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.

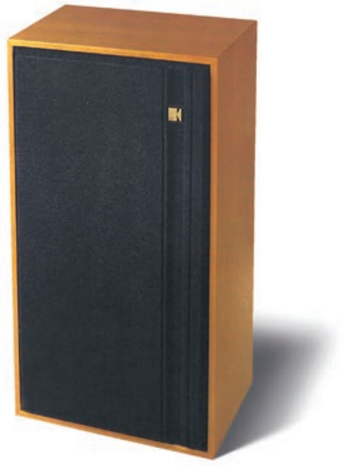
Because KEF wished to balance a number of client needs – the BBC's technical criteria, burgeoning audiophile tendencies, commercial realities and other considerations – a conscious decision was made to improve the performance of domestic loudspeakers with the aid of computers, while respecting various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

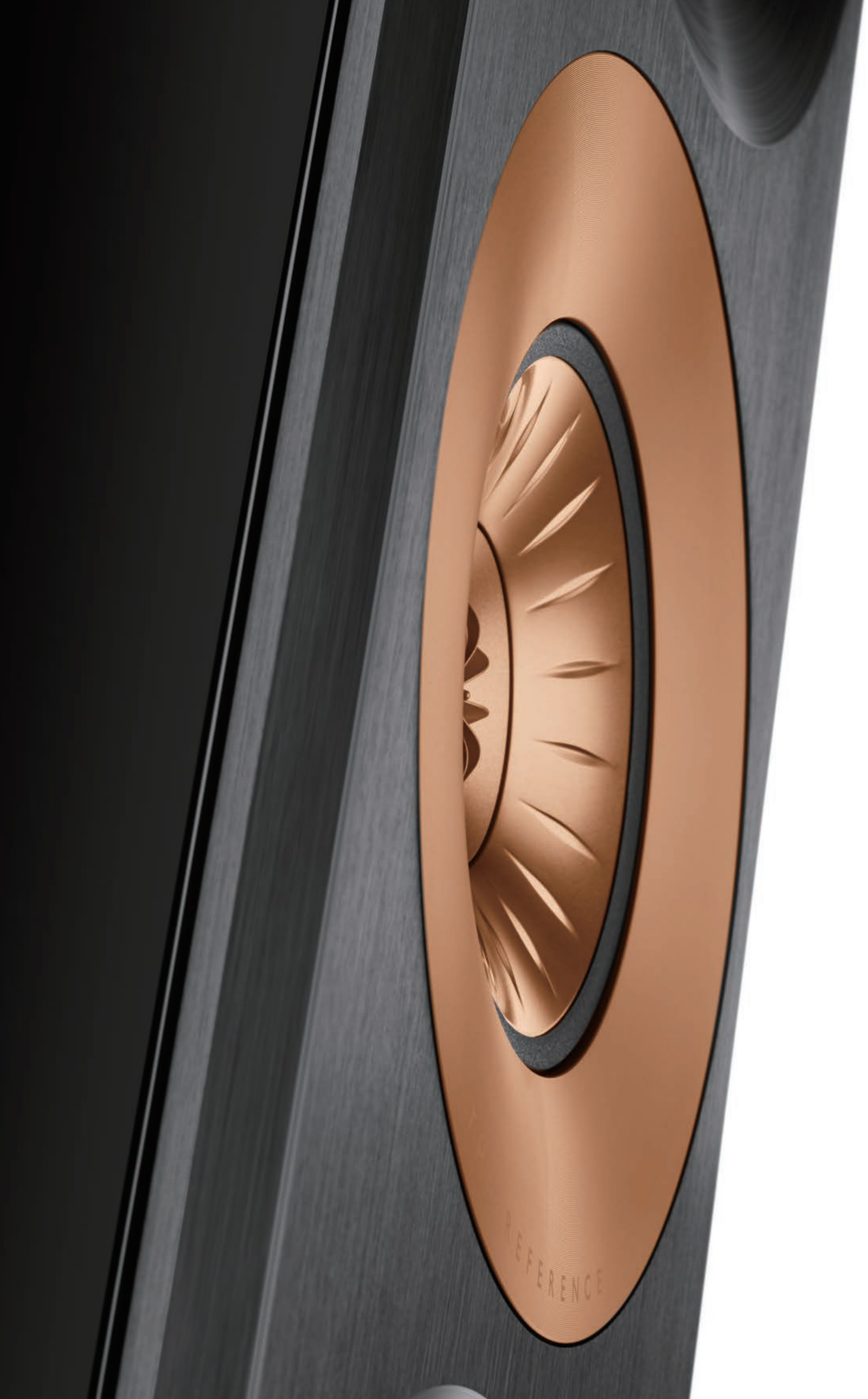
Controlling the loudspeakers' directivity is also key to avoiding loss of midrange and treble fidelity, which can happen when loudspeakers are placed in a real listening environment. One of the features of our auditory perception is that we are well used to hearing sounds that include reflections off 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.

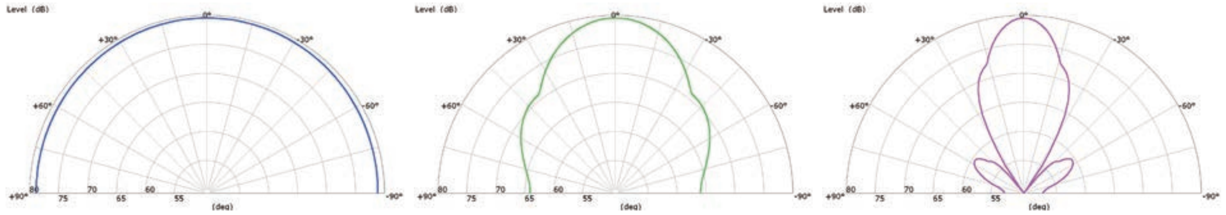


Figure 1. Polar dispersion of a 160mm ideal driver, with rigid flat piston mounted in infinite baffle, at 500Hz, 3kHz and 6kHz.

