Most information here was copied from http://www.diyaudio.com and copy/pasted in this document by user _Wim_ Hope you like it, and have fun building a fantastic speaker…

DIY Summa

- 5 inch mouth radius. a radius larger than 1/4 wavelength at the lowest frequency it is not necessary). So a rule of thumb is to target a radius of 1/4 lamda - which is hard to do.
- 2 inch radius on cabinet summa, 1 inch radius on smaller cabinets
- Crossover: I have before, I use a third order on the Summa and a psuedo second order on the Abbey and Nathan. You have to remember that the whole DE250 pass band is on a + 6 db/oct slope.
- 15TBX100's as the woofer
- Tripath TA2020 powered amp for compression driver?
- BMS driver are not recommended, phase plug to simple, parameters vary widely
- Zingali uses a Ciare woofer and B&C compression driver
- High frequency driver with 100ohm series resistor => resembles a current source => flatter HF response
- Type of foam: 30 ppi (pores per inch)reticulated foam
  - The reticulated foam sold for outdoor furniture upholstery is usually 35 ppi.
  - Air conditioning companies, water filter companies, pond filter companies, marine upholstery companies all carry various grades of polyurethane reticulated foam.
- Glue for foam plug: I use 3M "99" spray adhesive, but only right at the walls of course.
- I remove the screen as the foam does this same job. And Yes I retain the gasket material on the driver. These gaskets squeeze down to almost nothing.
- The throat region of the horn is much more critical than mentioned in the traditional horn literature if low diffraction (and low coloration) is an important criterion.
- Enclosure? : All of my systems use a CLD baffle. The baffle is the critical mounting for the drivers and this approach works well to minimize the transference of energy from the drivers into the box. But the air in the box couples very poorly to the physical structure and so with a decent amount of damping this source of excitation can be minimized to the point of insignificance.

  So when I tested for sound coming back through the cone, it was using the high quality B&C speakers that I use. I can't speak to other less robust devices, but then I don't use those.
- The waveguide size has more to do with how well it works than the crossover point. A 10" one does not control as well as the 12" which is not as good as the 15". Ideally it should be about 18" - I could make that size work perfectly. The 10" has a lot of flaws, but the size is attractive to most people.

  The woofer size determines the frequency at which the two source patterns mate up. As you would expect the 10" woofer mates to the 90° output of the waveguide at a higher frequency than the 12" and 15". The 15" is about 800-900 Hz the 12 about 1 kHz and the 10 about 1500 Hz.

From my data, the ideal would be an 18" waveguide with a 12" woofer. This would elliminate all the flaws that I see in practice. But such a combination would not be an attactive speaker for a lot of reasons. the next best, and what I would build for myself, is a 12" woofer and a 15" waveguide. D

- My speakers have always been B&C, even though I built some with TADs, I couldn't personaly afford the TADs. I have tried the DE500 and found it pretty much the same as the DE250, but twice the price. I don't actually expect that there could be much of an improvement on the DE250 within the existing marketplace. Most likely would be a BMS, but they would cost me a lot more and I don't see the potential advantage except maybe a little more energy above 15 kHz. You can just guess how much I think thats worth.

  I would love to improve the woofer because of its internal resonances, but nothing that I have tested is any better and only costs more. Again, kind of "what's the point". You see I don't believe anything that I read or see in terms of specs, I test everything myself under identical conditions. Then you find out that what's being said and shown as measurments doesn't really stand up under scrutiny.
- Listening distance: 10 center to center spacing of the drivers
- It is terribly wrong to EQ'd to an axial response since there are many things that happen on axis which should actually be ignored. I EQ to the power response in the forward +-30° direction (just as JBL and many others do). I almost completely ignore the axial response since I don't recommend listening on axis.
- Finally you will have a peak in the response due to the drivers resonance and this can be tamed with the LRC across the driver. I generally find that there is a second peak above the driver resonance which is due to the resonance in the waveguide. Another LRC helps tame this one. In some cases there is another, a third one, and another LRC works. But by the time you have three LRCs in parrallel with the drivers impedance things get pretty hairy and difficult to tune. Generally the values become so precise that they are hard to obtain. In the ESP line we resorted to custom inductors because this was the only way to get the correct values.
- I do measurements every 7.5 degrees. I first work on flattening the 22.5 degree line and then check the whole polar response to see that I haven't made things terribly worse at other locations. I do tend to ignore the axial curve and put a lot of weight on the total power flatness. I think that you need good detail out to 90 degrees. I don't worry much about behind the source. I believe that I can hear a power response peak even if its response near the axis is not that great. The second resonance that I talked about tends to be a power response peak (a peak in the response at all polar lines) but is not so obvious in the near axial curves. In fact, on axis, its a hole, which is logical if its a resonance of the waveguide. You can imagine raising this level to correct the axial hole, thus pushing the power response way up - net result, a horrible sound.
-

DIY Sub

- They have -6db points of something like 45hz and 100hz with a peak at 60hz
- They are about 14" x 16" x 18".
- *see 4 ports, but setup non-symmetrically. Is this intentional, or was that just for looks?* => Intentional and important. If they are all symmetrical then the

ports pick up the same standing wave in the front cavity. This way they all hit different spots and different modes. The four ports here can be longer than a single one in the center because that where the woofer magnet is. Longer ports allow for less sound leakage above the box tuning, which can be a big deal in these designs.

Also, four ports like this can do what one cannot and that is static flow of hot air. There will be a cross flow of air as this box heats up that will have no effect on the sound, but will help to cool the magnet. One port can't do that. They are small for damping, thus elliminating the need for foam to dampen the resonances.

- A properly built sub enclosure will have its first panel resonance well above the operating range, so there's no point in damping.
- The driver Q is about .25 - about as low as possible. In a bandpass design the Q that you have to worry about is the Helmholtz resonance Q of the front chamber. To make a small box, this Q will be high and needs to be dampen. Yes, there is some port compression due to the increase in resistance at higher velocities, but thats not a bad thing. At higher SPLs the Q goes down and the output level drops slightly, maybe 3 dB max. But at higher SPL we are far more sensitive to bass so a drop in bass level is not a bad thing and in fact could well be a good thing. I have been using this type of sub in my theater for about five years and I find it very desirable. The bass never seems to get of of control no matter how loud you make it.
- I think that the point is that things like linearity and frequency response don't matter much when all one wants from the sub is 20 -> 40 or 50 Hz. Basically it is just a big piston. I'm not even sure that long throw matters much if it is bandpass. It just needs to pump some air without breaking.
- Subwoofer placement: Actually its 1) in corner, a must 2) along an opposing wall someplace (lots of possibilities here), recommended 3) anywhere but a corner, away from the first two and if possible, above the center line.
- still only see the single mode/node discussion as being misleading. Even at a node a source will excite other nearby modes since all modes are excited by all frequencies to some extent. Then there is the direct sound, but careful study will show that this direct sound is in fact the contribution of all the modes (Welti got this wrong in his paper). This goes back to something that I said a long time ago that nobody accepted. Free space has to be thought of as a continuum where the modal density goes to infinity - not zero. Hence, even outdoors in free space the direct field is carried by the modes (which are now infinite in density) just as it is in a small room. But in a small room the modes get sparse and hence the ability for them to carry the energy goes down.

  The sound from a LF source does not - let me repeat - does not travel in all directions away from the source. (There is what is called an evanescent wave sent out, but this disapates in time and space exponentialy so it is a very small factor. The sound wave can travel only in a discrete number of directions defined by the modes that it excites. This means that the energy emitted by this source in a real room is not the same as the energy emmitted by this source in free space. This can be seen in the radiation impedance for a small room which is NOT the same as that for a source in free space.

  People want to think that you take a free space source and bring it into a room and that it emites sound the same way, but that this gets amplified by the modes at certain frequencies. This is not correct. The presence of the room changes everything and not until the source sees a high model density does it begin to behave as it does in free space.

- The Xo location can only be determined in-situ, but the slope can be decided. I would use 2 nd order even if I were designing the system myself. I have not found localization issues in any bandpass sub usage, but then these, acoustically, end up being 4th order LP. But that probably does not happen arround the cutoff since it would be unusual that the elctrical filter and the acoustic one were at the same frequencies. Thats why, to me, 4th order is too sharp if it designed such that all four poles are nearly the same frequency. Maybe thats why I like bandpass and 2nd order - eventual sharp cutoff, but more gradual at the cutoff.
- The rest - modal effects are significant well up to 150 Hz. in most rooms. As long as there are modal effects we need multiple sources to "tame" them. The mains are some of those sources and yes they do work better I have found as closed box (which is why thats all I sell anymore.) But its also correct that the mains LF sources are seldom if ever in desirable locations and never phased or leveled (as pointed out above). So while they might help at the very LFs they are not really all that effective. So the subs need to be "blended" into the existing sound field created by the mains. This is why I suggest overlaping and not HPing the mains. A smooth blend smoothes out the response and the main retain the imgae cues for propoer localization of the instruments. Done properly one never localizes on the subs only the mains as they should. In short, no I never HP the mains.
- That high damping reults in high decay is obvious, but its actually the modal interaction that results form the broader modes that I am interested in. I mean really, its all good - lower peaks, shorter decay, broader modes, beter coupling. Which is the "most important" is kind of moot. I dod not see the interest in whether or not EQ changes the decay rate - I just don't see the importance.

