How to build to the limits of possibility.

Comments
In this chapter you will be presented problems with a solution but also problems without.
I hope this paper will create a forum world-wide for discussion of these subjects and if possible form a brain trust giving a lot of proposals – good as bad – from which we all can be inspired.
It is time for us amateurs and those of the professionals interested in good sound to fight against that ignorance, carelessness, cost benefit and resign the manufacturer show, even when they demand a tremendous amount of money for there production. It seems as if all energy is put into appearance and design. Would you bye an ALFA ROMEO equipped with a CITROEN 2CV motor
Thinking back, I will point out the very popular AR-3 as an example of a construction, where energy was put into its working parts – no matter how it looked behind the front cover. It must be the ability to recreate sound that counts and not so much how it looks. If you focus on looks then there are lots of nice looking constructions to choose from.

The search of perfection.
In search of that you must have an open mind and look at all parts in and around the loudspeakers as bricks in a puzzle that must end to fit together.
The amplitude and phase versus frequency, determined by the crossover, units, cabinet and room, are most complex, and it will be impossible to make fit together perfectly. But it is with our hearing as with our eyes, there is a limit for how small a chink can be, for us to see it.
Our hearing is much more forgiving, as it works in time, where no stand still for further inspection is possible, but that doesn't change the fact, that the better the chinks are filled out the closer you come to the recorded event. ***When all is perfect, you just have to close your eyes to see.***

Our ***measuring scale*** - the dB scale - is a rather coarse one. Normally one would inspect for linearity of summation as the only possibility. This gives a rough result; as it doesn't

tell, how it is achieved. Every speaker can in broad outline be seen as a vector. What you measure is the resulting vector and that tells nothing about its origin.
A main problem by measuring is the fact, that ***you listen with two ears*** placed at your head at the top of your body. This enables you to hear the origin of the resulting vector dimly if you are not trained - clearly if you are.
In order to avoid that, all units must play in phase with each other very precisely focused at your listening position. You will experience this area, but despite that the reproduction is strangely still stable outside this area, if you keep your height for listening reasonably constant.
Therefore we need a complementary technique for measuring.
Using my filter topology and the special cases of it – The Linqwitz-Riley filters - every parameter is to calculate, so also ***a magnifying technique***, which is straightforward.
Turn the phase of one of the units. Now instead of a 6-dB summation, you have a more than 40-dB useful nullification, and that shows even the smallest anomaly in level as in phase. Especially the rounding of the shoulders is most revealing for mismatches.
This magnifying technique is also very good to find ***the actual acoustic centre***, of greatest importance when the units work together. This addition reach far wider in frequency than normally looked upon, and if you want filter functions with no overshoot the slope of cut-off only gradually reaches its maximum no matter its order.
When you shall find these centres, the unit must be filtered completely and take care, that the two units has the correct level according to theory at their common frequency.
Then you just find the height for the microphone, where the two units nullify each other best possible at that specific frequency. From your point of listening you then shall see the microphone pointing exactly at the middle between the centres of the two units' voice coils. If they don't you must push the unit apparently nearest the microphone backwards. When that is in order you measure a sweep on the two speakers

1. In phase
2. Out of phase

And compare the results with the theoretical curves. These curves can be calculated from worksheets to Mathcad 7 for the 3 and 4 way systems, which are present at these pages.
The formula for the calculation of curves used with this technique is straightforward.
You just find the value
20*log (H (jω)) for sufficiently small steps of ω e.g. 20 to 100 steps per octave.
Remember that ω=1 is the centre frequency.
You can also find graphs, which are calculated from the formulae for the 3-way version with a value for (a) equal 2.828427 except for 2nd and 4th order 2 way.
These curves are stored in an image file "Antiphase".
They are calculated from:

$$H(jw) = \frac{+w^4 + (a^2-2)\,w^2 + 1}{(w^4 - (a^2+2)w^2 + 1) + (-2aw^3 + 2aw)j}$$

You just have to change signs on the expression.

Figure 1. Bandpass antiphase (the +sign at $(a^2-2)\omega^2$ is changed to -)
Figure 2. Treble antiphase (the +sign at ω^4 is changed to -)
Figure 3. Summation of bass and treble ($(a^2-2)\omega^2$ is taken away)
Figure 4. Difference between bas and treble (as 3 but with - at ω^4)
Figure 5. Difference between bass and treble 2-way 2nd order Linqwitz-Rieley.
Figure 6. Difference between bass and treble 2-way 4th order Linqwitz-Rieley

If a demand rises, curves for the 4 way even 5 way will be established.

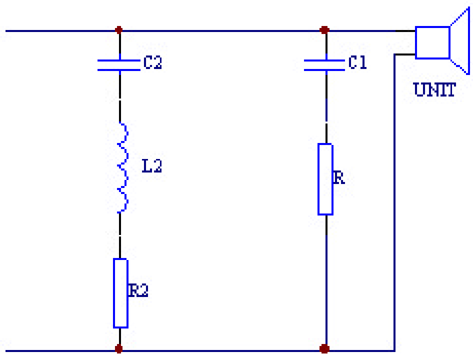
Using this technique, the need of fighting every peak occurring on the roll of down to at least -40 dB, has been shown. With the single unit you should go deeper, I would go for -70 dB ore more, but it stretches the demands on the units and measuring equipment beyond their normal ability.
The problem with units' range of frequency is by these high demands put into relief.
In some way the manufactures of loudspeaker units have forgotten the importance of geometry of the diaphragm. New material for the diaphragm pops up, but for me to see, they all are formed into the same shape, and that can't be right.
It is of course the cheapest way, but given the basket some thought, it should be a minor task to get possibility for displacement of the magnet assembly, and thereby freedom of shape for the diaphragm. As an example you could examine the construction of this part used by "Audio Technology".
The main problem with the geometry is the roll off towards the treble. This is heavily related to the slope of the diaphragm and the front suspension as well. Private research in this area on an 8 inch unit, has shown it possible to construct a really full range loudspeaker with very good treble reproduction as well, and still have minimum phase behaviour
I can in this paper only present, what I think right. I'm not infallible, but by many years work with loudspeakers and related topics, with no commercial interest to stop me at a given development, I feel to have got a deep understanding or intuition of, what is right and what is wrong concerning sound reproduction. The whole process seems so complicated and is yet rather simple. It is the process of doing it all correctly in the long chain from source of voltage to loudspeaker and further to the listener, that is so overwhelming.
My last private project will result in a four-way loudspeaker for me and friends to toy with for the rest of our lives, and give us the opportunity at last to enjoy music, without the ever irritating noise annoying us through the years. This construction is open, it means that, it can be changed in a multitude of ways, only the cabinets will be the same, though their working manner also can be changed from closed gradually to bass reflex, if that should come into question for some.
This property is build into the construction to be free to try every new components and loudspeaker units, when ever they turn up, or are made by own hand. As the demands for every single loudspeaker is determined in all detail, this can be done. All level and position is to be optimised, the dividing network easy to reach, the important baffle

curvature is changeable and so on. Realising that it's a never-ending work, I call it 'The No End Loudspeaker'. That construction will hopefully come on these pages in the year 2000, and will be my final statement of a 4-way loudspeaker based on the closed cabinet.
First now after 25 years work with loudspeakers, I finally believe to know in all detail how to build this most complicated thing to satisfaction. Former models are still playing to their owners' joy. But the real break through where ***every part*** is optimised is for now at a temporary stand still, waiting for the last parts to be made. A real problem for me is to stop new ideas to pop up, but my friends hopefully will hold me focused at the project.

