The Geddes Loudspeaker System Design Philosophy

The following white paper outlines in broad strokes the design rational for Earl Geddes line of loudspeaker systems. By most accounts these loudspeakers are among the best loudspeakers in the world – at any price. That is quite a claim – the following white paper explains why this is believed to be true. (Personal accolades can be seen in the paper on the web site.)

The first loudspeaker in Geddes line of loudspeakers is the **Summa**. This loudspeaker represents a culmination of nearly five decades of loudspeaker design experience, in-depth technical knowledge and a significant amount of research, both public and proprietary, into both the physical design of the systems as well as the perceptual design. It should be emphasized that Geddes' approach is a *systems* one where he includes the full bandwidth of audible sound, the room, and the listener themselves. This is quite a unique departure from the usual procedure of just buying a pair of loudspeakers, putting them in an arbitrary room and hoping for the best. These speaker designs incorporate many new technologies and insights and each will be briefly described here. First, in order to understand why Geddes designs have been so well received it is important to understand the fundamentals of how we perceive sound, particularly in small rooms where Geddes designs excel. Only by understanding what is important in the perception of sound can one truly understand what makes these designs sound so good.

Background and Problems

Most audio reproduction loudspeaker systems are in small rooms (size less than a small auditorium). Dr. Geddes has spent his life optimizing sound reproduction for this kind of room – a typical home listening room. The problems for audio playback in small rooms are substantial – looking at it scientifically, it really isn't a very good place to do critical listening, but, unfortunately, it is almost always what we are stuck with! Small rooms have acoustic problems that are unique, differing substantially from the acoustics problems in larger rooms. The acoustics of large rooms have been studied extensively and are well understood and the solutions to its problems are very well known. But unfortunately a great deal of this knowledge is not applicable to small rooms and its unique set of problems. The small room problems have not had nearly the same level of attention applied to them as the large room. Practical solutions to a small room's acoustical problems are only now starting to evolve.

There are two fundamental concepts that need to be understood in order to develop a sound system that is optimized for a small room. The first is how the sound field in the small room is established and the second is how the human hearing system perceives this sound field. The first is an acoustics problem and the second is a psycho-acoustics one. It takes a considerable knowledge of both these fields to make a speaker system that will perform well in a small space.

The acoustics of small rooms – Low Frequencies

The acoustics problem in a small room has two major facets to it, the low frequency region and the high frequency region. These are excited and perceived quite differently. The transition between the two is room size and damping dependent, but for a typical home listening room this transition is between 100 and 200 Hz. In a large room the entire audio frequency range of sound behaves basically the same, but in small rooms, the low frequencies have a very distinct modal behavior and must be dealt with separately from the higher frequencies. Basically, above about 200 Hz there are so many modes - even in a small room - that all rooms behave alike in these frequency regions and modal effects are not relevant. In essence, once a room is larger than a few wavelengths its sound field starts to behave like every other room as far as its modes and wave propagation are concerned. Basically, modes are unimportant at the higher frequencies because there are so many of them.

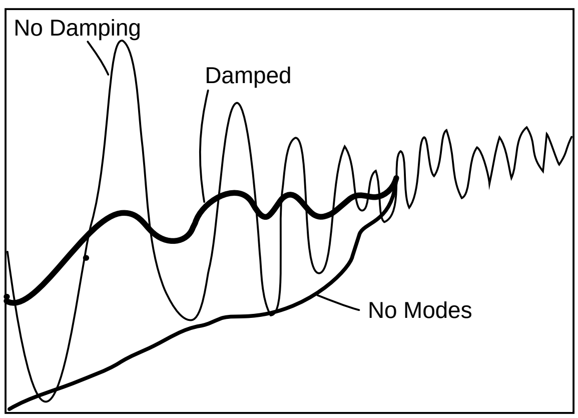

Figure (1) Hypothetical room responses for three different acoustical situations.

On the other hand, consider the low frequency region in a small room. If this room has a small amount of damping it will have a very discrete modal structure with an associated resonant modal sound character. No single source placed in this room would ever sound correct – not dipoles, monopoles, low frequency horns. In essence, the room dominates the situation over the source characteristics making all bass sounds in this region colored and not lifelike. Each mode accentuates the sound level when it is excited and diminishes the level where there are no modes. The sound is either “boomy” or “dead” with no happy medium in between.

