion resulting from using the ideal 160mm and 25mm drivers together in a two way loudspeaker. The tweeter is positioned 200mm away from the woofer. The first figure shows the result when a 1st order crossover is used and the second figure with a 4th order. At first inspection the dispersion is reasonable at high or low frequencies when only one of the two drivers operates. In the midrange region, when both drivers contribute to the loudspeaker output, the dispersion pattern is very poor due to interference between the two drivers. Some deep nulls are seen in the directional response.

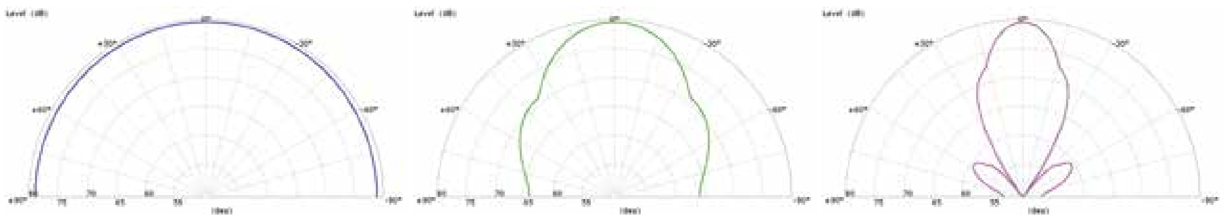


Figure 1. Polar dispersion of a 160mm ideal driver, with rigid flat piston mounted in infinite baffle, at 500Hz, 3kHz and 6kHz.

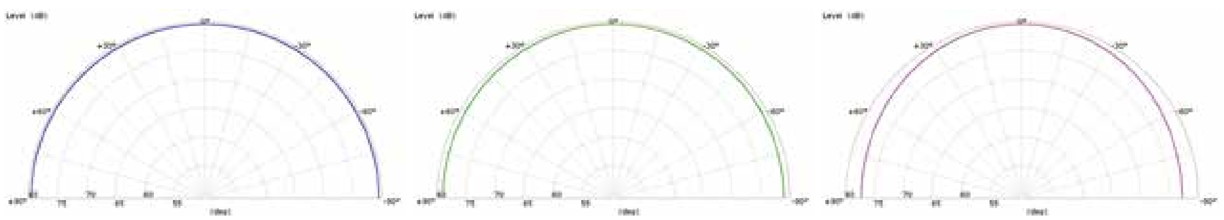


Figure 2. Polar dispersion of a 25mm ideal driver, with rigid flat piston mounted in infinite baffle, at 500Hz, 3kHz and 6kHz.

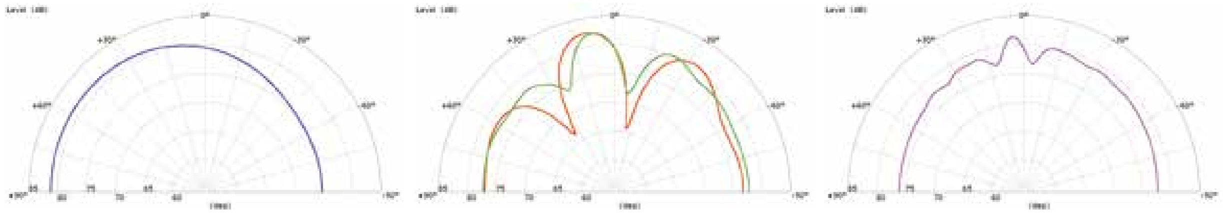


Figure 3. Polar dispersion of theoretical system with ideal 25mm driver 200mm to the right of an ideal 160mm driver with 1st order Butterworth crossover at 2kHz, polar plots at 500Hz 3kHz (green) 2kHz (red) and 6kHz.

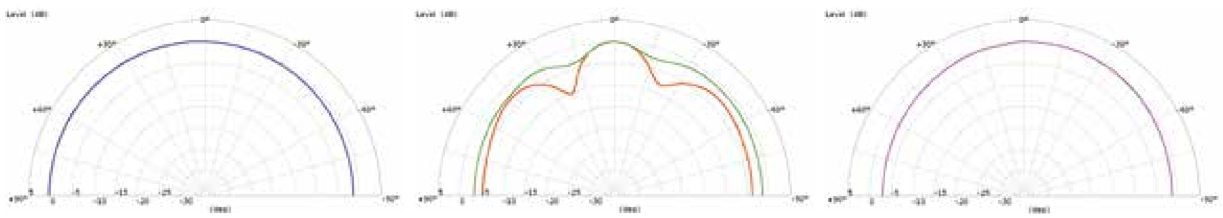


Figure 4. Polar dispersion of theoretical system with ideal 25mm driver 200mm to the right of an ideal 160mm driver with 4th order Linkwitz Riley crossover at 2kHz, polar plots at 500Hz 3kHz (green) 2kHz (red) and 6kHz.

Another behaviour can be noted on the high and low frequency graphs and is particularly obvious on the 4th order crossover: at low frequencies the dispersion is skewed to the left, towards the location of the low frequency driver, and at high frequencies the dispersion is skewed to the right towards the high frequency driver.

The KEF Uni-Q[1] driver is the first step in the solution to these issues. The tweeter is placed at the acoustic centre of the midrange driver. This immediately overcomes the issue of a coherent source location over the operating range of the tweeter and the midrange. In addition, unlike other concentric tweeter and midrange solutions, the tweeter dispersion is carefully matched to the midrange using both the KEF tangerine waveguide[2] and the shape of the midrange cone, acting as a waveguide. Because of the matched dispersion and the shared source location, the crossover design is much simpler than in a conventional system. No interference dips or lobeing occurs. The result is an array of two drivers that is totally coherent and with almost ideal dispersion and response characteristics.

For The Reference a new Uni-Q driver has been designed using technology from the Blade loudspeaker, the details of the new drivers are outlined in section 4.4. Figure 5 shows the frontal horizontal polar measured data for the prototype Reference 5 loudspeakers. At each frequency the response is symmetrical and smooth, with increasing frequency the directivity gently and monotonically narrows.

The Reference Uni-Q is designed to cover the frequency range from 350Hz upwards. This allows the Uni-Q array to cover the entire critical upper six octaves of the audio band whilst at the same time not requiring excessive excursion of the midrange cone. This is very important as with the tweeter and midrange driver in such close proximity there is the potential for interaction to occur if the midrange movement was not kept to an insignificant level. This 350Hz cut off also allows the midrange driver size to be chosen based on optimal dispersion matching with the tweeter, rather than based on the bass output requirements.

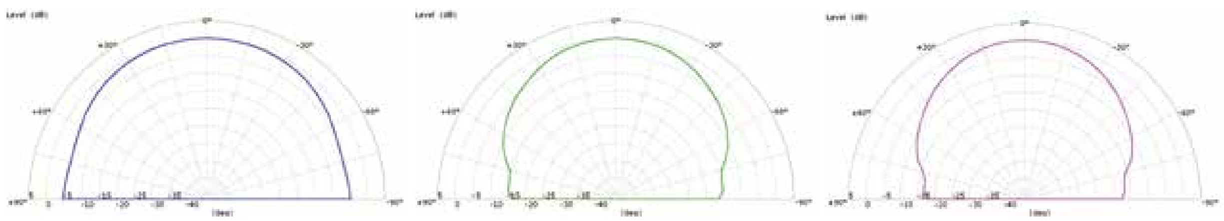


Figure 5. Horizontal polar dispersion measurements of prototype Reference 5 system with crossover, polar plots at 500Hz 3kHz and 6kHz.

1 Please see Uni-Q section in Appendices for more details.
2 Please see Tangerine Waveguide section in Appendices for more details.

4.1.2 3-way design approach

The frequency range below 350Hz is handled by dedicated bass drivers making The Reference systems 3-way designs. With each additional splitting of the audio bandwidth the complexity of the system increases and the more difficult it is to make a loudspeaker with a coherent overall output. Two way designs normally require the midrange and the low frequencies to be reproduced by a single driver. Usually this results in a compromise, choosing a smaller low/mid driver gives better dispersion and a better behaved diaphragm but will limit the bass output level. A larger cone could be used to give more maximum bass output but normally at the expense of dispersion and response smoothness at the top end of the LF/MF driver. By contrast, with a 4-way design the additional complexity is hard to justify compared to a 3-way design. Firstly, in a 4-way design it typically becomes necessary to use a very low bottom crossover point and this is a particular challenge as it increases the crossover complexity significantly in terms of number of components. Secondly, each crossover point adds an inevitable amount of time smearing[3] to the system and this becomes particularly problematic with a low crossover point. Thirdly, the bass efficiency of the loudspeaker is predominantly determined by the cabinet volume available to the low frequency driver - adding a lower mid section to create a 4-way loudspeaker uses additional cabinet volume that could be otherwise used for the bass section. Finally, the system dispersion at low-mid frequencies is largely determined by the driver positions and adding another set of drivers makes it much more challenging to position all the drivers in locations where they will sum effectively to give a good overall dispersion. The caveat to a 3-way design is that the drivers must cover a larger frequency range compared to the 4-way design. However, with modern state-of-the-art transducers this is achievable.

4.1.3 Controlling dispersion at the bass to mid crossover

All models in The Reference range use 6.5inch drivers to handle the low frequencies of the loudspeaker output. This driver has been developed especially for the new range, the full details are outlined in section 4.4. There are several reasons to use the same size driver across the loudspeaker range. Firstly, the 6.5inch driver is easily capable of being used up to the LF/MF crossover frequency of 350Hz, which is the same across all models. Indeed the new driver has a diaphragm that remains rigid more than two octaves above this frequency. Secondly, in the mid and low frequency region, from 100Hz to 600Hz, the loudspeaker directivity is largely determined by shape of the loudspeaker cabinet and driver positions. The use of a consistent 6.5inch LF driver size allows all models to use identical cabinet widths resulting in horizontal dispersion across the range which is almost exactly the same. Finally, for the larger models several drivers are used together and share the input power equally. Each 6.5inch driver is designed to be extremely linear and cope with exceptionally high power individually. Consequently this configuration is able to play at higher power levels with lower distortion than would be possible if fewer larger drivers were used.

A common misconception is that only large drivers are able to efficiently produce deep bass. This is not the case. Thiele and Small (formerly of KEF) showed in their seminal series of papers in the 1970s that it is only the cabinet volume that limits the efficiency of the bass output of a loudspeaker [1] [2] [3] [4]. It is, however, necessary to move a large volume of air in order to produce bass at high sound levels. The multiple drivers help in this aspect too, the four 6.5inch bass drivers in Reference 5 have equivalent radiating area to a 12inch subwoofer driver and extremely large excursion capability.

Below 500Hz, the drivers themselves all have wide dispersion and it is the cabinet, driver locations and the crossover design that determine the system dispersion. In all of the models the 6.5inch LF drivers are placed as close as possible to the MF driver of the Uni-Q in order to minimise lobeing and interference dips at the lower crossover. The floorstanding models use a symmetrical driver layout first described by Joseph D'Appolito [5] and designed to avoid vertical off-axis lobeing at

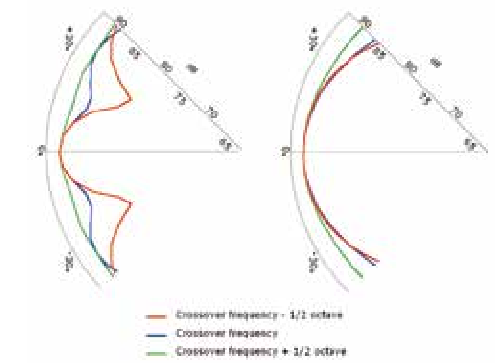


Figure 6. D'Appolito polar response comparison, left a conventional arrangement using a tweeter and two midrange drivers, right with a Uni-Q driver and four low frequency drivers as on Reference 5.

3 This refers to the non-constant group delay resulting from causal active, passive or digital crossover filters. FIR filtering allows other options but has problems of its own, such as pre-ringing, latency and frequency resolution at low frequencies.

crossover. The configuration is most frequently used to crossover between a pair of midrange drivers to a tweeter. However, in this circumstance it is not normally possible to avoid off-axis interference dips close to the listening axis in the vertical polar response as typically the inter driver spacing is significant compared to the acoustic wavelength. With The Reference D'Appolito layout the crossover frequency is much lower and consequently the acoustic wavelength is large enough that interference dips in the vertical response are pushed well away from the listening axis. The D'Appolito arrangement also has the additional benefit that the apparent acoustic source does not shift away from the position of the Uni-Q driver over the entire frequency range of the loudspeaker.

4.1.4 Cabinet diffraction

The shape of the loudspeaker cabinet has a very large effect on the smoothness of the directivity and frequency response of a loudspeaker. Edges and openings can scatter the sound from the drivers, the tweeter in particular, and this results in irregularities in the response that are typically focused at the main listening position. Figure 7 illustrates the underlying physical behaviour: when the sound from the driver hits a discontinuity, such as the edge of the cabinet, the sound will scatter and be re-radiated in all directions. If this sound reaches the listener it will arrive momentarily later than the direct sound from the driver. At the frequency that this arrival time difference is half a wavelength the sound from the two paths will destructively sum resulting in a null in the loudspeaker response.

Olsen was the first to look at this diffraction effect

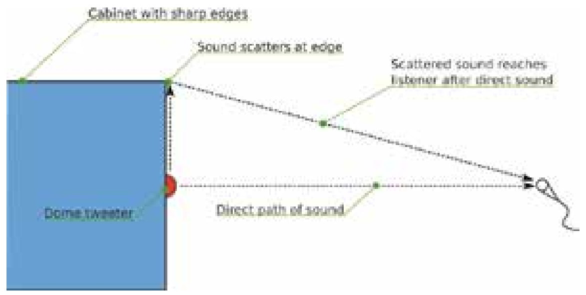


Figure 7. Diagrammatic explanation of diffraction effect.

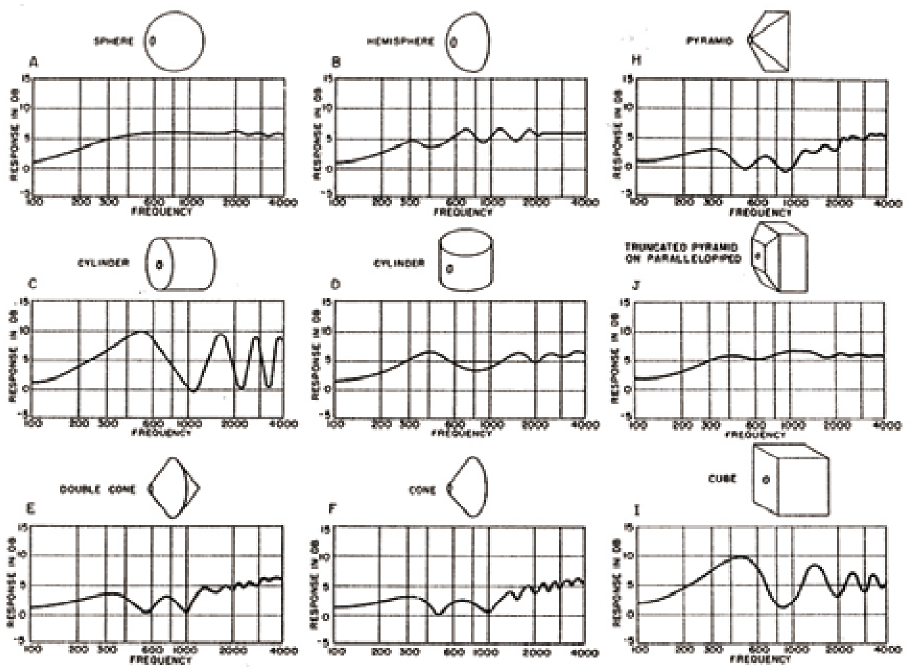


Figure 8. Reproduction of Olsen's classic loudspeaker enclosure baffle-diffraction experiment.

in detail and his classic experiments in cabinet shape are reproduced in Figure 8 [6, p.23]. These graphs are widely reproduced [7, p.318][8, p.347]. and give a good initial guideline for the preferred shape of the loudspeaker cabinet to minimise this effect. However, there is an important effect that is not demonstrated in Olsen's classic experiments: the effect of the driver dispersion on the diffraction effect. For example,

Figure 9 shows the modelled response of two different size ideal drivers located at the face centre of a 20cm cube. The red curve is the response for a 25mm driver and the blue curve is the response for a 160mm driver. With the 160mm driver the response above 2kHz is much smoother than for the 25mm driver. The raw dispersion of these drivers was shown in Figure 1 and 2 above, note that the location of the dip for this cube is 3kHz which corresponds to the second polar plot for each driver size. At 3kHz the 25mm driver has an almost omnidirectional polar response, this means a strong sound wave reaches the edge of the cube and consequently the diffraction effect is severe. The 160mm driver is a little more directional and, as much less sound reaches the edge, the diffraction effect is much less severe.

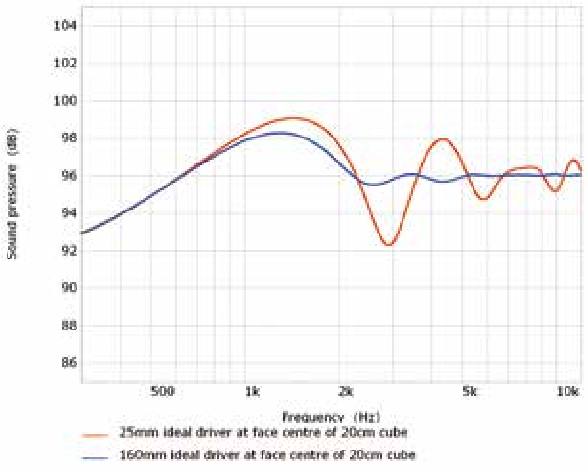


Figure 9. Comparison of the diffraction effect with radiators of different size placed in the centre of a cubic enclosure

As was discussed above, the very wide dispersion of a baffle mounted 25mm tweeter in the lower treble region is actually a problem in terms of directivity matching with the midrange driver at crossover. It is also something of a worst case in terms of diffraction because it fully “illuminates” the edges of the loudspeaker cabinet in the 3kHz region where the first diffraction dip is typically seen. A less often mentioned benefit of the Uni-Q driver is that the tweeter dispersion in the lower treble is controlled by the wave-guide of the Uni-Q driver and the diffraction problem is lessened.

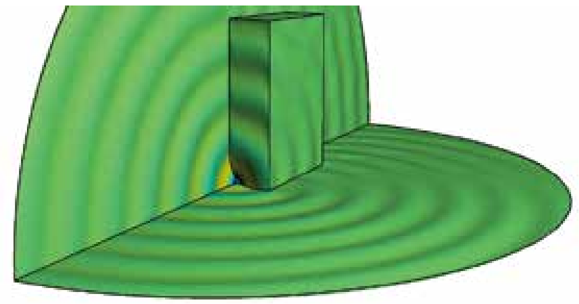


Figure 10. FEA modelled acoustic pressure on, and around, cabinet surface due to MF driver output.

BEM analysis was used to simulate the cabinet diffraction in detail with the full driver and geometry details. Out of this work the “Shadow Flare” was developed. The Shadow Flare is a shallow waveguide that smoothly blends the Uni-Q driver into the baffle of The Reference cabinets. This waveguide further controls the dispersion of the Uni-Q tweeter and midrange driver and creates an acoustic shadow in the region of the cabinet edge closest to the driver. This reduces the cabinet diffraction effect significantly to the extent that little irregularity is seen on the tweeter or midrange driver responses when they are mounted in the system. For example, the effect on the tweeter response can be seen in Figure 11.

