MY SEARCH FOR THE IDEAL STEREO LOUDSPEAKER

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Stereo loudspeakers are usually operated in a reverberant environment and meant to create a phantom aural scene, a believable illusion in the listener's mind. But what exactly must the loudspeakers and the room do to optimally support such illusion and not distract from it? Loudspeakers always held a fascination for me and I designed, built and listened to many prototypes in a seemingly never-ending search for the ideal loudspeaker for my home stereo system. For thirty-seven years of my professional life I was involved with research and development of electronic test and measurement equipment covering the frequency range from 10 kHz to 22 GHz. In terms of wavelengths this is equivalent to acoustics from 0.01 Hz to 25 kHz. Loudspeakers have many analogies to electromagnetic antennas. I will talk about the progression in my understanding and design of loudspeakers. In the end it was primarily about the radiation pattern, though I did not recognize its overriding importance initially. The ideal stereo loudspeaker has a frequency independent polar response.

INTRODUCTION

Hearing is subjective. Hearing evolved for survival in natural environments full of sounds and reflections where aural illusions would be dangerous and must be avoided. But the essence of stereo is the creation of a believable aural scene using two loudspeakers in a room. My loudspeaker journey started with modification of commercial products and by questioning their design rational. I began to design box loudspeakers of various sizes and internals and with passive and active crossovers. Spurious box radiation proved difficult to avoid and became the impetus to investigate various configurations of open baffle speakers with conventional piston drivers. Today I know that stereo loudspeakers must have a frequency independent polar pattern in order to create an aural scene that excludes both, loudspeakers and listening room, from attention.

While working on electronic test equipment design in the Microwave R&D Laboratory of Hewlett-Packard Co. in the early 1960's many of the engineers pursued personal side projects in their spare time. So we designed, for example, our own FM receivers with pulse rate demodulation and phase-locked stereo decoders, pre-amplifiers, phono stages, very low noise moving coil preamps, solid-state power amplifiers and whatever was needed to assemble a top notch Hi-Fi system. HP actually encouraged such personal projects as long as they were on our own time and not commercial, figuring that what we learnt in the process would benefit the company, which it did.

1 LEARNING FROM THE PRO'S

Loudspeakers seemed far from our level of expertise and so we frequented the Palo Alto Hi-Fi store, listening to the latest loudspeakers. I bought what I thought I could afford and liked for its sound, though sometimes with trepidation about its visual impact at home. My first loudspeakers were a pair of KLH-6 (Figure 1).

Figure 1: Two-way loudspeaker with "acoustic suspension" woofer in a sealed box.

Removal of the grill cloth revealed a layout of drivers on the baffle, which immediately raised a question. Why was the tweeter mounted in the upper left corner on both speakers? We received the answer eventually: "It sounds best this way". The tweeter construction also was intriguing (Figure 2).

Figure 2: Tweeter with small dome and large paper surround

Next came a pair of ADVENT loudspeakers with more capable tweeters (Figure 3). Still, they would distort noticeably on loud piano passages.

Figure 3: Small ADVENT speaker with novel tweeter.

1.1 Making acoustic measurements

We had now begun to perform acoustic measurements indoors after Russ Riley had designed a parallel filter-bank audio analyser. (Figure 4). Lyman Miller had modified the electronics of an inexpensive electret-microphone capsule to lower distortion and measured the frequency response in comparison to a professional capacitor microphone. He is to this day an avid recording engineer with a sphere microphones and low noise microphone preamplifiers of his own design.

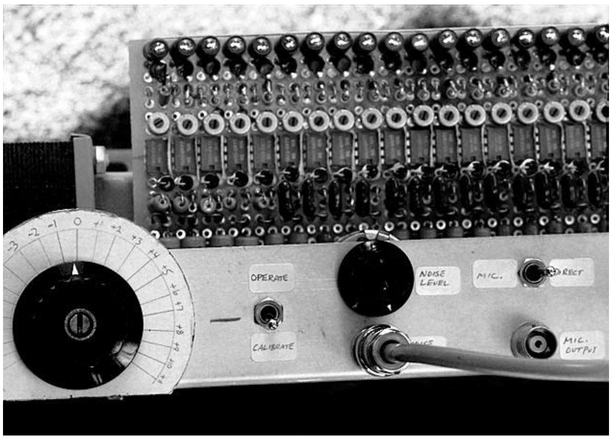

Figure 4: Audio analyzer using small light bulbs for power level indication.

The analyser was calibrated for equal light output from each filter when driven from a pink noise source. Noise was generated by a Germanium microwave point contact diode, which exhibits a 1/f corner above 10 MHz and thus plenty of signal in the audio range after some amplification. For acoustic measurements the signal from the noise source was applied to the loudspeaker via the power amplifier and the microphone output to the audio analyser. A variable gain stage with a potentiometer and scale calibrated in dB, allowed to adjust the brightness of the light bulbs to a reference illumination and thus to read the equivalent power level from each filter output. The resulting frequency response was plotted manually on graph paper.

We then used active line-level filters to equalise the frequency response. But what should be the target? It seemed obvious that the response at the listening location should be flat, except that doing so made the speaker sound too bright. The problem was that we had measured acoustic power in the room at the listening location and not just the direct response from the loudspeaker.