A common, but naïve, point of view is that minimizing the excitation of the modes will minimize this modal characteristic. While this may be true to some extent, it is none the less a sub-optimal approach and realistically it is not even desirable. Consider figure (1) – this figure shows three hypothetical room responses in a typical home listening room. One curve shows the response with no room damping – clearly an auditory disaster. The lowest curve is what would happen if somehow the source excited none of the low frequency modes. This lower curve is smooth, but it is also not physically realizable. While it is possible to excite “fewer” modes, it is not possible to avoid exciting any modes. Hence the smoothness of this “No Modes” curve is an illusion which is not obtainable in any real room. What one usually obtains when one tries to avoid "exciting the modes" is a greater gap between the excited modes and an even greater frequency response irregularity. "Exciting fewer modes" is not the answer.

The middle curve is what is recommended. By dampening the low frequency modes a smooth response with reasonably good energy gain is obtained. Studies published by Dr. Geddes have shown that dipoles do not produce any smoother response in well damped rooms than another other type of source, but, on the negative side dipoles can only produce a substantially lower energy output when compared to a monopole at these lower frequencies. The dipole requires a great deal more EQ and amplifier power - neither of which is desirable in the most cost effective system and only ends up being basically the same response variations as the monopole. Why pay more than you have to for anything when the end results are virtually the same?

It is important to note that the overall response of the heavily damped room basically does not require any narrow band EQ, although there could be some small amount of EQ that is desirable. Generally, however, a highly damped room has a low frequency energy falloff that is an unavoidable result of there being less and less modes to carry the energy with a substantial amount of absorption to take it away. Modes are what make a room a room as opposed to a free field and like it or not they are not a bad thing. So while the low frequency response can certainly be smoothed out with damping, there is usually a practical limit to how much LF absorption one can obtain. LF absorption can be very difficult to achieve. But when there is a great deal of LF loss then one must accept the fact that the energy or power response will diminish at low frequencies. This factor can be accounted for in the speaker design.

How one achieves good low frequency damping in a room is a topic unto itself and one addressed by Dr. Geddes in his book *Premium Home Theater*. The interested reader is referred there for more detail on this critical topic. Suffice it to say, room design is not an inconsequential part of any good audio system and achieving the optimum sound quality in any room requires substantial attention to its characteristics. Good loudspeakers are always good loudspeakers, but the room can add or subtract from the speakers attributes a great deal.

One thing is certain from this discussion; without some means of dealing with the low frequency modes in a small room, no speaker will sound optimum at these frequencies. You will hear the room, not the speakers. The source characteristics at these low frequencies is not a critical issue since the room virtually always dominates the situation. The important factor is the sources ability to produce sufficient energy at lower and lower frequencies. This places a substantial burden on the low frequency driver, especially at high Sound Pressure Levels (SPL) like those required in a Home Theater.

The Need for Multiple Subs

Is the low frequency playback in a small room doomed to poor sound? – no, not at all, but it does require a completely new concept and a unique approach that is quite contrary to previous thinking about low frequencies in rooms.

As noted before, the modal situation in a room is a matter of large peaks and dips that basically dominate the audio landscape against which the speakers, as sources, are placed. This landscape is different for every position in the room and for every position of the sub and its entirely different for every room. One cannot approach a problem like this as if there is a singular solution, a one size fits all so to speak. To make a long story short, we must look to statistics to solve this problem. We must look for ways that will reduce the variability of the peaks and dips in both the frequency response as well as the response around the room. The reason for this is that if the response is different at every point then EQing the sound at one point will only make it worse at another point – there is no EQ that can correct the situation except for a single point. On the other hand if we could reduce the variability of this response from point to point then we could apply some EQ and correct the entire problem. Statistically this comes about by using averages – by using multiple LF sources that will average out to a smoother spatial variation which can then be EQ'd for a total global solution. One can obsess about the best locations and number of sources, and papers have been written on this subject, but it all comes down to a simple approach. Use multiple subs placed at fairly random locations in the room – in other words place them wherever you want, just so long as they are not close to one another. It has been shown that this will smooth out the spatial variation of these sources thus allowing for a global EQ'd solution.

