KEF KEFTOPICS

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A Target Function approach to the design of filters

In 1971 KEF began experimenting with computer aided digital techniques for evaluating loudspeakers.

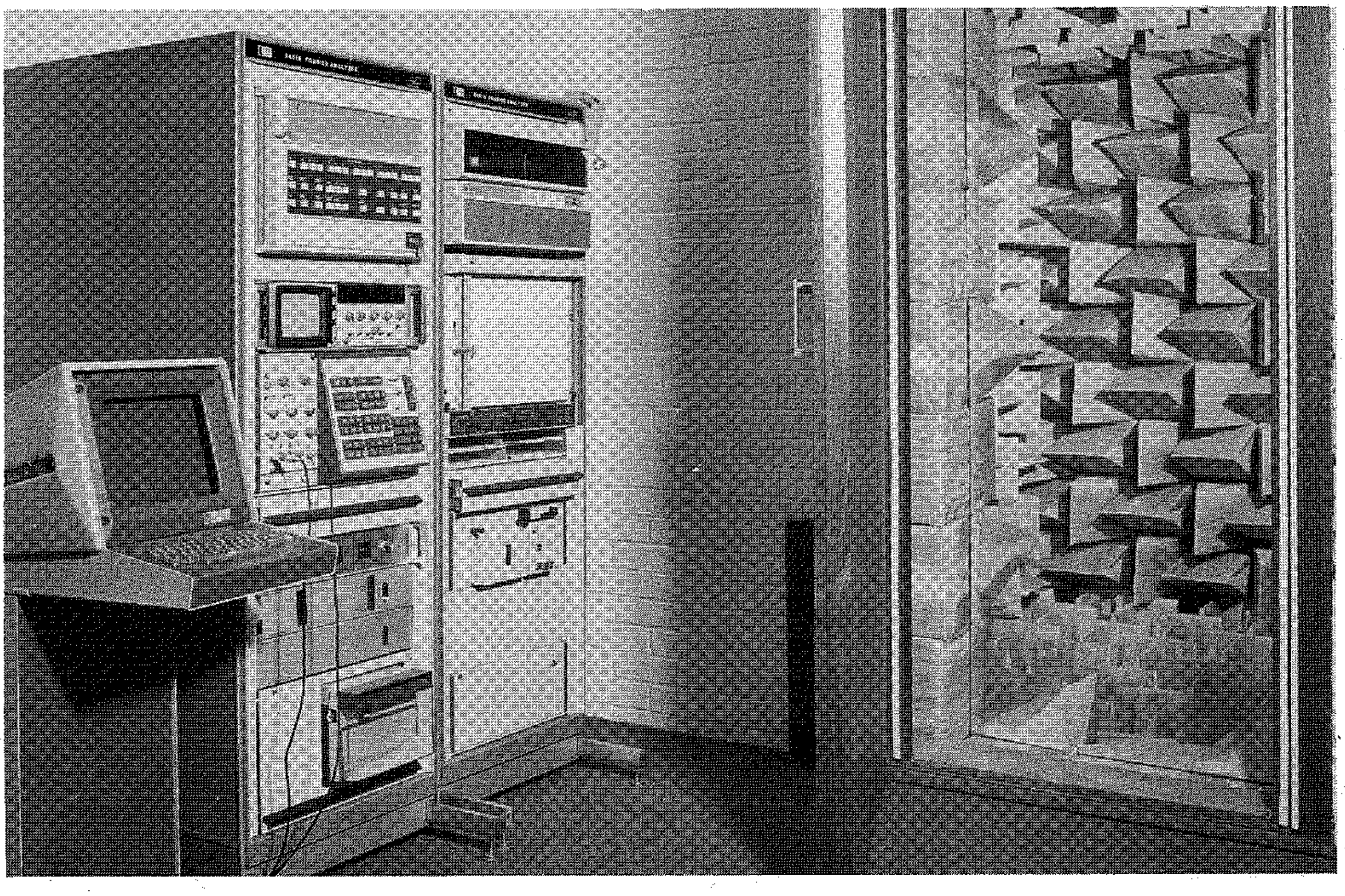
Since digital computation had become an accepted tool in almost all design studies, why shouldn't this very exact technique be applied to the design of high quality loudspeakers?

Obviously, the end product of any sophisticated and technical loudspeaker design has to be sound quality that is subjectively acceptable to the ear.