Loudspeaker impedance correction.
When you work with a dividing network, it is a must that the impedance of the loudspeaker has to be corrected to DC-resistance, not just linear as normally seen.
From measurement taken you must determine following.

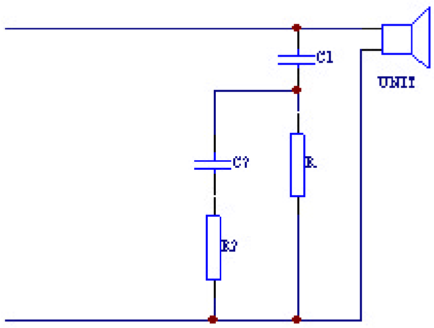
- DC-resistance =R
- Resonance frequency =Fr
- Resonance impedance =Zr
- Frequency for 2*R (voice coil inductance) =F2

From these you can calculate the impedance, for which you shall find the frequency just under and over the resonance frequency.
Look for $(\text{sqrt}(R^2+Zr^2))/2$ and find Fl(ow) and Fh(igh)
To control your reading sqrt (Fh*Fl) should be equal to Fr
You can now calculate values for the components in following circuit.



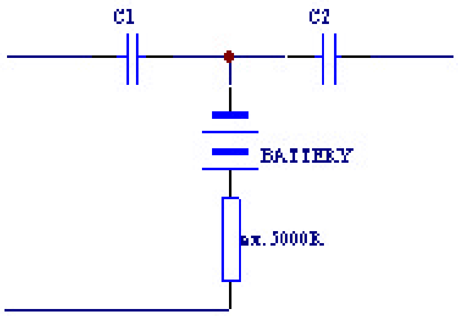
To calculate C1:
$p=\text{sqrt}(3*R^2) / (2*\pi*F2)$ and $q=R / (2*\pi*p)$ then
$C1=1000000 / (2*\pi*q*R)$ uF
To calculate L2, C2 and R2

$L2=1000*R^2 / ((Zr - R)*2*\pi*(Fh - Fl))$ mH
$C2=1000000000 / (L2*(2*\omega*Fr)^2)$ uF
$R2=R*Zr / (Zr - R)$
If all was perfect, these values should work, but it isn't. The voicecoil is placed into the magnetic field, surrounded by iron and perhaps copper.
You will therefore probably face mismatches with some or all calculated values, but you will know about where the values are.
It is advisable first to regulate the voice coil inductance, as this network influences the readings on resonance impedance. With copper around the voicecoil you must experiment, as this copper complicates this process.
The C1 works in the lower end and R in the higher end of the rise of impedance caused by the inductance in the voicecoil. You shouldn't be surprised if you have to change the calculated value on C1 and even double the R-value. This leads to a further rise of impedance to be compensated again to look:



The values depend on the unit used, so you have to experiment.
To my experience great care should be taken to achieve exactly DC resistance. If not - you will have hidden reactance to react with your filter components, sometimes leading to rise of level, more than that received from the unit without any components attached. This extra energy is created by the components within the unit together with the network, and should obviously be avoided. It is a resonance.
Another strange parameter is the need of quality for the components used for this part. Despite they're coupled in series with a resistor, they must be of very high quality.
When this is fixed, you should measure again, to get the value needed for correction of rise of impedance on resonance frequency. Here copper again can interfere, so if that is present the inductance must be raised by a factor 1.4 and capacitance lowered by a divisor 1.4. (This value can vary dependant on factory)
You probably can't avoid use of electrolytic capacitors of the bipolar type. If you need high values you can connect two polar capacitors in series negative to negative or positive to positive (dependant of can connection) to reach that property, beware of voltage - that doesn't double and should be chosen high. (100 v or more)
The calculated value using C1 and C2 should be C1*C2/(C1+C2) and remember to take

off their plastic cover.
You can improve the performance of this coupling by adding negative/positive voltage (dependant on polarity chosen) to their connection. See fig.



C1 and C2 will, if the battery voltage is chosen high enough, never reach zero voltage, where the memory effect comes forward. It is a cheap trick for you to try.
The coil could be a toroid coil with airgap in the iron, and able to handle high current, or the former described air-toroid dependent on frequency - iron for low and air for high frequencies. (Over 400 Hz).
The resistor must be able to handle high effect and its calculated value has to be lowered with that DC-resistance present in the correctional coil. Some like to place all impedance inside the coil, but then beware the warmth generated.
But you are not finished yet. You should test how it all behaves under dynamic conditions. For that you should play some music, at a sound level you normally prefer, for an hour or two. Then measure again and look for changes due to rise of temperature in the parts, and correct if necessary.
On could believe this part of the work to end here, but it isn't, you’ll probably have to return to this circuit to make slight modifications, to optimise the roll off of the unit, when the filter components is attached. It all is heavily connected, as the output from the unit still is dependent of its Qt, even if it is electrically corrected. If it comes through you can lower the impedance in the correctional circuit.
It is a most tiresome work to get it all right, but there are no ways around these adjustments to get it all in order.

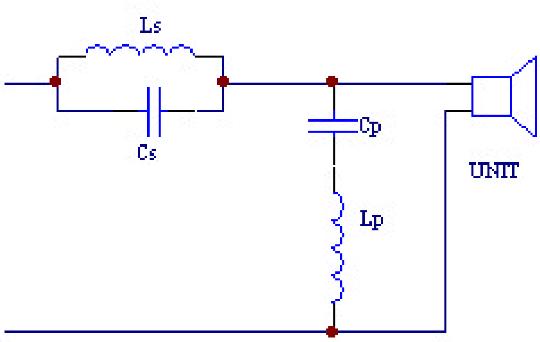
The bass response.
The manufacturer of loudspeakers does not treat this part of the frequency band in a proper way. ***It is simply not done right***. One problem with bass response seems to be our brain’s tendency to manipulate with that. What you apparently hear is not what your ear register. It seems as if, parts of the bass response are overheard and others amplified by the brain. This could be related to the complicated turn of phase on modern loudspeakers. Experiments indicate that the capacitive turn of phase is disturbing for the

