# RESONANCES IN AUDIO THEY'RE BAD THINGS - AREN'T THEY?

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## 1 INTRODUCTION

Almost every musical instrument uses resonance to generate or amplify the sound output. Imagine listening to a violin with no body. So, it seems, some resonances are necessary whilst others are taboo. Maybe we should think of them as either good or bad resonances.

It was Rice and Kellogg who published a paper entitled "Notes on the Development of a New Type of Hornless Loud Speaker" in 1925, with a description of an instrument of the piston type they had recently developed. It consisted of a lightweight conical diaphragm driven by a moving coil and they deduced that it would generate a flat pressure as well as a flat power response, at least up to the point where the diaphragm started to beam. Engineers like simple ideas and the concept of the "rigid piston" seemed the ideal solution for a loudspeaker – and we've been stuck with it ever since. Keeping the piston rigid also meant that we would avoid any resonances in-band, since these were regarded as bad resonances.

This paper looks at the difference between good resonances and bad ones. The journey starts at Celestion, where the first scanning laser interferometer in the loudspeaker industry was designed and built.

It finishes in the present day, by describing the invention of a loudspeaker which uses resonances as an integral part of the design – but only the good ones, of course.

# 2 CELESTION – THE EARLY YEARS

I first joined Celestion in 1977 where the order of the day was cones - much like those used by Rice and Kellogg in the twenties. It had been pretty much established that all cone resonances were bad things – apart (funnily enough) form guitar loudspeakers where these were positively encouraged (obviously good resonances in this case).

It was believed – and quite rightly so – that the secret to the loudspeaker behaviour was vested in the diaphragm behaviour. Get this right and you were well on the way to a high quality loudspeaker system. For many years loudspeaker engineers had tinkered with the geometry, material and treatment of diaphragms in order to improve performance, but had mostly relied on the measured on-axis SPL as their yardstick.

In order to build ideal drivers it was essential to be able to know what the behaviour of the diaphragm was, up to the highest frequencies to be reproduced. To be able to be confident that the diaphragm was indeed behaving as intended required some form of instrument that could check the velocity over the whole of the radiating surface. I had been introduced to laser interferometry by Peter Fryer, a few years earlier, who was at Wharfedale at the same time as me. His system showed fringe patterns which gave indications of the relative motion of the various parts of the diaphragm. Figure 1 clearly shows the effect of lead-out wires on a driver, taken from the AES 50<sup>th</sup> Convention in 1975<sup>1</sup>.

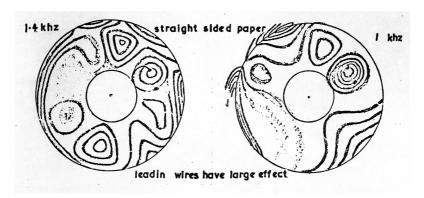


Figure 1 Holographic fringe plots of lead-out wires

The system did work, but required special facilities and needed expertise to run it.

### 2.1 Laser Interferometer

What was needed was a day-to-day instrument that normal loudspeaker engineers could use. We had heard of a system that Harwell were using to check ultrasonic transducers that in turn were used to do NDT on nuclear reactor vessels. A trip to Harwell AERE outside Oxford led Celestion to see if their laser interferometer would do the job. We had imagined scanning the laser over the surface of the diaphragm and replaying, in slow-motion, the diaphragm shape at any chosen frequency. We were advised that the signal to noise would be so low and scanning would take so long that this idea wasn't practical. Such advice was ignored and Harwell agreed to sell their experimental laboratory laser interferometer. Harwell used this money to build themselves a newer version, so we thought that we got quite a bargain. The resident engineering team at Celestion then set about making the idea work.

It was in 1980 that the first scans were made. By May of 1980 we presented a paper to the AES<sup>2</sup> describing the system. It created quite a stir. So much so that we started to get interests from other researchers working in completely different fields.

## 2.2 Lasers spread far and wide

The first researcher who contacted us wanted to look at the sound absorbing characteristics of plant leaves. It was in 1982 when Maurice Martens contacted us and described his work. He wanted to design sound absorbers made from plants. These would be environmentally friendly, look pretty and of relatively low initial cost. What he didn't know was the absorbing characteristics of different plant species. Traditional methods were to fill anechoic rooms with mature plants and measure the transmission characteristics. This needs a lot of plants and a big anechoic chamber.

His idea was to find the natural modal frequencies of various leaf species and relate these to anechoic measurements<sup>3</sup>. It seemed to him that at a natural mode of the leaf energy would be absorbed.

