Review of the Loudspeaker Systems

Summary

On this website around fifteen articles have been published about loudspeaker systems. Seven of them treat elaborate descriptions of high end - middle/treble units consisting of (a number of) full range speakers. They all take the frequency range of 400 Hz to 20 kHz. The low notes have been reproduced forever by a Motional FeedBack (MFB) system to complete the two-way systems.
The units described are compatible so that they could easily be compared to each other.

The usage of full range speakers for the middle/treble units is to avoid a cross over around 2 kHz (being within the most sensitive range of human hearing) to some kind of tweeter. The physical distance between such (dome) tweeter and the squawker always produces side lobs apart from the differences in radiation patterns of the units in question. I have never heard a satisfying solution with those configure-tions even not with coaxial speakers.

Generally, full range piston speakers, that cover 400 Hz to 20 kHz, are small (2**''**) with small light voice coils, so they are not able to handle much power. This means that a number of small full range speakers have to be put in an array. Such an array should be designed carefully to obtain a reasonable radiation pattern. With ESLs (Electro Static Loudspeakers), this is nearly impossible.

The exception is the BMR (Balanced Mode Radiator), also called 'bending wave loudspeaker'. This speaker behaves like the "perfect point source". Around 2005 a practical flat diaphragm 3**''** loudspeaker has been developed which has a substantially flat on-axis pressure response, and a smooth and extend-ed power response as well, so it behaves like a "perfect point source". Furthermore it can handle some 20 watt so that it must not be configured into an array!! It seems to be the ultimate solution for the wide range speaker covering 400 Hz to 20 kHz.

Timescale

Far back in the previous century, I did built myself many two-way end three-way systems with Philips loudspeakers. The AD8061 was nearly always in the system together with all kind of tweeters: dome tweeters, flat tweeters and little cone speakers. See also:*'De Keuze van Dynamische Luidsprekers en Cross-overfilters*'. The sound was always much less than that from the ESL63s of my friend: Henk ten Pierick †.

ESL (Electro Static Loudspeaker)

The bending came with the construction of my ESLs (400 – 20,000 Hz) which I describe in: *'ESL + MFB = the best of 2 worlds?*' on this site. They did sound great, even better than the ESL63 because of the lows from the MFB-boxes. I listened to them for many, many years!
There are <u>three</u> disadvantages however: the bad <u>radiation pattern</u> that creates a small sweat spot, the lack of <u>dynamics</u>: they cannot play loud enough and they need <u>maintenance</u>!

Wide band speaker arrays

Small 2**''** wide-band piston loudspeakers often do have an acceptable radiation pattern, but they cannot handle enough power to reproduce the frequency range of, say, 400 Hz to 20 kHz in a high-end system at 100 $dB_{\mathbf{SPL}}$. This could be solved with a composition of these small speakers in arrays. This is not simple and enquires design tools (like LEEP) to establish a desirable radiation pattern so that the sweet spot becomes large. See: '*High End Cardioid Loudspeaker Array*', end '*High End Circular Cardioid Loud-speaker Array*'. These solutions offer a good sound and great dynamics. The stereo image is very precise. Sometimes however the produced sound 'sticks' a bit to one of the arrays. The sound becomes hard/harsh if the system is played loudly.

Omniwave

Leo de Klerk makes OmniwAve systems, which are assembled, from four 'homebrew' bending wave speakers (see: 'De rondstralende luidspreker', on this site). The idea is to produce a phantom stereo image from two phantom mono images. He calls that mono phantom images 'inaudible loudspeakers'. Such OmniwAve systems could very well be combined with live music instruments in a concert hall for real time enhancement. They sound very nice in such environments. However, the 'homebrew' speakers are difficult to copy. Therefore, I tried to build an OmniwAve system with BMRs. I think that this should very well be possible (see: 'OmniwAve with BMRs') but I leave it further to Leo.

BMR (Balanced Mode Radiator)

Three BMRs of different sizes have been investigated. The difference in size is small as the sound from them. They have been put in a quasi dipole arrangement and in closed boxes as the **min 12** of Cambridge Audio. See: 'BMR in afgeronde kast', 'OmniwAve met BMRs?' '2 BMRs in een baffle' and 'BMRs in an Array'.

The sequence of the listed speaker systems not only show the progress in time but more or less that of the sound quality. Afterwards I could have spared myself the effort, but the BMR is known for only ten years and scarcely applied. They are that cheep and so relevant for this goal that there is no justification to build any other system from now on.