1.2 Refining commercial loudspeakers

More serious investigation into loudspeaker and room behaviour began for me with the ESS 7 (Figure 5). It sounded better in the store and also when I had brought it home than my equalised ADVENT speakers. Yet the new speakers did not measure particularly well (Figure 6). The drivers were arranged in an illogical pattern, the box had a mysterious air pressure relieve opening.

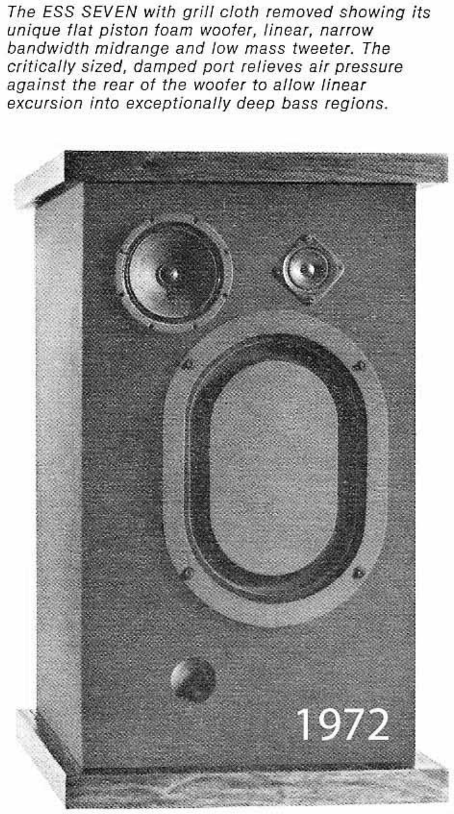

Figure 5: ESS 7 three-way loudspeaker
with vented box

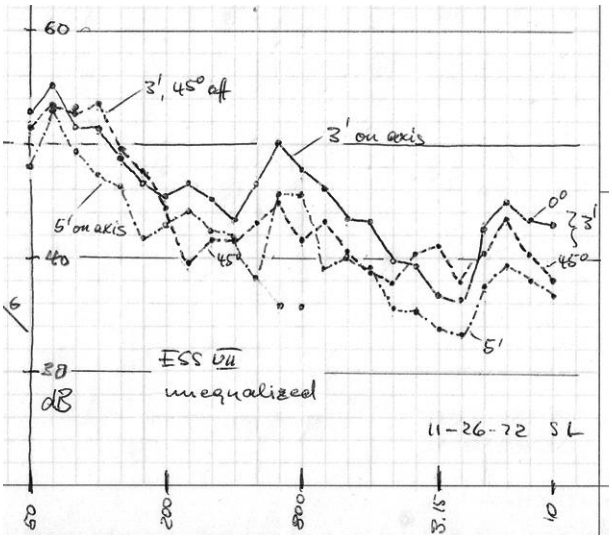

Figure 6: Frequency response of ESS 7 measured with pink noise source and a filter-bank audio analyzer.

I found out soon that the sound could be improved by replacing the internal stuffing with long fibre wool. In the end it turned out that the midrange and woofer drivers were the prime contributors to the speakers surprising clarity. Measurements of box panel vibration using a phono cartridge as velocity transducer revealed resonances in the mid frequency range. The large surface area made the vibrations audibly more significant than those from the small box speakers that Russ Riley was experimenting with (Figure 7).

Figure 7: Small 2-way loudspeaker DITTON 10

The smaller speakers suffered from insufficient bass output but handled electronic boost of the bass quite well, if it was done precisely by pole-zero compensation. This form of equalisation led to the single IC biquad circuit (Figure 8), often called “Linkwitz Transform” [1].

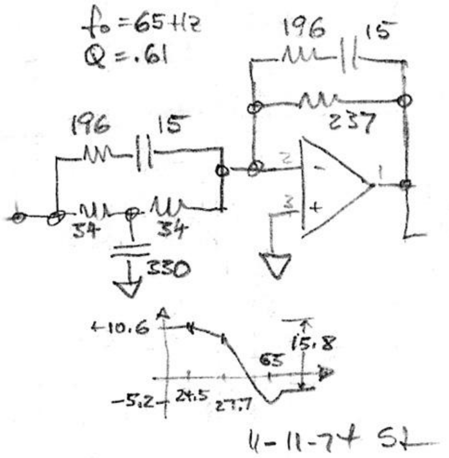

Figure 8: Biquad circuit to control the low frequency roll-off behavior of a sealed box woofer or tweeter

We observed that sealed box bass had less audible overhang than a vented or passive radiator system and that equalisation could effectively compensate for a small internal box volume provided that the drivers had sufficient excursion capability and cone area for the desired sound volume.

I met Laurie Fincham, R&D manager of KEF, in 1973 when he came with his team to HP for training on the new FFT Analyzer, which they had bought for acoustic and vibration measurements. I saw this as my chance to get answers to all our loudspeaker design questions, only to find out that they had some of the same questions. Many airmail letters and packages went back and forth between California and Kent in the following years and we came up with answers. Crossovers and group delay distortions were among the hot topics at the time.