brain’s work. This also goes for too much inductive turn.
It is a general opinion, that bass under a certain frequency is impossible to point out. To my experience this is untrue, at least when playing recorded music. I know very well that deep bass has a tendency to be recorded mono as the wavelength is long, but the same tendency is repeated played back on your loudspeaker. The bass loudspeaker is further filtered upwards and thereby delivers parts of sound easily to point out. So for the sound stage to form in a proper way, you **must** have a stereo pair of bass units. Furthermore the level of bass is just as critical as the level of treble.
Lots of energy has been put into the bass reflex solution, with its magnitude of problems. I started with horns, left them and have tried all the different systems through the years, with a long stop for now at the closed system.
For ***me it must be full range***, and here it's fare more a question of quality than of quantity. To my experience the brain will do the rest, when it has understandable sound to build on. Only the closed box, with slight modifications, has the right potential to work together with your room for listening. That's a main fact often forgotten.
They do have a bad reputation, based on wrongdoing in the past. This primary in an attempt at in magical way to conjure bass from small cabinets, which led to that fat pumping bass response, bringing AR in the hall of fame.
This attempt is now done again with the bass reflex idea.
The real problem with bass is its deep dependency of room behaviour, as the loudspeaker is a box placed in another box (the room), for the bass waves of sound to see.
Rooms in general build up anti wise a soft curved 12-dB roll of, which is created perfectly with the closed box. This build up is dependent on room size and reflection coefficients; why it's impossible to develop a loudspeaker in general. It can only be optimised to the specific location. Even though there are a certain line of uniformity in our living rooms, the loudspeaker manufacturer still develop loudspeakers to play in the open, if they care of bass at all. It is, I’m sorry to say an overlooked parameter.
The level of bass is really critical, why it must be to regulate both for level and decay downward in frequency.
In order to have standing waves under some control, you must use great magnet power. The loudspeaker units also serve as microphones when playing. When standing waves build up in the room, the bass loudspeaker generates a signal reflecting that, for the amplifier to see as a rise of impedance, whereby the level is lowered. It is so to say self-repairing, but to force them to do that properly, the magnet power should be very high. This also fights bass output by creating a low Qt. So how to conjure bass again.
There are different solutions for that, but they all demand closed cabinets, very low resonance frequency and low Qt for the loudspeaker with enclosure.
Units present on the market in general don't fulfil these demands. There are a few that can be rebuild, but adding a magnet on the backside of the magnet construction might help sufficiently. For that to be the iron part must be plane, in conflict with a good part of units on the market.

The simple and the best to regulate

With great magnet power you also should have higher sensibility, if it isn't wasted in

widening the airgap.
It is possible to ***lower this sensibility, and in same time keep the bass output intact.***
The need for that is of course dependent on room and bass response.
The real advantage in the solution, is its ability to be optimised to perfection to fit into your room, but the gained efficiency and perhaps a little more will be lost.
Connecting a second equal unit in parallel will increase the energy with 6 dB.
This again has a price in doubling the volume of cabinet and halving the impedance so the amplifier must deliver the double amount of current. This shouldn't create real problem, as smaller cabinets often are placed on a stand. The volume within the stand should be the volume for you to use. The impedance is halved, but if you start with 8 ohm you reach 4 ohm, easily for modern equipment to work with. (The actual DC resistance is lower (6 and 3))
The circuit is straightforward:



The network is constructed to have linear impedance, dampening in normal way (Rs and Rp). The Ls and Cp parts outbalance each other.
At low frequency Ls is open and Cp closed, at high frequency Ls is closed and Cp open. Thereby the resistors dampen the higher frequency response and the reactive components lead the deep bass through.
The phase turn of this network forms an inductively turned low “hill” on the phase curve, and it is the placement of this “hill” that is critical.
The mechanical parts forming the resonance frequency form likewise around the resonance frequency phase turns. Under Fr an inductive “hill” and over Fr a capacitive “anti hill”, these two curves nullify each other at Fr. It is into this “anti hill’s” upper part, the of the network created “hill”, must be placed, trying to straiten it out.
You have to find this point of your loudspeaker, dependant of resonance frequency and Qt. According to my calculations it can be found with sufficient precision from the expression:
Frequency for maximal capacitive turn Fm=Fres * 1.1/sqrt (q) when q is smaller or equal with 0.5. (That it is 1.1 and not 1 is caused by the influence of the inductive part)
You should work with low Q-values.
The matrix construction helps considerable on this point. With Q-values over 0.5 you must increase the magnet power by attaching an extra magnet, or try to soften the

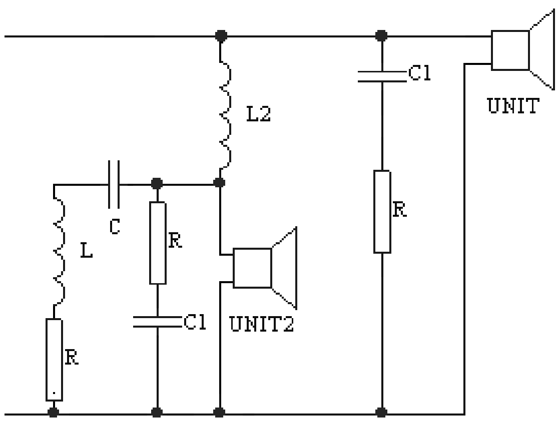
suspensions. The front suspension can be treated with silicon oil, used to protect the rubber lists on your car. This helps enormously on the dependency of temperature some modern rubber/plastic types have. Concerning this front suspension, it should be soft, but regrettably the foam, often used for that, doesn't last, but break down in structure. Also here this silicon oil helps on its durability.
From measurements in your room you should know how many dB you want the very deep bass level (under the lowest standing wave) to be risen. You know the loudspeakers resistance and the frequency of maximal capacitive turn.
With these you should calculate.
Loudspeaker resistance =R
Wanted damping in dB =dB (shall be negative)
$Rs= R*(1 - 10^{(dB/20)})$
$Rp= R*(R/Rs - 1))$
Frequency for maximal capacity turn = Fm
$c=1/(1000*R^2)$
$f=sqrt (1000*Rs/(c*Rp)) / (2*\pi)$
Ls=f/Fm mH
Cp=c*f/Fm * 1000000 uF

Control: 3.2 ohm, -8 dB, and 50 Hz gives
Rs=1.926, Rp=2.117, Ls=9.716, Cp=948.785
The inductor coil must have very low DC-resistance.
With this circuit attached, you further can regulate on the size of Cp, to change the slope of dampening. If you decrease the value of this part, you can smooth the work of the network. This could be chosen, if your living room is of the reverberate type. But you have to experiment with that.

The electronic way.
If you drive your bass unit with an amplifier of it own, there is the electronic solution. It's called the Linquitz-Grainer network. It's an active part with which, you - within some limits -can decide the behaviour, your loudspeaker will seem to have concerning Qt and Fr. But most important it will nullify the turn of phase created by your bass unit. It has been used to conjure bass up from small boxes with high Qt. That is to use it in a very wrong manner. It will correct the input all right, but will do nothing to the tendency of a high Q unit to vibrate at its resonance frequency generated by the standing waves. It came forward, was used and disappeared again, as it couldn’t correct a 15 inch bass unit in a 20 litre cabinet, and is to my knowledge rather unknown.
Also for that to work properly, you must have low Qt.
You must search the literature for that. If interest arises I will put it on this page.
An interesting use of this would be to straighten the units all to have the same resonance frequency and Qt, and then use theoretical filters, this only when electronic filtering is used,

The charming way.
With this you again must use two loudspeakers. But this time you don't connect them in

parallel in the normal way.
The first of them must be seen as a single unit and treated, as should it be the only loudspeaker of its kind.
The second unit is corrected for impedance variation as normal to form a resistor electrically.
If the two units share the same volume, the impedance variation must be found with both units driven simultaneously. If they are identical, you can measure both connected in parallel and double the read values. If they are different you must use a stereo amplifier and tone generator, and calculate the impedance using a voltmeter and an ampere meter for each of them.
The LCR circuit - to correct for the rise of impedance around resonance frequency for the first unit- compensates and thereby reflects the magnetic power seen from the voice coil. Thereby it also reflects the decay of level, as current through this circuit. So what you do is simple. You substitute the resistor with the second unit - also impedance compensated -, throw the capacitor away, as it will damage the electrical Q of the second unit, and change the inductor L2 to a type able to handle high current. In other words with very, very low DC-resistance.



