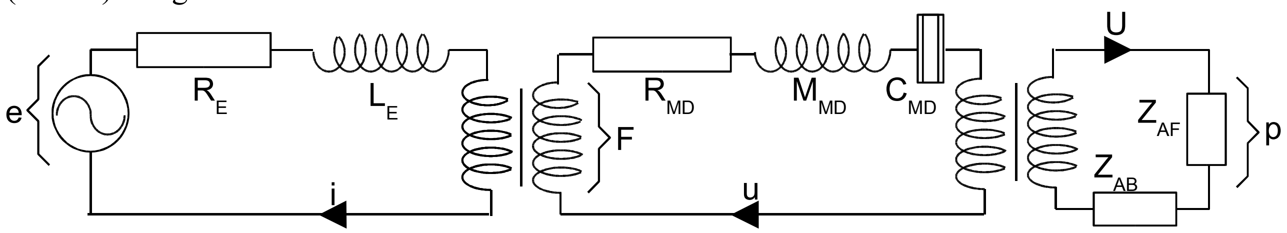
Designing a Vented Cabinet

1. Building a general model

Any loudspeaker system can be modelled using an analogue circuit similar to the one used previously to model the bass driver alone. The circuit initially consists of three sections:

- The electrical section, representing the electrical characteristics of the voice coil.
- The mechanical section, representing the mechanical characteristics of the driver.
- The acoustical section, representing the air load and the cabinet characteristics.

We will first produce a general model, and then move on to adapt this for the bass reflex (vented) design. The initial circuit is as follows:

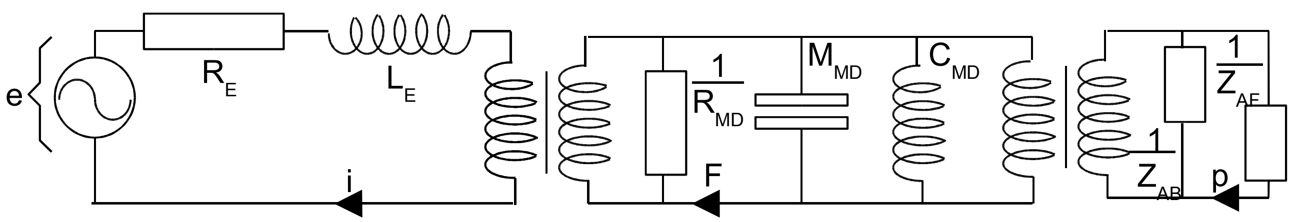


This provides a logical starting point for building a more useful model. Power enters the circuit as an electrical PD of **e** Volts on the left of the diagram – it finally leaves on the right as an acoustic pressure of **p** Pascals. The components marked $\mathbf{Z_{AF}}$ and $\mathbf{Z_{AB}}$ are the impedances of the air to the front and rear of the driver respectively (note that they are *impedances*, despite looking like resistors). These are as yet undefined, as we have yet to consider the acoustic loading on the driver. When we do, $\mathbf{Z_{AB}}$ will represent the effect of the enclosure and vent, whilst $\mathbf{Z_{AF}}$ will represent the free air to the front of the unit – the impedance of which determines the acoustic pressure radiated. Before we can elaborate on what $\mathbf{Z_{AF}}$ and $\mathbf{Z_{AB}}$ may be, we first need to put the rest of the circuit into the acoustic domain.

The three domains (electrical, mechanical and acoustical) are separated by transformers (theoretical ones, not real). For each transition between domains, there is an equation linking a quantity in one domain to a quantity in the other. For instance, between the mechanical and acoustical domains, the equation is $U = u S_D$. In reality this equation relates the velocity of the cone, its area and the volume velocity of the air moved by it. In the model, however, it gives the ratio of the current in the mechanical domain (**u**) to that in the acoustical (**U**). The ratio of the currents either side of a transformer is determined by the turns ratio of the transformer – in this case **U** must be $\mathbf{S_D}$ times larger than **u**, so the **u** side of the transformer must have $\mathbf{S_D}$ times more turns then the **U** side (this ratio would be the other way around if **u** and **U** where voltages). Expressed another way, this means that ratio of turns on this transformer is $\mathbf{S_D}$:1.

The formula governing the electrical to mechanical transition is $F = BLi$ – this relates the current in the voice coil, **i,** to the force acting upon it **F**. In the model, this equates to the ratio between the current in the electrical domain (**i**) and the voltage in the mechanical (**F**). However, there is a problem with this: Whilst there is a simple relationship between the voltages or currents either side of a transformer, calculating the voltage on one side from the current on the other is not so easy. It would be far simpler if **i** and **F** where *both* currents; this can be achieved by transferring the mechanical and acoustical domain sections into the *mobility analogue*. This will turn the voltage **F** into a current, making the first transformer easier to handle.

The new circuit is:

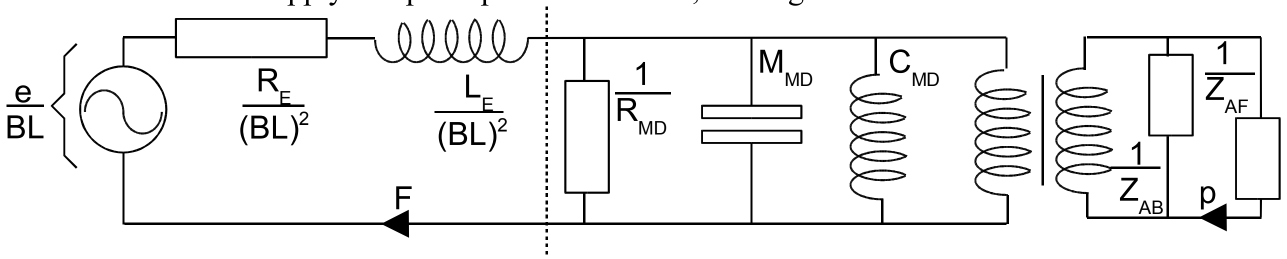


The latter two sections (mechanical and acoustical domains) are now in the mobility analogue, and the Voltage **F** has now become a current. It is important to note that the turns ratio of the second transformer, which started as $\mathbf{S_D}$:1 has now become 1:$\mathbf{S_D}$, as voltage and current have swapped. It can now clearly be seen that since $F = BL\,i$ the current must be **BL** times *higher* on the **F** side of the transformer then the **i** side. This implies the **F** side must have **BL** time *fewer* turns, and so a ratio of **BL**:1 is required.