2 DESIGNING MY OWN BOX-SPEAKERS

It was apparent to us by now that loudspeakers should be physically small to minimise spurious panel radiation. A narrow front baffle would yield wide horizontal dispersion, drivers become forward directional with increasing frequency and radiation in vertical direction was a function of crossover and driver spacing. I now experimented with my own box speakers, which led to a satellite/subwoofer system (Figure 9). It served in my home for many years and was duplicated in many places, DIY style, after I published a detailed description in Wireless World magazine [2, 3]. The satellite speakers eventually were changed to a taller box with an M-T-M layout of drivers for increased output capability and to evaluate different types of crossover filters for audible effects. In an MTM arrangement the maximum addition of driver outputs occurs always on the tweeter axis, regardless of phase differences between tweeter and midrange outputs.

Figure 9: Hanging satellite speakers and subwoofer

The subwoofer was placed in a bookshelf, was internally braced and then pressed down by the weight of books to suppress radiation from the panels. Left and right woofer channels were summed to mono, because stereo was supposed not to matter at low frequencies, but more importantly it reduced cone displacement by cancelling vertical rumble from phono playback. The satellite speakers were hanging at seated ear height and about 1 m from the bookshelf. Placing the speakers into their storage places, into the shelf and flush with the books, degraded imaging.
Internal construction and bracing of the satellites was extensive in order to increase stiffness and to push any panel resonances into the tweeter frequency range where they are less likely to be excited (Figure 10). Internal air cavity resonances were suppressed with long fibre wool as much as possible without decreasing low frequency output. The thin woofer cone is not much of a sound barrier for any remaining air resonances and acts more like the skin on a drum.

The woofer driver was mounted by its magnet and mechanically decoupled from the front panel except for a foam air seal (Figure 11). The driver basket was stamped and formed out of sheet metal and was staked to the massive magnet assembly. When the basket was tightened with screws to the front baffle, then the compliance of the basket and the mass of the magnet formed a spring-mass system with a high-Q resonance in the region between 200 Hz and 400 Hz depending upon driver size. Mounting by the magnet eliminates the resonance.

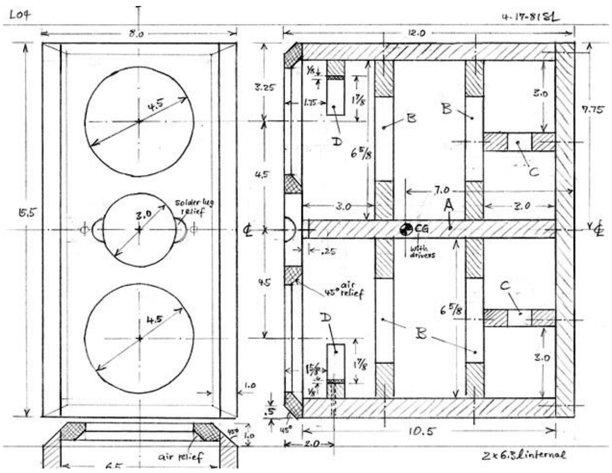

Figure 10: Internal box construction

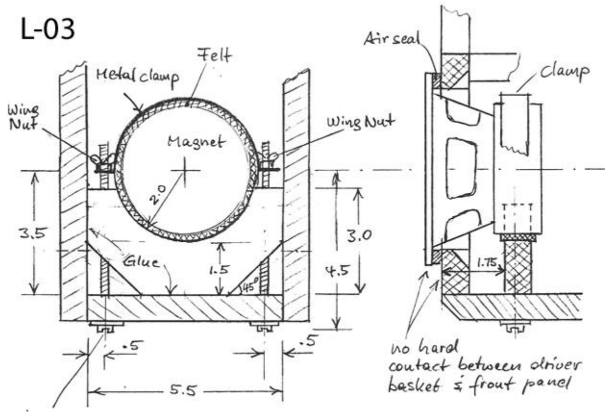

Figure 11: Magnet mounting of driver to eliminate panel-mount resonance.

2.1 Shaped toneburst testing

Our acoustic measurement capability was increased with the design of a generator that produced a 5-cycle sinewave burst with raised-cosine envelope and covering a constant percentage bandwidth [4, 5]. The burst frequency was continuously variable to find resonant behaviour in the frequency response of a driver under test as seen in the decay of the burst envelope (Figure 12). The peak level of the envelope was plotted versus frequency, giving a graphical presentation that correlated strongly with listening impression. The burst test was used before FFT analysers with post-processing of the impulse response became available. Shaped burst signals are used to this day to safely test for and hear the onset of audible distortion of a loudspeaker when driven with increasing voltage. When used for room acoustic tests the time domain behaviour of room reflections versus burst frequency is readily seen in the envelope (ETC) of the displayed waveform.