However, KEF felt that the process of assembling each and every component of a loudspeaker system could be followed objectively and with an end target response in mind. Provided this target response was sensible in practical acoustical terms, it would be a worthy and wholly justifiable project.

Essentially, a loudspeaker system should possess a flat amplitude — frequency response, such that it will introduce no colouration into the programme it is reproducing; it should be able to reproduce at realistic sound levels; and it should be carefully phase compensated to avoid any inter unit time delays.

This paper indicates how the target approach was applied to the design of filters that act as dividing networks in loudspeaker systems.



Why do loudspeakers need filters?

Until the late 1940's, attempts to improve the quality of loudspeakers centred on the production of single drive units which would be capable of reproducing the entire audio frequency spectrum.

These attempts were continually foiled by problems of directivity and phase distortions, so eventually, attention was turned to loudspeaker systems with more than one drive unit.

Since then, drive units have been developed to reproduce certain predetermined bands in the spectrum. This, however, raised an immediate problem: given a single input voltage from the amplifier, containing frequencies over the whole spectrum, how would the appropriate voltages and frequencies be allocated to their drive units?

This fundamental problem could only be solved by filters.

The complications.

The first few attempts were simple high and low pass filters which treated the drive units as fixed resistance devices. At certain frequencies, these filters caused the input voltage to decay for one drive unit and simultaneously to build up for another. These are called crossover points.

However, drive units are not fixed resistance devices. Their impedance varies with frequency and therefore so does the amplitude of the drive units reaction to the input voltage.

This effect immediately introduces an imbalance in the outputs of the drive units. The loudspeaker system will therefore not have a flat amplitude — frequency response.

Ideally, the filter sections in the loudspeaker system should modify the input voltage to the corresponding drive units so that at some chosen listening position, the outputs from all the drive units will add together to give a flat amplitude — frequency response.

This will only occur if the pressure outputs have both the correct amplitude and relative phase, particularly in the crossover regions.

KEF's approach.

Clearly, to design the pressure outputs correctly, one needs to know as much as possible about all components of the system before attempting to combine them.

The search for more detailed accurate information forms the backbone of KEF's target design approach.

The critical information required is the amplitude — frequency and phase — frequency response of the drive units and their associated filter sections.

Measuring amplitude response has been fairly straightforward for many years, but measuring phase response, particularly for drive units has not been easy.

The main problem has been the time delay between the reaction of the drive unit and the collection of the resultant information.

KEF's method, however, overcame these problems by measuring the drive unit's impulse response.

A short, square-wave burst of energy containing frequencies over the entire audio spectrum is fired at the drive unit. Since the signal is so short, this experiment need not be conducted under anechoic conditions because the drive unit has reacted and settled down again long before any echoes (which would give misleading information) arrive at the measuring position.

The loudspeaker's response to this tone burst is collected by a microphone and stored for analysis. By taking many readings and then averaging, both the experimental and observational errors can be minimised.

The analysis requires a digital computer and a sophisticated mathematical technique called Fourier Analysis. Suffice it to say, by this technique, both the amplitude and phase response can be computed from the impulse response.

While all of this may appear somewhat theoretical, it is worth noting a significant point. The impulse response of a drive unit, or the corresponding amplitude and phase response are complete descriptions of the linear behaviour of that drive unit.

Knowing this behaviour means knowing how the drive unit will respond to any signal – be it steady state or transient in nature. Clearly this sort of knowledge is vital.

Next to be investigated was the relationship between phase and amplitude response. Ideal drive units should have a minimum phase shift response associated with a particular amplitude response, as this will introduce the least amount of relative time delay over the frequency range covered.

The results of this research were surprising. KEF current production drive units proved to be essentially minimum phase shift devices.

This is most fortunate, since in a minimum phase shift device, the amplitude and phase characteristics are uniquely related in a way that is acceptable to the human ear.

In terms of the design of filter networks this means that the drive unit can be fully described in terms of its overall amplitude response.

The Target Function.

KEF define the target function as the desired amplitude – frequency response of a particular minimum phase shift drive unit, combined with its minimum phase shift filter section.