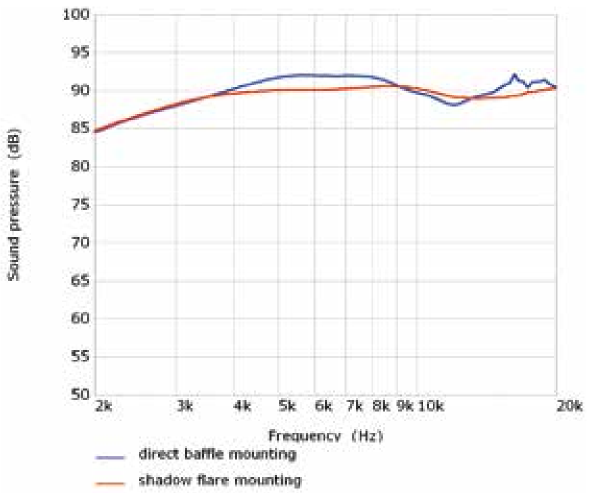


Figure 11. Uni-Q driver HF response with and without shadow flare to control the diffraction effect.

In addition to the cabinet edges, other discontinuities can cause diffraction issues. For this reason the low frequency drivers have been designed to be as low profile as possible so that they present as little deviation from the flat front baffle as possible. The port exits are located on the rear of the loudspeakers and one of the reasons for this is to minimise the diffraction. This location also serves another purpose as it greatly reduces the audibility of any remaining port midrange leakage as fully outlined in section 4.2.

4.1.5 Overall dispersion performance

Figure 12 shows a set of frequency responses for an early prototype of Reference 5. The curves shown in this plot are the figures of merit for assessing loudspeakers as suggested by the research work of Floyd Toole [9] [10]. The curves are measured at 96 data points per octave without any smoothing. Based on these figures of merit Toole was able to predict real listener preference with a remarkable accuracy. This research work is an important endorsement of the KEF philosophy of focusing strongly on the loudspeaker dispersion. The interested reader is directed to Toole’s many publications for full explanation of how to interpret the data. Briefly, this set of data shows that the loudspeaker response is exceptionally flat and smooth, free from resonance and that the dispersion is smooth and well controlled. Note that there is no change in any of the responses at the crossover frequencies of 350Hz and 2500Hz.

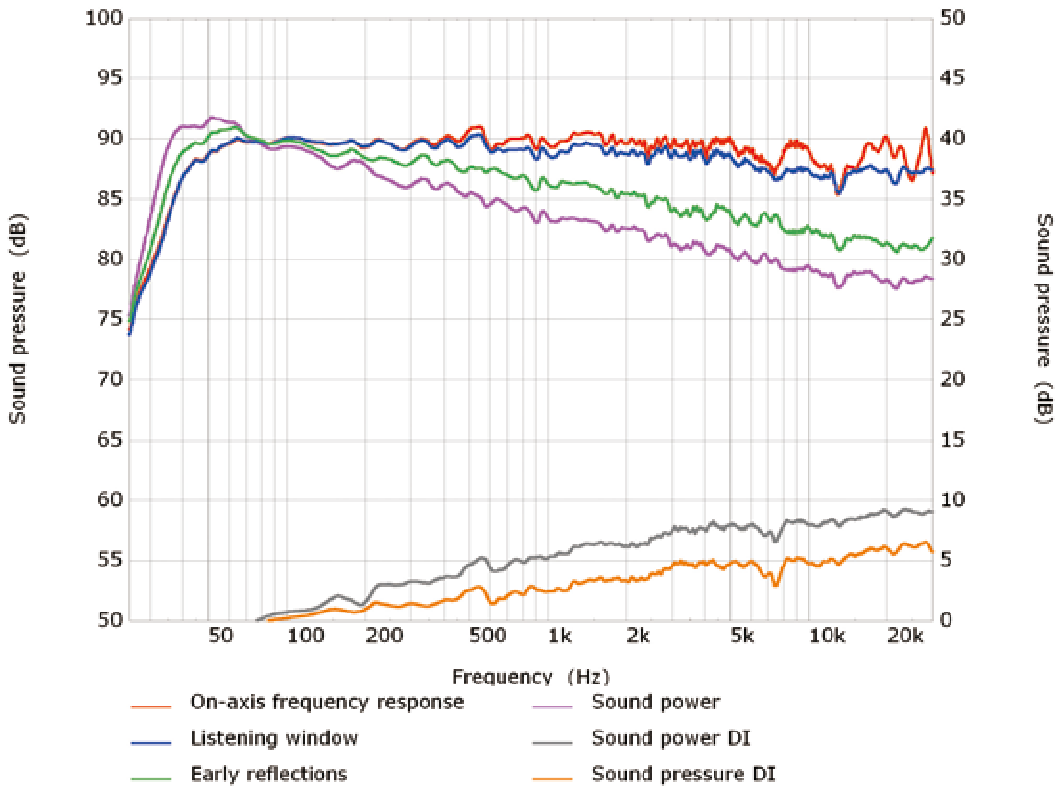


Figure 12. Family of directional response curves for early Reference 5 prototype.

4.2 Low frequency response

The main loudspeakers in The Reference use ported enclosure designs. This section outlines the reason for this choice and how this maximises the performance of the loudspeakers for the size and drivers used.

One might ask the question, why do we need a loudspeaker enclosure at all, what purpose does it serve? The answer is really quite simple: it is because when the cone of the loudspeaker driver moves forward it creates just as much sound at the back of the driver as the front. Figure 13 illustrates this behaviour, without an enclosure the positive acoustic pressure created at the front of the driver is cancelled by the negative pressure created at the rear of the driver. This cancellation effect is extremely effective at low frequencies, consequently a loudspeaker driver in free air outputs virtually no bass even at maximum input power.

The enclosure is required to contain the negative rear acoustic pressure to prevent it from interfering with the sound from the front of the driver. The simplest design of enclosure is the sealed-box, which simply contains the rear radiation in a completely closed cabinet as shown in Figure 14.

The enclosure, however, changes the loading experienced by the loudspeaker driver. When the cone is displaced, the pressure change inside the enclosure results in an additional restoring force which pushes the cone back towards the rest position. Effectively the enclosure behaves like an additional stiffness is connected to the cone - the smaller the enclosure the greater this stiffness. This is the reason why a small loudspeaker cannot produce deep bass efficiently. The effect is very dramatic, for example Figure 15 shows the modelled response of a single 160mm bass driver which has been optimised to work well without any cabinet loading, given the label driver A[4]. In an extremely large 100L box this driver can give a huge amount of bass efficiency (-3dB point at 28Hz) however, once placed into a more reasonable cabinet of 15L the response is very poor and all of the bass extension is gone due to the additional stiffness of the cabinet. Another response is shown, this time for a driver optimised to work as well as possible in the available 15L, labelled driver B. This achieves quite a tidy frequency response and bass extension down to approximately 50Hz (-3dB).

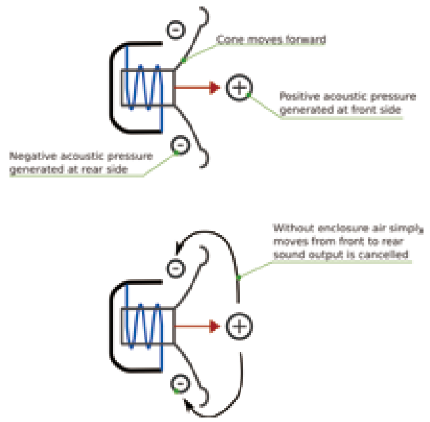


Figure 13. Family of directional response curves for early Reference 5 prototype.

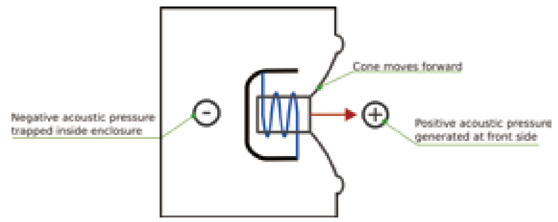


Figure 14. Illustration of a closed box loudspeaker.

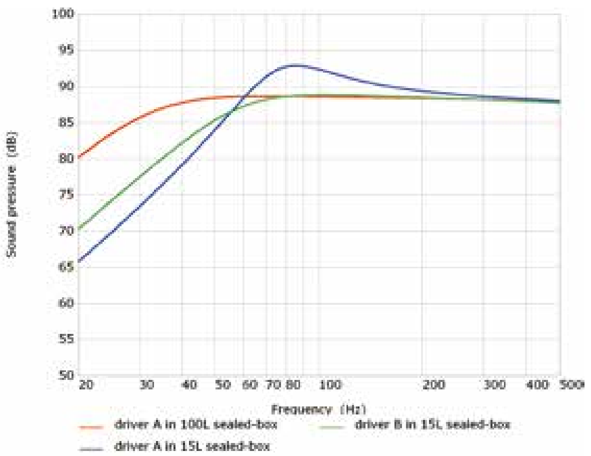


Figure 15. Variation in frequency response of a 160mm bass driver as the rear enclosure volume is changed.

4 Note that driver A parameters are probably unachievable, for example the total moving mass is only 6g and the resonance is around 5Hz.

A ported enclosure includes a tuned port or vent connecting the inside of the enclosure to the listening environment. The port allows the bass to be augmented around the region that the vent is tuned and allows greater bass extension from a given size of loudspeaker. Figure 16 shows the simulated response of a 160mm driver placed in a 15L ported enclosure compared to the previously shown optimal closed box 160mm driver. The -3dB point has been extended from around 50Hz to approximately 38Hz. Perceptually this is a very large change and the ported enclosure will sound far more extended in the bass than the closed box.

For many listeners the ported version will be much more favourable than the closed due to the additional bass extension, however, the additional bass extension is not without compromise. With the addition of the port to the low frequency system, the system order has been increased and as a result the transient response is worsened. For example Figures 17 and 18 show the response of the closed and ported systems to a low frequency toneburst input, the difference in the temporal response is quite clearly seen. This creates something of a dilemma as, depending on their personal preference and room characteristics and loudspeaker position, some listeners will prefer the ported response while others will prefer the closed box response.

For The Reference, as a solution to this issue, the main loudspeakers are supplied with two different length port liners. The shorter of the two liners results in a loudspeaker response similar to that shown in the ported system above. Fitting the longer liner results in a frequency response similar to that shown in Figure 19, the same closed box response is shown for easy comparison with Figure 16. This low frequency alignment is specifically designed to roll off very slowly and gently in the upper bass octaves. In many listening rooms this will compensate for the natural bass augmentation due to the closest room boundaries[5].

The toneburst response with the longer port liner is shown in Figure 20. Comparison with Figures 17 and 18 show that the temporal response is now quite close to the closed box system yet the bass efficiency around 30-40Hz is usefully augmented by the low port output. An additional benefit of this approach, other than the adjustability, is that the port helps to control the driver excursion even in the lower tuning mode. Figure 21

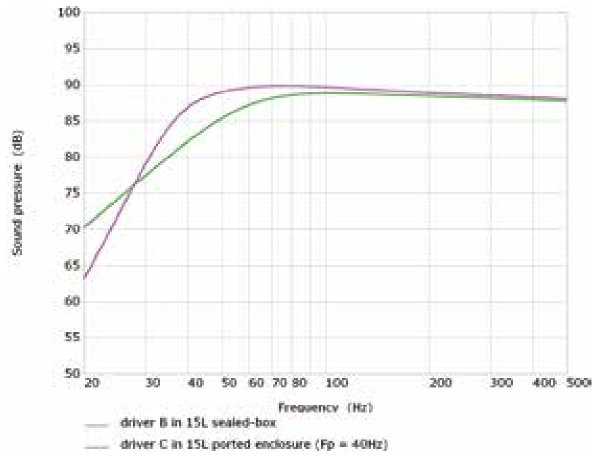


Figure 16. Comparison of two 15L loudspeaker systems each with a single 160mm LF driver and using a sealed and ported box design.

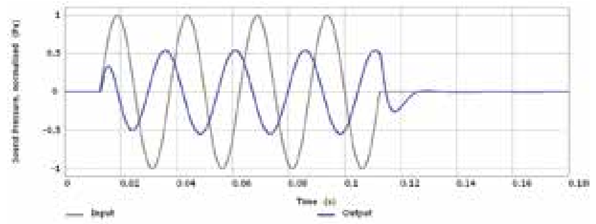


Figure 17. Response of closed box loudspeaker shown in Figure 16 to a 40Hz toneburst.

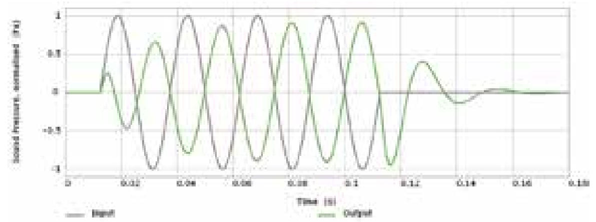


Figure 18. Response of ported loudspeaker shown in Figure 16 to a 40Hz toneburst

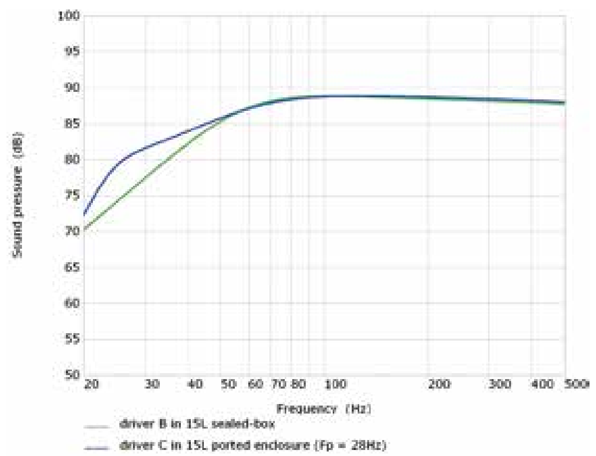


Figure 19. Comparison of two 15L loudspeaker systems each with a single 160mm LF driver and using a sealed and ported box design.

5 Please see appendix VII on the acoustics of listening rooms for more details.

6 A standard four string bass tuning sets the open bottom string fundamental to approximately 41Hz. The less common five and six string tuning typically use a tuning of around 30Hz. In most music the energy content below

shows the driver excursion at an input voltage of 2.83V for each of the three systems. Over much of the bass region the ported systems require the driver to move less, particularly in the region around 30-80Hz which typically has the highest energy content in modern music[6]

The user is left to experiment to find the best options according to their personal taste and listening environment. With the floor standing models, which have two ports, it is also possible to use intermediate tunings by using one long and one short port. This allows a finer degree of control over the bass response.

4.2.1 Optimising the port behaviour

The models in the section above show the theoretical response of various different types of low frequency loudspeaker design. The behaviour of the cabinet and port are simplified. This is very useful for gauging the overall performance and optimising a particular design. However, it is very important to account for the "higher order" effects that occur in a real loudspeaker. The port, in particular, presents a real engineering challenge and The Reference incorporates several design approaches and technologies to ensure that the real life port behaviour is close to the theoretical ideal.

Port flow

Figure 22 shows the peak port velocity versus frequency for the 40Hz tuned ported system that was described above for an input level of 10V rms (around 25 watts). The velocity close to the port tuning frequency is surprisingly high - more than 15 metres per second (approximately 33mph or 54kmph). At these velocity magnitudes it is important to carefully consider the port airflow in order to avoid port turbulence. When turbulence occurs in the port the efficiency drops dramatically and the bass output is severely compressed. Additionally the turbulent flow generates noise that the listener may hear.

There is a great deal of existing research into fluid flow in the field of vehicle aerodynamics. Unfortunately these studies are often not very relevant to port flow problems as turbulence is either inevitable in such applications, or the priority is to maximise the flow efficiency and whether the flow is laminar or turbulent is of little consequence. Indeed some famous and ingenious methods for reducing drag, such as the texturing on a golf ball, function by inducing turbulent flow. However, there are some excellent studies of

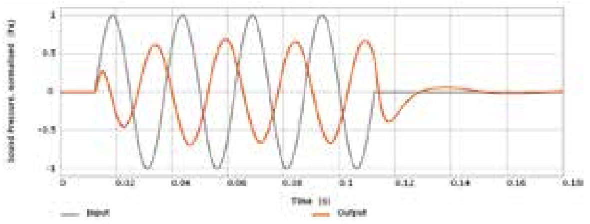


Figure 20. Response of ported loudspeaker shown in Figure 19 to a 40Hz toneburst.

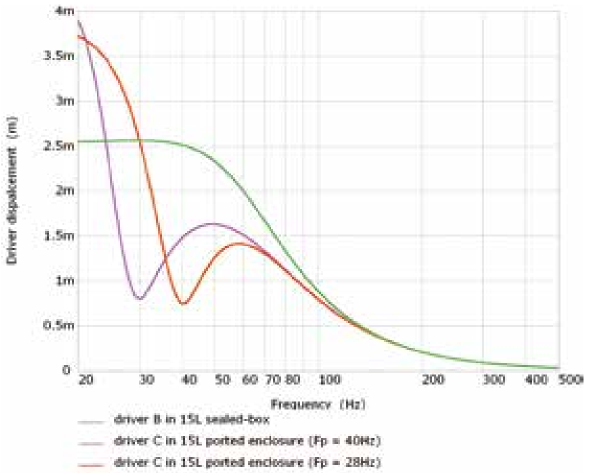


Figure 21. Lumped circuit representation of a generic electromagnetic loudspeaker.

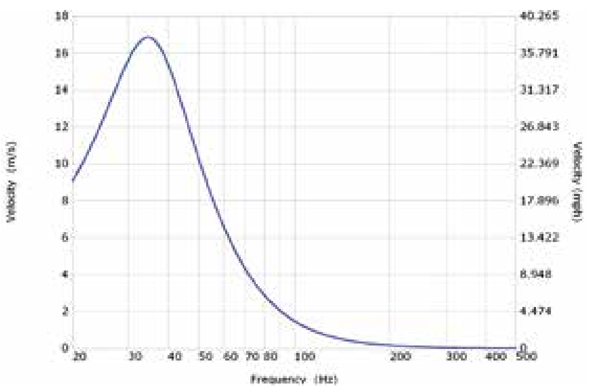


Figure 22. Peak port air velocity for a theoretical 15L speaker at an input level of 10V rms (approx 25Watts).

port flow published by the AES, in particular Salvatti, Devantier and Button published a very comprehensive summary of experiments on ports of different types [11]. They show quite clearly that a smooth walled port results in less output compression and noise than a textured port, and they also demonstrate that the port performance can be greatly improved with careful design of the profile and flares.

30Hz is significantly lower than rest of the bass region.

At KEF the airflow in a loudspeaker port can be computer modelled using Computational Fluid Dynamics (CFD). Using this tool a flare profile has been developed that is optimised to delay the onset of turbulence to high output levels. This flare profile is used on the ports of The Reference. Figures 23 and 24 show some of the flow results from the CFD analysis for different types of port. The optimised port shape, on the right in each case, shows a very even flow pressure through the entire port tube. By comparison, on the unoptimised ports you can clearly see turbulent vortices inside the ports that will lead to power compression and noise during use.

Port locations

The theoretically ideal behaviour of the port is dependent upon the air in the loudspeaker enclosure acting like a perfect acoustic compliance, or spring, and the port itself acting like a perfect acoustic mass, or single united plug of air. The behaviour of a real loudspeaker enclosure and port is somewhat more complex. The result of this is that, in addition to the desired output at low frequencies, the port can also output significant sound in the mid-band. In particular, enclosure standing wave resonances can quite easily leak out from the port if care is not taken with the design of the enclosure and the placement of the port.

For example, Figure 25 shows the modelled response of the low frequency section in a ported loudspeaker system that includes standing wave resonances in the enclosure. In this model there is little acoustic wadding in the enclosure to make it very easy to see the resulting behaviour as the port location is changed.