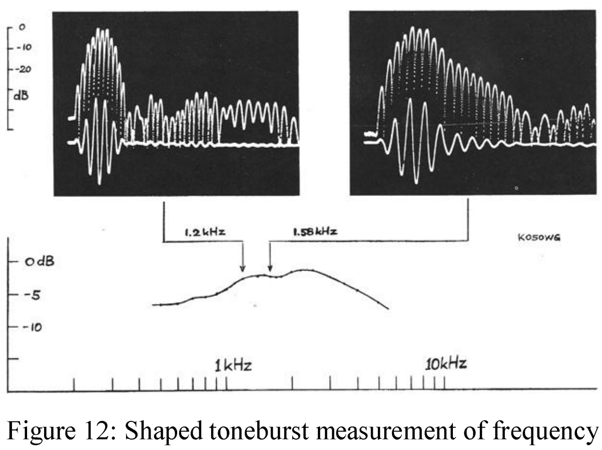

Figure 12: Shaped toneburst measurement of frequency response and time domain behaviour.

2.2 Crossovers and polar response

The driver selection and their layout on the front baffle of the ESS 7 (Figure 5) or of the Celestion (Figure 7) loudspeaker in combination with their crossover filters was driven by achieving a desirable listening experience. The design process was highly empirical at the time, relying mostly on a pair of "golden ears". When looked at as an antenna design problem, then each driver is like an antenna element with its own polar magnitude and phase frequency response and the separation of elements determines the path length differences to a distant observation point. The different drivers add according to their relative phase and magnitude at the observation point in space. Furthermore when each driver is driven via a magnitude and phase changing filter, such as a crossover network, then that will further affect the output summation.

In our opinion drivers should have been aligned along a vertical line and as close together as possible in (Figure 5) and (Figure 7) in order to minimise path length differences in both horizontal and vertical planes. Also the driver layout should be symmetrical for left and right speakers for balanced stereo imaging. Wide dispersion is more important in the horizontal plane. People are usually seated and a change in sound when standing up may be acceptable.

Lowpass and highpass filters of odd-order Butterworth crossover networks are in phase-quadrature at all frequencies, while even-order Linkwitz-Riley filters are in-phase at all frequencies [6, 7]. This has the effect that L and H outputs from an acoustic LR crossover add maximally when their path lengths to a listener are identical, which is on the line of symmetry. If, for example, the drivers are spaced by one wavelength λ at the crossover frequency, then at +/-30^0 off-axis the path lengths differ by $\lambda/2$ and the outputs cancel (Figure 13).

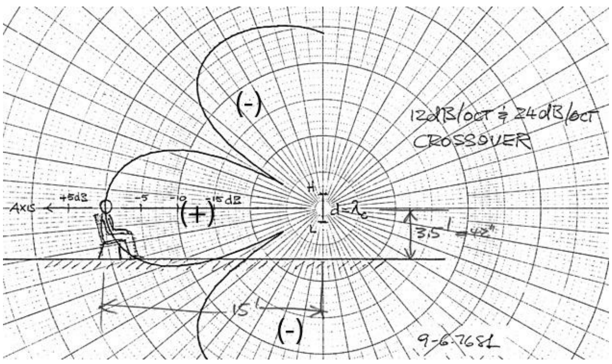

Figure 13: Vertical polar pattern of two point sources separated by λ at the crossover frequency for a Linkwitz-Riley filter.

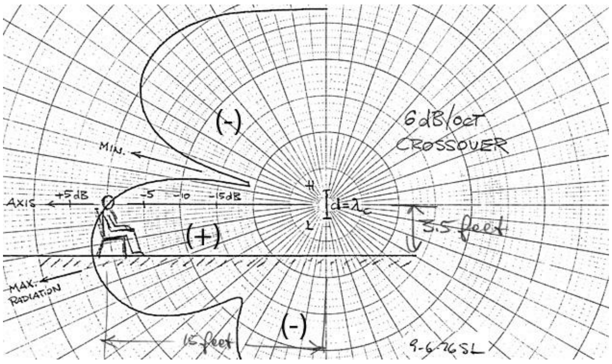

Figure 14: Vertical polar pattern of two point sources separated by λ at the crossover frequency for a Butterworth filter.

Under the same conditions the output addition for a Butterworth filter produces an interference null at $+14.5^0$ when the L output lags by an additional 90^0 i.e. by $\lambda/4$. Maximum output of +3 dB occurs at -14.5^0 when M and L are in phase (Figure 14). The axis of maximum radiation points even further downwards, if the acoustic centre of L lies behind that of H. The offset can be compensated by adding electrical delay or phase lag to H.

Radiation in the vertical plane is dominated by the separation of drivers in combination with the polar patterns of the individual drivers. In the horizontal plane where L and H are equidistant, baffle edge diffraction and the polar patterns of drivers dominate the off-axis response. Overall a box loudspeaker is omni-directional at low frequencies and becomes increasingly forward directional with shorter wavelengths and higher frequencies (Figure 15). A smaller baffle area and smaller drivers extend the frequency range before beaming begins. Maintaining a flat frequency response on-axis means that less acoustic power is radiated into the room as frequency increases. Power varies typically more than 10 dB between low and high frequencies and with it the timbre of the room reverberated sound. Also proportionally too much power is available to excite low frequency room modes.

