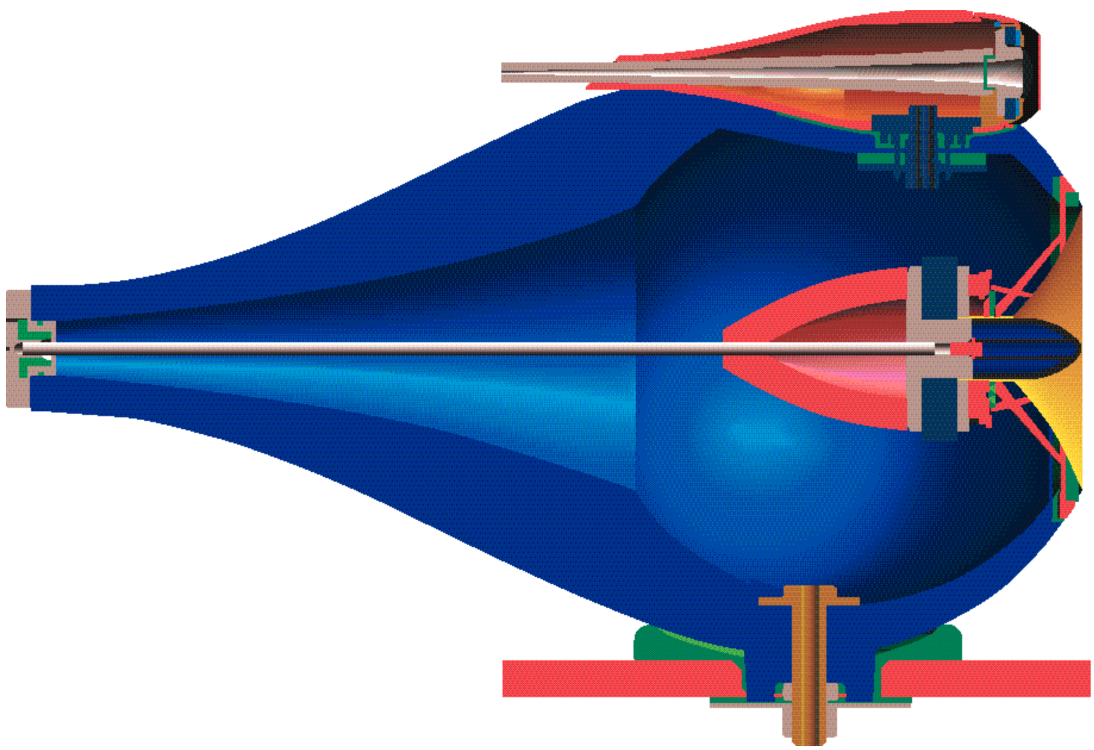
Nautilus™ 800 Series

Development of the Nautilus™ 801 Loudspeaker



LISTEN AND YOU'LL SEE

Introduction

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The original Nautilus™ loudspeaker was the result of a 3 year research and development programme for which the brief was simply to get as close to a perfect loudspeaker as possible by investigating all aspects of loudspeaker design. The engineer responsible for the project, Lawrence Dickie, defined the perfect loudspeaker as "one that neither adds anything to nor takes anything away from the input signal."

The resulting product was esoteric, both in looks and price and would always be out of reach to all but the very affluent. Prior to Nautilus™, the flagship of the B&W catalogue was represented by the Matrix™ 801 Series 3, a development of a model originally introduced in 1979 and for many years regarded as the de facto standard for monitoring in classical music recording studios.

The Nautilus™ clearly demonstrated a level of performance in certain areas an order better than the Matrix™ 801. The latter fell short of the output level requirements of the studios which, due to commercial pressures, could no longer restrict their activities just to classical music. There was therefore a need to develop a loudspeaker that incorporated the qualities attributed to Nautilus™, could play at significantly higher sound levels and would be affordable by a wider customer base.

This paper outlines the results of the three year programme to develop the Nautilus™ 801 loudspeaker.

PROJECT BRIEF

The following attributes were required of the new loudspeaker:

- Frequency response: At least as extended as Matrix™ 801 Series 3 (-6dB frequencies 23Hz and 30kHz)
- Maximum output level: ≥120dB peak spl at 1m
- Total harmonic distortion: <0.5% above 100Hz

Qualities indefinable in terms of simple numbers that had to be addressed included:

- Imaging (to be capable of focusing a stable centre image when listened to away from the "ideal" centre position and portraying good depth when such information is present in the recording).
- Lack of coloration (to be at least equal to the Nautilus™).
- Bass transient performance (so called "slam").
- Freedom from compression (dynamic range).
- Quality of performance should not be compromised by the type of music fed to the loudspeaker.

Restrictions placed on the design were:

- It should be no more than a 3-way system.
- It should be entirely passive.
- Styling should be domestically acceptable to a wider customer base than the Nautilus™ and reflect the heritage of the Matrix™ 801.

As the new loudspeaker was to be a development of both the Nautilus™ and Matrix™801, a résumé of both will be a useful precursor to the description of the development programme.

Matrix™ 801 Series 3

OVERVIEW

This was the last in a line of Model 801 variants, all with the common layout of separate bass enclosure, midrange "head" and tweeter mounted on top of the head. The separation of the enclosures allowed optimisation of the construction of each to suit the relevant frequency range and improved the off-axis response and thence the imaging when compared to more conventional box loudspeakers. The excellent imaging and low coloration of the system led to its adoption as the reference loudspeaker in many of the major classical music recording studios. However, the relatively low sensitivity and maximum output level, together with the relatively ponderous bass quality on impulsive type signals had begun to restrict the system's appeal when compared to Nautilus™ and other competitive loudspeakers.

DRIVE UNITS

Tweeter

The 25mm (1in) metal dome of this drive unit maintains pistonic motion to beyond the accepted limits of human audibility, having its first resonance peak at 27kHz. Its development involved the company's first use of Finite Element Analysis (see appendix IV). The natural response shape falls with increasing frequency and the level between 15kHz and 20kHz would fall to an unacceptably low level except for the influence of the "phase ring" placed in front of the diaphragm. This concentrates radiated sound energy on the unit's axis at the expense of the off-axis energy. Its use does detract slightly from the impression of openness in the sound stage.

Mid-range

The 125mm (5in) diameter unit features a woven Kevlar® cone. When Kevlar® was first used as a loudspeaker cone material in the B&W DM6, it was recognised as offering superior performance to other materials available at the time. However, it is only recently that we have begun to understand why this is so (See appendix i for a more thorough description of the performance of woven Kevlar® cones and the measuring techniques used). The relatively small diameter of the drive unit, whilst giving excellent dispersion up to the crossover to the tweeter, does restrict the sensitivity and maximum output.

Bass

The 300mm (12in) long throw unit was designed to have a very low resonance frequency in order to extend the bass cut-off frequency as low as possible. The desire for a low resonance frequency results in a relatively soft outer cone suspension (surround). The cone is made from a PVC based material. Both the cone and surround have a tendency to deform at high levels under the influence of the high pressures inside the cabinet. This is thought to have a bearing on the bass transient performance of the loudspeaker.

ENCLOSURES

Tweeter

The design of the tweeter enclosure is based on the fact that, if it is barely wider than the radiating surface itself, the interference from waves diffracting off the outer edge is extremely low. The result is a smooth response and wide dispersion from the small radiating area. The coupling to the mid-range enclosure below includes a flexible mount to prevent any high frequency vibrations being transmitted to the lower structure.

Mid-range

The small frontal area maintains a wide dispersion throughout the unit's operating range and the effects of diffraction are reduced by chamfering the edges. The cosmetic outer plastic shell is reinforced by an inner layer of Fibrecrete (mineral loaded concrete) that improves the stiffness and adds mass to the enclosure. This and the anti-vibration mounting of the drive unit ensures an extremely low level of sound radiated by the enclosure itself via both reaction forces on the drive unit's chassis and transmission of sound from inside the enclosure generated at the rear of the drive unit. Like the tweeter, the mid-range enclosure is flexibly mounted to the cabinet below.

Bass

The ported cabinet features Matrix™ bracing, more fully described in appendix II. The use of Matrix™ reduces the level of radiation from the cabinet panels by around 45dB compared to the unbraced cabinet.

CROSSOVER

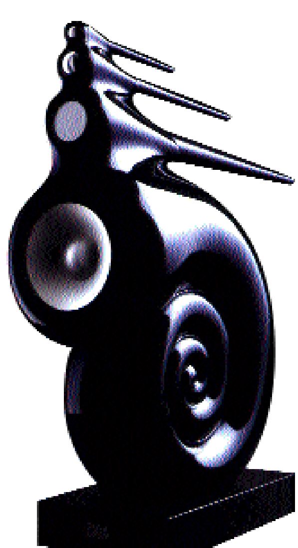
Mounted on a single printed circuit board located on the underside of the bass cabinet, the components include a proliferation of electrolytic capacitors and cored inductors, other than in the tweeter section, which limit the linear power handling of the system.

The Nautilus™ Loudspeaker

OVERVIEW

The Nautilus™ loudspeaker not only looks different but sounds different from any other speaker and is a classic example of the principle of "form following function".

The premise that the speaker should add nothing to nor take anything away from the input signal, translated into scientific terms, means that it should be totally free of resonances, in both the enclosures and in the drive units themselves. The techniques used to achieve these requirements necessitated a 4-way system with active crossover and an unusual enclosure design.



DRIVE UNITS

The requirement for the units to be resonance free translates into a desire for them all be perfect pistons within their pass bands and up to two octaves above them as well. As a result of the decision to use perfect piston motion, a minimum number of four drive units is dictated by the need to maintain as nearly a constant directivity as possible – because perfect pistons become increasingly directional at high frequencies – and because at a high enough frequency the diaphragms cease to behave in a pistonic manner, displaying high Q resonances.

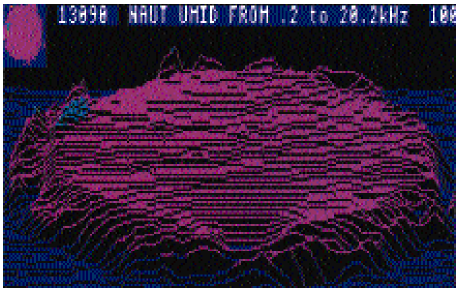
Perfect Piston Drive Units

At first sight "perfect piston" drive units, (ie those that move solidly without bending and with a total freedom from resonances), would appear to satisfy the ultimate requirement for perfect sound reproduction. That is, within their linear motion limits they add nothing to the sound and take nothing away – they simply reproduce exactly what is fed to them, and only what is fed to them, so perfectly converting electrical energy into acoustic energy. However, one must have regard to the fact that the loudspeaker has to convert a one dimensional electrical signal into a three dimensional sound field. Beaming or directivity effects, limit the useful bandwidth of the drive unit.

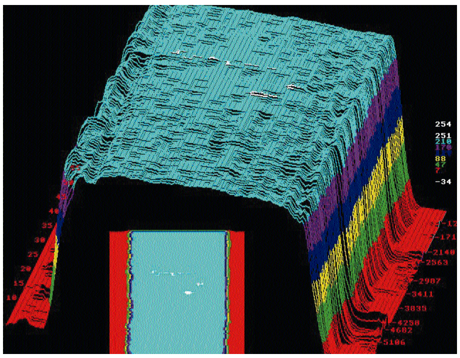
A perfect piston at low frequencies, where it is small relative to a wave-length, and if it is placed in a small box to cover up the radiation from the back of the piston, will radiate sound equally in all directions. As the frequency increases, and consequently as the wavelength decreases, more and more sound is radiated in the forward direction and less in the rearward direction. When the wavelength is equal to the circumference of the driver, a definite forward beam begins to form with a set of side lobes in the directivity pattern. As the frequency is further increased this beam becomes narrower and narrower and more intense, until finally all the sound radiated from the driver will go straight forward, forming a beam having the same diameter as the drive unit itself.

In order to maintain a uniform off-axis response, one must restrict the unit's bandwidth to below that frequency where the wavelength is equal to the circumference of the diaphragm, crossing over to a smaller unit above this. Much research has indeed shown that a constant directivity pattern is synonymous with the perception of good hi-fi, since the speaker will not only be radiating an equal amount of sound in the forward direction at all frequencies, but will also be coupling into the rest of the room, via its radiation in other directions, in a similar manner at all frequencies. (ie: Its energy response as distinct from its on axis frequency response will be more nearly constant with frequency).

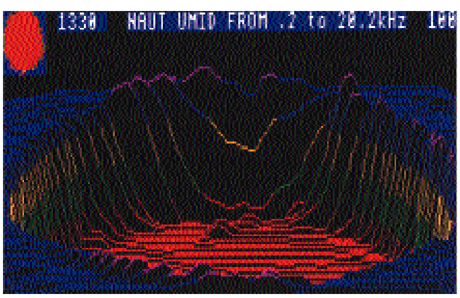
The Nautilus™ drivers are all perfect pistons within their pass bands and for 2 to 2½ octaves above as well. Since there are 4 drive units, we are able to cross over to another driver before the forward beam becomes troublesome – and incidentally also before the first break up resonance comes close enough to the pass band to be audible, even after attenuation through the crossover network.



1 Whole Cone Single Frequency Scan
Nautilus Drive Unit. Frequency at Top Left Hand Side.



2 Frequency Slice Plot 100Hz to 6000Hz
Nautilus Upper Mid – good to 6Khz.
Low Frequency Slice at rear. High Frequency Slice at front



3 Upper Mid going down below fundamental

Tweeter

The moving parts are similar to those used in the Matrix™801. However, the need to use a phase ring to support the response between 15kHz and 20kHz is removed by having the luxury of an active filter, enabling better openness to be realised. The dome is further stiffened by the use of a carbon fibre ring, ensuring it behaves as a perfect piston all the way up to 25kHz. Directivity at even the top end of the band is not a problem because the diaphragm is only 25mm in diameter.

The magnet structure has an exponentially tapering hole through the centre pole as the start of the rear tube (see '**ENCLOSURES**').

Upper Mid-range

This 50mm diameter dome diaphragm behaves as a perfect piston right up to 14kHz. However, since it does break up above this frequency it cannot be used as a tweeter on its own. Also the relatively large dome begins to produce a pronounced forward beam if used much above 4kHz. It is also not really large enough to go down much below 500Hz or so, since its excursion would begin to reach unacceptable levels. In short it is an excellent upper mid driver which must be used in conjunction with the tweeter above and the lower mid-range unit below. (Fig 4)

Upper Mid Range Unit Frequency Plots

- Unit is a perfect piston from 0.2kHz to 14kHz a typical "still" is shown from the animation (Fig 1)
- Dome goes downwards below its fundamental resonance and upwards above it (Fig 3)
- This illustrates the phase change that occurs through the fundamental resonant frequency
- The frequency slice plot (Fig 2) illustrates this near perfect pistonic behaviour

Lower Mid-range

This unit is also a perfect piston within its operating band and for about 2 octaves above it. The flat front of the diaphragm is beneficial in preventing perturbation of the upper mid-range unit's response. It was found that a cone unit adjacent to the upper mid-range unit introduced a cavity resonance. If, however, this were the only mid range unit, the first break-up resonance would lie just outside the operating band and would be audible. Fig 5 illustrates this:

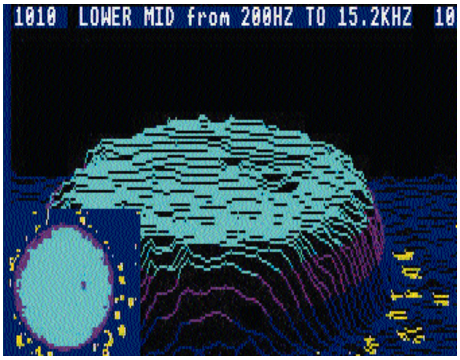
Nautilus Lower Mid Range Unit
(Frequency Plot)

- The Cone goes downwards below its resonance and upwards above it: A typical "Still" is shown. (Fig 4)
- This illustrates the phase change that occurs as it goes through its fundamental resonance. It is a perfect piston up to 3.5kHz.
- There is a powerful resonance at 3.5kHz. shown. (Fig 5)
- In The Nautilus it is crossed over to the upper mid unit more than two octaves below this.
- The Slice plot (Fig 6) illustrates perfect piston motion over its pass band range and 2 octaves above.

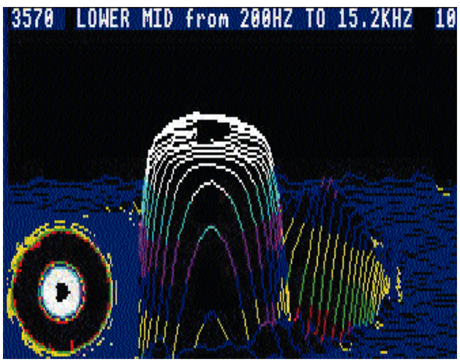
Bass

Once again this unit's diaphragm is a perfect piston within its operating range, but this cannot be extended much higher in frequency before its first break-up resonance also starts to be come a problem, as in the case of the lower mid-range above.

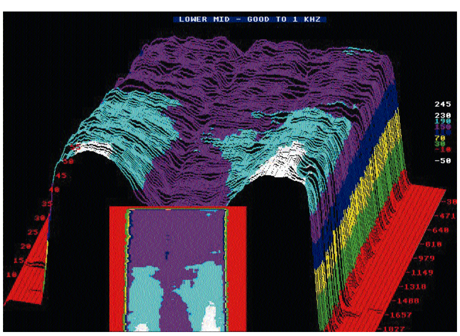
The slice plot (Fig 7) illustrates pistonic motion up to crossover and for 2 octaves above it, when the outer edges of the driver start to move more then the inner regions.



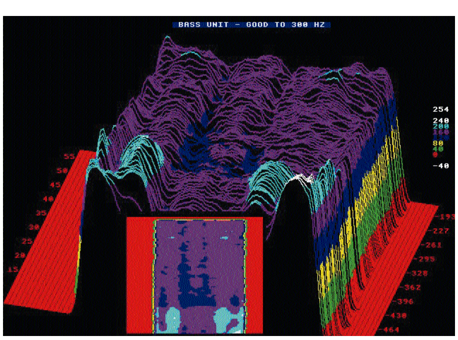
4 Whole Cone Single Frequency Scan
Nautilus Drive Unit. Frequency at Top Left Hand Side



5 Whole Cone Single Frequency Scan
Nautilus Drive Unit. Frequency at Top Left Hand Side



6 Frequency Slice Plot 300Hz to 2000Hz
Nautilus Lower Mid – good to 1kHz
Low Frequency Slice at rear. High Frequency Slice at front.



7 Frequency Slice Plot 100Hz to 500Hz
Nautilus Bass Unit – good to 3kHz
Low Frequency Slice at rear. High Frequency Slice at front.

ENCLOSURES

The whole frontal profile of the enclosure is smoothly rounded to avoid diffraction effects.

Tweeter, upper mid-range and lower mid-range

All three of these units employ straight exponentially tapering tubes to absorb the radiation from the rear of the diaphragms and all feature profiled holes through the centre of the magnet structure as the start of the tube. See appendix III for a fuller explanation of the principle of tube loading.

Bass

The bass unit also fires into a tapering tube, but in this case the tube is coiled to save space. This type of loading results in an over-damped bass alignment. The roll-off is gradual but starts at a relatively high frequency which requires an active crossover to provide the necessary equalisation.

CROSSOVER

The active crossover frees the designer from the restrictions of limited drive unit sensitivity and response shape, so it is a simple matter to provide any equalisation required and balance the system correctly. In addition it allows the use of all-pass filters to alter the time delay of the drive units. This avoids the need to step the baffles which might lead to irregularities in the response.

The Nautilus™ crossover employs subtractive filters. The low-pass section is a Bessel alignment designed for maximally flat time delay. The overall phase characteristic of total response is matched to this delay and the high-pass is derived by subtracting the low-pass from this total all-pass.

SUMMARY

Nautilus™ Technology may be summarised as follows:

- Rear tapered tubes lead the sound away from the rear of the driver to be totally absorbed.
- Completely rounded shapes everywhere limit box edge diffraction effects.
- Perfect piston drivers ensure faithful reproduction within their pass bands.
- Dome shaped or flat fronted drivers ensure minimum re-radiation of sound from adjacent drivers.
- Totally over damped bass driver ensures no trace of more normal ringing at the bass resonance frequency.
  (Editor's note – this statement is not true if the electrical EQ restores a more normal alignment as the EQ will have its own group delay)
- Totally damping out the bass resonance dictates active equalisation of the bass response.
- Active subtractive crossover networks ensure totally flat crossover regions – amplitude and phase
- Maximally flat delay characteristic used for the crossover roll off.
  (Editor's note – the maximally flat delay characteristic of Bessel filters is in the pass band for low-pass, but in the stop band for high[pass)

Development of Nautilus™ 801

OVERVIEW

It was desired to incorporate as much recognised Nautilus™ technology into the new system, but it was clear that a wholesale transfer of technology would not yield acceptable results because:

- A 4-way system was not an option, therefore the mid-range unit at least would need to encompass a frequency range greater than could be successfully achieved using a single pistonic drive unit working into a simple tapered tube.
- The use of a tapered tube at low frequencies, with its overdamped characteristic, could not be employed, as active equalisation was not an option.

In addition, each drive unit would be required to have higher sensitivity than its equivalent in either the Nautilus™ or Matrix™801 Series 3 in order to realise the desired maximum output with achievable power handling.

DRIVE UNITS

Tweeter

The tweeter motor system needed to be physically as small as possible to maintain the small frontal area essential for avoiding diffraction problems. In addition, maximum sensitivity was required to maximise the output capability. There was also the need to bore the pole piece to allow the radiation from the rear of the diaphragm to travel through to the tapered tube attached to the rear of the pole piece which absorbs the unwanted energy.

The eventual solution to this problem proved to be quite complex. The pole piece was undercut level with the underside of the top plate to make the fringing field more symmetrical and concentrate more flux in the gap. It was also profiled at the tip, following the radius of the dome diaphragm, to prevent unwanted Helmholtz resonances occurring behind the dome. The bore through the pole piece is of an exponential profile to match the profile of the rear tube and blend seamlessly.

