Pipeline

Pipeline Transmission Line Line Array

A conventional dynamic loudspeaker has a more or less spherical radiation pattern. If a number of identical drivers are placed in a vertical line, then you get a totally different radiation pattern. Instead of the spherical radiation pattern, you get a controlled cylindrical pattern. This has the advantage that vertical room resonances are minimized because the room is excited over more than half the height of the room, so you get an even pressure response.

Characteristics

Besides the controlled radiation pattern, a line-array has a number of other interesting characteristics. Due to the cylindrical radiation pattern of mid- and high frequencies there are less reflections from the floor and ceiling so more direct and therefore less indirect sound reaches the listener. This results in a more stable and better focussed image. The image of a line-array is therefore room-filling and constant over a wide listening angle. Also by combining a large number of drivers in a loudspeaker the maximum dynamic range is increased and the level of distortion is decreased. After all, each individual driver has to produce only a small part of the total sound pressure level. Another positive characteristic of a line-array is that the total cone-area is divided over lots of small, individually driven masses. This increases the perception of "speed" and dynamic expression. The Pipe-Line's full-range driver has a cone area of only 30cm2 and a moving mass of a mere 3,1 grammes! Each enclosure uses 16 of these drivers which results in a total cone area of about 480cm2 which is equal to that of a single 12" woofer while the ratio moving-mass Mms / drive force Bl remains the same. In this case a nice high acceleration factor of 935! As with any loudspeaker system, a line-array also has it's limitations. Seeing that each driver of the array produces the same frequency range, the theory predicts a cylindrically shaped sound source that only because spherical at a very large distance. In practice (in a normal living-room) you listen to a line-array in the near-field in which you have one large problem: due to the difference from each individual driver to your ear, you get cancellation at high frequencies. A solution would be to build a curved array (a sort of banana shape) but that would only be ideal for one distance, that of the radius of the array. This would make the sweet-spot rather small so this would make the system a one-person-only set-up, not very hospitable. Furthermore a line-array (like any speaker) has the "baffle-step-problem". Any loudspeaker, once mounted in an enclosure, has a stepped down low-frequency response, usually about 5 to 6dB's quieter than the higher frequencies. With most loudspeakers this step occurs around 500 to 800Hz depending on the width of the enclosure.

The little wonder

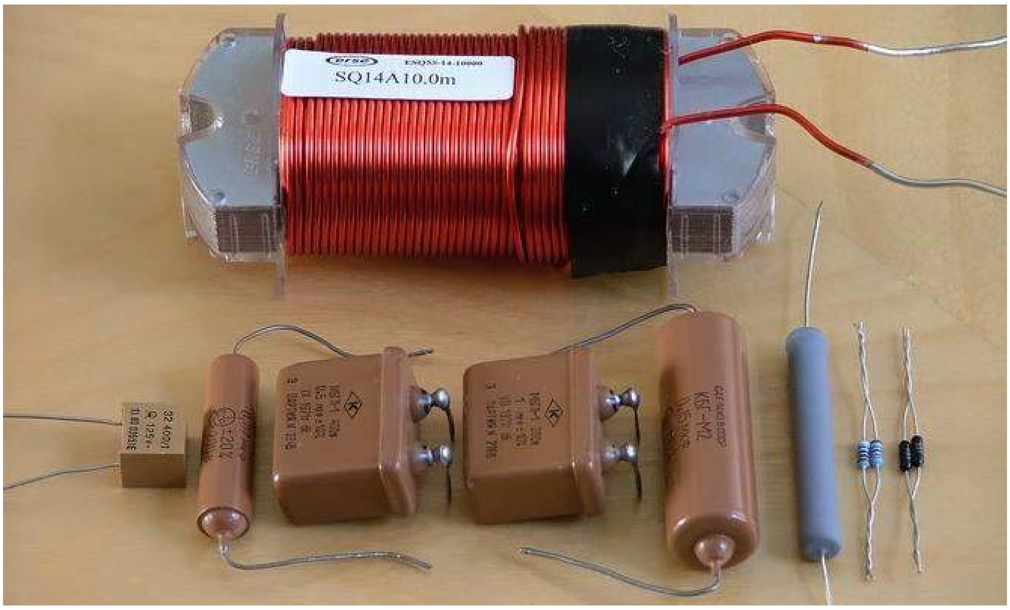
The full-range driver used in the Pipe-Line is the Chinese built Hi-Vi (Swans) B3S. A total of 32 pieces are used to create one pair of loudspeakers. It is a small driver with a faceplate of 78x78mm and a one-piece convex cone made from bronze anodised aluminium. There is no phase-plug or dust-cap in the centre of the cone. The cone is held into position by a rubber surround and on the rear we see a relatively large magnet shielded system. The drivers are mounted on the rear of the baffle so that they can "breathe" enough. Were they to be mounted from the front, then there would be a large possibility that you would close off the rear of the driver! Such a small driver is of course not very efficient and a single driver only produces 82dB's for 1 watt input power. The resonance frequency (80Hz) is rather low for such a small driver but still don't expect earthquake bass from this full-range unit. With an X-max of +/-3mm there is room for some dynamics especially considering there will be a total of 32 of these little chaps in the final system. The frame is a standard quality pressed steel job with a few extra ventilation openings here and there. The voice-coil former can be seen between the rear suspension and the front pole-plate so compression inside the motor-system should be minimalised.