In 1985 Konrad Konradsson contacted Celestion with a view to using the Laser Interferometer to study the behaviour of tympanic membranes. At least this was a bit closer to reproduced sound – so quite interesting. Konrad would bring samples of inner ears which had been donated by willing volunteers. This couldn't be done with live specimens since the laser could not scan down the ear canal. Anyway the sound levels would more than likely cause some long term hearing problems. The tympanic membrane was coated with a fine white powder and scanned just like a treble unit. Excitation was from an external sound source<sup>4</sup>.

What came out of this work was that the first mode of the tympanic membrane was typically at around 550 - 600 Hz. This is exactly where the Fletcher and Munson equal loudness contours have their minimum.

It was fascinating to confirm that the ear drum becomes modal above this frequency. So, your primary audio receptor has resonances from 550 Hz upwards. I have to assume that these belong to the good resonance family.

# 2.3 Lasers used for loudspeaker diaphragms

The laser system was developed from a single phase one-shot device into one which was more useful. In later versions you could manually adjust the driving frequency, look for changes in the output at the diaphragm centre and then perform a complete surface scan. A pair of mirrors controlled the position of the laser beam in the X and Y axes, allowing a full surface scan to be made. Changing the reference phase and re-scanning built up a set of snapshots which were subsequently animated to give a very realistic image of the actual displaced diaphragm shape. The early instrument showed only a single phase plot, captured on a flat-bed plotter, and a couple of plots are reproduced here for reference.

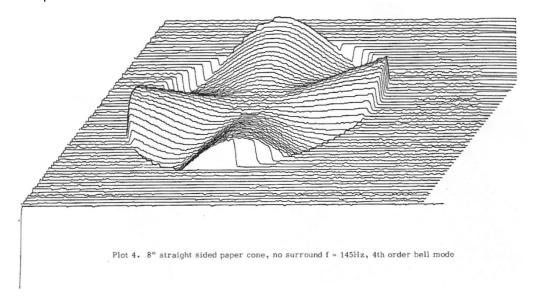


Figure 2 8inch edgeless cone<sup>2</sup>

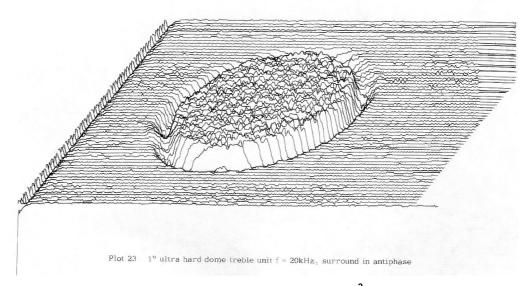


Figure 3 Beryllium metal dome<sup>2</sup>

Diaphragm modes, surround resonances, lead-out wires and dust caps were all shown for what they were and more importantly what they did. It was more than an eye —opener. It led engineers to re-think what makes a good drive unit and led to some new construction methods.

The early laser system was updated by a computer-based replacement, using an 8-bit CPU. The new system was capable of scanning twelve different phases at each frequency, and the results displayed on an oscilloscope screen.

If pistonic was to be the target, then this measuring device heralded the era of the metal dome. Not much else was affordable and as good in terms of stiffness to weight ration as metal domes. Celestion started with a copper foil which was electrochemically grown using a metal plating technique. This was followed by cold forged aluminum foil and then by titanium. The object was to try to adjust the dome geometry in order to put the first mode as high in frequency as possible. It seemed to set a trend and many followed.

Making a pistonic bass unit seemed a much more challenging task and few have succeeded in making an affordable piston of any size which can operate over the full bandwidth. Here it seems you need a balance between diaphragm geometry and damping to keep the resonances within control.

## 3 WHY MIGHT THE PISTON NOT BE IDEAL?

Whole generations of drive units were now available which had mechanical behaviours which were well understood and controlled to a level which had not been done previously. But there was a snag. Controlling the mechanical structure would give an acoustic performance which was predictable – but unfortunately it was limited.

Making a pistonic driver is ignoring the very acoustic medium that it couples to. A perfect piston couples to the air above it in a way which is completely characterized – it has a flat pressure and power response up to a frequency determined by its size. The drive unit beams because of the destructive interference patterns in the air above it. What we actually need is to consider the structure and the air as a coupled system.

## 3.1 Modal loudspeakers

This brings us to modal loudspeakers – those devices where the natural resonances in the diaphragm are arranged to support the output - rather than generate destructive interference.