The permanent magnet was manufactured from nickel plated Neodymium Iron Boron (NdFeB) to maximise the energy product for the size of magnet. The top plate also required intricate geometry to balance the fringing field and maintain the linearity of flux distribution throughout the magnetic air gap. It included location pins with which to accurately align the diaphragm assembly.

Copper clad aluminium ribbon wire was employed for the voice coil. Aluminium has a lower density than copper which reduces the moving mass. In a tweeter, this reduction in mass more than compensates the lowering of sensitivity resulting from the higher resistivity of aluminium. The ribbon wire (which is rectangular in crosssection) is used to eliminate all air gaps that are a consequence of using round section wire. This ideal packing factor maximises the volume of conductor present in the magnetic gap and significantly raises the sensitivity.

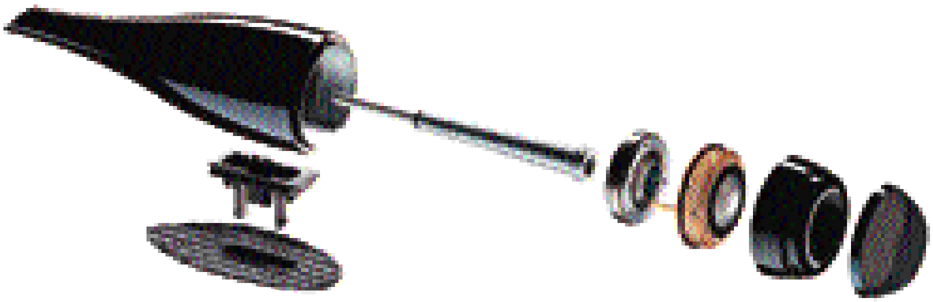
The wire is wound on a Kapton® former to minimise mass and also to eliminate eddy currents within the magnetic gap which would be present if using a conductive aluminium former.

One of the main causes of failure in earlier tweeters was fracture of the leadout wires which were extensions of the winding wire. This had to be addressed, especially as the unit would be required to reach higher output levels. The answer was to use beryllium-copper strips, which allow repeated flexure without fracture.

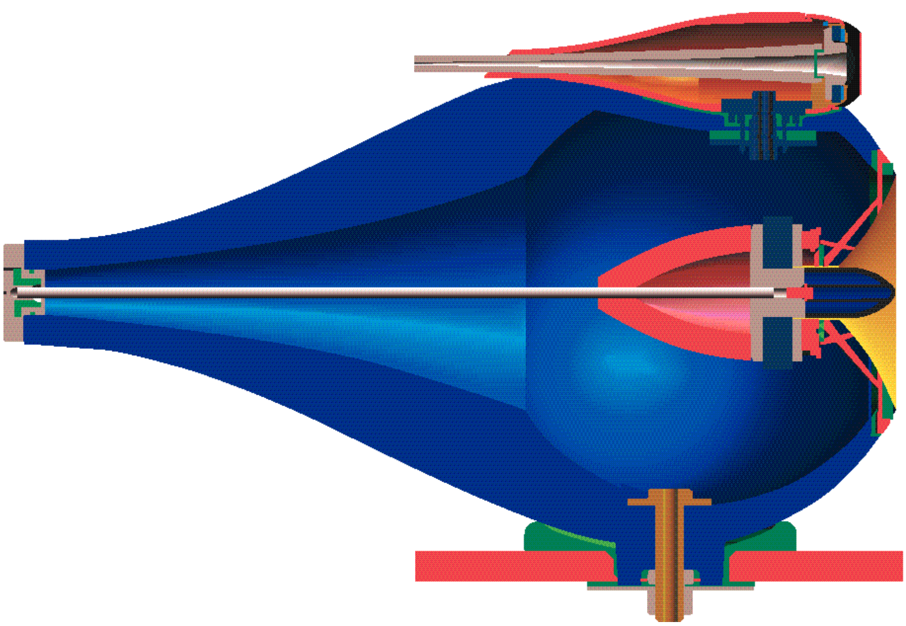
To minimise the harmonic distortion within the tweeter, the moving assembly needed to be designed such that all the components exhibited linear behaviour throughout the pass-band. Also, the surrounding geometrical environment needed to be designed so that it did not introduce distortion mechanisms arising from diffraction or reflection.

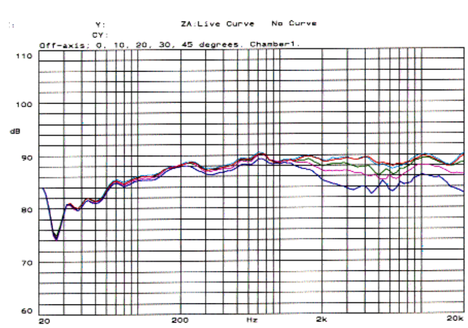
As in both the Nautilus™ and the Matrix™801, the dome is manufactured from anodised aluminium alloy, which offers one of the best strength-to-weight ratios available. This ensures that the break-up frequency of the dome remains outside the audible bandwidth.

Orthodox tweeters use a small roll surround to allow the dome diaphragm to move to and fro. However, it was discovered, using a laser Doppler velocimeter, that the roll surround is responsible for a large proportion of



8 Sectioned View of Nautilus 800 Tweeter





9 Frequency Response Showing 15°, 30°, 45° and 60° Horizontal Dispersion

distortion within the operational bandwidth of the unit. The aluminium dome exhibits pistonic motion throughout its bandwidth and beyond, whilst a conventional roll-surround does not move pistonically – there are numerous vibration modes present.

A new suspension was developed using environmentally stable foam. Working in compression rather than rolling as a normal surround, it avoided these resonance problems. Furthermore, the foam surround led to a useful reduction in the moving mass of the moving parts, which further increased the sensitivity of the unit.

The phase ring on the front of the tweeter, familiar on the Matrix™801, was dispensed with, as the inherent sensitivity of the unit allowed equalisation to the required system sensitivity without having to raise the level at extreme high frequencies. The dome therefore radiates without any obstacle, improving the dispersion at the top of its pass band. The use of a short horn in front of the unit was considered as a way of raising the sensitivity. This, however, would also lead to a restriction of the dispersion and it was felt that the unit had sufficient sensitivity without.

The tweeter does not use any screws in its assembly. The whole assembly is held together by the mating of the front cap and the housing. The absence of screws offers a clean and uncluttered environment around the diaphragm. (Fig 8)

The geometry of the front cap and housing was carefully profiled to avoid discontinuities and to minimise the frontal area.

The exclusion of a horn and the painstaking design of the curvature of the front cap results in an outstanding off-axis response. (Fig 9)

The end result was a tweeter having an inherent 96dB spl sensitivity, low Total Harmonic Distortion (THD) and excellent dispersion.

Mid-range

It has already been stated that the use of a pistonic diaphragm over the required pass band of a single mid-range unit would be impossible to achieve, given the physical parameters of currently available materials. Woven Kevlar® has proved to be an outstanding diaphragm material, when correctly doped with resins and damping compounds (see appendix I). Nevertheless, it was felt that improvements could be made to the performance of existing units.

Laser measurements had shown that, although the cone behaviour was improved by using Kevlar®, the behaviour of outer roll surrounds, which provide a significant proportion of the total radiating area, left much to be desired. In particular, a phenomenon known as the "surround resonance" tends to occur right in the middle of the mid-range where the surround moves in the opposite direction to the cone. This phenomenon manifests itself as either a peak or a dip in the frequency response between 700Hz and 2kHz depending on the size of the diaphragm, the surround and their relative areas.

Laser plots and frequency measurements always showed the following features at the surround resonance frequency:

1. The whole of the surround moves the opposite way to the whole of the cone.
2. The movement of the cone at this resonance is in the same direction as that of the cone at neighbouring frequencies – but of greater amplitude.
3. The frequency response shows a peak if the extra movement of the cone at resonance more than compensates for the opposing movement of the surround in terms of volume velocity.
4. The response will show a dip if the reverse surround volume velocity is greater.
5. The effect will still be present, but not noticeable in the steady state response, if the areas and relative movements of the cone and surround exactly balance.

All resonances in the cone / surround combination can be thought of as standing waves. These start out as bending waves moving out from the voice coil, they travel up the cone to a point where they are reflected and then move back down the cone to the voice coil again and so on.

If an whole number of wavelengths exactly fits into this "out and back" travel path, then peaks in the wave on the way out coincide with peaks in the returning wave and a resonance or standing wave results.

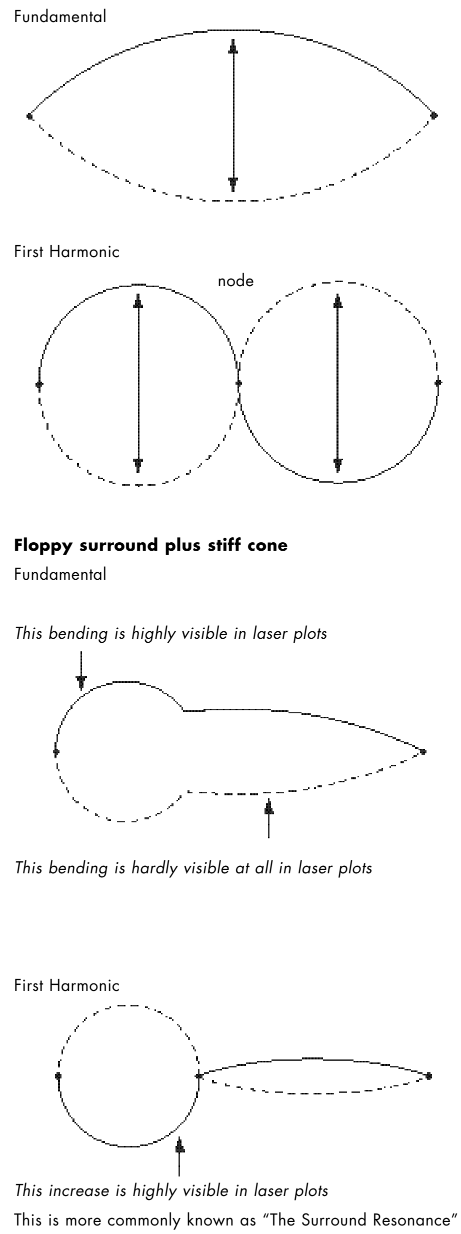
The lowest resonance occurs when an area between the cone neck and the outer edge of the surround simply moves up and down and the cone neck and outer edge are (relatively) stationary. The first harmonic occurs when this "in between area is also stationary and the area closer to the neck centre moves in the opposite direction to that nearer the surround. This first harmonic is the standing wave or resonance which is thought to be responsible for the so called "surround resonance".
(Fig 10)

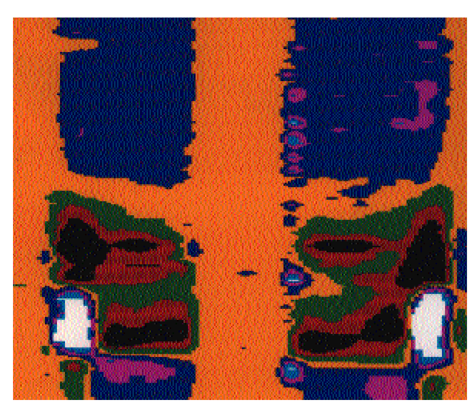
With a loudspeaker diaphragm there are two distinct areas – the cone and the surround. The cone is relatively stiff. It will therefore not bend very much and the speed of bending waves is much faster in it. The surround is relatively much more flexible. It will therefore move a great deal and bending waves travel much slower in it.

When there is a standing wave in the cone / surround combination therefore, the cone will appear to move one way almost as a solid piston with a slightly increased movement at the centre, whereas the surround will move the opposite way a great deal more, with a clearly visible maximum amount of movement at its centre. The position of the zero movement node will be near to the cone surround junction because of the faster bending wave velocity in the cone material. Laser progression plot measurements showed that travel times of bending waves from the voice coil to the surround are in the right order of magnitude for this to be the correct explanation for this phenomenon.
(Fig11)

Furthermore, measurements of a very stiff metal cone speaker show up not only the familiar surround resonance, but also the lower frequency fundamental, (ie when both the surround and the cone appear to move in the same direction, but more than they would normally, particularly the surround). This is shown up at point a on the laser plot. The more usual surround resonance shows up at point b. The third and higher harmonics of this standing wave system will appear at higher frequencies than those shown here.

10 One Continuous Material





11 Contoured Slice Plot of Surround Resonances
a The Fundamental of the Surround Resonance (black – black – yellow – black – black)
b The First Harmonic i.e. Surround Resonance (white – black – yellow – black – white)

In fact all speakers with relatively flexible surrounds show this fundamental mode behaviour. However, it is not usually spotted from the response curve because it will only slightly affect level in the order of 0.5dB or so. Laser plots, when taken sufficiently low down in frequency, invariably show the surround moving more than the cone does at this fundamental bending wave modal frequency.

Since this is a bending wave reflection phenomenon, it can be cured in four places:

1 At the voice coil / cone junction
2 At the surround / cone junction
3 At the surround / chassis junction
4 In the material of the cone and / or the surround

Taking each of these in turn:

1 At the voice coil / cone junction.
Associated with all wave transmission mechanisms there is a so called "characteristic impedance". A good analogy is a TV antenna coaxial cable, where the characteristic impedance is 75W. If the end of the TV cable is terminated with a 75W resistive load, no energy will be reflected back down the cable at all. (The standing wave ratio will be zero).

If the cable is shorted out completely, no energy is absorbed at all and 100% of it is reflected back, but out of phase with the incoming wave. Conversely, if the cable is left completely open circuit (ie nothing is attached to it at all), all the energy will be reflected again, but this time it will be in phase with the incoming wave.

Thus as you change the load from just below to just above the characteristic impedance, the phase of the reflected wave will change by 180º and, more importantly, the conditions for standing waves to form will be altered and the resonance will now occur at a significantly changed frequency, much lower in the "shorted out" case.

With speaker cones the characteristic impedance to bending waves at the voice coil can be changed simply by altering the coil mass. As the changes in mass take the bending waves towards their characteristic impedance, the surround resonance amplitude falls to zero at exact impedance matching. Then the frequency at which the surround resonance occurs will change significantly by, and determined by, the half wavelength that the impedance change represents, and its amplitude will then increase as the impedance is altered further in the same direction. This is a major change indeed.

If bending waves were not dispersive, (ie if their speed did not change significantly with frequency), the frequency of the surround resonance would be doubled. Since they are dispersive however, the frequency is changed by a factor closer to $\sqrt{2}$ instead.

2 At the cone / surround junction.
The effect of this junction is more complex.

If the bending wave impedances are the same on both sides of the junction, the wave will pass through without reflection. However, damping placed at the junction can absorb some of the energy as it passes through.

If the junction represents a short circuit, then all the energy will be reflected back out of phase and none will pass into the surround at all. Standing waves will then form in the cone on its own and will occur when the phase changed wave fits into the cone's radial dimensions.

If the junction represents an open circuit, (ie if the cone moves in a totally unrestrained manner), then once again all the energy will be reflected, but in phase this time. No energy will pass into the surround and standing waves will form in the cone alone when the non-phase-changed wave fits into the cone dimensions.

In reality, the impedance of the junction will be neither exactly open or closed circuit and certainly will not remain constant as the frequency is changed, due to the different dispersive characteristics of the surround and the cone materials. So there will be standing waves in the cone and in the surround and also in the cone plus the surround, all occurring at different frequencies and different amplitudes.

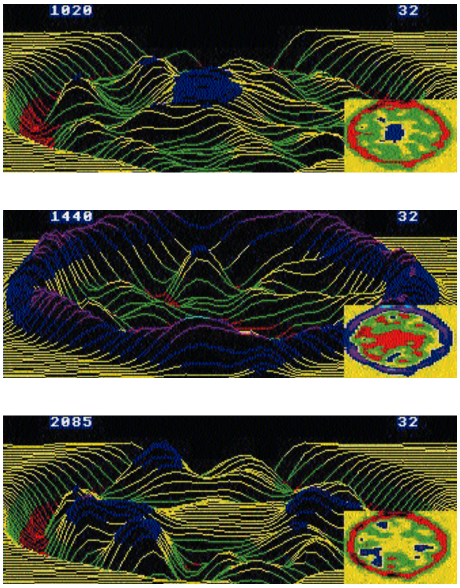
On the whole it is thought better for all the energy to pass into the surround of a more normal cone / surround case, as it is possible to conceive that most of the energy can be absorbed in a suitable surround.

3 At the surround chassis junction.
These resonances can be reduced by placing a ring of flexible absorber at the surround / chassis junction. This prevents any bending wave energy being reflected back into the surround and hence the cone – so stopping this phenomenon dead. However, this cure only works if virtually all the bending wave energy passes from the cone into the surround. Impulse Progression plots show that this is not often the case, so the effectiveness of this method is not guaranteed.

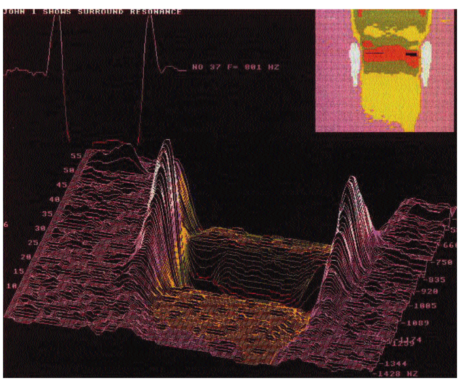
4 In the cone or surround material.
If the bending waves can be absorbed anywhere in the system, this standing wave phenomenon will be cured. This can be done by improving the damping within the cone material itself. Furthermore this has the added bonus of damping out the more usual cone resonances as well, (i.e. those standing waves in the cone only – not in the combination of cone plus surround which we are considering here).

Experimentation with different resin mixes for Kevlar® cones has shown this up most markedly. A slight limitation to this damping approach is that the frequency response of cones with excessive damping, and probably lower stiffness as a consequence, will not be so well maintained to higher frequencies.

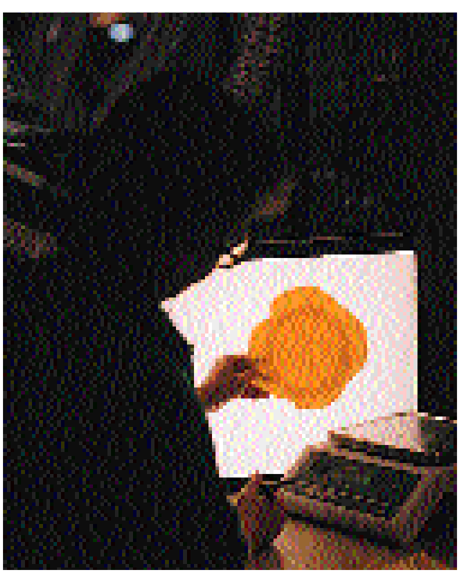
There is also another negative effect from making the surround material too lossy – especially for bass cones – and that is that they lose their "punch" or "attack". This is because a highly damped surround tends to be more massive than a less damped one and has poor resistance to deformation under the effects of pressure within the cabinet. This means that their mass tends to delay the initial acceleration of certainly the outer regions of the cone and possibly the whole cone in response to an input impulse. The effect is a perception of slow, unexciting bass when a highly damped surround is used in a bass unit, and a limited higher-frequency response when it is used in a mid range unit.



12 Effect of Floppy Surround (Impulse Plot)



13 Effect of Floppy Surround (Slice Plot)



SUMMARY

- The so-called surround resonance is nothing more than the first harmonic of the overall coil / cone / surround bending wave resonance phenomenon, when exactly one complete bending wavelength will fit into the combined distance from coil to chassis.
- The fundamental of this resonance always exists, but usually goes unnoticed since it only marginally affects the steady state response.
- So-called break up resonances of the cone and the surround are the result of the remaining harmonics of this phenomenon.
- The resonances may be cured in a combination of at least 4 different ways, possibly the best one being to match the impedances at the cone surround junction and lead the energy in to the surround to be totally absorbed if possible.
- Similar phenomena occur for each wave type or propagation mode.

Fig 12 illustrates the surround behaviour showing the motion continuing for a long time after the input impulse has ceased.
Note that:

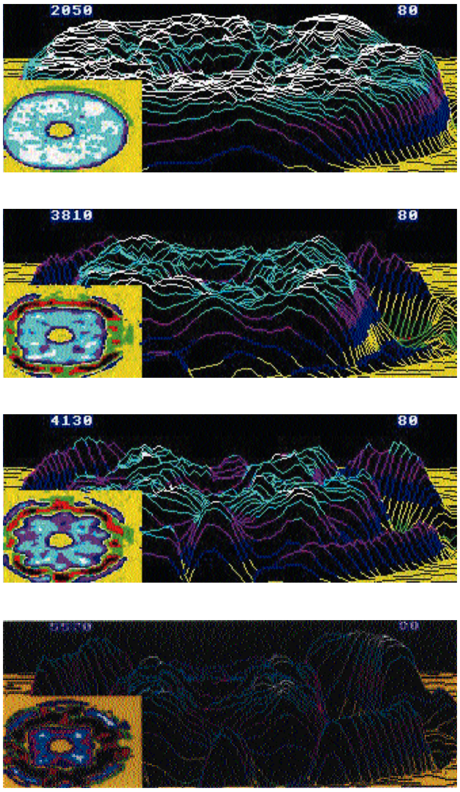
- The impulse travels rapidly to the surround.
- The surround carries on slowly oscillating long after the impulse has passed.
- The two curves shown here illustrate the surround in an upwards and a downwards movement.

This graphically illustrates how surround resonances can carry on radiating long after the impulse (and therefore the music), has ceased. This greatly colours the sound. If the surround radiation can be removed, as in the new Nautilus™801 mid-range, the sound quality is greatly improved as a result.

Fig 13 shows a picture of a laser slice plot with low frequencies at the back and high frequencies at the front.

The "surround resonance" is seen clearly in the middle, with the cone going downwards much more than at neighbouring frequencies and the surround going upwards at the resonance, whereas it behaves properly to each side of the resonance i.e. it acts as a "lever arm" having no resonant behaviour of its own.