The actual implementation of this is in general not trivial, because each sub has gain and phase and low pass filter points that can be set and it only makes sense to set this in a manner that optimizes the frequency response smoothness. In fact, it has been found that, in general, three subs operating in unison with a pair of Summas will yield an acceptably flat frequency response throughout the listening room without the use of any EQ at all. This can, however, only be done with a judicious choice in the parameters of the subs which requires some measurements of the real situation in the actual room.

The details of this procedure are beyond the scope of this article, but are available elsewhere. GedLee also offers this service to those who buy their subs.

The bottom line is that at low frequencies one is force to smooth out the modal response by using several sources placed throughout the room and optimized to work together as a system. In this way the rather disastrous LF situation found in most rooms can be tamed resulting in a very pleasing sound quality in this most problematic area.

The acoustics of small rooms – Mid-High Frequencies

At mid and high frequencies, there are several factors that must be considered in the loudspeaker design that will present some formidable problems. At these frequencies there are enough modes that the sound field and wave motion basically acts like a ray or beam of sound moving from the source and reflecting off of walls etc. – this is called the geometric or ray acoustics region. It is well know that reflections can be good and bad and it is critical that a loudspeaker designer understand the difference. One must look to psychoacoustics to determine what is required of the loudspeaker.

There are a couple of characteristics of the ear that need to be understood in order to understand what is important in small room acoustics at mid to high frequencies. These are mostly related to how the human ear perceives reflections and diffraction. The perception of reflections and diffraction is a highly complex topic, but there are a few principles that are most important here:

The earlier and the greater in level the first room reflections are, the worse they are. This aspect of sound perception is controversial. Some believe that all reflections are good because they increase the listeners feeling of space – they increase the spaciousness of the sound. While it is certainly true that all reflections add to spaciousness, the very early ones (< 10 ms.) do so at the sake of imaging and coloration. There is no contention that reflections > 20 ms are positive and perceived as early reverberation and acoustic spaciousness within the space. In small rooms, the first reflections from an arbitrary source, mainly omni-directional, will never occur later than 10-20 ms (basically this is the definition of a small room), hence the first reflections in small rooms must be thought of as a serious problem that causes coloration and image blurring. These reflections must be considered in the design and should be also be considered in the room as well.

Reflections become less of a problem as coloration and image shift at lower frequencies. Below about 500 Hz. early reflections are not as much of an issue. The ear has a longer integration time at lower frequencies and it has a poorer ability to localize resulting in a lower sensitivity to early reflections. Image localization is strongly weighted towards the higher frequencies.

A reflected signal that arrives at the opposite ear from the direct sound is less perceptible as coloration and image shift than if both signals arrive at the same ear. This is because of head shadowing above about 500 Hz and the fact that our ears can process signals between them. When the two signals arrive at the same ear, the signals are physically merged in space even before they enter the ear and no amount of auditory processing can separate them. When these signals arrive at different ears, the auditory processing system can diminish the adverse effects of these early reflections through cognitive processing between the ears.

Very early reflection-like signals can be quite audible due to non constant group delay. Diffraction acts much like reflection, but is usually frequency dependent – far more so than a typical reflection. Diffraction can occur from any nearby discontinuity be it on the cabinet or the room itself. These very early reflections/diffractions can have a pronounced audible effect far greater than what their frequency response effect would indicate. This is because of temporal masking effects in the ear. This audibility can also be level dependent making it a non-linear distortion-like effect. Diffraction cannot be electronically corrected except at a single point in space. Only acoustical corrections to diffraction (like not generating any) can have a global effect.

From an acoustics reproduction standpoint then, the loudspeaker system design must help to provide as much delay as possible in the early reflections and allow for speaker placement and orientation such that the earliest reflections occur at opposite ears rather than the same ear. This needs to be done above about 500 Hz. Below 500 Hz other factors, such as room characteristics and our hearing mechanism, may dictate an entirely different approach. The cabinet and nearby room geometry must be such as to minimize the generation of diffraction.