My first introduction to these was in 1996 in Huntingdon. Cloaked in secrecy I was ushered into a darkened room – along with a few others – who had also signed a Non-Disclosure Agreement – to hear this new technology. It was a bit rough around the edges but the two large 3ft x 2ft panels showed clear promise. This was the start of NXT.

The idea is quite neat. If you force a flat structure close to its centre then on-axis you will get a flat pressure response – provided that the force has a low mass associated with it. The natural modes of the free panel do not radiate on-axis but contribute to the off-axis output. Combine these two outputs properly and you have a wide bandwidth, wide directivity loudspeaker. The basic concept does have some limitations though. A heavy driving coil upsets the on-axis response and low frequencies are limited by the size of the panel. Early large panels showed excellent results, but the trend to smaller loudspeakers was already gathering momentum.

#### 3.2 Resonances or modes?

These first devices were diffuse down into the mid-band because of their size, but later designs which grew smaller were less so. Right from the start we tended to call these devices modal rather then resonant – simply because everyone knows that resonances are bad things.

## 4 THE BALANCED MODE RADIATOR

Delivering low frequencies had always been a problem. The Holy Grail for loudspeaker engineers was to find a way to have low frequencies, wide bandwidth and wide directivity all at the same time. Because work on making an NXT panel have high modal density was so focused, delivering wide directivity, wide bandwidth and low frequencies - altogether, didn't look possible.

It was during a time that I was asked to prepare a training session on a particular aspect of NXT that it suddenly dawned on me. Maybe we shouldn't be looking at a densely modal panel, but consider a sparsely modal device. I wanted to investigate such a device, so went right back to the beginning. To reduce the number of modes I needed to consider a one-dimensional object, which have fewer modes than a rectangular panel. Such a device could be either a long beam or a circular panel and I chose circular symmetry for ease. The results of this analysis were published in the AES <sup>5</sup> as well as presented to the IOA <sup>6</sup>, and the background is presented here.

#### 4.1 THE FREE DISC

For a modal object, the total response is made up of the sum of the outputs from all the individual modes. In the case of a flat disc, the on-axis pressure is a sum of the pistonic response and the modal contributions. In using a free disc, the modal contributions have a zero mean volume velocity, so they do not contribute to the on-axis response.

If we now apply an "ideal force" to this disc, that is a force which acts at a point and has no mass or damping, we can convert it into an "ideal loudspeaker". This loudspeaker has a substantially flat on-axis frequency response and wide directivity, indicated by a smooth extended power response.

## 4.2 FEA MODELLING

In order to evaluate the free disc case, as well as the proposed solutions it was decided to use the numerical methods of Finite Element Analysis (FEA) and the Boundary Element Method (BEM). In combination, these techniques can be used to calculate the complete responses. Using a fully coupled model, a series of axisymmetric Finite Element Analyses were run using PAFEC. Shell elements were use to model the diaphragm. The far field region was modelled using a boundary element formulation, and an intermediate region of fluid finite elements connected the structure to the boundary elements. Any possible ill-conditioning from the coupled boundary region was eliminated by using the Burton-Miller formulation. Both on-axis pressure and radiated acoustic power were evaluated for a range of frequencies from 100 Hz to 10 kHz (the acoustic power is a convenient way to gauge the overall directivity of the device in a single curve). No structural damping was used in any of the analyses.

#### 4.3 Basic model

The disc used in the FEA model was 1mm thick aluminium, 144.7mm in diameter. A monolith avoids most of the concerns over the shear limitations that a composite might have, and the disc size was chosen to have four modes in the band of interest. The basic mesh is shown in Figure 4. A parameterised approach allowed changes to the model to be made readily, and facilitated automatically meshing to new parameter sets.

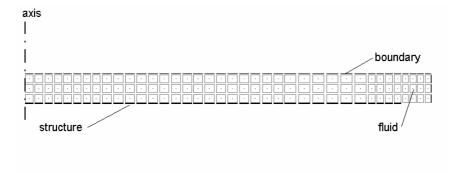


Figure 4 Mesh of axisymmetric FEA model

## 4.4 Ideal voice-coil

The first model was used to confirm the expected behaviour of a rigid piston, set into an infinite baffle, by driving the centre of a disc with a mass-less force. All the structural freedoms were repeated, constraining the disc to be a piston. Results, shown in Figure 5 are as expected, with a flat on-axis response and a falling power response, indicating beaming at high frequencies. The turnover point confirms that the power response is dictated by the overall diameter of the piston.

