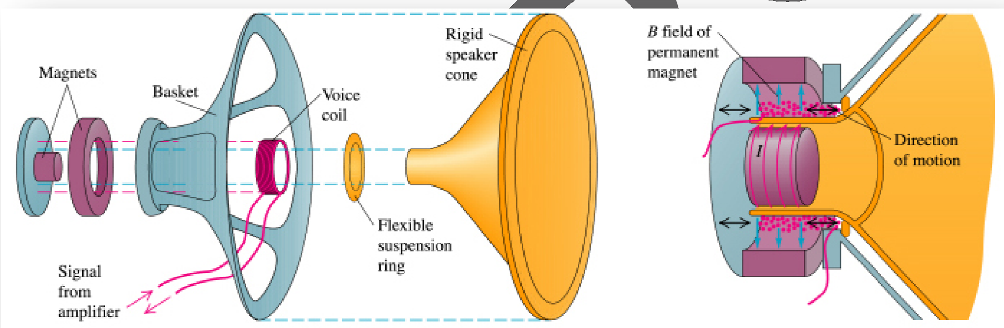
A nearly complete case for the horn

This paper attempts to describe, in least complex scientific terms, the challenge of converting an electrical audio signal to an acoustic one. In doing so it also attempts to explain the rationale behind the principles of horn loading and why Earo bases most of its designs on this method.

Beginning with the driver

You are invited to join in on a journey that will take you to some insights on why speakers are designed they way they are and why old school wisdom can be combined with state of the art technology to create outstanding realism in audio reproduction.

We need to begin where the rubber hits the tarmac, with the loudspeaker driver. It appears to be simple in concept and implementation, that's deceptive as we will show. In the picture below we can structure the speaker into two significant section, the frame and the moving parts.



The frame consists of the basket which is the driver chassis. It holds the magnetic circuit and the two flexible joints that allow the moving parts to move. The lower moving part both centers and acts as a return spring to the cone assembly. It is termed “spider” which is strange as it does not resemble one at all. At the other end of the cone is the other joint suitably termed the “surround”. We won't dig too deep into the exciting world of the magnetic circuitry and will rest to say that regardless of what source of magnetism is used, its purpose is to direct as much as possible of it into a circular airgap in which the voicecoil moves. The coil is wound on a “former” which in turn is permanently fixed to the lower end of the cone. The hole that thus appears at the point where the former /coil is affixed to the cone is normally covered by a dust cap. It´s purpose is to, yes, stop dirt entering the confined space between the coil and magnetic circuitry. On speakers that have a wider frequency range the dust cap may be replaced by a pointed tip called “bullet” or more correctly “phase plug”. The purpose is threefold, firstly it ensures that the wave-front is more uniformly produced assisting in improved spatial linearity. Secondly it effectively halves the diameter of the cone w.r.t to the issue of beaming and thirdly, it avoids issues related to the pocket of air that is compressed and expanded beneath the dust cap, such as noise.

A similar function is performed in the small sub-cone, often called a “whizzer”. Its purpose is also to provide a small radius cone in order to reduce the effect of the wave-front becoming flat rather than spherical, thus aiding in reducing the beaming effect.

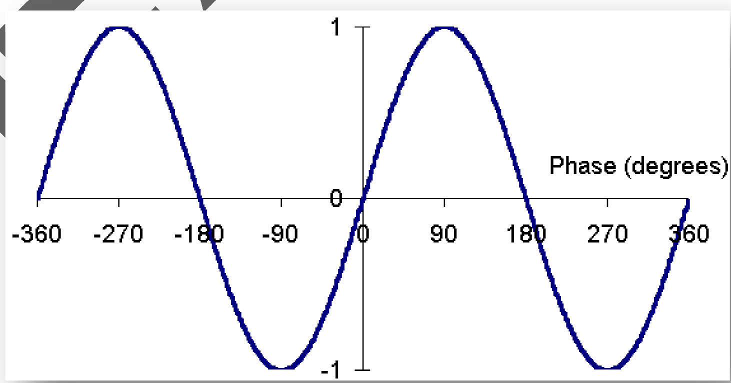
We will argue later on that for horns using a single driver, the total mass of cone and coil is ideally as low as possible. Instead, we can mention a few facts about the spider and surround. The materials used and their properties are essential in acquiring the exact performance of the driver. The spider is a spring and must act to return the cone to its resting position. Both the surround and spider must also provide hermetic seals while ensuring that the cone travel is correctly centered. The surround must also have such properties that the wave traveling in the cone, generated by the coil, terminates at the surround and is not reflected back. It´s about high level of materials knowledge and not an application of what you had in the scrap box. Surely you will now look at a speaker driver with different eyes?

Next, we need to get a grasp of what sound is.

Sensation of sound

When molecules of air enter the outer ear they will eventually hit the eardrum, here the movement of the molecules will be transferred to the eardrum and in the inner ear be sensed by the hairs of the basilar membrane. From there on it´s electrical signals to the brain.

The sensation of sound is proportional to the **average** velocity of the moved air. To know the actual displacement of the air all we need to do is to integrate velocity over a defined time period. By the word “average” it can be surmised that the motion of said air is not constant, if it was we would not hear anything at all. Instead, if the change is in the range of approximately 20 to 16000 times per second (Hz) we can hear it. Sound does not have to be a cycle of a certain frequency, it can be completely asymmetric in the shape of an impulse or transient. Regardless, in all cases there must be a rapid change in air-pressure that propagates outward from the source of sound.

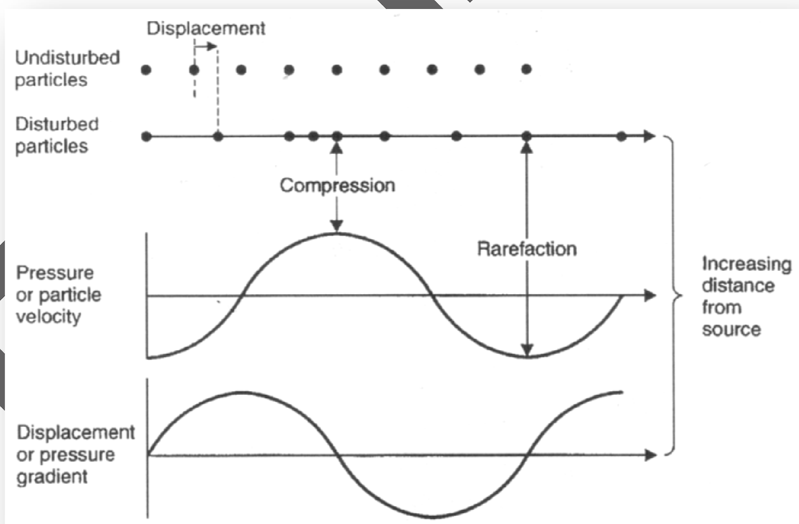


Picture shows a sinewave over two full cycles. The X axis shows the angular relationship but could also be time as frequency =1/t. The Y axis is the level or amplitude.

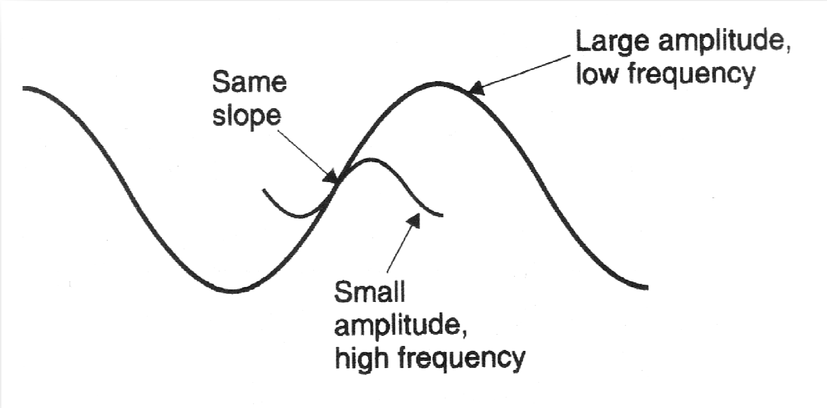
But lets look at a steady tone so that we may understand what happens in terms of pressure and velocity. Picture in your mind that the tone in the picture above is a sine wave driving a speaker. The cone will follow the electrical signal. Each peak, negative or positive, marks the point where the cone reaches its endpoint, slows to a stop and begins going back again. At this very peak the cone is neither going outward or inward. The power amp is driving maximum current against the spring of the cone and the air it has pushed, the cone is thus producing maximum pressure. Conversely, the part of the sine wave that crosses over from one polarity to the other is the point where the cone is moving fastest. This part of the sine wave has a clear attribute of great interest, a slope. We can see that the slope is a function of frequency and amplitude. From this it is clear that the higher the frequency the more vertical is the slope. Thus, remembering that sound pressure is the average volume of air per time unit, it is may be seen that as frequency goes up more sound pressure is generated if the amplitude is unchanged. Herein lies the underlying physical relationship that forces speaker driver units to become larger with decreasing frequency.