The transformers must now all be eliminated, to bring everything into the acoustical domain (where the cabinet is). This involves adjusting the component values such that, with the transformer removed, they present impedances equal to those seen with the transformer in place. To explain: If a load of *x* ohms where driven through a transformer of turns ratio *a:b* , the load would appear as if its impedance where $x\left(\frac{a}{b}\right)^2$.

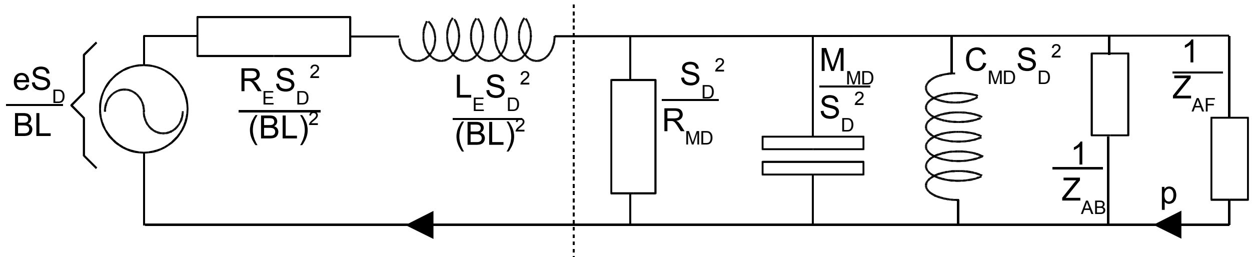
Therefore if the load where replaced with a new load, of the value given by the formula, and the transformer where removed, the source would not be able to tell the difference!

We will now apply this principle to our model, starting with the leftmost transformer:



All the impedances have now been reduced by a factor of $(\mathbf{BL})^2$, putting all everything that was in the electrical domain into the mechanical. Notice that the voltage source, **e**, has also been reduced – but by a factor of **BL**, not $(\mathbf{BL})^2$. This is because a voltage source seen through a transformer is only modified by the turns ratio, not by its square as an impedance is. It is important to remember that the section to the left of the dashed line is still in the impedance analogue (everything to the right is in mobility). This will need to be converted later.

The second transformer can now be removed in a similar fashion, to give:



The circuit is now *nearly* in a state where it will be useful to us. The only change that now need be made is to transform the right-hand side into the impedance analogue. However, it will actually be easier to bring the whole system into the *mobility* domain, and *then* find the dual to get back into impedance. This means that the "electrical domain" part of the system, to the left, needs to be replaced by a mobility-analogue equivalent.

This can be achieved by applying Norton's theorem, which states that any voltage source with a series impedance and be replaced by a current source with a shunting impedance – the two systems will appear identical to whatever they drive. The important thing to note is that a voltage source has zero impedance whereas a current source appears open circuit: This means that, to preserve the source impedance of the circuit, the series and shunting impedances must be *the same*. Thus we can see that our equivalent circuit must consist of a constant-current source in parallel with the impedances of $\mathbf{R_E}$ and $\mathbf{L_E}$. The series network of $\mathbf{R_E}$ and $\mathbf{L_E}$ presents a combined impedance of $Z = \frac{R_E S_D^2}{(BL)^2} + j\omega \frac{L_E S_D^2}{(BL)^2}$ where ω is the angular frequency of the driving signal.

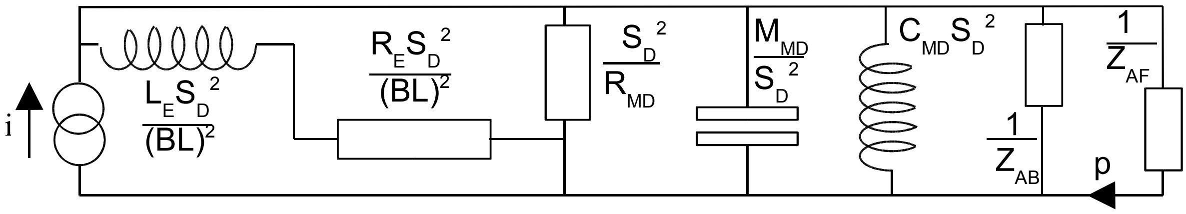
(that is, the resistance of the $\mathbf{R_E}$ component, plus the inductive reactance of the $\mathbf{L_E}$ component)

In order to be equivalent, these two circuits must deliver the same short-circuit current: Finding this current for the *original* (voltage-source) circuit will tell us the value of the current source in the *new* circuit. The current is simply given by voltage over impedance – the only complication here being that the impedance (and hence the current) depends on frequency. The maximum current that can be supplied by the voltage-source-and-series-impedance configuration is then:

$$i=\frac{V}{Z}=\frac{e\dfrac{S_D}{BL}}{\dfrac{R_E S_D{}^2}{(BL)^2}+j\omega\dfrac{L_E S_D{}^2}{(BL)^2}}=\frac{e}{\dfrac{R_E S_D}{BL}+j\omega\dfrac{L_E S_D}{BL}}=\frac{BLe}{S_D(R_E+j\omega L_E)}$$

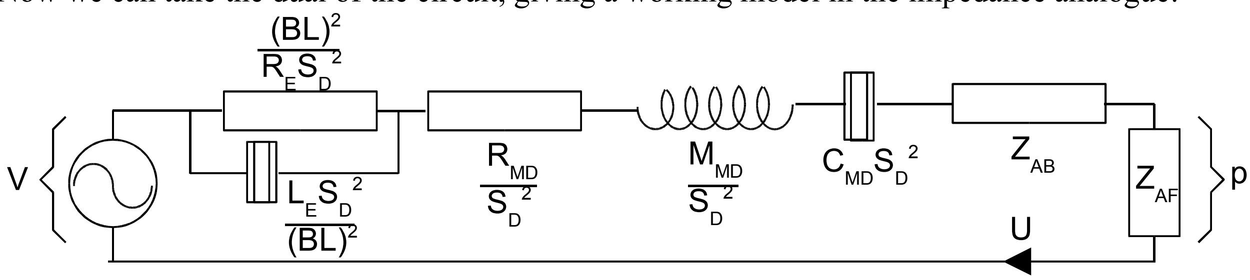
And so this *must* be the current limit of the constant-current source.