An obvious question always comes up – “Why not just make all the reflections and modes go away? Doesn’t this solve many of these problems?” That approach is (unfortunately) used in a great many situations, but it is far less than ideal. Without real room reverberation the perception of the playback is dead, lifeless, in acoustics parlance it lacks *spaciousness* or *ambiance* – the feeling of being engulfed in an acoustically spatial environment. If a non-echoic space were desirable then an anechoic chamber would be the ideal listening room, but as anyone who has ever listened to speakers in this kind of space will tell you, it really isn't a good listening environment. One is always aware that they are listening to speakers – the room adds nothing. It's something like listening to headphones, which admittedly some people like.

To achieve good *spaciousness* in a room requires a multiplicity of lateral reflections (vertical reflections don't really contribute much) arriving from many directions, i.e. a diffuse sound field. To get the feeling of spaciousness in a small room it must be live, which presents a problem. How does one minimize the early reflections, damp the low frequencies and still allow for a "multiplicity" of later reflections and reverberation? These seem to be completely contradictory requirements. However, with proper loudspeaker and room design and loudspeaker placement this can happen.

It appears then that the best rooms for serious listening would have a large amount of low frequency absorption accompanied by very low amounts of high frequency absorption. This is exactly the opposite of what is usually achieved in most rooms. Also consider the fact that absorbing material placed in a small room is orders of magnitude more effective than this same material placed in a larger room. This happens because virtually all sound absorption takes place at the enclosure boundaries and a sound wave in a small room strikes these boundaries an order of magnitude more times in a given period of time than it does in a large room. Thus, even small amounts of sound absorption in a small room can lead to an over-damped condition - especially at higher frequencies. This over-damped situation is a big problem in many small rooms.

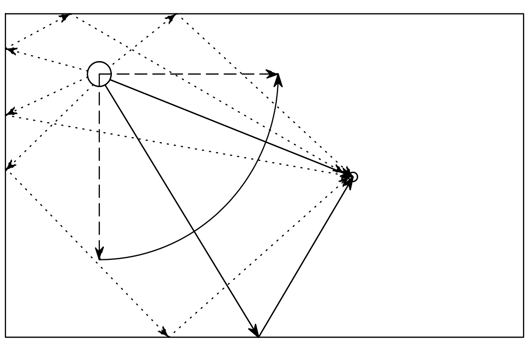

Figure (2) A room with an omni and directional source showing room reflections

Now consider the situation shown in Figure (2) at the left where an omni-directional source and a directional source (with a 90° coverage pattern) are placed at the same location in a typical small room. (Only a two dimensional example is considered here - in three dimensions the example would be even worse!). The omni-directional sound rays are shown as dotted lines while the solid lines represent the directional sources sound rays. It is easy to see that the higher directivity gives a far clearer - less cluttered – pattern of early reflections. Best of all, the directional source can be pointed in such a way that the first reflection actually arrives at the ear opposite the direct arrival. For the directional source, the secondary (after the first) reflection arrivals are virtually all lateral and behind the listener – a good thing. The omni source has several very early reflections arriving at the same ear and several more arriving at the opposite ear, all within the early 10-20 ms. time frame. Taking into account the secondary later reflections and the omni source can be seen to have a proliferation of early reflections when compared to the directional source. Dipoles help this situation, but are not as effective as a truly directional source.

Now comes the really crucial part. In order to perform this “trick” for optimizing the early reflections in a small room, two specific source characteristics are required. First, the source directivity must be less than about 90° and second, the listener is not actually on the sources axis! In other words the direct sound, the first arrival sound, is not the axial sound. To achieve a flat response at the listener in this configuration, the loudspeaker must have a flat frequency response off axis. This is virtually never the case with most loudspeakers. Most loudspeakers with a smooth flat axial response will usually not work very well in this configuration. This type of characteristic in a loudspeaker is call Constant or Controlled Directivity - CD. Constant in that the directivity of the source does not change with frequency or stated another way, the source has a basically flat response at angles away from the axis.