  Ideally, in my mind, I would heavily damp the LFs in a small room, but then add back LF reverberation - i.e increase the decay rate. However, please note here that this added reverb is broadband NOT modal - big difference.
-

DIY Summa questions

- Gedlee is using 18db/oct, is this by any chance inspired by the minimum group delay (as suggested by JMLC)
- What would be the downside of using thicker foam (100ppi)
  - Possibility for increased directivity
  - More pronounced damping of the homs
- extent the foam beyond the mouth, eg by placing a sheet of foam in front of the speaker (like a sort of speaker grill)
  - this could also help with diffraction effects at mouth, baffle sides
- further optimization with piece-wise continuos function any results?
- What would the phase-plug of a HOMless driver look like? What is the best existing driver for lowest generation of HOM?

Driver Exit Angles (total opening angle = waveguide angle x 2)

BMS 4552ND: +-23.6° (measured by Mavo)
BMS 4540ND: 14° (recommended by Patrick Bateman)

Info jzagaja
18sound:
NSD1095N, ND1090 = 27°
NSD1480N = 10°   (1.4”)
B&C
1"
DE250 14.6°
DE10 7.7°
DE12 24°
DE400TN 20.7°
DE400 31°
DE500 17°
DE200 9.9°
2"
DE85TN 34.5°
DE750TN 22°
DE950TN 17°
BEYMA:
CP750Nd – 24º
CP850Nd – 7º
CP755Nd – 12º60’
SMC65Nd – 15º
CP385Nd – 16º30’
SMC225Nd – 13º1’

Horn Drivers

Question: recognize sonic character from driver

Hi Paul, just talked to Mike at Radian on the phone yesterday. Without breaking any NDA's (they sell drivers to several $80,000/pr high-end vendors), the 850-PB 2" is favored for its relaxed, open midrange and performance almost up the top of the range (it is frequently used with supertweeters in ultra-fi applications). When ultimate HF extension is more important, the smaller 1.4" 835-PB is favored for slightly sweeter HF at the expense of a bit less midrange power-handling.

Apparently, the real differences sonically are the compression ratio and diaphragm material - higher compression ratios are more "focussed" and intense in the midrange, and lower compression ratios are more relaxed sounding. With a 3" diaphragm and a 2" exit, the 850-PB has a lower compression ratio than the 835-PB, with its 1.4" exit. This alters the sonic presentation.

I asked specifically about the 950-PB (4" diaphragm & neo) vs the 850-PB (3" diaphragm & ceramic) and the implied much lower distortion for the 950-PB mentioned on the web-page, but Mike said the 950-PB is only about 2.5 to 3 dB lower than the 850-PB - and both are many DB lower than the competition from JBL and TAD. The larger diaphragm is why the 950-PB has a somewhat lumpier extreme

HF compared to the smoothness of the 850-PB.

Mike mentioned that some Ti diaphragms have a distortion peak around 2 kHz, compared to aluminum. The people who have commented negatively on the sonics of Ti are reacting to this Ti abberation in the midrange. Reading between the lines, Ti is selected by CD manufacturers for extreme HF performance - at the expense of midrange, where high-purity aluminum is at its best. According to Mike, Beryllium is (marginally) best, but is notoriously difficult to fabricate and is apparently prone to sudden failure.

I didn't realize this, but the 5312 coax (the one Mike feels is their most advanced and best-sounding coax) actually uses a 2" exit CD from Radian's 651PB/760PB/850PB compression-driver series - not the smaller-exit CD's used by other coax vendors. Yes, Radian makes the Hemp Acoustics series of coaxes, which use non-Radian hemp cones and slightly different CD diaphragms.

Question: optimizations of the typical horn driver

You're welcome - good paper, isn't it? Most of it far above my head, but it's clear the author has done a thorough analysis - a little reminiscent of the original Richard Small doctoral thesis papers. Have to give credit to Australian Universities for supporting audio research.

Dr. Geddes, I was wondering about one of the things you mentioned over in the Waveguide thread. Perhaps I misread it, but it appeared that a major source of HOM's was simply the fact the throat size was nonzero - in other words, the smaller the throat, the less the HOM's. Is this a correct reading?

In traditional horn theory, second-order nonlinear distortion is assumed to primarily originate from the throat geometry, but as described in the 2002 Voishvillo paper, the major source of second-order air nonlinearity is actually between the diaphragm and the rear surface of the phase plug, and other small-dimension parts of the phase plug.

If throat size is an open variable, and you had a free hand designing the phase plug, would smaller throats have better performance in an OS waveguide - say, half or quarter-inch? Or would that be flirting with second-order distortion from air nonlinearities?

John Sheerin and John Janowitz, thanks for the commentary about the distortion curves of the Beyma drivers. 0.1% third-harmonic distortion at 95 dB SPL seemed too good to be true. That puts the BMS 18N850 in a more favorable light, particularly considering the astonishing drive level. You could actually plug the 18N850 into the wall in North America (1800 watts at 60 Hz and 120V RMS) and it would survive for a little while. I wouldn't want to be in the same town, though.

As for a potential TD 18-incher, I'd vote for 40~50 mm Xdamage, as with the Beyma and BMS drivers. It would be nice not to have to babysit the driver against wayward LF content from movies or techno CD's. Performance somewhere in the Beyma and BMS league would be desirable (highish Fs, Qts, and efficiency), and I'd vote for silicone/Aquaplas-damped double spiders to minimize undesirable side-to-side rocking modes. Double spiders also offer the option of push-pull distortion cancellation by reverse-mounting one of them, which seems like a clever and easy-to-do idea.

One of the things I'll be mentioning to Great Plains Audio is applying Aquaplas (now called Antivibe) to the tangential surround of the aluminum compression driver diaphragm. JBL applies Aquaplas/Antivibe to the whole diaphragm when you buy a 435Be, but I feel this is a mistake. The part of the diaphragm with the most chaotic radiation is obviously the surround, and it's an area where mass-damping and outright suppression of radiation is desirable. That is NOT true of the diaphragm dome, where low-as-possible mass and uniform emission into the phase plug assembly are primary goals. Raising the mass of the diaphragm is extremely undesirable, since it depresses efficiency and decreases HF extension.

Since the dome of the diaphragm and its surround operate in completely different ways, and in fact have completely different functions, it only makes sense to treat them differently, rather than applying damping goo to the whole thing. Adding a bit of mass damping to the surround seems like a good idea, and is likely to improve the mechanical termination between the moving diaphragm and stationary mounting ring.

Well, to be honest, I think there's a lot of mysticism about WECO, Altec, JBL, and TAD. True, they made great, classic loudspeakers, but physics still applies to them, and there are areas of design that were overlooked or ignored.

It hardly seems controversial to treat the surround and spider differently than the diaphragm - this has been standard practice for direct-radiators for 80 years. The task of the surround and spider is to assure pistonic motion, prevent side-to-side rocking motions, minimize nonlinear distortion, and minimize spurious emission. Rice & Kellogg were aware of all these requirements.

The aluminum tangential surround was originally chosen to provide maximum excursion linearity for a near-full-range WECO theater driver. Due to the limited HF content of movie soundtracks, spurious emission above 8 kHz was not a concern, but power-handling was very important, with only one speaker system behind the screen, and very large theaters.

Adding mass to a diaphragm always lowers efficiency, lowers the mass rolloff point, and lowers the maximum HF extension of a driver. In direct-radiator tweeters, the amount of damping goo to apply to a silk-dome tweeter is a judgement call between leaving in resonances and too much depressing of efficiency and HF extension. Similarly, the reason the Bextrene drivers of the Seventies were so woefully low in efficiency (85 dB/metre typically) wasn't the Bextrene cone itself, but the very generous application of damping goo to quiet down the 1.5 kHz and higher modes - raw Bextrene is actually very resonant. The BBC developed the inherently lossy polypropylene cone specifically so they wouldn't have to mess with applying damping goo to the cone.

Now that optical soundtracks are no longer in use, and compression drivers are expected to cover the 8~20 kHz range, it exposes problems with diaphragm breakup and surround spurious emission - the same problems seen in direct-radiators for the last 80 years. The problem with compression drivers is that they are much bigger than the equivalent direct-radiator tweeter, so these HF problems creep in at lower frequencies.

Spurious emission from the surround is a problem for direct-radiator tweeters, and as shown by the 1979 Murray paper, is a problem for compression drivers as well. I guess the reason I have a slight problem with applying Aquaplas to a beryllium diaphragm is that after decades of PR about how wonderful beryllium is, now it has to be treated with the same kind of damping goo as other, less awesome drivers using titanium or aluminum diaphragms. The plain fact is that large-format compression drivers just aren't at their best above 8 kHz - the HF cutoff of old optical soundtracks, which is what the WECO and Altec compression drivers were originally designed for.

I've listened to the big TAD speakers (with my favorite recordings) over several months, and liked them, but wasn't bowled over by them. HF and extreme HF were good but not great. So I'm not really in the beryllium camp, despite the obviously superior measurements. I'm looking forward to auditioning a K2, but I'm not expecting extreme HF better than anything I've heard before.

CD Horns

Introduction:

I have listened to horns for more than 40 years. Basically thats all I have ever owned. They have their pro's and cons, but I like so many, found that they had a sound quality that can sometimes grate on your nerves - its often called harshness and I find this an appropriate term. Now, not I, nor anyone else, can tell you exactly what harshness is or how to measure it, but certainly no one will deny that it is real.

I spent a great deal of effort over the last several decades on trying to understand this poor quality, because, quite frankly, in every other aspect horns beat all other types of HF sources hands down.