Replacing the constant-voltage system with its constant-current equivalent gives:



(The current is just indicated as **i**, for clarity)

Now we can take the dual of the circuit, giving a working model in the impedance analogue:



The voltage **V** which drives the circuit is exactly the same in value as the current **i** in its previous incarnation – the formula above can be used to calculate it from **e**, **BL**, $\mathbf{R_E}$, $\mathbf{L_E}$ and $\mathbf{S_D}$.

As we are concerned here with designing the bass response of the cabinet, we will never be using frequencies over a few tens of Hertz. At such frequencies the capacitance representing $\mathbf{L_E}$ will appear close to open circuit, and so can be left out of our calculations – the represents the negligible inductive reactance of the voice coil at such low frequencies.

We now have a model which represents a general loudspeaker system. With appropriate networks substituted for $\mathbf{Z_{AB}}$ and $\mathbf{Z_{AF}}$, any cabinet type can be simulated. In the next section we will consider the effect of the vented cabinet, and build this into the model.

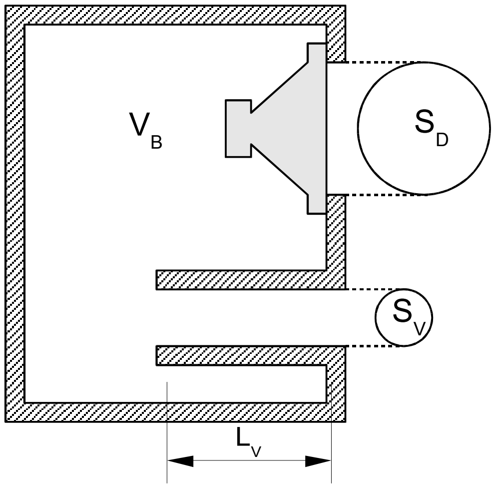
2. The bass reflex system

The bass reflex system works by installing the bass driver in a cabinet which is sealed save for a duct connecting the its interior with the free air outside. This duct, termed the *vent* or *port*, contains an air mass which acts as a resonator - at its resonant point the acoustic output from this this re-enforces the output from the driver. This raises the bass output of the cabinet, lowering the -3dB cut-off point. In addition, the action of the vent creates a greater SPL within the enclosure, and this opposes (and therefore damps) the motion of the driver cone – given the right conditions, this can increase the power handling of the driver at low frequencies.

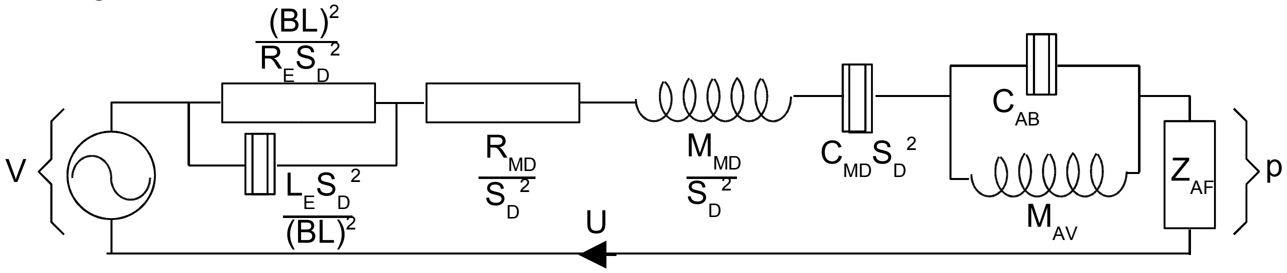
The addition of the vent effectively changes the cabinet from a second-order to a fourth-order highpass filter – this is the disadvantage of the bass reflex system. The increase in bass output comes at the expense of a 24dB/octave roll-off below resonance, which often sounds less pleasing than the 12dB/octave given by a sealed enclosure.

The layout and dimensions of our bass reflex cabinet are described on the following page.

There are three parameters which need to be set for a vented system: The vent's length and area, **L_V** and **S_V**, and the volume of the cabinet, **V_B**.
Although in the diagram the vent is shown opening towards the front of the cabinet, it can equally well face backwards or to the side, depending upon which is more convenient for the layout of the enclosure. Only one driver is shown in this diagram, although more may be present – our system will have two. The placement of the tweeter may also influence the final design of the cabinet.



However, before we can consider the reality of the enclosure, we must build the vent and cabinet into out model.
Both of these factors act on the driver's rear wave – the front radiates directly to free air. Air displaced by the rear of the cone has two options: It may cause a change in pressure within the box, or it may escape out of the vent. In order to change the pressure within the box, it must act against the pressure already present – that is, the volume of gas in the box is *compliant*. In order to escape through the vent, it must move the volume or air already present within the vent – this volume is *massive*. Thus we have to alternate paths, which implies a parallel connection, one of which is compliant and the other massive. In out electrical analogue these correspond to capacitive and inductive components, which together make up the back load on the driver. These are represented in the diagram below:



In order to use this model, we must know the values (in acoustical units) of the mass of air within the vent, and the compliance of the air within the box. In reality, there are also resistances (dampings) associated with both the massive and compliant paths – however these are small, and as they are very hard to quantify, we shall ignore them.