CD is a well know design criteria for large venue systems, but it is almost non-existent in home high-fidelity and Home Theater loudspeaker systems. There are two reasons for this; the use of cone speakers and physical size. Piston sources (cone or dome loudspeakers) can never be CD and CD simply cannot be done in a small radiating area. It takes space and area to control sound radiation - there is simply no way around this fact of physics. Hence, for sound systems in small rooms, bigger really is better. Virtually all small speakers are omni-directional. Somehow it just seems that the larger speakers of the past sounded better than the multitude of mini-cubes and mini-tower speakers of today. Small speakers do have their place, but not as sources for critical listening or for creating the higher SPLs desired in a Home Theater. They are easy and inexpensive to build, hence their attractiveness, but they are not ideal in a small room.

Summary of Requirements.

The requirements for the design are now clear. Above about 500 Hz, the speaker system must posses a narrow, 90° or less, coverage pattern which must be constant with angle and frequency. Stated in another way, the Directivity Index (DI) should be above 9 dB above about 500 Hz and should be flat with frequency. The system can have an increasing energy output capability (but not necessarily the SPL at a given point) at lower frequencies to produce the increasing energy demand of low frequencies without overloading. The polar pattern below 500 Hz is not as important as it is above so it makes sense to have the coverage control go down to about this frequency. In other words, the DI should fall to about 3 dB below 500 Hz. The sources must be oriented in the room in such a way as to avoid a reflection off of the walls nearest to them. The room has a substantial effect and all loudspeakers will perform better in a well designed room, but a well designed loudspeaker will perform even better. The corollary to that is that in a poor room the poorly designed loudspeaker will sound very bad while the better design will still sound good. A good room makes a good loudspeaker great! Good loudspeaker design is always a benefit, regardless of the room, but good room design makes the difference between a good and a great sounding system.

Finally the system design and its placement must be such as to minimize diffraction from the cabinet as well as nearby obstacles. Narrow directivity is a significant aid in this regard.

Solutions and Further Problems

Now that it is know what the loudspeaker must do, the immediate question becomes how to achieve it.

The CD part of the design poses the biggest obstacle. That's because piston sources do not have constant directivity and so they cannot be used, at least not above 500 Hz. Arrays of sources are sometimes used as a solution to this problem and often stated to have the ability to achieve CD, but this is not the case. Arrays do not posses the attribute of CD. Indeed, of all known sources only horns have this characteristic. The problem is that horns and compression drivers are well know to produce substantial coloration and perceived distortion. Dr. Earl Geddes of ***GedLee LLC*** has spent a lifetime on this later problem; how to make a horn and compression driver sound more natural - not harsh and offensive. This effort has paid off as the first listen to the **Summa** will show.

In 1991 Dr. Geddes gave the first in a long series of papers on the subject of horns. In it he defined how classic horns had significant failings in their theoretical foundations that led to serious shortcomings in their performance. He proposed a new theory, *Waveguide Theory*, which was free from these shortcomings. The waveguides that derive from his new theory are currently used in various forms throughout the industry. A notable result of the failure of horn theory is the failure to predict and define what have become known as Higher Order Modes (HOM). HOM are waves that do not travel down the axis of the device but instead propagate by bouncing off of the walls - much like the wave from a rock thrown in water would travel down a narrow channel by reflecting off of the sides. HOM generation should be seen as a serious drawback of horn theory and a significant limitation to their ability to sound natural.

Historically, CD was accomplished in a horn by using diffraction – a significant source of HOM. A narrow slit in the device forces the sound waves to diffract into a wide coverage angle which is then controlled with a basically straight sided horn. The downside of this approach is that the diffraction causes two problems. The first is the refection of energy from the diffraction slit back down the horn, which creates a standing wave resonance and its associated coloration and the second is the the substantial amount of HOM that are created.

So why are HOM so bad? It's because they travel a path down the body of the device that is longer than the direct path. In essence HOM are a form of very very early reflections. The ear effectively masks problems in the frequency domain (as MP3 techniques can attest) but it does not mask as well in the time domain. It is very sensitive to these internal diffractions and HOM and they are quite perceptible. The rise in this perception with level is what makes the horn sound like it is distorting nonlinearly. The HOM are in fact the reason that horns have their poor sonic reputation.