Some thirty years ago I started to study horns in detail, but it didn't take too long to figure out that analytically (mathematically) they were sadly lacking. There were some asumptions made that everyone knew weren't true, but everyone just kept on using them anyways. I wasn't satisfied with this lack of understanding so I dug a little deeper. And then it dawned on me that what was needed was a more complete mathematical description of the performance of a horn - and waveguide theory was born (thats now all in print and well accepted). This new approach predicted some radically new (for the time, about 20 years ago) profiles. There were some early attempts at using the new concepts but they weren't very sucessful and for all practical purposes the ideas laid dormant for a number of years. Some of the concepts were begining to be used, like mouth radi to reduce mouth diffraction, etc. but no one was actually making true waveguides according to my original work.

When I left Ford about ten years ago I decided to renew my interest in the concepts. I built some waveguides and low and behold, they actually did sound better - to me at least. But they also measured better. Back in the early 90's, John Eargle, who was good friend and had a strong interest in my work, commented that JBL had built some waveguides and found that the impedance (electrical) of the devices was very smooth - none of the multiple ripples found in the diffraction horns of the era. He commented on how much of an advantage this was to a passive crossover. Years later I was to see this advantage in practice.

While the waveguide themselfs were a big improvement, it was not until I tried a foam plug that I really heard something that got my attention. ANYONE who has tried foam will tell you that the difference isn't small - its major. Needless to say I was intrigued. I wanted to understand why this simple device worked so well. As I looked into the situation further I begain to put together a corherent concept of what might be at play here - namely the HOM (I had discovered that distortion wasn't a factor as I had though at first - see the B&C paper). The new theory had predicted that HOM would exist and there is now little doubt that they do, but is this the sole answer to what makes the new devices sound the way that they do? Quite honestly, I am not sure.

ALL of the data that "I" have says that the HOM and the internal reflection reductions are what is making the difference. However, I am still NOT convinced that this is ALL there is to the story. That HOMs are part of the story, I have no doubt, that they are the whole story, I am far less confident. In fact I have some pet theories on what else it might be, but alas those are not for public consumption at this point.

So in a nutshell, I have spent nearly 30 years trying to improve the sound of a horn. I believe that I have done that. What exactly is it that I have done to make this difference? - first, better contours than have less diffraction and edge treatments that create less reflections and diffraction. These things CAN be measured and with some experience its possible to see them in a set of data. I see them, but clearly not everyone else does.

Second there is the foam plug. Exactly what this does is not yet clear, but damping of the undesirable waves is certainly part of it. But there might be other aspects to it as well. Measureing these effects IS NOT easy and even I have not found the "smoking gun" as to exactly what is going on. That the foam plug is not some audiophool psuedo-science can only be truely stated by those who have actually experinced it. I have not heard a single person say that they did not hear a difference, and further, that it was not an improvement. But of course this later data is all circumstantial at this point. And its likely to stay that way for a fairly long time. Thats just the way audio is.

Question: what is an OS Waveguide

It turns out that OS waveguides are quite constrained by the math. Once you have a driver and then the coverage pattern, the rest is fixed. OS waveguides do not have a predicted "cutoff" so the low end tends to be dictated by the compression driver. The coverage angle can only be held down to where the mouth dimension is too small to control it. In the ESP15 (a Summa) this coverage narrows a bit at about 1000 - 2000 Hz. The waveguide is about 16 inches across. In the smaller ESP12 and ESP10 the waveguide is smaller with a notable raising of the lowest frequency of coverage. It is really important in these devices to have a sizable waveguide - too small is a lot of compromises.

So the design procedure is simple. Pick your Comp driver, and then your coverage pattern then the contour becomes:

y(x) = sqrt(throat radius^2 + x^2 Tan(coverage_angle)^2)

x is the distance along the axis. Note that at x = 0 the angle is zero and the radius is "throat_radius".

For extreme accuracy, which appears to be important, one wants the initial angle and radius of the wavegiude to match the exit angle and radius of the driver. This is tricky, but one who is competent at design can work out the correct numbers from the above equation. I did it in MathCAD. I can generate the contour for any radius and exit angle, (I would have to charge a fee for that). Its not an intractable problem however and trial and error on a piece of paper or in a spreadsheet can get you what you need.

Note that larger throats on the compression driver invariably lead to a lower frequency of falloff when the waveguide is true CD. Many drivers show good HF response out to 20 kHz on a plane wave or on a non-CD horn, but when put on a true CD device like an OS waveguide the response dies above 10-12 kHz for a 1.5 " driver and 9-10 kHz for a 2" driver. This is why I have only used 1" drivers. I have not found one that goes out far enough in any larger throat sizes.
Geddes

Question: what is Constant Directivity?

CD not not mean "uniform in angle, it means uniform in frequency".

I think that its important to understand that Constant Directivity does not mean that the sound stays at the same level as one moves off axis - and then somehow falls to zero at the coverage angle. Waveguides have a continuous drop in level - independent of frequency however - as one moves off axis up until the coverage angle and then the drop is steeper.

This slow drop with angle is exactly what one needs off axis in the toe-in configuration.

The wider the angle of the device (as above) the faster the initial falloff with angle and the slower beyond the coverage angle and this tedns to not be frequency independent - in other words the wider the coverage angle the more the polar response looks like a piston - not surprising.

At about 90 degree coverage (45 degree wall angle) one gets just about the ideal angular falloff. Narrower than this and within the coverage its not falling fast enough, but then it drops like a stone. Wider than this and the falloff with angle is too great.

Question: impotance of CD, Constant Directivity?

Another point that I think needs to be understood is the impotance of CD, Constant Directivity. We must consider that without CD we cannot have a flat power response and a flat axial response.

Most reserchers agree that the power response is very important for tone coloration while the direct response tends to be the major factor in imaging. The industry is all too focused on getting a flat "axial" response, but to me this is probably the least important measurement.

In a polar diagram, the axial response respresents a very small portion of the radiated sound field, its a small disk at the center, but the off axis points represent every greater areas - anuluses (sp?) of increasing area. The axial point is therefor the least significant point for the power response - it has almost no effect on the power response. Further, there are very good reasons for one to not be directly on-axis of the loudspeaker (another topic), and the classic "sweat-spot" approach to sound design is kind of hedonistic. In a home theater there can be six or eight people listening - a sweat spot is simply not viable in that venue, and lets face thats the venue of the future.

For these reasons the power response and the polar responses must be smooth and flat even if the axial response is not. In my deisgns I pretty mush ignore the axial response seeking to get the best 22.5 degree response with smooth and flat polar/power response. Typically the axial response is not ideal in this scenario.

Now in a small room the situation is even more constrained. Thats because the sound system in a small room needs to avoid the very close by room boundaries to as great an extent as possible. The very early reflections and the lack of a gap between the direct sound and the reveberant field will create confusion in the image and a poor timbre of the sound. This means that in addition to needing CD, we need CD with a very narrow coverage angle - not a trivial task.

In my years of research I have only ever found one way to get CD and narrow directivity at the same time and that is with a horn. But classic horns sounded, if not terrible, certainly collored, distorted, not good! I spent nearly 20 years on this problem since I could see that horns were the solution to the CD and narrow coverage problem, but only if we could solve the sound quality issues.

The solution, that I have found, is a waveguide, with a foam plug. This device has the sound quality of the very best direct radiators, but a much better pattern control - true CD. Then lest we not forget about signal power and power compression. A compression driver will have a fraction of the power compression of a small tweeter and loads more headroom. Whats not to like!!

Our imaging and timbre perceptions are nearly dominated by the sound above 1 kHz. The musical content, the rythm, etc. are carried by those frequencies below 1 kHz, but our "perception" is inordinately weighted by the response above 1 kHz. There is a very good reason for this psychoacoustically, and it has to do with the way the nureons fire in the ear (an interesting topic in and of itself, but the important point to note is that we process sound differently above and below about 1 kHz).

So getting the 1 kHz. and up right is paramount to a good perception of coloration and imaging. I feel that far too little attention is paid to this critical region in the market place because it is here that we see all kinds of problems and yet it is here that we should be the most concerned.

I view sound system design in three major frequency ranges - low frequencies, where modal effects and the room dominates, there is no imaging or psychoacoustics to worry about, its simply a matter of adequite output and smooth spatial and frequency response (more on this in another thread); 200 Hz - 1000 Hz, probably the most forgiving of the three regions, our auditory system is only just begining to be capable of resolving spatial aspects (localization) and it is not yet very good at resolution of time delays, reflections and frequency response. If you are going to compromise something do it here as it will have the least noticable effect. Above 1 kHz is where we live as far as music is concerned. This region is ultra sensitive to time delays, reflections, frequency response, diffraction, all the things that tend to mess up coloration and imaging. Mess up this frequency region and you won't be able to recover the sound quality. Here is not where you want to make compromises for sound quality.

Question: calculate lower boundary directivity vs. frequency

On the ESP web site it references Keele's Asymptotic Model. As I understand it, it allows us to calculate at what frequency a CD wave guide exhibits constant directivity behavior. Below this frequency, there is a narrowing of directivity to 2/3 of the wave guide's included angle. Link to ESP wave guide page:
http://sound.westhost.com/articles/waveguides1.htm
The formula is given as: F = Kk / á * w Where á = included wall angle, Kk = 25.306 x 10^3 and w = mouth width (meters). This is for rectangular wave guides. For an axis-symmetrical circular wave guide, the Kk constant is given as 29.707 x 10^3. This is the frequency at which the wave guide will exhibit constant directivity.
To find the frequency where directivity narrows to 2/3 of the wall angle the formula: 10^ (Log (f) - 0.176) is given; "f" being the frequency at which the wave guide exhibits constant directivity.
1.) Are the above formulas valid from your experience?
2.) At what frequency do you place the crossover? Do you place is before the dip in directivity, at the lowest dip in directivity, or only after constant directivity behavior begins?
3.) What is the ideal place to crossover to the wave guide in the example below?