The mass of air in the vent is, as we might expect, the product of its volume and density – that is $M_{MV} = L_V S_V \rho_0$. As we saw earlier, when the mass of the cone was transferred into acoustical units, it was reduced by its area squared. The same applies here; the acoustic mass of the gas in the vent is equal to its mechanical mass over the square of its area. That is, $M_{AV} = \frac{L_V \rho_0}{S_V}$

The compliance of a volume of air is given by $C_{AB} = \frac{V_B}{\rho_0 C^2}$ where **ρ_0** is the density of air, and **C** is the speed of sound in it.

This is all the information we need in order to model the rear load of the driver.

3 The front load & radiated pressure

The SPL generated by a loudspeaker depends on the acoustic pressure that it radiates. Just as voltage depends upon current and impedance, acoustic pressure depends on volume velocity and the acoustic impedance of the air. The acoustic impedance if air, however, has the peculiar property that it is a resistance that depends on frequency: It is in fact proportional to the frequency, causing the SPL from a source producing a constant volume velocity to rise by 6dB/octave.

If the source produces a volume velocity of **U** at a frequency of **f**, the pressure at **r** meters is given by: $p=U\dfrac{\rho_0 f}{r}$

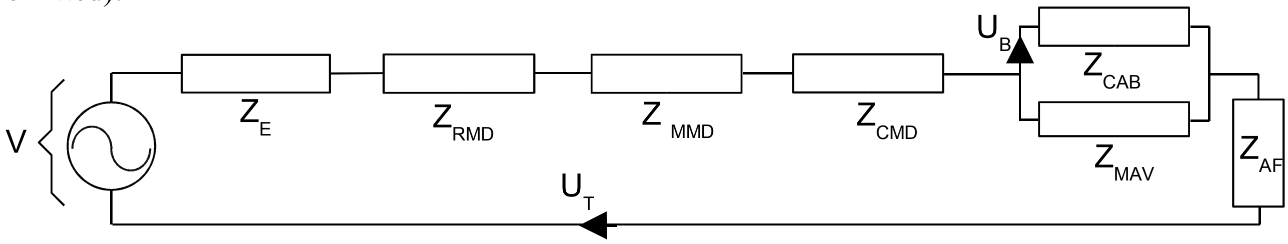
So in order to find the SPL radiated by our vented loudspeaker enclosure, all we need to do is calculate the volume velocity that is radiated from it.

This *total* volume velocity is equal to that radiated from the front of the driver, less that radiated from the port. But why do they not add together? The reason for this is simply to do with the signing convention. All of the sound radiated from the vent is the *rear* wave of the driver, and so must be in anti-phase with the front wave – in which case they *must* cancel. They will re-enforce at some frequencies, but this is taken into account by the phase shift through the vent.

If we inspect the circuit on the previous page, we can see that current that flows through $\mathbf{Z_{AF}}$ (that is, out the front of the driver) but not through $\mathbf{M_{AV}}$ (out of the vent) must pass through $\mathbf{C_{AB}}$ (the compliance of the box). This difference current, through $\mathbf{C_{AB}}$, is the total volume velocity that we have just discussed.

At first this may seem rather odd: How can the pressure radiated to the outside world depend upon a volume of air that never leaves the enclosure? However, thinking logically, the answer is simple – and holds true for any boxed loudspeaker system. For every unit of air that is radiated from the enclosure, an equal amount must leave its interior (and of course, vice versa). Thus the volume velocity radiated to the environment always equals the volume velocity that is compressing and rarefying the air within the cabinet. Different *pressure* changes will be generated inside and outside the enclosure, but the *amount* of air moving will be the same.

We can now finally set about deriving an expression for the pressure radiated from the enclosure. To help, here is a simplified diagram of the model. All components have been replaced by simple impedances, the values of which are described below (the inductance of the voice coil has been omitted):



$$V=\frac{BLe}{S_D(R_E+j2\pi f L_E)}\quad Z_E=\frac{(BL)^2}{R_E S_D{}^2}\quad Z_{RMD}=\frac{R_{MD}}{S_D{}^2}\quad Z_{MMD}=\frac{j2\pi f M_{MD}}{S_D{}^2}\quad Z_{CMD}=\frac{1}{j2\pi C_{MD} S_D{}^2}$$

$$Z_{CAB}=\frac{\rho_0 C^2}{j2\pi f V_B}\quad Z_{MAV}=\frac{j2\pi f L_V \rho_0}{S_V}\quad Z_{AF}=\frac{2\pi f^2 \rho_0}{C}+\sqrt{\frac{512}{9}}\times\frac{j f \rho_0}{\sqrt{\pi S_D}}$$

The current $\mathbf{U_T}$ through the circuit is given by:

$$U_T=\frac{V}{Z_{TOTAL}}=\frac{V}{Z_E+Z_{RMD}+Z_{MMD}+Z_{CMD}+Z_{AF}+\dfrac{1}{\dfrac{1}{Z_{CAB}}+\dfrac{1}{Z_{MAV}}}}$$

The this current splits between $\mathbf{Z_{CAB}}$ and $\mathbf{Z_{MAV}}$ in the ratio $\mathbf{Z_{MAV} : Z_{CAB}}$ (that is, the larger one lakes the least share of the current). This means that the current through $\mathbf{Z_{CAB}}$ is:

$$U_{CAB}=\frac{U_T\times Z_{MAV}}{Z_{MAV}+Z_{CAB}}=\frac{V\,Z_{MAF}}{\left(Z_E+Z_{RMD}+Z_{MMD}+Z_{CMD}+Z_{AF}+\frac{1}{\frac{1}{Z_{CAB}}+\frac{1}{Z_{MAV}}}\right)(Z_{MAV}+Z_{CAB})}$$

The pressure at 1 meter an then be found by:

$$p=U_{CAB}\rho_0 f$$

While this expression is hardly elegant, it can be implemented easily enough using a spreadsheet or modelling package – doing so will provide a way to experiment with different port and box sizes, without building scale models.