So how is the HOM problem solved in the **Geddes Loudspeaker designs**? This is done in two ways. The first is to design a waveguide (using Dr. Geddes theories) that has a minimum of HOM to begin with. Dr. Geddes has proven that all contours generate HOM, but some more than others. For an axi-symmetric device, the Oblate Spheriodal (OS) waveguide can be shown to be the contour that generates the least amount of HOM, although HOM do still exist even in an OS waveguide.

Over the last several years Dr. Geddes has continued to work on ways to further reduce the HOM in his waveguides. This has resulted in the *Refractive Sound Plug* or RSP (™ and Patent Pending). This plug is made of very low density open cell polyurethane foam. The idea is to absorb the HOM as they travel down the waveguide. Since the HOM travel a longer path in the device than the axial wave, the HOM will experience more sound attenuation in the foam than the axial wave. Further, since all waveguides have some reflection at the mouth (a large radius will reduce this, but not eliminate it) the RSP also helps to eliminate this standing wave. In essence the RSP allows only the desired direct wave to propagate freely and effectively attenuates all other waves. The result is a device that has significantly diminished all forms of internal diffraction, resonance and HOM. This allows for a sound as natural as any piston source yet possesses characteristics unobtainable by any piston type source.

So why not use a Tractrix horn? This question will surely come up. The answer is simply to ask: Why use a Tractrix? It can be shown that the OS waveguide has the optimal shape for the suppression of HOM. What <u>advantage</u> would a Tractrix offer? Or any other profile for that matter. The OS waveguide is ideal in this application - why use anything else?

With the higher frequencies under control, this leaves the mid frequency aspects of the problem to be addressed. First it is essential to match the directivity of the woofer to that of the waveguide. This is a necessary requirement, but one that is often ignored. It is not enough to simply use a CD device in the system, one must insure that the entire system is CD and this means that the LF driver must have the same polar pattern, at the crossover, as the waveguide. A simple investigation will show that only large woofers can provide this feature since a smaller woofer would require the crossover point to be much too high before the polar pattern was at or below 90°. When a piston woofer is used, CD can only go down to just below the crossover point, so the crossover point needs to be as low as practicable. A 15″ driver is the smallest device that can be used if a crossover point below 1 kHz is to be achieved. The woofer in the **Summa** system is 6 dB down at 45° at 800 Hz. Below this frequency their directivity begins to widen. Matching a 15″ woofer to a waveguide at 800 Hz is a workable solution. A narrower directivity than 90°, or a lower crossover point would require an even larger diameter woofer, like an 18″ and an even larger enclosure. The matching of a 15″ driver with a 90° waveguide at 800 Hz seems to be, if not the ideal choice, certainly one of the few workable ones and optimal for a reasonable sized system.

The smaller Geddes systems have to compromise on the above requirements, mostly at the crossover. As the LF driver and the waveguide gets smaller the crossover point must move upward and the control of the directivity become slightly less than the full size system. The Summa appears to be nearly ideal, but the Abbey is surprisingly close given its dramatically smaller size (and cost).

In order to be CD in all angles around the system (vertical and horizontal), the waveguide must be axi-symmetric, simply because the woofer is axi-symmetric. The waveguide will be found to require a mouth size that is exactly the same area as the 15″ − which is not a coincidence. Smaller mouths do not allow the coverage pattern control down to the crossover point and a larger mouth would not offer a significant advantage. It is clear that for optimal performance in a small room the size and composition of the system components are not really a matter of free choice, but are dictated by the system requirements. Smaller enclosures require a compromise in system performance – most notably directivity control. Larger components might offer a slight advantage, but the current choice seems to be the best compromise – the sweet spot if you will.

Housing these components so as not to degrade the imaging and low coloration that has been achieved by the component design requires a unique approach. The cabinet edges will need to have minimal diffraction if they are not to obscure the low diffraction in the waveguide. If the edge has a radius of ¼ wavelength of the sound, then above this frequency little to no diffraction will occur while below this frequency diffraction will occur, much like a form of low-pass filter. A sphere can be shown to have the least diffraction of any object with a given volume since no enclose of a given volume can have larger corner radii.