Transmission Line

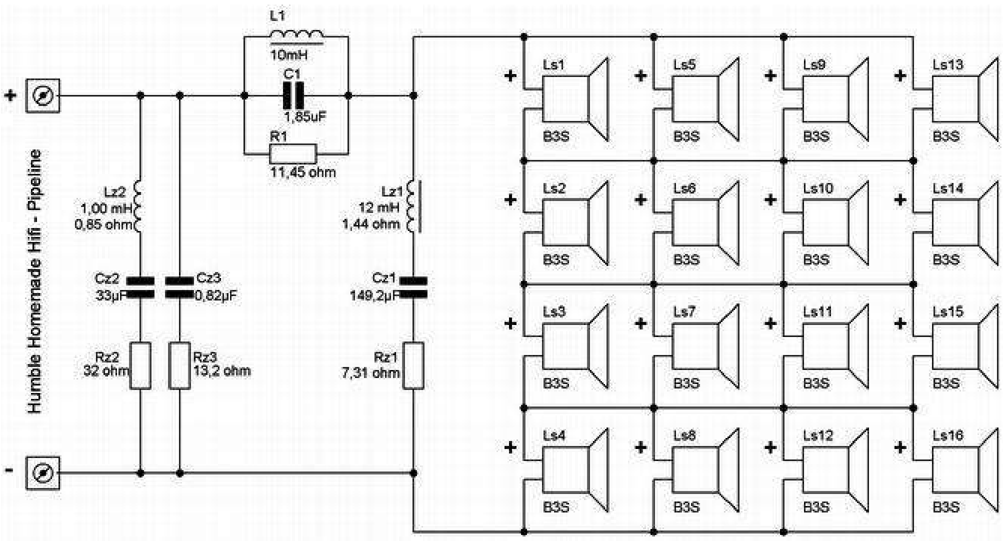
The cabinet in which these drivers are mounted is also a line: a transmission-line. Basically it is an straight "organ-pipe" that is open at the top to make maximum use of the low frequency energy radiated from the enclosure. The internal height of 150cm should coincide with a quarter wave length of about 57Hz, low enough for such small drivers. To create a stable construction, the cabinet is placed on a base-plate of 435x330mm with adjustable spikes at each corner. All internal walls except the baffle are covered with Intertechnik Damping-30. This is a 3cm thick damping pad made from recycled cotton. The lower two-thirds of the cabinet is lightly filled with Intertechnik Sonofil. The combination of the cabinet with the array of drivers results in a -3dB point of about 80Hz and a -6dB point of about 60Hz. The cabinets are made of 18mm mdf except for the baffle which is CNC-milled from a piece of 12mm thick Corian artificial stone. CNC-milling of the baffle makes life a lot easier because there are 32 driver holes to cut out and 172 holes for the mounting screws, etc! I used black Hex-head bolts with brass rings that matched nicely with the black Corian and bronze coloured drivers. A higher resolution drawing is available on request.