4 Selecting alignment

Having secured a model of the enclosure we intend to build, we must now decide on its dimensions. One way of doing this would be to select likely-looking numbers and put them into the model (perhaps using existing loudspeakers as a guide?). However this is neither very scientific nor very efficient – it would doubtless take a lot of experimentation before a suitable alignment was found.

Fortunately a lot of the work has been done for us by R.H. Small, who did much research into enclosure design in the 1970s. As part of his work he published a chart from which it is possible to calculate box and vent dimensions, given values for $\mathbf{Q_{TS}}$ and $\mathbf{F_S}$ and $\mathbf{C_{AS}}$. This chart will make at least a good starting point for the design of our enclosure.

The input values for the Celestion T4625 are:

- $\mathbf{Q_{TS}}$ = 0.41
- $\mathbf{F_S}$ = 58.5
- $\mathbf{C_{AS}}$ = 6.59×10^{-8} (calculated by the division of $\mathbf{C_{MS}}$ by $\mathbf{S_D}^2$)

Starting with $\mathbf{Q_{TS}}$, a value for the ratio α may be found: This gives the ratio between the acoustic compliance of the driver, $\mathbf{C_{AS}}$, and that of the box, $\mathbf{C_{AB}}$. In this case α happened to come out as *one*, making $\mathbf{C_{AS}}$ and $\mathbf{C_{AB}}$ the same.

Since we know $\mathbf{C_{AS}}$, we can calculate the compliance of the air within the box, and hence its volume – the two are linked by the expression:

$C_{AB}=\frac{V_B}{\rho_0 C^2}$ The required volume of air is therefore $6.59 \times 10^{-8} \times 1.28 \times 340^2 = 9.756 \times 10^{-3}$ m³

The line used to find α can be extended to meet another curve – this intersection can be read off to give a value *h*: This is the the ratio between the resonant frequencies of the box and the driver. In this case *h* comes out as 0.9 – placing the box resonance $\mathbf{F_S}$ at 58.5 × 0.9 = 52.65Hz.

The resonant frequency of the box is dictated by the compliance of the air contained within it and the mass of the air in the vent. Since the compliance is known, the acoustic mass of the vent may now be calculated. The box resonance is given by:

$F_B=\frac{1}{2\pi}\sqrt{\frac{1}{C_{AB}M_{AV}}}$ Thus $\mathbf{M_{AV}}$ is given by $[\,(52.65 \times 2 \times \pi)^2 \times 6.594 \times 10^{-8}\,]^{-1} = 138.58$

Now a port must be designed to contain the appropriate air mass – this depends on both the length and area of the port, according to the formula:

$$M_{AV}=\frac{\rho_0 L}{S_V}$$

There are an infinite number of length/area combinations which could (theoretically) be used, but some are more practical than others.

A common problem encountered with bass reflex cabinets is that high air velocities through the port produce turbulence, which can be audible as “chuffing”. As a rule of thumb, in order to eliminate this effect, the port area should be at least one quarter of the area of the driver cone. A round port (which are by far the most common, at least in domestic equipment) having the convenient diameter of 5cm has an area of 1.96×10^{-3} m^2, which is almost exactly a quarter of $\mathbf{S_D}$, and so should be suitable.

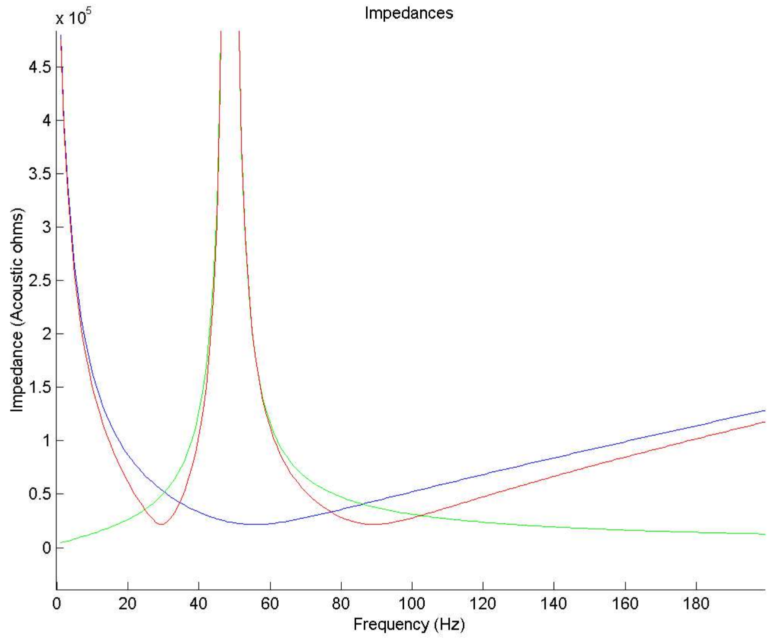
If this area is applied to the above formula, the length can be found to be:
$138.58 \times 1.96 \times 10^{-3} \div 1.28 = 0.2125$m.

We now have the box volume and port dimensions which we may put into our model. This *should* represent an ideally aligned system, however it is worth checking that it does provide the desired results.

5 Modelling the system

The set of equations derived in section 3 where implemented using Matlab (which handles complex mathematics transparently). The measured loudspeaker parameters from the first part of the assignment and the enclosure dimensions calculated from Small's chart in the previous section where used as input data.