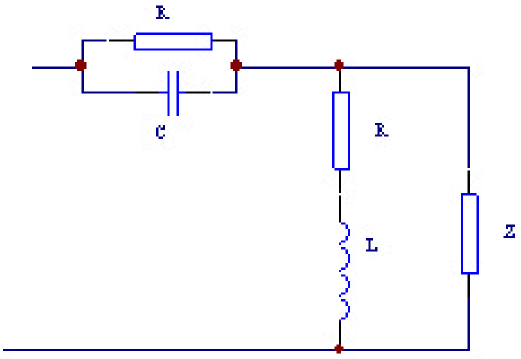
You now have a new single loudspeaker (both seen as one) with a 6 dB lift in the lower end, if your amplifier doubles current when the load is halved.
This method is filling the decay caused by the great magnet power beautifully. The filter function used consists of the mechanical capacitor inside UNIT and the passive L2, and is therefore of first order. You can, if you like, make it to a mild second order by connecting a further C1 parallel to UNIT2.
This capacitor you may choose with or without. Listen carefully, if you can hear the difference, and decide from that.
Again there is a price to pay, as the two units are coupled in parallel at the lowest frequencies. But in normal music there are not much energy there, so you shouldn't care. I call it charming, because even the smallest detail in the bass response is heard. It somehow is so easy to listen to, so charming and in my ears so right in its richness of detail in the bass, seldom heard in reproduced music.

When playing very loud there may be problems. If the two units are too different and they share the same volume, they are connected electrically and mechanically in the lowest band. Take care of their equality and you should have no problems, or don't play very loud - it isn't a disco loudspeaker - just loud.
This solution can be chosen with to different sized unit if your room for listening is large, as the amplification done by the room, then will lie low in frequency. You also may choose two enclosures and experiment if they should be sealed or connected with an airflow resistance. Their equality of Qt and resonance frequency is important and can be altered by loosing suspensions, adding weight or adjust magnetic power.

Comments on peaks in the frequency response and measurement
Variation in frequency response is normal unavoidable. The unit forms it, but also reflections on the enclosure play their part. These variations are easy to measure, but they are not as easy to extract the right information from.
Working seriously with loudspeakers you must have lots of damping material around treble and midrange, in order to hide the cabinet and it edges from these two loudspeakers whole/upper band.
That fits perfectly with the need for these two units to be pushed backwards to place their acoustic centres directly over the bass unit's, so there is good space to be filled.
Actually the ***acoustic centres should form an arc of a circle*** with the distance to your listening position as radius, and the units should point directly towards you.
Don't listen to these people, who tell you to listen of axes. If they recommend that, you can be sure something to be very wrong with the loudspeaker. That, not you I hope, can fool only persons, who weight the measurements done by a microphone.
By measurements you should use different distances between microphone and unit. When you examine this row of curves you will see peaks/dips which are steady and others that move in frequency. The "movers" should be judged only from measurements taken with long distance to the unit, as the movement in frequency is caused by the geometry in and around the unit together with the distance to the microphone.
You should fight the irregularities by mechanical means, as far as you can, before you regulate them electrically.
It might be help for you, to find the sources of these, if you look at the transport of sound near the speaker not as sound but as a stream of air. That it actually is so, you can examine by the flame of a candle. It is therefore clear, that to avoid turbulence and reflections the surface must be very smooth. It is also obvious that the front suspension must be a main problem together with the sudden disappearances of support from the baffle.
This work is so closely connected to the practical making of a loudspeaker, which you have to wait to see used on a construction.

It is so common only to concentrate at the part of high level from a loudspeaker, as if their roll off is of no importance. I'm convinced that to be wrong. Again an opinion based on wrongdoing.
Mechanical filter parts within the unit you can use as well as electrical ones. You can

even change them magnetically, by adding weight or by softening the suspensions.
The lower roll off and the area, where it works as a piston, is normally ***the most perfect parts of the loudspeakers response***, so why not use that whole. The only problem is the units' behaviour at resonance frequency, but that can be solved.
The front suspension has a tendency to form second order distortion by the uncontrolled deflections of the diaphragm at higher Qt than 0.5 – even smaller. That can be improved on, and the rest is not an enemy of sound as the level from the bandpass and highpass loudspeakers hopefully are low there, so also the distortion,
When 12-dB filter is used, resistors must cancel the electric components; to let the mechanical components take over.



By inspecting the C and L dependency of frequency you can get an idea of the size of the resistors, where you must stop the capacitor to close and the inductor to open.
With 24-dB filter you should be able to use just two components. It is not so straight forward, as it sounds, and you probably also must work with the correctional network on resonance frequency, to make it all add up.
This is another way of looking at the whole process. Start with the roll off part to get that in the right level with a minimum of parts and then work further with the curved part and be free to vary the load impedance for these parts to see. In this way you will be surprised, how peaks and dips are connected and corrected at once. It is really a troublesome work, but what you gain in reducing the number of components, to achieve the wanted curve, will reward you in sound quality. To my experience you should start with the treble and let that part decide how the other units should behave. It is much easier to form them than the treble.
But again - all this is best illustrated in practise on coming constructions.

How to fight peaks in an impedance linear way

Sometimes you have peaks where your filter components are open or have low impedance. Then you can't use variation of load. In these cases you have to compensate the peak in a way for the filter not to see. It must have linear impedance.
Now peaks are seldom symmetrical, so you must examine the peak for that, and weight

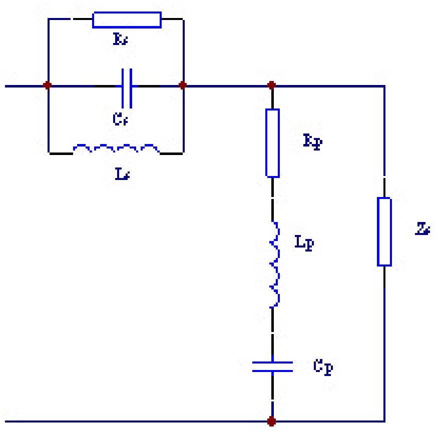
which side of the peak you'll find most important, and then make a drawing of the peak symmetric, from which you can extract the needed information.

Loudspeaker impedance at DC =Zs
Wanted damping in dB = p (p must be negative)
Then serial resistor Rs = $Zs*(1-10^{(p/20)})$ Ohm
Parallel resistor Rp = Zs*(rht/rs - 1) Ohm

You now must calculate a new level within the peak for which you must find two frequencies: Flow and Fhigh.

Level = $-p+20*\log(\mathrm{sqrt}(((10^{(p/20)})^2+1)/2))$
To control your readings sqrt (Fl * Fh) = peak frequency
q=Fpeak / (Fh - Fl)
Wp = $2*\pi*$Fpeak
Lp = $q*rp/\omega p$ *1000 mH
Cp = $1/\omega p/q/Rp$ * 1000000 uF
Ls = $Rs/\omega p/q$ *1000 mH
Cs = $q/\omega p/Rs$ *1000000 uF

The circuit:



The components used in such a circuit must be selected with care. The Lp coil can have some of the Rp built in. For the Cs and Cp parts you must avoid electrolytic, as they don't follow theory precisely. The Ls coil must have very low DC-resistance. When the circuit is built, then test it for variation of impedance, a resistor should replace zs, and the measurement must give a totally straight line.
You should by all power avoid use of this circuit, but some loudspeakers are tormented by very high peak level, say more than 6-dB, very hard to silence, then this circuit may

be used. It works miraculously well on the peaks beyond hearing ability from metallic domes, and should be placed as near as possible to the unit, to work at its best.