Hence, as frequency decreases the amplitude (cone excursion) or area of radiation (cone size) must increase for sound pressure to be equal. Thus, for low frequency we need a large radiating surfaces whilst for higher smaller ones suffice.

Two observations are key here; first the maximum velocity is the steepness at the crossover. Secondly, velocity and displacement are a quarter of a cycle apart, ie. 90 degrees phase shifted. Keep this in mind as we move along.



In the two pictures above and below (courtesy John Watkinson, "The art of sound reproduction") we may see the relationship between pressure and velocity of sound. Also seen is how the slope is a function of frequency and amplitude and why larger drivers are required for lower frequencies



Those of you that have experience in electrics may already have recognized some relationships here and made the connection. As an example, the utility company has a continuing issue with power consumers that load the network with other than pure resistive loads. Typical resistive loads are heaters and incandescent lamps. Typical non-resistive loads are motors and fluorescent lights. These reactive loads are called so because they in fact react to power fed to them by shifting the 90 degree relationship between current and voltage. When this occurs power is lost at the point of load, part of it is reflected back into the utility network causing all sorts of undesired issues.

Why all this talk about the utility grid? Well acoustic and electrical parameters can be modeled and they use the same mathematical and physical relationships! This really helps understanding what goes on. Sound pressure (U) or volume velocity (i.e perception of loudness), is in electrical terms equal to current (I) whilst the pressures (Pa), or force has an electrical cousin named voltage (V).

We can then apply a bit of OHMS law and soon note that the resistance is V/I and in acoustical parameters R= P/U

It is a gross simplification to use R here, because we are dealing with an AC signal and shift in between current and voltage means the R is actually something depicted by Z which means impedance. With a constant and fixed relationship of 90 degrees between the two, impedance is said to be purely resistive.

This situation is not always so, we have studied a sound wave that is planar. This occurs at some distance from the source. We need to move on to the next section to understand sound propagation.

Since horn loudspeakers are all about making a correct interface between the speaker driver and the air, getting down to the physics is key in understanding what goes on.

Sound propagation

Understanding how sound propagates, or radiates from a source is fundamental requirement for the loudspeaker designer and also very useful for the audiophile to have a grasp of. We will dwell on two phenomena, reflections and wave theory, they are closely related.

First we need to talk about a dimensionless parameter which makes discussing sound a bit easier. The wave number, *k*, describes the behavior of sound in relation to the wavelength. Since the audible sound spectrum has wavelengths from 17mm to 17meters, a lot can happen in conjunction with our listening environment. *k* is the relationship between a sounds wavelength and a size of an object in the path of sounds propagation. Sound behaves so that when the wavelength is greater than the obstacles size, sound passes through it (or around in fact). Whereas when the wavelength is small w.r.t the object, it is reflected. However when the two are near equal and k is close to one, interesting things occur. What, will become clearer as we go on.

Understanding the concept of wave number is very helpful in understanding how sound propagates and thus behaves in real physical world.

Lets consider a space with no boundaries, no floor, no walls and ceiling. A sound source becomes like a light bulb in a (perfectly) black room. Light radiates in all directions and nothing bounces back. You can see the bulb but you cant see anything else. Are you with us so far? Now if the room does have any reflective surface it will add a sense of dimension, the direct reflection and the shades aid the brain to interpret and build an impression of a reality. In fact, the perception of colour is defined by the wavelength(s) that a surface may reflect. White reflects a broad spectrum whilst a colored surface has a more limited range (bandwidth or spectrum) of reflections. So k can be found to exist in the metaphor of light as well and may make it easier to understand.

A speaker in a room acts similarly. Listening to a speaker in an anechoic chamber (acoustically infinite space) is an odd experience, there are no reflections and we are not used to this. What we hear is a point source and only that.

Since it is rare that reflective surfaces including objects in room, reflect equally at all frequencies there will also be shade or color, of what arrives back from the reflection. In other words the frequency response of the perceived sound is altered. We have introduced the concept on tonality. In a musical instrument, tonality is the quality factor whilst in a loudspeaker it ought to be the opposite.

The other mechanism occurring is that the room will, to varying degrees, produce unique mixes of reflections in any given physical location. In plain language, there will be a different frequency response in all locations in the room.

Since frequency response in a room is a product of delayed sound being mixed with direct the result is both amplitude and time (phase) shifts. So we have two mechanisms that color the listening experience.

The big question arises, is it an advantage or disadvantage to have the rooms acoustics involved or not? We will get around answering this question as we learn more.

Being this close to a soundsource that radiates as a single point means that we will be hit by waves that are spherical, a 3-dimensional version of the pebble hitting a surface of water.

The ripples propagate from one point uniformly, frequency (pitch) is constant but with amplitude decaying geometrically

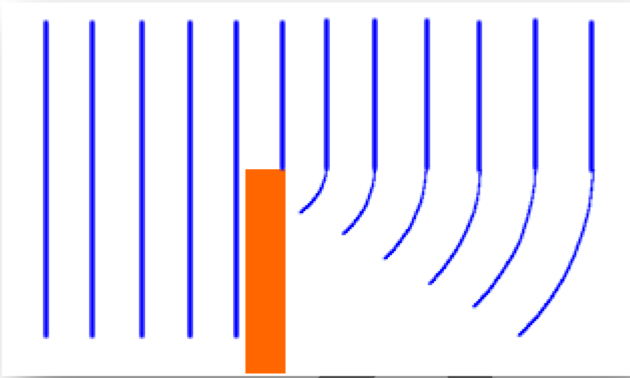
Wave number also lets us understand how sound propagates off a speaker drivers membrane. With a frequency being low in relationship to the diameter of the membrane the propagation is a spherically shaped wave. As frequency rises the propagating wave becomes more planar causing an effect called *beaming*. The result is that with increasing frequency the directivity increases. This is one reason why in loudspeaker designs that desire to have a broad sound dispersion in both planes, must have gradually smaller drivers or else resort to acoustic lenses. In fact, these drivers can be made smaller without penalty in sound pressure level (SPL) as we discovered earlier, by finding out the relationship between sound pressure and frequency.

But is it always a design goal to have a wide dispersion and several drivers?

Not at all, for instance in a public address system it is highly desirable to be able to have very close control over how sound propagates why the speaker systems are designed to in a controlled manner restrict dispersion. More advanced designs exploit the opportunity to by active processing alter phase relationship in between drivers in a cabinet to create a planar wave propagation. In a home audiophile situation there is a benefit in having a directed, planar source that directs energy towards the listener rather than into all of the spaces in the room thus exaggerating the effects of room nodes (reflections and standing waves). This will not do anything good in home cinema where several individuals need to share the spatial effects produced in the creative process in the making of the feature. But then this is not an audiophile situation either.

Edge diffraction and Baffle step

Having talked about the wave number, k we now should have a feeling that the use of the word "sound" is broad definition and to be clear, one need to be more specific to be understood.

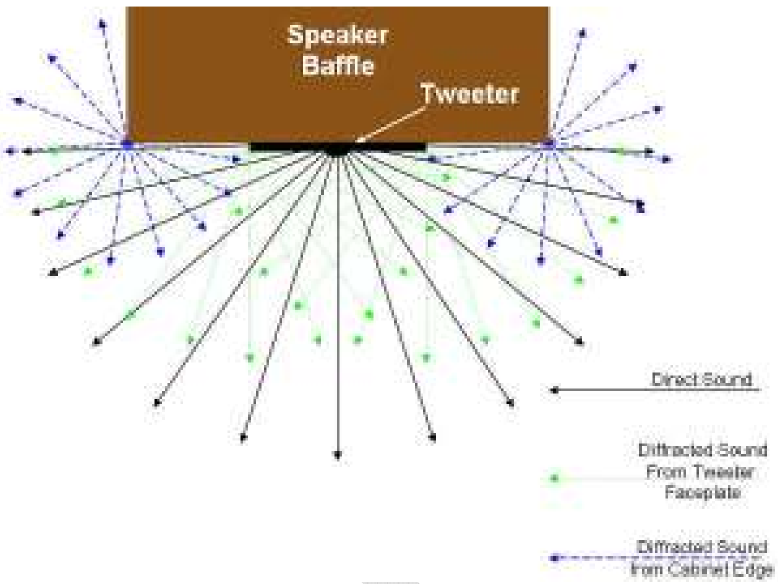


When k is close to one, neither transparency nor reflection occurs but diffraction

The impedance model can be used to explain an effect that is related to the wave number. As a wavefront propagates from the speaker driver it may be spherically shaped or linear. If spherical, the wavefront will meet the surface of the baffle and see a load that is related to the hemisphere it meets. As the wave rolls along it will eventually meet the edges of the cabinet. If the driver is centrally located (forbid!) then all of the energy will at the same time suddenly realize that the world is in fact larger.