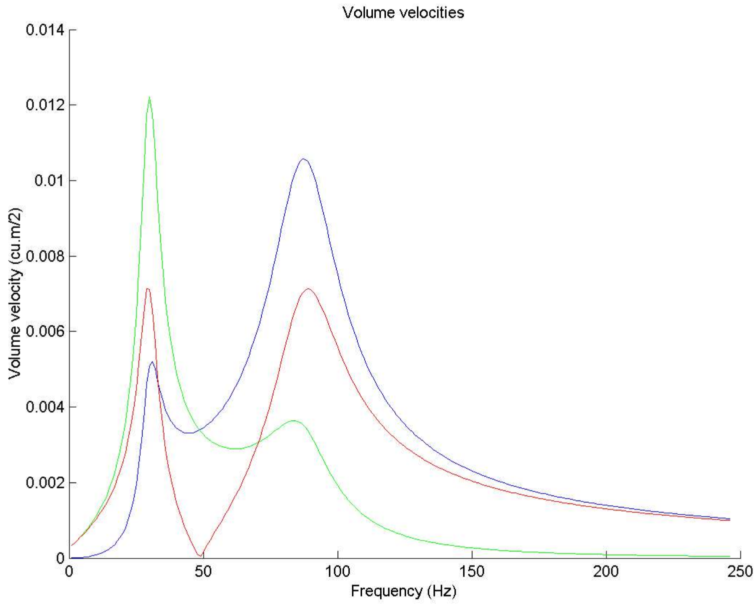
Using Matlab it is possible to graph any of the variables or parts of the circuit in order to ensure that the system is performing as expected. The following are the results of the model:



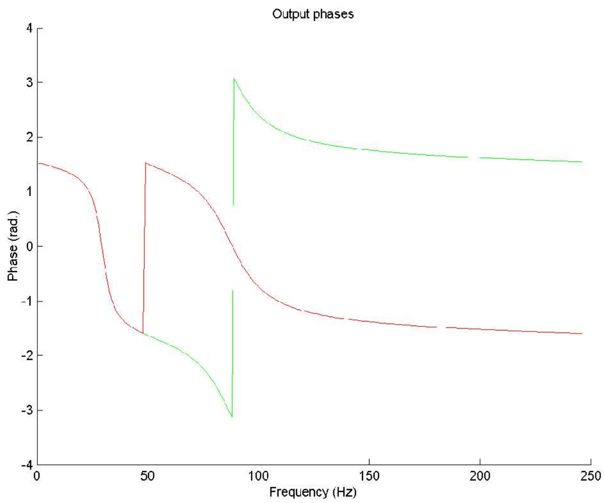
Firstly, this set of plots shows the impedances (magnitudes) of different section of the system versus frequency. The green line is the box impedance, blue is the driver and red the impedance of the system as a whole ($\mathbf{Z_T}$).

It is clear that the driver (with the series connection of reactors) acts, as we might expect, as a bandpass filter. The enclosure however, (represented as a parallel tuned circuit) has a band-stop characteristic. In the model this peak ascends to many billions of acoustic ohms – this is merely because there is no damping of the box resonance built into the model (as it is rather hard to quantify). It can be seen from the relative positions of the peak/trough in the box/driver impedances that the box resonances is, as expected, a few Hertz lower than the driver resonance point.

Here we see the volume velocities flowing in different parts of the system. The green line represents the volume velocity flowing through the port to free air, the red line shows that radiated by the cone, and the blue that which remains inside the enclosure – which as we have seen – is equal to the total radiated by the cabinet.



As expected, the motion of the cone (and hence the volume velocity radiated from it) is heavily damped at the box's Helmholz resonance point, $\mathbf{F_B}$. At this point the total volume radiated from the cabinet is equal *only* to that flowing through the port. Above the Helmholz resonance the driver's front wave and the port radiation are in phase and the total output is greater than either of them individually. Above the upper coupled resonance, just below 100Hz, the mass of the air in the vent causes its output to drop off sharply (so that by 200Hz the system behaves like a sealed box). Below the Helmholz point the driver's front wave and the port are out of phase and therefore cancel – so that, despite the high volume velocity through the port, output is low. Moving lower again, past the lower coupled resonance around 30Hz, the driver's motion is limited by the compliance of its suspension and so the system's output drops to rapidly zero. The only problem with this plot is the driver output line around $\mathbf{F_B}$: The peaks either side are of equal height, as they should be, but the rises on either side are not symmetrical.



To confirm our ideas about the way that the the port and driver radiations interact, the plot to the left shows the phase of the driver rear wave (red) and the port radiation (green) relative to the electrical input. It is clear that the two waves remain about 3 radians out of phase down to the box resonance at 50Hz – that is, the port and driver *rear* are in anti-phase and therefore the port re-enforces the driver's output to its surroundings.

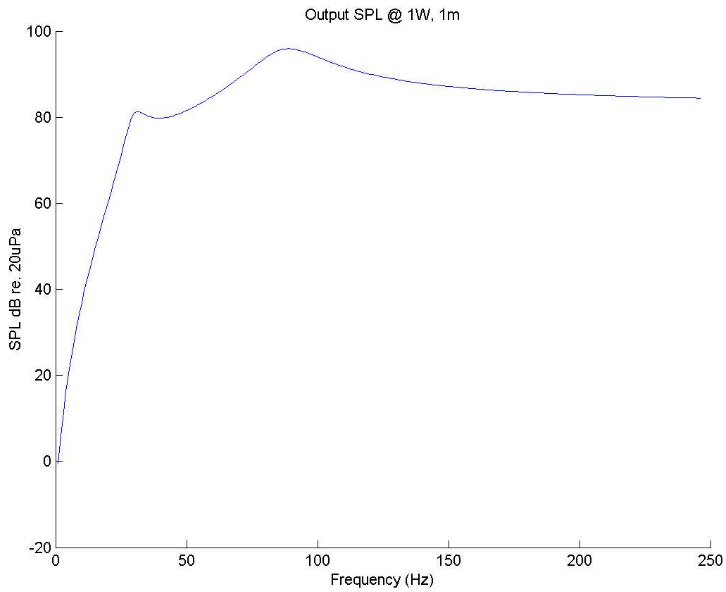
These graphs illustrate that the components of the bass reflex enclosure are working together as they should – but what does it sound like?