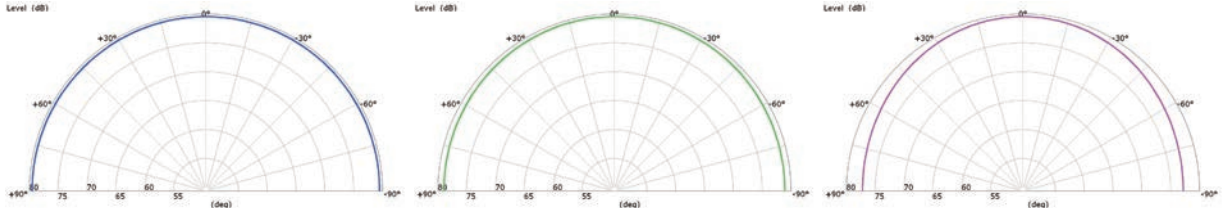


Figure 2. Polar dispersion of a 25mm ideal driver, with rigid flat piston mounted in infinite baffle, at 500Hz, 3kHz and 6kHz.

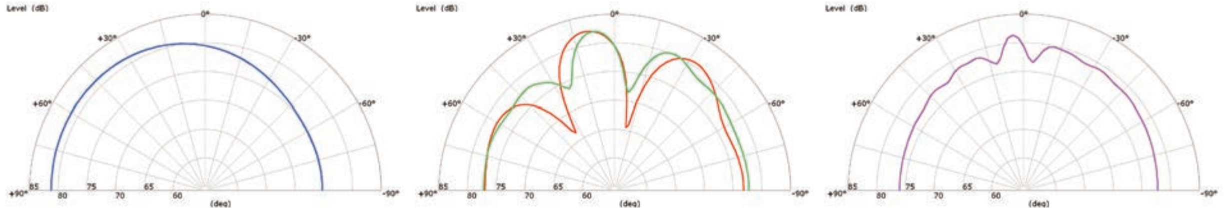


Figure 3. Polar dispersion of theoretical system with ideal 25mm driver 200mm to the right of an ideal 160mm driver with 1st order Butterworth crossover at 2kHz, polar plots at 500Hz 3kHz (green) 2kHz (red) and 6kHz.

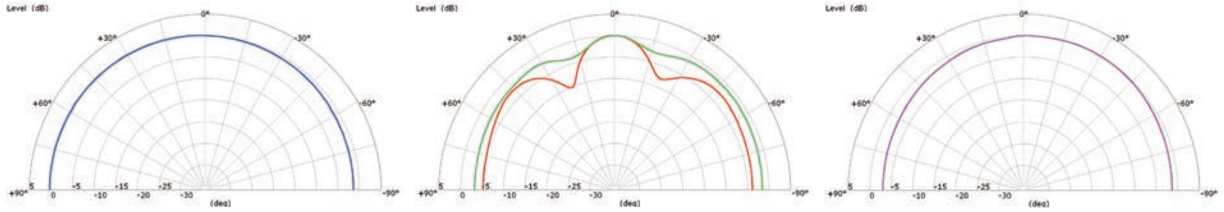


Figure 4. Polar dispersion of theoretical system with ideal 25mm driver 200mm to the right of an ideal 160mm driver with 4th order Linkwitz Riley crossover at 2kHz, polar plots at 500Hz 3kHz (green) 2kHz (red) and 6kHz.

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.

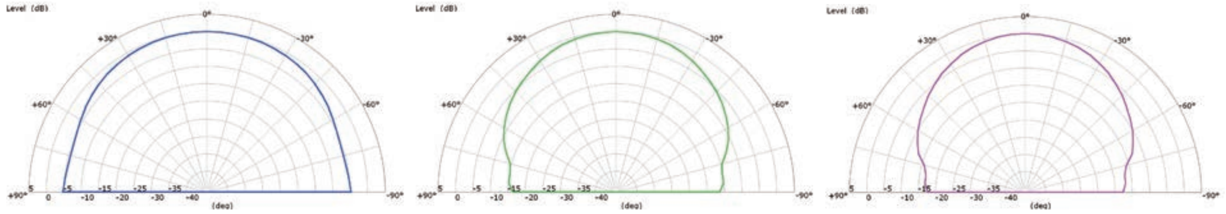


Figure 5. Horizontal polar dispersion measurements of prototype Reference 5 system with crossover, polar plots at 500Hz 3kHz and 6kHz.

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[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 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.

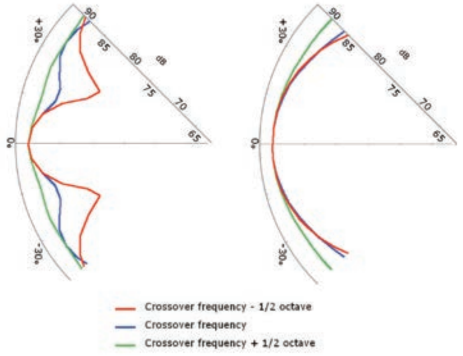

Figure 6. D'Appolito polar response comparison, left a conventional arrangement using a tweeter and two midrange drivers, right with a Uni-Q driver and four low frequency drivers as on Reference 5.