If S(f) is the frequency response of the drive unit in its mounting without any network; T(f) is the target function, and H(f) is the filter frequency response, then:

$$T(f) = H(f).\,S(f) \quad \text{or} \quad H(f) = \frac{T(f)}{S(f)}$$

Since the amplitude response S(f) which describes the complete linear behaviour of the drive unit, is known, and T(f) is defined, the problem is now to synthesise a filter section with a response H(f).

Other problems.

Apart from the variation of resistance of a drive unit with frequency which has already been mentioned, there are other considerations.

Each drive unit has its own fundamental resonance frequency at which the impedance of the voice coil suddenly and dramatically changes.

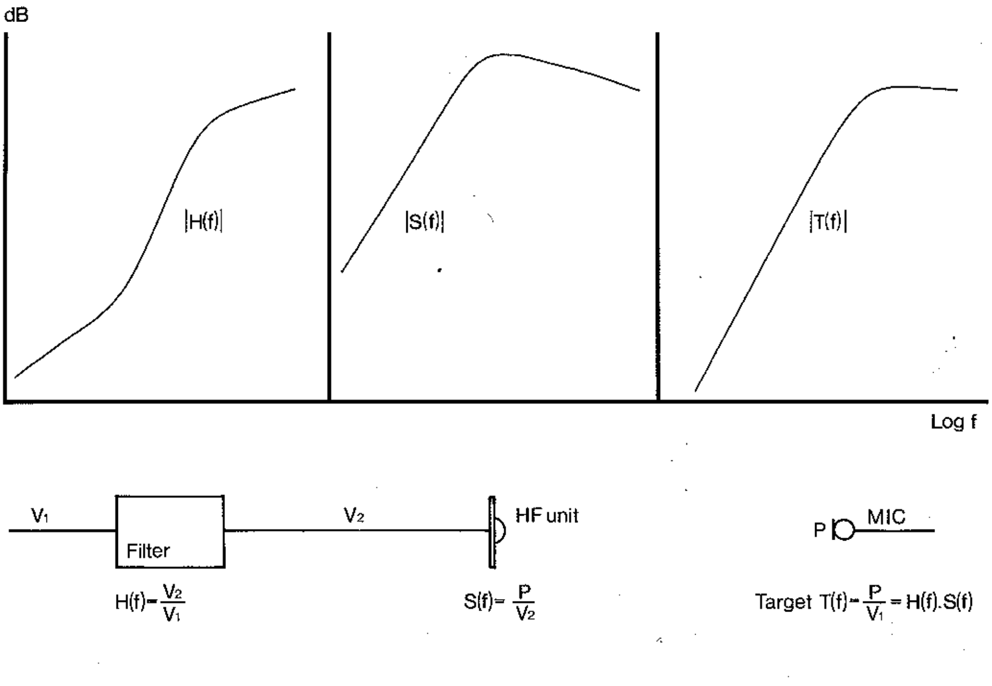
At certain frequencies, the voice coil inductance can resonate with electrical components in the filter network.

The slope of the filter response in the crossover regions should be smooth, since any irregularities will introduce colouration into the sound.

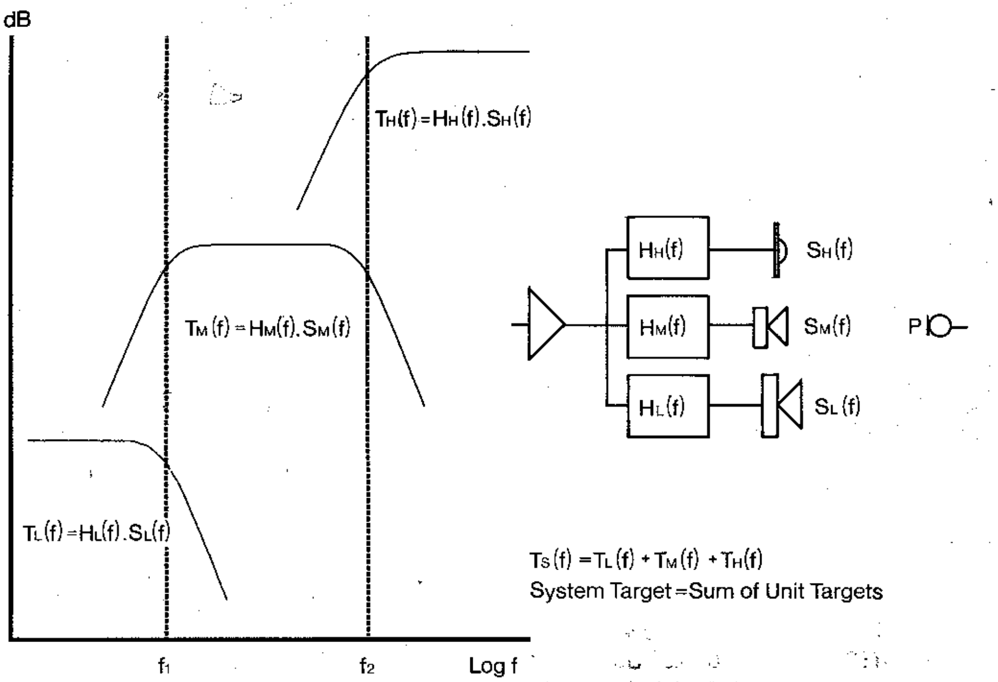
The Loudspeaker system and the practical realisation of a target function.