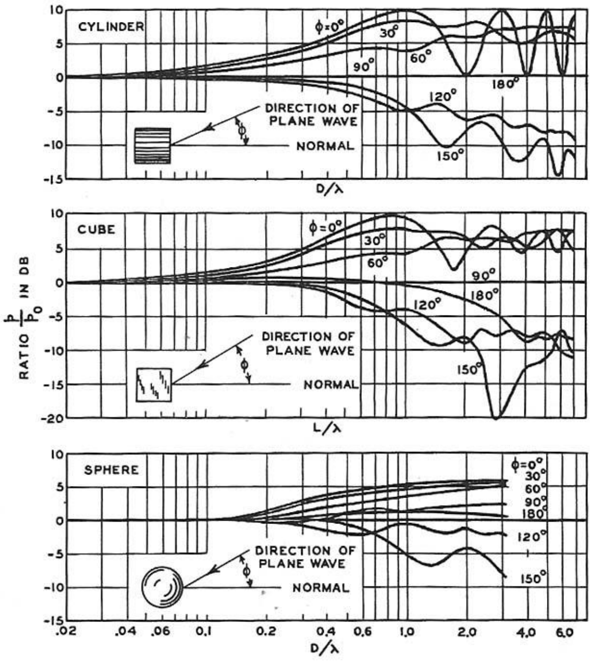

Figure 15: Off-axis radiation from a point source on a cylinder, cube and sphere [8].

3 OPEN-BAFFLE LOUDSPEAKERS

In 1984 I became involved with setting up a satellite down-link for an A/V presentation with a large audience. The venue was an indoor basket ball court. The existing PA system was not useable for listening to long speeches or folkloric music due to the high degree of sound reverberation. I decided to design an open-back line source of 2.5 m length, using twelve 6.5 inch drivers (Figure 16). At high frequencies only the four innermost speakers are active. The on-axis frequency response was equalised outdoors with the speakers firing down the street in front of our house. The column loudspeaker was placed between the edge of a large projection screen and a podium with microphone for the local announcer. Acoustic feedback was very low with the microphone placed essentially in the dipole null. The directivity of the line source provided excellent intelligibility even in the far corner of the court. The audience in front was not overwhelmed by sound because the bottom of the source was raised above head height.

The line source was built in two pieces for easy transportation. When I took it home I set it up as a pair of stereo loudspeakers in my living room to hear what it would sound like. I really surprised myself.

Figure 16: Column loudspeakers with open back

Despite a roughness in timbre they had an openness of sound, which reminded me of the Quad ESL-63 (Figure17), an electrostatic loudspeaker that I admired very much for its sound and design [9]. But the problem with electrostatic loudspeakers is the weak motor force, which requires to low mass, large area radiating surfaces to obtain adequate sound output from small excursions. Dynamic capability is usually lacking at bass frequencies and high frequencies are radiated in multiple beams. Room placement is notoriously difficult. Yet the sound is open and transparent, providing a natural experience.

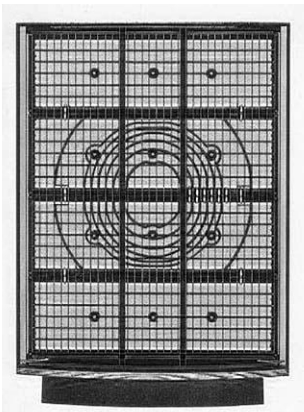

Figure 17: Electrostatic loudspeaker with concentric rings to control the radiation pattern.

I decided to find out if the desirable characteristics of the planar ESL could be preserved and the undesirable dynamic and beaming limitations overcome, by using acoustically small piston drivers in an open baffle. Those drivers then must have very large excursion capability to compensate after equalisation for the front-to-back acoustic short circuit. After extensive experimentation, measurement and electronic equalisation I came up with a 3-way open baffle loudspeaker with sealed box woofers (Figure 18). The woofer channels were summed to monaural into a sealed box woofer.

The open-baffle speakers rendered a sound stage that was more 3-dimensional than that of the box speakers in (Figure 9). There the stage appeared like the view through a narrow horizontal window. Here the stage had both depth and height.

Figure 18: Open-baffle loudspeaker with sealed box center woofers.

Figure 19: Rear view showing magnet mounted lower midrange drivers and rear facing tweeter.

An open-baffle speaker has several advantages over a box speaker. Most important is the constructive use of the rear radiated sound power to provide directivity instead of trying to dissipate it inside a box. Directivity reduces sidewall reflections at the listening location depending upon the toe-in (Figure 20).

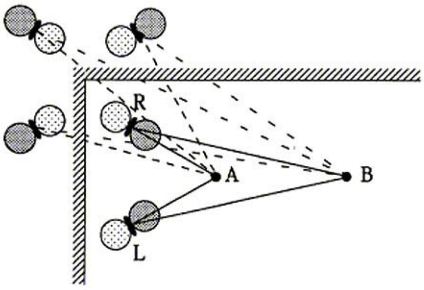

Figure 20: Corner reflections for a dipole.

Total radiated power for a flat on-axis response is less from an open-baffle speaker and more constant with frequency than for a box loudspeaker (Figure 21).