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.

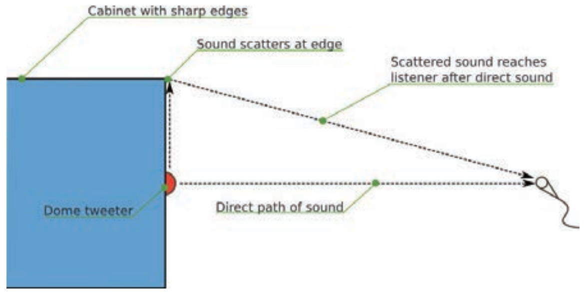

Figure 7. Diagrammatic explanation of diffraction effect.

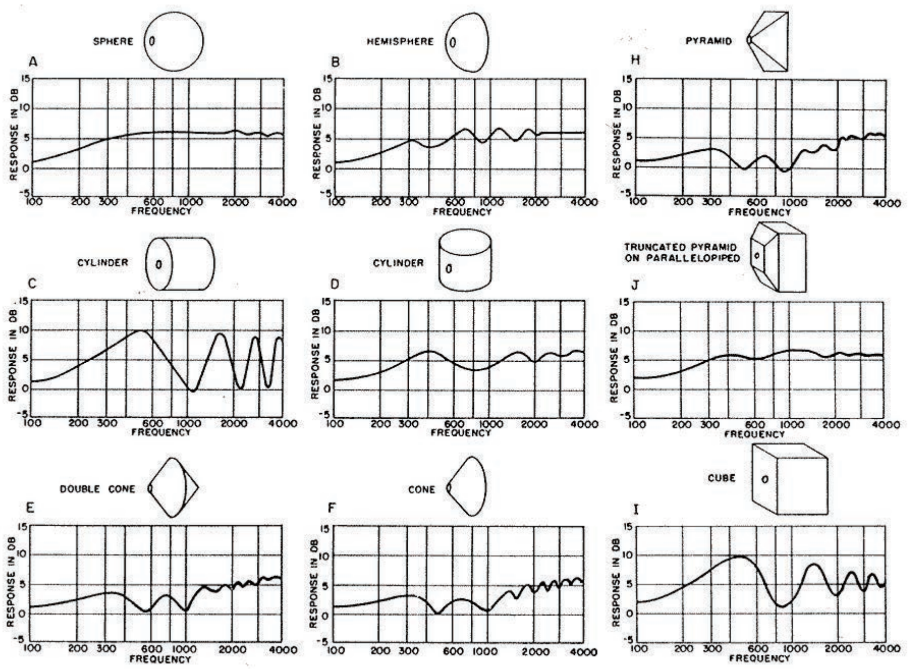

Figure 8. Reproduction of Olsen's classic loudspeaker enclosure baffle-diffraction experiment.

Olsen was the first to look at this diffraction effect 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,

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[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.

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.

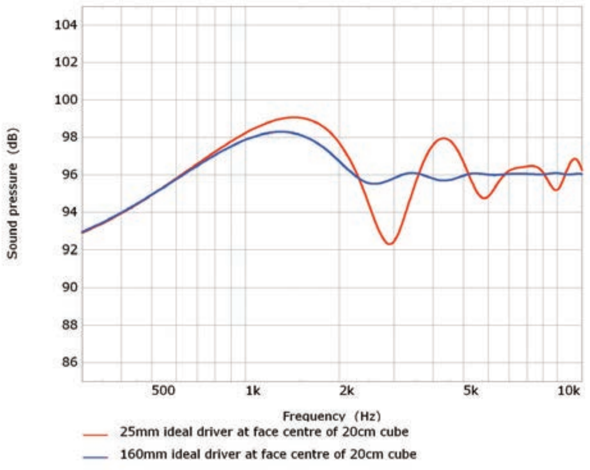

Figure 9. Comparison of the diffraction effect with radiators of different size placed in the centre of a cubic enclosure

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.

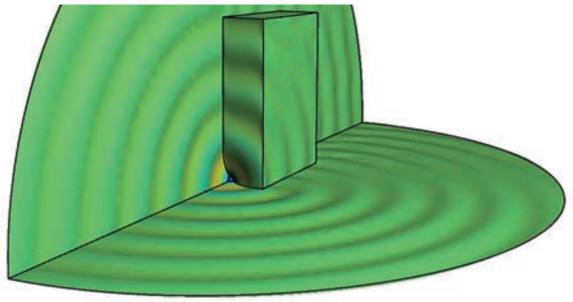

Figure 10. FEA modelled acoustic pressure on, and around, cabinet surface due to MF driver output.

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.

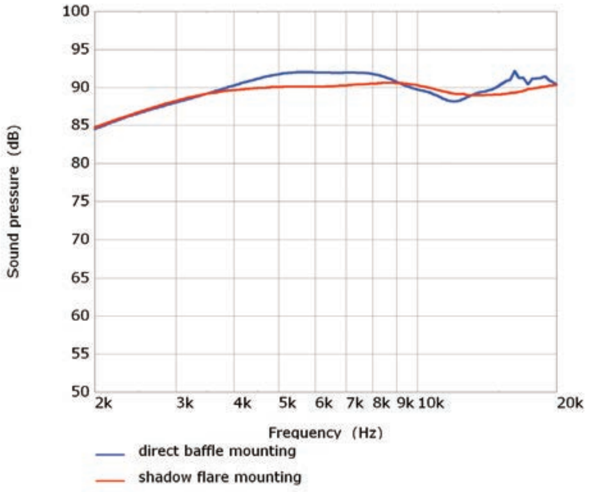

Figure 11. Uni-Q driver HF response with and without shadow flare to control the diffraction effect.