The overall frequency response of a loudspeaker system is given by the sum of the individual target functions produced by the various drive units and associated filter sections.

The filter is a passive array of electronic components connected between



Practical realisation of Target Function for HF unit



System Target Function

the amplifier and the drive unit. It generally has three functions.

Firstly, to provide equalisation for the non-flat frequency response of the drive unit both in the bandwidth and in the crossover regions.

Secondly, to produce the desired slope and roll-off characteristic.

Thirdly, to maintain correct impedance matching between amplifier and drive unit.

One class of filter having suitable characteristics for target functions is the Butterworth, or maximally flat type.

The first order configuration, which has a 6dB per octave slope, seems to be attractive because of its flat amplitude response, flat phase response and constant power response.

However, the first order filter suffers from three disadvantages.

Firstly, because of the shallow slope, it becomes necessary to control the individual target functions over three octaves either side of the drive units allocated bandwidth. Therefore for a mid range unit covering a band from 300 to 3000Hz (over three octaves) the filter would need to provide amplitude equalisation over more than nine octaves – not at all a practical solution.

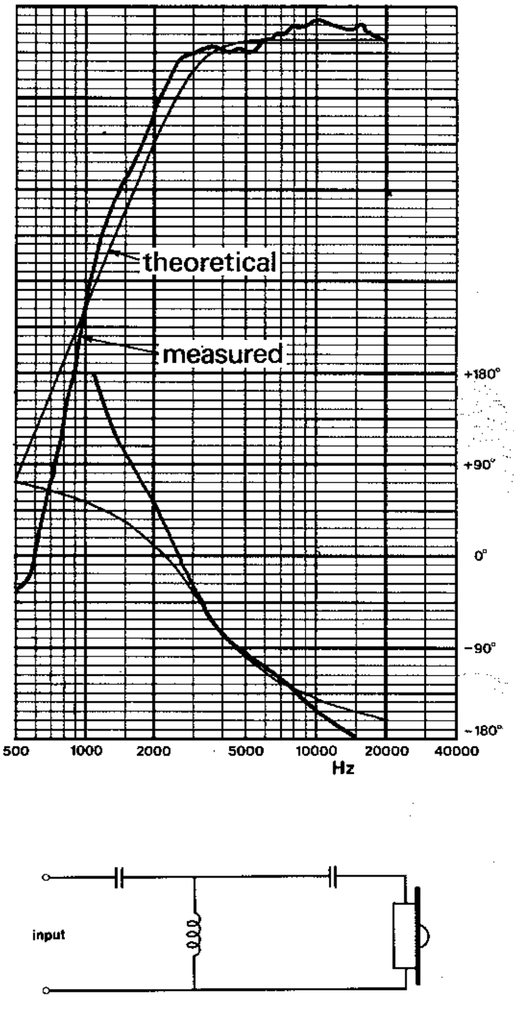
Secondly, many drive units possess cut off slopes steeper than 6dB per octave, so the filter would have to provide positive slope or boost to flatten the response even before any filter shaping.

Thirdly, the single series inductor or capacitor of a first order device is not sufficient to maintain a good match between the amplifier and the changing resistive load of the speaker system.

The 3rd order Butterworth Filter.

As a solution to these problems, the 3rd order Butterworth filter provides the best compromise.

Figure 3 shows a conventional 3rd order high pass network and its measured response compared with the theoretical. At high frequencies, the response is affected by the voice coil inductance, which at 5kHz resonates with the second capacitor in the filter.