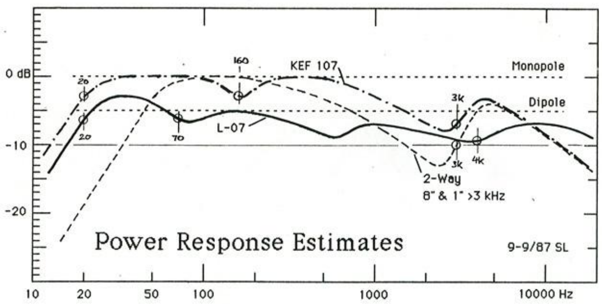

Figure 21: Estimated power response of L-07 open-baffle speaker compared to 2-way and 3-way box loudspeakers.

3.1 Adding a dipole woofer

The speaker system of (Figure 9) used a box woofer because I thought that a dipole woofer would need to be very large and therefore could not be placed in my living room. A colleague at HP, Brian Elliott, had designed an H-frame open-baffle woofer (Figure 22). He had stacked three of these modules against each sidewall of his listening room for a total of twelve 12 inch drivers. These woofers delivered the most realistic and effortless bass reproduction that I had ever heard.

Don Barringer, at the time a recording engineer for the US Marine Band in Washington, combined two of the modules in (Figure 22) with his built of the loudspeakers in (Figure 9). He immediately reported satisfactory results, which encouraged me to add H-frame woofers to my own system, confirming his observations and making me wonder why they sounded more realistic than my bass box [10].

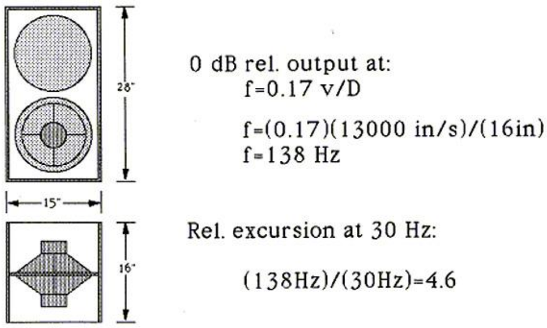

Figure 22: Open-baffle, H-frame woofer. Cone excursion requirement relative to a sealed box woofer.

3.2 Almost full-range dipole designs

A follow-on open-baffle speaker design was based on a M-T-M midrange and tweeter arrangement and a reduced width H-frame woofer for a 3-way dipole loudspeaker (Figure 23).

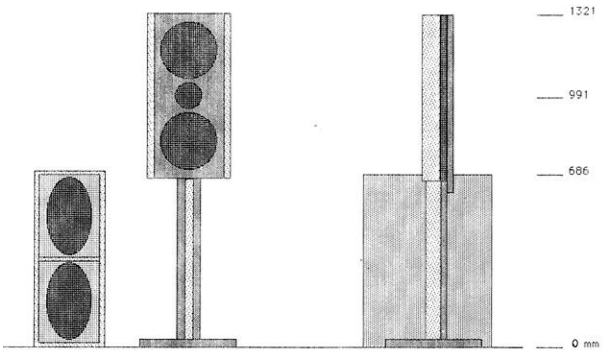

Figure 23: Compact 3-way dipole loudspeaker [11, 12].

The rear tweeter was eliminated, because high frequency reflections from the wall behind the speaker seemed problematic. Also the wide dispersion of the tweeter compared to the narrow dispersion of the two midranges around the crossover frequency would seem to balance out the total radiated power. The design concept led to a line of open-baffle loudspeakers, which I designed for Audio Artistry. They were highly praised in the Audiophile Press and still have their followers today. The flagship was the "Beethoven-Grand" (Figure 24). It used the best drivers I could find in terms of low non-linear distortion and energy storage.

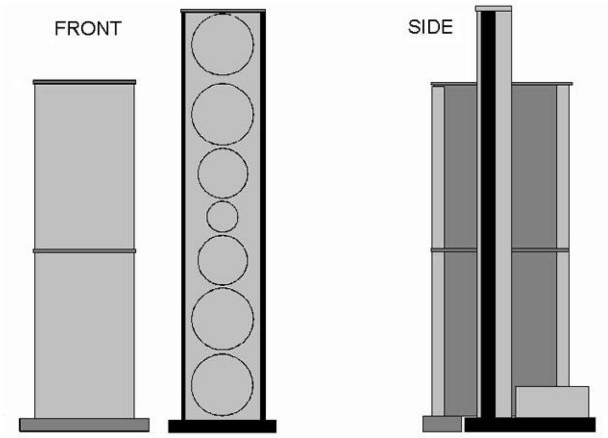

Figure 24: "Beethoven-Grand" by Audio Artistry

The single 1inch tweeter was flanked by two 8 inch upper midrange drivers, followed by four lower midrange 10 inch drivers. Each woofer used four 12 inch drivers. They were mounted at an angle to reduce baffle width. The drivers altered their orientation in a push-pull sequence to reduce even-order distortion.

After I retired from Audio Artistry I launched my website www.linkwitzlab.com to publish what I had learned about the benefits of open baffle speakers with piston drivers. I loved how they sounded in my room and that they would sound essentially the same in another room when set up with a minimum distance from large reflecting surfaces. Other music lovers should have access to this technology, which even today is hard to find in commercial products. So I designed and offered construction plans for a new speaker, which illustrated my best practices. Again the woofers were in separate enclosures. I found strong interest and immediate acceptance of the design concept in the DIY community worldwide. The difference to a box loudspeaker was so clearly audible.