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.

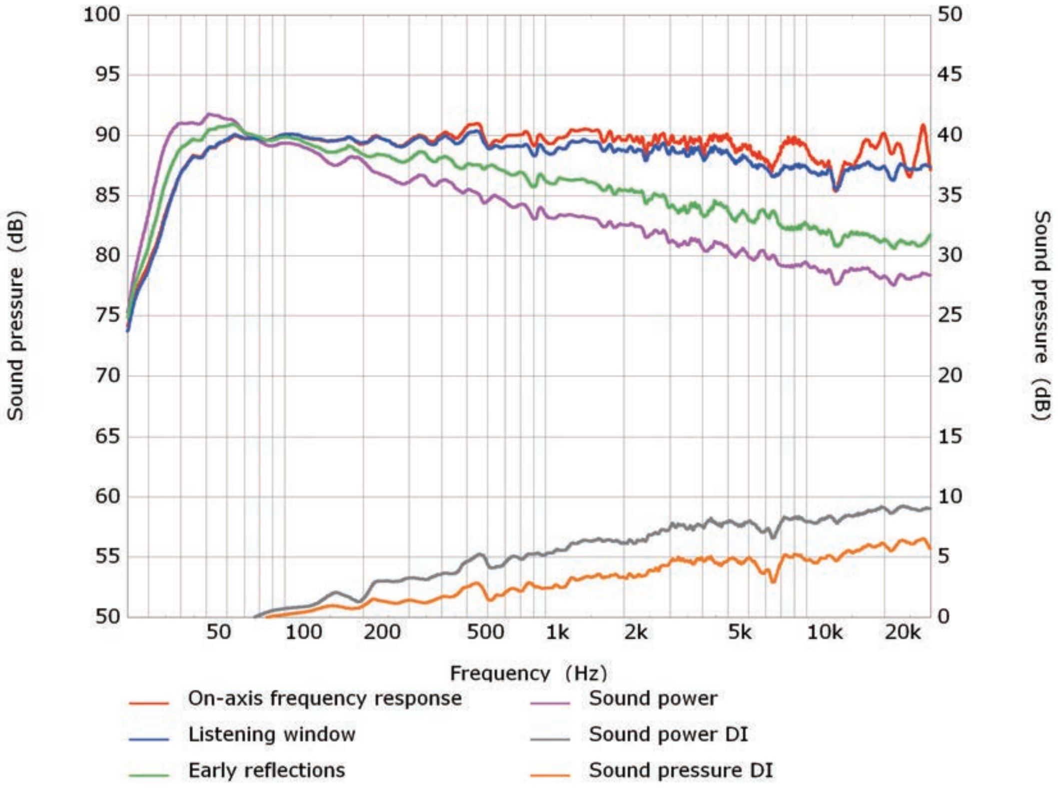

Figure 12. Family of directional response curves for early Reference 5 prototype.

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.

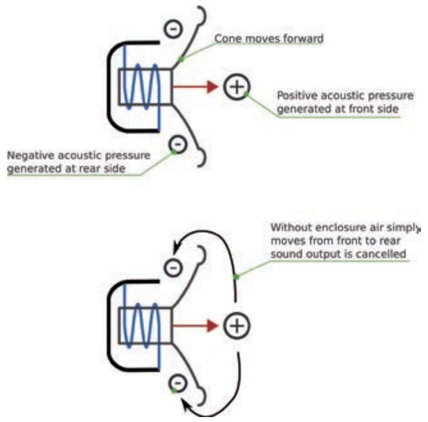

Figure 13. Family of directional response curves for early Reference 5 prototype.

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.

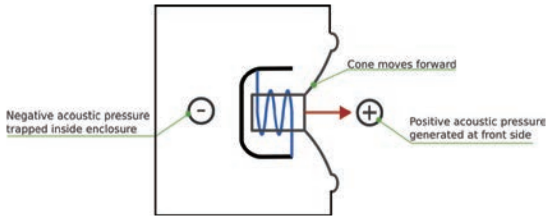

Figure 14. Illustration of a closed box loudspeaker.

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).

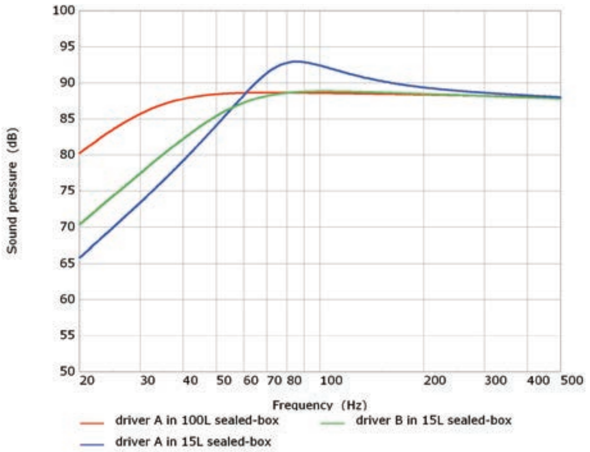

Figure 15. Variation in frequency response of a 160mm bass driver as the rear enclosure volume is changed.

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[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.