If the volume velocity from the cone wins out we get a peak in the response whereas if the volume velocity from the surround wins out we get a dip in the response. If such a resonance ever appears in the pass band then corrective measures have to be taken – e.g. any one of those described above.



14 Whole Cone Single Frequency Plots
Top Picture at Low Frequency, Bottom at high Frequency
Note: 4-way symmetry and smoothly reducing central area

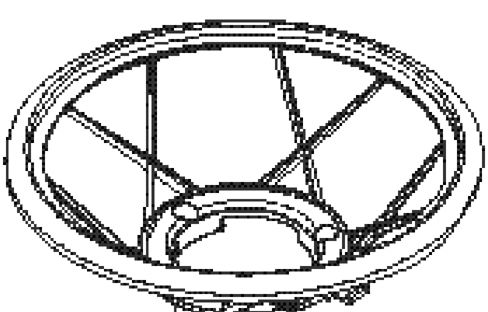
THE SURROUND RESONANCE – APPLICABILITY TO THE NEW "SURROUNDLESS" KEVLAR® CONE

The new "surroundless" Kevlar® cone takes the work described in the mid-range section and matches the impedance of the Kevlar®, to the suspension at 8 places around the circumference spaced by 45º and roughly corresponding to half way between the warp or weft and the bias directions. The surround material itself evolved with experimentation. Early tests with silicone rubber rings drastically reduced the surround resonance effect, but only gave a clue to the potential of matching the impedance, and also resulted in too high a drive unit fundamental resonance frequency. In the end an open cell PVC material having specific levels of plasticiser and cell size realised the desired parameters.

As is well shown in laser plots, the Kevlar® cone, although circular in form, behaves physically as if it were square. In line with the warp and weft, the material is at its least stiff and the bending waves travel at their slowest. In the bias direction, the material is stiffest and the bending wave velocity is at its maximum. At the interim points of matched impedance, energy is passed directly into the surround and there is no reflection. However, on either side of these points the impedance of the surround is either above or below the characteristic impedance of the Kevlar® and there are therefore two different conditions for reflection.

The reflections at points where the cone impedance is below the characteristic impedance of the surround, exhibit phase reversal and so appear at a lower frequency than those at points on the other side of the characteristic impedance match, where the waves are reflected in phase. In the far field, the peaks in the resonance pattern on one side cancel out the dips in the pattern on the other side and vice versa.

Note in this case that the surround, being of small surface area and operating in compression at the rear of the cone edge, gives rise to a minimum of radiation of itself and is devoid of the internal resonance effects that a normal roll surround would exhibit.



Bearing the above behaviour in mind when looking at single-frequency whole-cone plots of such a cone, we see that the resonances always appear to combine a bell mode, (having four lobes i.e. 4 regions going downwards and 4 regions going upwards), with a whole series of axisymmetric modes, (modes in concentric circles where each successive circle of the cone is moving the opposite way to the previous one). This means that the outer regions of the cone, when the low frequency perfect piston region is exceeded, always behave like a number of octopoles, (4 point sources in phase and 4 interspersed point sources out of phase), which are very inefficient at radiating, leaving the inner regions to dominate both the on-axis frequency response and the directivity, which is consequently maintained nearly constant over a wide frequency range.
(Fig 14)

This analysis will be true, only provided that the acoustic environment in which the cone unit finds itself submerged, both inside the box and externally, is symmetrical itself. If there are sufficiently strong local changes in internal acoustic impedance, such as those found in ordinary boxes, the requirements for "cancelling symmetry" will no longer be fulfilled and the performance of the cone / box combination will deteriorate. Consequently, the mid-range enclosure was designed to maintain symmetry round the drive unit.

A further possible junction – between dust cap and cone – was avoided by the use of a central, bullet-shaped dispersion modifier. This device improves the off-axis response of the diaphragm.

The motor system

Having removed the effects of the surround, the deficiencies of standard motor systems became increasingly obvious. Harmonic distortion caused by the non-linear inductance of the voice coil, was reduced by the addition of a copper sheath on the centre pole piece. This also reduced the overall value of the inductance and usefully extended the high-frequency response of the unit.

The chassis

Drive units built using a standard chassis sounded "closed in", despite having a good steady state response.
Laser measurements of the diaphragm behaviour also failed to show any major

deficiencies. However, once the voice coil / diaphragm / surround design had been refined, response decay measurements showed clear reflections off the chassis structure. Previously, a frequency peak caused by a coil to cone mismatch had exactly coincided with a response dip caused by a reflection.

A new chassis was designed having as large an open area as possible. In addition the rim of the chassis was feathered at the rear and shaped at both the front and rear to smoothly align with the walls of the enclosure. The thin section legs were criss-crossed for maximum strength and five were used to avoid interaction with the four-fold symmetry of the woven Kevlar® cone.

Bass

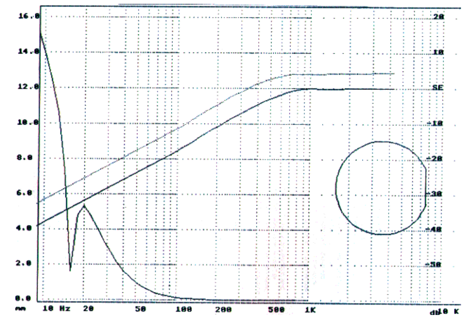
Advantages of Producing a Coherent Wavefront

One remarkable fact was consistently noticed during the development of the new 380mm bass unit for the Nautilus™801, and that was that a single large and stiff bass cone always sounded better than a number of smaller cones, even though they may well have had the same aggregated properties. One possible explanation for this is the concept of the production of a "coherent wave front". This will be produced by a single large very stiff cone, which can couple with the air in a uniform manner over the whole of its surface area unaffected by differences in loading over that area. This behaviour is to be compared with that of several cones which, even though they may be closely spaced, will still leave gaps of "uncoupled air" between them. The very stiff cone material of the large single driver, which is a thick sandwich of Kevlar® reinforced paper fibres with a very stiff skin, makes it less responsive to local changes of acoustic impedance or unbalanced modal pressures either behind or in front of the cone.

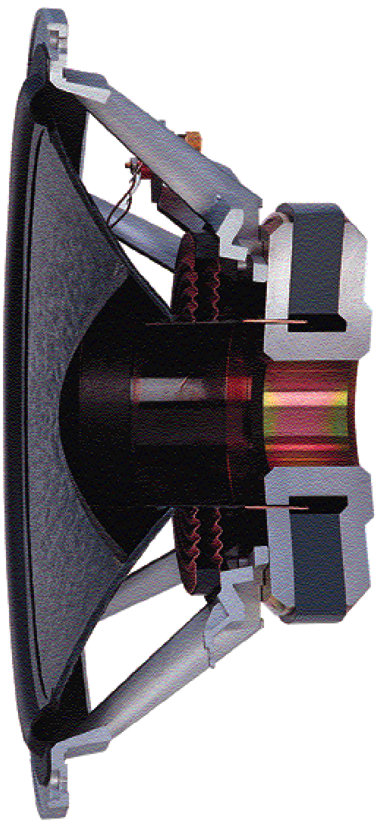
A "coherent wavefront" simply means there is either a constant or a smoothly changing phase relationship between neighbouring parts of the wavefront. So even if two drivers are relatively close together compared to a wavelength. Also, even if their contributions are equivalent to a single large driver, their different acoustic environments will mean that their outputs are slightly different, in terms of both amplitude and phase.

Furthermore, the air between the drive units is not being driven at all and this will translate into a change in phase across the resulting wavefront as the air tries to "fill in" the lost contribution. One can postulate that, at low frequencies, air can "spill off the edges" of the individual cones more easily in an array of small cones, which obviously have more edges for it to spill off, than from a single large one. For instance, two 12 inch drivers have a combined circumference of 1630mm whereas one 15 inch cone has a circumference of only 1037 mm. It is also interesting to note in this context that the radiation resistance and reactance at low frequencies of one 15 inch cone, is actually larger than that of two 12 inch cones, even though the area of the 15 inch cone is 0.02m² smaller than two 12 inch cones. This is because the change from a steadily rising radiation resistance characteristic at low frequencies, to a horizontal one at high frequencies, occurs at a lower frequency with one large driver than with two smaller ones (Fig 15). Mutual radiation impedance effects will redress this imbalance to some extent, provided that the two individual drivers are close enough together for one driver to acoustically load the other, though it will only be totally redressed if the array of small drivers produces a totally contiguous surface in all directions.

With a less stiff cone, localised changes in rear (or front) pressure due to modes, or the presence of obstructions which prevent the free flow of air, will mean that the cone will flex in response to this differential loading during each vibration cycle. This flexure, quite apart from upsetting the resulting wavefront, may well ultimately cause the demise of the drive unit itself which may eventually even fall apart due to the repeated stresses. This is especially true of light cones trying to compress the air in very small boxes, but the principle can obviously be extended to large boxes having rear obstructions – such as a Matrix™ indeed – which inevitably impede the free flow of large volumes of air which occur at low frequencies. All these effects are, of course, highly audible and conspire to remove the "punch" or timing effects which are often used to describe "fast" bass.



15 Radiation Resistance Curves



The New 15 inch Bass Unit

The rationale of using a single large pulp-coned driver for the bass of the Nautilus™801 was not immediately apparent. Early attempts at producing prodigious bass from domestically acceptable enclosures, utilised multiple drivers in slim Matrix™ cabinets.
The use of light alloy cones was also propounded at this stage; on paper they seemed to be the superior choice for a bass driver, working well within its piston region as on the Nautilus™. While the measured results looked very promising, subjective assessment failed to excite; the speed and timing of multiples were always inferior to a single driver of equivalent area, but aluminium cones larger than 8″ sounded distinctly metallic and muddled when mounted in Matrix™ cabinets. Parallel work on subwoofers involved the evaluation of various bass drivers for possible future use, the majority of these were pulp coned.

Drivers with light, rigid piston cones regularly won the listening tests for "slam" and timing but would often provide insufficient subjective depth because they were too light. Drivers with the best low frequency performance would often have a denser, more heavily doped pulp cone that was actually no stiffer than it's lighter counterparts.

In collaboration with our cone / surround suppliers, we developed a low density kapok cone pulp with a very high stiffness / mass ratio. It has a large proportion of Kevlar® and stiffening resin, and the required mass is obtained through slow deposition rather than high compression. This means that while the resulting cones have a fairly poor surface finish and take a long time to cure, their internal structure is an open network of long, undeformed fibres. This composition is enhanced by a surface impregnation of hard thermosetting resin, meaning that the overall structure is effectively a thick laminate, with a very high resistance to the static bending stresses caused by cabinet pressure over the whole cone, and the local variations of radiation impedance that can encourage the formation of audible bell modes and localised flexure.

When bell modes do occur, their high Q makes them particularly problematic and they can provide a clearly audible sonic signature of the cone material used at working bandwidths a couple of octaves below break-up. Coherent, axisymmetric radiation appears to be critical to accurate musical timing, but unless the enclosure is perfectly circular, or the piston perfectly stiff, there will be localised flexure and thus divergence from this ideal. The Achilles heel of large aluminium cones is that for a given mass, aluminium, or for that matter any solid sheet material, is much thinner and thus poorly equipped to cope with the pure bending moments that can arise in an application like a Matrix™ cabinet that does not present a truly even loading. The 12 inch Nautilus™ bass driver, in a lossy tubular transmission line with streamlined chassis legs and a very low crossover frequency, is perhaps near the dimensional limits of conventional aluminium cone applications.

Used in a non-circular reflex cabinet – Matrix™ or not – a bass cone will also be subject to a degree of largely incoherent cabinet reflections that can re-excite cone modes and / or, by coincidence frequency effects, re-emerge through the cone. The improved bending stiffness and damping quality of thick pulp provides higher transmission losses and thus greater isolation from delayed reflections that contribute to overhang. Incidentally, listening tests of different cones in an experimental circular test chamber, confirmed that under truly axisymmetric loading conditions, the minimisation of localised flexure and bell modes partially reduced the audibility of compositional differences.

Improvements to cone behaviour allowed the sonic contributions of the surround – always a compromise – to become clear. Soft, heavy rubber that had traditionally helped to damp cone modes was now serving only to blur timing and would visibly deform when cabinet pressure was elevated by the reflex port. With less internal loss required, the surround could be made lighter and stiffer. Objectively, this raised the frequency of the first surround mode and reduced its effect on the cone's pistonic output. Subjectively, we now had 12 inch bass drivers aligned almost identically to old Matrix™801 bass drivers, (and their legendary extension!), but with timing and control that was in another league.

It is perhaps important to note here that the vast improvements to bass performance resulted largely from a better appreciation of the mechanical requirements of pumping air, rather than controlling the travelling wave phenomena beyond the passband. "Pistonic" motion to an elevated frequency is often seen as panacea for a good bass driver, but it is no measure of truly pistonic behaviour from DC, where simply E and not E/r2 matters (where E = Young's Modulus and r = density).

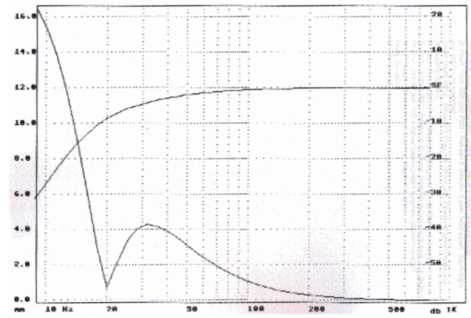
While the qualitative improvements were encouraging, the Matrix™ 801 replacement was required to have much higher sensitivity and headroom. The new bass drivers would play louder without audible protestation but, (without entering an explanation on alignments), you cannot change the laws of physics. Only more cone area and a bigger box would really suffice. Simply using two of the improved 12 inch units proved to be disappointing. Even though sensitivity and headroom were approaching the required levels, perceived scale, timing and speed were poor. The psychoacoustics of bass is sometimes difficult to quantify but, particularly at high levels, the body's contribution to space and time location means that the brain is even less easily fooled.

Two large drivers operating in free field conditions will produce a largely coherent wavefront comparable to a point source at a position equidistant from the two sources. Divergence from the central position will produce an anomaly in arrival times. Now place the same two drivers in an environment with at least one near field boundary at a normal to the array, the two drivers now experience different driving point impedances, their outputs will vary in magnitude and phase and the coherency of the wavefront is now further degraded. Take into account the effects of the inconsistent directivity of a large two element line source on a multimodal room environment, as well as a host of other reservations, and the arguments for using multiple drivers become less clear cut when large amounts of air are to be displaced in a domestic environment, and there is floor space for a single unit. A single circular cone is simply the most spatially efficient dynamic radiator available. This was part of the rationale behind the development of the 15 inch unit.

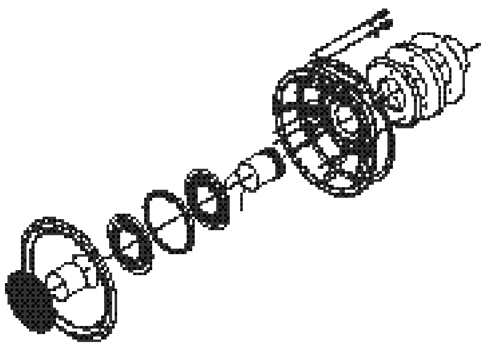
The new mid-range unit was larger and thus usable to lower frequencies, meaning that the inherently lower bandwidth of a 15" would be adequate even in a passive application. What would be crucial to the driver's success would be its ability to function as a rigid piston, and not a collection of cone areas disjointed by flexure, much like multiple drivers. A great starting point was of course the new thick pulp cones. These were mated to the obligatory stiff rubber surround and 4 inch diameter Nautilus™ specification voice coil, spider and motor system. The chassis was a long awaited, Nautilus™-inspired, alloy casting with ultra stiff aerofoil section legs, with due consideration paid to minimising rear reflections.

Initial results were highly encouraging. Measurements in a rectangular Matrix™ cabinet showed a very smooth response to beyond an octave from the proposed crossover frequency, with the first significant spikes an octave beyond this and well down in amplitude. The surround resonance also seemed to be highly controlled, this being another advantage of large cones; they have proportionately less surround area for a given excursion capability. Subjectively, the transient performance of the driver trounced the twin 12 inch unit set-up, despite a slightly smaller total radiating area. However, bottom end control was poor. More work was needed to get the correct alignment and it was clear that a single spider was not up to the job of controlling the required masses reliably.

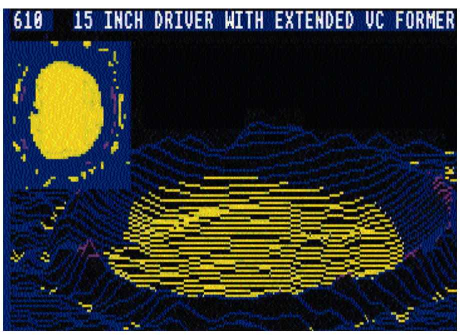
Cone mass was increased by thickening, (and thus further stiffening), the cone, the shore hardness of the surround material was raised, the coil was lengthened for even greater linearity and a double, mirrored spider construction was employed to ensure that axial piston motion was maintained at all levels. The resulting drivers had much of the breathtaking speed and depth that was required and is exhibited by the final design, but reliability of the spider-coil former assembly became a problem, as the limits of the new driver were enjoyed. Normally the card sleeves used to stiffen formers are perfectly adequate for bonding a spider to. But even with two spider glue joints spreading the load, the intense shear forces involved at high excursions, were sufficient to delaminate the surface of the card and allow the spider to shift and tear from



16 Nautilus™ 801 Bass alignment diagram (15" driver)



17 Exploded diagram of the 15″ bass driver



18 New Nautilus 15 inch Bass unit Full Cone Plot
Shows cone is a piston at 610Hz

unbroken joints. Massively stiff carbon fibre filament wound former sleeves, a totally new application, were employed to solve this problem and provide still greater rigidity and robustness at the heart of the cone/coil assembly.

Traditional cotton-phenolic spider material was also supplanted, its stability and strength proving to be insufficient. A new Nomex® fibre based material was employed. The large excursions at high currents meant that even the coil lead-out wires had to be scrutinised. Long term tests proved a special braid construction to be the only choice. Together, these measures help to ensure consistent performance and reliability.

Testing on the early 12 inch prototypes had shown the dust cap to have a significant influence on quality, with hard, oversized dust caps proving to offer the best performance with pure bass. However, the dust cap dome resonance increases in relative magnitude and decreases in frequency as it is enlarged, putting an effective ceiling on the size of dustcap used in the bass of a quality three-way system. While the isolated response of a deep-coned bass driver with a minimal dust cap may be impressive, it will degrade the acoustic output of a nearby mid-range source due to concavity effects, and is also troublesome to time-align, meaning that integration would otherwise be improved by a large dust cap.

In answer to these problems, the Nautilus™801 bass driver uses a unique construction, (the “mushroom” assembly), whereby the oversized carbon fibre dome is driven directly by the elongated carbon fibre former, as well as being attached conventionally to the cone. The problem dome resonance is now minimised and shifted far out of band. The acoustic response of the entire cone assembly is also improved; the first axisymmetric cone mode is raised slightly, due to improved coupling to the coil, and the impedance mismatch reflection from the cone / surround termination is reduced, as is the surround resonance, due to the staggered wave propagation times from the coil to the cone / dome junction. The overall resistance to localised flexure is of course vastly augmented by the new construction. Radially, the cone is now effectively linked by a triangulation to the coil former, and the cross-sectional stiffness of the coil former itself is transformed as it is properly capped. Another ingredient in the creation of hugely clean and tight yet powerful bass driver. (Figs 17, 18)

- Note this unit is only used up to 400Hz or so
- The cone is a piston until well above its operating band

ENCLOSURES

Tweeter

The unit incorporates straightforward Nautilus™ technology through the use of a tapered tube, filled with wadding attached to the rear of the unit and matching the hole through the pole (See appendix III).

The exponential profile has been designed to ensure that the cut-off frequency of the tube, is low enough to absorb all the energy in the operational bandwidth of the tweeter, but allowing a shorter tube than in the Nautilus™. It also allows the absorptive wadding to be packed loosely at the mouth of the tube and to become gradually compressed towards the end. This allows the sound energy radiating from the rear of the dome, to pass through the pole piece and into the tube without being reflected back up towards the dome. This variation in packing density ensures that the acoustic impedance is varied smoothly, and that there are no sudden changes which would cause such a reflection of energy. As the passband of the tweeter is similar to that in Nautilus™, the onset of cross modes in the tube is not a problem, occurring well above audibility in the human ear.

A secondary use for the tube is as a heat sink. The small dimensions of the magnet assembly, result in a lower thermal mass than that in the Matrix™801. Making the tube of aluminium and ensuring a good thermal bond to the magnet back plate, significantly reduced the operating temperature of the unit. When fed music from a 600W amplifier run just below clipping, the operating temperature is reduced by around 20C. In fact the tweeter was found to be capable of withstanding unclipped high frequency peaks from an amplifier rated up to 1kW, without the coil burning out.

The tweeter / tube combination is housed in an outer die-cast shell which defines the outer housing of the unit. The tweeter diaphragm only moves a maximum of 0.5mm. Therefore, it is crucial to isolate it from mechanical energy arising elsewhere in the system. To this end, the tweeter and tube are held in the housing with rubber O-rings, the housing in turn is decoupled from the mid-range enclosure below by the use of two isolator pads of Raychem Iso-Path™ material. (Fig 19)

The top isolator has been shaped to sit in the scallop of the mid-range head enclosure and cradle the underside of the tweeter housing. Raised ribs have been designed into this isolator to create maximum compliance at this interface, in order to absorb any energy transmission between the mid-range head enclosure and tweeter body.

The bottom isolator sits between the connector and the underside of the mid-range head enclosure to ensure that both the sections of the Molex™ connector are isolated from the mid-range head enclosure. The tweeter is allowed to 'float' free and reproduce the input signal without any external interference.