That this effect occurs is quite true. Don did a lot of good work in this area for its time, but that was some time ago. His methods were somewhat rough as well as his capabilities somewhat limited so, at best, we have to assume his formulas to be approximations. How good are these approximations? I don't have a good feeling for that especially as applied to a symmetric waveguide like I use.

My crossovers tend to be below the dip in directivity, but very close to the dip.

Ideally one would want to crossover above 850 Hz in your example. But that example is far more optimistic than I have found to be the case in practice. On a 15" 90 degree included angle the dip in directivity is about 1 kHz or a little bit higher. This would imply that the Keele formula that you show is quite optimistic for a waveguide, which is not at all surprising since Keele used diffraction devices in all his work. A diffraction device has a much more controllable and predictable polar pattern. Its just that it also has a lot of diffraction and standing waves to achieve this predictable control. Control versus sound quality, thats the tradeoff.

Hence while I would love to always work above the "dip" it is not feasible in my designs (which prioritize sound quality for nice plots), I have to deal with the dip and do the best that I can to minimize its effects.
=>
The formula outlined are called "Keeles asymptotic model" by Hendrickson and Ureda in the Manta ray horn reference, I have just converted them from imperial to metric units.

Johansen in his paper shows that they are fairly accurate for rectangular mouth horns as originally published, but need to be modified somewhat for axis symetry devices.
There is also an effect due to the flare shape near the mouth;- basically if you make

a mouth section with a larger flare than the conical inner section the width for the same cut off can be reduced by 1.12, and if the total device consists of three conical sections the factor is around 1.24, the measurements I have made showing these hold up reasonably well if circular arcs replace the outer section in the first case, or constitute the whole device in the second.

As I have previously stated these are simple approximate formula that can give you a good idea of the form of a working device with a bit of calculator bashing, and measurements of actual devices and systems using them show that they are perfectly adequate for most practical purposes.
As far as directivity goes remember that two in phase sources 6db. down at the same frequencty sum flat and if you take one device away the result is 6db. Down, i.e. if one device has no output at all the maximum discrepancy is 6db. How many speaker builders can honestly say that about their creations?

Question: vertical lobes?

There are two unavoidable polar response lobe holes in the vertical direction - this is the worst area for power response because of this, but it's not as bad as many, if not most speakers. A lot of time was spent with the crossover to optimize this response aberation. One of the lobe "holes" is aimed at the floor bounce.

Question: Time alignment?

From the impulse response, the time alignment of the two drivers is quite close - the difference is in the usecs. The combination of physical offset and time delay from the LP filter makes the matchup of the delays almost exact. The woofer could go forward an inch or so to be ideal, or the waveguide back, but thats very difficult to do at this point and for a few usecs its probably not worth it.

Question: Whats HOM?

HOM - Higher Order Mode, its a term that I coined to define waves that propagate in a waveguide that do not go down the axis, but travel by bouncing off of the walls. They are not predicted by the Horn Equation, so most people didn't even know that they existed (I was the first person to hypothesize there existance). The Waveguide Theory predicts them, and low and behold, it turns out that they are quite significant to audibility. Minimizing them yields a far better sound quality. But with "horns" its not possible to minimize them because you don't know what to do - the equations aren't rigorous enough to predict them so they are simply ignored.

HOM are an alternate allowed form of wave propagation

When a wave propagates down a waveguide it bends to keep itself in contact with and parallel to the walls - this is the boundary condition. No matter what, at some wavenumber, this cannot be true and a second wave is estabished to enforce the boundary condition. Now this wave can be called "scatering from the walls, or diffraction from the walls, whatever, the the net result is the presence of a second mode of wave propagation, one that reflects from the boundary. I call this wave an HOM, others seem to want to call it something else. OK, but I'll continue to call them HOM because that's what they are. Its not really so complicated.

Question: Mathematics of HOM?

If I measure the wavefront at the mouth of a horn, then I know exactly how it will radiate. Now this wavefront will be the sum over all the modes in a plane aperature, Bessel functions if it is a plane. But these modes are NOT the modes of the waveguide as those are Spherical Harmonics. There is a relationshp between the Spherical Harmonics and the Bessel functions and one could then calculate the wavefront in terms of the Spherical Harmonics. The lowest order mode is that of a uniform velocity profile on a spherical surface. This is the "main mode" of wave propagation in a conical waveguide - all waveguides become conical at the mouth. Now if, and this will almost always be true, there is deviation from a perfectly uniform wavefront on the hypothetical spherical surface in the mouth, then this deviation has to be the result of HOM. It would be an almost insurmountable task to derive how much of the HOM at the mouth was due to the three possible sources of HOM creation 1) the driver diaphragm modes 2) the phase plug and interface, 3) the waveguide itself.

Question: sources of HOM

There are three sources of HOM
1) diaphragm non-pistonic motion
2) phase plug and horn driver interface (which to me are the same thing)
3) the horn itself
In any given device with any given driver these three sources could come in any mixture and further this mixture would be frequency dependent. The fact that there is so much confusion about HOM is evidence of the difficulty in understanding them.
Geddes

Question: most important source of HOM

BUT, here is some data that makes the issues at least somewhat clearer. 1) identical drivers on a poor horn will sound bad, hence the HOM in the driver are not dominate because 2) the same driver on a good waveguide sounds better, hence the HOM created by the waveguide or horn must be a strong contributing factor. Finally 3) adding foam does not change the creation of HOM by either the driver or the horn/phaseplug, but does improve the sound, hence the foam must have an effect on something that causes the poor sound quality. Is HOM reduction the whole story? I doubt it. Are the HOM contributed by the waveguide a factor, of that I have little doubt and no one that I know of has studied this as much as I have.
=> **Horn**

Question: How to measure HOM?

HOM can be measured. In underwater acoustics they call it "matched field processing". Under water sound is propagating in a waveguide. Measuring the acoustic field with several geophones at different depths makes it possible to sort out the vertical wavenumber spectrum of the sound. This can be done by taking a Fourier transform over the array and thereby sorting out the different modes. For a horn it may be more difficult. You need to measure the spatial variation at the mouth of the horn by moving the microphone in steps across the mouth. Then a Fourier transform of there measurements at a given frequency will give you the wavenumber

spectrum, kx and the different HOM's. Not quite sure how the near field evanescent waves will influence the results though.
SEH

That is quite correct. I worked in underwater sound at Penn State where the torpedo sonar heads were developed. I know HOW to do it, I just don't have the capability. And in the end we already know that we don't want them and measureing them won't change that. At B&C I showed them how to measure the HOM from the drivers using microphones along a plane wave tube. The HOM in the horn could be done the same way. Since its axisymmetric, you only need a line of point measurements and you can sort out everything.

Theoretically you can measure the far field and calculate the mouth velocities - acoustic holography - and from that you could calculate the modes. But this method is highly prone to errors and singular or near singular matrices in the inversion process - we tried this several year ago. WIth enough computer power however the matrices could be analyzed with SVD, but basically it all requires a tremendous amount of effort, which in the end doesn't change what you want to do.

Question: were possible to subtract out the direct wave, what would the spectra of the remaining HOM's look like?

It would likely be very spiky, not flat or smooth at all. The modes have sharp high-Q cut-ins which increase in amplitude as the mode number goes up. So the response would be anything but smooth. At a high enough mode number it would be virtually a very sharp high-Q resonance - much like the LF modes in a totally undamped room. Yes, higher and sharper Q as the frequency goes up, more dense too. In a 1" driver there are maybe 5-8 modes in the audible band. For a 2" this would likely quadruple.
=> This means the HOM's depend not on diameter of the throat, but on its area.
(I presume 1~20 kHz) ...
=> Actually I meant to 10 kHz. To 20 KHz the numbers would at least quadruple. HOMs would be maximally audible in the range from 1 kHz to 10 kHz, peaking - predictably - at about 3 kHz.

Question: Are these modes (highly) directional, or do they emerge from the entire mouth of the horn or waveguide and diverge smoothly outward?

The answer is more like the later. Each mode will have a different directivity from every other one AND this will change with frequency. Clearly a situation that would be difficult to analyze. Thats why I simply take the posotion that I need to minimize these undesirable attributes and not worry too much about the details of how they propagate. If there aren't any, or they have been reduced as low as possible, then how they propagate is academic. So I really haven't studied the directivity of them very much. Gotfried Bueler did some work in this a few years back at AES. He showed that they were indeed complex in the patterns that they radiated. His work was purely experimental.

Question: how does an axi-symmetric horn produce non-axi-symmetric HOMs?

The colorful 2D graphs are very detailed pressure maps of the output coming from the two test horns. The reference level is taken at 0 degrees, and is dark red. Yellow is -8 dB down from the zero-axis level, and deep blue is -20 dB down.

The measurements are taken with an automated MLSSA system and an X-Y traverse system to move the microphone across an XY axis in front of the rigidly mounted horn. As you can see from the captions, the microphone is stepped across 300mm in each direction.

It was the non-axi-symmetric results that were the most interesting to me. When I did the HOM solutions for the OS waveguide, I assumed axisymmetry and always wondered about this. The reason is that the non-axi-symmetry modes occur at frequencies well below the axisymmetric ones, so they are a more seriuos problem. Theoretically they should not exist in a purely axi-symmetric system, but Morgan shows very clearly that they do exist. How this happens is most curiuos indeed.