In impedance lingo this is construed as a mismatch and as told elsewhere, mismatches cause reflections. What actually happens and it is quite audible, is that all those edges become radiators of sound. When k is close to one, that is wavelength and object is near equal then the edge diffraction is most pronounced. Thus edge diffraction is not a problem in the low end but it is in the mid and treble range. A typical effect of edge diffraction is the smearing of the stereo image, the speaker is no longer a point source. Clearly, the more drivers on the baffle the more edge diffraction will happen. The ideal cabinet from this aspect would be spherical with only one driver, nice but not practical. This is however the most *acoustically transparent* solution.

What can then be done to maximize acoustical transparency? Several things, firstly you can make sure that the wave can propagate out of the driver as cleanly as possible. When air-racing was big in the 30´s some understood the benefit of flush rivets on the airfoils. Same thing here, care pays. Also, by not placing the driver centered, the smearing is spread over a broader range. The cabinet can be made to have a narrow frontal area, which is clear trend these days but this only solves the problem in one dimension. It can be seen that the benefit of a speaker driver that is allowed to beam, that is produce a linear wavefront and thus exhibits narrow dispersion, is that it clearly directs most of the energy forward and not across the baffle thus lower edge diffraction to begin with.



Edge diffraction messes up the speakers polar response, making it look like the back of a porcupine.

The baffle step is closely related effect and is a most significant issue in the placement of speaker in a room. Here it is the effect of the radiation space and it is frequency dependent. What we don't get in edge diffraction we get as loss of low end as the apparent radiated space increases. Placing a speaker far away from a wall increases the radiated space, moving close to a wall reduces it. A corner furthermore reduces radiated space, increasing bass. This is not to say a speaker should be placed here nor there, however the designer has taken placement into account such that a correct tonal balance is achieved when recommendations on placement are followed. Generally , there are benefits in placement away from large objects that can smear the image. I.e, put the speaker away from walls or recess it into it.

A medium and an actuator

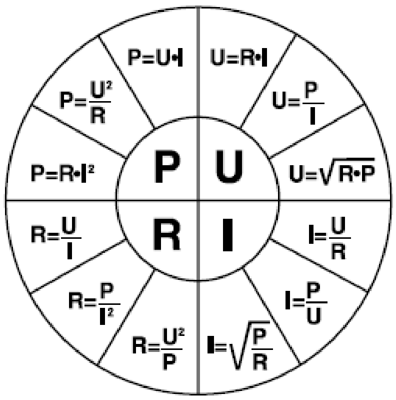
In communication science the term "transmission line" is a medium that can transport energy from one point to another and theoretically do so without a loss.

The practical implementation in the world of optics and electrics include the coaxial cable, an optical fiber, waveguides, striplines and electrical power lines.

The underlying principle is based on the physics suggesting that for this optimal (lowest loss) energy transfer to occur, the source and termination impedances must be equal.

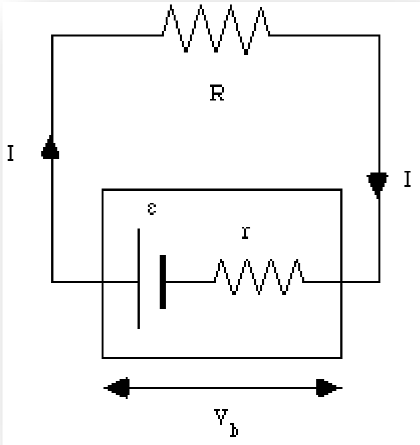
The key word here is "impedance" as this now indicates involvement of alternating currents and voltages and that there is a frequency involved. This in turn translates to the involvement of wavelengths.

Early on in this paper we established that "impedance" is the keyword so lets spend some more time to get acquainted with it. In electrics, semantically, impedance can be described as an "AC resistance". However impedance-like relationships are everywhere in nature so its not by any means an electrically related model although most often found here. Having an intuitive sense for impedance helps model and understand a great many scientific occurrences. Describing impedance can be a lengthy exercise involving real and imaginary dimensions (when current/voltage phase angle is not at a 90 degree relationship) and some even less exciting mathematics. Lets dodge this, and instead imagine impedance as a resistance at a specific frequency. We can get away with this for the time being, as in speaker design, it is only over the real resistive part that sound is produced, not the reactive.



P stands for power, U for voltage, R for resistance and I for current. This electrical model has many analogies in other disciplines giving an intuitive sense for physiscs.

To get a grip of resistance we can study OHM´s law and do this is conjunction with the example of a car battery. The battery is our source. We know these things can deliver a lot of current as they are used to turn an engine against friction and compression while providing current to the engine management system to get it running. But what rules how much current? This is governed by the battery's inner resistance, Ri (r in the picture). Imagine the battery as two components, a voltage source in series with Ri.



If you apply OHM´s law here where P is voltage multiplied by current, you will find that when the battery is loaded (R in the picture) by something with a resistance equaling the battery Ri, maximum power is produced. Note POWER, not current and not voltage, but the product of the two is at its peak when source and destination R`s are equal matches.

In a battery, where electrical energy is converted from chemical energy, this condition is finite. The Ri will rise as the chemicals neutralize, why the battery looses its charge. In the picture below the relationship of charge and discharge against time. Note the typical non-linear relationship which explain why so suddenly a car a battery can appear to go flat.

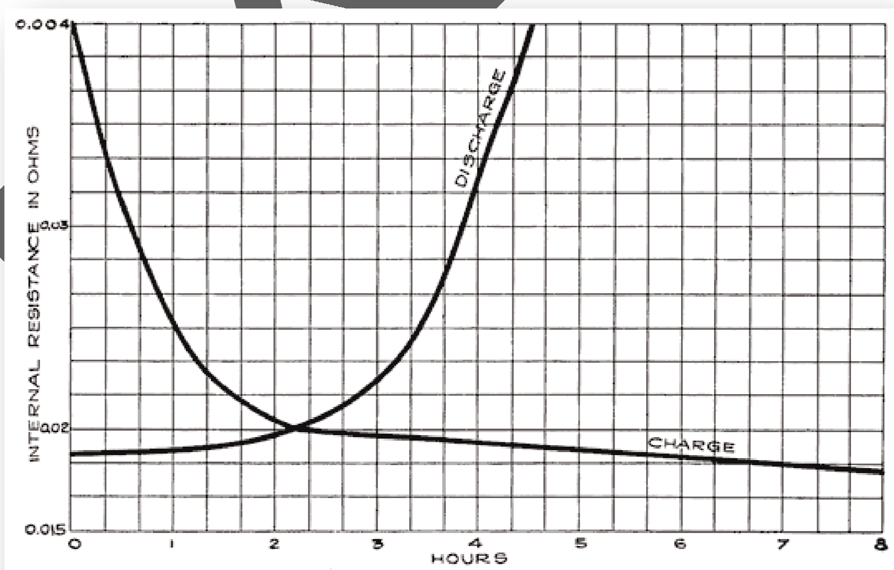


Fig. 26. Changes in Internal Resistance During Charge and Discharge

Between the source and destination there must be a *conduit* of some kind, normally this is realized by a cable (later we will look at another type of conduit, air). This too has a resistance and will also be a component of the transmission system. The cables resistance is a part of the entire chain of source and load resistances and will also dissipate some of the energy produced by the battery. But, in an ideal world, the conduit has no resistance (i.e no losses) which means all of the energy would be dissipated at the destination.

In a situation where we are dealing with not pure DC but alternating currents, things become a little bit more interesting. When an ideal match between source and destination exists maximum energy is absorbed at the destination. When a mismatch exists (the load is reactive) part of the energy is *reflected* back (like a mirror) to the source and thus becomes wasted. It also distorts the transmitted signal. The effects of this vary but in the case of conveying digital information, which btw can only be transmitted using an analog signal, poor matching can cause timing errors or even worse, incorrect interpretations of high and low levels. It is conceivable that the reception circuits for such signals may have limits in ability to cope with mismatches that can be exceeded and eventually become audible.