How to soften the rear suspension.

This part of the loudspeaker is often causing dips and peaks on the impedance curve and thereby, when passive filtering is used, also effect the level of the loudspeaker. The suspension should be much softer than normal seen, but serves also as protection, when the loudspeaker is used in bass reflex systems.

Inside the closed cabinets the air acts as a spring and resists the movement of the loudspeaker. Therefore the loudspeaker here should have extremely low resonance frequency in free air, to purify the spring behaviour to be that of the air alone, and in same time achieve the lowest possible resonance frequency and Qt. The spring character of the air is further more to be slightly regulated by incorporating an air flow resistor.

The easy way:

You simply massage it with your thumbs to more softness. You could further burn some holes in it with a solder tip, and let it be with that.

The troublesome but fare the best way:

Don’t try this, unless you are a skilled person for handwork and know loudspeaker mechanics by heart.

You must take the unit apart, by use of some solvents and patience. You should end up with following parts.

1. Magnet
2. Basket (if it is possible to take it apart from the magnet)
3. Diaphragm with voicecoil flex-wires and front suspension

From the rear part of the basket you should hacksaw away all parts disturbing the flow of air and unnecessary for the reassembling.

The magnet and modified basket is assembled again, and supporting parts of wood are glued on the sides of the magnet and basket to form support for the basket and for the new suspension.

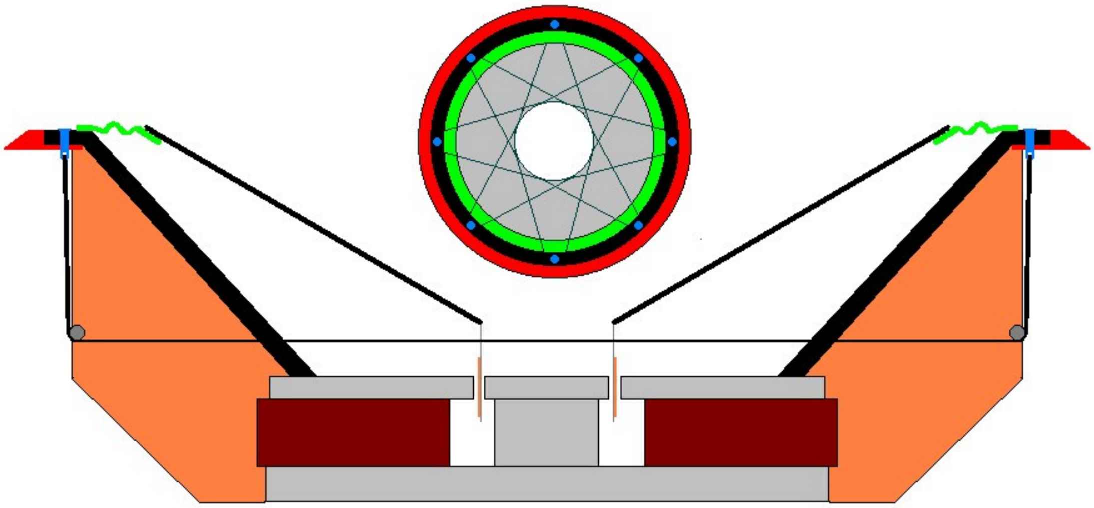
The mounting holes in the front ring of the basket are used for regulation of the steering wire, so you must cut a new ring for mounting purpose. This gives you possibility to create more space at the backside of the unit. Even small compression here causes problems.

It all are assembled, connections soldered, and a woven nylon wire put into place (see Fig) and glued to the diaphragm or the voicecoil. If for the voicecoil a metallic form is used, you should beware the heat built up playing loud, why you must use glue capable to withstand this heat.

For twisting the wire a screw fitting the hole is shortened down and made flat with a hole for the wire. This part must reach through the material for the basket and the part for fastening the unit to the baffle.

You can in this way regulate the tightness of the wire supporting the voice coils placement in the magnet.

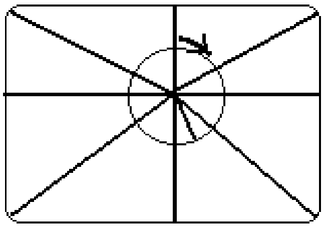
By this method, it is possible to reduce the need of air around the voice coil to as little as one tenth of a millimetre. My stepson has speakers of that accuracy, he plays very loud and has had no problem. ***It works***.



How to find the mechanical filter's components.
This procedure must first take place, when your unit has played for some time, and you are satisfied with its working manner. Even the slightest change will effect the size of the mechanical components.
When the unit is corrected for rise of impedance around resonance frequency, that circuit is part of a system, like that for correction for peaks. Therefore you can calculate the mechanical components from the components of your correctional network.
R = loudspeaker DC-resistance. Rs is the measured impedance at resonance.
You have Rp, Cp and Lp from the correctional network and find the components inside the loudspeaker.
$Cs = Lp * 1000 / R^2$ uF
$Ls = Cp * R^2 / 1000$ mH
For control: $Rs = R*Rp/(Rp-R)$ Ohm.
From this you can write a transfer function - second order high pass.

Method of measuring.
This is misleading and complicated, if you use a normal enclosure and room, as what you think you measure isn't.
Measurement done by the manufacturer is of no use for you, as the difference in basic conditions is so different. One thing you can be sure of, yours will be worse.
The cabinet, in which the loudspeaker is placed, and the room itself serve as reflectors. The reflections mix with the sound from the loudspeaker unit, why the graph can be said to be useless. But there are ways around that
To get an idea of how the unit itself measures, you must place the microphone in the

nearfield (1 to 5 mm) of the diaphragm. This method doesn't tell the truth for the upper end of the unit, where dimensions and runtime of sound will interfere, but from the frequency where the ½ wavelength is bigger than the diameter of the diaphragm you get a very precise trustworthy measurement.
In order to measure the rest of the frequency band, you must silence the cabinet with rock- or glasswool, and perform a row of measurements from different distances. The room will disturb, but by examining the graphs, you will get a reasonable good idea of the units properties.
On order to find the sources of irregularities a calculator must be at hand, and it is advisable, before you measure, to find the frequency for ½ the wavelength for all distances form which to expect disturbance.



Irregularities are either stable or moving in frequency dependant of distance between unit and points of reflection and microphone.
The "movers" are created by change of geometry between microphone and unit and must be optimised to listen position. These should by all power be corrected mechanically to be stable by damping points of reflection with SOFT felt or rock wool placed on the baffle.
The stable peaks need action on the unit, according the principle "Try and Error", and can be a rather costly affair, considered the amount of unit wasted.
It is less complicated to go for electrical solutions, even if they have a bad reputation. It is much more a question of quality of the components used. Here you must choose between the Devil and the deep blue sea. Some like these peaks as they add more pace and drive to it all. I don't, I find them disturbing.
To get a stable curve to work from, you may measure from a distance of 15 to 20 cm and find what is missing in its upper end comparing with measurements from distance (2 to 3 m), and remember to account for that mismatch in your work.
The graph must follow the wanted graph form this distance, but not be measured from it.
When you believe it to be right, you should measure the unit from different distances and see if the tendency is invariant of distance.
When this is done for every single unit, you must control how they work together.
You must work with two units at a time, and have your microphone placed equally spaced from the units acoustic centres. From every measuring distance you must perform 4 measurements of level.