Its a real mystery, but one that I had a lot of evidence to suspect would be present. I first realized this after a conversation with Don Keele. He was tell me that when they rotated the driver on a waveguide they got a different polar pattern! Thats when I came to realize that real devices were anything but ideal.

It would be of great interest to me to study where these asymmetries come from. Alas, nobody does advanced work like this anymore.

Question: Would some surface modulation (beyond the OS waveguide smooth contour) might serve to further enhance suppression of HOM propagation, and or enhance the DI of the guides?

(example from microwave transmission)

The answer would be yes, but of limited value. The effect would be small since the depth of any surface treatment would be small compared to a wavelength at anything but the very highest frequencies. But in effect, the way in which the foam plug is mounted will do exactly what you suggest, albeit more in a random form than a controlled on. The glue at the boundaries will create a rough surface which will tend to scatter the HOM as they inping upon it.

Question: how is hom generated in a horn

the OS is fed with a plane wave and this plane wave is bent into a spherical wave. This bending in and of itself does not result in diffraction - thats a key point that you seem to be missing. There is a given amount of bending of the wavefront that is allowed without diffraction. Any deviation of the walls from this "allowed" contour of minimum diffraction wil create HOM waves to "make up the difference". Diffraction is the cause, the HOM are the result. But it is incorrect to think that all wavefront curvature creates diffraction and hence HOM. In practice, it would not be feasible to create the conditions under which absolutely NO dffraction would occur, but it could be done. The point is simply that curvature does not mean diffraction and HOM, its not that simple.

Question: the best compression driver?

No, I don't have a lot of drivers measured. From what I have done, there is not a large difference, the waveguide tends to dominate the response and the driver differences show up mostly at the high end because of diaphragm breakup. In some drivers there is a voice coil leak "hole" (like the TAD drivers) and this always shows up, but for similarly constructed drivers like the JBLs, Beymas, B&Cs, etc. the results are very very similar.

Question: Exit angle of the driver?

It seems to me like a phase plug inside the driver, changes exit angles considerably In which case the side angles of the exit doesnt really tell much about actual exit angle But if you use the flare rate exit angle caused by phase plug, it will cause abrupt change in slope rate of the exit troath angle If manufactor gives information about exit angles, do they deal with this issue, or will such spec be totally unreliable If above holds any truth, the short steep OS throath transistion begins to make sense =>
For example for a certain Celestion CD-driver an optimum entry angle (of a horn/wg) is 25deg and this is not the driver's wall exit angle (which even is negative in this example), rather it continues the cross-section expansion rate -- which looks right to me as long as we are below the critical frequency when wavelenghts start to be important.=>
It is all wavelength dependent. If the waves are much longer than the dimensions involved then the angles don't matter much. But when the wavelengths from the diaphragm get to be comparable to dimension then these angles all matter. So there is no one right answer. What Tinitus suggests is correct at the lower freqs, but its much more complicated than that at the upper freqs. Its comlicated enough that only a numerical sim could actually sort it out. The simplified lumped parameter approach that is usually applied to phase plug design is only approximate at the upper freqs.

Question: type of wavefront for conical /OS?

For any coordinate system there is a shape to the coordinates at any location. For a spherical waveguide this shape is always a sphere, but its radius varies. For OS, for example, it is flat at the origin and very nearly spherical at larger values of the "radial" coordinate.

Now if the wavefront does not match the shpe of the coordinate system at its "entrance" then it has to be "fit" to the boundary conditions by taking a sum of terms, or modes, such that this sum fits the boundary condition. When there is a perfect fit, this is the classical 1P concept. For a spherical waveguide only a spherical wavefront at the throat will be composed of a single "mode", namely a radially propagating spherical wave. Any other shape will require HOM (that loathsome term defining non-ideal wave propagation conditions) created right at the entrance and these HOM will then propagate to the mouth as waves that bounce off of the walls and travel a longer path length, thus delaying them in time.

Now a perfectly flat wavefront at the throat of an OS waveguide will not require any HOM to "fit" the boundary condition and a "nearly" pure 1P wave will propagate ending up as a spherical wave at the mouth. However, the math of the OS coordinate system requires what is usually called "leakage". By this I mean that the main mode continuously leaks into HOM as it propagates. This is very small effect at LFs, but gets more pronounced as the frequencies go up. At any rate the actual effects are quite complicated and mathematically very hard to compute.

A flat wavefront at the throat of a spherical waveguide however, will have HOM right from the start, at all frequencies. There will be HOMs in evidence at very LFs and they will increase in level all the way up to the HFs. A dome will be a "better" fit to a shperical waveguide than a flat wavefront, but still not ideal. There will be HOM present at all frequencies.

The least HOM will be generated by a flat wavefront - from a compression driver - into a OS waveguide. Regardless of what others here may say, this is the situation - like it or not.

Question: Opening angle of a horn

Narrower angles improve loading at LF and reduce the HOM, but cause massive growth in size as highlighted above.

The OS waveguide is defined for any angle - see the equation. The narrower the angle the lower in frequency it will load. You can't have wide angle and low frequency loading in a waveguide any more than you can in any other device. Think of it as "flare rate" if you have too (although this is technically incorrect). The higher the flare rate the higher the "cutoff" (again a technical misnomer).

As a general rule, a 60° waveguide would be -6dB at 60° (+-30°). If the waveguide mouth is too small, this will not hold true at the lower frequencies. It will narrow and then widen as the frequency goes down.

This is exactly where the problem is. Lets say you require a match of the polar responses at the crossover (to me this is essential, an absolute requirement that tends to drive everything else). If you make the mouth too small then you need an even bigger driver below to match this pattern. If you go lower in frequency to get back to the original coverage pattern then you don't have CD anymore as the pattern is wide, narrows then widens again. So you have to go up in frequency to get to the point where the pattern stabalizes to its design intent.

I might have stated this in a confusing manner. Its not that the lower angle reduces the HOM, it moves them higher in frequency, which is better because HOM are like modes in a room. They start out widely spaced and then get denser and denser as the frequency goes up. Thus the density of them will be lower for a given frequency band as the angle narrows.

While the tube has the fewest HOM what it has are the most delayed and they would tend to have the greatest gain. A coincidence effect is at work here. My paper on Waveguide Theory Revisted clearly shows this effect. The gain rises with narrow angles, its reactive component increases and hence its group delay will increase - all subjectively bad things since audibility depends on level and delay. One aspect gets greater (level and group delay) while the other gets smaller (density).

Now is there a sweat spot? Thats a very good question. I would guess that the answer is also quite complex and it would take some serious investigations to sort it out. Based on the physics I would also guess that 45° is either the best or the worst - the min or the max. Judging by the sound quality of the waveguides that I have heard, I would guess its the best compromise.

A very interesting question - one that I will have to think about. Maybe in addition to the OS being the optimal shape, 45° is the optimal angle. This later aspect would have been complete luck on my part as I had never thought about this before.

Question: Reason for axial holes?

The point that Earl makes is that it is precisely because the wavefront is a very good resemblance to spherical that complete phase cancellation can occur and produce a center minimum. =>
The point that Earl makes is that it is precisely because the wavefront is a very good resemblance to spherical that complete phase cancellation can occur and produce a center minimum.
=>
I already KNOW what causes the dip and I've posted it here a number of times. Its the mouth diffraction which adds out of phase from the direct sound at precisely one frequency when precisely on axis. An elliptical mouth will make this go away.

Question: horn opening angle vs amout of HOM?

If I understand correctly, the sharp transition angle at the throat is a major, if not the major factor in generating HOM's, It would make sense then, that the wider the angle, the more HOM's would be generated near the throat. Whether or not this is correct, my experience is that the foam in the throat is the most important.

The HOMs have higher "cut-in" frequencies with narrower angles and hence within the audio bandwidth, for a give throat size, as the angle decreses the HOM content decreases. But Mark is correct, building such a device is impractical. For a miriad of reasons 45° seems to be a sort of optimal angle.

Question: how was the contour derived

First I noticed that the profile of an OS wave guide is a hyperbola, at least in the throat and initial expansion regions before the profile is modified to blend smoothly into the baffle surface. This profile happens to be the that of flow or stream lines for the potential flow solution through an orifice.

It would seem that the OS wave guide is basically the right 1/2 of that picture with the profile set to a streamline that asymptotes to the correct exit angle (defining the profile hyperbola). So wouldn't the ideal diaphragm shape be that of an OS, i.e. the shape of a surface of constant velocity potential?
=>
Absolutly correct. And it was exactly your picture (In Skudryks Foundations of Acoustics) that led me to the OS waveguide and the entire theory. I, like you, realized that this figure showed an exact solution of the problem in 3 dimensions and did not require the assumptions that limited Horn Theory. I later found that such a solution had already been performed by Freehaufer at MIT under the direction of Phillip Morse (not at all surprising!). He used a unique instrument called a "Differential Analyzer" which could solve differential equations mechanically. Today the numerical solutions have been done in Numerical Recipes and other texts. I used those solution techniques when I did the second paper identifying the HOMs and how they would be generated and propagate.