Mid-range

It was known that a simple Nautilus™ tapered tube would not work well over the bandwidth required of the drive unit (see appendix III), so other enclosure configurations were investigated. In particular, a sphere is well known to give a smooth diffraction-free exterior shape. Initial experimentation was directed at investigating how the performance was affected by the size of the sphere and how the drive unit was mounted in it. In all cases, the spheres were constructed from a GRP outer shell, lined on the inside with Fibrecrete.

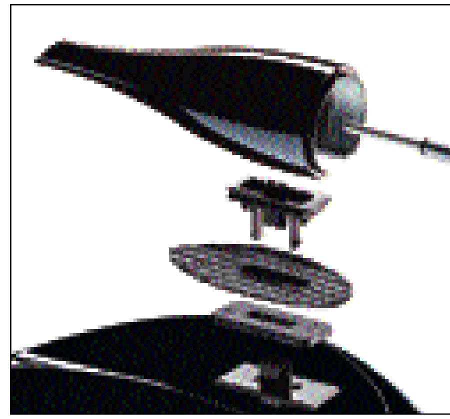
A sphere of around 300mm proved to be the best size. Any larger and imaging seemed to be impaired. Any smaller and the unit sounded "closed in". This is a difficult phrase to explain succinctly, but is akin to having traces of the effect obtained by cupping the hands round one's mouth when speaking. The same effect is heard if tubes are used beyond the frequency of the onset of cross modes and this gave a clue to what was happening in the sphere. In fact the way the drive unit chassis fitted into the sphere was found to have a similar effect which was only removed if both the inside and outside surfaces of the sphere blended smoothly with the rim of the chassis. This involved the sphere having a thin wall close to the unit and so to maintain overall stiffness of the enclosure, internal and external spherical profiles were offset, with the internal sphere centre being brought forward.

The sphere supports strong internal cross modes, evident in delayed response waterfall plots. These could only be damped by using copious amounts of wadding inside the sphere. Listening tests, however, revealed a loss of transparency with this approach, even when the waterfall plots apparently indicated a good result.

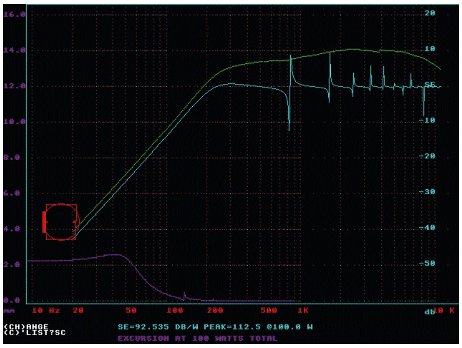
The combination of a sphere and an inverse horn was then investigated both experimentally and theoretically. Stuart Nevill undertook measurements of spheres closed and open, with and without absorption being present and then added a convenient Nautilus™ style rear tube which had an open end and was either empty or filled with absorption. Meanwhile Peter Fryer made some theoretical predictions of these systems to see which tallied with the reality.

To a reasonable degree of approximation a sphere can be modelled by a pipe of length and diameter equal to the diameter of the sphere and having the same volume. When this is done the simulated cone output agrees remarkably well with the measured output for a cone in a closed sphere or cubical box. The only differences between the box and the sphere appear to be in the frequencies of the harmonics.

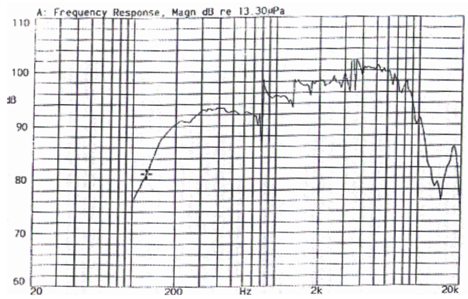
In a sphere these follow Bessel function zero crossings, which will be closer together than those for a pipe whose resonances essentially follow a sine wave's zero crossings. However, the form and amplitude of the response shape is remarkably similar! Exact analytic modelling of a sphere adds only a little extra accuracy to these features.



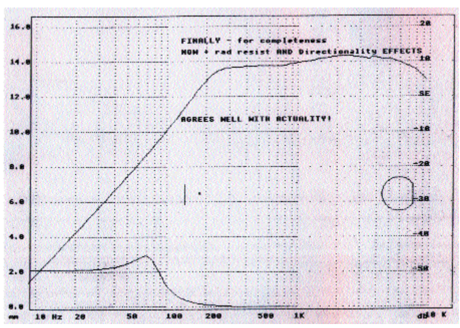
19 Tweeter Section Showing De-Coupling Detail



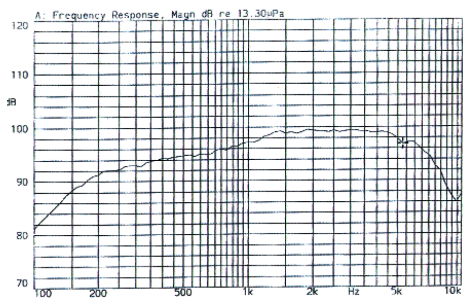
20 Box and Sphere Internal & External Effects



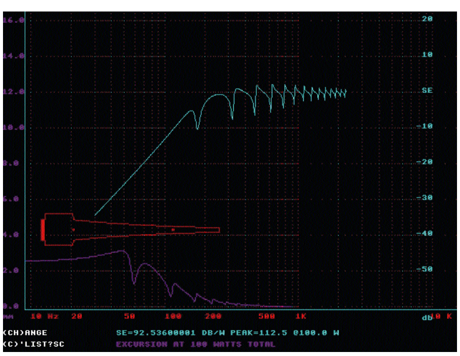
23 New Midrange Unit (measured)
In a Sphere – no absorption present – 1/24 Octave



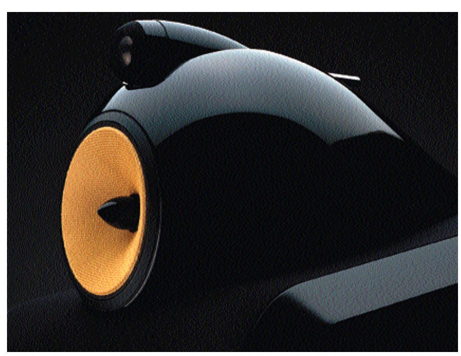
21 Prediction of simple sphere and piston responce including external acoustic effects



24 New Midrange Unit (measured)
In a Sphere with a Tube with absorption in Sphere and Tube



22 Long Rear Inv Horn + short Tube



When a tube is added to the back of the sphere, the combination might be expected to behave like two tubes in series. To model this involved the modification of existing modelling routines to allow for a change in diameter at the intersection of the two tubes. Previously there had only been the possibility of a smooth joint, with the end of the first tube having the same diameter as the beginning of the second tube. However, when this routine was used to simulate a sphere, (modelled as a real box), with a pipe attached, the theoretical results did not tally with the experimental results at all. The resonances in the real example corresponded only to those found in the pipe part of the combination, rather than to those of the box and the pipe taken separately, and to those of the combination of the two – which the theoretical modelling showed. (Figs 20, 21, 22)

Changing the modelling of the sphere to a simple lumped parameter volume but keeping a "proper" tube plugged into this gave much more realistic results which agreed closely with experiment.

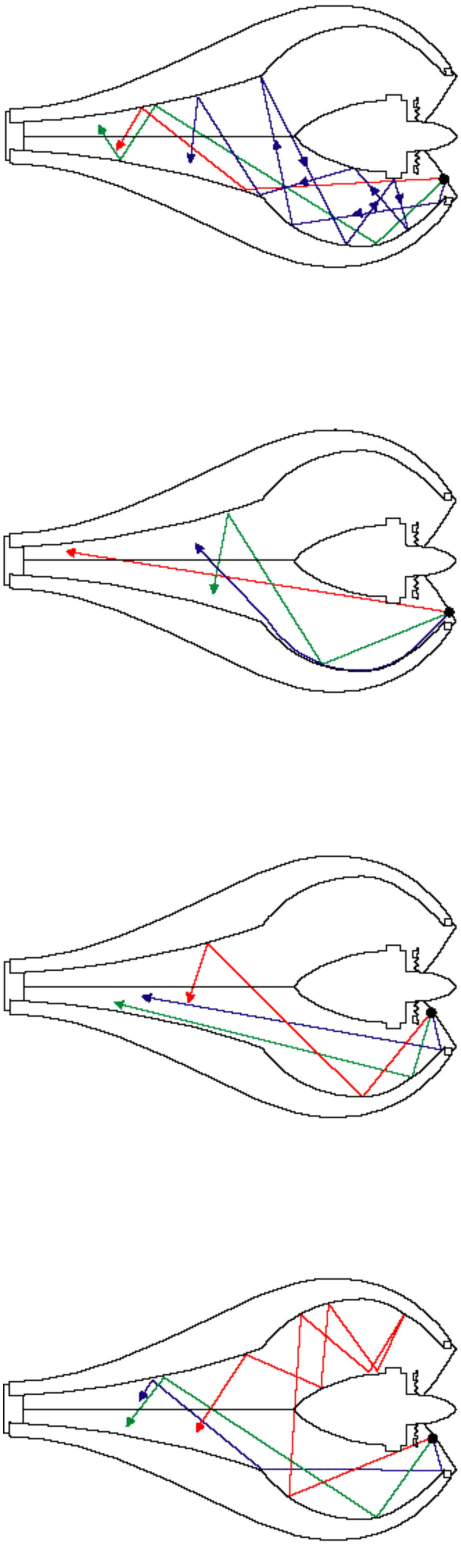
It seems that an empty sphere, on its own, at the rear of the speaker acts as a high Q resonance with Bessel function harmonics. However, as soon as a hole is cut into the sphere, particularly if it has the same diameter as the driver, and a Nautilus™ style tube is plugged into it, then the sphere resonance is greatly defused, as it were, and the sphere then behaves almost completely like a lumped parameter spring, even with no absorption present in either sphere or pipe! We are then only left with traces of the resonances due to the length of the pipe alone. Merely cutting a hole in the sphere, with no tube present, gives rise to a classic Helmholtz resonator, which has the usual low frequency resonance, but which does not stop the sphere's internal resonances at high frequencies to anything like the same extent.

The next step, having proved that the analytic modelling tallies remarkably well with the measurements (Figs 23 and 24), was to see how much the various dimensions of the tube could be pared down. Reducing the length to around 300mm and increasing the taper rate from -3 to -11 gives similar results to the long tube with minimal absorption present, though now the fundamental tube resonance frequency is shifted into the region where the "shorting effect" of the lumped parameter volume reduces its amplitude.

If the diameter of the large end of the tube is reduced to be smaller than that of the driver, the effect of this appears to be deleterious. This is because the acoustic impedances of the near end of the tube and of the matching section of the sphere both have to be matched to the impedance of the driver opposite to them in the sphere for maximum power transfer from sphere to tube. Also, If the taper rate is increased further, the horn cut-off frequency will be correspondingly increased, and with it the changes in acoustic impedance that this represents. So these changes will intrude into the pass band and so the beneficial effects may be expected to decrease. Furthermore, a closed-end horn appears to be better than an open ended horn, partly because the fundamental is at a higher frequency and is therefore shorted out more by the sphere. (Fig 25)

The combination of GRP and Fibrecrete used to construct the experimental enclosures did not lend itself to mass production and so an alternative material was required. It had to be stiff and heavy to minimise vibration levels in the walls and be mouldable into the complicated shape required. The material chosen – Marlan®, a synthetic resin – proved ideal for the application when used with the decoupling techniques described below.

It appears, then, that the combination of a sphere plus a tapering tube is greatly superior to a simple sphere or a tube on its own. Furthermore, when an easily achievable value of absorption is placed in both the sphere and the tube, closed at the far end, the remaining small resonance effects present with the empty sphere and tube combination are removed. The ability to reduce the amount of internal wadding maintains transparency on listening.



25 Some simplified vector representations of the radiation from a diaphragm visualised as a number of omnidirectional high frequency virtual point sources. This shows how the sphere/tube combination minimises reflections back to the diaphragm.
Note: lossy wadding is placed in sphere and tube and diaphragm's rear suspension disperses/absorbs coil area reflections

Bass

It was explained above why the use of tube loading is not applicable to the Nautilus™801 bass cabinet. Therefore the Matrix™ technology used in the previous 800 Series has been employed (see appendix II). Compared to the previous Series, however, the inertness of the cabinet has been improved by increasing panel thickness to 38mm, thus significantly adding to the mass. In addition, smoothly curving the rear surface greatly adds stiffness to the cabinet, and gives an interior shape which shows fewer internal acoustic resonance modes, since there are fewer parallel internal surfaces available to support the undesirable acoustic standing waves. The combination of an internal Matrix™ construction together with both a massive and stiff external "skin" makes the combination uniquely resistant not only to sound transmission from inside to outside, but also to intrinsic cabinet structural modes.

Bending thin wood laminations under heat and pressure is widely used in the furniture industry for the manufacture of chairs. However the ability to accurately match and join two such curved panels together without a witness grove and to maintain the accuracy required to fit the Matrix™ panels inside is beyond the capability of may suppliers. Special storage conditions for the raw laminations, with controlled temperatures and humidity are essential and sophisticated CNC 5-axis routing machines are required to shape the edges and cut-outs of the curved panels.

FlowPort™

It has long been realised that the movement of air in and out of tuning ports, which may represent quite a considerable physical displacement of the plug of air, often causes “chuffing” noises as the air interacts with the discontinuities found at the internal and external ends of the tuning port tube. These noises occur as turbulence is formed at the discontinuities. Even when the inside and outside ends of the tube are given smoothly rounded profiles, the problem is not totally cured, though it is mollified. The reflex port is a well established device to improve the bass response of a transducer in an otherwise sealed box of finite dimensions. As the power handling, excursion and linearity of bass drivers have steadily improved over the years, the limitations of a simple tuned port have become apparent. At low levels the behaviour of the air in the tube can be correctly approximated to a solid, massy piston bouncing on a known air volume and at a specific tuning frequency; a readily predictable and essentially acoustic problem.

At higher levels, aerodynamic effects become increasingly important and the associated losses mean that a given rise in bass driver input level will yield a smaller rise in clean port output level. This also means that the port is not reducing the excursion of the bass driver as effectively and the system will thus behave increasingly like a lossy sealed box design; the combined effect is known as "port compression" and can often create an ultimate ceiling to achievable bass levels.

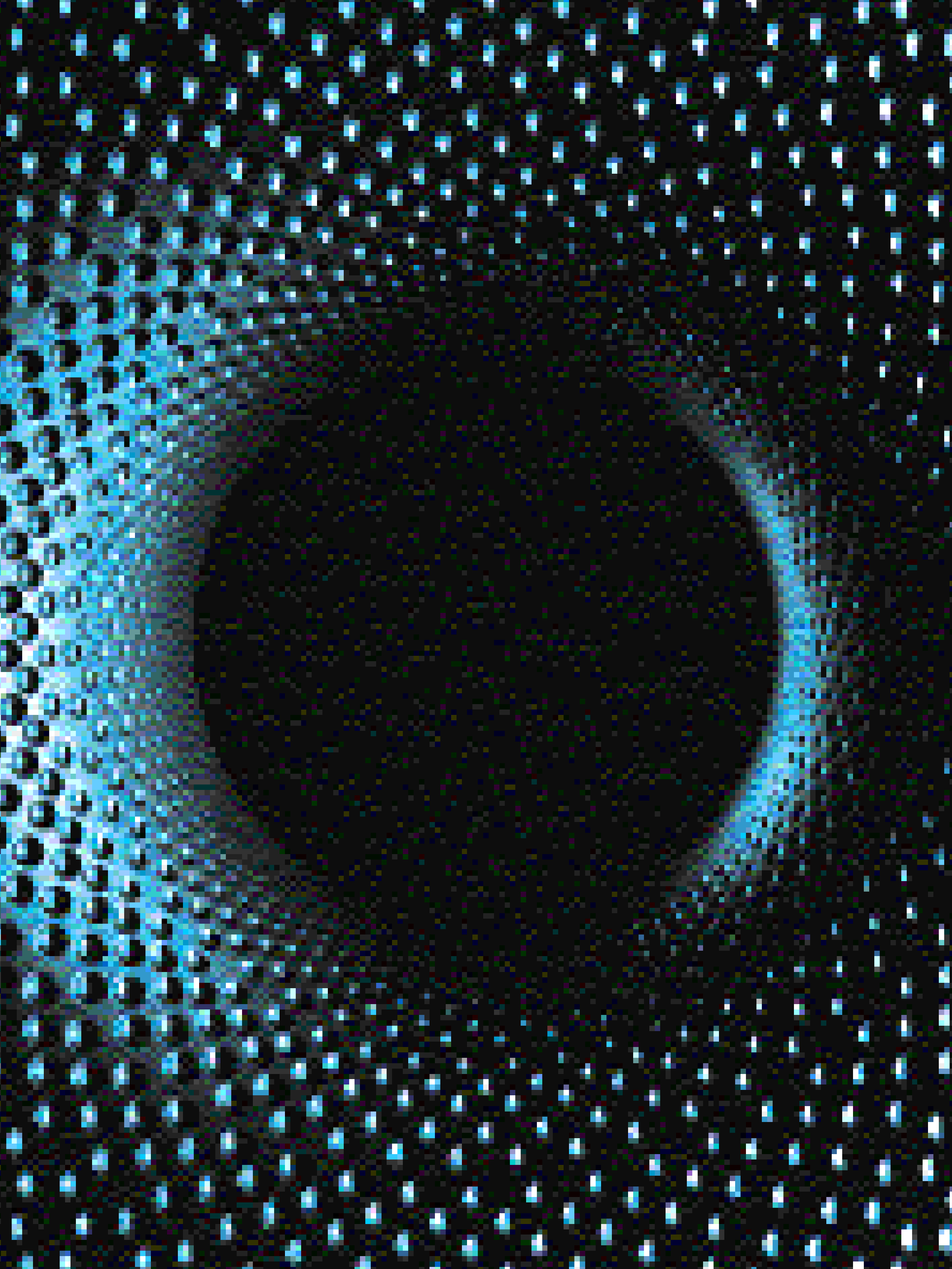
Well before any ceiling is reached, the energy losses associated with port compression cause problems; and it is the way energy is lost rather than the amount lost that causes serious acoustic problems.

At very low velocities, and with a perfect entry, air travelling through a real port tube will pass smoothly along streamlines that do not interfere with each other. Close to the walls of the tube is a thin boundary layer caused by skin friction with a relatively high velocity gradient. It provides the transition between the stationary walls and the moving air. Laminae of air rub against each other causing pressure drag through noiseless viscous losses. These are minimal at low levels but increase at a geometric rate in proportion to velocity. At high enough velocities, if the tube is excessively long and rough (or just very rough), the high shearing energies in the boundary layer can make it turbulent, which may be heard as windage noise, particularly because it can excite the organ-pipe resonances of the tube.

Far more serious problems occur when a laminar airflow tries to leave the tube at high velocities. If the curvature of the diffuser (flare) is too sharp, the minimal momentum of the air at the base of the laminar boundary layer is insufficient to pass the resulting sharp, adverse pressure gradient without stopping or stagnation. Slightly downstream, the pressure gradient (higher velocity with lower pressure to lower velocity with higher pressure) causes the flow at the base of the boundary to reverse and a turbulent eddy is created in the form of a rotating torus, (this is how smoke rings can be blown). The boundary layer now becomes the region that is between the eddy and the main flow but it has now separated from the surface of the diffuser. It tries to follow the pressure gradient formed by the turbulence but may form more eddies trying to do so, and so on.

The turbulent wake thus created is responsible for the “chuffing” noises that even gently flared ports can produce under some conditions. The separation can sometimes be so extreme that a turbulent jet can hit a listener at some distance from a speaker.

The aerodynamics of reflex ports is actually rather complex and somewhat unusual in that it involves alternating flow in two different pressure regimes (at and below port resonance), three octaves of the frequency spectrum (different systems have different tunings), completely indeterminate starting conditions and well over 100dB of level difference. Aerodynamics research into reflex ports at B&W is still in its infancy. Classical wind tunnel work is very difficult because the alternating flow makes a mockery of smoke trails. Recent work with Computational Fluid Dynamics has shown that ports are very difficult to model accurately, not least because of the large number of variables, but also because the flow regime is



influenced so heavily by small scale turbulence creation, which is less well understood than large scale fully developed turbulence (more is known about how aircraft stay in the air than how midge flies do).

Therefore, work has been largely empirical, using comparative rather than absolute benchmarks, because it is difficult to make reliable measurements of turbulent noise. Theoretical predictions of air velocities down the port were checked with a new Doppler measurement system, to establish the kind of flow regime operating around chuffing levels in terms of the Reynolds number, (a dimensionless indicator of turbulence levels). This showed that, with care, it was possible to maintain laminar flow down the port tube, but that air could detach from the flares at fairly modest levels. Simply making the flares more gentle would not guarantee silence.

Anyone studying aerodynamics will soon learn that turbulence is not always a problem. In fact, many aerodynamicists engineer turbulence to their advantage (indeed, some aircraft would not stay in the air without it). If a boundary layer is turbulent prior to the stagnation point it will be less inclined to separate because the base layer has increased kinetic energy. This means that the surface flow can be swept further downstream before pressure conditions stagnate it and the lower pressure in the layer that results from the higher velocities within the eddies adheres the main flow to the surface profile better. Thus, small scale turbulence can be used to delay the large scale turbulence caused by separation.