Spiderman

Here we are then: we've got a loudspeaker with not much bass due to the baffle-step and not much treble due to cancellation. It sounds like a megaphone and doesn't look very promising but there is a solution for every problem: Spiderman! No, just joking, I mean a correction network. This can be done by means of active correction but I chose for the easiest and cheapest solution by means of a passive LCR-network directly in the signal-path of the line-array drivers. This does kill some efficiency, but we had too much energy in the midrange anyway. At a first glance the crossover schematic looks rather complex but at a second view it only consists of correction networks, one to correct the acoustic frequency response (L1+C1+R1), three to correct the electrical impedance response. L1 lets through frequencies below a certain point, C1 lets through high frequencies above a certain point. R1 determines how strong L1 and C1 work. If R1 were to be infinitely large then all the energy would be distributed between L1 and C1 and nothing would go to the midrange in between. If R1 were to be infinitely small (zero) then all the energy would pass the network without altering the frequency response. So by playing with the value of R1 you can “tune-in” the amount of midrange to match that of the bass and treble. By playing with the values of L1 and C1 you can adjust the bandwidth in which R1 will work. The quality of these three components is of crucial importance for the final tonal character of the system. For inductor L1 I chose one with very low Rdc so bass gets through without too much loss. You are rewarded by stronger bass with better definition compared to a cheap inductor with high Rdc. Capacitor C1 is made from a combination of several smaller values wired in parallel. I used some New Old Stock paper-in-oil types (in this case ex Russian military types from 1971) and a 0,1uF Jantzen Audio Superior Z-Cap. For the bypass I used a small 32nF Silver-Mica capacitor to bring out the top-end of the full-range drivers nicely. Resistor R1 is also built-up from several values wired in parallel including some 0,1% precision metal film types that seemed to add just that little bit of extra micro-detailing. The value of L1 is rather large and therefore sets in quite early.



Because the loudspeaker / cabinet combination shows an impedance peak around 120Hz, this peak had to be flattened first by Lz1/Cz1/Rz1 so that L1 could do it's jog properly. Without Lz1/Cz1/Rz1 the bass would have a bump in the output and sound muddy and boxy. It is important that the total resistance of this impedance correction network should stay the same if you decide to use a different inductor, so the value of Rz1 and Rdc of the inductor must always add up to the same total value. There are two more impedance correction networks connected across the input terminals. These are used to create a smooth overall system impedance so that the loudspeaker isn't critical to the amplifier connected. So you can use these loudspeakers with just about any amplifier on the planet that produces at least 10 watts per channel.



Crossover parts list

L1 = 10mH Erse Super-Q inductor - Rdc = 0,26 ohms

Lz1 = 12mH iron-core inductor - Rdc = 1,44 ohms

Lz2 = 1,0mH air-core inductor - Rdc = 0,85 ohms

C1 = 1,72uF New Old Stock Paper-in-Oil + 0,032uF Silver Mica + 0,1uF Jantzen Audio Superior Z-Cap; total value = 1,85uF

Cz1 = 100uF + 47uF bi-polar electrolytic + 2,2uF Mundorf M-Cap; total value = 149,2uF

Cz2 = 33uF Mundorf M-Cap

Cz3 = 0,82uF Mundorf M-Cap

R1 = 12 ohms / 10 watts MOX + 4x 1000 ohm 0,1% Precision metalfilm; total value = 11,45 ohms

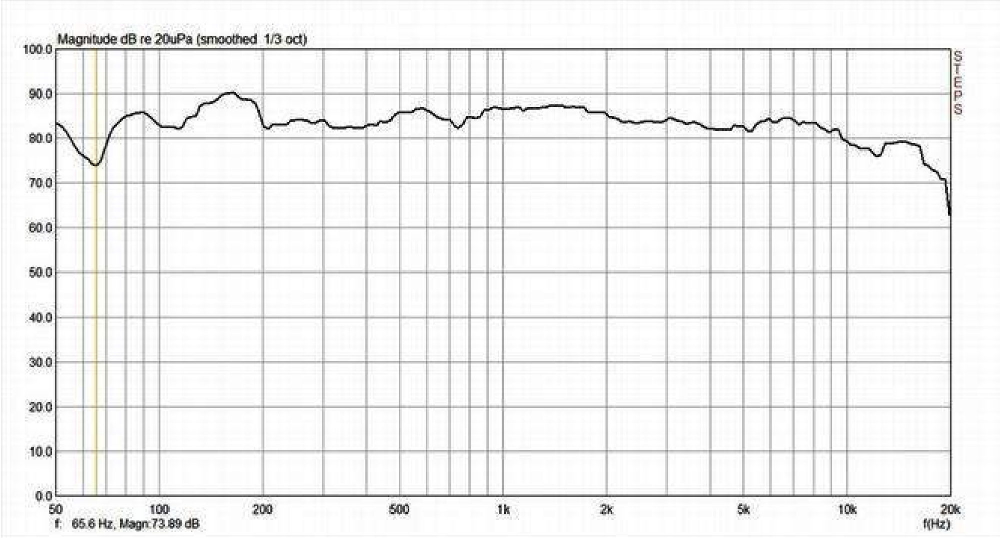
Rz1 = 1 27 + 27 + 27 + 39 ohms / 10 watts MOX parallel; total value = 7,31 ohms