1. First unit.
2. Second unit.
3. First and second in phase
4. First and second in antiphase

From this curves in comparison with the calculated ones you should be able to get the two units to work correct together. You must undoubtedly modify your filters, but so it is aiming at perfection.
You must do this procedure with every two unit in your system.
Reflection on the floor will disturb measurements in the bass. You can be free of that by placing the loudspeaker enclosure on its side, for the loudspeaker unit to come as close as possible to the floor, and measure the bass from that position. Raised again you will still have the problem, but when playing it needs time to build up, so the transient will come through in correct manner, and that count most..
When all this work is done, your loudspeaker probably doesn't look nice. The next work is how to reduce and/or cover the needed damping material. That I'll leave to you, but remember to measure and measure again under this process.

The magnet
This part is easy to make yourself, if you have access to a turning lathe and a very powerful magnetiser.
Years ago, before high-powered amplifiers came forward, much work were done, to increase the efficiency of the loudspeaker. Lowther fullrange units put into a great horn-construction, was a masterwork from this time not to mention Klipsh and Tannoy. The magnet was very cleverly constructed. But a war in Africa put a spoke in the wheel for deliverance of cobalt. The alnico was replaced with ferrite, which in many ways is easier to handle. The precision wasn't so necessary any more and the row of so-called improvements placed their deteriorating trace on production. These "improvements" were much more directed on the yield of the invested capital than of the quality. Onto that came the closed box's believed lower demand on Q-value for the bass unit, which was a helping hand for the bad development.
You can't get bass from a powerful unit, was the general opinion and still is.
Two very important facts aren't taken into account.

1. The loudspeakers are placed in a room.
2. The moving mass must be high, and the suspensions soft to keep the resonance frequency low.

It is really strange to me, the amount of energy put into the reproduction of treble, which ought to exist hidden, where the bass, which also should exist hidden, is totally forgotten. It's the least troublesome to calculate and make.
A room has reverberation governed by dimensions and reflection-coefficients of the different surfaces but also the frequency of the signal, when the wavelength is comparable to the dimensions for the room. Then resonance builds up causing great variation in sound-pressure dependant of position in the room. The summarised effect of this in the bass area can in effect be compared to a mild inverse 12-dB low-pass filter. This effect should be taken in account with the loudspeaker's slope of cut off, why the closed box is the only useable when room and loudspeaker are looked upon together.
The room was compared to a mild inverse filter, why the bass loudspeaker also must have a mild decay - a mirror image of the rooms builds up. To achieve that, the magnet power must be very high. With the loudspeaker placed in a closed box the decay probably still is too quick, so further regulation is needed. This is described in "The bass

response".
A further argument for great magnet power for the bass unit is the simple fact that a loudspeaker also acts as a microphone - also described under "The bass response"
The power parameter - talking loudspeakers, is the Bl product. "B" the magnet power and "l" the length of wire in the magnet gab is what creates the force factor. Lots of other parameters interfere, but let's stick to these two for now.
To increase the length of the wire you must lower the amount of insulation to squeeze one or two windings more on the coil – seen on single wounded coils with flat wire.
But there is more to gain by increasing the diameter or height of the coil present in the magnet gap. That leads us to the construction of the magnet as the main point.
The normal production of magnets for loudspeaker is too much guided by low expenses. It will cost more to make them better. Modern technology with CNC-steering makes it now within economical reach to make the parts softly shaped to avoid sharp edges. The nature of the magnetomotive force is to establish a magnetic field reaching far out. Ferromagnetic material attracts this field to be concentrated in the airgap. The iron normal used should have high permeability (ultra low level of carbon), but again that costs - so the demand for this product has sunk so low, that production to my knowledge has stopped. We have to be content with, what we can get. Therefore the iron must be thick. It is normal to use 6-8 mm plates – It is far too thin, I use 20 mm.
The centre tap is and must be the keyhole in the magnetic circuit, in order to establish as many field lines in the airgap as possible. Therefore the dimensions of the centre-tap dictates the greatest height of the airgap.
The cross section forms a circle with the area of $r^2*\pi$.
The receiving band of the front-plate with a gap of 1 mm is $(r+1)*2*\pi$*height of iron.
The maximum height is therefore: r^2 **/ (r+1)*2**
Examples: r=20 mm, height =9.5 mm r=25 mm, height=12 mm
As a rule of thumb the height can be set to a quarter of the diameter of the centre-tap.
That the height of the front-plate often is seen thinner, is an attempt to saturate the iron around the voicecoil. For that other types of magnets as Alnico or Neodymium must be used. Again the price is high

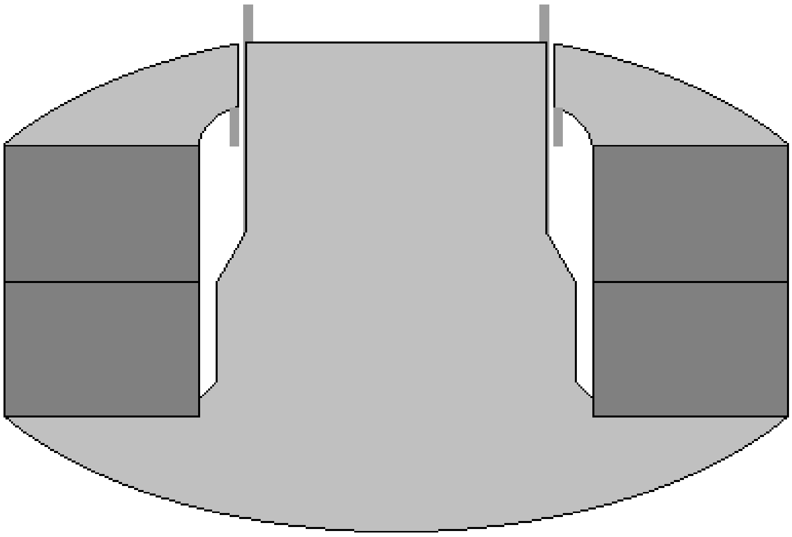


Figure: The magnet construction

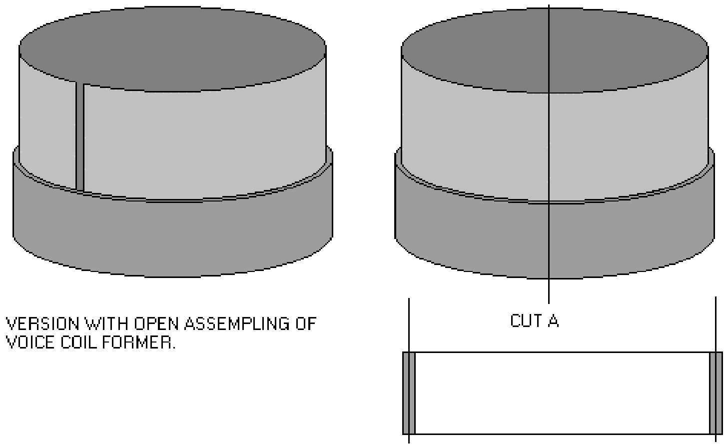
As seen from the figure, a construction of copper-rings fights the rise of impedance caused by the voice-coil. Likewise these rings fight variation of the permanent magnetic field. I'm not sure if more could be reached by expanding them to follow the inner curve on the front-plate and cover more of the top of centre-tap. I'm neither sure if the iron structure could be even better. It would be better, if we could get the corners of the magnet embedded into the iron. But the variation in dimension for the magnet makes it very difficult in a production. It should be possible if the iron parts aren't cleaned for the rest of iron always formed by the cutting process, until after assembling. Then this rest can be bent over the corner of the magnet and thereby be adjusted to the actual magnet. It is handwork, risky for your fingers and therefore expensive.
A further investigation into this, demands access to an expensive computer program, I don't have.
The magnet must be seen as a foreign body placed where pressure is high, why its form isn't trivial.
I have used this construction for some years now, and have gained 4-5 dB by that.