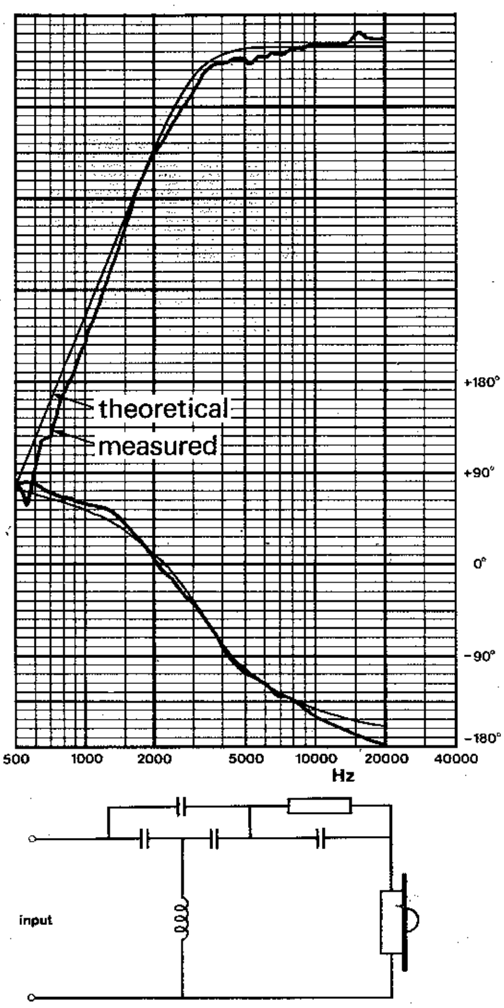


Conventional 3rd order high pass network

Fig 3

From 3kHz downwards the roll-off of the drive unit is incorrect and ultimately falls at nearly 30dB per octave below the fundamental resonance frequency, whereas the theoretical slope for a 3rd order network is 18dB per octave.

However in Figure 4, the practical realisation of an acoustic Butterworth network shows a much closer fit to the theoretical – being within ±1dB over most of the frequency range 500-20,000Hz.



Practical realisation of acoustic Butterworth circuit

Fig 4

This filter with its slope of 18dB per octave reduces the overlap region between drive units to only one octave which is much more practical.

Although it differs from the first order Butterworth in that its phase response is that of a first order all pass network, it has been proved that this is not a disadvantage, as the difference in phase response is not detected by the human ear.

Dispersion and the ideal listening position.

Results from research of this kind are only valid at a single listening position – that of the microphone during the experiments.

However, in practice, providing the individual drive units have sufficiently broad dispersion over their whole bandwidth, the response will be substantially the same over a wide horizon.

But because of the unavoidable geometrical separation of the drive units on the front baffle, dispersion in the vertical plane will not be so good. It turns out to be about ±2dB of the on-axis response over a small angle of about ±3° of the principal axis.

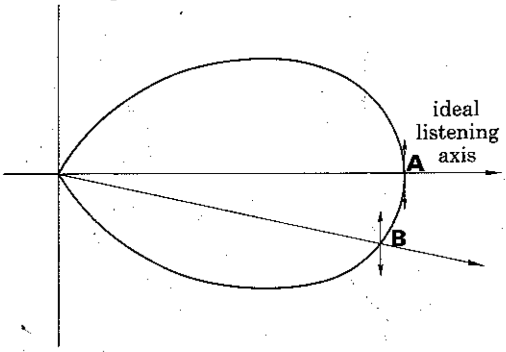
Therefore, an important feature of precisely designed systems is that the desired system target-function can only be obtained over a limited listening area.

It is therefore essential for the listening axis to coincide with the target design axis, because although off axis sound contributes to the general ambience and tonal quality, it is the direct sound alone which is responsible for the image forming properties of a loudspeaker system.

The total sound leaving the loudspeaker system is generally illustrated by a polar sound pressure level response which usually has a different shape around the vertical axis to the horizontal axis.

The overall (vector) axis of the polar diagram should ideally coincide with the listening axis, particularly in the vertical plane, as this will minimise the changes in sound pressure level heard by the listener in moving slightly up and down.

Polar Response. Sliced through the vertical axis.



A listener at position A experiences only slight variations in spl with vertical movement.