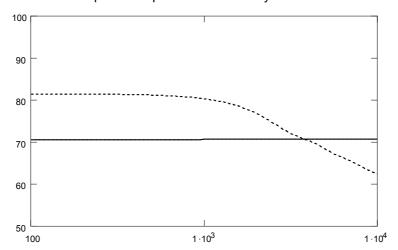


Figure 5: Piston driven with mass-less point force. Solid line: dB SPL (on-axis sound pressure level), Dotted line: dB SWL (sound power level), versus log (frequency).

When the repeated freedoms are removed, then the disc becomes a modal radiator and, using an otherwise identical model, and has the response shown in Figure 6. The on-axis pressure remains substantially flat, but the fall-off in sound power has been dramatically improved. It can be seen clearly that the panel size no longer controls the directivity. However, there are panel modes visible in the simulation results, partly because the model uses no mechanical damping.

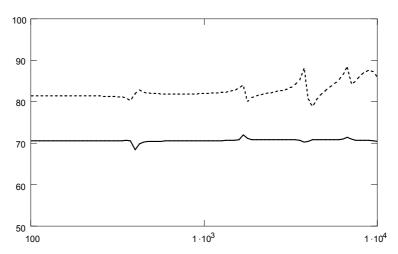


Figure 6: Modal disc driven with mass-less point force. Solid line: dB SPL. Dotted line: dB SWL

For all practical loudspeakers, the force would be supplied by a voice-coil with a finite diameter, depending upon the design and power handling requirements. Prior art would indicate that a good place to position the voice coil would be on the nodal circle of the first bending mode, which is at a diameter ratio of 0.68. As with the previous case of a point force, the mass is still set to zero and the result is shown in Figure 7.

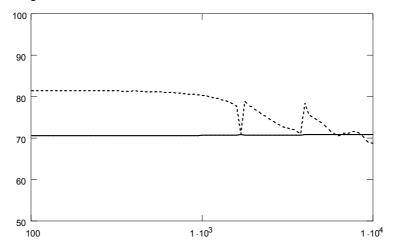


Figure 7 Modal disc driven with mass-less coil, Solid line: dB SPL. Dotted line: dB SWL

The uniform sound pressure is preserved, and the sound power falls off at high frequencies, but noticeably less than in the piston case (compare this result with Figure 2), confirming that the power response is now related to the voice-coil size not the panel diameter.

## 4.5 Practical voice-coil

Unfortunately, for all practical purposes, a force always has mass and normal voice-coils have finite diameters, rather than delivering a force at a point. The action of driving the disc at a given diameter, with a mass associated with it, unbalances the free disc and gives rise to errors in both the pressure and power responses.

When the ideal voice-coil is replaced with a practical example (which has mass and damping), as well as adding a roll surround, then the disc is unable to behave like the free disc prototype. The result is a loudspeaker which has an uneven on-axis response and a poor power response. It has been customary to treat these designs with some form of applied damping in order to make them

useable. To show this effect, the next two figures show the previous model with 2-gram coil (Figure 8), and then an 8-gram coil (Figure 9).

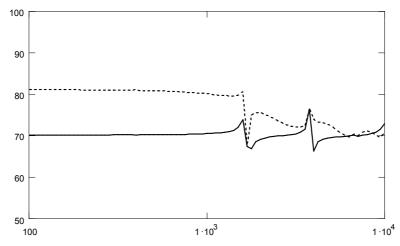


Figure 8 Modal disc driven with a 2 gm coil, Solid line: dB SPL, Dotted line: dB SWL

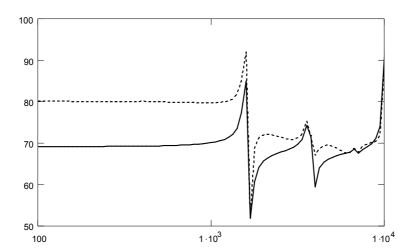


Figure 9 Modal disc driven with an 8 gm coil, Solid line: dB SPL, Dotted line: dB SWL

## 4.6 BALANCING THE MODES

Adding a practical voice-coil has "spoiled" the naturally flat response by distorting the mode shapes. The hypothesis was formed; that if the original mode shapes could be restored, then the desirable acoustic properties of the free disc would also be restored.

## 4.7 Adding balancing masses

In an attempt to recover the free disc modes shapes that had been distorted by adding mass, it was proposed that additional masses might be used to balance the necessary mass of the voice coil. But, how could a number of modes be considered simultaneously, since the nodal circles would change positions for every mode shape?