The advent of woofer drivers with very large excursion capability allowed me to integrate the woofer with the midrange and tweeter in the same cabinet (Figure 25). The contour of the side baffle suppresses the typical $\lambda/4$ cavity resonance of the H-frame woofer and is also intended to give a more pleasing shape to the "ORION" loudspeaker. Many hundreds of them have been built DIY style all around the world and to great acclaim.

Trusting Don Barringer's astute hearing ability and comparing his observations to my own refined the ORION over time [13, 14]. First was the addition of a bracket to mount the midrange driver by its magnet for vibration decoupling from the front baffle. Later came the addition of a rear tweeter and changes in equalisation .

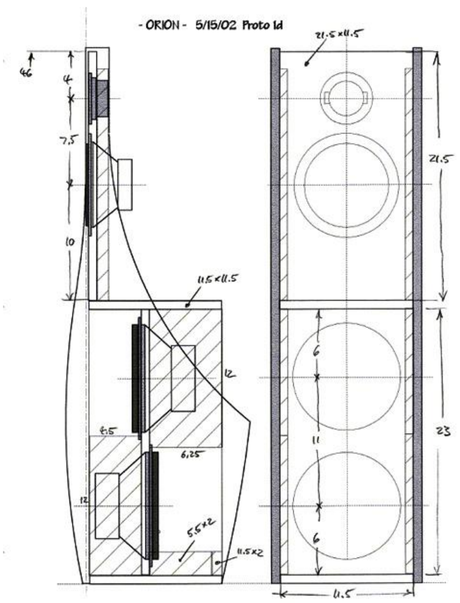

Figure 25: Woofer, midrange and tweeter integrated in one open baffle

Figure 26: The shape of the side baffle meets acoustic and cosmetic requirements.

Figure 27: Wide dispersion 2-way loudspeaker

4 THE MONOPOLE AS PROTOTYPE

The reintroduction of the rear tweeter was prompted by a series of events. I had built a rectangular box as a 10:1 scale model of a room in order to study the spatial distribution and excitation of resonant modes. A unique 2 inch tweeter with a 200 Hz free-air resonance was suitable to configure monopole and a dipole sources. They had to provide sufficient output down to 50 Hz, equivalent to 5 Hz in a real room. The test results for mode distribution and strength met my expectations, except that I had built a reverberation chamber. I had not realised that a 10:1 model of a real room with RT_{60} = 500 ms must have RT_{60} = 50 ms for the model.

The tweeter gave me the idea and possibility to build a loudspeaker with minimal baffles in order to study diffraction effects. Compared to the ORION in (Figure 25) the "PLUTO" loudspeaker in (Figure 27) and (Figure 28) shows diffraction only above 3 kHz when radiation changes gradually from being omni-directional to forward directional. The crossover frequency between the two small drivers is at a low at 1 kHz.

I was surprised how similar the two loudspeakers sounded, though their radiation patterns are quite different. I attributed this to the similarity in timbre of the reverberated sound in the room. PLUTO though gave a more realistic rendering of high-pitched voice. The addition of a rear tweeter to ORION improved the situation. Clearly the power response of the loudspeaker was more important than I had observed so far. Consequently I postulated that the monopole, the pulsating sphere, the omni-directional and acoustically small radiator with its frequency independent polar pattern is the basic prototype of a loudspeaker for rendering stereo in reverberant spaces. PLUTO is a reasonably close approximation [15]. A dipole, a cardioid or any other directional loudspeaker design with frequency independent radiation pattern would be an even more appropriate prototype. Greater directivity brings a higher ratio of direct to reverberant sound and is the desirable direction to pursue.

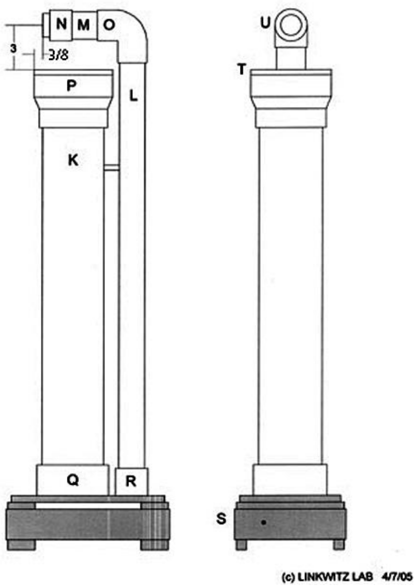

Figure 28: Mostly omni-directional loudspeaker "PLUTO" with sealed woofer and tweeter enclosures

Figure 29: Open baffle construction for a full-range dipole loudspeaker [16].

5 THE DIPOLE AS IDEAL PROTOTYPE

The wide front baffle of ORION (Figure 25), which had been thought to be necessary for adequate acoustic output from an 8 inch driver in combination with a 1 inch dome tweeter had led to compromises in radiation pattern. After much experimentation, testing and measuring the polar response for different driver combinations and baffle shapes, I concluded that an upper midrange driver on a narrow baffle had to be added (Figure 29). Two 1" dome tweeters separated by a relatively wide baffle form a dipole like polar pattern. The tweeters see each other only at large off-axis angles, being directional of their own due to their size in the frequency range of use above 7 kHz. Upper and lower midrange driver outputs are combined by a 1st order 1 kHz crossover filter and form a very wide bandwidth dipole up to 7 kHz. A V-frame dipole woofer takes over below 120 Hz.