Rz2 = 47 + 100 ohms / 10 watts MOX parallel; total value = 32 ohms

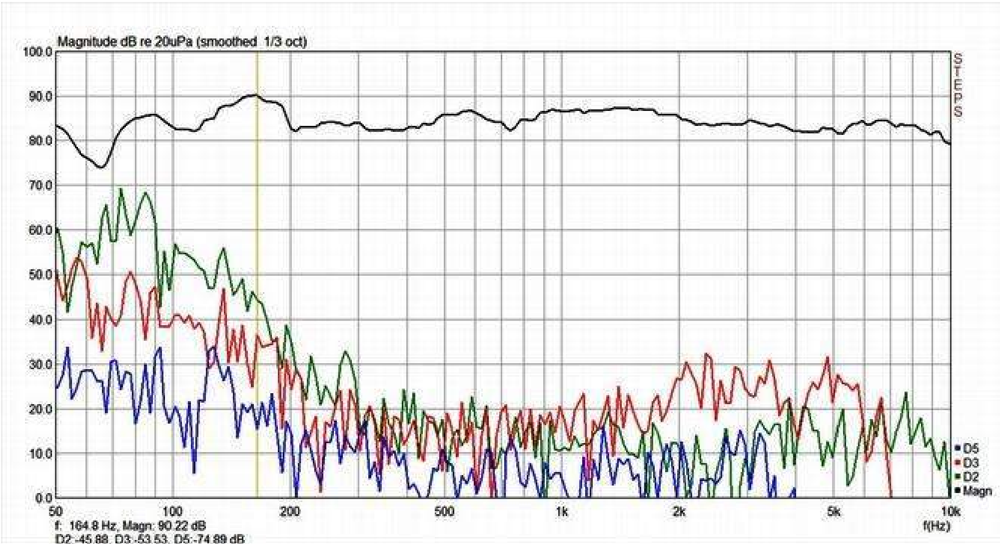
Rz3 = 22 + 33 ohms / 10 watts MOX parallel; total value = 13,2 ohms

Listening and measurements

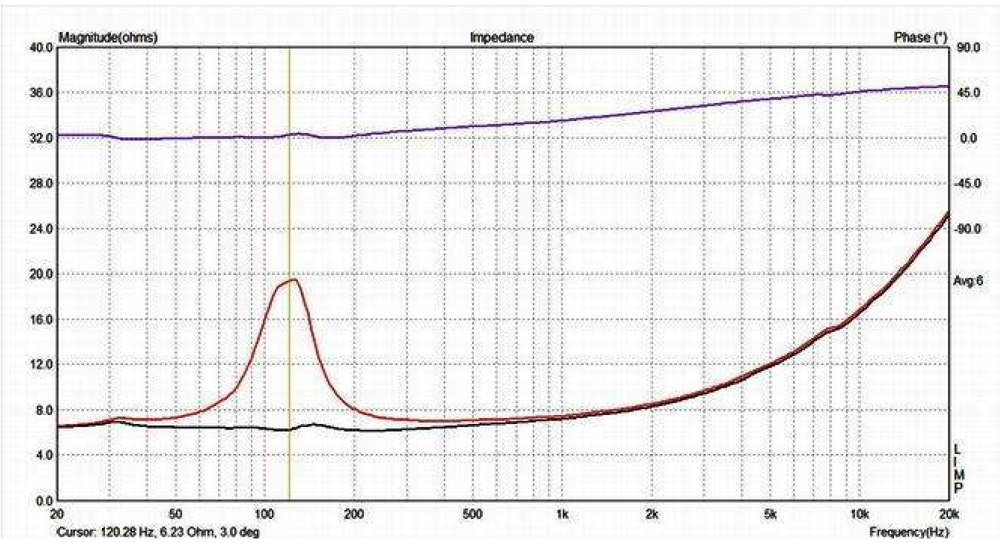
Every new loudspeaker (and the Pipe-Line is no exception) needs some time to burn-in. Fresh out of the box they sound tinny with poor bass definition and the sound seems to be stuck to the speakers. After about 40 hours of normal use, things start to fall into place. You can speed up the burn-in period by running some pink-noise 24/7 for about 2 days. An important thing when positioning these loudspeakers is that the listening distance shouldn't be too small otherwise the cancellation at higher frequencies will cause problems. A rule of thumb is that the listening distance should more than two times the height of the array. In this case the the distance between the centres of the two outer drivers is 120cm, so the listening distance should be at least 240cm, preferably a little more (I listen at 320cm). The loudspeakers are best positioned free-standing and turned far enough at the listener that the inner sides of the cabinets can still be seen. The size of the image can be dialled-in by experimenting with the amount of toe-in, just a few degrees can make quite a difference. Once the loudspeakers are positioned correctly, they create a large spatial image with big dynamic shades. A sort of wall of sound filled with subtle details and nuances. If pin-point imaging is your thing, then maybe a different speaker would be your choice, but even so, this loudspeaker creates a realistic 3D picture in which all the individual instruments and voices can easily be pointed out. With good orchestral recordings there is not only good left to right imaging but also there is depth in front of and behind the speakers. The little full-range drivers produce a solid bass foundation, only the bottom 2 octaves are a little shy. So if you want to use these in a high-end Home Theatre set-up then I would advise to add an active sub-woofer. You can't expect everything from an 8cm full-range driver! The top-end of the spectrum is direct but never harsh, it reveals heaps of detail without getting over bright. It lets you take a deep look into the recording and takes you on a musical discovery journey. This time the frequency curve is measured at 3 metres distance instead of the standard 1 meter measurement because otherwise the comb-filter effect would be visible above 4kHz. This 3 metre corresponds with the perceived response at the listening seat. It shows a well balanced response with about 85dB efficiency. There is a slight roll-off towards the top-end, but that is normal when measuring at larger distances.