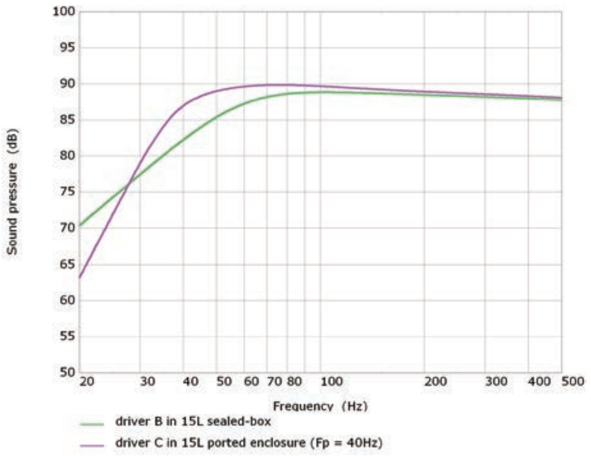

Figure 16. Comparison of two 15L loudspeaker systems each with a single 160mm LF driver and using a sealed and ported box design.

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.

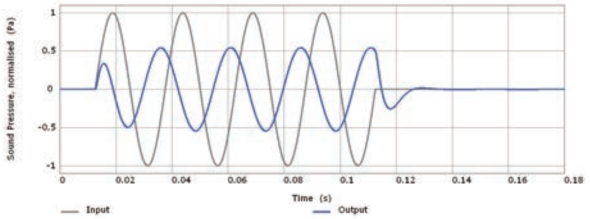

Figure 17. Response of closed box loudspeaker shown in Figure 16 to a 40Hz toneburst.

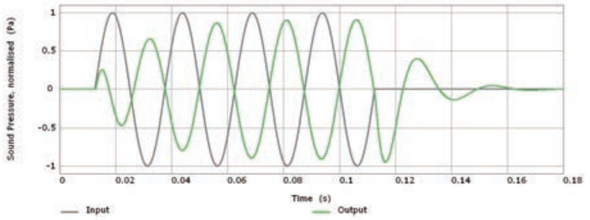

Figure 18. Response of ported loudspeaker shown in Figure 16 to a 40Hz toneburst

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

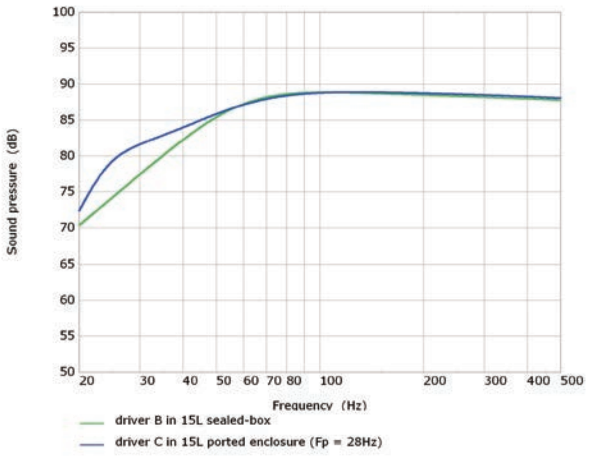

Figure 19. Comparison of two 15L loudspeaker systems each with a single 160mm LF driver and using a sealed and ported box design.

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[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 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.

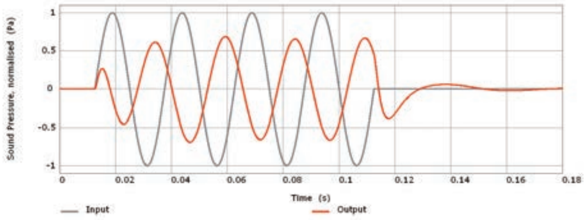

Figure 20. Response of ported loudspeaker shown in Figure 19 to a 40Hz toneburst.

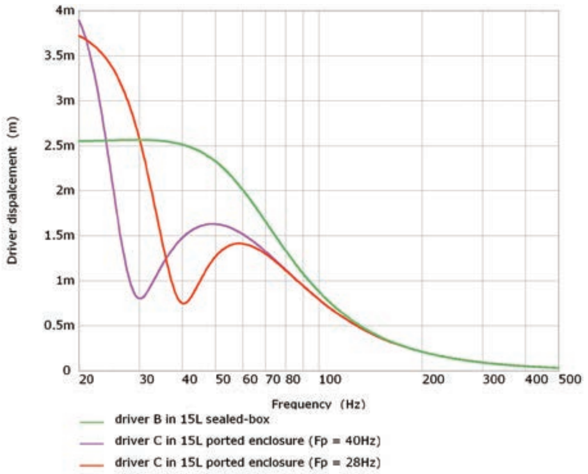

Figure 21. Lumped circuit representation of a generic electromagnetic loudspeaker.

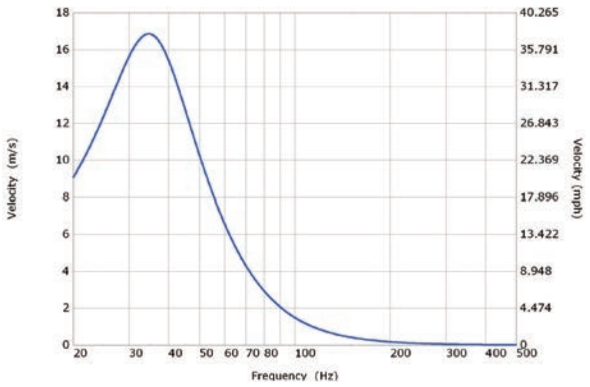

Figure 22. Peak port air velocity for a theoretical 15L speaker at an input level of 10V rms (approx 25Watts).

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.