I call the loudspeaker "LX521" (Figure 30) because I had arrived at a useable baffle shape on May 5, 2012 and it is the 21st speaker that I designed over the years. I consider the LX521 to be a satisfyingly close approximation to a dipole, which in itself is the prototype for a directional loudspeaker that is ideal for rendering stereo in a reverberant environment.

Figure 30: LX521 dipole loudspeaker.

Only dipoles are directional to the lowest frequencies. I have observed that loudspeakers, which are omni-directional at low frequencies and then become highly directional below a kHz, such as some horn, coax or waveguide speakers, tend to image like headphones at a distance. A phantom centre voice may appear even in front of the line between the speakers. The aural scene is warped and shifts back into left and right speakers with small lateral head movements. With dipole or monopole loudspeakers the aural scene is smoothly distributed between and sometimes beyond the physical sources.

6 THE ROOM

With PLUTO-2.1 and LX521 I now have two loudspeakers that I can live with for a long time, enjoying the best that stereo is capable of in terms of spatial rendering and realism. I am convinced that few people have ever experienced this. In particular recording engineers should hear their work fully revealed to appreciate what has been accomplished and what more could be done. The listening room is always involved when stereo is reproduced over loudspeakers. It is most important that the loudspeakers are set up with a minimum distance of 1 m from large reflecting surfaces. In that case the reflected signal is delayed by more than 5 ms and the listener's brain can treat the room's response separately from the direct sound coming from the speakers to the ears (Figure 31). The room should not be overdamped, be diffusive behind the loudspeakers and absorptive behind the listener [17-23].

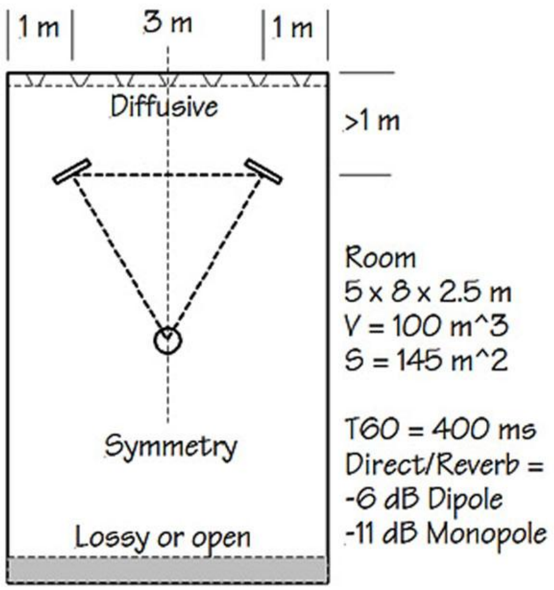

Figure 31: Loudspeaker, listener and room configuration for rendering stereo.

Monopole speakers require a shorter listening distance and thus a smaller loudspeaker-listener triangle than dipole speakers for equal phantom scene detail. The ratio of direct to reverberated sound should be at least –6 dB (Figure 32). A normal living space can fulfil the setup conditions quite adequately and without special room treatment products (Figure 33).

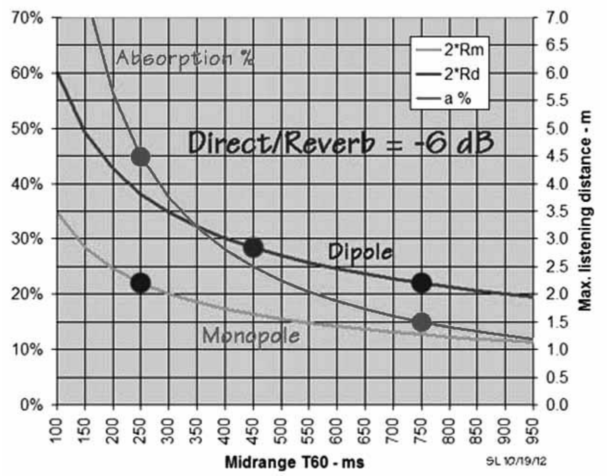

Figure 32: Maximum listening distance and absorbent room surface area for a given reverberation time.

Figure 33: My living and listening room.

7 CONCLUSIONS

My search for the ideal stereo loudspeaker has come to a satisfying end with the conclusion that it must be a loudspeaker with constant directivity. The ideal can be approached over most of the frequency range, but it takes some effort. Hardly any of today's consumer loudspeakers or professional studio monitors come close to the ideal. Thus the potential of 2-channel stereo is rarely realised. Audio has moved on to surround sound, which in most cases is merely overpowering the brain's natural directional hearing processes with large numbers of loudspeakers, source directions and sound effects. One can only hope that rendering stereo takes a last step forward with constant directivity loudspeakers, which engage the room and the brain's discriminatory ability in a natural way.

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