This begs for a moment of digressing. Why can the Hifi industry claim to produce brands of cables for conveying digital information that sound "better"? Well, only if the source and destination devices have some type of mismatch to begin with. Digital signals, or ones and zeros benefit of not having any quality aspects tied to them. If such effects occur it should instead be a clear cue for the assuming audiophile to return both items of equipment and let the service department establish which device does not conform to standards.

The impedance matching model is very useful in understanding day to day occurrences, particularly as much of what we do in practical implementation of technology is about attaining impedance matching. Many real life examples are all around us. For instance we know how well travels sound across open water. The reason for this is that the water is a good reflector due to the mismatch between sound source and the hard surface of the water. For the same reason, this is why a diver will hear very well the whine of a motorboats propeller while below the water surface, but not the engine itself as it is radiating into open space of air. With light, a really poor impedance match is normally referred to as a mirror. Whereas a good one is a perfect black body, an imaginary object so black it cannot reflect at all. Instead it absorbs all the energy falling upon it.

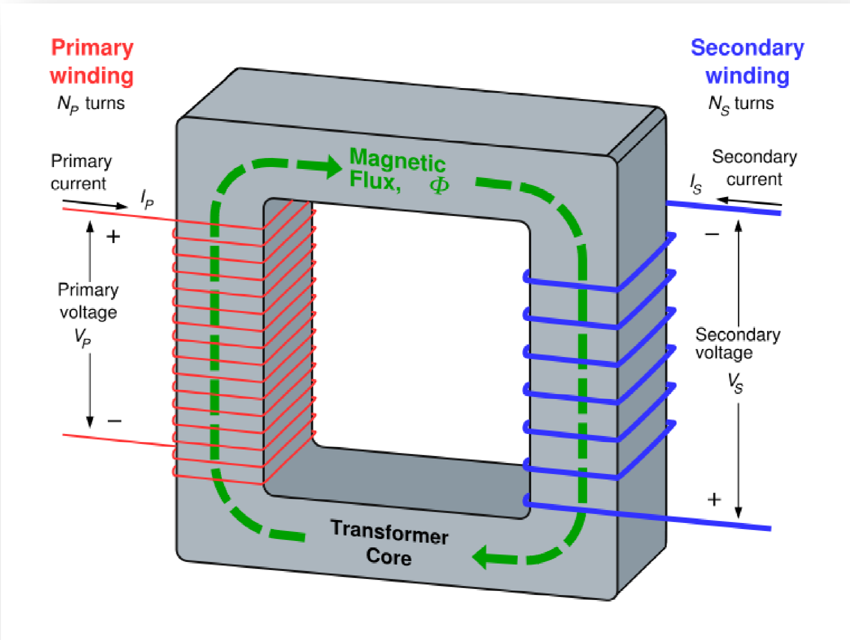
Getting down to it

So then, what about a speaker and its ability to convert electrical energy to mechanical energy (kinetic energy) and put this into the air so we can enjoy it?

Lets take an example by looking at a common 12inch woofer. Typically the moving parts may have a mass of around 40grams. What we are interested in is to find out the woofers source impedance so that we can discover how it matches the destination impedance. Don't be misled here, the impedance is not the one of the voicecoil but the one of the moving mechanical part the of the cone. The voicecoils impedance is the electrical interface to the power amplifier and not of primary interest here just now.

It may be a new concept and challenging to grasp initially, but anything being an actuator, that is producing movement from some other energy source, also has a source impedance and with it an inner resistance (Ri). To get the maximum power from this source we need to match the load so that energy is produced where we want it. For a loudspeaker driver we must know the source impedance in order to understand how efficiently it makes sound from an electrical signal.

Its useful to take a look at a common impedance matching device in the electrical domain, the transformer. See picture below.



The simple transformer converts electrical AC signals to electromagnetic and then back to electrical. It has two sides, the primary and secondary side. The relationship between the two sides is the ratio and represented by the number of windings in each coil. With a ratio of 10:1, 100 volts on the primary produces 10 volts on the secondary. The power must be constant, less losses. This means if we draw 10 ampere at the secondary 10 volt output, the primary must be capable of providing 1 amp of current with the 100volt input. Without a transformer the load would be totally mismatched to the source, our transformer solves the conundrum and the price of conversion losses is often a small cost to pay.

Back to our 12” loudspeaker driver, as we are dealing with impedance and it is only over the resistive part of the complex function where the (sound) energy is generated, we need to study this at a single frequency. Lets look at what happens at 440Hz, the flat A key. Working the calculator where cone mass and frequency are factored, the source impedance is approximately 100 ohms. Thus we know that 100ohms is the internal source impedance of what the air sees with the specific mass and area of cone at the given frequency.

So then, if the speaker cone is the source, what is the destination load? It´s the air the speaker moves. This is why the area of the cone is a significant part of the equation, to find the load one must calculate the mass of the air over that area and displacement the cone will make as it moves to produce sound, I.e a volume.

Cranking the calculator again turns up around 8 ohms of impedance that the air loads the cone. Remember that for maximum power to be produced the two impedances must be equal. Not exactly two equal numbers there. Loading a source of 100ohms by 8 ohms is more like a short circuit. A greater than 1:10 mismatch, why not much of the energy is to become audible. With a constant speaker diameter, this mismatch becomes greater as frequency falls. We got 8 ohms with a standard 12” woofer at 400Hz, imagine how poorly this match will be at 40Hz? This we know already empirically, small speaker drivers have trouble to produce enough sound pressure at lower frequencies.

This example of source and load mismatch is the mathematical evidence of the inefficiency (frequency dependent) of conventional loudspeakers where the air is driven directly by the loudspeaker cone.

The purpose of the horn is to act as an acoustical transformer. It´s job is to convert the speaker drivers high pressure and small amplitude at the horn throat, to low pressure and large amplitude at the horn mouth.

Typically, a conventional closed cavity or reflex type speakers capability to convert electrical energy to sound, is in the range of 0,5 to 1%. A horn, being and impedance converter, will achieve 15 to 40% depending on its range and other design parameters. The broader the range, the less efficient it becomes. Horns of the type discussed here (Earo designs) have a range of 3 to 3,5 octaves.

What happens next makes matters worse for the conventional loudspeaker system.

Worthwhile repeating, we found that the impedance mismatch between cone and air increases as frequency falls. Since the cone size is part of the equation, enlarging it improves matters a bit for the lower frequencies. But doing so does nothing good for the remaining part of the audio spectrum, a larger driver has trouble reproducing higher frequencies. The reason for this is that it becomes increasingly difficult to make mass move as frequency rises. Mass must increase with cone size.

It is worth noting that achieving a low mass of the cone is also an important factor in another sense. What is the best match to air? Well it would be air itself since mass is equal, intuitively the lighter the cones mass, the closer it becomes to the density of air. Thus from an impedance matching point of view a light cone is desired but for a conventional speaker design it must also have a large area to handle low frequencies. But that’s a contradiction as size and mass goes hand in hand. The ideal speaker driver has a large cone with no mass and no self-resonance. So, we have established that wehave two physical parameters that work opposed, how choose?

We know that sound pressure is a function of moving mass (air weights 1,2 kg per cubic meter) it means we need to move a volume of air per time unit, or liters and gallons of air per second.

However, we just identified that the state of the art mechanism using a directly to air interfaced vibrating membrane is rather inefficient. This is where designers of speakers part ways. Most take the path of lesser resistance (!) and allow the drive units to have a longer throw. I.e the cones become like pistons moving more air. An analogy with hydraulics and the physics in “communicating vessels” springs to mind, remember those from High School? Or, they make use of larger diameter drive units. Lately, the two appear to be combined as is evidenced in many car audio installations and home theater systems. Add yet another trick to increase the production of air per time unit, the resonator as used in bass-reflex designs. One has now introduced several strategies that have taken the design further away from its desired goal of efficient and accurate energy transfer.

At least its cheap…or… is it?

Whilst it may be seen from the argument above that longer throw appears to reach the same end result of moving an equal amount of volume of air per time unit, the argument does not take into account what distortion actually is. First of all, the definition of distortion is the introduction of a non-linear transfer function. A transfer function, graphically presented, is a straight line in an X-Y graph where input is on one axis and output on the other. Hence, the cone must be able to move in perfect accordance with the electrical signal fed to it. As longer throw is required in the design of such a driver, design challenges begin to emerge. First of all the magnetic flux gap, in where the coil travels back and forth, must be absolutely uniform and since it is so much longer it also means that very large/powerful magnets must be used. Since the flux is spread over a longer distance the magnetic efficiency is not necessarily increased. On the moving side, the voicecoil must be very large and capable of dealing with high currents. All of this introduces serious amounts of heat that must be ventilated in order to ensure reliability. To support all of this heavy duty copper accelerating a cone a long distance, the entire membrane, spider (the corrugated fabric attached to the bottom of the cone) and surround begin to take on rather heavy duty form. All of this increases mass, accelerating mass has a square relationship with velocity why the electrical power required increases in the same scale. Physics’ don’t dance along with the designer here.