The geometry can be taken back to the origin, in which case the ideal source is a flat piston. An actual flat piston was used in the first OS waveguide experiments that proved that this geometry was indeed ideal for a flat piston. When the source is not flat other approachs must be use and the best source that I know of for that is in fact Chapter 6 of Audio Transducers.
=>
Thanks for the conformation, Earl. One other thing, agreed that at the origin the potential surface is flat. However, The thing that has me wondering is that a surface of constant velocity potential is not (necessarily) a surface of constant velocity. This is certainly the case here since the potential surfaces are elliptical. Similarly, a surface of constant velocity would not be orthogonal to the stream lines, thus the velocity vector would not be perpendicular a surface of constant velocity. The point being that it would seem (nearly) impossible to actually construct a diaphragm which had pistonic motion and had the velocity vectors correctly aligned with the flow lines. Is this the origin of at least some of the HOMs? Also, are there other sources of HOMs, for example, arising at higher frequency due to relative comparison of wave length to WG dimensions, sort of like the HOM that are present in a constant area duct went the wave length is on the order of the duct cross sectional dimensions or smaller?
=>
The wave equation that I use is in terms of pressure not velocity potential, hence the two variables are pressure and pressure gradient which is velocity. I suppose that it could be done in a velocity potential form (that is more common in CFD than acoustics), but that's not the way I did it nor have I seen that done. The fact that the pressure gradient at the origin is a function of the distance from the axis is described in my past writings which shows how a uniform velocity will still generate HOMs. This function also changes with frequency and so it is highly unlikely that BOTH the velocity contour and the change in frequency could be achieved simultaneously. However, I do show how one could surpress any desired mode by velocity shaping, and one of my patents describes how to do this.

The nonorthogonal nature of the velocity to the pressure is the reason for the HOMs that are generated within the device. And all devices where the wavefront shape is changing will have these. The HOMs that are created by a mismatch of the source to the duct are another issue. Those HOMs are created by the boundary conditions at the throat.

The HOM do have a cut-in phenomina exacty like waves in a duct and they exist, just like in the duct, depending on how the source is aligned with the duct. A duct fed with a flat plane wave will not have HOMs even if it is possible for them to exist. Everything depends on how the source fits onto the duct.

Question: how was the contour derived

In any analysis it is possible to break a wavefront down in terms of an infinite set of plane waves, this is precisely what is done in Optics in "K-Space". So the motion of a wavefront in any duct could be done as a sum of plane waves, or it could also be done as a sum of spherical waves - this is BEM. But these techniques obscure what is happening precisely since any wavefront requires an infinite number of waves to define. Better, in a known waveguide contour, is to solve for the set of allowable wavefronts which then requires only one or two or at most a hlf dozen waves to be defined. This later set is precisely what the OS waveguide solves for. That minimum set of waves that can exactly define any wavefront allowed in a waveguide. This set is important because it inherently seperates the main wave from the HOM which are dispersive and allows for us to maximize the main one and minimize the unwanted HOM waves. No other technique allows for this simplification.

Question: how was the contour derived

For minimum diffraction from waveguide walls, there should be zero second derivative of the walls, or straight sides, which equates to a conical waveguide. However, as Earl has pointed out, a spherical wavefront must be applied to the throat of a conical waveguide from the source to meet his objectives.
There being only planar wavefront sources, curved sides have to be used to morph the planar wavefront of a driver at the throat into a spherical wavefront as the wavefront approaches the conical asymptotes of the waveguide. For a wall curved to have the smoothest transition to minimize diffraction (HOMS) from the throat to the conical asymptote, it has to have the minimum second dervative (or CHANGE in slope) possible, that being the OS curvature.

Question: how was the contour derived

The OS contour is based on a solution to the full 3 dimensional wave equation. The wavefronts are, by definition, iso-phase, but that does not mean parallel in common understanding. They are parallel in the sense that they travel along and perpendicular to a set of orthogonal coordinates, which in a non-Euclidean geometry sense is the deffinition of parrallel.

The OS solutions are analytically exact in full three dimensions and as such they are the only true solution of the wave equation for a flared contour, all others being only approximations, some better than others.

Question: minimize the second derivative???

Try this: If you take a stiff wire and hold it at 0 degrees and at the throat radius and then take the other end and hold it at the design angle, say 45 degrees, it will form the shape of an OS waveguide. The wires stiffness will minimize the second derivative (the change in slope).
Geddes

Question: further optimization of the os contout?

Dr. Geddes, can You specify any kind of magnitude for the second derivative (SD) anywhere on the "curve" before it gets bad ?
I mean, I calculated some WG´s for fun and their SD´s and compared with other curves that i managed to get below the initial SD of the OS, but not to zero.
=>
You can get the 2nd derivative to go to zero (it does for the OS at larger x) but starts out lower than the OS for a function that is piecewise continuous, but not for a continuous function. That is the sense at which the OS is optimum and a catenoid. I too was able to get a lower 2nd D with piece-wise continuos function and I have to admit that this idea is interesting. I'm not willing to give away the functions just yet.

I do suspect that the 2nd D integrated form zero to the waveguides end will be lowest for the OS for any continuous function, but the math behind its derivation does not allow for piecewise continuos functions. Hence allowing them might result in some interesting ideas.

I did this work after reading a post by Jean-Michel who compared his 2nd D to the OS, and, as suspected, it was misleading. His curves do not start with zero slope and so they won't match the throat of the driver and this will generate HOMs but will not appear in his curves. It is possible to set the slope of the early exponential section to match the 6 degrees of the driver, but this yields a very slowly flaring device which would need to be exceedingly long to yield a mouth of any appreciable diameter. Thus he adds a wide flare at the mouth which adds lots of 2nd D and diffraction. If you make an exponential horn that initially flares as fast as an OS then it has much higher 2nd D all the way along its length.

SO the question becomes, is diffraction better at the throat than at the mouth? I suggest that with the foam that I use the answer is "absolutely" because the HOM created at the throat get absorbed while the ones at the mouth do not. If you postpone the diffraction to the mouth, then there is no way to minimize it. Bottom line to me; what I am currently doing still appears to be the ideal as all other approachs will yield more diffraction at the listener.

Question: illustrating how the OS curve generates the least diffraction

Think of it this way. Diffraction is created by a change in the slope of the waveguides bounding surface. The amount of difraction therefor depends on the second derivative of this bounding curve. If you calculate that curve that has to start out straight (to match the driver, assuming a plane wave) and end up at some angle and has the smallest second derivative, you will find that the curve that minimizes this function is in fact the OS contour. Its a simple double integration. Its called a catenoid, and is also the minimum surface area connecting a flat disk to another larger flat disk. where the initial slope at the smaller disk is given.
A straight sided cone has, of course, zero second derivative, but it doesn't meet the condition of zero initial slope. So a cone is minimum, but only if you can somehow get a spherical wavefront at its opening. That would not be easy even if at all possible.
Geddes

Question: nearfield vs farfield

As to the nearfield to farfield, the nearfield does determine the farfield, but not everything seen in the nearfield will propagate to the farfield. For example do a run of a piston or cone in a baffle in the near field and then compare it to the farfield. You will see that many details seen in the nearfield do not propagate to the far field. The farfield tends to be an average of the nearfield where the details as washed out. Much of the fine structure that you show for the OS waveguide will NOT propagate to the farfield. These effects are called Evanescent Waves and they are "complex" (in the mathematical sense) and have wavefunctions that are real exponentials - they dampen out exponentially with propagation.

=> All my measurements of direct radiating drivers show near field response much soother than far field measurements. I even took the time gradually increase distance from probably 1mm or so and out until I see no significant change in SPL trend. I'd be willing to post my data if you will do the same.
=> You are talking about frequency smoothness and I am talking about spatial smoothness. Yes, in general, the nearfield is smoother in frequency and rougher spatially than the far field. If you get a spatially smooth nearfield then you are doing something wrong.

Question: how to get an elliptical contour?

So elliptical waveguide molds may be cost prohibitive at the moment, but out of curiosity what does a proper one look like? Is it as simple as calculating the contour repeatedly while varying the exit angle as you rotate around from the 90 degree side to say a 40 degree top? And should the distance from throat to exit stay the same meaning these would need a curved fronted cabinet?

Quite perceptive. You are absolutely correct about the contour.

People have long thought that the cross sectional area of the elliptical had to satay the same as it went from round to an ellipse. This ends up with something that doesn't work very well. The idea of maintaining the cross section comes from Horn theory since it deals with areas. But in my theory only the contour shape matters, the cross section is irrelavent. So one just uses the same equation at each rotational angle, but with a different theta.

It would be better to have each path length a different length as this will minimize the coherence of a standing wave either in the mouth or along the length. This is why I would not expect an elliptical waveguide to exhibit an axial hole in the response.

There is a lot to like about the elliptical, except its cost. It would easily double the cost of the current waveguides which are already expensive. An Abbey with an elliptical 90 x 40 would be quite attractive to me.

Question: how does the foam work?

The foam will be a very gradual smooth (nothing resonant) roll-off of the high end and almost nothing below about 6-7 kHz. Its major effect will be on waves that pass through the foam several times, like reflections, and waves that travel longer distances, like HOMs. This is why it works so well. Its effect upon the main wave is minimal, but its effect on the aberations (distortions if you like) will be major. A real nice net improvement overall.

I early subjective tests of the foam (with no EQ), some, like myself, liked the change, but others didn't like the HF loss. When I corrected the HF loss with EQ everyone liked the effect.

The foam used by Earl Geddes possess most probably a fractal nature... Relevant lectures should be find using "poroacoustics" as keyword =>
The answer is most certainly yes, I have and continue to view the reproduction problem as one of a fractal nature, although I use the term entropy as its more appropriate. I have hypothesized that the ear likes a certain amount of entropy in its signals because whenever we increase the entropy the ear seems to like it.

Jean-Micheal is correct in that the foam is a fractal structure and will impart a small degree of entropy to the signal (in addition to its absorptive characteristics). This may be why it sounds so good when in fact the measurements indicate only a small change.

