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. Moreover, the electric impedance as function of the frequency is far from flat and shows a not to ignore dip and peak near the cross over frequency.
I have never heard a satisfying solution with those configurations 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 extended power response as well, so it behaves like a "perfect point source". Furthermore it can handle some 20 watt so that it needs not to 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 range speaker arrays

Small 2**''** wide-range 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 Loudspeaker 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 BMR-types 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 them: 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

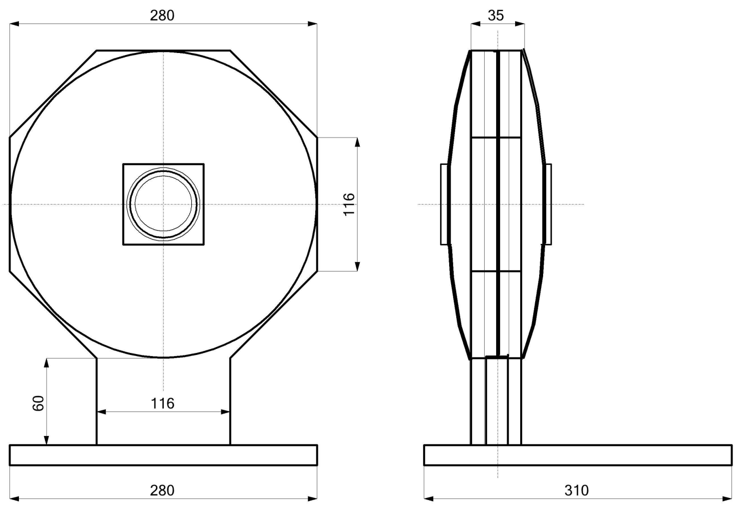
It became immediately clear that any array of loudspeakers causes interference problems if it has not been carefully designed like the circular- and linear arrays with the small VISATON speakers on this site. BMRs are no exception, so two or more BMRs in one and the same plane prescribe a complicated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

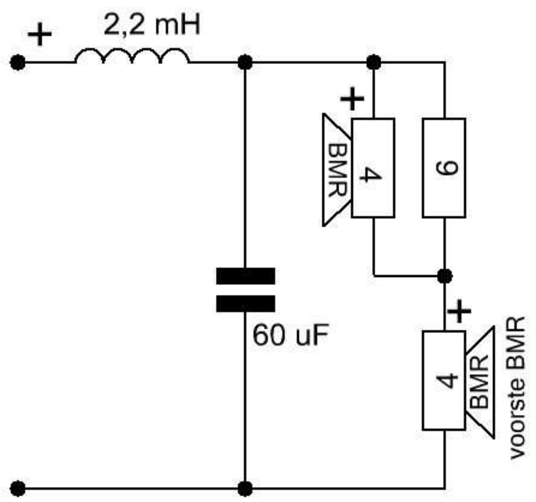
This has been done for the frequency characteristic up till now, BUT, could it be imaginable to hear the right phase and decide for the right values of the cross over L and C of the mid/high units? The variable L and a switchable C from the LC-bank could be useful.

In the mean time I prepared an L- and C-bank to realise a temporary cross over filter. They have been equipped with rotary selector switches for small capacitance/induction steps which is nice if working with an audio spectrum analyser or just for listening.
Measuring the amplitude – frequency characteristic, I find **60 μF** met **2 mH**. The <u>polarity</u> of the two ways can hardly be determined. This became clear with listening to noise from one channel at a time *and walking through the room!* **But,** the idea to find the best match in one stroke by listening to noise (brown nor pink) was just a farce!

Nevertheless with the found L and C value in the amplitude domain and the 'right' phase by listening, the stereo phantom image of a piano became what it should be: the bass came from the same direction as the discant, and a Steinway-D could easily be recognised! If not, the recording is left to be desired.

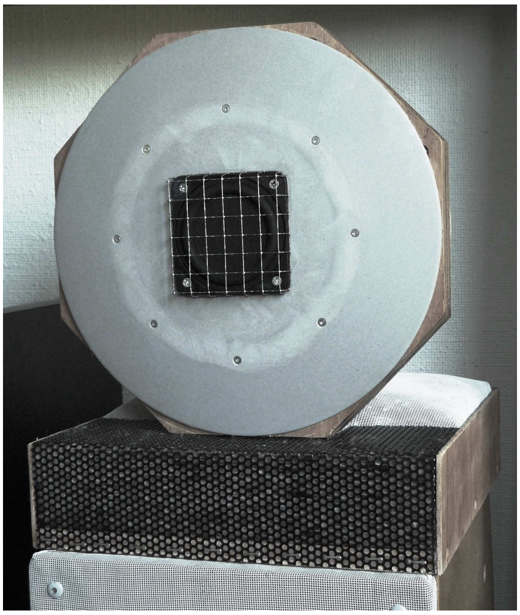
The Finishing Touch

The eventually cross over filter looks like:



The 'voorste BMR' means: the BMR in the front of the baffle. The other one is at the rear side, which has been shunted by a resistor of 6 Ω. The effect is that the front speaker radiates about 2.5 dB more than the rear speaker which effects the total radiation pattern to give a better sound image.

To enhance this once more, the footplate should be covered with a thick wool blanket to avoid reflections. This acts apon the transparency.



The BMRs compared to the ESLs.

Is the difference with my ESLs of that time so great? After all the great pains described on this site, the ESLs have been pulled back for a wile. They sound still great to me. I did not hear them with the SSA120 amp before. On the sweet spot the position of the instruments in the phantom image is more precise as with the BMRs. The phantom stage is smaller (no any sound comes from outside the two ESLs) but the placement of the instruments (and other sound sources) is that accurate it never could be heard live! However, listening to 'Gruppen for three orchestras' (Karlheinz Stockhausen) in which is no melody nor rhythm, only dynamics and placement of the instruments, it has its charms.
Bear in mind however that the radiation pattern of ESLs is a dipole: the back side of the membrane moves in the opposite direction of the front side, so an ESL does not function as an omnidirectional aerial. The sweet spot becomes unpractically small.
Listening to eg. the Berlage Saxophone Quartet in 'Search of Freedom', I prefer the BMRs: I can walk through my room and still get the suggestion to bee in the room with the players.

The dynamics and the loudness of the ESLs compared to the BMRs did not really frustrate (-6dB). Mind that the BMRs have been put in series.

27-9-2018

A week later Pieter asked me if my ESLs were still available. He would like to listen to them again. Beside Pieter Meijer, came Gertjan Groot Hulze.
They had better memories about the ESLs which had not played for years…! The reproduction was disappointing. Returning to the '2 BMRs in a Baffle' was a relieve. But, still there is a tendency of sound-sticking to the speakers! This was mainly the case with close mice-ing as with Diana Krall (track 4 on the CD: The girl in the other room).
On Sunday 7-9'18 I listened to 'Diskotabel' over the FM-radio and I did not notice any tending to this sticking…. so…

What next?

One is never through with working on sound systems! Could there be any problem with my amplifier? Comparing the SSA120 with the SSA30 proved that there must be something wrong with my SSA120 !!! The SSA30 sounded much better!
Eventually C2 in the feedback (to make the amplification x1 at DC to reduce the offset voltage at the loudspeaker terminals) of the SSA120 has been short circuited, which seemed to be the bad guy!

15-11-2018