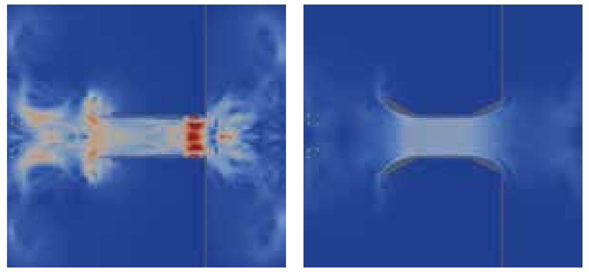


Figure 23. Instantaneous flow pressure contour of straight tube port (left) compared to optimised port (right) computed using CFD, note that both ports are shown on the same colour scale and at the same instant. For this analysis the peak velocity is 10ms.

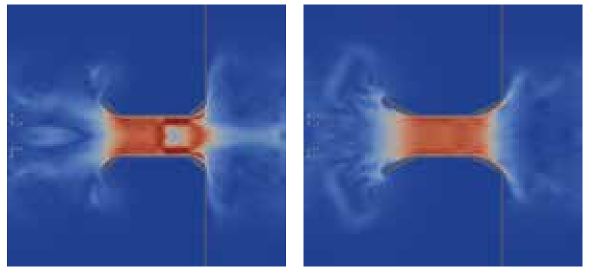


Figure 24. Instantaneous flow pressure contour of unoptimised port (left) compared to optimised port (right) computed using CFD, note that both ports are shown on the same colour scale and at the same instant. For this analysis the peak velocity is 15ms.

The left hand chart shows the port output with an unoptimised location, the right with the port location optimised. The reduction in the midrange leakage through the port of the first standing wave resonance, at 450Hz, is dramatic. The output of the second is relatively unaffected in this case and the third resonance is also reduced in level. Note that the location of the port in this case has been specifically optimised to control the leakage of the 450Hz resonance as this is the most difficult to control by other means.

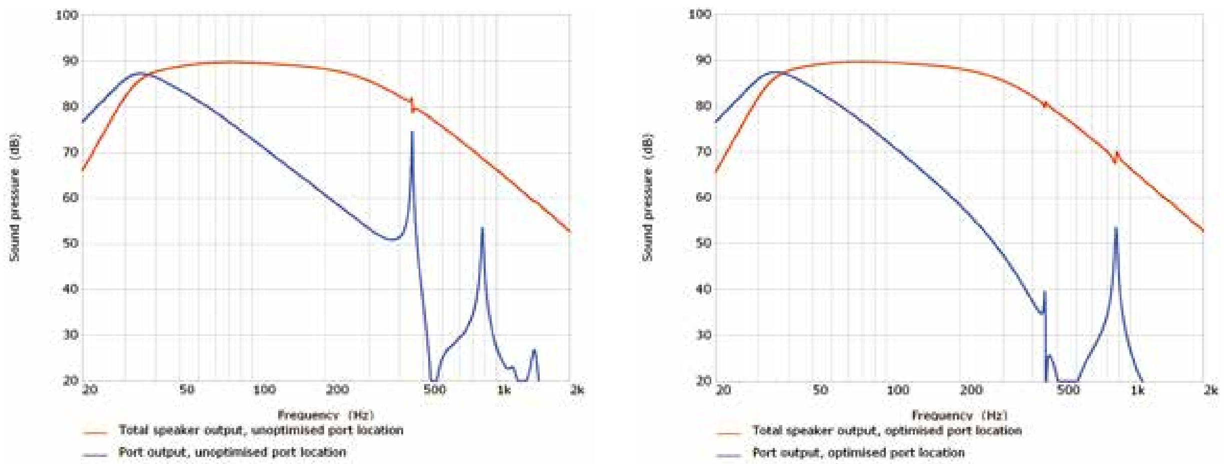


Figure 25. Simulated comparison of the loudspeaker output and port output with unoptimised port location (left) and optimised port location (right).

Adding wadding to the optimised port version further reduces the midrange leakage as shown in Figure 26. Eagle eyed readers will also note that as wadding is added the small "glitches" on the system response, due to coupling between the driver and the standing wave resonances, also disappear. Another design feature of The Reference is that the ports are located on the back of the loudspeakers. This is intentionally done for a few reasons. One of these is that it makes any midrange leakage through the port even more difficult to hear from the listening position as the majority of the midrange port output is directed away from the listener. The rear placement effect is also shown in Figure 26. Note that on this figure the midrange leakage from the port is now at an extremely low level compared to the loudspeaker's main output, which is approximately 90dB.

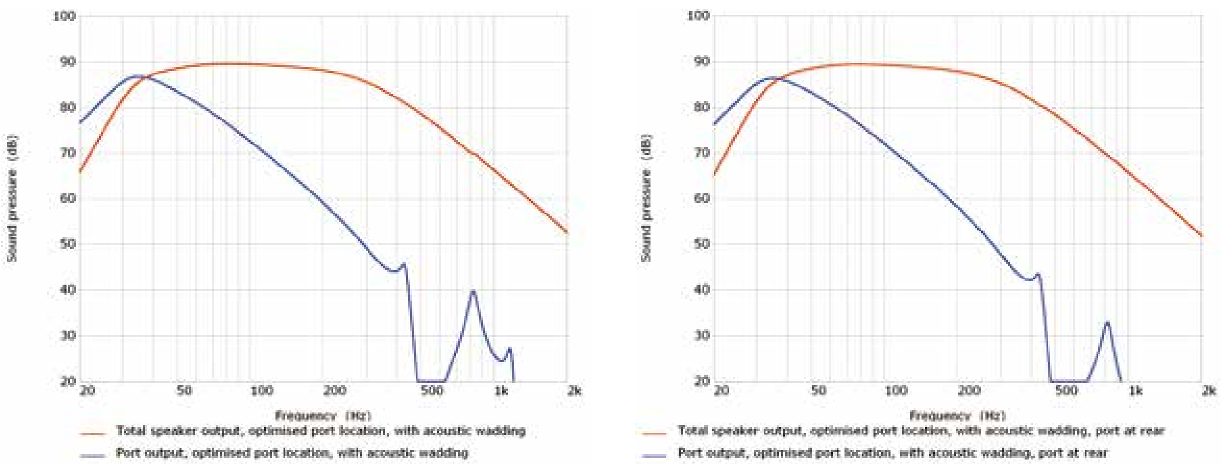


Figure 26. Simulated comparison of the loudspeaker output and port output with optimised port location and enclosure acoustic wadding, port at front (left) and port at rear (right).

Port standing waves

In order to minimise turbulence and bass output at high levels it is necessary to use a large and long port. However, the port itself has standing waves due to the sudden change in the acoustic environment at the inner and outer ends. This is the same type of longitudinal resonance that is seen in a pipe organ. Just like a pipe organ, a longer port/pipe has a lower fundamental standing wave resonance. Figure 27 shows the response of the modelled system discussed above with a small, medium and large port. The small port behaves exactly the same as the models shown above, whereas with the medium and large ports additional resonances have appeared in the port output. These resonances correspond to the fundamental and harmonics of the port standing waves.

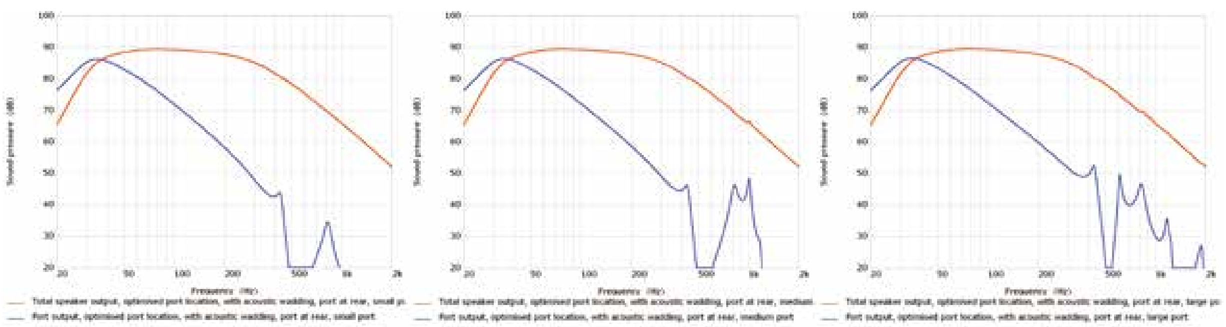


Figure 27. Simulated comparison of the loudspeaker output and port output with optimised port location and enclosure acoustic wadding, port at rear and small (left) medium (middle) and large (right) port.

Reducing the magnitude of the longitudinal resonances cannot simply be achieved by filling the port with acoustic foam since this would reduce output in the bass region and prevent an efficient alignment. An alternative method to control the longitudinal resonance was devised for the LS50, by creating a port with flexible walls. For The Reference this is achieved by using a soft foam insert to form the port walls. At midrange frequencies the soft walls of the port flex and dissipate energy from the resonances. This can be clearly seen in Figure 28, which shows the modelled acoustic pressure and port wall displacement at the first standing wave frequency of the port. The right hand result, with the flexible walls has significantly lower pressure at the centre of the port. The acoustic pressure at the surface of the port causes a compressional wave to form in the soft wall material, the energy is gently absorbed as the compressional wave travels into the port wall.

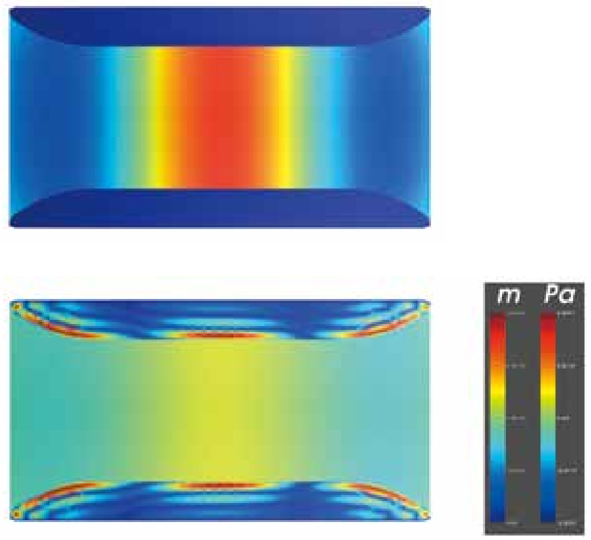


Figure 28. Comparison of the FEA modelled pressure magnitude in the loudspeaker port at the first standing wave with rigid walls (above) and flexible walls (below).

Figure 29 shows a comparison of the acoustic pressure in the centre of the two modelled ports. The peak pressure at the 900Hz first standing-wave resonance is reduced by approximately 20dB compared to the solid port.

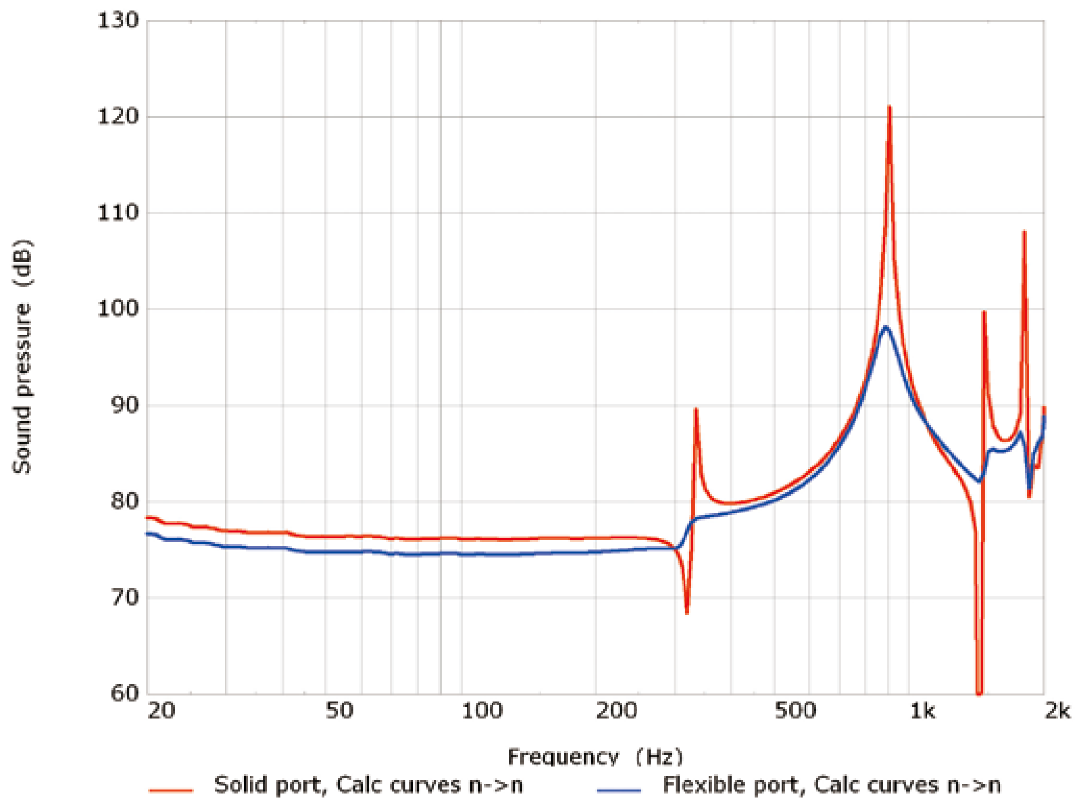


Figure 29. Comparison of the FEA modelled internal acoustic pressure in a solid and flexible port

4.3 Internal acoustics and vibration control

4.3.1 Controlling enclosure standing wave resonances

The sound inside a rigid enclosure reflects from the enclosure surfaces with very little attenuation. This situation leads to standing wave resonances. The simplest explanation of a standing wave resonance is to consider the sound between two parallel walls as illustrated in Figure 30. In this example the source plays a steady tone and radiated sound travels towards the first wall, reflects towards the second wall and then reflects again arriving back at the source. The sound travelling on the reflected path arrives back at the origin a short time later. If this time difference corresponds to a multiple of a period of the radiated tone, then the radiated sound is re-enforced by the reflection. Every cycle adds more energy into the system and the sound pressure gets higher and higher.

The reason that this phenomenon is called a standing wave is that eventually the sound pressure between the two boundaries falls into a steady oscillating pattern and no travelling wave behaviour is visible. Figure 31 shows the standing wave pattern that occurs when the time delay is one wave period and the corresponding air particle velocity.

A common misconception is that if the walls of an enclosure are not parallel the standing waves will not occur. This is not correct, for example, figure 32 shows the FEA computed first standing wave pattern in an enclosure with parallel and non parallel walls. Even though the walls are not parallel there is still an acoustic path that sets up a standing wave situation. With parallel walls, the path is the same for all frequencies, which means that all the standing wave resonances occur at frequency multiples. When the walls are not parallel this will not be the case but there are still just as many standing waves in the undamped enclosure.

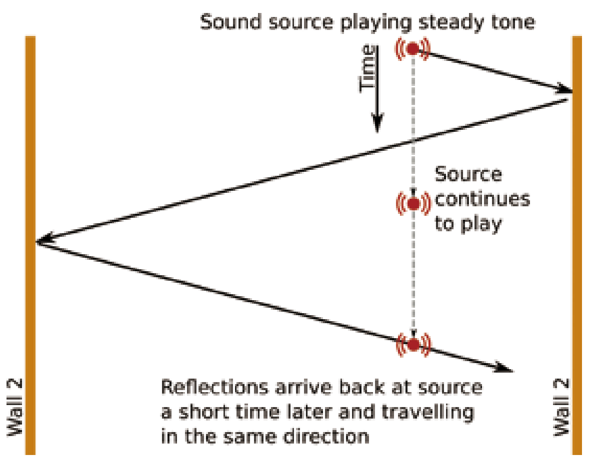


Figure 30. Illustration of standing wave resonance mechanism.

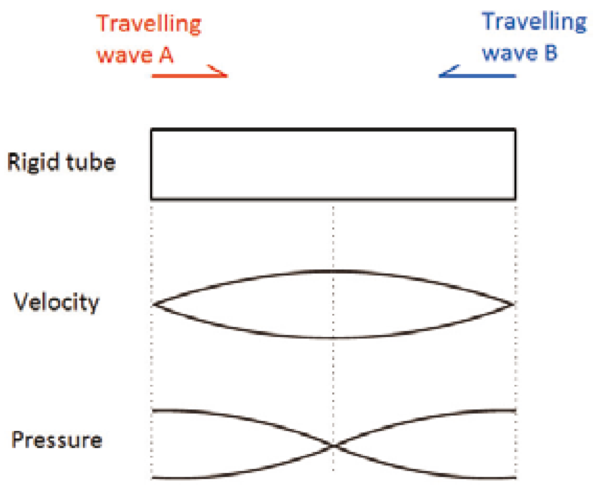


Figure 31. First standing wave pattern (or mode shape) between two parallel rigid walls.

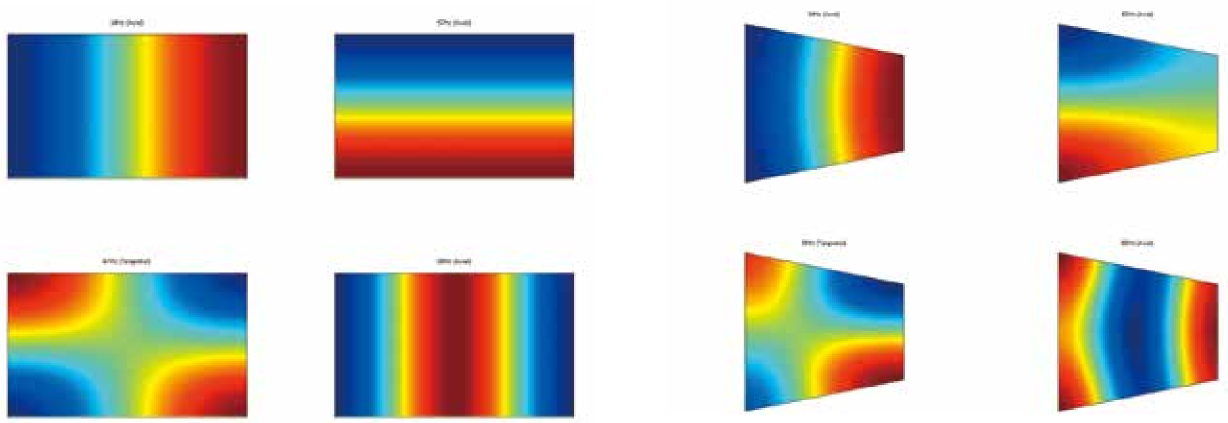


Figure 32. First four standing waves of a rectangular enclosure (left) compared to a trapezoidal enclosure (right)

It is inevitable that standing waves will occur in the LF enclosures within the bandwidth of the driver's output as the enclosure is large compared to a wavelength at the upper end of the low frequency bandwidth. When an enclosure standing wave is excited, very high acoustic pressures are generated in the enclosure. The effect on the sound output is predominantly due to two effects. Firstly, the enclosure acoustic loading on the LF driver is changed due to this high pressure acting on the back of the driver cone, this typically causes a glitch or dip in the driver output. Secondly, the high acoustic pressure can be radiated through the ports[7].

As with the ports, a great deal can be achieved by carefully optimising the shape of the enclosure and the location of the drivers. The driver location is particularly important as it is the driver that is the sound source that causes the standing wave to form. In addition, acoustical damping material can be added to the enclosure.

The size of the enclosure is of particular importance. The largest dimension of the enclosure determines the frequency of the first standing wave resonance. If the enclosure is large then the first standing wave will be relatively lower in frequency. This is not desirable as it is much more difficult to dampen a low frequency standing wave with acoustic wadding. In addition, the effect of an enclosure standing wave on the driver cone motion is greater if the standing wave frequency is closer to the port and driver fundamental resonance. Because of this, The Reference LF enclosures are partitioned into smaller enclosures to push the standing wave frequencies higher.