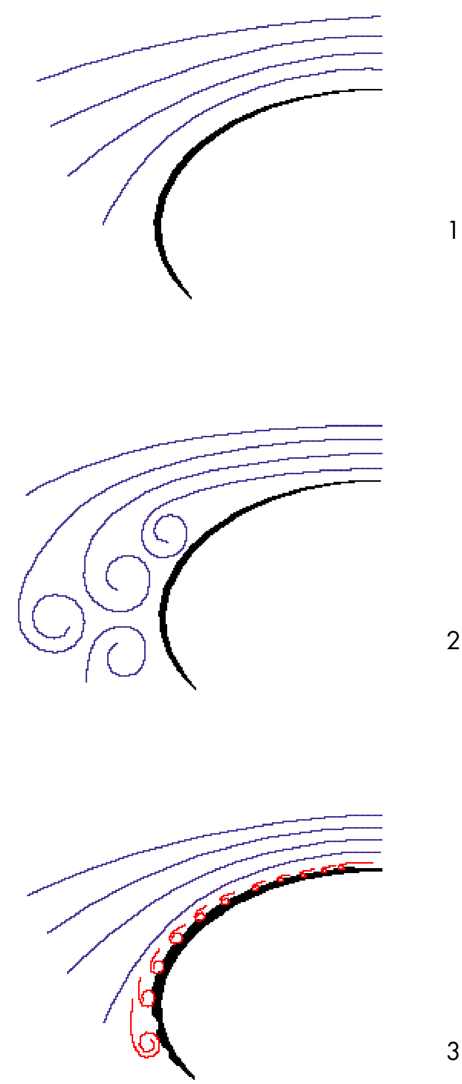
Artificially creating turbulence in the air moving down the tube can delay the onset of chuffing to higher bass unit input levels but, problem windage noise happens far earlier, especially as turbulent air is sucked back in to the port as the flow alternates. In addition, the thickened boundary layer effectively constricts the flow causing pressure drag and thus port compression; this constriction also alters the effective area of the port which in turn affects the Helmholtz tuning. Thus it is otherwise desirable to delay the onset of turbulent flow down the tube to as high a level as possible. A more optimal solution would thus be to use a smooth tube and limit artificial turbulence creation to the problematic stagnation area. (Fig 26)

It is quite easy to produce turbulence where it is needed; aircraft use vortex generators, (vertical strakes) ahead of separation points. These strakes project into the main flow and are very effective, but when the same technique is applied to port flares it creates too much windage noise at lower levels. Enter the golf ball; it can travel twice as far as an equivalent smooth ball because of its distinctive dimpled surface. The dimples are very carefully shaped to produce tiny separation points and favourable conditions for the creation of vortices within them. The ball is thus covered by a thin turbulent boundary layer that moves the separation point further round the ball. This decreases the ball's wake and hence its drag, and it was this technology that was used to improve the performance of the port flares.

Because a round port flare is axisymmetric, it was first thought that a series of rings with the cross section of a dimple might work (and be easier to prototype). However, the regular vortices formed simply became the new separation points and at lower levels there was audible windage noise because they were so abrupt. So real, pseudo random dimples were tried on the surface of the flare; these immediately improved the chuffing phenomenon as predicted, but there was still windage noise caused by deep dimples at the edge of the tube where flow velocities were highest. These were filled but at the expense of earlier separation levels.

By a process of experimentation, the size, shape and distribution of the dimples were refined to maximise headroom and minimise windage noise. Small, smooth dimples are thus used where velocities are highest and larger, more abrupt dimples are used where velocities are lower. This greatly refines the exit flow regime and also ensures that a minimum of turbulence is carried back down the tube when the flow is reversed. It was found unnecessary to make the dimples totally random over the whole flare, but as long as they are locally irregular, perceptible windage noise is incoherent and unobtrusive. In the case of the Nautilus™801, the port is downfiring, so more windage noise is acceptable and the dimples are optimised for maximum high level flow.

In use, the dimpled ports delay the nuisance chuffing noise to significantly higher levels. However, and perhaps of even greater importance, when large scale separation does occur the resulting turbulence is far more incoherent and thus less apparent. A reduction of 6dB in certain regions of the noise spectrum was measured, particularly around the problem organ pipe frequencies. Port compression is also decreased and the tuning frequency is more stable at higher levels.



26 Representation of streamlines exiting port flare.
1 Laminar Airflow following curvature of flare
2 Higher velocity turbulent airflow separates from surface of flare causing large scale eddy formation
3 Small scale turbulence due to dimples encourages laminar streamlines to remain attached to boundary

DECOUPLING

Having achieved excellent cabinets for each of the drivers independently, it is important that vibrations and radiation from neighbouring drive units do not leak into the cabinets of other drive units. There now follows a brief report, which illustrates what is important when it comes to theoretical calculations of decoupling.

The Effects of the Decoupling Spring on Speaker Performance

The effects of the decoupling spring between the magnet chassis and the speaker box have been investigated theoretically and experimentally verified.

The main results of this are hardly surprising – the resonance frequency of the magnet plus chassis on the decoupling spring stiffness should be as low as possible to avoid any undesirable effects on the speaker's response in its pass band. Also, if the resonance frequency of the magnet plus chassis on the decoupling spring, is the same as that of the speaker cone on the combined stiffnesses of its suspension and the air in the box, then there is no effect on the speaker's response. This latter effect would drift in production and is probably not a desirable solution to the decoupling problem. The effects of the spring losses on the response have also been investigated.

Method

As in time honoured style, an equivalent circuit was derived by inspection for the three masses and three springs and one force generator, all fully floating, which this complete system comprises. The fact that all three masses were fully floating contributed to the difficulty of finding the correct equivalent circuit, because a reference to ground is always required for equivalent circuit analysis.

Once a mechanical circuit had been derived, the "Voltage" electrical model was derived from that mechanical circuit with capacitors being equivalent to masses and inductors to springs. Then the dual of that circuit was produced to form the "impedance" model, (inductors = masses, capacitors = springs), which was analysed to produce the volume velocities, (i.e. currents), flowing into all the parts of the system These are used to give the output response taking everything into consideration.

Verification

A simple speaker system was built comprising of a magnet and a cone of the same mass as the chassis plus the magnet. This was suspended on rubber bands to comprise a fully floating system. The nearfield sound pressure output of the cone was measured

(a) with the magnet equal to the cone mass and

(b) with the magnet being much heavier than the cone (which is the normal situation)

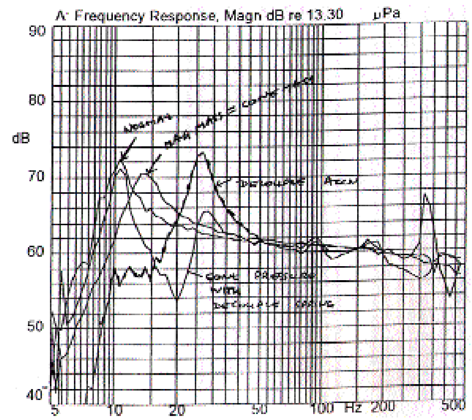
Next, the whole system was supported on a large lump of rubber and this constituted the "decoupling spring" of a more normal speaker system, (this test system had no box of course)

The acceleration at various places in the system was also measured with a small accelerometer. The frequency responses of the accelerations of these parts of the system are related to the frequency response shapes of the volume velocities of the same parts and thus to any radiation which may take place. Of course, parts like the decoupling spring will not actually do any radiating, but other parts like the speaker cone itself will. Thus accelerations of the parts of the system may be compared with each other, just like the acoustic outputs of some parts may be. (Fig 27)

Points to notice are:

- The shift of 1.414 in the basic resonance of the system when the magnet mass changes from "large" to the same as the speaker cone.
- The dip, followed by a peak in the cone output with decoupling spring present. The dip occurs at the frequency at which the magnet plus chassis resonates on the stiffness of the spring
- The output of the cone at high frequencies remains unchanged over a wide range of magnet masses.

This occurs because the voice coil produces a force which it exerts equally in both directions, i.e. into the cone and into the magnet system. They then exhibit an appropriate acceleration according to F=M*A with F being the same in both directions and the magnet and cone masses determining the accelerations and hence the radiation from both directions!



27 The results show a high degree of correspondence between the measured system and an equivalent theoretical system

The effect of the decoupling spring losses was then investigated. As expected when the losses are total, (i.e. Q decouple very small), the system defaults to a normal speaker plus chassis plus enclosure. As Q decouple is increased the dip and peak in the response become more and more pronounced. The effective Q of the lump of test rubber was approximately 20, which gave the correct ratio between peak and dip in the response.

Next, a more normal speaker system was investigated and families of curves were produced varying such parameters as: Q decouple, the frequency of the magnet resonating on the decoupling spring, and so on. From these curves it becomes apparent that either the decoupling resonance should be very low, or it should be the same as that of the cone mass resonating on the combined stiffnesses of its suspension and the air in the enclosure.

Finally, the volume velocity being fed into the cabinet was plotted – this of course is the reason for decoupling the magnet in the first place and it should be reduced as much as possible within the pass band. Curves are shown with the decouple frequency being 1Hz and Q decouple being 20. It is seen that very little volume velocity is being fed into the cabinet. Comparing this when Q decouple is small and F decouple is 20 shows the size of the relative problem that proper decoupling will solve.

So decoupling is effective if it is correctly applied, the big danger being that the resonance of the magnet on the decoupling spring may fall within the pass band. If that is the case, there will be a peak and a dip in the response the size of which depends on the decoupling spring losses. Of course, the larger the losses the less effective the decoupling will be.

Use of decoupling in the Nautilus™801

The Nautilus™801 follows the original Matrix™801 and Nautilus™ systems in that it uses extensive vibration isolation to minimise cabinet resonances and driver interactions. The techniques used have drawn from B&W research work which has shown, both theoretically and practically, the benefits of decoupling. This work has also shown if, how and where it is best applied as well as the problems that can arise if performed incorrectly. Good isolation between components is relatively easy to achieve on the laboratory bench, but engineering it into a rugged product would have been far more difficult without the application of a new material.

It is vital to ensure that the fundamental spring/mass resonance of any reliable decoupling scheme is below the operating frequency range of the speaker drive unit. If this is achieved then any damping control is unnecessary and even undesirable. To achieve the lowest resonance the spring must be soft and the mass must be high. The trouble has always been getting a soft spring to support a high mass within tolerances. The mid-range driver isolation of the Nautilus™801 was a particular challenge and one which drove the search for new materials.

Despite the huge mass and stiffness of the mid-range enclosure, decoupling the mid-range driver from it produces huge reductions in cabinet vibration. However, there must be a complete seal which fits within the required acoustic shape and point contacts, (fixing screws are undesirable because they unnecessarily excite higher order modes).

A tensioned rod system was devised to hold the driver against the cabinet with complete axisymmetry. It was then necessary to find a very compliant material to mate them. Foams were easily soft enough but unable to support a load long-term. Heavily plasticised materials could be found with the low Shore hardness required but leached or crept over long-term tests and were often too lossy to be effective across the band. Unfortunately, a fairly large gulf separated the hardness of these materials from the usable, homogenous, stable rubbers; even the softest silicones.

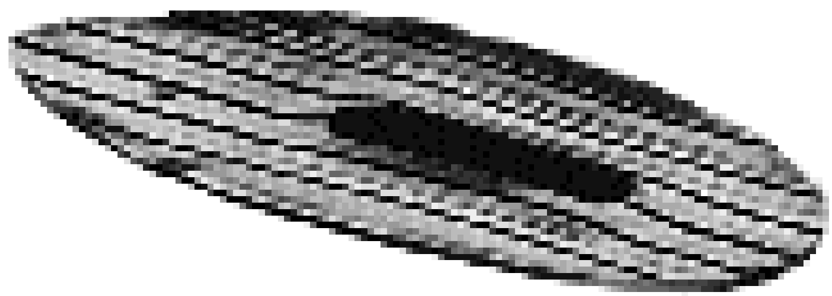
Armed with an idealised specification, the purchasing team succeeded in a more extensive search for super-soft rubber suppliers. Raychem Corporation in America had long been specialising in cross-linked dielectric gels but had just developed an ultra-soft, injection mouldable thermoplastic rubber using their cross-linked gel techniques. This material, since named IsoPath™, is effectively a liquid suspended with complete stability in a polymer molecular matrix. Its mechanical properties are a reflection of this structure. It is very compliant in shear and stretch but has poor compressibility when confined, much like a liquid but unlike conventional elastomers. And while it is freely elastic at low frequencies, it exhibits more viscoelastic behaviour at much higher frequencies, (it has a high tan delta product), giving it a useful transmission loss.

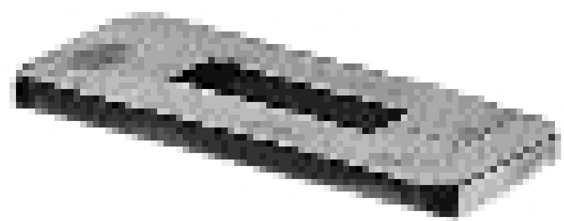
IsoPath™ is thus employed in an L-section gasket on the rim of the mid-range chassis. The mass is supported by the thin edging which provides shear freedom and the tension is held against a thicker compressive region, the unconstrained edge of which follows the critical internal profile of the mid-range cabinet. IsoPath™ is also employed at the other end of the tensioned rod to provide a fully floating assembly with the required single degree of freedom at a frequency well below the pass band. Additional mass is coupled to the mid-range magnet to further lower the resonance frequency and reduce chassis displacement, this being of additional importance because the cone impedance matching relies on the chassis as a virtual ground. The high loss of IsoPath™ at these elevated frequencies is thus invaluable, because it provides damping control of any structural modes in the chassis itself.

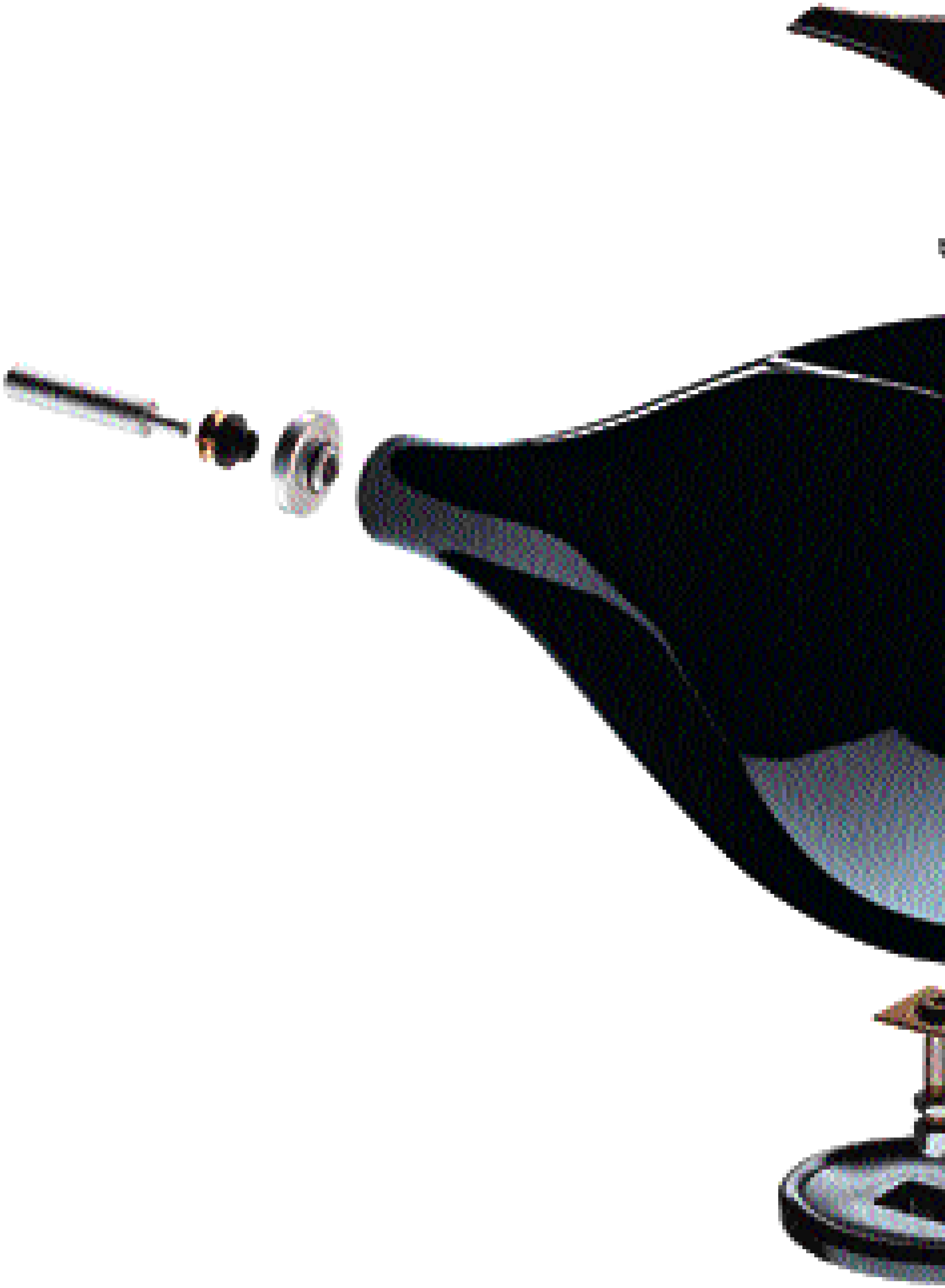
To ensure that low-frequency vibration from the bass driver does not excite the mid-range decoupling resonance, and to shift cabinet on cabinet resonances below the bandwidth of the bass driver, the weighty mid-range head is supported on a bed of IsoPath™ at its base and at the rear of the tailpipe. A similar scheme is employed for the tweeter/head interface. The tweeter motor is satisfactorily isolated from the heavy cast housing by conventional rubber "O" rings, but the entire housing is decoupled by a sculpted IsoPath™

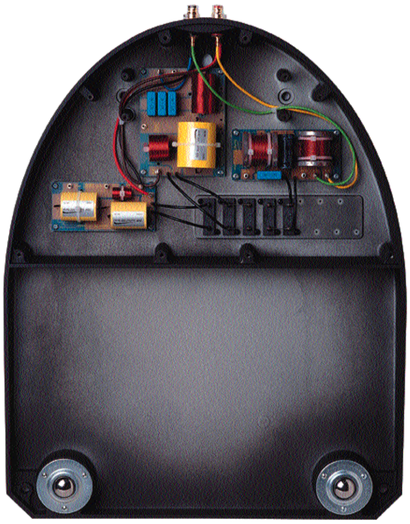
interface. Isolation is provided by shear freedom, but a ribbed surface profile is employed to improve compressibility and thus ensure that even the rocking modes are well below band, (yet kept marginally above the mid-range cabinet's fundamental resonance frequency).

It will of course be noted that the bass driver is not decoupled from its cabinet. This design decision was made fairly early in the development stage because the complications necessary to do the job correctly would have been impractical and perhaps unwarranted. Unlike the mid-range and treble drivers, the bass driver reacts against a large air stiffness, so any orthodox compliance scheme would involve losses or in-band resonances unless the mass of the magnet was unreasonably high, (in passing, note that limited magnet excursion has no effect on the acoustic output of a mass-controlled diaphragm, as in the mid and treble schemes). However, decoupling cannot be ignored because it will occur in a mechanical system at some frequency whether it is desired or not. If that frequency cannot be brought below band then it is best to raise it above. To this end, the hugely stiff bass chassis is bolted firmly into the rigid Matrix™ cabinet, which ensures that the first chassis/cabinet resonances are just above band and that there is minimal lost motion between the two. The Matrix™ construction of the cabinet has long been proven to be acoustically inert, so any vibration energy imparted on the cabinet has no serious panel modes to excite and is quickly dissipated. Lost motion to the acoustic environment is also minimised because the cabinet and plinth assembly are massive enough to become a virtual earth for the driver to react against.









CROSSOVER

Nothing shows up the importance of listening more than in the "tweaking" of the crossover network and the selection of the components chosen for use within it. It is clear that slight changes in balance between the drive units of as little as one 0.1dB are in fact clearly audible, and may affect the perception of such low level effects as "ambience" and "spaciousness", let alone the solidity and positional exactness of the stereo illusion itself. Slight trends and slope differentials between drive units of a similar magnitude are audible. The addition of even tiny amounts of extra distortion from inferior capacitors or dust iron inductor cores may mask some low-level effect that is now being revealed by intrinsically better drive units.

We have been surprised by the differences in sound possible using similarly specified components. For example, early listening tests with different makes and types of resistors resulted in the use of thin-film types in the Nautilus™801. These require substantial heat sinks and thus screwed directly to the underside of the cast aluminium plinth in which the crossover is housed. Capacitors have also proved interesting in their effect on sound quality. Although it is often difficult to pin-point the exact reason for preferring one to another, there is usually no doubt as to which is best. A good example is the use of multiple parallel radial lead types in preference to a single axial lead component. This phenomenon also extended to the use of polypropylene by-pass capacitors in parallel with electrolytics for high capacitance values.

By far the greatest improvement over previous models has been through the use of air-cored inductors. Whereas the distortion does not appear to be compromised by the use of different types of capacitor, the elimination of inductor cores has a marked effect on measured distortion. Even at low levels, there is a 3:1 reduction in the crossover's contribution which, combined with the low-distortion mid-range drive unit, gives unheard-of resolution. At higher levels, of course, the absence of core saturation means an even greater advantage.

Fine tuning by listening has shown that we are capable of hearing localised changes in level of a small fraction of a decibel, thanks to the enhanced resolution of these speakers. Differences in the ancillary equipment such as amplifiers, pre- amplifiers and cables are significantly clearer now than ever before, so we have gone to great lengths to make sure that these do not cloud the selection of speaker components, (eg we have typically used a passive potentiometer in place of a pre-amplifier, although we have the best available).

Final approval of components was not accomplished until the speaker had been tried in a number of different locations, and with a wide range of musical input. All this makes the selection of mean drivers very important, and also requires a greater consistency in production, to take full advantage of the system's performance.

Each filter section of the crossover is mounted on its own individual printed circuit board, which enables physical separation to reduce magnetic interaction between inductors. This maintains the benefits of bi-wiring. The inductors are designed for high power use without saturation, all but the largest values being air cored types. Except for very large values, capacitors are of the polypropylene dielectric type to maintain the highest possible signal quality to the drivers. In the 801 and the 802 the crossovers are mounted in the underside of the cast aluminium plinth and the resistors of the crossover are screwed directly to the plinth itself, which act as a large heatsink.

Each system can be bi-wired, the bass units are fed to one pair of terminals and the mid-range and tweeter to the other. The terminals themselves are manufactured from high quality gold plated brass alloy and are fully insulated in clear plastic.