The voice coil.
For this little part, which is placed in the heart of it all, there seems to be a lot of general misunderstandings.
The winding form was earlier made out of paper, but by modern time's higher demands on effect-capacity other materials have come into use.

1. Aluminium – this material is formed from a rectangular piece.
   At the point where the two ends meet, the form is open – a problem.
   As it is a metal - vortex current arises, which effects the free displacement of the diaphragm.
2. Kapton - this is also formed from a rectangular piece.
   Besides it is plastic, and that I avoid by all power.
3. Fibreglass is used on professional units, but it is heavy and too thick.

The material for the winding form should be light, thin, strong, be able to resist heat, be an isolator and possible to wind to a thin-layered cylinder to avoid the open assembling. The only solution, I can think of, is thin paper impregnated and wound with water glass.

This material could also be the glue that fastens the wire. To secure best possible cooling, the coil should be wound the way Lowther uses - one layer inside and one outside for better balance. The open suspension I suggest and the copper rings coloured black will help too.
If that is the right solution, I don't know for sure, but it's a possibility. Another material could be aluminium oxide with diamond structure or glass but how that can be manufactured I really don't know.



Another problem with the voice coil, I have stumbled over, is a missing ability to deliver energy to the diaphragm at a certain high frequency. This phenomenon is related to the length from the bottom of the voice coil to where the diaphragm is placed on the form. It seems as if this point forms a node, unable to deliver energy. If the distance is e.g. 4 cm, then nullification will occur about 4 kHz. At 1.1 cm it will happen at 15 kHz. Lowther seems to have had the same problem, as the assembling diaphragm/ voice-coil is unique. A simpler solution is needed. It will not be possible to construct a good middle tone loudspeaker in the normal way, if this problem remains unsolved.

The last problem is the wire itself.
It has positive temperature coefficient. This means that when playing, the impedance vary with warmth. The passive filter is disturbed, as is the amplifier, why transients in the music are dampened. Solutions involving the amplifier have been suggested, but why don't go directly to the cause of the problem and develop a wire material with a coefficient around zero?
Modern science should have no problem to develop an alloy fulfilling that demand, it has succeeded with alloys for resistors, so why not try for conductors. The price shouldn't count much, as the amount of wire for this particular purpose is low per unit.
A solution, in which I personally believe, would be a mix of copper or silver and graphite/nanotubes.
The net temperature coefficient would be very near zero – done right.
Graphite fibres could be woven to at thin string, and the metal put on/in by an electrolytic process. The technique works, but it is impossible to develop to the perfection for a single person and besides I don't have the knowledge of metallurgy needed.

I really would prefer it the other way around – copper/silver with a layer of graphite, but I don’t know if that is possible.
From literature I know of another possible way to do it. By nano-technology it is possible to create nanotubes. These are developed from a in 1985 discovered family of carbon crystals called buckyballs or fullerenes. Much research has been done into this area to find practical use of these new forms. It is possible to bind atoms to this molecule as every single carbon molecule is connected with two single- and one double binding. It should therefore be possible to develop a great variety of conductors. Hopefully one of these would have low resistance and small temperature coefficient. The future will show if it is possible. The market for such a product is present worldwide.
So much for the voice coil, which by its mechanical connection lead us to the next problematic part – the diaphragm.

The diaphragm and the front suspension
For those who really seek the true sound, these parts have to come into consideration. The practical and mathematical skill for you to have is now far beyond normal. But let’s give it a try.
Nature has a solution for nearly anything, if you can find it - so also for the loudspeaker diaphragm. It is said about the diaphragm, that is must be stiff – and light, but that is a qualified truth.
The weight is dependent of the force moving it, the position of the resonance frequency and its Qt and the efficiency wanted.
It is your wish of the final result that determines that parameter. It is a question of balance and not just one parameter.
The stiffness is a parameter on which I can agree but disagree with how it is practised. It is normal to use stiff material, but that causes problems at the first break up point, where the stiffness energises that break up beyond possibility of regulation.
NB! By precise adjustment the first break up of the diaphragm and the nullification from the voice coil could be placed at the same frequency and by care probably outbalance one another.
For me to see, the stiffness must be created in shape as well as in the material, which therefore must have high strength of stretch but not necessarily be stiff. This property is characteristic of kevlar and carbon fibre. *(Fibreglass could be considered as a cheap material to use for experimentation.)*
These fibres can be bought in various qualities and densities in one - two- or tree-axial weaving. It can be formed in any shape you like and kept there by use of hard hardening epoxy. The hardening process must happen slowly and the amount of epoxy left within the weaving kept to a minimum. This material can also be found mixed with thermal plastic material to be formed easily by warmth.
The shape of such a diaphragm must be ***guided by mathematics***.
In the same way, that the shortest distance between two points is a straight line, closed lines will form an area between them, a surface, which can have many shapes. One of these surfaces is the smallest, the “minimum surface”.
A loudspeaker diaphragm can be said to consist of two circles with a connecting

surface.
The minimum surface is by nature formed automatically by a film of soap, which always will form a surface as small as possible when the gravitational force is accounted for. The ideal form can be described mathematically:

Radius voice coil: r
Radius rim: R
The searched curve can now be drawn on a millimetre paper with the voice coil centre placed in point (0,0). The height follows the x-axe and the growing radius of the diaphragm follows the y-axe.
The amount of points (height, radius of diaphragm) = (h, rd) can be calculated from:
$Hmax = r*LN (R/r + sqrt (R^2/r^2 -1))$
$Rd = r*cosh ((h*arccosh (R/r)/Hmax)$

This curve is idealised as the angel between the diaphragm and the voice coil is zero, whereby the transfer of sound to the diaphragm should be perfect. But the speed of sound in the material for the diaphragm is a major factor to be cared for.
You have the freedom to use any point of height as place for the connection of the voice coil. Then the radius of voicecoil and Hmax are uncertain. This leads to another set of formulas, where R, r and Hmax are fixed.