The quantity and position of the acoustic wadding is critical for optimal performance. If too little wadding is added to the loudspeaker the standing wave resonances will not be suppressed. If too much wadding is added the acoustic output from the port will be restricted and the driver motion dampened. For the most efficient damping, the wadding needs to be positioned where the air velocity is highest. Referring to Figure 31, at the enclosure walls the air velocity is zero, it is much more effective to locate the wadding towards the centre of the enclosure. For example Figure 33 shows the effect of two different wadding placements upon the first standing wave resonance of a tube shaped enclosure. In both instances the same quantity of wadding is used, however the placement at the centre of the tube is

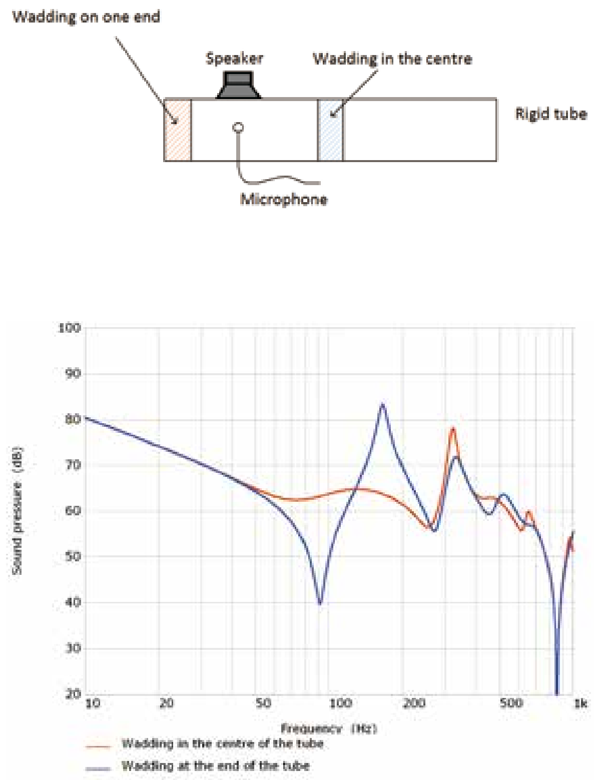


Figure 33. The effect of wadding placement on the attenuation of standing waves.

7 The transmission of these high pressures through the cabinet walls is relatively minimal [12].

much more effective, particularly at the lowest standing wave.

Figure 34 shows some real measurements from a prototype of Reference 5 with no wadding and with optimised wadding in the enclosure. These measurements are taken with the microphone very close to one of the low frequency drivers to clearly show the standing wave effects, the dip at 35Hz is due to the port. Note that even before the wadding is added to the enclosure the behaviour is very good, this is due to the carefully designed enclosure and driver locations. Once the wadding has been added the overall response is smoother, the dip at 400Hz (corresponding to the first standing wave) is suppressed and there is little change to the low frequency response save for a slight shift in the port tuning frequency.

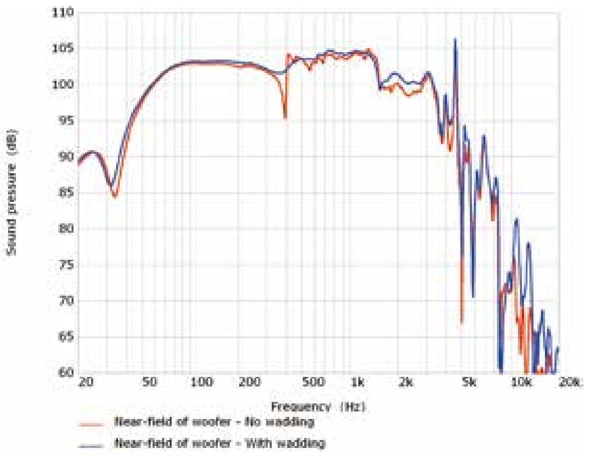


Figure 34. Nearfield measurements of an LF driver from a prototype of Reference 5 showing the response with and without wadding added to the enclosure.

4.3.2 Cabinet vibration control

To reproduce the most realistic listening experience, the sound needs to come purely from the drivers and not from the cabinet. Any resonant vibration from the cabinet panels will add unwanted distortion to the music. This means that the cabinet walls of a loudspeaker ought to be as inert as possible. The construction of The Reference cabinets has always been a strength of the range. The new models maintain this tradition and are constructed in thick high density wood the extensive internal bracing. Conventional internal bracing aims precisely to stiffen and support the inside structure as well as each panel. Figure 35 illustrates the effect of adding a pair of braces, crossing behind the driver, compared to a box without any bracing. The braced box has raised the resonance from 600Hz to 1.6kHz but the amplitude of the peak still remains the same. Adding stiffness to the cabinet will always have this effect, to reduce the severity of the resonance it is necessary to add damping.

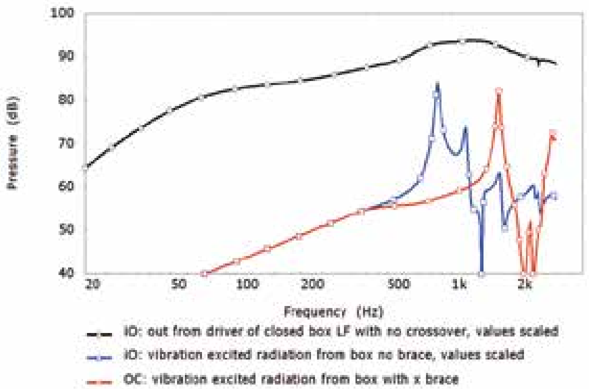


Figure 35. Closed box FEA predicted output from diaphragm and walls with and without x-brace.

During the development of the KEF LS50, it was found that by adding material with high mechanical resistance and low stiffness between the walls, baffle, driver and brace a dramatic amount of damping could be added to the cabinet. Figure 36 shows the same loudspeaker previously shown in Figure 35 using this damped bracing configuration. The peak at 1.6kHz is dramatically reduced in amplitude by about 30dB. The Reference uses this technology throughout the range in order to minimise the cabinet vibration.

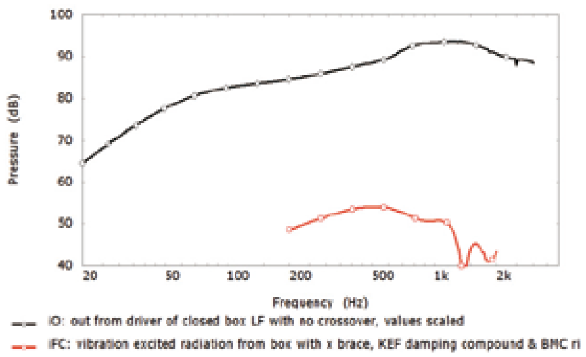


Figure 36. Closed box FEA predicted output from diaphragm and walls with constrained layer of damping material between brace and walls.

Figure 37 shows a cross section of Reference 5, the cabinet walls are supported by extensive internal bracing. These braces are connected to the cabinet walls using a layer of damping material. The drivers are also braced by the internal structure and connected using a layer of damping material to further control the cabinet vibration. The front baffle of the cabinet is constructed in an exceptionally strong laminated aluminium and resin composite, this adds a great deal of strength and mass to the cabinets. The baffle also forms part of the damping arrangement and is connected resiliently to the rest of the cabinet using high loss pads.

Figure 37. CAD section of Reference 5 showing extensive internal damped bracing.

4.4 Driver details

4.4.1 Uni-Q Driver

The Reference uses a specially developed 11th generation Uni-Q driver array. This particular Uni-Q is only used in The Reference and incorporates many features found in that of the flagship KEF Blade loudspeaker. Figure 38 shows a CAD rendering of the new driver, with the midrange motor shown in isolation on the right. In particular, note how open the rear of the driver is, this is done intentionally in order to maximise the rear venting of the midrange. On the right hand image the large aluminium ring on the top of the motor system is visible, another identical ring is buried in the magnet system to help to control the midrange distortion. The design of this driver is outlined in more detail in the next few paragraphs.

Figure 38. CAD rendering of The Reference Uni-Q, right hand image shows midrange motor in isolation.

The Tweeter

The tweeter in the new Uni-Q is closely based on that of the KEF Blade. It uses a powerful neodymium motor system with a ring magnet and a copper cap to reduce distortion. The rear acoustic design of the tweeter is critical for low distortion performance. The new Reference tweeter uses a large central vent which carries the rear acoustic radiation gently away from the back of the dome. More information can be found in Appendix V.

The dome itself is constructed from aluminium and uses KEF's patented stiffened dome design. This is a unique technology which enables the 25mm dome to extend into the ultrasonic bandwidth, more information can be found in Appendix IV.

The acoustic design around the tweeter dome is absolutely critical to the performance of the 11th generation Uni-Q design. It has taken most of KEF's 20 plus years of experience in designing the Uni-Q to fully understand and fully optimise this area.

The first stage in getting the optimal performance from the tweeter is to match the shape of the tweeter dome to the surrounding waveguide. If these two parts are not matched then large response irregularities can appear in the tweeter output. More information is given in Appendix III. Once the waveguide and dome geometries are matched, then the waveguide can be fully utilised to control the tweeter dispersion and to match this to the midrange driver. When combined with the optimal dome shape technology, the overall tweeter response is - unlike a conventional baffle-mounted tweeter - completely free from off-axis nulls. This difference can be clearly seen in Figure 39. This optimal geometry is patented, and hence only found in KEF Uni-Q loudspeakers.

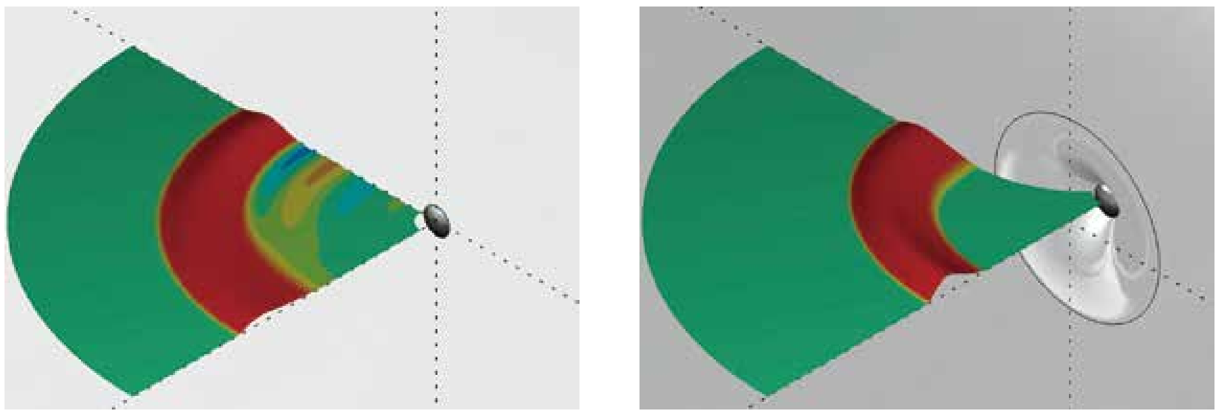


Figure 39. Comparison of a HF transient radiated by a conventional 25mm dome on baffle (left) and a optimally shaped 25mm dome in a waveguide (right).

The tangerine waveguide is another key piece of technology that is used by KEF to optimise the high frequency performance of the Uni-Q driver. A dome is close to the ideal shape for a loudspeaker diaphragm, particularly when placed in an optimal dome and waveguide configuration. However, a dome tweeter does not quite have the correct surface velocity to be the perfect radiator – when a signal is played through the tweeter the dome moves in one axis. Because the angle of the dome surface relative to this motion is greater towards the edge of the dome, this means that the surface-normal velocity is lower towards the dome perimeter. Ideally the surface-normal velocity would be constant over the entire dome surface. This difference is illustrated in figure 40.

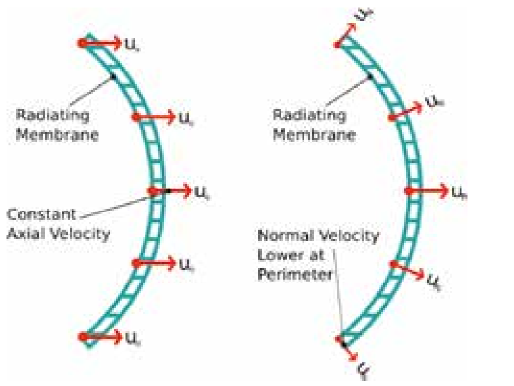


Figure 40. Comparison of tweeter dome axial and surface normal velocity.

The purpose of the tangerine waveguide is to correct for this difference between the real normal velocity and ideal normal velocity by directing the output from the dome into a small chamber under the tangerine waveguide and then to control the expansion of the sound into the waveguide through specially shaped channels. The design of the channels is a very complex process and required extensive use of FEA computer models in addition to involved mathematical analysis. A still from one of the computer models is shown in Figure 42 along with the final production part.

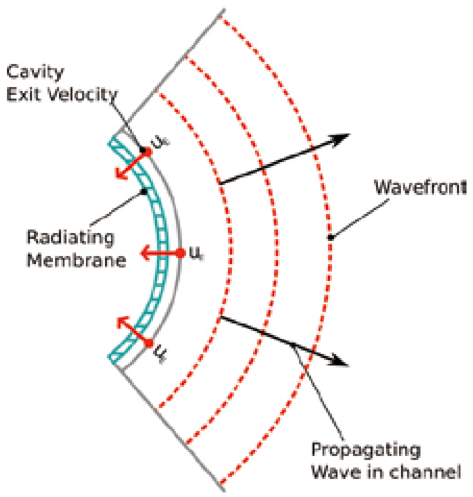


Figure 41. Illustration of the target behaviour of the tangerine waveguide.

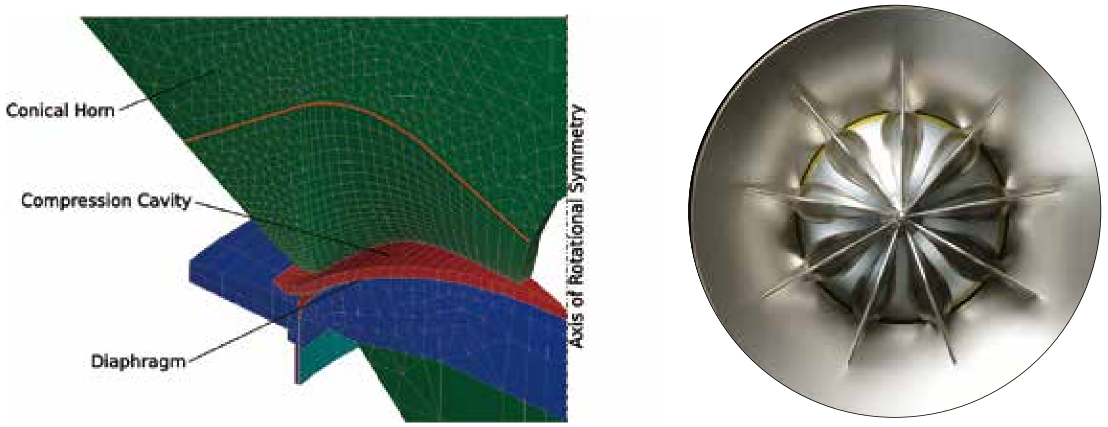


Figure 42. FEA computer model of the tangerine waveguide (left), final production waveguide (right).

Figure 43 shows the frequency response of a prototype of The Reference tweeter measured in a test waveguide. Also shown in this figure is a computer model of the same device. Note that the two responses are in very good agreement. The tweeter response is exceptionally smooth without any trace of resonance or interference. The sensitivity of the tweeter is greater at the lower end of the device due to the waveguide which actually assists in the acoustic coupling of the tweeter dome. In the 2kHz region, which is a critical area in most music material, the sensitivity is much higher than a conventional tweeter. In the crossover this downward tilt is corrected but the beneficial result is that the tweeter runs "cool" as less power is delivered to the tweeter at the lower end. For example at 2kHz almost one quarter of the power is fed to the tweeter, this is a huge advantage in terms of the maximum output and the linearity of the treble.

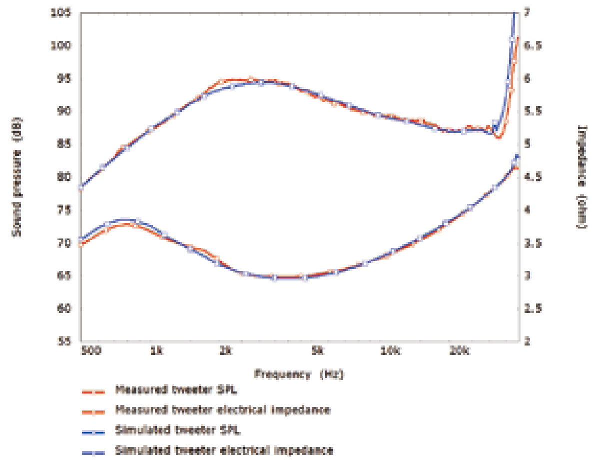


Figure 43. Frequency response of a prototype of The Reference tweeter measured in a test waveguide

In the full loudspeakers, the tweeter "waveguide" incorporates several physical parts and it is absolutely critical that these are designed to disturb the tweeter output as little as possible. The tangerine waveguide, the midrange cone, the midrange surround, the shadow flare and the baffle all form a single continuous waveguide surface. At each junction between two parts on this surface the gap is carefully controlled to minimise any effect and the parts meet exactly tangentially. The midrange surround uses a Z-Flex design so that it too forms this surface as smoothly as possible (see Appendix VI for more details).

The Midrange

The midrange driver has a difficult task: it must bridge the gap between the low and high frequencies, covering the range over which our ears are most sensitive. It is therefore exceptionally important that a midrange driver is well behaved above the working range, otherwise they might conflict with the tweeter output. The shape of the midrange driver cone, surround and surrounding objects are very important. In addition, the midrange driver cone experiences much higher vibrational forces than a bass driver cone because if the range over which it operates.

The Uni-Q uses a 5inch aluminium cone driver for the midrange output. The size of this driver is carefully optimised so that the dispersion of the tweeter and the midrange matches as well as possible at crossover. Aluminium is a very good choice of material for a loudspeaker diaphragm as it is both stiff and light and also easy to form into a complex shape. The stiffness and low mass mean that the cone operates as a rigidly body over the entire midrange region. This is unlike some competitor drivers where the cone operates in resonance over much of the working bandwidth. The benefit of a rigid body operation is that the sound radiated from the cone is free from irregularities due to differences in how the cone vibrates at different frequencies, providing the best possible coherence to the radiated sound. The rigid operation is also very important in controlling the dispersion of the driver. The target is for the dispersion of the midrange driver to slowly narrow monotonically as frequency increases. When a cone enters breakup the dispersion characteristic changes and typically becomes wider, this sudden change in dispersion is not desirable.

The issue with metal diaphragms is that when they do enter breakup, as they have little internal damping, very large irregularities in the response occur. These can be easily 15dB or more in magnitude. This is large enough to be a problem even if breakup occurs well above the crossover frequency. It is possible to add damping material directly to the cone to control these resonances, however, this is not a good solution as this direct damping application is very heavy and this results in a driver with low sensitivity.

KEF use a unique technology called cone neck control to avoid the traditional breakup problems of metal cone drivers. With cone neck control the cone is not rigidly connected to the voice coil of the driver. A resilient high damping link is used to connect the two parts together. This link is carefully designed and fine tuned with the help of computer modelling so that within the band of the driver the force from the voice coil is fully transferred to the cone. Above crossover, however, the resilient link begins to flex and to damp the cone motion. The effect on the driver response is quite dramatic. The breakup peak from the driver is reduced by around 15dB and the driver response is considerably smoother. The penalty is a small mass increase in the moving parts, albeit much less than using a direct damping approach. Figure 45 shows the modelled frequency response of a midrange driver with and without cone neck control technology.