In order to balance a number of modes in the operating frequency range we need to determine a set of average nodal positions. These average nodal positions are those which encompass the highest mode order of interest and all the lower order modes.

By evaluating the real part of panel's mechanical admittance Re (Ym) it is possible to determine the average nodal positions. In the case of the first bending mode, this is shown in Figure 10.

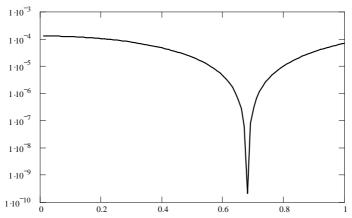


Figure 10 Real part of Ym, for the first disc bending mode, Re (Ym) s/Nm versus relative radial position.

The admittance curve exhibits a clearly define dip at the 0.68 ratio, confirming the expected result when only the first bending resonance is present.

Compare this with Figure 11, which is the result of adding a second bending resonance into the admittance calculation. Although both the first and second modes contribute to the admittance, the dip from the first mode has almost completely disappeared.

This result gives an indication of how we might rebalance our disc which was unbalanced when the mass of the voice-coil was introduced. Since the two dips in Figure 8 represent regions of low velocity, if two scaled masses could be added at these two positions, then the second mode shape would be preserved. Not only that, but these two masses would form a couple which would also satisfy the admittance curve for the first mode. This technique can be extended to any mode order, and adding masses at the "average nodal positions" of the highest mode automatically recreates the free disc mode shapes at this mode and all the lower modes.

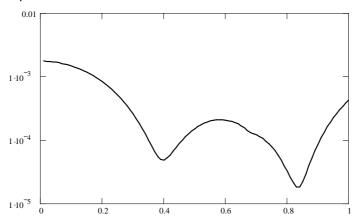


Figure 11 Real part of Ym, for the first two disc bending modes together, Re (Ym) s/Nm versus relative radial position.

When the voice-coil is positioned at one of these "average nodal positions", either the inner or outer, a second mass can be added at the other location, recovering the response of the free disc up to and including the highest mode balanced.

This method can be adopted for any number of modes, but typically up to 4 or 5 modes would

normally be sufficient. For 5-modes, the ratio of lowest to highest modes is 1:27. If the first mode was set at 1 kHz, then the output would be "fixed" up to 27 kHz. Likewise, putting the first mode at 2 kHz would "fix" the device up to 54 kHz.

A 4-mode "fix" would have masses located at four "average nodal positions", with one of these being the voice-coil. Since we are trying to re-create a free disc, the outer average nodal position is inboard of the disc periphery and it is here that the roll surround would normally be fixed, thus providing the outer balancing mass. The two remaining masses would take the form of rings fixed to the disc's surface. For a 2 mode "fix" the coil and roll suspension provide all the necessary masses and no extra parts are needed. This is the simplest implementation to make. Unfortunately, any additional masses will always incur some loss of sensitivity, but the characteristics of the on-axis response and power output should be similar to those of a free disc, up to the highest balanced mode. We termed this radiator a "Balanced Mode Radiator", or "BMR".

## 4.8 Analysis after balancing

The two examples used in Figure 8 and Figure 9 were modified to include masses to balance the voice coil mass. The BMR solution to the 2-gram voice coil is shown in Figure 12. The 8-gram case might appear to be much more difficult to balance, but the BMR method still works, as shown in Figure 13. Both solutions have no structural damping, so that the disc modes can be seen more clearly. The residual damping present in most practical materials typically renders these modes harmless.

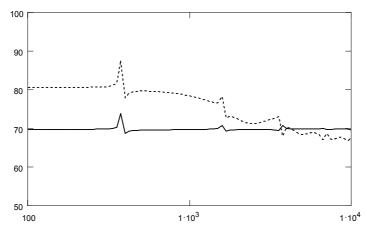


Figure 12 BMR modal disc solution, driven with a 2 gm coil. Solid line: dB SPL. Dotted line: dB SWL

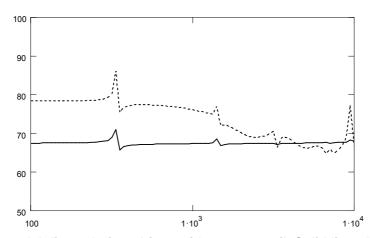


Figure 13 BMR modal disc solution, driven with an 8 gm coil. Solid line: dB SPL. Dotted line: dB SWL

# 5 A PRACTICAL BMR

# 5.1 Example

Previous results in this paper have shown that the high frequency output of a BMR is not governed by the panel diameter, but rather the coil size and number of balanced modes. The geometry of the "average nodal positions" dictates that there are a given number of panel diameters, once the voice-coil size is chosen, although increased disc diameters allow for more modes to be balanced.