Longer throw brings with it also another problem and its effect may vary depending on actual implementation but the underlying issue remains. This effect relates to the complex motion of the cone whilst reproducing both a lower frequency and a higher one. This may be visualized by imagining the higher frequency being modulated by a lower one. In principle a type of Doppler effect is created whereby the higher frequency now is swept up and down by the lower modulation. To in detail explain what happens and how it is perceived requires in depth study into psychoacoustics but the phenomenon is known commonly as intermodulation distortion. Thus it is the unwanted by-product of two or more signals (same effect is used to make radios work and is then called heterodyning). Because the amplitude is much higher, the cone must move a longer distance within same time period as a short throw system, the more exaggerated this type of distortion becomes.

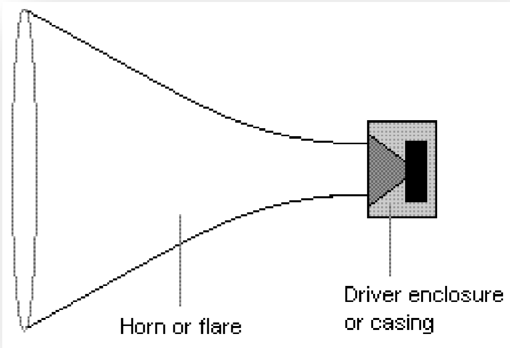
As is with compromises resting on poor foundation, new challenges are introduced along the path. With the longer throw and with increased mass the challenge becomes to make the membrane accelerate and decelerate as it attempts to follow the audio signal. Success in this endeavor is rare and can only be realized by brute force designs. The commercial availability of the newer type of high powered magnets (Neodymium) could help but instead the manufacturers take the cheaper way out and increase the size of the old ferrite magnets. Not within the scope of this document are the insights of the improved and controlled magnetic flux possible with these new types of magnets. With good magnetic circuit design and accuracy in engineering, the magnetic field may be highly focused around the coil assembly. Such drivers are identified not by their size and weight of magnets but how they don't attract paper clips, screws and keys or worse still, small magnetic debris entering the moving parts. Some high-end hifi attempts try to conceal the fundamentally flawed thinking by replacing permanent magnets with electromagnets. Electromagnets introduce new design challenges, the current must be very clean to not modulate the audio signal and the speakers begin to draw power and produce heat regardless if playing out or not. Not the greenest of solutions.

So what can be done?

Electrical engineers know that when impedances are not matched some device to achieve this is required or else the losses will be heavy. We saw how the transformer solves the problem. For instance, radio and antenna design is to a significant part an issue of impedance matching. The antenna designer attempts to convert electrical signals to electromagnetic ones so that they may use the Ether as medium. The source, a radio transmitter has a low impedance and the Ether has a higher one. Thus the antenna is an actuator matching the two impedances optimally. Some antennas are made to emit or receive energy only in a specific direction. Why waste energy when you can point it where it is supposed to go? Note that the best efficiency is reached when an antenna is matched to a single frequency…the broader bandwidth the antenna is designed for the less efficient it becomes. Sounds familiar?

Perhaps not so obvious is that the playback mechanics of the 78rpm record player is quite good even with todays standards. It was the recording itself that had the limitations. Its amazing that so much sound can be produced from a tiny stylus attached to the horn throat over a membrane. Very simple and very effective.

As we intuitively cup our hands around our mouth to direct speech we have created a simple impedance match. The old 78rpm gramophone managed to turn the tiny movement (but with very high acceleration) of the stylus into legible sound by connecting it to a small membrane at the throat of a funnel, such that music played at the other end, the mouth. It did so surprisingly well too. In fact, the "quality" is more related to the recording medium rather than the playback mechanics. One hundred years ago we understood the concept of impedance matching because we had to. We even understood why the rate (called flare) of the increase of area over the length of horn was significant. The horn is an acoustical transformer.



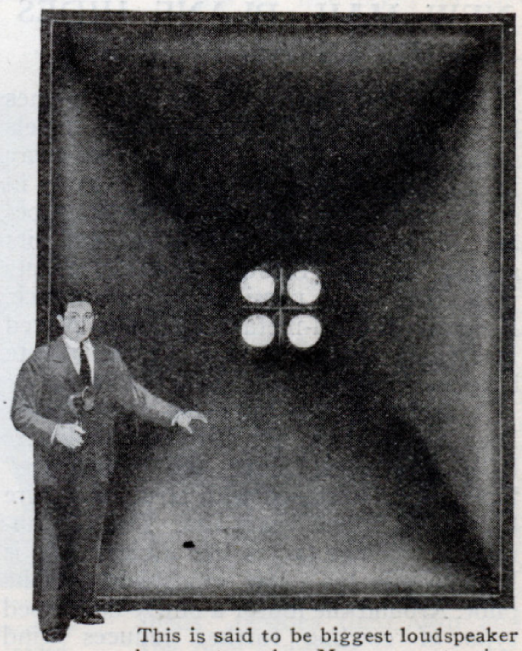
The horn transforms high pressure and low velocity movement from the speaker driver at the horn throat to low pressure and high velocity at the horn mouth by impedance matching.

Practical implementation of a theoretically correctly designed horn is challenging as its lowest frequency of reproduction requires a wavelength of said frequency in order for impedance of driver to air to be optimally matched. With the lowest note of a grand piano being at 27Hz, a technically uncompromised horn capable of this would have to be 12,7 meters (41,8 feet) in length, oops…

Even if we employ the method of making the ideal circular cross section of a horn into a square one and folding the horn inside a box, there is still no way 12 meters can be hidden conveniently in a spouse-friendly way. On the other hand, there is no point putting 27Hz into a room that is too small to reproduce this wavelength as room nodes will make the reproduction troublesome. Btw, nor can a grand piano do that. A musically acceptable compromise is in the 40Hz range. This brings us to less than 9meters. Still this is too long and the challenge is to reduce the size to acceptable dimensions. But this is not all, also growing with lower frequency is the area of the mouth, a 27Hz horn has the mouth size of a dual car garage door. Not a healthy place for either car or man. However, for the horn designer, there are many tools in the chest on how reduction in size may be accomplished, the results of which varies with the skills of the craftsman. Regardless, it is quite difficult to achieve acceptable results if the sizes are to be compared with conventional speakers.

LARGEST LOUDSPEAKER HORN FOR AUDITORIUM

DESIGNED for use in auditoriums, the biggest loudspeaker horn yet made has recently appeared on the market. Its twelve-foot opening gives it the appearance of the entrance to a tunnel into which an automobile could be driven. The claim is made for it that it will reënforce notes down to twenty-five vibrations a second and project it with no appreciable loss of tone quality, to the farthest corners of a large concert hall.



This is said to be biggest loudspeaker horn set made. Man suggests size.

Already in the 30´s most of the fundamental research was made into horns. With very low power availability from the amplifiers, efficiency in the loudspeaker was paramount. This art has died off with advent of higher power amps and commoditization of audio reproduction technology. The cost of this evolution being apparent loss of some opportunities for significant fidelity

We have learned that the efficiency of a loudspeaker can be increased by use of impedance matching. But why does this matter when power amplifiers are cheap and no matter how loud you play it cannot be noticed on your utility bill? It´s what it brings with it that matters. To understand better we need to, again, take a closer look at the driver unit.

To have control of how the cone moves, that is, how accurately it will follow the electrical signal, is a matter of the compliance (the spring) of the materials that support the membrane, it´s mass and the motor doing the work. Also the surrounding air influences the results. This motor is the combined effect of magnetic flux density and the properties of the coil and how much of it remains within the flux of the magnet during normal operation. Fundamentally, there is nothing complicated going on here. As the coil receives an electrical signal it attempts to move in the magnetic field until the electromagnetic force is balanced by the mass and mechanical spring. The mass and the spring will resist this motion but eventually the membrane begins to travel. Then, without forewarning, the electrical signal ceases driving the membrane in this direction. The membrane on the other hand follows Newton's first law and wants to continue along its established path. The mass has had energy put into to it and has inertia, propelling it forward releasing that energy.