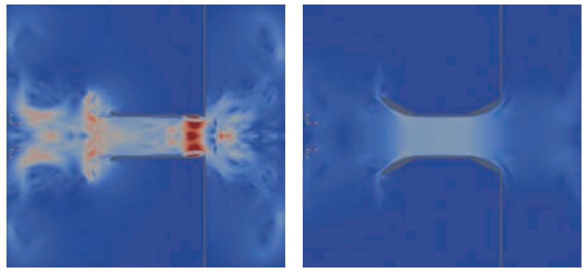

Figure 23. Instantaneous flow pressure contour of straight tube port (left) compared to optimised port (right) computed using CFD, note that both ports are shown on the same colour scale and at the same instant. For this analysis the peak velocity is 10ms.

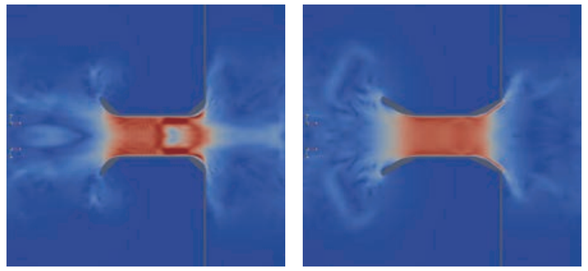

Figure 24. Instantaneous flow pressure contour of unoptimised port (left) compared to optimised port (right) computed using CFD, note that both ports are shown on the same colour scale and at the same instant. For this analysis the peak velocity is 15ms.

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.

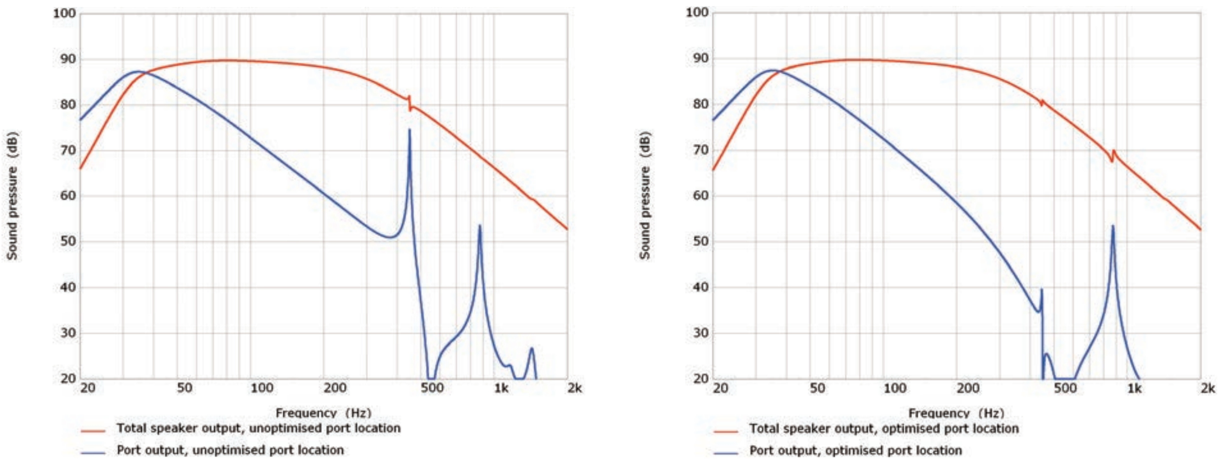

Figure 25. Simulated comparison of the loudspeaker output and port output with unoptimised port location (left) and optimised port location (right).

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.

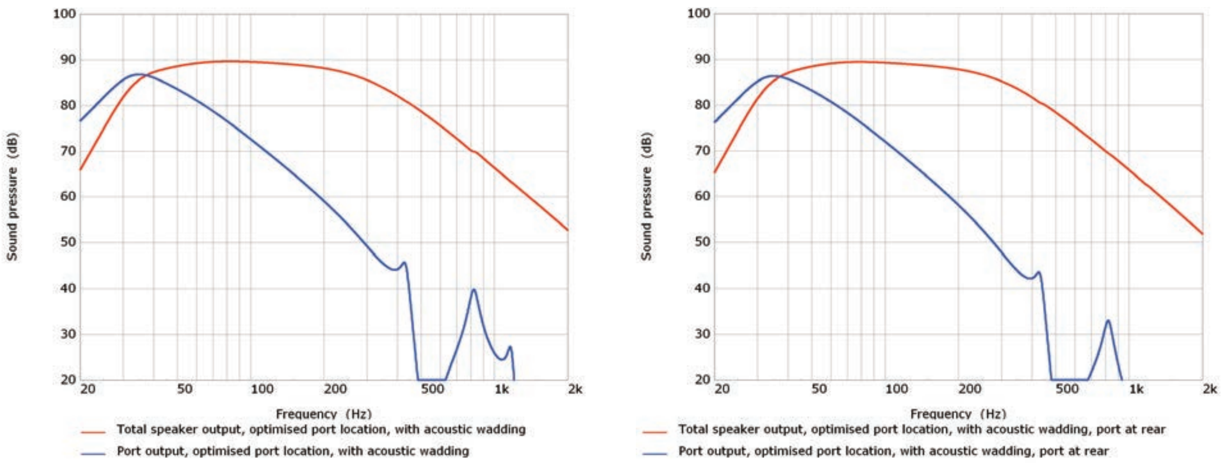

Figure 26. Simulated comparison of the loudspeaker output and port output with optimised port location and enclosure acoustic wadding, port at front (left) and port at rear (right).

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.