In the example used in this paper, the voice-coil was set at 32mm diameter and the BMR balanced using four modes. The coil was set at the third "average nodal position", whilst the surround fixed at the fourth position. These ratios are 0.2, 0.44, 0.69 and 0.910, giving a panel diameter of 46 mm. Figure 14 illustrates the loudspeaker construction. Masses at the two inner positions are not shown in this sectioned view for clarity.

In order to recreate the behaviour of the free disc the surround is not located at the disc edge, and is positioned behind the panel, so that the whole of the disc is able to radiate, reconstructing the complete wave front.

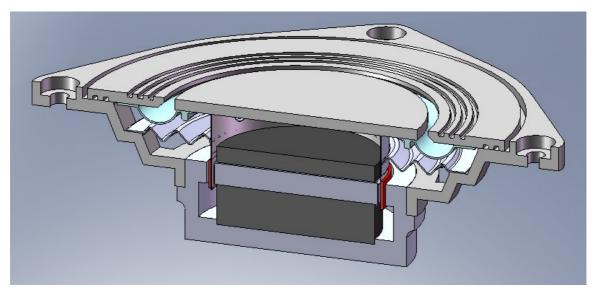


Figure 14 Sectional view of prototype BMR.

## 5.2 Panel material

The behaviour of the BMR is dominated by the overall mass distribution, leaving the panel stiffness to control the frequencies of the modes. These need to be set so that the balanced modes are in the frequency range of interest. The high frequency limit will be controlled by the first un-balanced mode, and panel shear must be sufficient to prevent this mode from dropping into the operating band.

## 6 RESULTS

A prototype, as described in section 5, was built and tested. The response curves for 2 V rms into 4  $\Omega$  (i.e. a nominal 1 W input) are shown in Figure 15. The responses were measured in a small, 2  $\pi$  anechoic room.

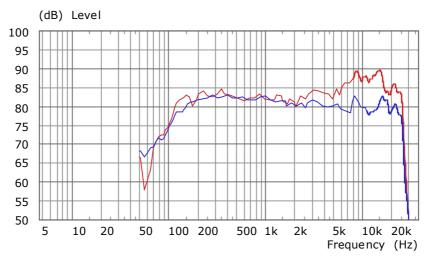


Figure 15 First prototype, 46mm BMR, Red = SPL on-axis SPL at 1 metre, Blue = SWL

Coincidence frequency for this panel is at 6.3 kHz, and is partly responsible for the increased output at high frequencies.

## 7 CONCLUSIONS

The underlying hypothesis, that the mode shapes of a free disc could be recreated by mass-loading a disc at its average nodal positions, was tested using a fully coupled FEA model. The results for a free disc, driven with a mass-less voice-coil, were compared to those of a balanced disc, driven with a practical voice-coil. Re-balancing the disc in this way gives substantially the same response characteristics as the free disc, although adding extra masses will inevitably lower the sensitivity.

The measurements of the prototype demonstrate the practicality of this new loudspeaker.

#### 8 REFERENCES

- 1 P. A. Fryer, 'Holographic Investigations of Speaker Vibrations', 50th AES Convention, London, 1975.
- G. Bank and G. T. Hathaway, 'A REVOLUTIONARY 3-D INTERFEROMETRIC VIBRATIONAL MODE DISPLAY', 66<sup>th</sup> AES Convention, May 1980, Los Angeles.
- M. J. M. Martens, J. A. M. van Huet and H. F. Linskens, 'Laser interferometer scanning of plant leaves in sound fields', Proc. Koninklijke Nederlandse Akademie van Wetenschappen, Series C, Volume 85 (2), 1982.
- 4 Konrad S. Kondrasson, Alf Ivarsson and Graham Bank, 'COMPUTERIZED LASER DOPPLER INTERFEROMTERIC SCANNING OF THE VIBRATING TYMPANIC MEMBRANE', Scand Audiol 16: 159-166, 1987.
- N Harris and G Bank, 'A balanced mode radiator (BMR)', 119<sup>th</sup> AES Convention, New York. October 2004.
- G Bank and N Harris., 'A balanced modal radiator (BMR)', Reproduced Sound Conference 21, Oxford, 2005.