What kind of system with BMRs?

As the latest loudspeaker project I have built me the ‘BMRs in een Array’ to investigate more precisely the behaviour of BMRs particularly that of the **BMR56XE N4R** of Cotswold Sound Systems.
It is not needed for power handling but only to investigate the correlation between a speaker configuration and the sound from it: spaciousness, stereo image (localisation of the sound sources in the phantom image) and the sound in general. For this the project: ‘BMRs in een Array’ has been harnessed.

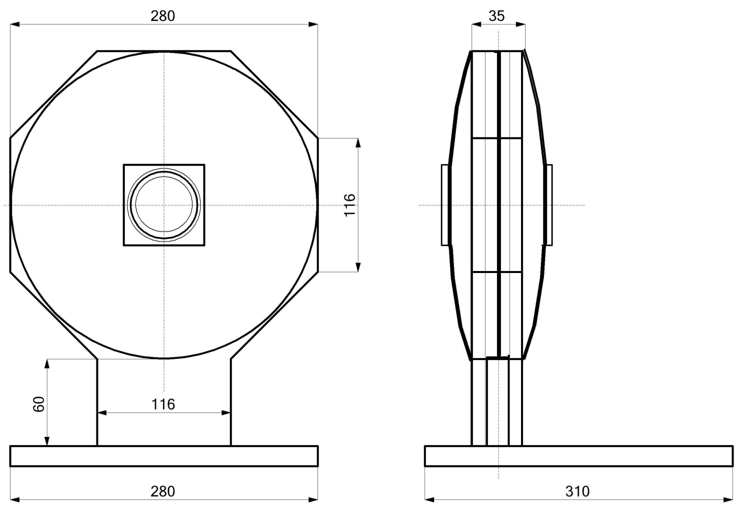
Array of BMRs

It became immediately clear that any array of loudspeakers causes interference problems if it has not been carefully designed like the circular- and linear arrays with the small VISATON speakers on this site. BMRs are no exception, so two or more BMRs in one and the same plane prescribe a complicated design.

One BMR

Because one BMR could handle the power for the high end of a two-way system with a cross over frequency at about 400 Hz, one speaker will satisfy for not too large rooms. The radiation diagram is that of a point source. With one BMR, a second order high pass filter and a woofer, a splendid HiFi system could be designed. It will sound great with a detailed stereo image.

Baffle step



Designing a speaker box with limited dimensions, one has to deal with the so called ‘baffle step’ (Olson).
If F_3 is the 3 dB point of the 6 dB step: F_3 = 115 / D if D …(m) is the diameter of the baffle.
<u>If a plane baffle is used, the baffle step could be counterbalanced with a second speaker at the rareside of the baffle!</u> Doing so, a more or less omni-directional speaker arises. It becomes at least a speaker that is difficult to localise which is desirable to obtain a nice spatial stereo image with two of them.
This system has been described in:
‘2 BMRs in a Baffle’ which became the far best speaker system I have ever build: the spaciousness together with a nice localisation of the stereo phantom image is unbeatable!

The cross over to the low notes MFB unit

Until now, all mid/high unites from 400 Hz to 20 kHz have been rounded with an MFB system from 400 to 40 Hz. How should this be executed?
The MFB system shows a roll off of about 24 dB/octave (see page 19 and 20 of High End Circular Cardioid Loudspeaker Array). The mid/high units should also show a slope of about 24 dB/octave for a good match.
One of the strange things is that <u>the phase of the cooperating connection</u> for a flat frequency characteristic on an audio spectrum analyser <u>hardly matters</u>! It also is difficult to <u>hear</u> the difference between the two in a stereo session. Nevertheless the basses of a piano seems to originate from a different place as the discant. This could be better. The phase characteristic seems to be of more importance than the frequency characteristic. Of course one could do phase measurements outdoors (a room is too small at 400 Hz) but could a <u>mono listen test</u> in the room with pink (or brown!) noise not satisfy?
Both units, the mid/high unit (at least that of the ‘2 BMRs in a Baffle’) and the MFB box do have an omni-directional radiation pattern at 400 Hz so that it must be possible to match the two.
This ‘tuning’ (with the L and C in the cross over filter) has to be executed in mono.

This has been done for the frequency characteristic up till now, BUT, could it be imaginable to hear the right phase and decide for the right values of the cross over L and C of the mid/high units? The variable L and the switchable C from the LC-bank could be useful.