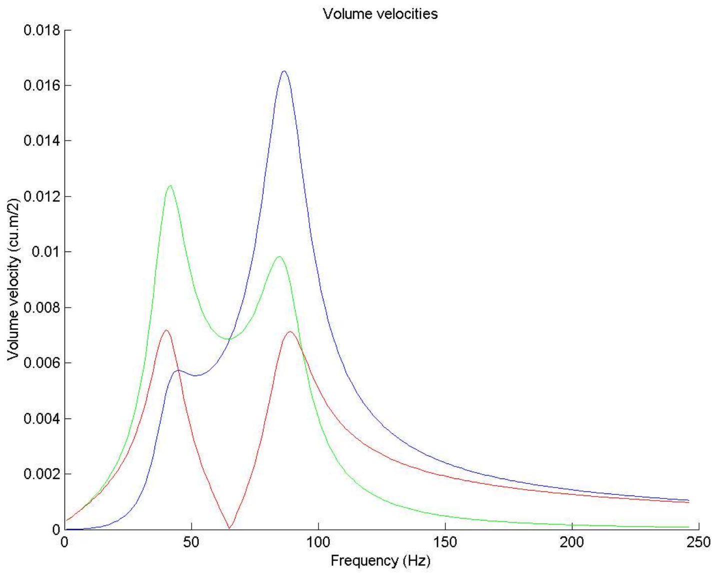
A graph of the output SPL at 1m for 1v input is shown on the following page.

Working downwards in frequency, we first we see a large peak of about 10dB at the upper coupled resonance, around 90Hz. The output then drops sharply towards the Helmholz resonance point. There is then a smaller, sharper, peak at the lower coupled resonance close to 30Hz. These two peaks should not be visible – the boost due to the vent/box resonance should coincide *exactly* with the point where the driver output starts to drop off. It is possible that $\mathbf{F_B}$ is too low, meaning that the damping on the cone does not start high enough – allowing the 90Hz peak to form due to cone resonance. It is also possible that the dual peaks are a symptom of the very sharp box resonance – due to the lack of damping in the model. This would mean that the box resonance is not of the same width as the driver resonance, and the two do not “fit” together well.

Fortunately, it takes little effort to revise the numbers in the Matlab script, and these can be adjusted to see if a more satisfactory alignment can be found.

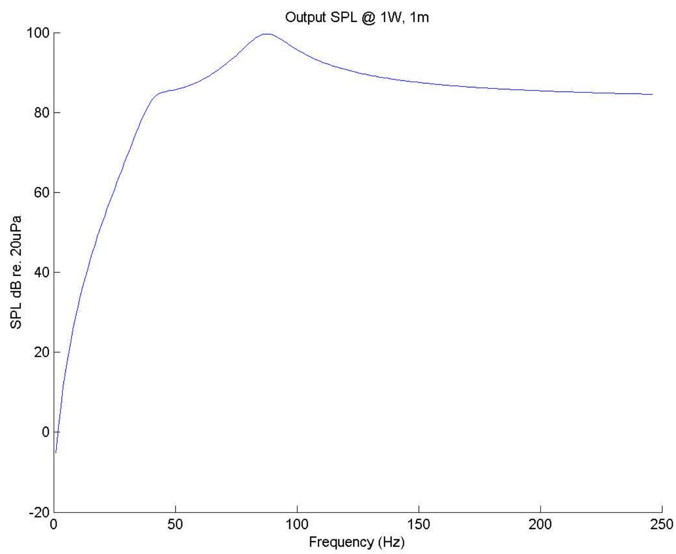
The ideal system would produce a flat response as far as possible, and then roll off as linearly as possible. This implies un-coloured reproduction of the bass registers.





The graph to the left shows the results of "tweaking" the enclosure and port dimensions slightly.
The box is now 0.015m3 in volume, and the port is only 8cm long.

This has had the effect of moving $\mathbf{F_B}$ higher, so that the driver output is more symmetrical about it. The port radiates *less* at the lower coupled resonance (which should reduce the small peak), but *more* at and above $\mathbf{F_B}$ – which should in turn increase output at these frequencies.

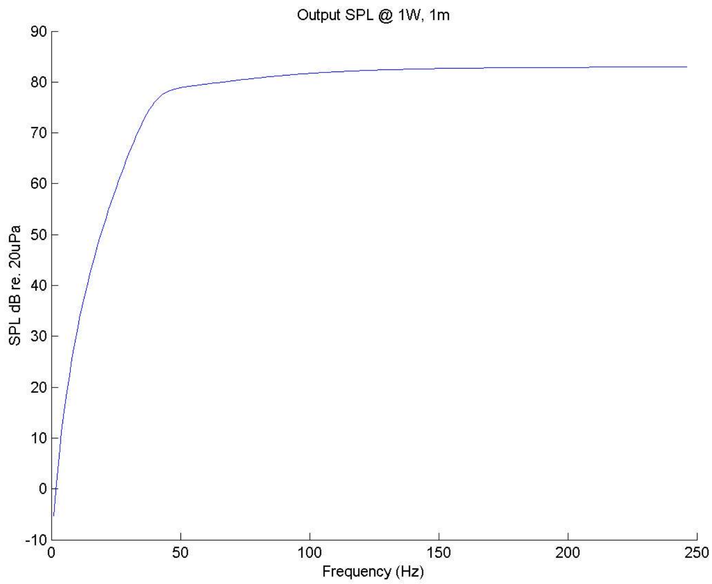


So how do these modifications affect the output SPL?
The frequency response is now without the small peak at the lower coupled resonance, but the peak at the upper is even larger.

The design of loudspeakers generally depends upon the intended application. The response shown here may be satisfactory for sound re-enforcement applications where quality can be traded off for SPL – however as we are only working with a 4" unit, it seems safe to assume we are aiming at the domestic Hi-fi market, and thus quality should be the primary concern.

While anomalies in frequency response can be filtered out, it should be remembered that they also imply a slow transient response, which results in a bass end lacking an clarity.

In order to avoid these effects in the frequency and time domains, it seems that the 90Hz peak should be damped out. It has been said that bass reflex enclosures should not be damped, as their operation relies on resonance within the cabinet: However in this case we appear to have an abundance of resonance, and damping some out can be only a good thing.



The graph to the right shows the result of adding 30000 acoustic ohms damping in series with the box compliance. The resonant peak has now clearly disappeared, although the bass extension remains the same.