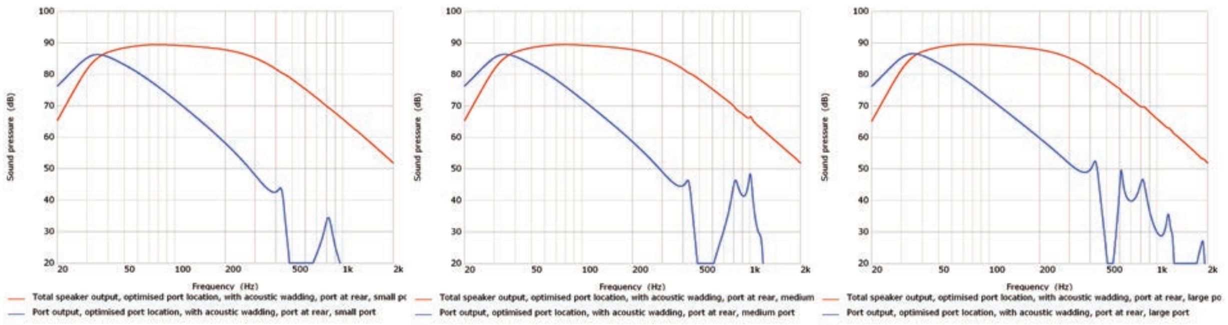

Figure 27. Simulated comparison of the loudspeaker output and port output with optimised port location and enclosure acoustic wadding, port at rear and small (left) medium (middle) and large (right) port.

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.

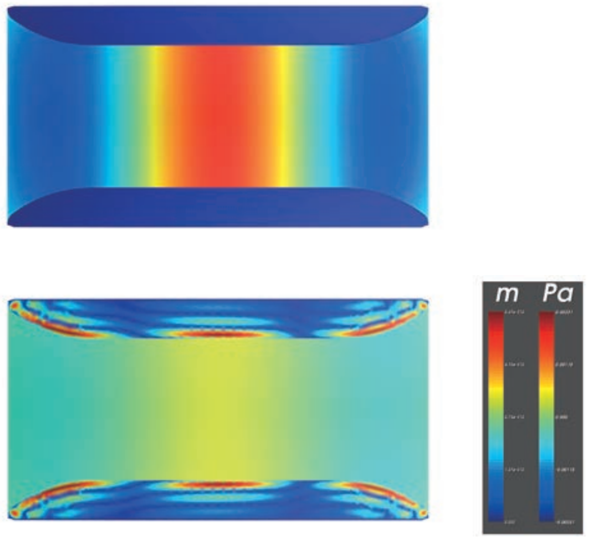

Figure 28. Comparison of the FEA modelled pressure magnitude in the loudspeaker port at the first standing wave with rigid walls (above) and flexible walls (below).

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.

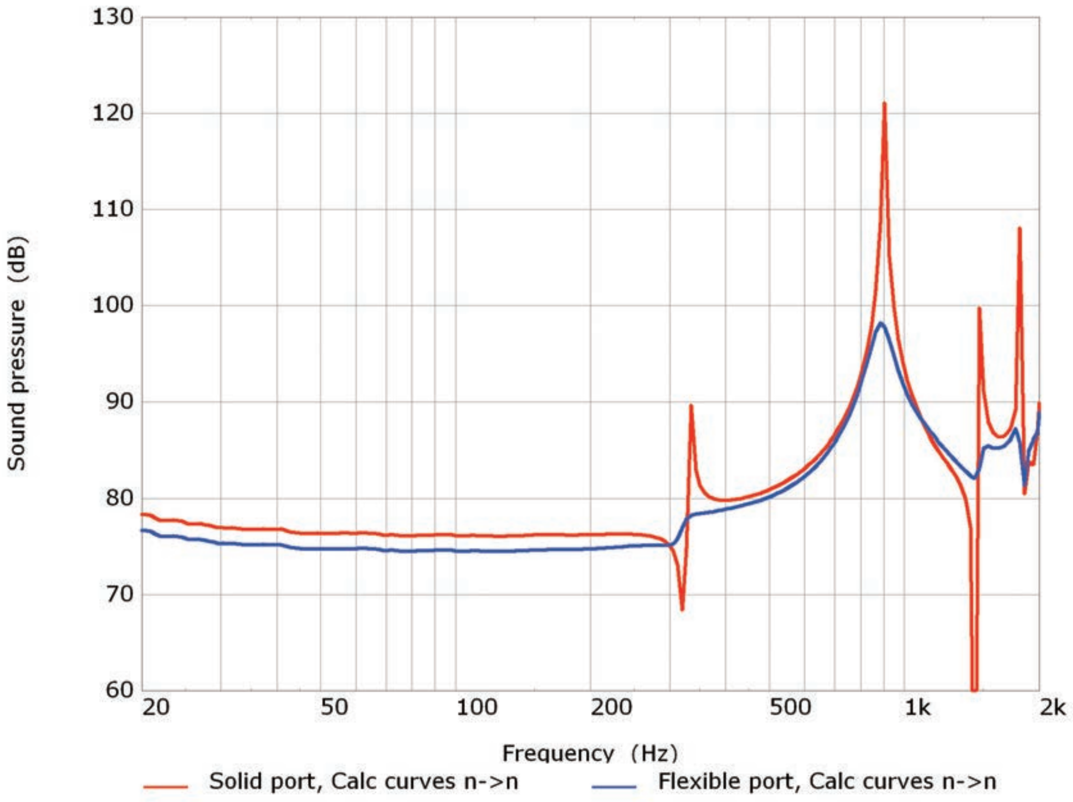

Figure 29. Comparison of the FEA modelled internal acoustic pressure in a solid and flexible port

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.