Performance

Loudspeaker performance is represented graphically in several ways and Figures 28a-e illustrate some of the common parameters. However, these graphs go only part of the way to fully describing the performance. Properties such as imaging, coloration, dynamics and bass "speed" or "slam" are difficult to describe graphically and can only be properly assessed through a series of listening tests.

It is interesting to note that, although the Thiele-Small bass alignment of the Nautilus™801 is virtually identical to that of the Matrix™801 Series 3, the two sound quite different. To directly represent this difference in terms of the usual linear and non-linear distortion mechanisms would be difficult. That is not to say that such measurements would not show up the differences, but mapping the measurements to the perceptual experience is not a well developed science.

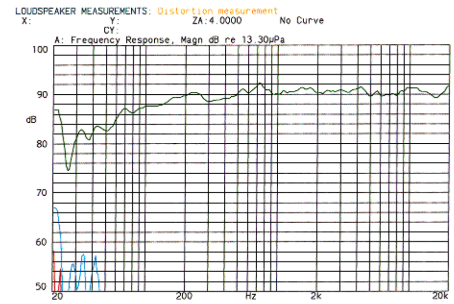
The imaging capabilities of the speaker are extremely well developed. The usual tendency for the central image to collapse to the nearer speaker when the listener is positioned off-centre is not present, although the image does, of course, move to one side. An image between the two speakers is present even for listeners positioned outside the area between the speakers. The representation of depth is excellent where the information is in the recording and the speakers are well spaced from the rear wall.

The low levels of coloration and distortion, especially with regard to the mid frequencies, create a very transparent transducer that enables the listener to readily perceive differences in ancillary equipment and the quality of the recorded programme. This degree of performance is maintained to high sound levels – the target output level of 120dB spl at 1m has been achieved – which has the effect of making the speaker sound less loud than it is actually playing, as distorted sound is perceived to be louder for the same measured level.

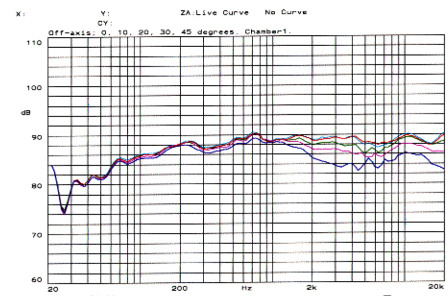
Credits

The creation of the Nautilus™801 is the result of the work of a dedicated team of designers and engineers and the input of the following is recognised:

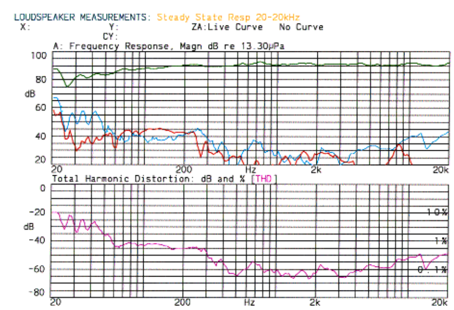
- *Steve Roe BSc*
  Development Director
  (Overall project responsibility)
- *Dr Peter Fryer BSc DIC PhD*
  Research Director
  (Laser techniques, diaphragm and enclosure analysis)
- *Stuart Nevill BEng*
  Research Engineer
  (Drive unit, Flowport, Sphere/tube and decoupling design)
- *Dr Gary Geaves BSc PhD*
  Research Engineer
  (Finite Element Analysis)
- *Graham Landick BSc*
  Research Engineer
  (Magnet systems and drive unit design)
- *Dr John Dibb BTech PhD*
  Senior Development Engineer
  (System design)
- *Steve Pearce*
  Development Engineer
  (System design)
- *Doug Standen*
  Design Engineer
  (Mechanical Design)
- *Morten Warren*
  Native Design
  (Industrial Design)



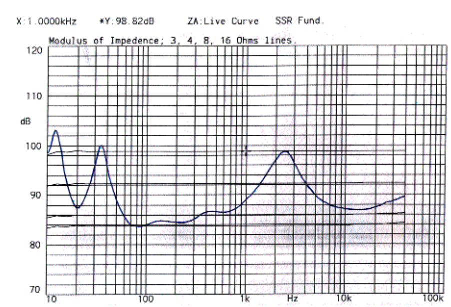
28a Nauitilus™801 frequency response



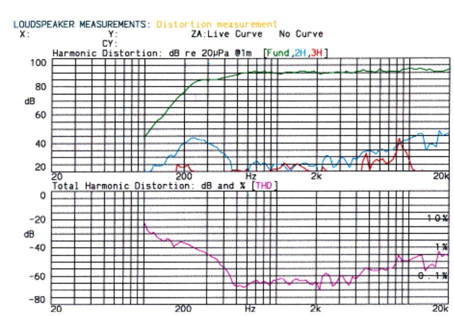
28b Off-axis responses



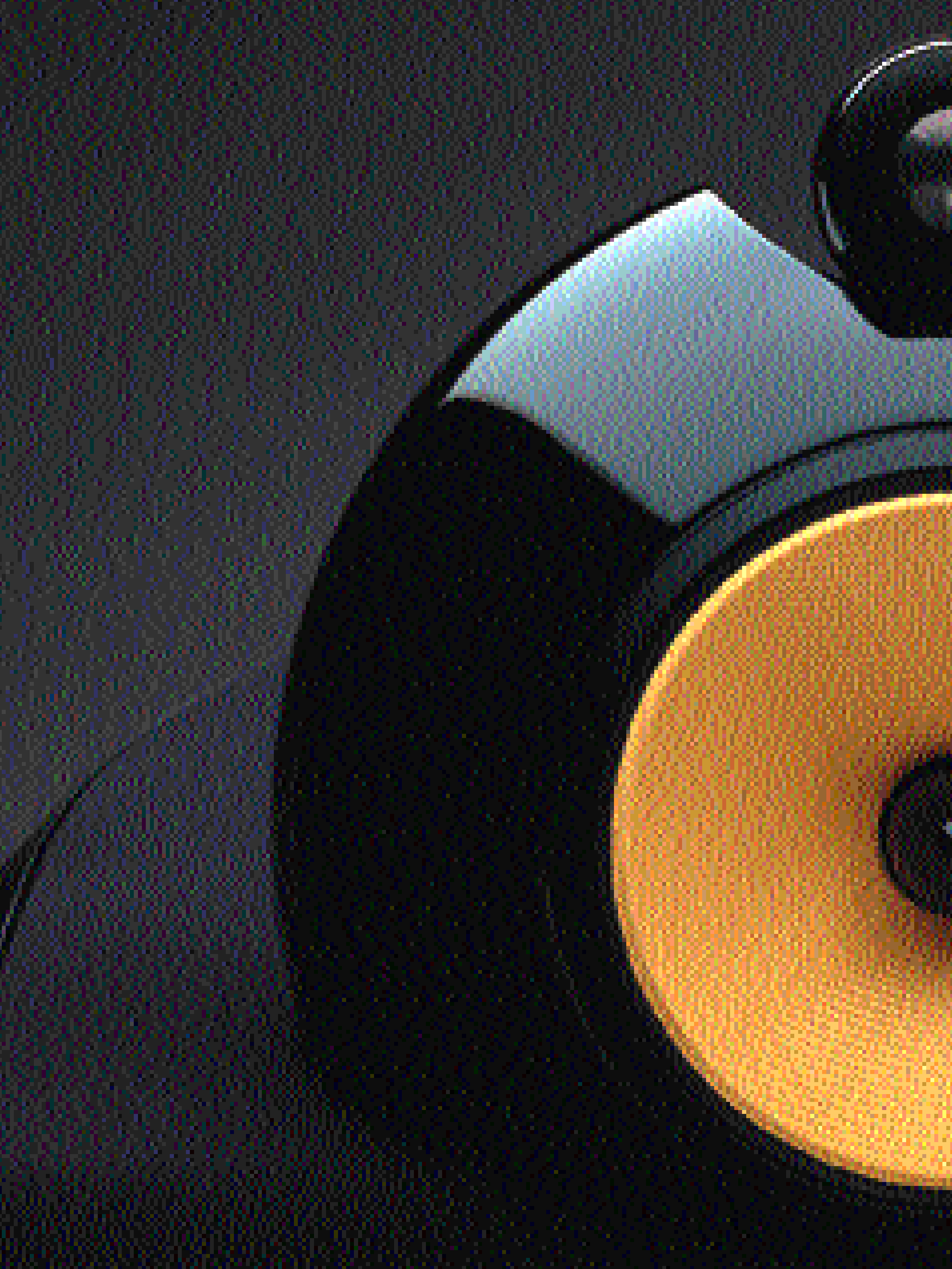
28c Distortion



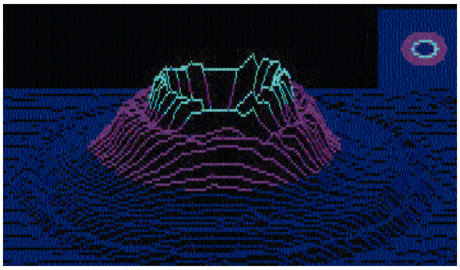
28d Impedance



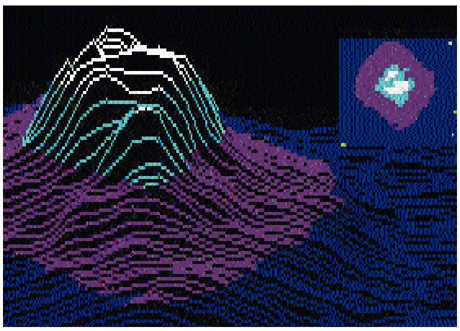
28e Loudspeaker measurments



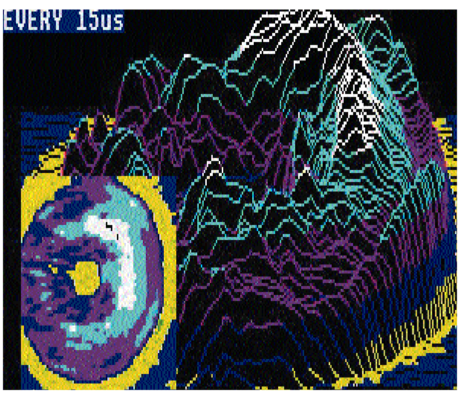
Appendices



29a



29b



29c Whole Cone Surface Impulse Progression Plot

30

APPENDIX I
THE USE OF WOVEN KEVLAR® AS A LOUDSPEAKER CONE MATERIAL

Introduction

We shall examine the difference in behaviour between a woven Kevlar® cone and a plastic one of similar dimensions.

Plastic Cone Impulse progression plot (Impulse Plot)

(Fig 29a)

We see that

- Impulse is fed into the voice coil in middle.
- A fast Compression Wave then speeds towards the surround
- A Slower Bending Wave follows on behind
- The Bending Wave is reflected from surround like sea wave hitting a sea wall
- Reflections are rotationally symmetrical and
- carry on back and forth for a long time
- The behaviour is totally rotationally symmetrical at any time
- No dust dome was present

Note: Because the waves are rotationally symmetrical, they all add up the same way in all radial directions. Whenever half a wavelength of the bending wave fits into the space from voice coil to a reflection point, (usually the surround and/or the chassis), we get a standing wave otherwise known as a resonance forming. This is always bad news for this cone as these waves, particularly at their standing wave frequencies, give out energy long after the original impulse has gone, and badly colour the sound.

Original Kevlar® Cone Impulse progression plot (Impulse Plot) (Fig 29b)

- The Fast Compression Wave moves at different speeds in different directions
- The Cone appears to be acoustically square, even though physically round
- Reflections at the surround and/or chassis, do not occur at the same time in all directions, and:
- Reflected waves do not form the same standing waves in all radial directions
- If properly designed, standing waves in one direction cancel out those in another!
- The Central (floppy) Dust Dome moves a great deal!
- After a time the surface of the cone appears to go more or less random in its behaviour

Note: Compression waves are much faster in Kevlar® than in Plastic, and there is hardly a trace of the bending waves that were present in the Plastic. Most of the energy is carried outwards by fast compression waves. Because they arrive at different times all around the circumference, any standing waves they form are not rotationally "coherent" and so do not add up in phase to produce troublesome resonances, in fact they tend to cancel out instead. The plot indicates in effect, that they are being reflected more or less at random from the surround into the cone, there to effectively cancel each other out.

Note: Arguments applying to matching bending wave characteristic impedances and the effect of this on any standing wave formation, apply equally well to matching compression wave characteristic impedances.

Plastic, Old Kevlar®, and "Surroundless" Kevlar® Cones Compared (Impulse Plot) (Figs 29a,b,c)

- 29a shows impulse moving from voice coil to surround on Plastic cone
- 29b shows impulse moving over Old Kevlar® Cone
- 29c shows impulse moving over surroundless Kevlar® Cone

Note Particularly

- Old Kevlar® is much faster than Plastic
- Surroundless Kevlar® is much faster than old Kevlar
- Old and New Kevlar® appear to be square acoustically, though round physically

The waves in the "surroundless" Kevlar® case are much faster than those in the original Kevlar® case we believe because the mass of the surround in the original case was sufficient to slow down the rising edge of the impulse response by simple F= m * a considerations. The mass of the surround was holding back the movement of the cone and more particularly the outer regions of the cone were being restrained. The superior rise time exhibited by the "surroundless" Kevlar cone translates into a greatly extended frequency response which is actually usable up to between 7 and 8kHz, a very high frequency for such a large cone, and then because of the "symmetrical resonance cancellation effect", merely falls off smoothly above this frequency.

Impulse Progression across a diameter "surroundless" Kevlar®
(Fig 30)

This single slice impulse progression plot shows:
- Lack of reflection of bending wave on LHS of the diameter
- Some reflection of the bending wave on the RHS of the diameter

The left hand side was half wave between the bias and the warp direction, where the impedance is matched, whereas the right hand side is some way from the correctly matched condition showing some waves being reflected. Note also the very fast passage of the initial wave from the voice coil to the surround.

Frequency whole cone plots
Plastic Cone (Frequency Plots)
(Figs 31a,b,c – upper image)

- Whole Cone Frequency plots from 0.2kHz to 10.2kHz
- Many resonances are seen – all axisymmetric i.e. the same all the way round the cone!

Note: The first resonance is seen at 210Hz – the lowest frequency displayed. This is where a half wavelength of the bending wave fits into the distance between the voice coil and the chassis thus including both the surround and the cone. This is the fundamental of the surround resonance which is thus the first harmonic on the standing bending wave. It is hardly ever noticed in frequency responses as its effect is usually minimal, even though the surround moves more than it ought to.

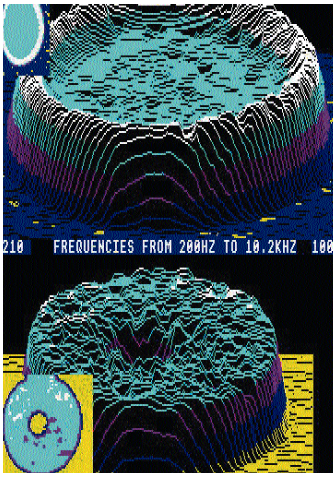
The next resonance above the fundamental appears to be only in the surround, when a circular dip is evident in the middle of it. This next major resonance, which is usually called "the surround resonance", occurs in fact when a whole wavelength of the bending wave fits into the whole distance between the voice coil and the chassis. The surround, being much floppier than the cone, appears to move a great deal more than the cone and since the standing wave is a whole bending wavelength, the surround goes backwards whereas the cone goes forwards only a great deal more so.

After these two initial standing waves or resonances, the cone and the surround exhibit more and more axisymmetric regions going upwards or downwards, until the whole cone is filled with concentric circles and the cone has ceased radiating at all. This sequence of resonances, which used to be called "cone break-up modes" merely occur when whole numbers of half wave-lengths of the bending, (or other types of waves), will fit into the total distance available from the voice coil to the chassis. After the "surround resonance" we see the cone itself progressively alters more and more as the frequency is increased and the errant behaviour appears to be no longer confined to the surround alone, even though in fact, it was never confined in this way even at low frequencies.

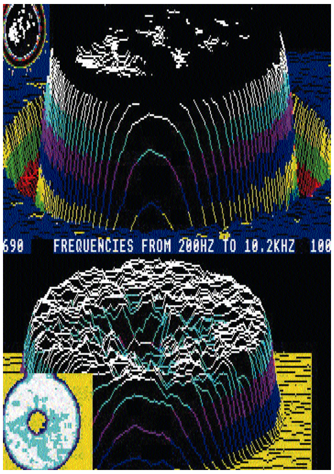
These resonances greatly colour the sound from speaker cones that
a) have floppy surrounds that radiate in their own respect and
b) that are totally axisymmetric thus allowing anything that happens in one direction to reinforce anything happening in another direction. This sort of thing, of course, does not happen with correctly designed Kevlar® Cones which are not axisymmetric.

New 801 Kevlar® Mid Range Unit
(Frequency Plots)
(Figs 31a,b,c – lower image)

- The unit shows no surround resonances at all
- It is a perfect piston up to about 3kHz
- The "break-up" pattern above that firstly shows the edge "giving way" slightly



31 Kevlar® and Plastic Drivers Compared (Frequency Plot)
a The Plastic driver shows many resonances all through the band and beyond

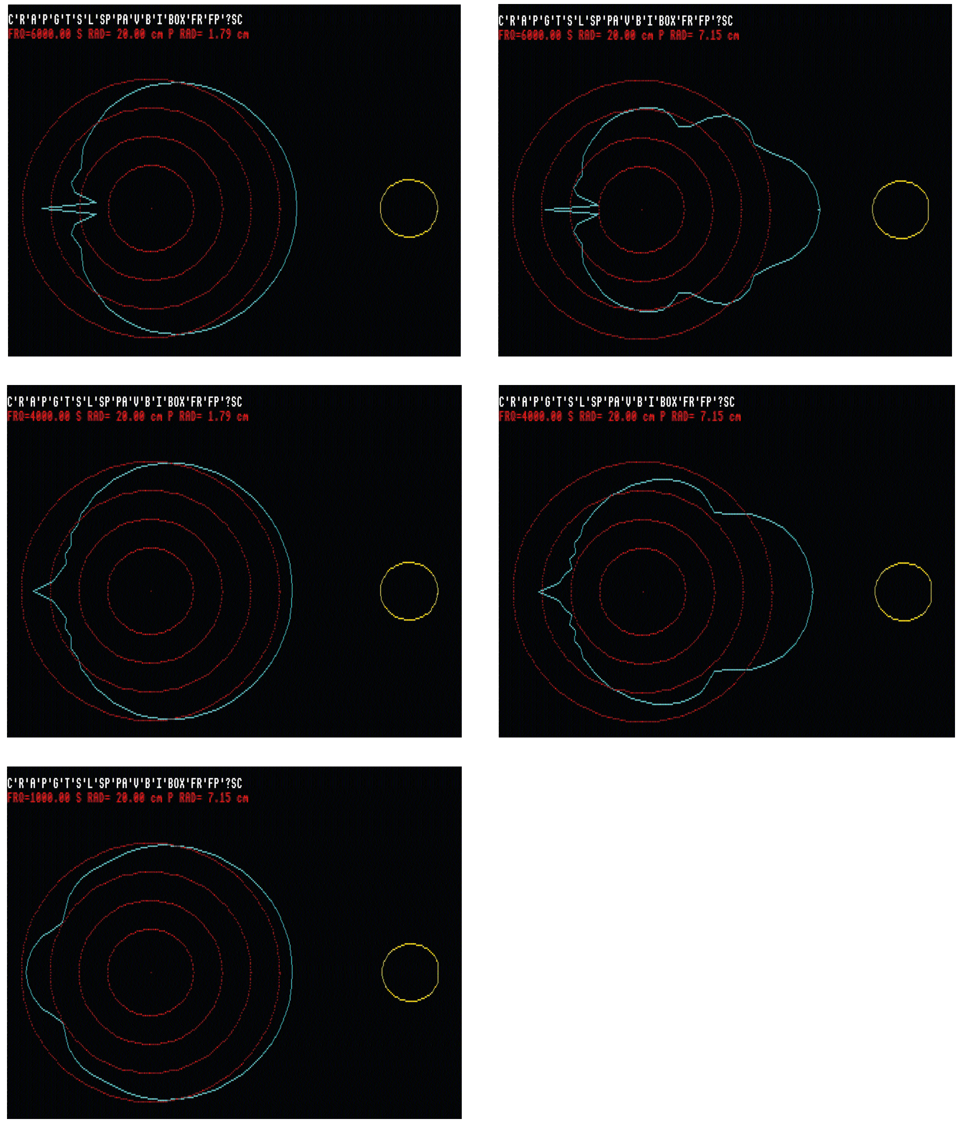


b The Kevlar® Driver's edges begin to give way above 3kHz

c The Kevlar® central area decreases steadily above 3kHz maintaining a constant directivity pattern

After that the unit shows “four fold symmetry” this means that:
Sectors going up and down in the outer regions cancel out leaving:
Radiating central area decreasing in size with frequency, so that the directivity is largely constant above 3kHz because:
The radiating area is roughly halved by 4kHz and
The radiating area is roughly reduced to one quarter by 6kHz (Fig 32)

Theoretical plots are shown of the directivity of cones the size of the new mid range unit and ones having one half and one quarter the radiating area at 4 and 6kHz respectively. Measurements of the frequency responses at various degrees off axis, confirm that the output there is upheld approximately as predicted.



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APPENDIX II – MATRIX™ CABINET CONSTRUCTION

The phenomenon of cabinet radiation has been recognised for as long as loudspeakers have been used in boxes. In theory, the function of the box is to act as a perfect obstruction to the acoustic field generated within it by the rear radiation from the drive units. Even vented enclosures rely on the panels containing, without deflection, the pressure element of the resonant action of the port.

Of course real materials have finite loss and stiffness and hence will deflect in an acoustic field, and the problem has been to minimise this movement by judicious use of the available materials within the constraints of economics and ergonomics.