The overall frequency response of the midrange driver is shown in Figure 46. The frequency response is very smooth and well controlled to well above the crossover frequency of approximately 2.5kHz.

Figure 44. Cone neck control resilient link between the cone and the voice coil.

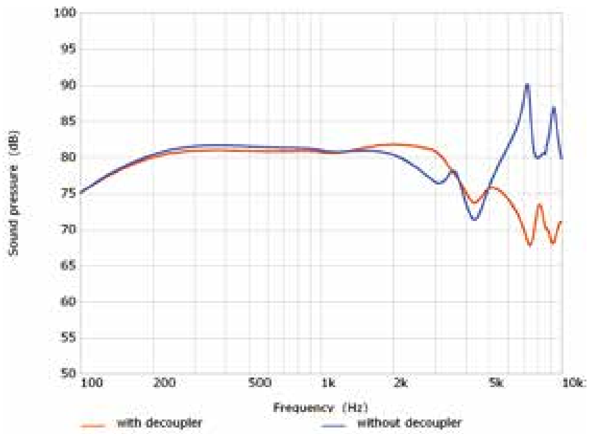


Figure 45. Midrange driver response with and without cone neck control technology.

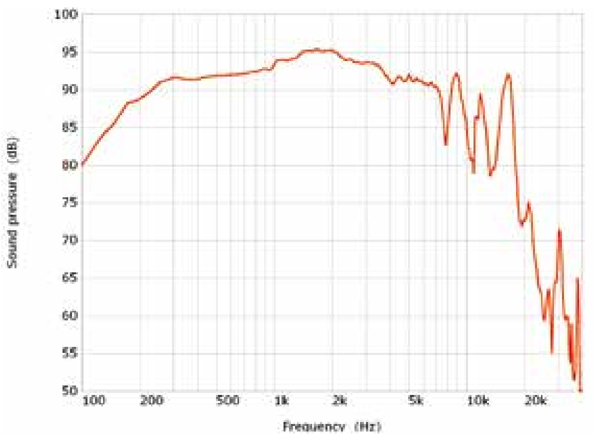


Figure 46. Axial frequency response of the midrange driver measured in 2pi at 1v/1m.

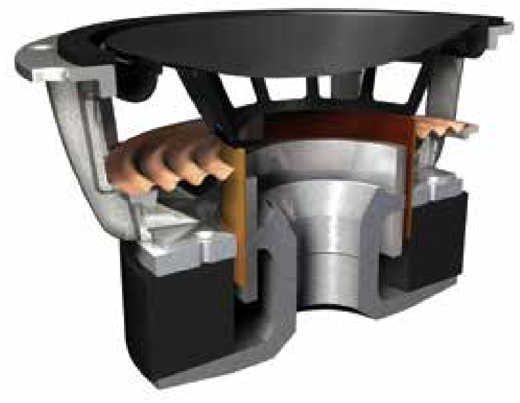
4.4.2 Low frequency drivers

The low frequency driver is shown in section in, right. Computer modelling was used extensively in the design of this driver in order to achieve the best possible performance. Rather than a conventional cone diaphragm, the driver uses a shallow aluminium disc. This gives the driver a much lower profile when placed in the loudspeaker baffle and ensures that it has the minimum effect on the response of the midrange and tweeter.

The rear of this disc is supported by a vented coupler. This is a unique technology to KEF that was first developed for the Blade loudspeaker. The vented coupler serves two purposes, firstly it allows very free movement of air away from the driver as the diaphragm moves. With a conventional driver, air can often be trapped inside the voice coil and this can lead to losses and distortion at high levels. At the centre of the motor system there is a large venting hole that further aids the movement of air at the rear of the driver. The geometry of the vented coupler is fine tuned using computer analysis so that it connects with the aluminium disc at a "nodal" position. A nodal position is a region on a structure where a particular resonance has no effect. By choosing a nodal position to drive the aluminium disc the first resonance of the disc is suppressed.

The voice coil on the bass drivers is particularly large compared to other drivers of a similar size. This is very important as at high levels a great deal of heat must be dissipated from the coil. A large voice coil has a fundamental advantage in this respect as for the same power input it will not become as hot, simply because it has a greater thermal mass. Secondly, the larger area aids the dissipation of this heat into the motor system and surrounding metal work. The overall result is much less power compression. The larger voice coil could present a problem in terms of mass, however to overcome this issue the coil uses aluminium wire.

The driver uses an undercut pole and an overhung voice coil design. Both of these features are to maximise the excursion capability of the driver for low distortion output even at high levels. The surround and suspension of the driver are also carefully designed with this in mind. In order to control the distortion in the upper bass and lower midrange region the motor system incorporates large aluminium Faraday rings above and below the magnetic gap.



Bass driver cross section

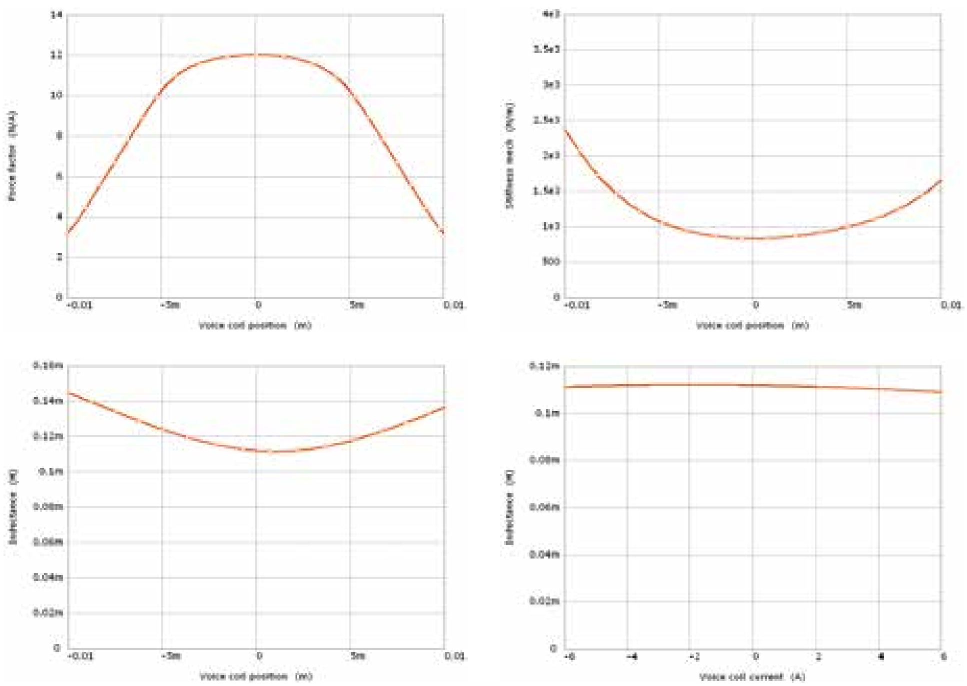


Figure 47. Non-linear LF driver parameters measured with Klippel analyser system, 16Ohm driver version.

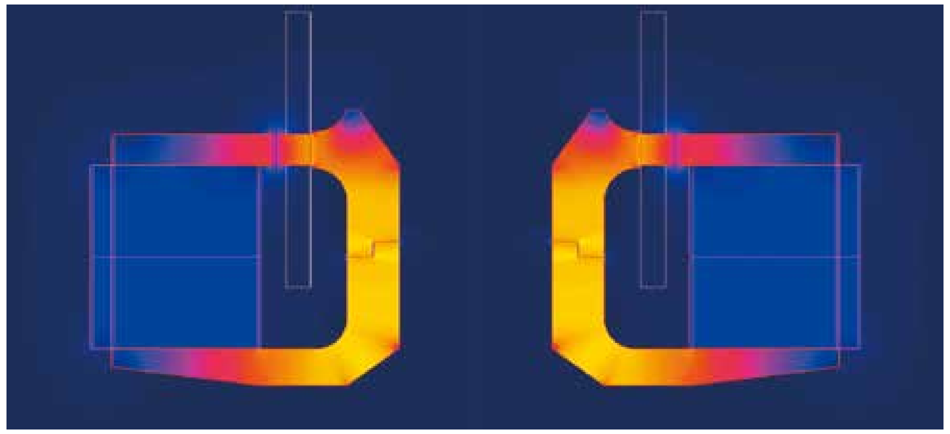


Figure 48. Static magnetic FEA simulation of LF driver motor with flux density shown as colour.

4.4.3 Motor system design

The Reference system and driver design has specifically focused on achieving the lowest possible distortion performance. Though there are several sources of non-linearity in loudspeakers, it is the driver motor system design that is most critical in achieving low distortion in the midband region where the diaphragm motion is small. At KEF, advanced FEA computer analysis methods are used to design the motor system and to analyse the distortion performance. For the Uni-Q driver, due to the close proximity of the mid and high frequency motors, the two magnetic systems must be designed to work together. Figure 49 shows a static magnetic FEA analysis of the MF and HF motor systems, the magnetic flux density due to the permanent neodymium magnets is shown. The field of each motor is concentrated in the magnetic gaps, there is little stray field. The undercut design on the midrange driver can be clearly seen, as can the large central venting hole used to accommodate the rear loading of the tweeter.

Reference motor systems use a mix of underhung (tweeter) and overhung (low frequency/midrange) voice coil designs to ensure that the the force from the motor system is not modulated with the motion of the driver diaphragm. In addition both motor systems incorporate carefully designed conductive regions. The tweeter has a copper sleeve on the inside of the magnetic gap, the low frequency and midrange use large aluminium rings above and below the gap. These conductive regions are inductively coupled to the voice coil and they are used to cancel the self-inductive behaviour of the voice coil. The placement and size of the rings are fine tuned using computer analysis to ensure that the voice coil inductance is not modulated with the voice coil position. In addition, the cancellation of the voice coil inductance itself is very effective in reducing distortion. This is because the magnetic moment generated by the voice coil will, if not controlled, modulate the permanent magnetic field. For example Figure 50 shows the magnetic flux density generated in the steel parts near the midrange motor gap due to the voice coil current, with the conductive rings in place the magnitude of this flux is significantly lower. Figure 51 shows the corresponding inductance for two voice coil positions.

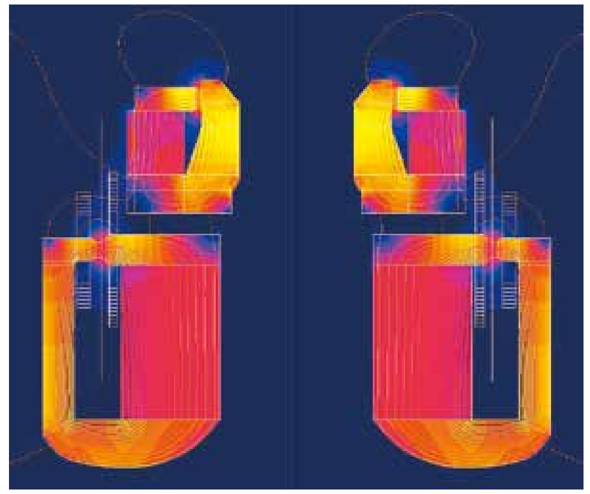


Figure 49. FEA magnetic analysis of Uni-Q motor showing flux density due to permanent magnets.

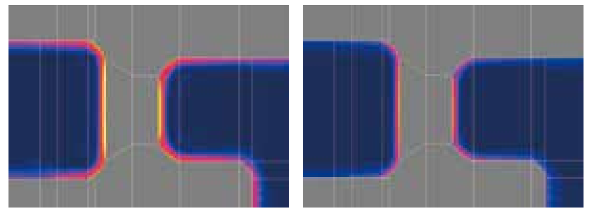


Figure 50. Flux density modulation of midrange motor system gap area at 2000Hz without conductive regions (left) and with conductive regions (right).

The result of this attention to detail in the design of the drivers and motor systems in general is that the distortion performance of the loudspeakers is exceptional. Figure 52 shows the percentage harmonic distortion of a prototype of Reference 5 at a 90dB/1m output level. The mid-band distortion is less than 0.07%.

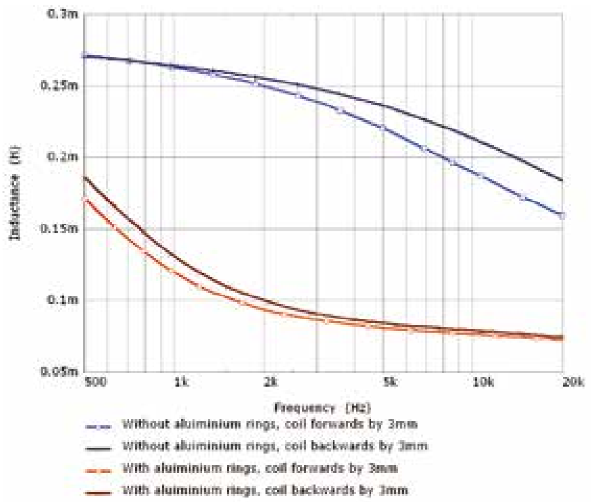


Figure 51. Midrange voice coil inductance versus frequency for motor system with and without aluminium rings.

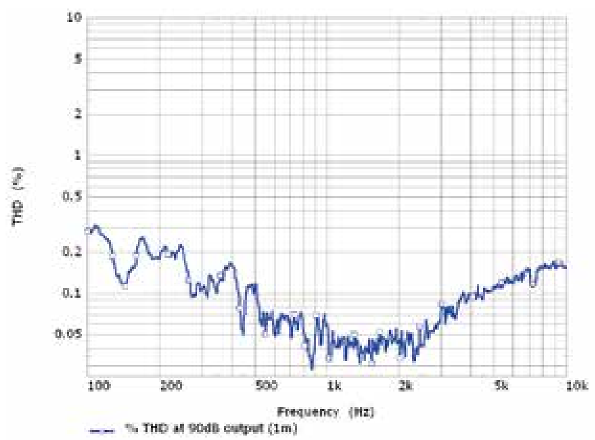


Figure 52. Measured THD of Reference 5 prototype as a percentage of the output SPL at a level of 90dB/1m.

4.5 Crossover design

The crossover is the electrical circuit that divides the input signal and sends appropriate parts to the relevant drivers in the loudspeaker system. The Reference loudspeakers are passive designs, this means that the electrical crossover circuit is not powered and the components must perform at the high power levels that are sent to the loudspeaker by the driving amplifier. Crossover design can be a complex subject and there are plenty of strong opinions of how it should be approached. The approach taken to the design of The Reference crossovers are outlined in the following sections.

4.5.1 Crossover component distortions

Passive crossovers are constructed from simple electronic components such as resistors, capacitors and inductors. Real resistors, capacitors and inductors are not ideal, and all introduce some level of distortion to the system. Selecting the correct types of each component can greatly affect the performance of the crossover, and choosing the wrong components can result in a crossover that introduces a significant amount of distortion into the loudspeaker system.

For The Reference, extensive objective testing of individual components were carried out to identify those with the lowest distortion. This study turned out some extremely surprising results. In many cases there was little correlation between the size, cost or published specification and the measured component distortion. For example, Figure 53 shows the distortion levels of three identical value inductors of a similar size, cost and DC resistance, but of different constructions. As can be seen, the worst inductor had 10 times more distortion than the best.

Figure 54 shows a comparison of the measured distortion for some different capacitors. Again, there were significant differences between different types. Using this testing allowed very quick identification of the best components for further investigation in listening tests.

This testing approach was also applied to fully constructed crossovers, for example Figure 55 shows the measured distortion of a prototype Reference 5 high frequency crossover circuit. Both curves are for the same circuit, but one has been built with a single poor component choice, whilst the other has been made using only low distortion components. This shows that a single bad component can introduce a large level of distortion, whilst correctly selecting the components can reduce the distortion generated by the crossover to a negligible level. By using this methodology through the whole crossover network, the result is a lower distortion and a better sounding loudspeaker system.

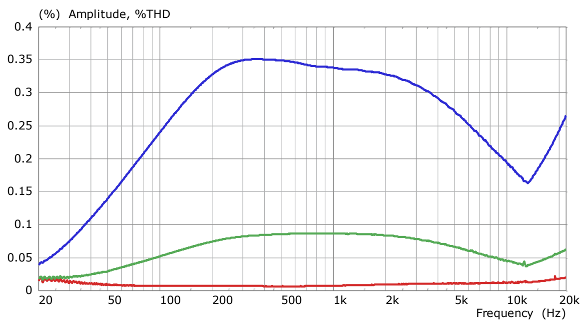


Figure 53. Measured component percentage THD at 20V input level for three inductors of the same inductance, similar cost and similar resistance but different constructions and manufacturers.

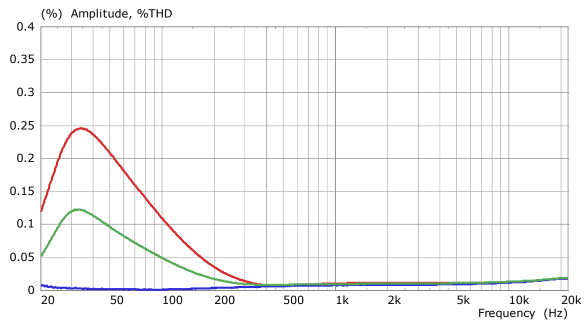


Figure 54. Measured component percentage THD at 20V input level for three capacitors of the same capacitance, but different constructions and manufacturers.

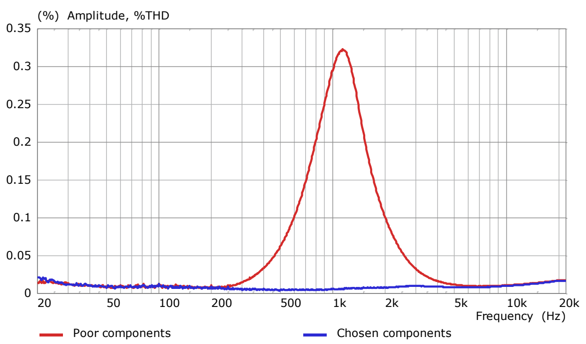


Figure 55. Distortion measurement of full HF crossover at 20V input level with two different component selections, note the component values and filter responses are identical.

4.5.2 Crossover filter order

Crossover filters have been widely studied and there are numerous publications on different approaches to their design. The most widely known are the classical Butterworth and Linkwitz-Riley types. Both of these families of filters can be implemented in either active electronics or with passive electronics[8]. These filters have the characteristic that when the output of a complimentary high-pass filter and low-pass filter are summed the resulting signal has a perfectly flat frequency response. In a loudspeaker the high-pass filter would feed the signal to the tweeter and the low pass filter would feed the signal to the woofer. The order of a filter determines how much attenuation is present in the filter stop band. The first four of these classical crossover designs can be seen in Figure 56, all filters result in a crossover frequency of exactly 1000Hz and result in a flat summed response. The effect of increasing filter order can be clearly seen - the 1st order high-pass filter provides 20dB of attenuation to the tweeter at 100Hz, whereas the 2nd order high-pass filter provides 40dB of attenuation to the tweeter at 100Hz.