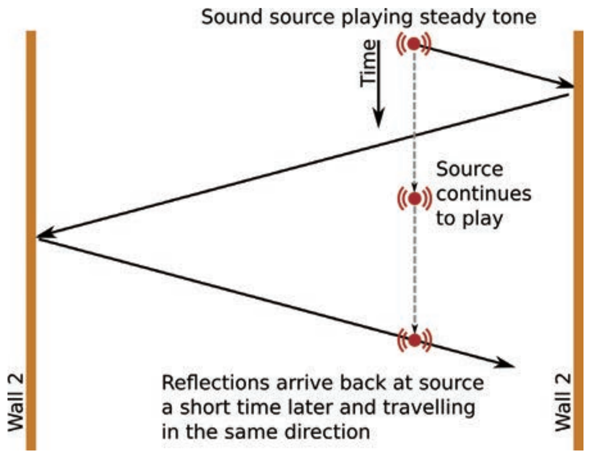

Figure 30. Illustration of standing wave resonance mechanism.

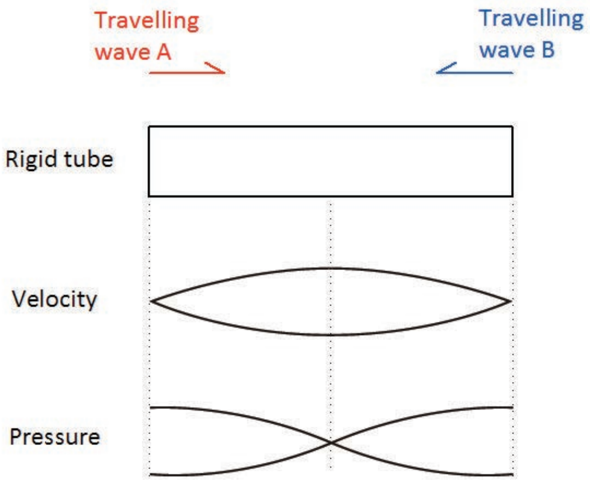

Figure 31. First standing wave pattern (or mode shape) between two parallel rigid walls.

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.

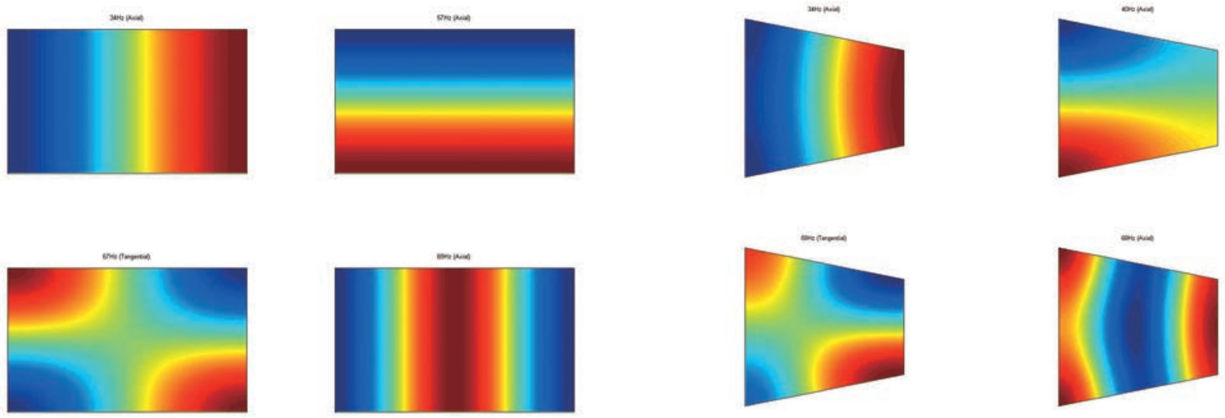

Figure 32. First four standing waves of a rectangular enclosure (left) compared to a trapezoidal enclosure (right)

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

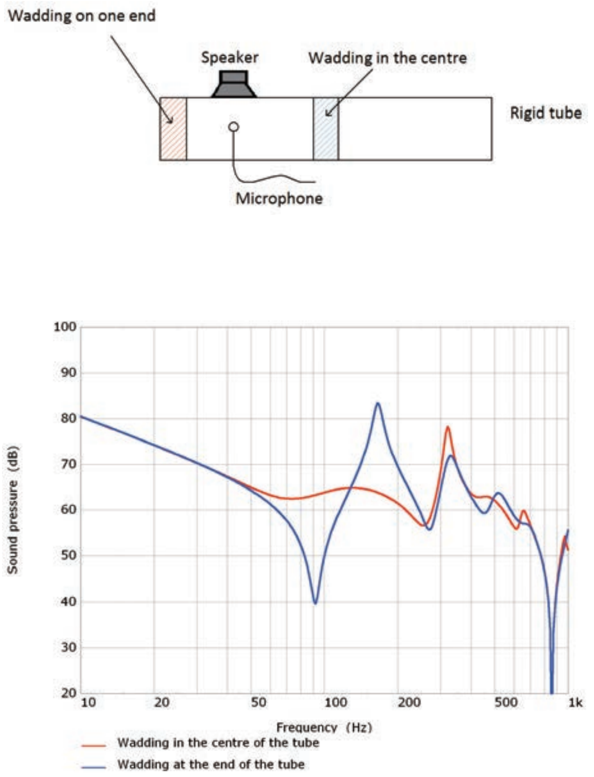

Figure 33. The effect of wadding placement on the attenuation of standing waves.

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