It follows from this that it is desirable to have low mass and a strong motor. That combination is crucial. It is also easy to imagine that the coil in the motor must always perform inside the magnetic field. The reason for this is that together with the properties of the actual amplifier used, a tighter electrical command is achieved if the magnetic assembly always has the coil within reach. A coil traveling outside the constraints of the magnetic flux is like a dolphin skipping out of water, not much control. This is why manufacturers produce operating limits in the product specifications.

Anything coming in between the coil and amplifier electrically is from this standpoint also a disadvantage. Yet, most speakers on the market have more or less complex arrays of electrical components that precisely do so, the filters. These passive filters are intended to direct a range of frequencies to two or more driver units each covering a section of the frequency response. This all sounds ok on paper, particularly the glossy kind you get in the Hifi shop. But, and a really big but, due to the reactive nature of these drive units, components are added to compensate for the ensuing loss of electrical matching. All in all, the filters have a non trivial impact on the end result. An entire chapter could be dedicated on this issue but the salient point is that these filters introduce severe non-ideal transfer function properties in the time domain. To begin with it is more or less impossible to realize passive filters that are ideal, its difficult enough in the active small signal pre-power amp domain as is where the designer has a great deal of control of relevant parameters. In the passive, speaker level domain, the filters are at the will of the reactive effects of the connected speaker drivers. The greatest contributor is the low frequency unit, particularly ones with great mass and strong electromagnetic assemblies. Such units will under short periods become generators that oppose what the power amp is instructing the driver to do, with reactive response.

So then, attempts to separate frequency response to two or more drive units and correct level differences over the spectrum in between said units in a speaker assembly, introduce detrimental effects in the time domain, an area where Earo is paying particular attention. For these reasons, it is truly non-scientific to argue for esoteric loudspeaker cables by a manufacturer that realizes its speaker systems using passive crossover filter. As band-aid, bi wiring may help to reduce some of the reactive effects of the low end affecting the upper end. At the end of the day, a passive crossover is not a first choice in the quest for accurate sound reproduction. With such speakers, changing cables will surely present an audible change, the question is then, what of what you hear is correct, any of it?

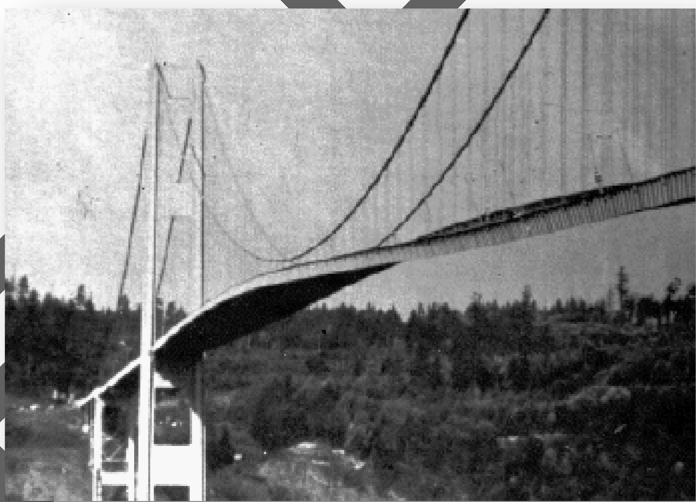
So as an engineer set out with ungrateful the task of designing a passive filter, the last thing you wish to do is load it with a loudspeaker driver. Still, if you insist doing this, you will end up with a filter that has a crossover point and phase response that moves around with the music played over it.

Fulfilling the discussed criteria for a driver unit makes it suitable for horn implementation. In the horn, the membrane moves only fractions of the magnitudes found in conventional speakers because there is a conversion from high pressure and low velocity to low pressure and high velocity.

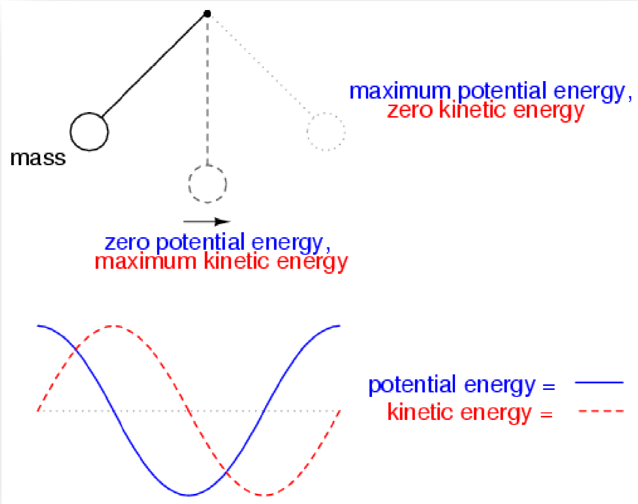
It should by now be intuitive for the reader to understand how this accounts for the detail and extraordinary transient response achieved. Following close in the horn design is the significant reduction of distortion, the key source of listening fatigue. But also of the fidelity that are attributed to horn designs.

Resonance

All real life objects has its own fundamental point of resonance, a speaker driver unit is no exception. Resonance is related to mass and some type of stiffness (spring) providing a reaction. The greater the mass, the lower the resonant frequency becomes. By the same token, the stiffer the spring the higher this frequency becomes. In a string instrument, the more the string is taught, the higher the pitch.



Not understanding and thus taking into account the existence of natural resonance has sometimes quite spectacular effects. Here, the bridge is brought to resonate by wind, eventually destroying itself.



A swinging pendulum translates potential energy into kinetic energy over its cycle. The swinging action can be projected into a sine curve revealing the 90 degree phase-shift in between the two states. This relationship is everywhere in nature and in science, understanding it, is very useful. The picture shows a simple harmonic motion, the basic oscillation.

We saw from above that it is desirable to have a lowest possible mass in the moving parts of the driver for the horn. But is this what we want for a conventional (reflex or sealed enclosure) loudspeaker?

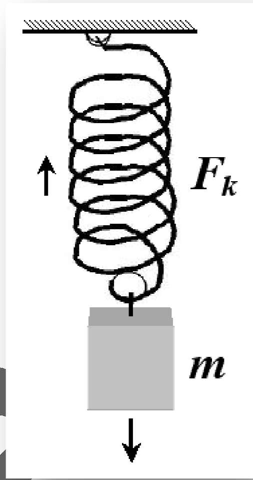
No, on the contrary. Lets take a look at the criteria the designer of conventional loudspeaker has to address. Since impedance matching (i.e energy transfer) was suboptimal here, we know we need to have a long throw to produce decent sound pressure and we learned that the larger the area of the cone the less poor the impedance matching becomes. But, this introduces the direct side effect of reducing reproduction of higher frequencies (due to increased mass). A long throw implies a lot of windings in the coil and with heavy gauge as the resistance must not rise above design goal so that the power amplifiers capability to drive the membrane is not reduced. Fulfilling this with the larger coil in the magnetic field becomes a challenge, both in size and cost but most in that the limits of linearity in the moving mechanical parts begin to show up. In order to not hinder the membrane acceleration, the compliance calls for a soft spring. This on the other hand is risky as the only thing controlling the cone when it loses its control due to it exiting from the magnetic field, is the damping of the air in the enclosure. Unfortunately this air does not offer a linear damping. Increasing the pressure in the enclosure does not require the same cone movement (energy) as reducing the pressure. The effect can be reduced by increasing the volume of the enclosure…however, then the damping of the membrane falls.

Again, non-linearity is another word for distortion.

Why non-linear? Air in an enclosure is a bit like stacking mattresses on top of one another. The one at the bottom will feel the combined mass of the ones above, the second one from bottom will feel the same, less one and so on. So, every time a mattress is put on the pile the pressure increases nonlinearly. Same effect is found on a really grand scale is how the mass of air acts on the earth, highest pressure at sea level and non-linearly decreasing with altitude.

The horn, on the other hand, never compresses air (of the type discussed here), it only moves it. In fact compressing air in a horn is a big no-no and something the designer is watchful over. Thus, the air in the horn exhibits a near resistive load to the speaker membrane and correctly designed, this load is linear. Since the driver is impedance matched it sees no reactive effects, reducing distortion.

This paragraph is about resonance so lets talk about this. When a driver reaches its point of mechanical resonance, its electrical impedance rises sharply. The result is that the power amplifier driving it is now increasingly mismatched to the speaker and ceases to transfer significant energy (current goes down due rise in AC resistance). The cone moves but is not under very good control.



You may have seen this before, even held a weight and spring in your hand and tried it. Imagine this again, you are holding the spring and very slowly lifting it up. The spring extends until it is stretched enough to lift the weight which then follows your slow movements. Now move your hand a little faster up and down. What happens is that the mass follows, but a little later than if the spring instead was string. At some point in increasing the speed of movement, the mass appears to move with ease and rapidly exactly together with your hand. You are now at the point of resonance. The stimuli is in phase with the motion. Increasing the speed of movement further and it appears as if the mass does not follow much at all.