I had tried to interest several driver manufacturers in making an increased entropy loudspeaker along the lines of a fractal structure, but alas, to those guys "no change is a good change".

Also when replying to the question about the modelisation of the foam Earl indicated that the BEM should should not be modelised as an acoustically resistive material but as a reactive material (complex impedance). So there is some good reason to think that a simple reduction of the SPL is not the only modification due to the foam=> What I said was that the wave velocity had to be complex (which means real and imaginary) NOT that the impedance had to be complex (which can also be real AND imaginary). When something is complex it need not be purely reactive, it can have a resistive part and in fact the resistive part can be the major part. And what if the foam does have a reactive part? So what? Its obvious from the data that that its not doing anything resonant.

Jean-Michel seems intent on not listening and confusing the discussion. I have never contradicted the claim that the HOM take a longer path, thats what they do. Hence the trace-velocity along the axis (as it is called in acoustics) is slower (longer path) although the wavefront moves at the same speed as all other waves. There is no dispersion (variable wave velocity) only diffraction (different wavefronts are created).

There was a study presented at AES on the directivity effect from the HOM by Gotfried Bueler at Achen. He concluded that they have a minimal effect on directivity and I would agree. But thats different than saying that they have a minimal effect on audibility. The two things are completely different.

Question: what type of foam is used

30 ppi reticulated polyurethane foam

Protectair is a completely open cell (reticulated) Polyether Poyurethane foam available in a range of 20 to 35 ppi.

Protectair 30 PPI

Engineered for filtration media in hydrolytically unstable envionments such as air conditioners and humidifiers.

I spent about a year trying out different materials and densities, even using "batting" as you suggest. I did not post or present these results, I doubt that I ever will. But its not as if I didn't do a lot of work to find the best option available. What I use is the best that I found.

The results are always a tradeoff between absorption of the main wave at HF and absorption of the HOM. Too much HF absorption is a problem in a CD waveguide because the eficiency already falls at -6 dB/oct and one cannot afford too much more loss that that. In the Summa the output of the DE250 at 10kHz is exactly the same as that of the woofer. Any more loss from the foam would require the woofer to be lowered in efficiency. One could argue that in the Summa there is so much output to spare that this is not an issue, but thats another discussion.

Question: where to place the foam?

If I understand correctly, the sharp transition angle at the throat is a major, if not the major factor in generating HOM's, It would make sense then, that the wider the angle, the more HOM's would be generated near the throat. Whether or not this is correct, my experience is that the foam in the throat is the most important. In my case, I put a foam plug all the way to the phase plug of a TAD 2001, and a little more than a third of the way out of a Unity (conical) horn. Without the plug, some harshness and slight "honk" is perceptible. With the short plug, it's gone. I don't notice a difference between that and the horn filled out to the front edge. More sensitive ears than mine may differ.

Question: how much loss is introduced by the foam?

There is no measurable loss at the low end from the foam, only the upper range at about 1-2 dB at 10 kHz.

Question: improved smoothness in CD ala LeCleach horn???

I would like to point out that it is the very large radius of Jean-Michels horn mouth that results in such smooth response. In theory all infinite horns and waveguides have perfectly smooth responses, there is nothing to cause otherwise. Its the mouth diffraction that is the biggest cuprite of non-ideal behavior. So if you want an improved OS waveguide, then adapt Jean-Michel's mouth profile to the OS. I use what could only be considered a minimu mouth radi simply because I have to be practical. I'm making reasonable cost and size systems and very large mouth radi are not feasible, nor is a free standing waveguide.
Geddes
Many "non Le Cléac'h" horns using smooth profiles exist. Not all of them deliver smooth response, smooth pulse response, smooth polar, smooth pressure fields. Smoothness of the profiles is not sufficient by itself in delivering smoothness of the results. Others characteristics are required one being a correct expansion law of the wavefronts area in order to obtain a very low reflection coefficient at all frequencies in the usable frequency range of the horn.
JMLC

Question: diffraction at horn mouth?

All horns or waveguides need low diffraction mouths not just OS. Based on my research and the results from the waveguides that I have tested it all comes down to the radius. The larger the better. I don't see how a varying radius would be of any benefit. If I had, I would have used one.

My response implies that there is no theoretical reason to vary the radius. I don't guess at things. Unless someone can give me some rational reason why varying the radius would work better why should I try it? Because someone elses "gut feeling or engineering intuition" says that I should test it? My speakers have been designed on strictly objective scientific principles. This approach has worked very well for me thus far and I see no reason to change now.

Diffraction depends only on the rate of change of the slope - in mathematical terms it is proportional to the second derivative of the contour. Thus the larger the radius the smaller the second derivative and the less the diffraction. There is no location dependence.

As a side issue, if you take the second derivative of the OS conyour you will see that it is a constant and once this constant is set the diffraction is set. There is no other contour which has this feature, which is why the OS is a catenoid with the minimum second derivative for a fixed starting slope. A cone has no second derivative, it vanishes, but its starting slope is fixed, it cannot be made to match the slope of the drivers exit. If it did then there would be no diffraction at all - in the waveguide that is, there could still be HOMs generated at the source end. An exponential has an increasing second derivative and hence it diffracts continuously along the device at an ever greater rate.

Question: vibration in horn?

The vibration modes as illustrated are more related to solid(ian) propagation. That's means that if the coil reproduces a pulse, a part of the energy will be transmitted through the body of the compression driver then through the material constituting the horn's wall. If the horn vibrate and specially near the mouth, then there will be some intermodulation between the waves emitted by the wall and the waves propagating through the air inside the horn.

Also in the worst case, we can see on the pulse response of a driver monted on a horn a small pulse arriving before the main pulse. This is because speed of sound through most solids is something around 10 times the speed of sound.

Most often people gives a small tapp with their finger to the mouth of a given horn and say "oh! it rrings like a bell". This is faintly useful. It is far more realistic to tapp the horn near the throat.

Now imagine you vibrate the horn near its throat, the vibration will strain (= induce deformation in) the wall of the horn. This is a very small deformation. The graphs I gave show that deformation multiplied by a large factor in order to see clearly how the wall of the horn vibrate for a given mode. A horn having a very thin wall will lead to larger modal vibrations than a horn having a thick wall but the shape of the deformation will be the same.
JMLC

Question: Reasons for Dip(s)?

In an axisymmetric waveguide there is a diffraction at the baffle edge. At some frequency the direct wave and diffracted wave will be exactly out of phase because of path length differences. This will cause a cancellation, but only directly on-axis. As the waveguide gets smaller, the diffraction gets greater and the path length difference gets smaller. So the hole moves higher in frequency and deeper in level. The data completely substantiates this, but Mr. Declerq did not seem to agree.

Question: Reasons for Dip(s)?

First, the wavefront at the mouth has to be coherent and virtually perfectly spherical. Any deviation from this will smear the dip making it invisible. For example a small bump along the sides of the waveguide to "trip up" the wave will virtually elliminate the dip. Of course this makes for more HOM and potentially a worse sounding device, but the dip is gone.

Second the dip depends on the size of the mouth. As the mouth gets larger the dip will get shallower and lower in frequency, until at a large enough mouth it will disappear even for a circular mouth. For a 90 x 90 waveguide this would be about 18-20".

An elliptical waveguide will have a smeared dip that will be shallower and broader depending on eccentricity but will still move lower and shallower with size.

Question: Equalizing CD (polar maps)?

CD requires a +6 dB/oct correction for any diaphragm operating above resonance since they will have a velocity falling at -6db/Oct. In its most general sense that what CD means, the axial response and the power/polar response track one another. No other type of source does this.
Geddes
Any horn or waveguide that provides constant directivity (or even close) will need 6dB/octave augmention above the driver's mass rolloff point, usually around 3-4kHz for a 1" compression driver.
Wayne

Question: Equalizing CD (polar maps)?

I design for the best average response up to about 30°, which is not really the total power response
Geddes
Excuse me if this is proprietary, but do you just a bunch of measurements up to 30º off axis, average that and use the average curve for designing your crossover?
Reply: Thats the basic idea, but it is done more by eye I suppose than a precise mathematical algorithm.

Question: to CD or not to CD? That's the question…

Your implication that CD does not provide for precise imaging is contradicted by the science and the data - the subjective response of the auditions of my designs. Subjective response is the only way to quantify imaging since it has no quantifiable measure, although we do know what impacts imaging. People universally feel that my CD waveguides have very precise imaging. Imaging is a direct field controlled phenomina and when the Very Early Reflections are properly handled through directivity then the imaging will be good if the direct field is flat and smooth. But later reflections have a strong impact on coloration because the ear integrates the sound from all locations when it evaluates the tone color and spatiousness. For this reason CD is required for low coloration, and directivity is required for good imaging. Your designs achieve directivity so they will image well, but they do not provide a flat reverberant field so they will not give a good sense of color or room spatiousness.
Geddes

Question: relationship between the intensity of diffracted wave and source intensity, edge shape and frequency

1) the dffraction is proportional to sounce intensity - it is linear
2) The sharper the edge the more diffraction. It is actually proportional to the 2nd derivative of the surface. A "knife edge" therefor is a worst case.
3) The higher the frequency for a given edge the greater the diffraction intensity.
Geddes

Question: Cut-off of a CD-horn

Cut-off is a concept from Horn Theory (a weak one IMO) that does not exist in Waveguide Theory. There is a coupling of the input impedance with the angle and this is something like a "cutoff" but no real "cutoff" exists. And of course this phenomina doesn't exist in the real world either. No horn or waveguide exhibits a true cutoff - only in theory. All devices transmit all frequencies, with more of less gain. Cut-off is a meaningless concept to me.
Geddes

Question: Length of a horn

Length is not a big factor and "no", length and loading have nothing to do with one another. A device can be too short and it won't have enough length to control the waveshaping function, but once its "long enough" the only thing that matters is the mouth size and edge treatment. If the mouth is large enough and the edge treated correctly then the length is completely arbirtary and the driver would have no idea if it was long or short.
Geddes

Question: Origin of vertical nulls

The vertical nulls are all about path length differences between woofer and tweeter.