Naturally, the effect of adding resistance into a circuit is low waste power – that is, lower the efficiency. The approximately 85dB (250Hz) produced at 1m for 1v input equates to a very respectable efficiency of around 97dB / Watt, which implies a significant bass lift over the reference-region efficiency of 83dB / Watt.

Applying this damping in practice would not be so easy, however. If the interior of the cabinet where waded with an absorbent, some damping would appear in series with the compliance and some in series with the vent mass. Added to the difficulty in measuring the damping properties of materials, this means that the only practical solution would be to set up an empty cabinet in the laboratory, and fill it with wadding until it gave the required response.

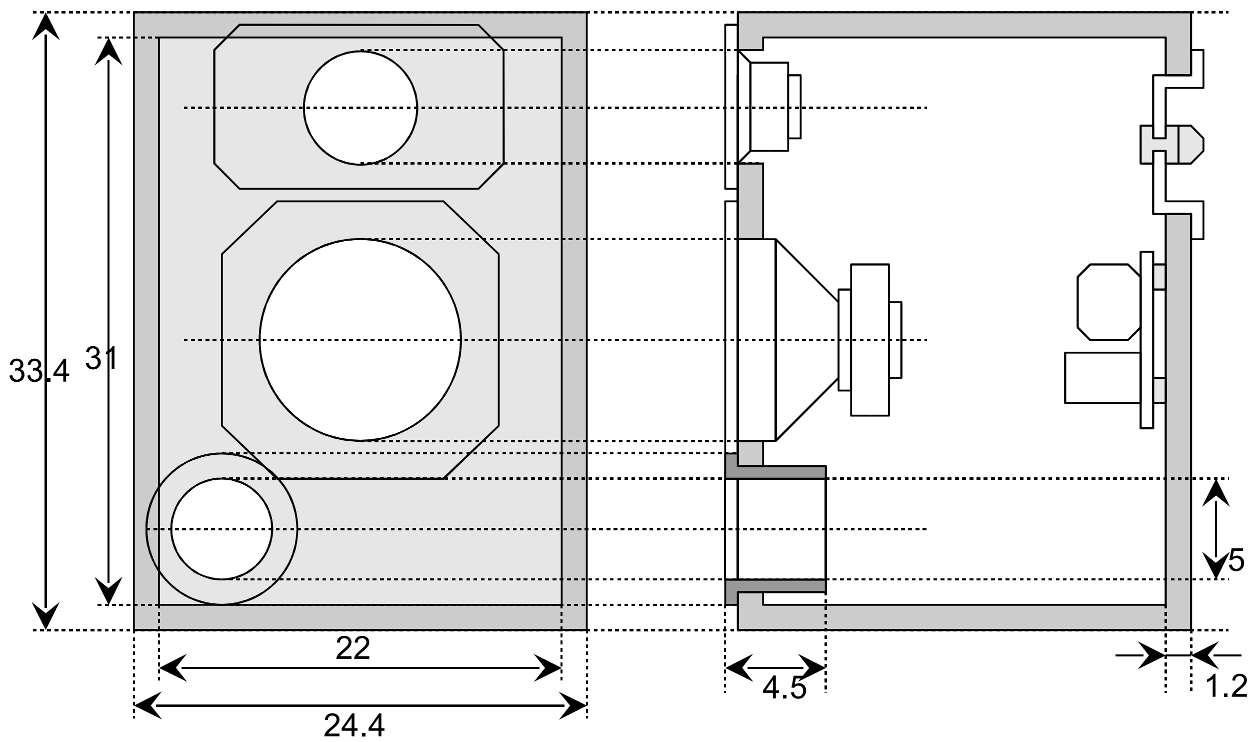
6 Physical construction

There are two small matters that need to be taken care of before we decide on the dimensions of the enclosure. Firstly, the port is to have a length of 0.08m – however, there are *end effects* associated with a volume of air leaving and entering a tube. The sections of air immediately outside the tube move with the air within it, effectively increasing its length. The amount by which the effective length is increased depends on the radius of the tube, and the nature of the tube's end – whether flanged or un-flanged. A bass port, fixed through the cabinet wall at one end and free at the other, has one end flanged (for which we add 0.85 times the radius) and the other not (an addition of 0.6 times the radius). Therefore to produce a port which behaves in the right way, we must remove $0.85r + 0.6r$ from its desired effective length. With a radius of 2.5cm, the corrected port length is now only 4.5cm.

The portion of the port which is present inside the enclosure detracts from its internal volume – so a volume equal to the volume of the port should be added in compensation. The volume of the port is 8.835×10^{-5} m^3, and added to the box volume, this gives 0.015088 m^3 (in all practical terms a negligible increase).

Materials for constructing enclosures should be rigid and have good damping properties – depending upon budget, this cabinet would probably be fabricated in MDF or chipboard, both of which are suitable. This board would be in the standard thickness of 12mm (½”). All joints should be glued with a viscose, slightly flexible, adhesive to prevent leaking and rattling. The cabinet could be manufactured in one fully-glued unit, with access to the crossover network and recess plate gained via the baffle cut-outs, with the drivers removed. Braces are often added across large panels to reduce resonances, however this is unlikely to be necessary in an enclosure of this size .

A practical design for such a cabinet is shown below.



¼ scale, all dimensions in centimetres.

The port faces towards the front of the unit to avoid interference from close objects (such as walls) and any phase lags that might be evident due to the time it takes for the wave to travel around the cabinet (although these would be small – about 60° worst case). Many designs *do* place the vent to the rear, so any detrimental effects must be small, but still worth avoiding.

The outlines and layout on the cabinet front are approximate, as the exact dimensions of the drivers where not available at the time the design was produced – however they serve the purpose.

The internal volume of the cabinet should be waded with the appropriate amount of absorbent (as determined in the laboratory). This should be of a type that will not compact with time and vibration. At least some absorbent should be present between the driver and the rear wall of the enclosure (where the crossover is mounted). This will prevent reflections from the back of the box passing through the cone and causing comb filtering in the output at 386Hz and its harmonics.

This concludes the design of the vented cabinet.