The cabinet for a 15″ loudspeaker needs a significant volume and a planar baffle to mount the components. The baffle must be at least 18″ wide to accommodate the waveguide and its mouth radius and it must be at least 33″ tall to accommodate the woofer and waveguide vertically aligned on top of one another (a requirement). Thus the front baffle needs to be at least 18″ by 33″. This does not allow for the use of a sphere (perhaps a Prolate Spheroid, but that's not too practical). Cabinet edge diffraction in the range of the waveguide would act to defeat the marvelous performance found in these devices. A radius of about 2″ is workable on the larger systems – larger would of course be better, but is not practical. The smaller systems can afford a radius of 1.5" and the Harper is .75".

After many trials (and failures), the fabrication technique that was chosen for the Summa is a fiberglass skin that is back-filled with a combination of wood fiber and dense polyurethane foam. This yields a strong rigid enclosure of reasonable weight with a very high amount of internal damping. The front baffle is 2″ thick and the waveguide is molded in and is a minimum of 2″ thick throughout. The front baffle is nearly as solid as a piece of rock. The rear portion of the cabinet has the same material composition in ½″ foam cored fiberglass walls. But on the interior it uses a second layer of material which is bonded to the outer layer with a high damping mastic thus creating a constrained layer damping effect. The enclosure is then cross braced in its interior. The cabinet itself is extremely rigid with absolutely no ringing to it at all. It is as if it were made of a solid piece of granite – but a whole lot lighter.

Most manufacturers make their cabinets out of wood and advertise the quality of the plywood (Baltic birch, etc.). But is this really what is important? How large are the cabinet corner radii? Do all of the edges have radiuses? Is the composition material non-resonant? Is there additional structural damping? In virtually every case the answer to these questions is going to be "No". In those cases where some of the answers are "Yes", the system cost is usually staggering. Gedlee has created the very best cabinet construction in a very cost effective manner. You pay for performance, not cosmetics. Unfortunately cabinet construction tends to dominate the system cost; hence it needs to be as cost effective as possible without compromising the overall system performance. These requirements do not always lead to the best looking or most stylish design.

Because the system design is assumed to have multiple subs, all of GHeddes designs are closed box. A closed box monopole matchs very well to the multiple sub approach that is recommended.

The crossovers are all computer optimized using a proprietary technique that maximizes the system performance over a coverage angle of 60°. Most crossovers are optimized for a single direction – the axis. What happens off-axis with this approach is usually not good. In a preferred Geddes setup, the listener is not actually seated on axis. The **Geddes Loudspeaker** crossovers uses high-grade components which are mechanically joined through terminal strip bars – the joints are not soldered. This improves reliability since under heat and stress a solder joints is the point of maximum-likelihood of failure. The components are spread across the entire back of the speaker - not clustered together on a circuit board - thus minimizing component interactions.

The performance data for all of the Geddes Loudspeakers can be seen at www.gedlee.com.

In Geddes designs the axial response is not necessarily the flattest response, although it is still within ±2 dB, but the 15° - 22.5° response is usually flat to within about ±1 dB. The response off-axis is extremely well controlled with no serious aberrations all the way out to ±90°. If other manufacturers ever dared to show you their off-axis responses the superior performance of this system would be obvious.

Some Likely Questions

After looking at the data on the web site there are surely some further questions that are sure to arise:

What is its power handling?

> These speakers can handle enough power to produce dangerously high SPL's in a small room. They are pro drivers designed to handle very high power.

What is the distortion level?

> There is a problem inherent with this question. GedLee has shown that THD and IMD are meaningless numbers in relating the perception of distortion. So what would be the point of measuring or showing them? Another point is that it would be extremely loud (read hearing damage) before these drivers were even close to an excursion domain where nonlinear distortion was even an issue.

Check points

The next time you go looking for or at a loudspeaker system ask yourself these questions:

1. How does it control directivity? To what coverage angle?
2. What does the DI look like?
3. How does it minimize room reflections?
4. Does the enclosure design minimize diffraction?
5. Does it have the capability to produce movie sound effects at theater levels?
6. How do I know that these things have been achieved?