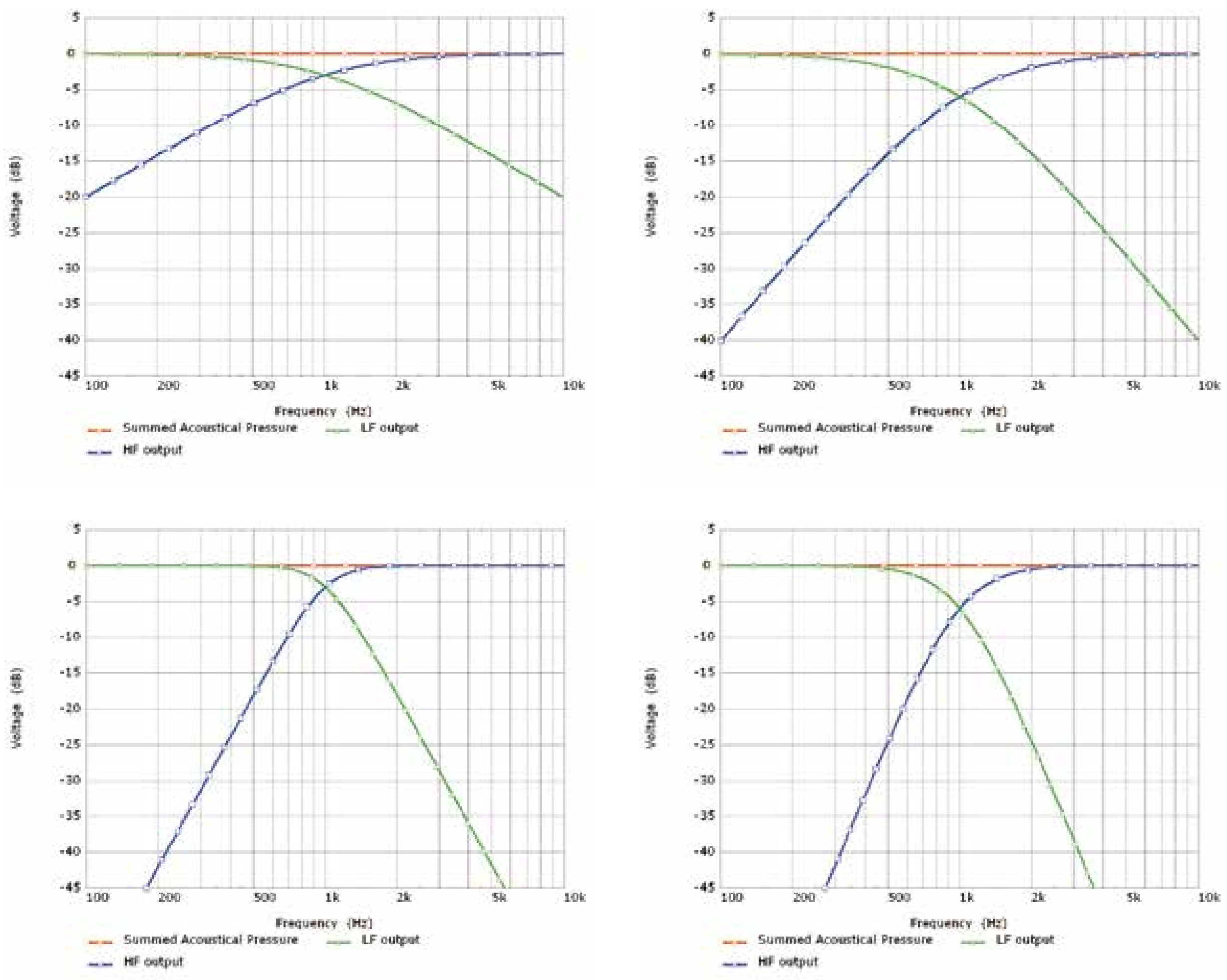


Figure 56. Classical crossover filter arrangements, from top left to bottom right, 1st order Butterworth, 2nd order Linkwitz-Riley, 3rd order Butterworth, 4th order Linkwitz-Riley.

8 This is only true given certain constraints on the load, typically in text books the component values required to achieve the classic filter types are given for a purely resistive load.

In terms of a loudspeaker one might think that a higher order will always be preferable, because the tweeter can be protected better from low frequency signals and the woofer signal can be curtailed before any high frequency cone breakup. However, there are several considerations that are not shown by the frequency level response. Firstly, a greater number of components are required for a higher order crossover and this makes it much more difficult to design a passive circuit that is completely transparent and distortion free. Secondly, with increasing filter order the summed signal becomes increasingly smeared in time. One way to look at this time smearing is in terms of the relative transmission delay of the crossover to different frequencies, this information is shown in Figure 57. It can be clearly seen that the time smearing is worse for the higher order crossovers. Note that this time smearing is inherent to the type of filter rather than the realisation of the filter. This means that irrespective of whether the filter is passive active or digital, a Butterworth 3rd order crossover filter will result in the same group delay characteristic.

Polarity inversion of the some of the drivers in a multi-way loudspeaker in order to avoid interference dips at the crossover frequency is commonplace. This is necessary because of the time smearing of the crossover filters and the drivers themselves. This can be a source of confusion. For example the classical 2nd order Linkwitz-Riley crossover requires one of the driver polarities to be reversed, whereas the 4th order Linkwitz-Riley does not. The time smearing of a second order crossover means that at the crossover frequency the low frequency signal is delayed by half a wave period and the polarity reversal is necessary to achieve proper summation between the two drivers. The 4th order crossover has more time smearing and the low frequency signal is delayed by a full wave period and hence no polarity reversal is required. There is no advantage to the fact that tweeter polarity is not inverted in the 4th order system, this is simply a consequence of the greater time smearing.

The 1st order Butterworth crossover filter has the interesting property that there is no time smearing. This fact, combined with it's simplicity, has made it very popular among audiophile loudspeakers. However, the lack of time smearing is something of a fallacy. While it is absolutely true that complimentary 1st order Butterworth filters will sum to a result without time smearing, in these classical crossover types the

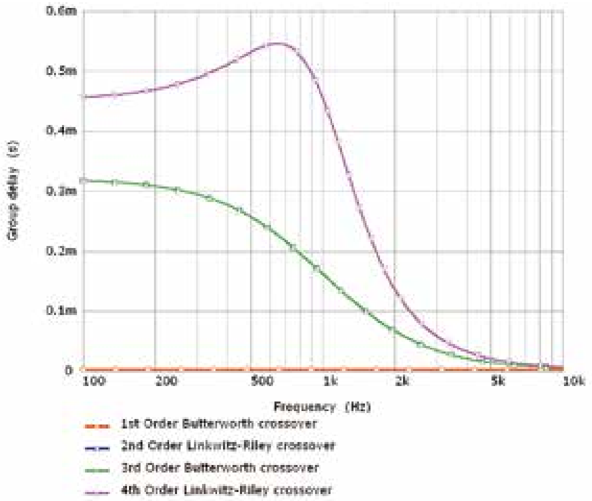


Figure 57. Group delay of classical crossover filter arrangements. Note that the group delay of the 2nd order Linkwitz-Riley and the 3rd order Butterworth crossover are identical

raw response of the loudspeaker drivers is completely neglected. In reality the raw driver responses must be taken into account. Figure 58 demonstrates this. The left figure shows the raw response of a theoretical tweeter with an exceptional response of -3dB at 300Hz. The centre and right hand figure show the overall response and group delay when a 1st order Butterworth crossover is used with this tweeter and a perfect woofer. Even with this exceptional tweeter response the overall frequency response is no longer flat and there is significant time smearing evident in the group delay plot. The reason that this approach did not work is that the response of the drivers must be taken into account when designing the crossover, it is the overall transfer function of crossover filter plus driver that counts. Unless the drivers themselves have no roll-off then it will not be possible to achieve the theoretical zero time smear of the 1st order Butterworth crossovers. In practice this is simply not possible.

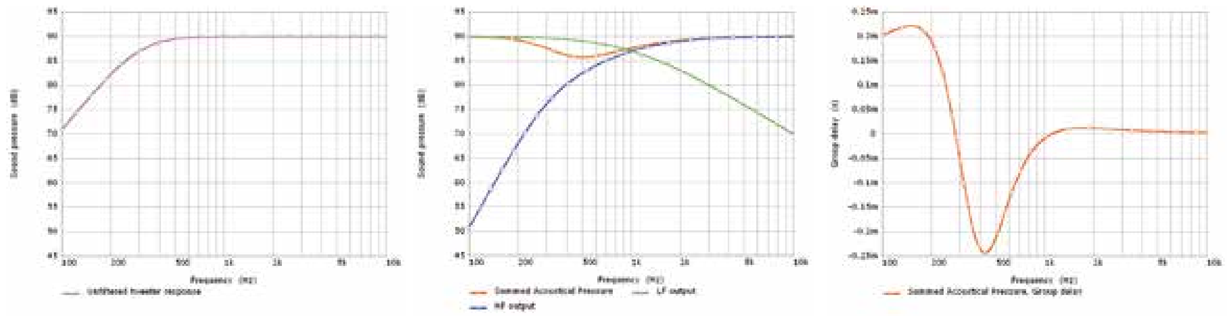


Figure 58. Raw tweeter response (left), 1st order Butterworth crossover using this tweeter (centre), overall group delay (right).

It is possible to achieve a reasonable response using this tweeter and a first order crossover, by inverting the tweeter polarity and manually optimising the filters, as shown in Figure 59, however note that the group delay is now comparable to a second order crossover.

An important point to note is that the time smearing from a crossover is dependent upon the crossover frequency. This is because the group delay introduced by a crossover is proportional to the wave period at the crossover frequency. This means that time smearing is much more problematic with low frequency crossovers. This is a significant reason why a 3-way design approach has been taken with The Reference, in order to avoid a very low crossover frequency necessary on a 4-way system. It is also one of the reasons why it is extremely difficult to crossover convincingly between main loudspeaker and a subwoofer.

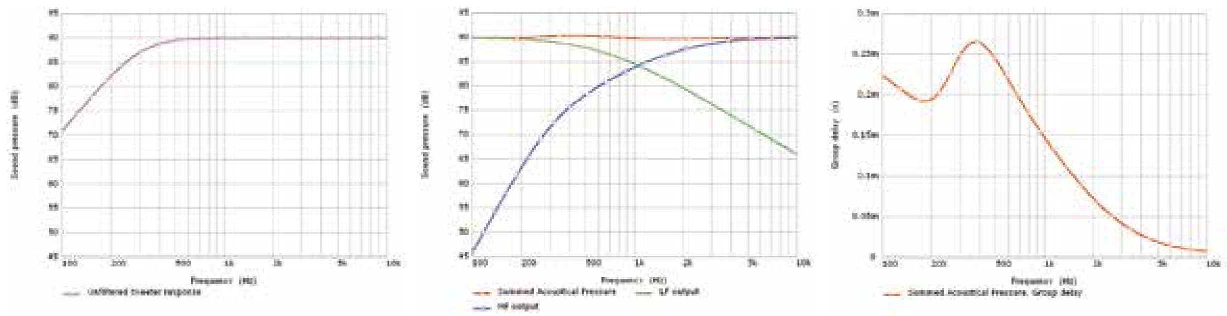


Figure 59. Raw tweeter response (left), manually optimised 1st order crossover using this tweeter (centre), overall group delay (right).

Using digital filters it is possible to achieve crossover filtering without any time smearing and with a high order roll off. The filters required to achieve this exhibit an unusual time domain impulse responses, an example is shown in Figure 60, with a pre-ringing before the main body of the impulse. There is some evidence to suggest that this pre-ringing is responsible for audible artefacts [13] [14]. Provided that the output from the drivers arrives at the listener in sync, the pre-ringing of the filters will cancel in the summation between the high and low frequency sections of the crossover. However, with most loudspeakers having separate treble and midrange drivers the relative arrival time of the tweeter and midrange signal is not constant and varies with listener position. Even with the listener located in a position where the sound arrives in sync from the two drivers the reflections from the room boundaries are likely to contain traces of the pre-ringing artefact.

With conventional loudspeaker systems the choice of crossover filter has a large effect on the dispersion of the loudspeaker. The dispersion of The Reference models, due to the Uni-Q driver and careful design of the low frequency directivity, is largely unaffected by the filter choice. This allows much more freedom to design a crossover with as little compromise as possible.

The Reference crossovers use a combination of 1st and 2nd order electrical filters. These filters do not follow any classical type but rather are carefully designed, using a combination of computer modelling and listening tests, to ensure good overall summation and equalisation of the natural driver responses. In some places a Thiele style notch is incorporated into some of the filter sections [15]. The emphasis has been to try and use the simplest possible filters that are able to fully control the drivers. Filters with an order greater than two have been avoided because of the additional components that they add to the signal path (see Figure 61)

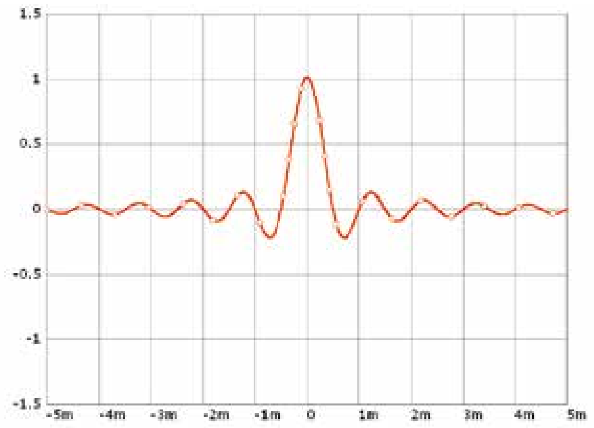


Figure 60. Example of a linear phase filter impulse response.

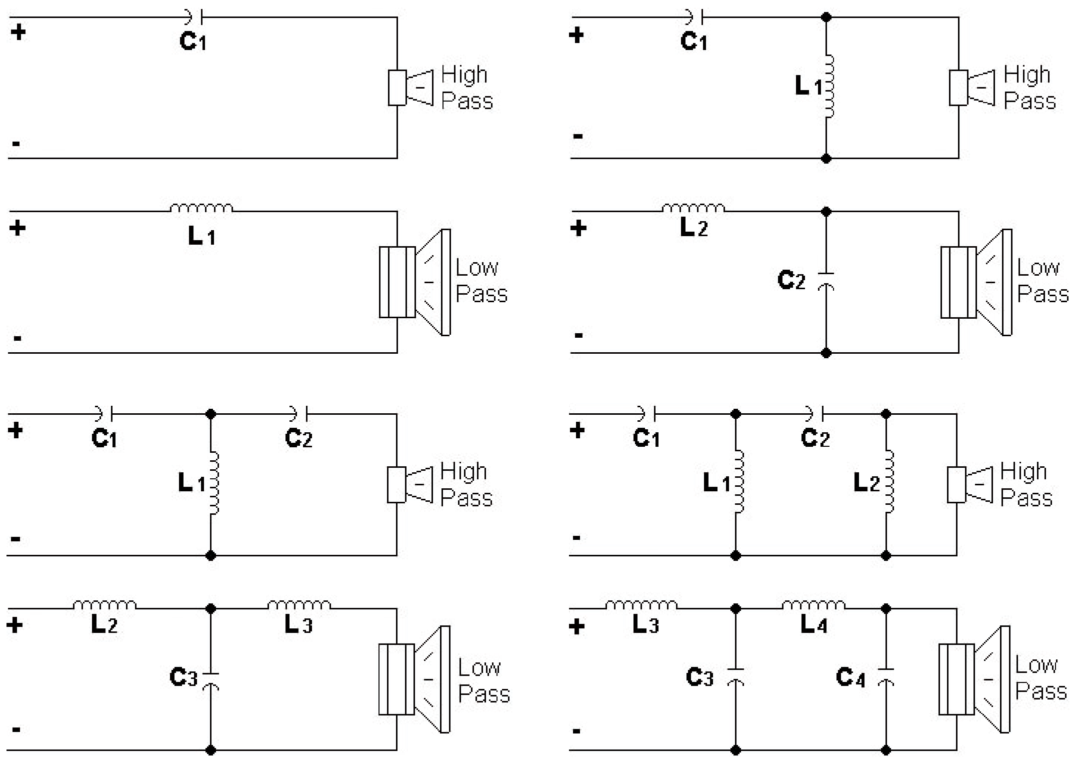


Figure 61. Passive crossover component layouts for 1st (top left), 2nd (top right), 3rd (bottom left) and 4th (bottom right) orders

4.5.3 Impedance conjugation and amplifier load

The Reference loudspeakers make use of impedance conjugation networks in the crossovers. Conjugate networks are required because real drivers have an electrical impedance that is not constant with frequency. Figure 62 shows a typical driver's electrical impedance magnitude compared to a resistor. With a resistive load, a passive filter circuit can achieve any of the classical filter responses exactly. Figure 62 demonstrates this for a 350Hz 2nd order Linkwitz-Riley crossover, with a resistive load the correct filter response is achieved. Also shown is the response achieved with the same passive filter loaded with the driver shown in Figure 62, due to the electrical interaction between the passive filter and the driver, the filter response target is not achieved. It is possible to get a slightly better match by manually optimising the crossover components, the result is shown in Figure 62. However, the version with the driver load still shows some response irregularity as well as attenuation of the very low frequency response.

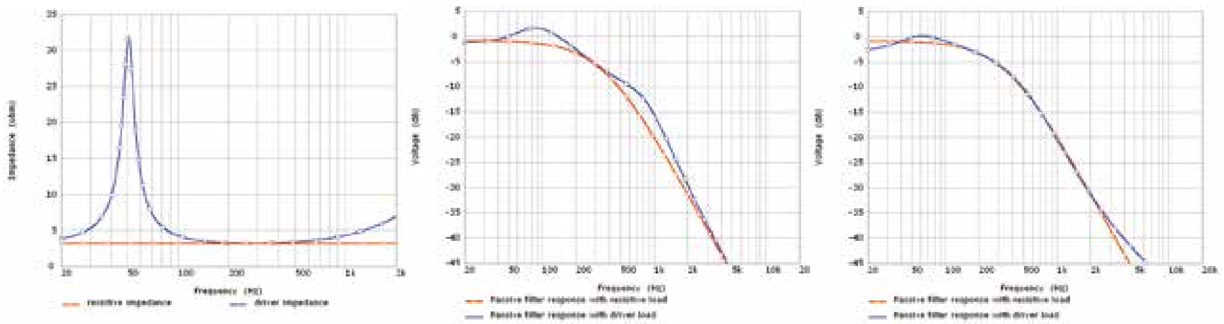


Figure 62. a. Comparison of electrical the impedance of a real loudspeaker driver and a purely resistive impedance (left).
b. Passive low-pass filter response with each load, filter designed to give 350Hz Linkwitz-Riley response with a 3.2 Ohm load (middle).
c. Passive low-pass filter response with resistive and real loudspeaker driver load, filter designed to give best match to 350Hz Linkwitz-Riley response in each case (right).

These interaction issues can be overcome by using impedance conjugation networks. This entails adding extra crossover components in parallel with the drivers to compensate for the natural driver electrical impedance. For example, figure 63 shows the input impedance for the same driver discussed above with the addition of a conjugation network to compensate for the impedance peak at driver resonance and for the rising impedance at high frequencies from the coil inductance. With both conjugation networks in place the electrical impedance of the speaker is almost perfectly resistive and consequently the response irregularity issues above completely disappear.