$Rd = c*cosh (h/c + arccosh (r/c))$ where c is found from the equation:
$Hmax = c*LN ((R/c + sqrt (R^2/c^2 - 1))/(r/c + sqrt (r^2/c^2 - 1)))$

The speed of sound is dependent of the material and the epoxy, but to make it even more complicated it is further dependent of frequency as well. The speed is somehow inversely dependent of frequency by a law I don’t know of, but that shouldn’t disturb so much, as it further again is a complex mix of longitudinal and transversal waves, that you can’t avoid measurements. You most learn from them.
It is clear, that near the voice coil where the curvature of the diaphragm is steep the higher frequencies with their lower speed should be transferred to sound. Adding an extra and smaller cone normally does that. This solution is catastrophic, as the loudspeaker thereby looses its minimum phase characteristic. One can - as done - forget it or stop caring, but striving for perfection I see no way without. The possibly right way to do this came from research on quite another matter -

The distribution of mass in the diaphragm.
When a cone-diaphragm is energised by a voice-coil, the energy is spread out in the diaphragm to a higher and higher mass formed by the greater area and constant thickness, and that can’t be right. The amount of energy per volume unit should be constant. This would be fulfilled if the thickness of the material were inversely related to the radius. Further the momentum seen from a hypothetical centre also should be constant, so we again must have mass (thickness) inversely related to radius. The thickness of the membrane must therefore be inversely related to r^2, for the diaphragm to be in balance with the force inserted. You know the thickness at the rim, and from

there you can calculate the thickness inwards. Now the question arose how to follow that rule, when the material was equally thick. The idea of producing a sandwich of differently sized 8 point star-shaped parts on the back or front of a Circular base, which in all followed that rule, was executed and measured.

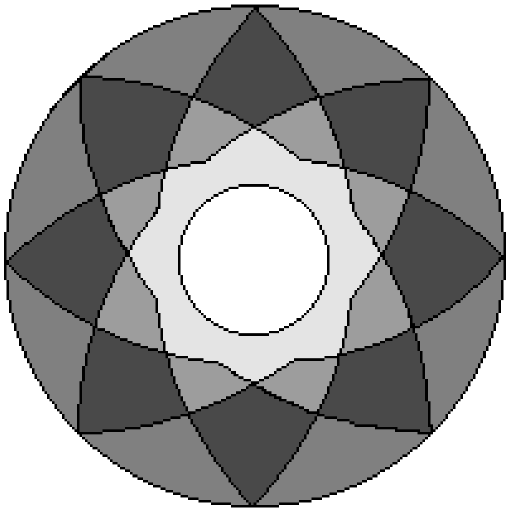


Figure: star-shaped cones drawn with a compass. It’s not all mathematically correct, but it will give you an idea of its construction.

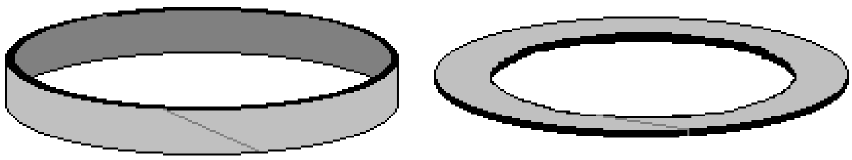
Big was the surprise, when this construction of the diaphragm showed to have ***extended frequency response***, despite its heavier mass. This was most satisfying for the bass-mid units I worked with. I just often used the construction for bass and didn’t think further. Years later a friend asked for my opinion on the possibility of making a full-tone diaphragm to use with a Lowther magnet. I considered it possible, having the star construction in mind.

The construction was changed to have the sandwich of three star shaped diaphragms on the front to form a four cones unit. All four cones are equally shaped, so the speed of sound is the same. The possibility for the high frequency to couple to the air is changed by the many endpoints of diaphragms. It was to be used with a PM2 magnet, but that was changed to PM6 and also a too high voicecoil was used. Despite that, its frequency response from 20 to 20 kHz was smooth with a minor dip from voice-coil nullification and within -10 dB at the end points- a beautiful bandpass function.

This result was marvellous from an 8-inch unit and it should be possible to scale down to 4-inch. A friend and I are in the middle of that experiment, but lack of time together

prolongs the process. The need for a bigger voice-coil than the calculated 20-mm disturbed more than expected and for now we have a deep nullification at 15 kHz. These problems will be solved, but measurements on this unit without front suspension have directed my attention to that part. The sound of this unit was so different, so clean compared to the same unit with front suspension, that this part ***simply must be silenced*** one way or another. Its deteriorating effect on sound is far greater, that I've even dreamt of.
I was unlucky also to work on some dome tweeters at the same time, and really it's a bad construction as well – even the expensive ones. So also that construction needs a closer look.

The front suspension.
The experimental 4-inch unit with the star-diaphragm had a very linear frequency response but with a dip around 1 kHz and nullification at 15 kHz without front suspension. It was beyond recognition with that part attached - still the null at 15 kHz wherever the microphone was placed. Therefore it ***must*** come from the voicecoil itself. Many different solutions for a suspension were tried and one fell out to be most interesting. A woollen string filling the space between diaphragm and basket left the upper part nearly untouched. It wasn't sufficiently airtight and silent. A mix of Merino and Angora wool was silent, and a tube was knitted. It has the advantage that, it can be knitted end to end, so it can't be seen or felt. The sound is wonderful, but it still isn't airtight, and every attempt to get it that until now, has punished us with disturbed frequency response. I hope it can be fixed one way or another, we just don't know how for now.
Shouldn't we succeed in this, a totally new way must be found. I have an idea of a flat suspension with a special willingness to move.
This new idea has now been tried with surprisingly good result. It has to be assembled and for that purpose soft glue used for diving suits works fine. This suspension is at start high but forced flat and held into that position by the gluing to the diaphragm. It has a tendency to seek out from this placement, but hasn't the power to move more than itself and the glue. It seemed to be a present from Heaven. It was by measurement really a surprise to se an untouched frequency response and at the same time have a tight suspension.



The material for this suspension, is foam of butyl used to tighten roofs with boards of Eternit. The adhesive side can easily be cut away with a warm thin wire to get a wanted thickness of 1.5 to 2 mm
This solution is most interesting and far the best to my experience, but from there to get a manufacturer even to try it is a long way, but ***here it is for you to try – it works.*** The

diaphragm and the basket must be glued to the in- and the outside of this suspension.

A dynamic tweeter – not a dome and not a cone.
I got this idea an evening after a longer discussion about the compression in domes, which a manufacturer had postulated his version not to have. That was a lie – of course! But was it possible to make at all?
Speculations led in different directions. As the discussion was started on a dome construction the thoughts was focused on that.
It should be possible to part its area to both sides of the voice coil, forming the well-known ring-radiator. This construction has been used for horns by JBL. But haven't been seen as an open construction.
This idea is presented as a sketch, but it should work and be without compression more than its closed chamber presents and can be chosen freely.
This version is inspired from the Vifa solution, Salute to them for daring try this other way of thinking. But shouldn't they improve it?

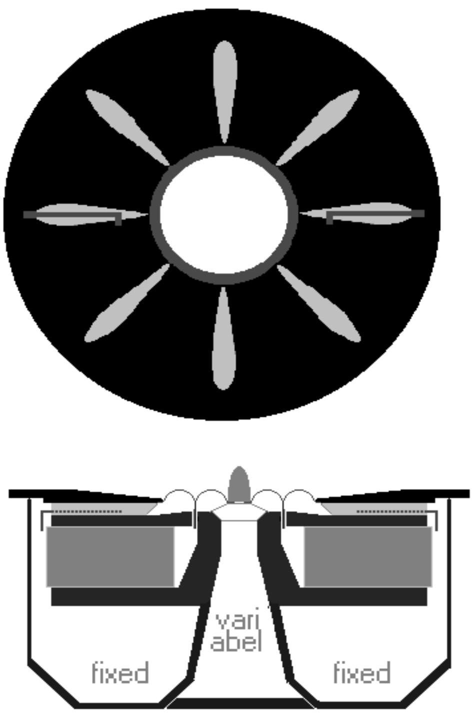


Fig. Ring radiator. The light blue supports are radial plates, so freedom for air is undisturbed.