In general, at low frequencies the stiffness of the walls dominates their behaviour, while at high frequencies it is their mass which rules. Between these two extremes they interact in a resonant manner which can grossly magnify and time smear the transmission at certain frequencies. A situation which is rendered tolerable by resistive losses or damping

In general one is trying to maximise all these variables, although at times it can make more sense to ensure that a resonance is outside the frequency band to be used, than it is to keep it subdued with the use of mass or damping.

The stiffness of a panel for a given mass can be increased dramatically by curving it. Hence axially loaded tubes and spheres have long been recognised as the most efficient users of materials, though in our rectilinear society they have usually been relegated to more exotic designs.

The mass of panels can be increased simply through the use of dense material. Bricks, lead and sand layers all offer increasing attenuation with frequency, but these are all definitely for the DIY enthusiast.

In the real world, cabinets were made of wood with various degrees of panel bracing and damping, like bituminous mats and suchlike, which also helped increase the mass. By increasing both the mass and the stiffness, the lowest point in the curve corresponding to the minimum transmission loss may be brought up to reasonable levels. (Fig 33)

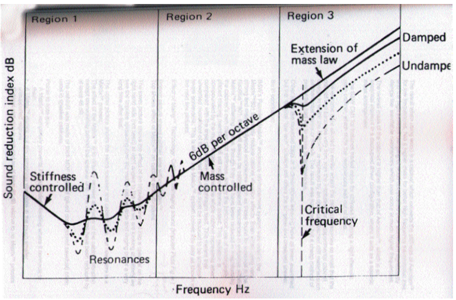
In 1983 Celestion revealed their SL600 which for the first time used Aerolam™ panels in which two sheets of material are separated by a honeycomb structure. This structure makes the best of a given mass and thickness of material by ensuring that all forces act in the plane of its sheet components. This greatly increases the transmission loss at low frequencies by extending the stiffness region of the curve upwards in frequency. The result was a stiff rather than light enclosure, with high frequency resonances near the coincidence critical frequency, which required damping with thin pads. Aerolam™ is, however, rather costly and difficult to work with, and the lightness of the panels makes the mass law part of the curve lower than usual, which allows high frequencies to be transmitted through the cabinet panels.

The Matrix™ approach to the problem is to extend the honeycomb principle of Aerolam™ to the full width of the enclosure. The walls are then being supported across their full area and, in the limit, require no bending stiffness at all, the displacement being entirely dependent on the longitudinal stiffness and acoustic velocity in the honeycomb.

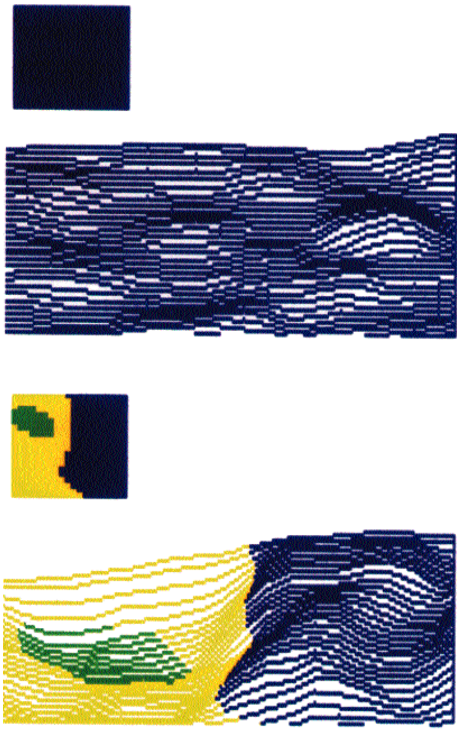
To provide the support of the three pairs of walls, an orthodox structure is preferable to the honeycomb. A wine box inspired the final structure of the Matrix™ which has now become a standard feature in all the top of the range B&W models.

The result when using wood is a cabinet which exceeds the stiffness of Aerolam™ for low frequencies, has a higher mass for better high frequency transmission loss, and the high inherent damping of wood composites over aluminium, significantly damps the inevitable resonances which now occur at much higher frequencies. The overall mass of the cabinet is higher than that of Aerolam™, which is an essential element in a decoupled driver configuration, where it acts as a seismic mass arrangement.

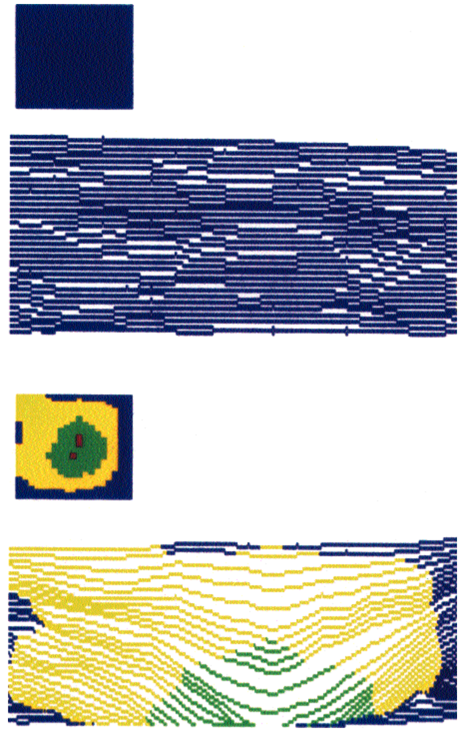
The B&W CM1 for instance traded mass for stiffness by using a rigid phenolic resin cabinet, which shifts the resonances well out of band.



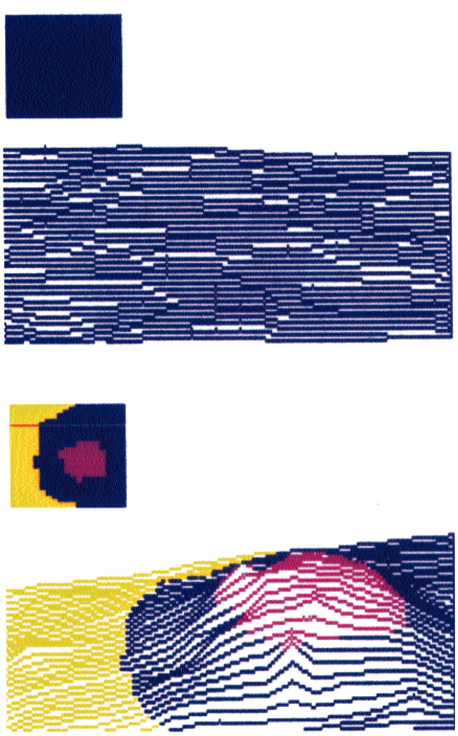
33 Mass Law + Stiffness + Coincidence



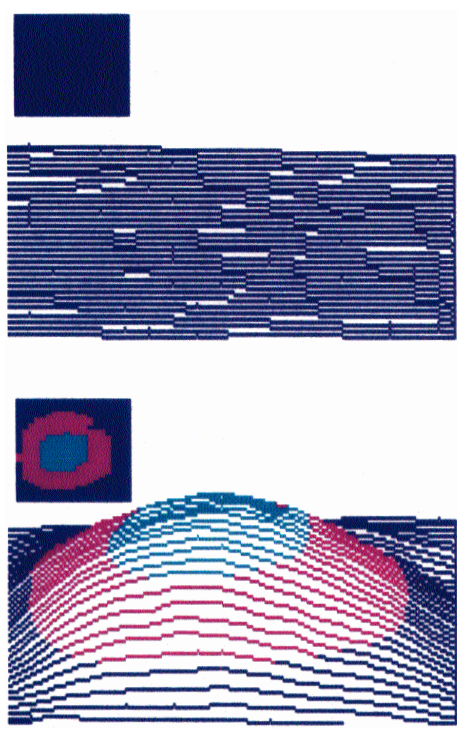
34a Whole Cabinet side Impulse Progression Plot, at 480 u-sec



34b Whole Cabinet side Impulse Progression Plot, at 1740 u-sec



34c Whole Cabinet side Impulse Progression Plot, at 2010 u-sec



34d Whole Cabinet side Impulse Progression Plot, at 3330 u-sec

Other ideas like moulded trays or 3-dimensional weaving have all been examined but the original "low-tech" solution still offers the simplest and cheapest answer.

So in conclusion:

B&W's Patented Matrix™ enclosure reinforcement system was used in the Matrix™801 to greatly reduce cabinet vibration effects. Even when using very strong and massive plywood enclosures such as those found for instance in the Nautilus™801, there will always be cabinet panel resonances in evidence without Matrix™ stiffening. These, though they may have been reduced in magnitude by the cabinet's own mass and stiffness, and by the judicious use of strategically placed bracing members, will still radiate to some extent. This radiation is compounded by the much larger area that the whole cabinet represents when compared with the area of the drive unit diaphragm itself. This means that unwanted movement of the cabinet's surface must be reduced to the smallest levels achievable. This is where the Matrix™ system comes into play as shown in the next few pictures.

Effect of Matrix™ Technology.
Cabinets damaging sound
(Impulse Plots, Figs 34a,b,c,d)
Lower image shows cabinet side without Matrix™.
Upper image shows effect of Matrix™ on cabinet side vibrations.

The lower picture shows the effect of an impulse being fed into a speaker on the right hand side, (not seen), at right angles to the side we are looking at. Initially we see the impulse spreading over the visible cabinet side and ringing on for a very long time after that.

The upper picture shows the same speaker and cabinet side, but now there is a Matrix™ present. The effect of the Matrix™ is to virtually remove all traces of the cabinet vibration except immediately behind the speaker unit. The vibrations have been reduced by at least 45dB.

The cabinet was made of thick MDF material, (glued compressed sawdust), often used in speaker cabinets.

Points to note are

- A Matrix™ completely prevents the box sides vibrating where it connects to them
- A Matrix™ damps out any remaining vibrations in the much smaller unsupported regions
- Cabinet radiation is reduced by at least 45dB relative to no Matrix™ present

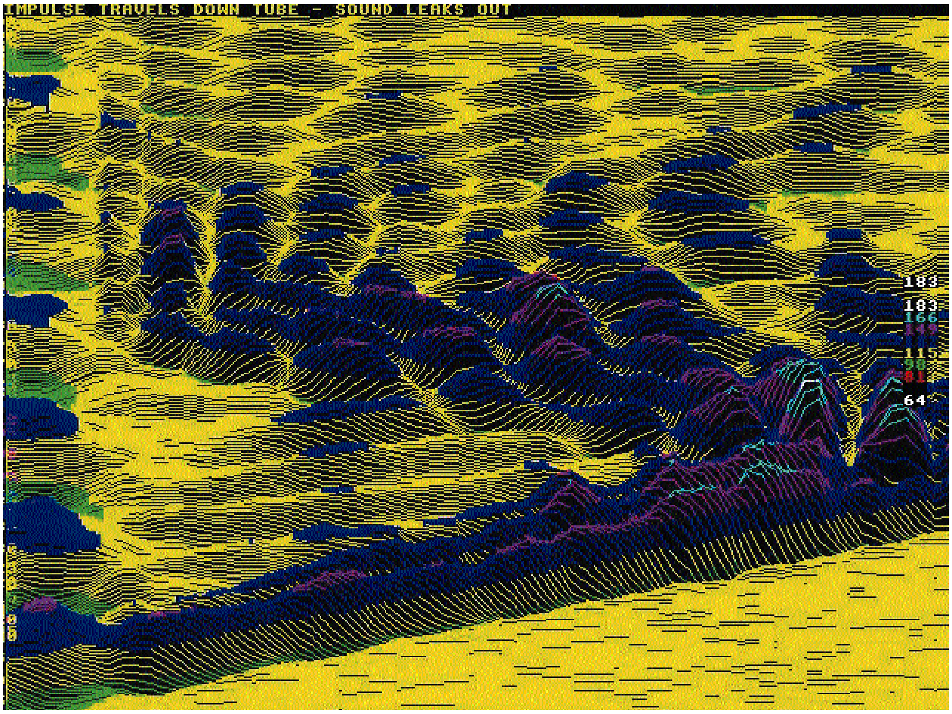
Effect of Insufficient Transmission loss inside to outside (Impulse Plot, Fig 35)

The plot shows the transmitted vibration of the side of a long tubular cabinet in response to an input impulse fed to a drive unit at the left hand side. So we have distance from left to right, and time moves away from us up the page.

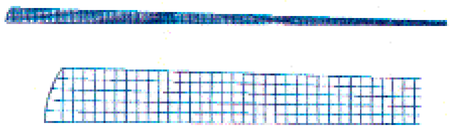
Points to note are:

- The initial impulse moves from the speaker at the left, along the tube from left to right
- It is reflected at the right hand side and moves back again along the tube from right to left
- The wave speed can be determined from the
- acute angle of the initial movement to the x axis
- This is the speed of sound in air, not in the material of the box

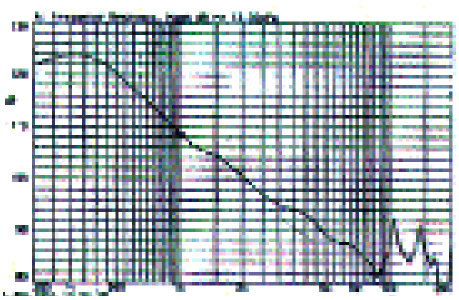
This clearly shows that the box is not massive enough to stop the sound from leaking out through the material of the box itself. Note that a vibration in the chassis of the driving speaker can be seen at the left hand side.



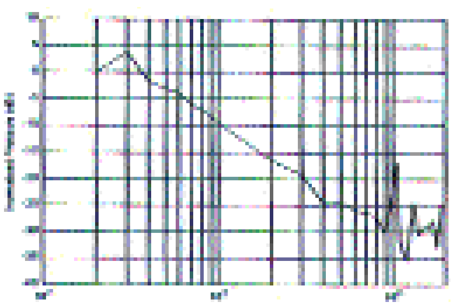
35 Single Slice Impulse Progression Plot – Box Side



36.1 The Finite Element Model
Upper – the entire mode, Lower – close-up of the driver end



a



b

36.2 Interior SPL responses **a** measurement and **b** simulation

APPENDIX III
TAPERED TUBE THEORY

Finite Element Analysis *Abstract*
The Finite Element Method was used to analyse the interior acoustic field of an inverted horn system. The finite element model was verified by comparing simulated and measured sound pressure level responses at an interior point. Contours of equal pressure phase were used to visualise the interior acoustic field. It is concluded that the frequency range over which the system can be used, is restricted by an upper frequency which is dependent on the diameter of the horn.

Introduction
A common source of distortion in loudspeaker systems is internal acoustic resonances of the cabinet. High pressures can build up behind drivers at internal resonance frequencies of the cabinet, which affect the movement of the driver, thus colouring the far-field sound. A solution to this problem is to use an inverted horn. A carefully designed inverted horn offers a smooth change in acoustic impedance which, when used in conjunction with an absorbent material, can result in a cabinet structure free of internal acoustic resonances.

In this paper results of Finite Element Analysis (FEA) of a prototype inverted horn designed for use in conjunction with an upper mid-range dome is reported. Initial listening tests of the prototype revealed distortion occurring at approximately 10kHz. The Finite Element Method (FEM) was used to isolate the cause of this distortion.

The finite element model is described in '*The Finite Element Model*'. In '*Verification of the Finite Element Model*' the model is verified against the sound pressure level (SPL) frequency response measured at an interior point. Finally, in '*Analysis of the Internal Pressure Field*' the interior sound field is analysed in more detail.

The Finite Element Model
The FEM is a mathematical technique that can be used to produce approximate solutions to partial differential equations. By utilising the FEM it is possible to carry out computer simulations of structural and acoustic systems. This paper is not intended to explain the FEM in detail. For detailed descriptions of the FEM applied to loudspeaker design see [Refs1+5], for example. The main point is that by applying the FEM, a virtual prototype can be constructed on a computer that can reflect the real world with sufficient accuracy to be useful in the design and analysis of loudspeaker systems. In this case a commercial FE package called PAFEC FE [Refs 6-7] was used to carry out the analysis.

The real prototype was composed of a 43mm diameter aluminium dome driver and an exponentially decreasing horn of length 680mm. The interior of the horn was filled with polyester fibre wadding, a material commonly used to absorb unwanted sound. Figure 36.1 shows the FE model of the inverted horn prototype. Note that the model was axisymmetric.

The model was composed of both structural and acoustic finite elements. Structural finite elements were used for the horn sides and the dome driver. The structural break-up of the dome driver was not modelled in detail and rigid motion (constant acceleration) was imposed. Detailed modelling of this component was considered unnecessary as initial measurements discounted structural break-up to be the cause of distortion occurring at 10kHz. Acoustic finite elements were used to model the wadding. (Fig 36)

Verification of the Finite Element Model
To gain confidence in the accuracy of the FE model a measurement of the spl response at an on-axis point approximately 50mm from the back of the driver was made using a microphone inserted through the side of the horn. This is compared to the simulated spl response at the same point in Figure 2. Clearly the general trend is the same in both cases – after an initial rise the responses decrease with frequency to approximately 10kHz where a spike occurs.

The differences between the measurement and the simulation probably arise because of:
- Small differences in measurement position.
- The simulated response being taken at a point whereas the microphone essentially averages over an area.
- Simplification of the geometry in the FE model.
- Uncertainty as to the acoustic impedance of the wadding.

However, the correspondence between the simulated and measured spl response is generally good and it is therefore assumed that the whole of the interior acoustic field is modelled with sufficient accuracy.

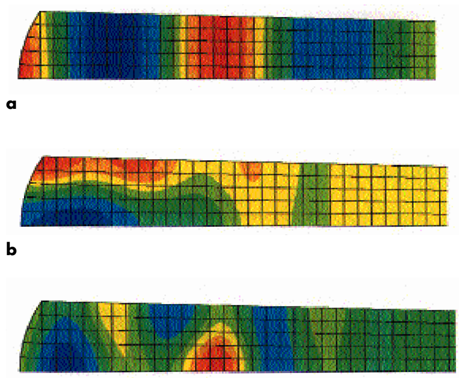
Analysis of the Internal Pressure Field
To gain insight into the nature of the interior acoustic field, animations of equal pressure magnitude and equal pressure phase were made. Such animations are a powerful means of visualising the interior acoustic field. Contours of equal pressure phase especially are significant because wave fronts move in directions normal to these. Snap shots of equal-pressure phase animations are shown in Fig 37.

At 5kHz it is clear that the inverted horn is behaving as expected – the direction of propagation is down the tube. However, at 10kHz there is clear evidence of a resonance across the diameter of the tube. This ties in well with the spl responses shown in Fig 36.2 a-b, where a spike is visible at approximately 10kHz. At 11kHz, it is apparent that the main direction of propagation is down the horn but with a component across the diameter (this is very clear in animations).

Concluding Discussion
The problem cross-diameter resonance could be removed or the effects reduced by using:
- a more effective wadding
- a tube placed in-side the horn
- a flatter radiator which would not excite diameter modes so strongly.

Each of these cases were analysed using the FEM and all resulted in either removing or reducing the 10kHz spike. However, none offer a realistic solution to the problem.

It is clear from the results presented in this paper that inverted horns are only effective up to a certain frequency dependent upon its diameter at the throat. This is perhaps an obvious result but by using the FEM the effects can be quantified and clearly visualised.



a
b
c

37 Simulated results showing contours of equal pressure phase **a** 5kHz, **b** 10kHz and **c** 11kHz

Sound Propagation down Nautilus Tubes
(Phases & Frequencies)
a Top plot shows a plane wave moving smoothly down a Nautilus tube at low frequencies.
b Middle Plot shows the wave at the cut on frequency of the first order mode of propagation. This occurs as a resonance across the tube width.
c Bottom plot shows the propagation above the cut on frequency. Energy then moves down the tube as a combination of plane waves and first order modes.

Note: all pipes and ducts will allow plane-only waves to pass down them below a certain frequency which depends on their cross sectional dimensions. Above a first critical cut-on frequency waves can also propagate in a zig zag fashion along the tube. The angle of the zig and zag changes with frequency and is at right angles to the length of the tube at the cut on frequency – which therefore shows up as a cross mode. Higher order modes of propagation also have their own cut on frequencies, which will also show up as cross resonances at higher and higher frequencies. For best effect therefore, Nautilus tubes can only be used up to the cut on frequency of their first higher order mode.
Ref: Book, Mechanical waveguides by Martin Redwood.

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APPENDIX IV
FINITE ELEMENT ANALYSIS

Introduction
Much work at B&W over the years has resulted in proprietary Finite Element and Boundary Element code capable of accurately predicting the vibration and acoustic behaviour of axisymmetrical shapes using Finite and Boundary Element Analysis. Papers listed (refs) include several on this topic. B&W proprietary code runs at more than 10 times the speed of any commercially available package and allows in-house optimisation packages based on "simulated annealing" to find the global minimum of any target function we may like to specify. The result of this is that a computer may be left to search through the whole relevant design space to find the best model that will fit our requirements – and it will not be trapped in any local minima along the way.

When non axisymmetric modelling is needed, commercial Finite Element packages have to be used and one of these, PAFEC, allowed the modelling of the total acoustics of normal Nautilus rearward inverted horns, though not the coupling of a lumped parameter speaker model at the near end. The tube was therefore excited with an "ideal" forced diaphragm for this exercise. As expected, this shows up the onset of the first higher order cross mode of propagation as a resonance across the mouth of the horn (ie where the speaker is situated). This graphically illustrates that rearward Nautilus horns may not be used on their own above this cut-on frequency, as this transverse mode is clearly audible through the speaker diaphragm.

The following reports from Gary Geaves cover the analysis of this phenomenon:

B&W Loudspeakers Ltd and Computer Simulation
Computer simulation based on the Finite Element and Boundary Element methods has been widely exploited in many diverse scientific and engineering applications. The Finite Element Method was developed in the 1950s to aid in the design of aircraft structures. Since then it has been applied to structural, thermal, electromagnetic, fluid flow and acoustic problems. In many industries, such as the automotive, it has been long regarded as an essential design tool. However, it is only relatively recently that sufficient computer power has become readily available and the underlying mathematical techniques sophisticated enough to be of use in the design of loudspeakers.