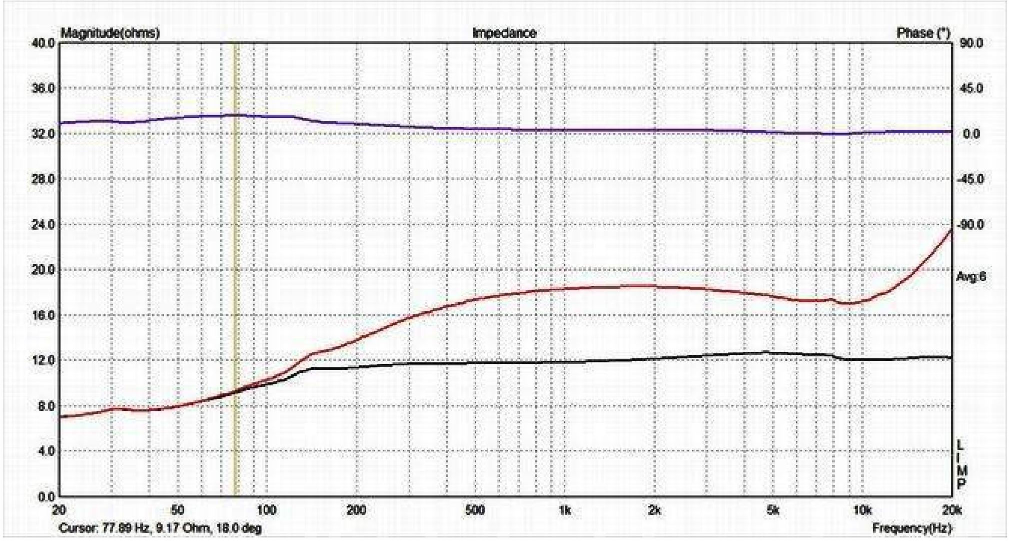
The distortion measurements show the 2^{nd}, 3^{rd} and 5^{th} harmonic distortion for 1 watt from 50Hz to 10kHz. In the bass the distortion rises but that was to be expected with such small woofers with a resonance frequency of 80Hz. Above about 150Hz the distortion is very low with only a slight rise between 2-6kHz (still only 0,25%). In the critical midrange the levels don't exceed 0,1%.



The impedance was measured at several moments during the build of the crossover. The first plot shows the system impedance with an impedance peak at about 120Hz (red curve). By adding Lz1/Cz1/Rz1 this peak is flattened (black curve) so that inductor L1 will work according to the book.



The second plot shows what happens when the notch filter L1/C1/R1 is put in place (red curve). The whole impedance from 100Hz to 10kHz is raised, the loudspeaker will therefore produce less energy in this region. The bass and top octave will thus be let through unaltered. When the last two correction networks are added you get the black curve: a ruler flat impedance of 12 ohms above 150Hz. Below 150Hz it drops to a safe 7 ohms. The corresponding phase curve is therefore also extremely fat varying between 0 and +18 degrees! That should be a very easy load for almost any amplifier.



Tony Gee, The Netherlands, January 2010