Moreover several things occur here, below the resonance the impedance is such that the response falls with 12dB per octave making it very difficult to get any sound pressure below this magic threshold. There are two mechanisms responsible for this but we will not go into this now. The result is that the speaker has reached its lowest point of reproduction. Since the driver unit for the conventional speaker required a soft compliance (the spring) the only parameter that the designer of conventional speakers systems has available to lower the resonance is to increase mass. Now, how good was mass again for impulse response and for matching to air?

Remember, impedance mismatch causes reactivity to rise and this causes distortion.

Increased mass can to a certain extent be compensated for by brute force. Larger coils and magnets in combination with larger power amplifiers can appear to do a job to cover up some of the discrepancies in creating desired sound pressure. This has been the trend in the Hifi community for some time now. However, this strategy does nothing good for the time domain issues discussed elsewhere in this paper. Nor does it make things less costly to produce.

Resonance is a physical occurrence; it is there and is neither good nor bad. In the (bass-) reflex speaker, another type of resonance is used to help reduce the effects of the sharp 12dB/octave speaker roll-off at the point of driver self resonance. The application of tuned resonators does not lower the speakers ability to reproduce sound, it only provides a means to enhance what is there, to amplify it. It does however have a flip side.

The implementation of bass-reflex involves a tuned cavity. The air inside the cavity is brought to resonance at its fundamental frequency as the speaker driver excites the air. This air has mass, the more mass the larger the cavity (i.e larger speaker). The air performs in the same way to Newton's first law, first it refuses to follow and when it does, it refuses to stop. Thus transients have no means to survive here. The little that passes through are delayed…

Several things happen here. By design most bass reflex speakers have a pronounced hump just before the frequency response falls sharply, the combined result is a sensation of rich bass. The second effect is that it that hangs around. Resonance is the killer of transients, thus picking out detail is in this soup is challenging. Luckily, only very few manufactures has taken this to the worst of heights, the double resonant cavity. Here the resonant point was broadened by use of several resonators giving the false impression of lots of deep bass from small enclosures.

We recognize these speaker systems in designer homes by their small satellite drivers and often well hidden bass unit with embedded amplifier. The commercial success of this principle is defies logic. Distinguishing an electric bass from an acoustic one is hard, it gets so bad that even then bass drum joins the soup of low frequencies.

Worse, the resonant cavity will continue to resonate at every multiple (harmonic) of the fundamental but at gradually falling levels. Thus the frequency response will demonstrate this too. Reflex speakers account for quite heavy coloring several octaves above the speakers reflex resonance.

The horn is fundamentally different; it does not make use of resonant techniques at all. On the contrary, a horn must not have any resonances whatsoever.

Compliance

We have elsewhere stated that the compliance of the cone in a driver for a horn often is stiffer than for closed or reflex speakers and that this is because the horn moves air, not pushing it around inside a box. As such it does not spring back why it needs to have this built into the driver assembly. Musical signals are not always symmetrical, i.e they are not a steady tone but one or a burst of transients.

Transients, the spikes of musical information, are asymmetrical and when the speaker is stimulated by such a spike there is less electrical signal that brings the cone back to resting position other than the compliance of the drivers assembly (plus part of the electrical amp-coil circuit). In the sealed and bass reflex enclosure, the compression of the air is partly responsible for this return mechanism and as stated before it is not a linear spring. Transients in music are highly informative, contrary to a steady tone they contain masses of information. Information that may be lost if the speaker fails to reproduce these impulse properly. In fact, it is arguably the single most important parameter of a speaker system, the temporal response.

Importance of voltage in the power amp

The voice-coil moving in the magnetic field is essentially a linear motor. But only linear when it remains fully inside the magnetic field. The coil is in fact also a coil in the electrical sense and exhibits inductance. Inductance is like mass but in the electrical domain. To move this mass rapidly requires a current to flow through the coil. Only a high voltage (pressure) can overcome this situation, the higher the voltage the faster it may happen. You can see that the amplifiers ability to not get intimidated by the coil inductance is related to its ability to at these point inside the cycle produce large voltage swings and have also a low internal resistance. Here a low DC resistance part in the coil is influential. So even if you don't play very loud and use little average power, power headroom is very important and you get this from a low source impedance from the power amplifiers side. A low source impedance in the power amp will be seen from the voicecoil as a short circuit to the back EMF (ElectroMotiveForce) produced by the voicecoil, the membrane mass and spider spring compliance. It acts as an effective damping reducing the motions not produced by the electrical signal (mass and spring). Normally, the more power an amp can produce, the lower is its source impedance.

Thus if everything is equal, if possible, using a smaller coil is an advantage. The recipe is a short throw, low mass membrane structure driven by a coil with low inductance and DC resistance in combination with strong magnetic flux density in a narrow gap. This makes the drive unit capable in following the varying electrical signals well allowing for a more true audio reproduction. Drivers for horns are quite different designs than those used in closed and reflex type boxes.

Slewing rate

This term is well known by designers of analog amplifiers. It relates to the ability to rapidly change voltage of a signal. In a sinusoidal signal, the rate of change of a voltage is not constant over time in a single frequency cycle. The largest acceleration of the voltage is around where the signal changes polarity, the zero crossings. The higher the frequency the greater this rate of change (slewing rate) becomes. As discussed early in this paper, this requirement does not disappear after the electrical signal becomes a mechanical motion. This parameter is obvious for the electronics designer but appears to be forgotten by the speaker design community. Again, this ability for a speaker driver to achieve adequate slewing rate is a compound of mass, frequency, compliance, magnetic flux density and amount of coil windings inside said flux plus some more. As has been argued so far in this paper, only a driver for a horn or impedance converting enclosure, may these parameters be optimized as they can be made to work together, not against one another.

Materials

Considerable effort has been put into applying advances in lightweight materials to the moving assembly of the driver unit (mid and low freq. drivers). This ranges from exotic alloys using both aluminium, titanium, beryllium and magnesium. In the non-metallic category, good use has been made of Aramid (Kevlar) and carbon fibers with some even more challenging uses of ceramics. Recently the application of plastics, most often polypropylene has found its way in the business of making sound.

Oddly, what still prevails in drivers for horns is the use of paper. Why? Well paper “sounds” good or it sounds perhaps less as it offers best compromise between mass, damping and stiffness. Paper cones usually employ special fibers, both organic and synthetic with or without doping compounds.

By now you should not be surprised that there is a scientific reason for this as well. Since the mechanical action on the cone is from the center via the voicecoil, load on the cone is uneven looking from center to edge. If you have seen an arrow exiting the bow in slow motion you have noted the wobble as the energy is projected from the arrows back to its front, reflecting back causing a resonant behavior. Similar things go on here. The coil force introduces a traveling wave in the membrane whose speed is ideally slightly lower than that of the air (speed of sound in the paper is lower). It so happens that paper meets these criteria such that the delay from center to rim introduces a spherical wave front from the speaker driver. Actually, when the frequency rises and its wavelength becomes shorter than the distance it travels through the cone, it can no longer produce a spherical wave front but a planar one. This is detailed explanations of the origin of beaming or directivity. The use of the whizzer cone is now obvious, it acts as a smaller cone for higher frequencies making spherical wave fronts.

A distinctive quality feature of a driver membrane is found in the use of a flare in the cone as this introduces three dimensional stiffness. It may even assist in producing a spherical wave front. But there is more going on here. Since any cone must have mass, there is a side effect that part of the energy is not only used to move the cone in the intended direction but it also travels as a wave motion through the paper until it arrives at the edge surround. The surround thus has an important function not only to guide, center and seal the cone but ensure that the wave is not reflected back into the voicecoil. An impedance issue? Certainly! Cone breakup is the situation when the membrane is driven by a high Disc-jockey to its extremes and where this energy creates chaotic vibrations, which of course are audible. Together with cone actually hitting the bottom of the magnetic assembly to the audible loss of dynamic response, these are the signs of overdriving a speaker unit.

A perfect solution

So then, is the horn the perfect solution for a loudspeaker?

Not in itself, as a technically correct horn has no commercial market due to its physical size. Instead it is as always a matter of making good compromises and apply good engineering standards, commonly it helps to have a total grasp of what challenges lay ahead. This is not simply about building a cabinet and getting a glossy veneer on it, but understanding structural mechanics, physics, acoustics including the psycho- one, electronics plus manufacturing methods as well as having a sense of aesthetics. Throwing in a handful of passion and more so of experience is a good start. Whilst the horn as principle, in our opinion, has more going for it than many conventional realizations, it is also amongst the most complex of loudspeaker principles to design and implement. The underlying theoretical and empirical work is daunting.