Engineers at B&W were quick to spot the potential of computer simulation in the design of loudspeakers, first becoming involved through collaborations with academic institutes in the mid 1980s. At that time, if one wanted to carry out simulation of an acoustic system, it was necessary to develop and code the algorithms from scratch. For this reason, B&W has proprietary code, written in the Fortran programming language, to solve a specific class of problem. This code is used routinely and is being constantly enhanced. It has also been used as the basis for an optimisation system that will automatically select designs fulfilling specified design criteria.

Recently, especially in the last five years, commercial, off the shelf, systems have become available that allow simulation of acoustic systems to be performed. With the introduction of such systems, computer simulation in the loudspeaker industry is becoming increasingly important with many other companies investing in the area. More recently, the task of porting the Fortran code to Matlab, a popular high level scientific programming language, has commenced at B&W. Though Matlab code is slow to execute in comparison to Fortran code, it has numerous high level features and in-built graphics routines that make it an ideal test bed for quickly trying out new ideas.

A detailed description of B&W Loudspeakers' research into and application of computer simulation to the design of loudspeakers may be found in the references section.

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APPENDIX V
LASER INTERFEROMETRY

Introduction
Various Laser test methods have been extensively used during the development of the new Nautilus™801. There now follows an introduction to the armoury of Laser test methods available.

B&W was the first speaker company to use a Harwell interferometer to make the vibrations in cones visible. Over the years there have been several significant improvements to the system; and one entirely new technique has been invented, which has great potential for helping to improve the design of cones and surrounds. Lasers are being used routinely during the development of new cones and surrounds at B&W.

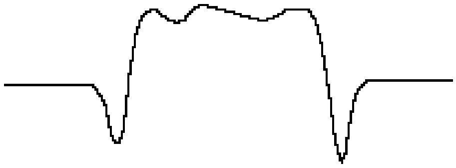
The four main ways that a laser may be used to show cone and cabinet vibrations are:
1. Single Slice single frequency plots (phase-sensitive or rectified)
2. Single Slice multiple frequency plots (phase-sensitive or rectified)
3. Impulse Progression Plots (Single slice animated or as a 3-D plot)
4. Whole cone plots of frequencies or impulses (as individual plots or computer animated)

In addition each of these techniques may be applied to the air itself in front of the cone and the speaker enclosure by utilising a very light diaphragm to represent the movement of the air in response to the sound radiated from the driver and the cabinet.

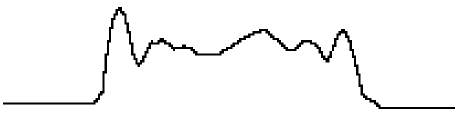
Computer animation techniques may be used to produce movies showing either the phase response at any single frequency over the whole cone or the progression of an impulse as it spreads out from the cone neck.

Single Slice – single frequency – phase-sensitive or rectified
The laser is used to plot single slices across the middle of the cone

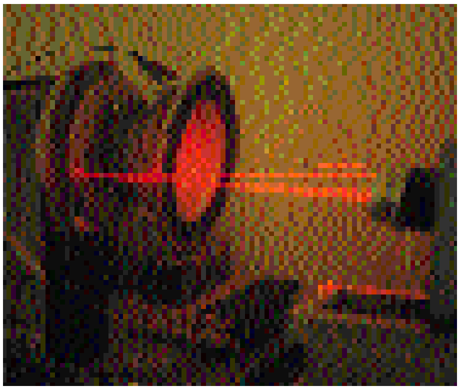
A diameter across the cone is scanned with the laser beam to produce two distinct types of plot as shown in Figures 38a and 38b. This is done at a single frequency of interest. On reflection from a moving object, the frequency of the laser light is changed by the cone movement due to the Doppler effect. The motion of the object is derived from that frequency shift by using what is in effect an FM radio. The output of this FM radio is then compared with the input signal to the speaker. If they are in phase with one another the result is plotted upwards on the screen. If they out of phase it is plotted downwards. The resulting phase-sensitive plot shows which parts of the cone are moving in the same direction as the voice coil and which parts are moving in the opposite direction – that is, it shows which parts are totally out of phase with the voice coil. The disadvantage of this type of plot is that those regions which are 90º out of phase with the voice coil do not show up at all. To overcome this disadvantage we have the 'rectified' plot, which disregards phase and always plots upwards if there is any cone movement at all. Clearly there is a need for both types of single slice plot for a compete picture of cone movement to be obtained.

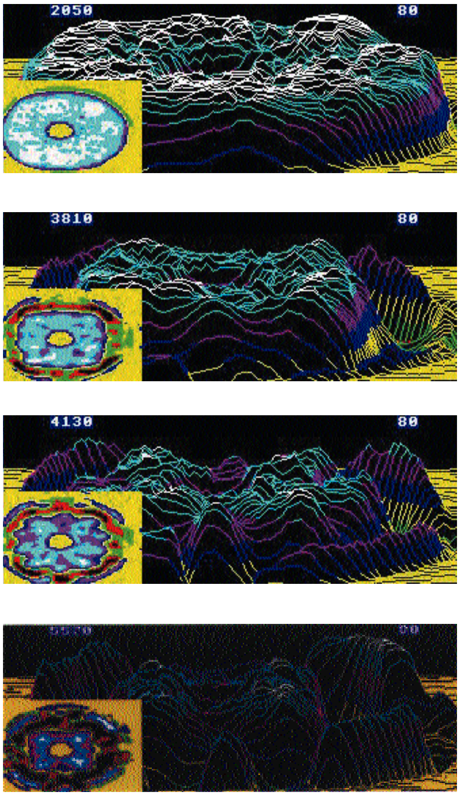


38a Phase Sensitive Slice Plot – Surround out of Phase

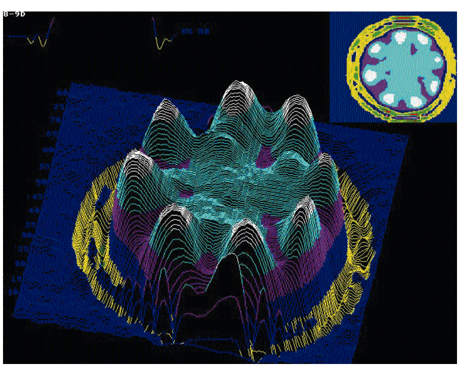


38b Rectified Slice Plot – No Phase Information





39 Whole Cone Single Frequency Plots
Top Picture at Low Frequency, Bottom at high Frequency
Note: 4-way symmetry and smoothly reducing central area



40 Phase sensitive whole cone plot.
Peaks and dips cancel out,
surround sound resonance does not.

Full speaker surface scan – single frequency – phase-sensitive or rectified

The second type of plot is produced by scanning the whole surface of the speaker, still at a single frequency of interest. On reflection the frequency of the laser light is still changed by the cone motion because of the Doppler effect. The motion is derived as before from the frequency shift by using the FM radio principle and the resulting information is now displayed in 3-D, producing mountain like plots as shown in figure 39+40. A perfect piston-like cone gives 'top hat' pictures, with every point on the cone moving exactly the same amount with so called 'piston' motion. The hills and dales produced by less than piston-like motion are easy to see, but are possibly less easy to interpret, and greatly aid the design and development process. Figure 39 shows a classic cone surround mode where the whole surround is moving out of phase with the cone causing a dip in the resulting frequency response. Figure 40 shows a totally symmetrical break-up pattern where for every region going upwards there is an equal and opposite region going downwards. The net result of this is that air is merely shunted around in the near field and very little sound energy finds its way into the far field. What we are then left with is any underlying perfect piston motion and the radiation from the central regions, which are still moving pistonically. When the hills and dales are equal in number and height, we have what is in effect a multipole source which, when the wavelength of sound in air is greater than that in the material, is very inefficient at radiating.

Just as with single frequency single slice plots, the scans may be done with phase-sensitive detection or with a rectified output. The surround resonance shows up as a characteristic "flan dish" shape with phase-sensitive detection, whereas with rectified plots only a slight difference will be seen from perfect piston motion.

Phase animated – single frequency – full-surface scan

The full-surface phase-sensitive detection method is used for this type of plot, except that instead of just one whole surface scan a set of up to 20 surface scans are carried out, each one being at a different phase though the vibration cycle. When these 20 pictures are placed in the computer's memory, a second programme allows all twenty pictures to be repeatedly mapped onto the screen. This gives the impression of animation and can often show up features in a speaker's response not clearly visible from a single frozen-phase whole-cone scan. For IBM PCs these can be produced as ".AVI" files which may be played with the standard Windows™ media viewer programme.

Phase Animated Single Frequency Plot (20 Phases) Fig 40

- Complete single-frequency phase-sensitive plot over whole cone
- A bell mode is evident, giving peaks and dips around cone's circumference
- A resonance in the surround can clearly be seen – it is the first harmonic
- The surround pulls tight at top dead centre (see single slice at the top left)
- The picture shown here is from top dead centre

Note: The bell mode has an equal number of sectors going upwards and downwards. These are also of equal amplitude and therefore their radiation cancels out in the near field (it's an "acoustically fast" multipole source). The surround resonance is not cancelled out by anything and is responsible for a major peak and dip in the speaker's response. The surround pulling tight will cause distortion in the resulting output sound.

Frequency animated full-surface scan

Either the phase-sensitive full-surface or the rectified full-surface type of plot is used for this technique, except that instead of just one whole surface plot being carried out at a single frequency, a set of 256 separate frequencies are used from 200Hz to 20kHz, spaced at approximately 80Hz intervals.

In this case the scanning mechanism leaves the laser spot at each of 126 x 126 positions on the cone surface, while 256 measurements are taken smoothly from 20kHz down to 200Hz at each of the 256 separate frequencies. The results are stored on a computer disc and the spot is then moved on to the next position. Complete surface plots for each frequency are disentangled from the set of results at a later date, using specially written software.

Just as with phase animated plots, the set of 256 separate frames can be projected onto the computer screen as a movie, graphically showing the development of surround and cone resonances which sweep in and out of view as the frequency is changed. The differences between the behaviour of say a Kevlar® cone with a bending wave impedance matched surround and a plastic cone with a conventional surround is very striking when both animated full-surface scans are visible on the screen at the same time.

Frequency Slice Plots
In this case instead of the whole surface of the cone being raster scanned at a single frequency like a TV picture, a single strip of the cone from edge to edge through the middle is repeatedly scanned. Starting at the rear of the plot and at a low frequency, the frequency is increased for each succeeding sweep plotting in front of the previous one, and the resulting 3-D plot shows a frequency history of the behaviour of that slice of cone as can be seen in figure 41. Once again this may be done using either phase-sensitive detection or the rectified method. Resonances and other problems may be seen at a glance, and particularly if the cone is axisymmetric. This may be the only kind of laser plot that is necessary to show everything that is wrong (or right) with the speaker cone / surround / coil combination.

The results of Finite Element Analysis are often plotted in this way as well, so it forms a very convenient test for the accuracy of finite element predictions of cone vibration behaviour.

Phase Animated Frequency Slice Plots (Fig 41)
Just as with the slice plot above, a complete set of scans is done from a low frequency to a high frequency at each of 20 different phases. The resulting 20 plots are then animated, revealing each slice's behaviour as it moves though a complete vibration cycle. This rather curious "frequency normalised" phase animated frequency slice plot shows the same 20 phases for each frequency, so the animation proceeds at the same rate for 20kHz as it would for 200Hz. Another form of plot, as yet to be produced, would cycle though the phases at 20kHz at 100 times the rate that they are cycled at 200Hz. It is doubtful though whether such a plot would be of any use. The frequency-normalised phase-animated slice plot gives the impression that energy passes down the plot from low frequencies to high frequencies, but this is merely an artefact of the normalisation.

Plots such as these show in graphic detail phenomena such as the progression of energy from the voice coil to the surround at high frequencies, just like waves moving down a length of rope being shaken up and down at one end. Also often seen is the surround lagging behind the movement of the otherwise piston-like cone by 90º.

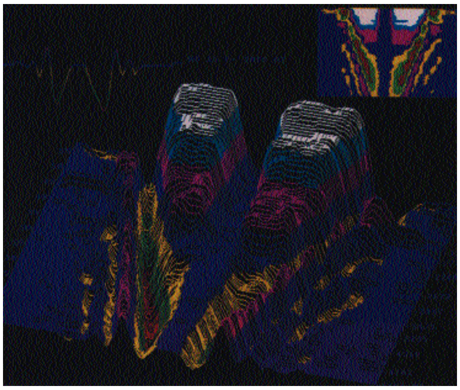
Phase Animated Frequency "Slices" (20 Phases) (Fig 41)

- Rear Slice is a scan across the middle of the cone at 0.5kHz
- Front Slice is a scan across the middle of the cone at 5.5kHz
- Each frame is at a different phase through a complete cycle at each slice's frequency
- This is a very bad cone showing major resonances and poor surround behaviour
- The single frame from the sequence shown here illustrates the surround phase lag at the rear

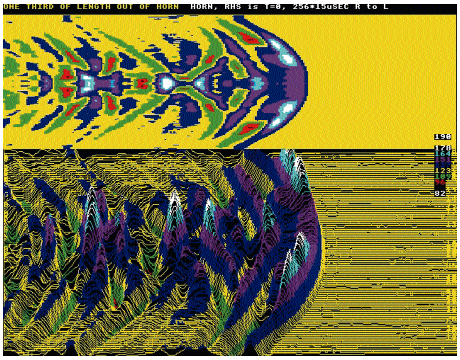
Note, the slice at the front, at 5.5kHz, is just like a rope being waved at one end. No sound will be radiated at this frequency. Even the slice at the back, at 0.5kHz, shows that the surround lags behind the cone (otherwise behaving like a piston) by 90º. Thus the surround radiation will cancel and reinforce the cone sound differently depending on the direction. On axis there will be little effect, but the directional pattern will suffer peaks and dips as a result. The major resonances produce peaks and dips in the speaker's response in all directions.

Impulse Progression Plots – in cones and in the air – slices or full cone
The latest technique in the formidable armoury of laser based measurement techniques available to B&W Engineers is called the Impulse Progression Plot.

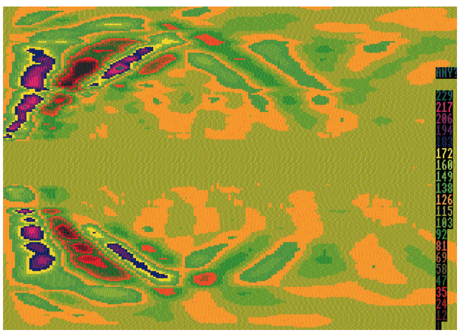
In this case the laser beam is pointed at a position on the cone and the speaker



41 Phase Sensitive Frequency



42 Wavefront Arrival Plot – Sound in the Air



43 Impulse Progression Plot – Cone Diameter

is fed with an impulse, rather than either a single frequency or a sequence of frequencies as in the previous methods. The resulting impulse response of the point is stored in the computer and the beam is moved on to the next point, where its impulse response is translated into digits.

For the single slice impulse progression plot, just a single line of impulse responses is used. For the whole cone case obviously, impulse responses from the whole cone surface have to be translated for processing later.

With a single line of impulses, a plot is produced across the whole diameter for each time interval of the sampling of the individual impulse responses. Each succeeding time interval may be produced as a 3-D plot, with time equals zero placed at any edge of the page (usually at the left hand side). The resulting cone behaviour is then displayed in 3-D as a time history from left to right (or whatever).

Alternatively, the single line may be displayed on the computer as a movie, and the line then appears to waggle up and down as the impulse progresses back and forth along it. However, perhaps the most useful presentation technique for the single slice impulse progression plot is as a coloured contour plot, with time equals zero at the left hand side – time therefore progresses from left to right, with the voice coil in the middle and the surround at top and bottom. An example of this is shown in figure 42.

This sort of presentation immediately reveals that there are several different types of wave motion conveying energy from the voice coil to the surround and back again. The most significant of these are bending waves and compression waves, the latter travelling much more quickly than the former. Figure 43 shows the impulse coming in at the voice coil in the centre. The cone begins to follow it and the wave progresses outwards in both directions to strike the surround near the edges of the picture. This looks very much like ripples produced when a stone is dropped into a round pond.

A computer animated version of this picture shows in graphic detail how much of the incoming bending wave is taken into the surround and how much is reflected back down into the cone again to form standing waves or resonances. The impulse may also be seen moving about in the surround itself, all the while being absorbed and reflected during its travels. The cone can clearly be seen to behave like a transmission line for bending waves (and for other types of waves as well) and for best effect should be terminated in the characteristic impedance of that line at both ends. This will ultimately produce minimum reflections and maximum absorption and so consequently fewer resonances and a cleaner sound output

Generalisations

As with all the above single slice types of plot, this single slice impulse progression plot is especially applicable to axisymmetrical cone structures.

For materials such as woven Kevlar®, the whole cone impulse progression plot has to be used. In this case a complete picture of the whole cone is built up from each of the sampled individual time elements making up the impulse responses at each point on the cone. These pictures are cycled from either the hard disk or the memory of the computer onto the screen, showing graphically an animation of the progression of the impulse across the whole of the cone surface.

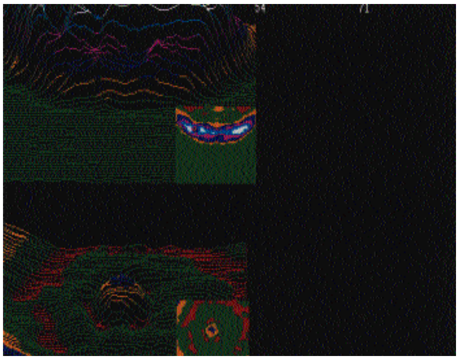
Plots of the motion of the air motion itself

As described in the paper "Laser Techniques in Loudspeaker Design including the Impulse Progression Plot", B&W have a Laser Doppler Velocimetric technique for observing the passage of waves across the surface of a speaker cone. As further shown in the Paper "New Pipe and Horn Modelling", this technique may be extended to measure air motion in order to discover which parts of the speaker cone do the radiating and which parts of the speaker box allow waves to be diffracted, so spoiling the resulting sound. The technique involves placing a very light, highly stretched clingfilm diaphragm lightly dusted with talcum powder in front of the driver. Although this diaphragm is like gossamer and is totally acoustically transparent, it does move with the passage of any sound wave, behaving almost like part of the air up to supersonic frequencies. Since it moves with the sound it may be observed with the Laser Doppler velocimeter which therefore detects the movement of the air itself. (Fig 43)

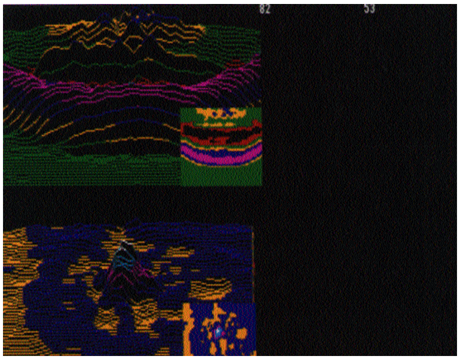
This technique may itself be used in a number of ways. The most obvious is to scan a whole diaphragm placed in front of a cone or speaker box. This allows us to observe the progression of the impulse across the air in front of the cone and across the surface of the box. The second way is to place the diaphragm in a succession of positions further and further away from the speaker or box and measure either a slice across the diaphragm, or indeed the whole diaphragm at each position. Computer processing then allows the passage of the impulse in the air to be viewed as it passes through each successive diaphragm, either as a succession of "stills" or as a computer animated movie. This is very useful for observing the transitions from near field behaviour close to the driver or speaker box to far field behaviour, which is what is usually perceived by the listener, and to pin point sources of diffraction and radiation within the total speaker system.

The passage of an impulse though a single diaphragm may be displayed as a 3-D plot, rather like the slice plot for displaying the behaviour of a single slice across a speaker diaphragm at different frequencies. In this case the first arrival is plotted at the right hand side and subsequent arrivals are displayed from right to left. We thus graphically see with this "wavefront progression plot", how energy comes along well after the original impulse has passed that point in space.

When observing the differences between the impulse behaviour of axisymmetric cones such as those made from plastic, with non axisymmetric cones made from woven Kevlar® fibres, the difference is striking. The sound energy radiated by the axisymmetric cone consists of an initial wavefront, which largely represents the music, followed by a series of other waves, which are not the same in all directions, representing the coloration of the cone and caused by the waves bounding backwards and forwards across the cone. Measuring the behaviour of the air can also be used to illustrate how cabinet edges really do produce copies of the original sound, but often out of phase with the original and delayed by the time it took for the sound to get from the driver to the diffracting sharp edge. (Figs 44, 45)



44 Phase Sensitive Frequency



45 Phase Sensitive Frequency

The science of sound

Visitors to B&W's Research Establishment in Steyning, UK, often compare it to a university department, set apart, as it is, from our day-to-day manufacturing operations. Indeed, our 'University of Sound' is widely regarded as one of the most sophisticated audio laboratories in Europe, home to a research and development team of 22 gifted scientists, engineers and technicians. Their work has enabled B&W to develop some of the world's most advanced measurement and forecasting techniques for loudspeaker design.

Our system of laser-doppler velocimetry, for example, allows us to measure, down to 1/3000 of a millimetre, the vibration of a loudspeaker cone or dome (B&W was the first hi-fi company in the world to adopt this technique). We have also developed computer-aided 'finite element analysis' programs, which enable us to build drive units, step by step, based on an accurate prediction of how the component parts of a loudspeaker will perform – separately or together.

Measurement techniques include colour-coded impulse scans, which show how sound waves behave when radiating outwards from a cone or dome; we have developed three-dimensional 'die-away' plots, which analyse patterns of stored energy which continues to radiate after a sound wave has passed. We can even measure sound patterns in the air around loudspeakers; and we have pioneered digital signal processing to analyse crossover performance and 'on-axis' response.

Finally, we call on subjective listening tests. Our team of experts analyse the performance of B&W loudspeakers in a variety of domestic and studio settings, allowing us to fine-tune each new model to create the unique B&W experience.

B&W's research institute in Sussex, England, is widely regarded as one of the most sophisticated audio laboratories in Europe

LISTEN AND YOU'LL SEE

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