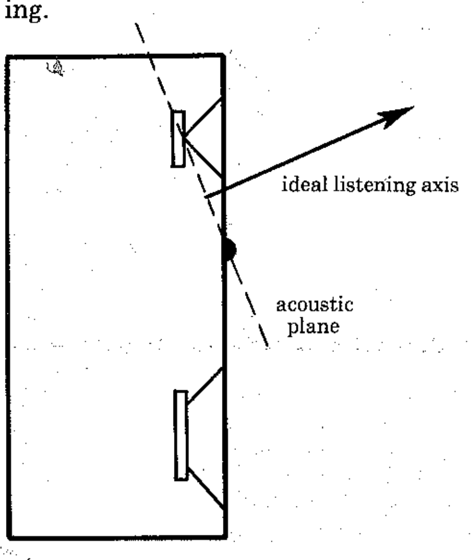
Clearly position B is not so good.

KEF realised that the direction of this axis was dictated by the acoustic plane, which connects the point sources of the drive units.

Leaving aside the bass unit, whose frequencies are not audibly directive anyway, and just considering the mid and treble units, KEF have turned this acoustic plane to their advantage.

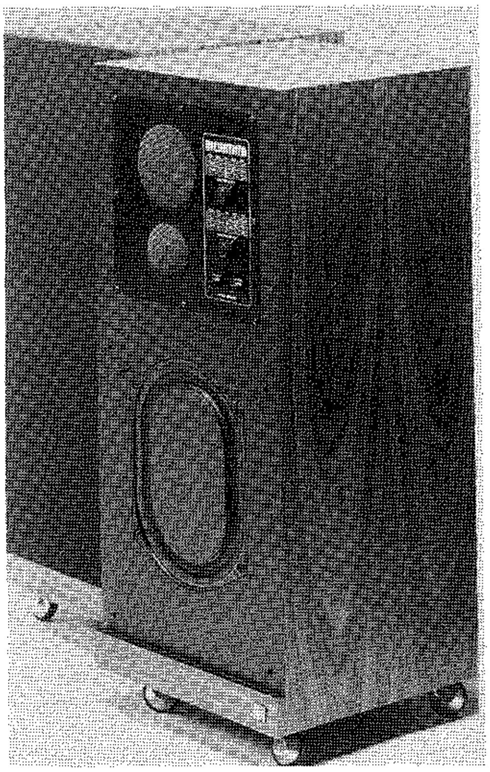
They have altered the geometry of these units in the Calinda and the Cantata, so that they are centre-line mounted, with the mid unit above the treble.

The effect is evident from this drawing.



The listener in an average listening position is now on the polar axis. In other loudspeakers with the treble above the mid, the best sounds are often heard on the floor.

Also, since the sound paths from mid and treble units are equal, the listener experiences no inter-unit time delay which would otherwise "blurr" the image.



KEF Cantatas

The centre line mounting improves dispersion in the horizontal plane. Because there is no relative inter-unit time delay it is wide, and the sound pressure level is very even over a broad arc.

For stereo listening, KEF have created a socially acceptable listening area which is both wide and deep. The old days of sitting in one fixed spot (big enough for one listener only) have gone.

Further research has produced a state of the art loudspeaker: The Reference Series Model 105, in which a number of features have been brought together by KEF's total system design approach.

The treble unit is above the mid unit, but staggered to compensate for inter-unit time delay.

In addition, the attitude of the mid and treble assembly is adjustable horizontally and vertically, so that once again the polar response axis may be made coincident with the listening axis.

In the Model 105, KEF have used an even more sophisticated filter, which in conjunction with the geometry of the drive units gives a more symmetrical polar response around the main axis.

One of the many benefits of this system approach is that a greater proportion of the direct (or early) sound contributes to the overall image.

The new filter and new bass enclosure design also provide a special bass loading effect, in which all three elements — electrical, acoustical and mechanical are ideally matched.

The result is a flat frequency response down to 38Hz at realistic volumes. This has previously been difficult to achieve since the impedance of the bass drive unit changes very sharply at such low frequencies. Most other loudspeakers, to avoid becoming excessively coloured at this depth, are designed to roll off before then in order to present a reasonable amplitude response.

The results of the 105 are startling. They represent another example of how KEF have brought sound theory and sophisticated mathematics in a very objective way to an inherently subjective field.

Reference Model 105

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