Leaving out compression type horns for their lack of fidelity (compression of air is the same as introducing distortion but it can be used to increase efficiency in PA scenarios), there are two types of horns that the designer has to make a decision for at the conception of a product. Back loaded or front loaded?

In the frontloaded horn the drive unit is not visible, it is lodged inside the enclosure. Typically this horn design is used for very large sound pressure levels and seldom found in domestic applications. The frontloaded horn is often combined with several horns working their own range of octaves. You will find these designs on large stages.

The backloaded horn is at first glance deceitfully simple. A drive unit can be seen and this is all there appears to be. However, the horn throat is driven off the back of the drive unit. The horn mouth area is a several times larger than the driver, an open area facing either forward or backward. The driver used is a wideband unit capable of reproducing the entire frequency range. Mostly, the lower end of the frequency range is horn loaded, where it is needed. The remaining range, typically above 300 to 400 Hertz is radiated directly from the drive unit.

We will mainly stay with the later type but it should be mentioned that the frontloaded horn is the easier method to get control over the range of parameters that affect the outcome. But it follows that the end result will have to be a series of stacked bandpass horns introducing a tall order of challenges in home environment. In a PA application these challenges are no constraints and is why they dominate the stages on all types of musical reinforcement where large SPL´s are required.

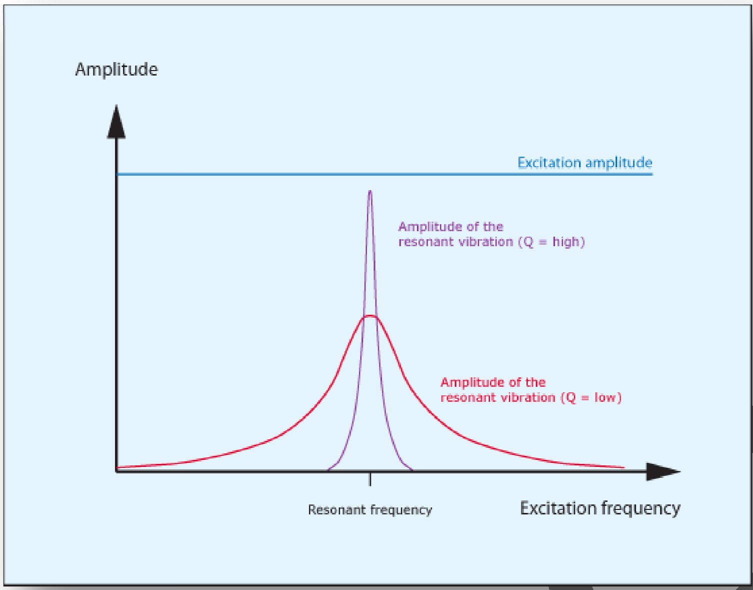
The backloaded horn is more elegant for home use. In fact it is a two way system where the back is the low end and fed to a horn section whereas the front radiates more or less directly to the air. As for choice of driver, it soon becomes apparent that anything greater than 200mm´s (8 inches) has no capability to produce the full frequency range and is the reason why most designs are centered around this max diameter. Particularly if the highest sound pressure is an important parameter, it isn't always so. The horn efficiency is a function of several parameters, one of them being its range, or bandwidth. Typically around three octaves is practical meaning that a 40Hz unit will begin to taper off at 320Hz. This is just fine as the 200mm driver begins to show its capability on the frontside of the cone. Efficiency over non-hornloaded designs is typically 15-25 times better. Wow, a green speaker?

It´s good to know what the flip side is before you set off, so lets begin to swallow the ugliest frog first, the sound from the back of the driver is in antiphase to the front side and must be delayed just enough so that it matches the driver at the mouth. This is done both mechanically in the design phase and in the Earo case additional processing in the digital domain is carried out. The horn is of course not full length but shortened, how much as well as rate of flare etc. is the designers secret. The effect of these altercations from the ideal is frequency dependent and when modeled, it shows up as a wavy difference in levels having a maximum at the point where the mouth and driver radiation crosses over. Thus, the horn is probably not at its full length, this would become 8,6 meters long for our 40Hz design, why the concatenation (reduction in length) also has a positive effect in lessening the time delay.

Other, superficially similar, enclosure principles are “transmission line” and the now so popular (why?) “dipole” speakers have a similar challenge w.r.t to antiphase radiation but none of the benefits of impedance conversion.

From a sensory point of view the two radiating surfaces, drive unit and horn mouth, will appear as one point source as our ability to locate sounds at 300Hz and below is poor. So we get away with that in the same manner as all other enclosure principles does, the low end placement does not have to be directly co-located with the remaining radiators.

Once over this step the designer has to deal with the aspects arising from brutally shortening the horn to one fourth (typically) of the ideal length (to a quarter wave) and folding it such that it may be implemented into a practical product. During this process it is most common to depart from the ideal circular cross section to a square one simply because the practical implementation and aesthetics demand it, who wants a Godzilla sized version of the 78RPM gramophone in the living room? This introduces new challenges such as the potential for unwanted resonances, efficiency losses and mismatches of various sorts. All of this can be modeled mathematically in simulation applications and therefore also dealt with properly already at design stage.



Q is a dimensionless factor in resonance where a high Q means the peak is high and bandwidth small. Even though incorrect use, Q is used for instance to describe the type of roll-off in a reflex speaker. Often the designer chooses the Q such that a pronounced gain occurs just before response rolls of. This accounts for the false impression of low bass found in such designs

One matter that does work well in favor for the horn is that the driver resonance frequency is not the end of the world as it is when choosing for a reflex or closed enclosure. Instead, we want the horn to not stop putting a load on the driver until well below its free air resonance. A loading (resistive and linear) of the air in the horn to the speaker driver thus lowers the free air resonance point and also broadens the peak. By doing so the large cone excursions occurring in resonance are damped by the horn, if the horns design limit is below the one of the driver. Using an automotive metaphor, even if the engine is off, the brakes work. This not only saves the driver from damage but also reduces its distortion in the frequency domain. In addition, the horn loading at or below resonance tends to lower the Q value of the drive unit, the resonant peak, making it flatter. The “Q” value is a dimensionless parameter that is the relationship between amplitude and bandwidth. The higher the Q the more peaked the response. At the point of resonance the Q typically rises sharply. In fact, this is how resonance is defined. However, the greatest benefit is that horn loading lets us choose driver parameters that are consistent with producing a full range driver rather than the opposite as those forced to be considered when designing closed or reflex enclosures.

Less obvious is the choice of function for the horn flare. Typically, exponential flare is common while geometric, parabolic, hyperbolic and linear flares exist together with the latest cry, tractrix. Each has its merits and need to be chosen for what they achieve in terms of impedance transformation against the plethora of criteria the designer has to take into consideration.

Since the front of the driver in a backloaded horn is terminated into plain air, the designer does not have the tool available to control the driver as if designing a frontloaded horn where the driver only radiates into a horn and the back being enclosed in a volume of air. This in fact makes the backloaded horn more challenging to design well. Therefore, focus has to be put on the design of the horn section behind the drive unit and to do so in conjunction with design opportunities in the analog and digital signal domain. Herein lies many challenges, but at the end of the day, when an inventory is taken over just which compromises have been made and which have not, that the true potential of this technology becomes evident.

A technically correct horn

By now this should sound like a strange proposition, it is. A technically correct full range horn not only becomes excruciatingly large en beyond spouse friendly, its questionable if the sonic benefit is warranted in the first place. Why? Well a technically correct horn should thus ideally be full-range and not a series of stacked horns with band-pass characteristics. However, technical correctness makes this horn then less efficient requiring larger than necessary cone excursions raising the distortion, thus by design removing one of the most important features of horns, their purity. So it goes on, until nobody is left without the realization that we live in a world where a good product is a product with the right compromises, this is to us a technically optimized design.

Summary of back-loaded horn benefits;

- Highly efficient, requiring relatively small power amps = less amp distortion
- Very low acoustic distortion due to impedance transformation - clearly audible
- High transient accuracy as result of small cone excursions - clearly audible
- Phase accurate - single driver = no crossovers – clearly audible
- No resonant principles used = detailed sound – clearly audible
- Very good dynamic response = large SPL without compression – clearly audible
- Generally narrow dispersion (directive) reducing room influences – clearly audible

/end/

<u>References;</u>

Art of sound reproduction – J.Watkinson

Acoustical Engineering - Harry F. Olson

Horn Loudspeaker Design , 1974, J.Dinsdale