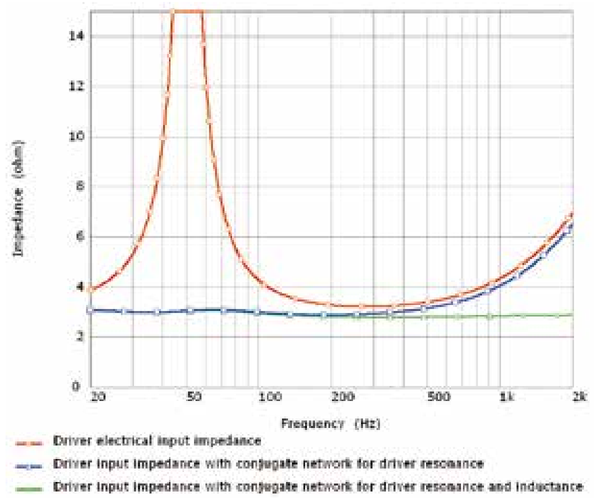
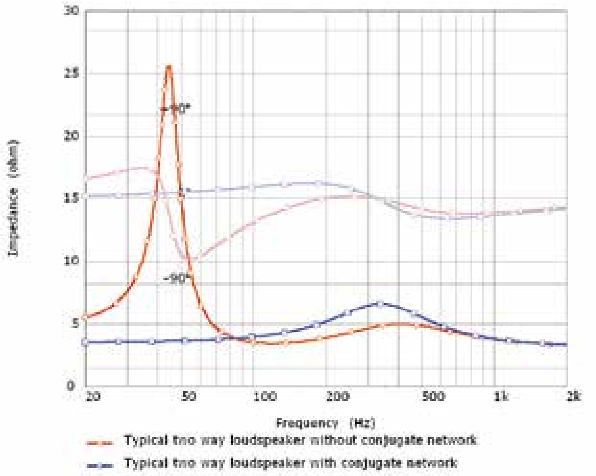


Figure 63. Typical loudspeaker driver electrical input impedance with impedance conjugation components

There is also a big change in the amplifier loading when an impedance conjugation network is added to a crossover. Figure 64 shows this amplifier load difference for a typical two-way closed-box loudspeaker system as an impedance magnitude and phase chart. This type of figure is widely used in various publications to aid consumers to asses the relative driving difficulty when comparing loudspeakers. Unfortunately this information does not lend itself to easy comparison as both low impedance magnitude and high impedance phase lead to difficult loudspeaker loads. For example, considering Figure 64, the impedance magnitude of the version with the conjugate network is lower at 60Hz whereas the version without a conjugate network has a significantly higher impedance phase angle. Another approach to looking at the impedance data has been suggested by Keith Howard [16], based on the work of Benjamin [17]. This approach presents the equivalent purely resistive load that would result in the same peak power dissipation in the output stage of a class B amplifier, thus combining impedance magnitude and phase into one figure of merit. This measure is called the “equivalent peak dissipation resistance” (EPDR), plotted in Figure 64 for the same two loudspeakers. From this plot it is clear that the loudspeaker without the conjugation network is a significantly more difficult load to drive. Not only does this mean that a loudspeaker with impedance conjugation may be used with a wider range of amplifiers, but it also means that the loudspeaker will perform better with any given amplifier because less power is dissipated in the amplifier output stage in trying to drive the awkward load.



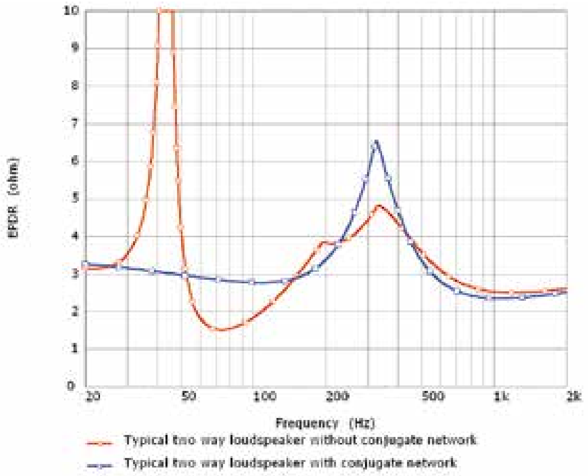
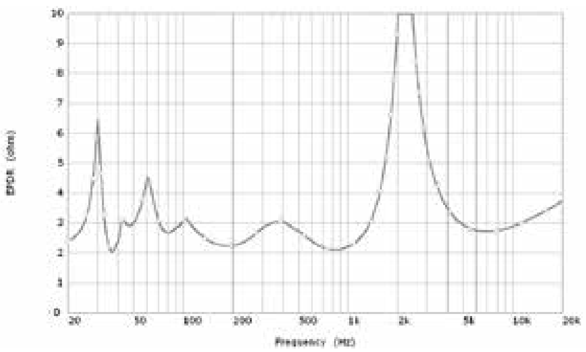


Figure 64. a. Typical closed-box two-way loudspeaker system electrical input impedance with and without a conjugate network in the crossover (left). b. Equivalent Peak Dissipation Resistance for the same two systems (right).

There is also a clear disadvantage to using impedance conjugation networks and that is that the number of components in the crossover is increased significantly. The impact of this is that there is more opportunity for the crossover to generate artefacts and distortion into the signal. For The Reference impedance conjugation has been used selectively where necessary to control the crossover to loudspeaker interface and to ensure that the loudspeaker load is reasonable for a good quality amplifier. By selective application of this approach it is possible to benefit from the advantages described above while not adding too many additional components to the crossover.

The result is a good balance between the different compromises whilst still retaining a relatively easy load for the amplifier. The EPDR of an early prototype of Reference 5 is shown in Figure 65 below in comparison to a competitor loudspeaker.



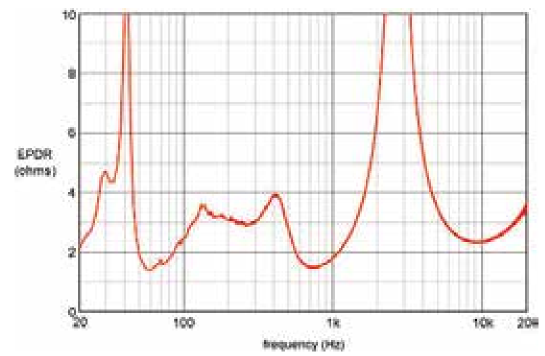


Figure 65. EPDR of Reference 5 prototype system (left) compared to competitor loudspeaker.

5 Voicing the loudspeakers

The crossovers were initially designed from measured responses of the individual drivers mounted in the final enclosure and based on the underlying approach outlined in section 4.5. Listening tests were used extensively to arrive at the final design for each model.

The listening sessions for the early prototype loudspeakers spanned several months and involved a large number of experienced KEF personnel, each with their own personal music tastes and preferences. The listening process is very much integrated with the technical engineering. Concerns raised during listening were rigorously investigated using various measurement techniques to try and identify any underlying physical cause. In numerous cases a clear correlation was found between objective data and subjective conjecture. This approach, though time consuming, is very effective. When issues are raised during listening it is often tempting to immediately change the loudspeaker voicing in an attempt to improve the loudspeaker’s character. However, if an underlying problem exists then it will remain unresolved and the loudspeaker’s balance will be compromised in order to disguise it.

Particular attention was paid to the component selection and the crossover layout. The short-listed components following the component distortion measurements described in section 4.5.1 were individually auditioned to ensure they did not limit the perceived sound quality. The capacitors for the higher-frequency section are vibration damped with mastic, to prevent sonic deterioration due to vibration. The crossovers are split into two different circuit boards and mounted apart from one another inside the loudspeaker. The crossover circuit boards are decoupled from the cabinet walls to minimise vibration transfer to the components. Each board only contains three inductors in order that they can be oriented in perpendicular planes to minimise cross talk.

Gratifyingly, as this systematic procedure progressed the preferred subjective performance merged with the balance that gave the best objective measurements. In particular the lowest distortion and the smoothest and flattest on and off-axis frequency response.

6 Summary

The Reference has been developed with a strong belief that the only way to design a truly exceptional loudspeaker is through rigorous engineering and attention to detail. The strength and depth of the methods described in this paper give just a small window into the meticulous approach necessary to achieve best possible performance.

Loudspeaker design is a great challenge for the engineer, uniquely combining several different interacting physical mechanisms. Allied to this is the astonishing sensitivity of the auditory system, which dictates the loudspeaker must have an extremely wide bandwidth, high dynamic range, exceptional linearity and controlled dispersion. There are few other devices that require the same level of fine control over the behaviour of physically moving parts. This complexity perhaps explains why "state of the art" in loudspeakers is still advancing despite being a 90-year-old discipline.

Judgement and compromise are a large part of the loudspeaker engineer's skills. It is often the case that the pursuit of one particular performance aspect is to the detriment of another. It is only with experience and technical innovation that the optimum operating points can be found. KEF has a long history of innovation and acoustic technology development, and this is the critical foundation to the development of this range.

The result is a range of loudspeakers that are themselves with little identifiable character, "without drama, without exaggeration, without artifice", such that the focus of the listener falls solely upon the music.

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Appendix I. Uni-Q

Of the many landmark innovations KEF has pioneered, arguably the greatest of all is the Uni-Q® point source driver array, with its outstanding acoustic clarity and off-axis dispersion. Through continuous innovation and development since 1988, the Uni-Q driver array achieves a level of sound quality over a broad area simply not achieved from conventional speakers.

Sounds come at you so naturally that it seems as if the musicians or actors are actually there in front of you. Whether you are in the centre of the room or off to one side, recordings sound real and convincing. Uni-Q achieves this because, unlike conventional speakers, the sound that is critical to the experience comes from the same point in space, and is produced in a controlled and continuous way over the whole audio range.

It is not easy to produce a convincing and realistic illusion of a live performance because the sound from a high quality loudspeaker does not come from a single source, or drive unit. Two or more units are required to faithfully reproduce the entire audio spectrum from the low bass produced by a concert organ or a cinema explosion, to the delicate nuances of the human voice in the midrange right up to the shimmering treble of cymbals. Most loudspeakers have the midrange and treble drive units mounted one above the other, so the sound is coming from two different places, causing audio 'confusion' and losing the chance of achieving a truly natural sound. With Uni-Q, the midrange and treble units are mounted at precisely the same point in space – allowing them to integrate perfectly and create the ideal sound field for the listener to experience a convincingly natural sound.

I.I. Acoustical point source

It has been well known for many years in the audio industry that one of the ideal forms for a loudspeaker is the 'point source'- where all the sound is radiated from the same point in space. To do this, the drive units (for example, the bass and treble units in a two-way system) need to be mounted so that their acoustic centres are at the same place. The problem in achieving this was the sheer physical size of the treble unit, which prevented it from fitting in the centre of the bass unit. Various forms of co-axial units emerged where the tweeter was mounted either in front of or behind the acoustic centre of the bass unit but these have significant drawbacks. The key to the invention of Uni-Q was the arrival on the market of a new magnetic material called Neodymium-Iron-Boron, which has ten times the magnetic strength of a conventional ferrite magnet. This material allowed a high sensitivity treble unit to be made small enough to fit within the voice coil diameter of a typical bass unit and so be placed at the precise point where the acoustic sources are 'coincident'.

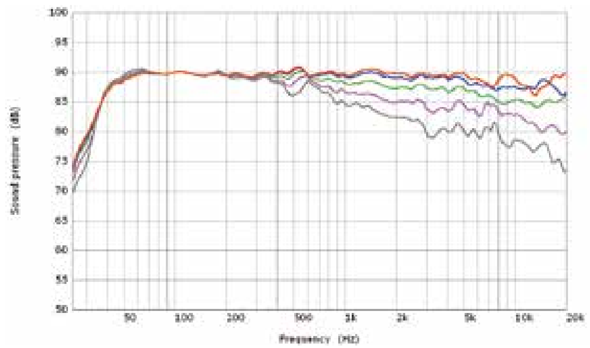

Figure 66. On and off-axis response curves of a well designed Uni-Q system.

With the acoustic centres at the same point in space the acoustic outputs of the bass and treble units are 'time-aligned' in all directions allowing the designer to achieve perfect integration between the units not just on one axis, as is the situation with vertically separated units, but in all directions. The first advantage of Uni-Q, therefore, is the lack of the vertical interference pattern of separated bass and treble units, which restricts the region of high quality sound output to only +/- 10 degrees above and below the principle axis. This same effect not only limits the vertical listening area but also produces a dip in the total energy output in the bass/treble crossover region, causing a distortion of the reverberant energy in the listening room. In Uni-Q systems this effect is completely eliminated.

I.II. Matched Directivity

The second advantage of Uni-Q is what we call 'matched directivity'. With the treble unit mounted at the centre of the bass driver's cone, its directivity (the spread of sound away from the main axis) is governed by the angle of the cone, which also largely determines the directivity of the bass driver. So with the coincident mounting of the two units, the directivity of the treble unit is adjusted to be virtually the same as that of the bass driver. As a listener moves away from the main axis, the output of the treble unit falls off at approximately the same rate as that of the bass unit, thus improving the uniformity of tonal balance across the listening area, and improving the off-axis stereo imaging. The listener is not, therefore, as limited to a central 'sweet-spot' as with conventional speakers. And, of course, the same is also happening in the vertical plane, so the reverberant energy in the listening room maintains an even balance, adding realistic ambience to the sound without introducing tonal colorations. The directivity is often referred to in engineering terms as the 'Q', and the 'Unifying' of the 'Q' gives rise to the name 'Uni-Q.

From a listener's perspective, the combination of the matched directivity and precise time alignment in all directions gives significantly improved stereo imaging over a wide listening area, the realism of which is enhanced by the even balance of the reverberant energy within the listening room.

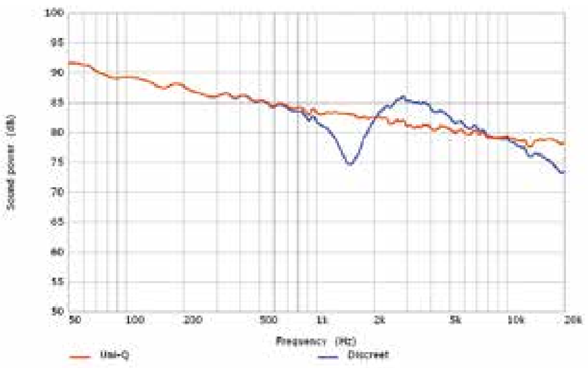

Figure 67. Comparison of power response of a Uni-Q and discreet loudspeaker system.

Appendix II. Tangerine waveguide

The Tangerine Waveguide is a patented KEF Technology which is now used in a number of products throughout the range. The technology was developed following research work into compression drivers, which are used in high power systems for concerts [18].

Compression drivers are very susceptible to acoustic resonances which occur in front of the tweeter dome. Whilst looking into the behaviour of compression drivers in detail, it was realised that the source of these acoustic resonances is also present in a normal direct radiating tweeter. The Tangerine Waveguide is designed to compensate for these problems, improving the coupling between the tweeter dome and the air [19].

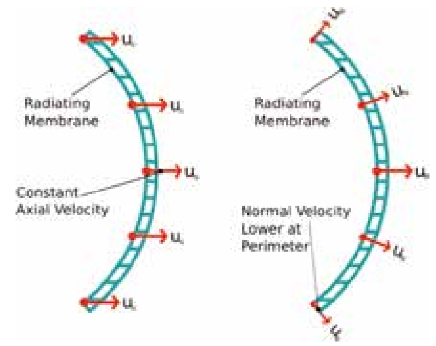

Figure 68. Comparison of tweeter dome axial and surface normal velocity.

As illustrated in Figure 68, because the dome moves in one direction, in reaction to the force from the motor system, the surface normal velocity at towards the dome perimeter is smaller. This is because the dome surface here is at an angle compared to the direction of motion. Ideally the surface normal velocity would be the same over the entire dome surface. This ideal behaviour is not possible because it would require the dome surface to stretch. The Tangerine Waveguide acts like an acoustic lens to adapt the surface velocity of the dome to the correct acoustic wave front to propagate in the Uni-Q waveguides.

Figure 69. Tangerine waveguide in position on top of tweeter dome.

With the Tangerine in place, the tweeter dispersion is improved at the very top of the audio band and the sensitivity is increased. The sensitivity increase is due to improved impedance matching between the mechanical and acoustical systems.

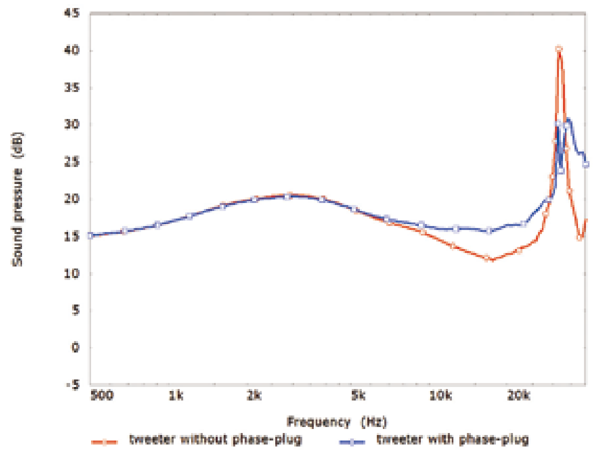

Figure 70. Simulated 1m axis pressure response of thetweeter when driven with a constant unit harmonic acceleration with and without Tangerine waveguide.

Appendix III. Optimal dome shape

It is a common misconception in audio that the perfect environment for a tweeter is a plain flat baffle and that any waveguide or discontinuity will always degrade the performance even if carefully designed. Recent work by KEF engineers has shown this not to be the case [20].

In fact, if exactly the right shape of dome and waveguide are used together their combined performance can beat the conventional ideal of tweeter in flat baffle. This is a KEF patented technology.

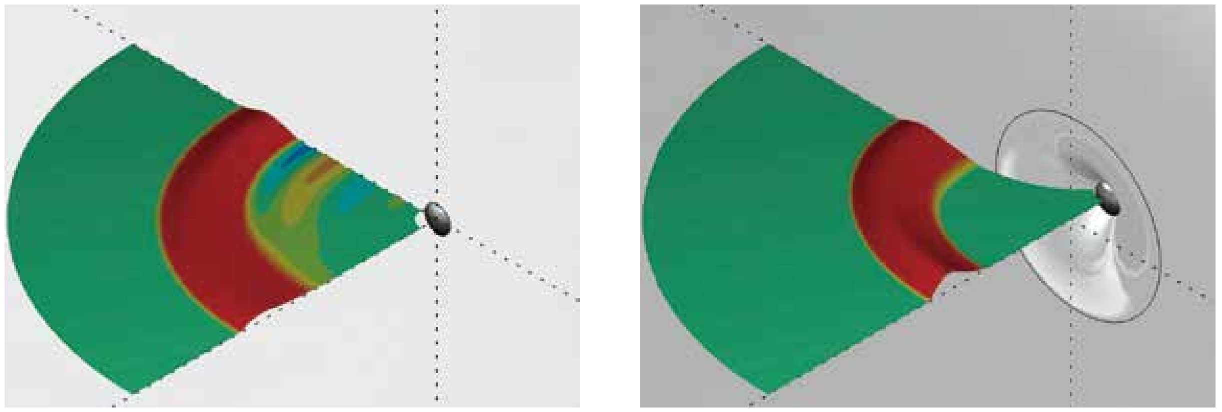

Figure 71. Comparison of a HF transient radiated by a conventional 25mm dome on baffle (left) and an optimally shaped 25mm dome in a waveguide (right).

The soundfield in front of two tweeters has been modelled using Finite Element Analysis, the results are seen in Figure 71. A short pulse is sent to the tweeters and the corresponding sound wave can be seen radiating through the air. On the left is a model of a simple dome tweeter in a baffle; on the right a model of a dome tweeter and waveguide in the same configuration found in the KEF Uni-Q. The dome and waveguide model on the right uses the patented KEF Optimum Dome and waveguide geometry. With the tweeter mounted directly on the baffle the off axis response shows irregularity. The version with the waveguide does not show this irregularity.

This is because the solid boundaries of the waveguide and baffle behave similarly to acoustical mirrors. The waveguide and dome are arranged so that this mirroring causes the dome to begin to behave similarly to a pulsating spherical source. A pulsating spherical source is the ideal radiator as it has a perfectly omnidirectional dispersion. When used in conjunction with the Tangerine Waveguide, the Uni-Q tweeter approaches this acoustical ideal. The result is a tweeter with a much wider dispersion and more consistent off axis behaviour.

Appendix IV. Stiffened dome

When a tweeter is designed for use in a Uni-Q driver array the shape of the dome must be set for optimum acoustic performance, as discussed in Appendix III. However, the acoustics is only half of the story; to achieve the highest possible level of performance the dome must be carefully constructed mechanically to remain as rigid as possible.

The tweeter dome is constructed from extremely thin aluminium. Aluminium is chosen as it has a remarkably high stiffness combined with very low density. This is important for the tweeter dome because for best performance the dome must move rigidly without deforming even at and above the highest frequencies that we can hear. The highest frequency that a human can hear is approximately 20,000Hz, this means that we can hear sounds which repeat themselves as fast as twenty thousand times a second. In order to move this quickly the tweeter undergoes extremely high levels of acceleration. At normal listening levels the tweeter hits peak acceleration levels of around 10,000m/s2. This is equivalent to 1000G of acceleration - more than 300 times that experienced during the launch of the Space Shuttle.

At such high levels of acceleration it is extremely difficult for the tweeter dome to remain rigid. The inertia due to the mass of the dome material itself can easily generate enough stress to deform the dome during normal use. The acceleration of the dome increases with frequency and ultimately there is a maximum frequency above which the dome is unable to remain rigid. The dome motion, including these deformations, is too small to see with the naked eye but it is possible to see them using a laser to record and amplify the motion as shown below.

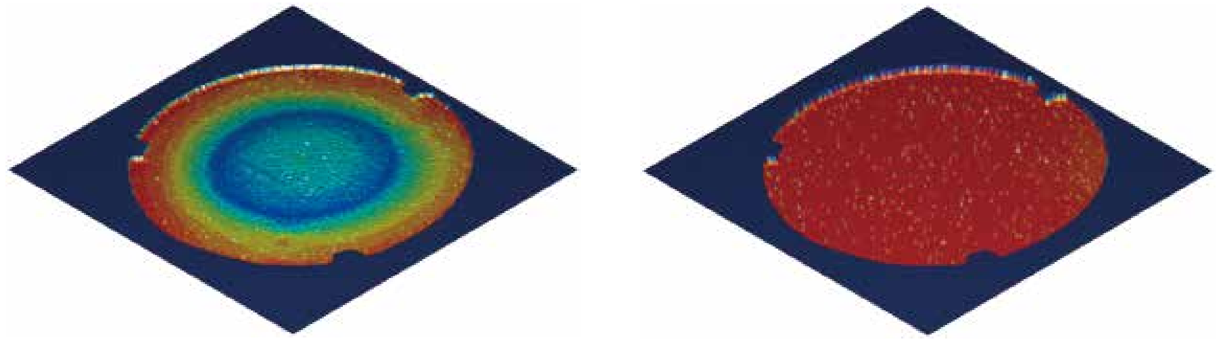

Figure 72. Laser vibrometer scan of a rigidly moving tweeter dome (right) and a flexing tweeter dome (left)

IV.I. Optimum Geometry

Research work was carried out at KEF in the 1990s to determine the dome shape that gave maximum resistance to the acceleration force and hence maximised the operational bandwidth. The study concluded that the optimum shape for the dome was an ellipse. In-fact, the optimum dome shape to resist the acceleration forces and remain rigid is quite close to the shape of the rounded end of a chicken egg. Using this shape it is possible to improve the bandwidth that a dome can be used over by around 75% compared to a conventional dome shape. Unfortunately, recent research work tells us that the optimum shape that we require for the acoustics of the dome is a spherical cap [20].

KEF research has resulted in two optimum shapes of dome for different uses:

1. The elliptical shape, optimum for the mechanics
2. The spherical cap, optimum for the acoustics

IV.II. The Solution

The Stiffened Dome geometry enables both of these optimum shapes to be used at the same time, resulting in the best possible mechanical and acoustical performance of dome. The Tweeter dome is made from two parts: one elliptical, one a spherical cap. These two shapes are superimposed, one placed on top of the other, forming the patented KEF Stiffened Dome.

At the edge of the dome the two shapes form a triangle. The triangle is a fundamentally strong shape and the edge of the dome is normally the weakest part. Triangles are widely used in many engineering structures because of their inherent strength. The Stiffened Dome gives a far higher performance than either the elliptical dome or the spherical cap shape alone.

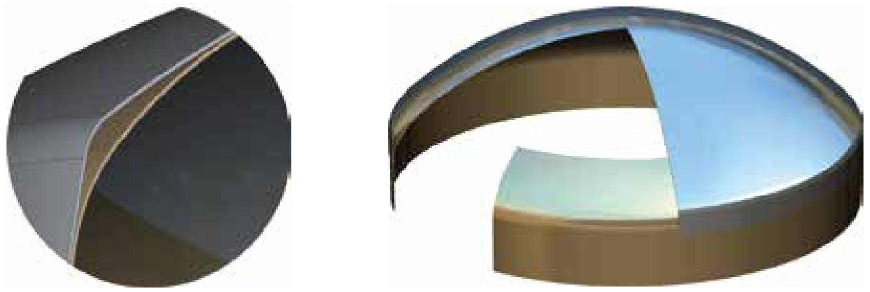

Figure 73. 3D CAD sectional views of the tweeter dome and extended former that meet to form a triangular stiffening member at the dome edge. This patented technology is the KEF stiffened dome.

Appendix V. Vented tweeter

The dome of a tweeter vibrates the air around it. In front of the dome this vibration is propagated away as sound. To the rear is an enclosed pocket of air. If the enclosure is too small the air undergoes large compressions and expansions. It will behave in a non-linear manner and cause distortion. In a larger enclosure the compressions and expansions are relatively small. The air in this case will act as a linear spring and the sound we hear will have much lower distortion.

The challenge is to design this enclosure in a way that avoids standing wave resonance problems. The KEF solution is to use a sealed narrow duct behind the dome. Inside this duct acoustical damping material is added in order to gently absorb the rearward wave from the tweeter dome. The quantity and type of the damping material is carefully fine tuned to give the best possible absorption of the rear wave.

Figure 74 shows the behaviour of the tweeter vent. Rear radiation from the dome travels down the duct and is gently absorbed in the acoustic damping material.

Figure 74. FEA simulation of the air in the rear of the tweeter.

Appendix VI. Z-flex surround

The surround is a critical component of any bass/midrange driver. The designer must carefully choose the material and shape so as to avoid irregularities in the midband response due to resonance in the surround whilst at the same time allowing sufficient excursion of the cone in order to reproduce bass frequencies. Most modern drivers use a half roll design typically moulded from butyl rubber. The half-roll design can often perform well if carefully designed, however, KEF's computer modelling techniques allow us to investigate some more adventurous possibilities. The Z-Flex surround is the result.

The Z-Flex Surround has a big impact on the high frequency performance of the Uni-Q Driver Array. Ideally, the Uni-Q tweeter would be in the throat of a perfectly smooth waveguide. Under these circumstances, provided the dome is correctly shaped, the driver will give smooth hemispherical "point-source" radiation and higher sensitivity than a tweeter in a flat baffle. Previous Uni-Q designs have used conventional half roll surrounds but with this arrangement the ideal situation is compromised. The waveguide created by the cone and surround is not perfectly smooth: it has an abrupt discontinuity at the surround. This discontinuity causes diffraction and secondary radiation which smears the sound from the tweeter.

When the Z-Flex surround is mounted in the cabinet and correctly trimmed it does not form a discontinuity as a conventional surround does. In-fact, it creates a close approximation to the ideal smooth waveguide. The performance of the waveguide is extremely good and very little diffraction is generated.

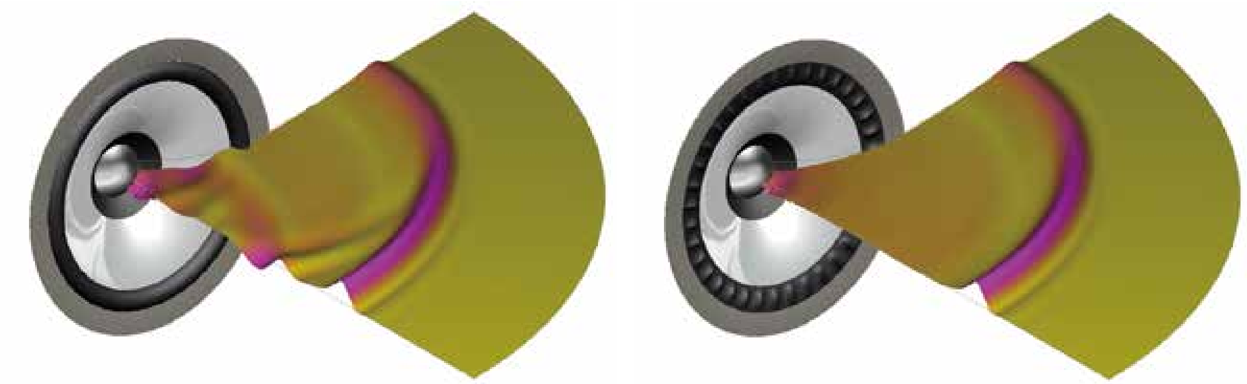

Figure 75. FEA simulated tweeter impulse response with conventional surround (left) and with Z-flex surround (right).

Appendix VII. Room modes and loudspeaker positioning

After the loudspeaker, the audio signal goes through one more process before reaching the ears: propagating across the room. This can have a significant effect on the overall performance of the loudspeaker, especially at low frequencies where room modes can play a prominent role in the clarity and evenness of the sound. No two rooms are identical, and the effects are difficult to predict. However, these low frequency problems can usually be significantly reduced through careful positioning of the loudspeaker and listener.

VII.I. What is a room mode?

Room modes occur at frequencies where the wavelength of the sound coincides with a dimension of a room, resulting in a standing wave resonance. Longer dimensions have lower frequency room modes than shorter ones. They are typically problematic below 200Hz, and occur in all 3 dimensions of the room. The standing waves formed in rooms are acoustically similar to those formed in the loudspeaker cabinet, as explained in Section 4.3.1.

Figure 76 shows the pressure of a room mode along a length of a room. The strength of the mode varies in a sinusoidal shape, being stronger at the walls due to the build-up of sound pressure. There is a node in the centre of the room where the pressure of the mode is zero. The pressures either side of the node are opposite phases, such that the total pressure in the mode remains zero.

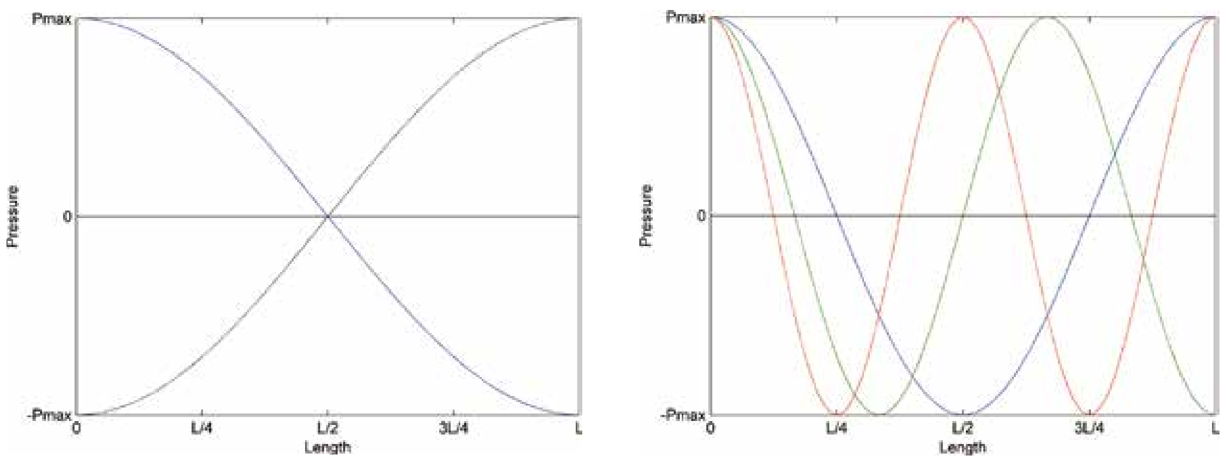

Figure 76. Modal pressure distributions between parallel walls for first (left) and second, third and forth modes (right).

Modes will also occur at integer multiples of the original resonant frequency (e.g. 20Hz, 40Hz, 60Hz, 80Hz…). At these frequencies an additional half sine wave can be fitted into the length of the room. The first mode will be half a sine wave (180 degrees), the second a whole sine wave (360 degrees), the third one and a half (540 degrees) and so on. Figure 76 shows the next 3 room modes, fitting between one and two wavelengths along the length. As can be seen the positions of the nodes and anti-nodes varies with the different resonances. When the loudspeaker and listener positions are moved around the room, the effect of the room modes will change depending on their proximity to a pressure maxima or minima.

The modes described so far are axial room modes, involving standing waves formed between a pair of parallel opposite walls. These are the strongest modal resonances in a room. If the room dimensions are integer multiples of each other, the modal resonances of each dimension will occur at the same frequency. This gives a low modal density with fewer but stronger modal resonances, which is more audible than a greater number of weaker resonances. The stronger resonances are harder to control, which is why shapes such as cubes are avoided when designing listening rooms.

Modes will also form between two pairs of walls such as the length and width, called a tangential mode. Similarly, modes can form across all three dimensions, called an oblique mode. Tangential and oblique modes are weaker than axial modes, but can reinforce resonances if they coincide in frequency. If room modes are strong in a room, then they can be damped using acoustic treatment, although large quantities are usually needed due to the long wavelengths at low frequencies.

VII.II. The effect of modes on loudspeaker responses

When a loudspeaker is placed in a room, if it is near to a pressure maxima of a mode (the point at which the magnitude of the pressure is at its largest value) it will excite it strongly. If the loudspeaker is placed near to a modal node (the point where the pressure is zero), then the mode will be excited much less. Excited modes result in resonant peaks in the response coinciding with the modal frequencies, and ringing in the time domain. Between the peaks nulls are often seen in the frequency response and occur due to destructive interference between adjacent room resonances. The depth of these nulls is greatly dependent upon the speaker position. In many ways these nulls can be more audible than the peaks as they tend to result in a perceived lack of bass.

Figure 77 shows a real in-room measurement. On the left of the figure a standard frequency response plot is shown, on the right a time frequency representation of the same data. Comparing the two, it can be seen that the peaks in the frequency response coincide with time domain ringing. Similarly, the dips in the frequency response result in dips in the time response. Note that the decay time associated with these irregularities is far greater than any of the other time smearing discussed in the body of this paper. These measurements are of a worst case scenario with the loudspeaker in the corner, where it excites all the room modes

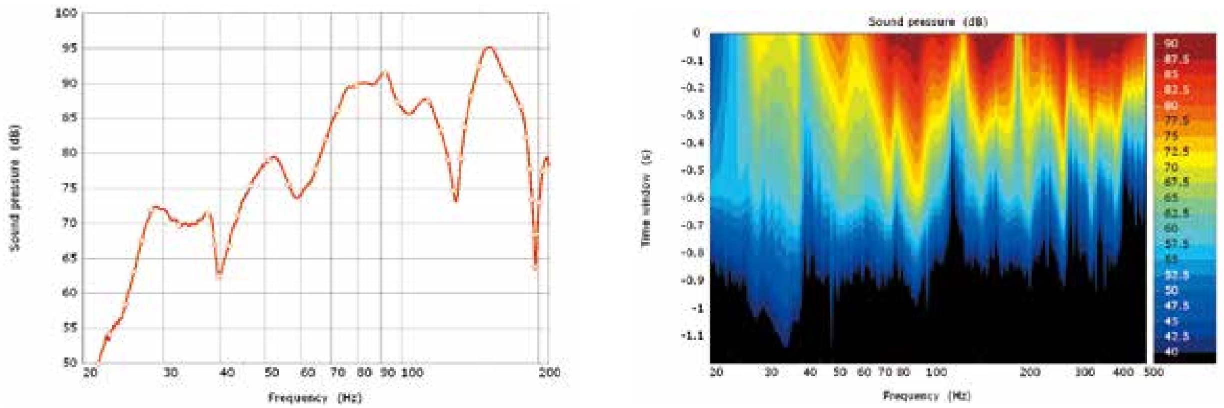

Figure 77. In-room loudspeaker response shown as frequency response chart (left) and short time Fourier response (right).

VII.III. Positioning

The effects of room modes can never truly be avoided. They can however be significantly reduced through careful positioning of the loudspeakers in the room. Small changes to the loudspeaker position can have a very large effect. For example figure 78 shows the response of a floor-standing loudspeaker measured from the listening position for two different positions in the room. The two positions are only 0.5m apart from one another yet the change in the response is as much as 7dB. The sound of the loudspeaker would be very different in these two locations. It is tricky to provide general advice that will work in all rooms, though there are some tips on speaker placement in appendix VIII. The best approach is to experiment with different loudspeaker and listener locations to find a layout with a balanced overall sound. In some rooms, changing the speaker location by as little as 10cm can make a big difference.

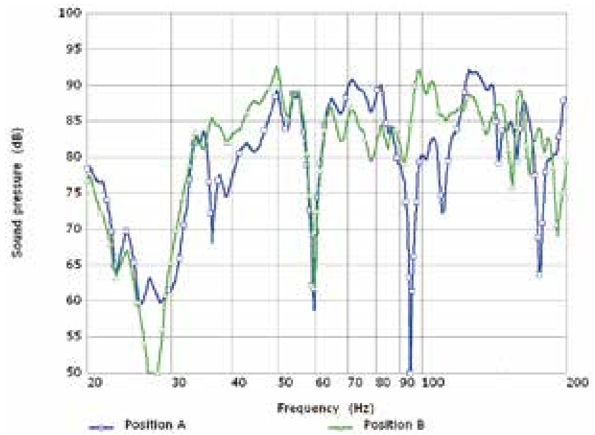

Figure 78. In-room loudspeaker measurements for two different loudspeaker positions 0.5m apart.

Appendix VIII. General loudspeaker positioning tips

- The speakers should be placed between 2 and 4m apart and the listener should be positioned exactly equidistant from the loudspeakers, and at approximately the same distance the speakers are apart.

- Ideally a wall should be directly behind the loudspeakers (the “front” wall). Both speakers should be the same distance from this wall.

- The distance between the speakers and the front wall is normally the most sensitive parameter for low frequency tuning. It is recommended that the listener fine-tune the loudspeaker to rear wall spacing to find the optimal distance. Normally this distance would be between 0.3m and 1.5m to the rear of the loudspeakers.

- Avoid placing the listening position in close proximity to any of the walls.

- Symmetry is very important for optimal stereo. Ideally the side walls, to the left and the right of the loudspeakers, should be at the same distance and of the same construction.

- The side wall has only a small effect on the timbre of a well designed loudspeaker provided it is not closer than around 1m. Ideally the side wall should be between 1 and 3m from the loudspeaker.

- If acoustically treating the room, it can be very helpful to add diffusion (uneven reflective surfaces) to the side walls. This helps to reduce timbral imbalance due to the side walls while maintaining a good stereo image.

- Toe-in can help to fine tune the stereo image and the perceived high frequency energy. Maximum high frequency energy will reach the listener when the tweeters are pointing directly at the listening position. With Uni-Q the balance does not suffer if you listen off axis, so using a flatter setup with less toe can be a very good option for lively rooms.

- Use a tape measure to ensure your spacings are exactly the same, the ear is very sensitive to arrival time.

- If it is not working, consider a complete change of approach. By using a different wall behind the loudspeakers you will most likely end up with a different listening position too. The listening position is just as important as the speaker positions.

- Bass traps can help a great deal to tame room mode resonances which will tend to improve bass overhang. However, they need to be physically large to have any effect on deep bass. Beware of poorly designed traps. A typical bass trap to effect frequencies below 80Hz needs to have a volume of approximately 0.5m3.

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