At the center of the forward lobe (which may or may NOT be along the baffle normal), the woofer and tweeter are exactly in phase with one another in the crossover band. Pick a frequency in the overlap band, and there will be a position (hopefully out in front of the speaker) where there is zero phase difference between woofer and tweeter. This marks the center of the forward lobe. In actuality, this position shifts a little through the overlap band because of the phase shift of the crossover. However, given a reasonably narrow overlap, the center of the forward lobe can be thought of as a fixed position.

Now move upwards from this position and you are getting further away from the woofer, closer to the tweeter. This can be expressed in percentages of a cycle. At high frequencies, it may even represent a multi-cycle shift, but for speakers like we are talking about, the crossover is carefully designed so the shift is less than a cycle through a fairly large angle. When the movement puts the listener closer to the tweeter by exactly 1/2 wavelength, that's the point where the upper null forms. Moving downward, when you get to the place where the woofer is closer to the listener than the tweeter by exatly 1/2 wavelength, the lower null forms.

What you'll see when you do measurements is the sound is very good through a range of +/- 1/4 wavelength. When the difference gets closer to about 1/3 wavelength on either side, the nulls begin to form. Between about 1/3 and 1/2 wavelength, the null grows from a shallow dip to a strong notch. Then as you travel further going from 1/2 to 2/3 wavelength, the notch subsides, becoming more of a dip and then relaxing. By the time you've moved all the way out to a full cycle shift, there is no notch, in fact, the wavefront is (generally) constructive again. However, this is a full cycle shift, and you would probably consider that to be "dirty". Sine waves will combine constructively even if passing through a full cycle shift, but aperiodic waves won't.
Wayne

Question: Conical Horns?

It is true that you need a spherical wavefront to drive a conical corn without HOM’s. There are a few 1 inch drivers that can do this well up to about 60 –70 degrees, like a BMS 4550 and others of that geometry. Also, several 18 sound compression drivers are very good. At least one 1.4 inch drivers can on narrower horns.
You can see the real problem if you plot out the path lengths in the phase plug, they generally produce a converging wave front at summation.

The up side is, a conical horn if driven properly has (down to some size angle relationship) constant directivity and truly emulates a patch of spherical radiation.

Question: Conical Horns vs Waveguides

ALL waveguides are conical at larger radi just as ALL wavefronts are spherical at larger radi - not matter how they start out. Thus even though an OS appears to differ from a cone only at the throat - and this is indeed true - it is exactly at the throat that it matters the most. Hence what appears to be a small difference is in fact a huge difference.

Question: Paint?

Look for Acrylic Enamel (rust-o-leum) and use an Epoxy primer. They have water based and regular. I use water based, but its not as good, just a lot easier to use. I buy bulk acrylic from an outlet in Madison Wis. and mix my own colors to the clear. I make all my own hobby paint as this stuff is ridiculously expensive - about $1 / ml.

Question: Speaker placement

Toole and I both recommend damping the wall behind the speakers. If this wall is very well damped then the speakers can be quite close. But in general away from the front wall is a good idea if this is possible. But better the speakers sit against a wall than the listeners. How much performance degradation s impossible to say since the situation is so room specific.

I try to keep the space arround the speakers a s "clutter" free as possible. Anything, TV's, bookcases, etc. can cause diffraction even if they are at the sides of the speakers. If the speakers are toed-in as I suggest then nothing should be between the speakers as this will diffract the sound. How much? Hard to say - I only know that the more I tried to eliminate these difractions the better the image got.

I always keep the wall behind the listeners highly reflective as this increases the spaciousness effect quite a bit. But you need to be fairly far from this back wall if you are not to have early reflections from it at the listeners position.

No, from "THE ROLE OF THE INITIALTIME DELAY GAP IN THE ACOUSTIC DESIGN OF CONTROL ROOMS FOR RECORDING OR REINFORCEMENT SYSTEMS BY DON DAVIS" (AES paper 1547):

"Tho acoustic goal in the control room is to insure that its ITD is made
longer than the ITD of the studio. This allows the ITD of the studio
to be reproduced acoustically in the control room, unmasked by early
control room reflections. Thus a series of options opens. [...]
The most obvious option is to make the entire front half of the control
room non-reflective at the geometric acoustic frequencies. Then, by
proper spacing of the listener relative to the rear half of the room,
which is made reflective and diffuse, control room ITDs of from 5 or
less msec to in some cases over 40 msec are viable alternatives."

Question: subs?

One of the key distinctions in what I recommend and the previous researchers is the use of overlaped mains and subs. This is unique but I find (and Wayne seems to agree) that this works well. It adds more "subs" to the mix, while perhaps requiring a little more capabilities from the mains. In my systems this is never an issue, in others it will be.

Psychoactoustics

Question: how does localization wok

Localization is done by three mechanisms ITD, ILD and pinnae. ITD, or phase localization is primarily a mid frequency localization, while both Intensity (level) difference and pinnae mechanisms are primarily High frequency mechanisms. Most research that I have seen claim ITD produces phase ambiguity above a certain frequency, let’s say 1800Hz, and then the brain used ILD and pinnae for localization. If the brain shuts off ITD functionality, wouldn’t just moving a horn’s crossover over this frequency reduce HOM perception?

There is no question that people misuse the precidence effect for delays less than about 2 ms. Blauert and Kuttruff both make a clear distinction in delay effects at about this time scale as the same rules don't apply above and below. Blauert discusses these very early delays, and Kutruff just ignores them because they don;t happen in concert halls. But he is very clear that the rules for larger delays don't apply at these short time intervals.
(localization => imaging)

Question: Importance of directivity index

Dr. Geddes I still did not clearly understand why is necessary to hold DI above 8 in 1KHz+ range. I assume it is from the reason to supress reflections in lateral direction. But on the other hand in B.Moore's book An Introduction to Psychology of Hearing there is written that the precedence effect and other mechanisms in auditory system supress those reflection very efectively. Maybe that is why we are still able localise source in higly reverbant fields.
I'm asking because in case of very early reflections (under 1ms) and image location for source level and time diferences you are both in agreement. Or I'm misssing something?
Second question which bother me is a imbalance between monopole lows and highly directive heights. Without acoustical treatment in room it should have a deteriorative effect on timbre, isn't it?

Very good questions.

There is no clear cut widely agreed upon answer - I differ with Toole who differs with Moore, etc. etc. I think the bottom line here is that the precidence effect simply defines the predominate perceived direction of the signal when reflections exist. The readily available literature does not imply anything about the sound quality impacts of those very same reflections - Blauert makes this point very strongly. Toole quoted me unavailable (internal) research that says that very early reflections "are not a problem" yet I can quote my own work which shows a highly significant impact on sound quality from very early reflections and diffraction. So take your pick!! Although

I do think that Moore would agree that the presidence effect says nothing about sound quality.

In my work I have found that for good sound quality, i.e. accuracy of reproduction, influences from the speakers or the room should be minimized for at least several ms after the direct sound arrival. After 10 ms., the more energy the better. A quick thought will show that the narrower the directivity the more the particular influence of early refelctions from the room is minimized and the greater the later reflected energy is maximized by proper choice of room absorption (i.e. almost nothing behind the listener, but dead behind the speakers - Yea, I know that this is the exact opposit of many peoples recomendations).

Now virtually everyone appears to agree that as the frequency goes lower, our ability to discriminate group delay, reflections, source location, etc. drops as the frequency drops. This means that the problems associated with early reflections and diffraction will not be as significant at lower frequencies as they are at higher frequencies. We can argue about whether the directivity needs to be controlled down to 500 Hz. or 350 Hz. or 200 Hz., but I don't think that there can be an arguement that one needs to control the directivity down to the lowest frequencies.

The choice of how far down to control the directivity then becomes one of practicality. It's not a completely impossible task to do it to 500 Hz., although its tough enough, 350 Hz. and things are getting pretty big. 200 Hz. well thats going towards the impossible and meaningless in a small room.

I have paper designs that can do this control to 350 Hz. (but this system would be impractical to build) and the Summa does it to about 500 Hz. Most speakers aren't really constant directivity so its impossible to compare them to one that actually is CD.

Finally to you last question - as the directivity of the system goes down, the sound absorption should go up (which again is the exact opposit of what is usually done) such that a very large amount of sound absorption is required below about 100 Hz. Again, difficult, but not impossible - I do it in all my designs.

Then put about three subs randomly arround the room and you will reach Nirvana! With the right CD or DVD of course. Junk is junk on any system, although I've heard some systems that were so bad that they even made junk CDs sound better!!

Question: Imaging vs spatiousness

If clarity must be traded for spaciousness, then perhaps what we really need is ambient surround channels, rather than compromised two-loudspeaker stereo.
Newell
I have recently been having some side E-mails with a guy in Belgium and Jens Blauert about very early reflections, imaging and spatiuosness. Dr. Blauert does believe that there is a tradeoff that has to be made between spatiousness and image as regards the <10-15 ms reflections. That one can't have it both ways. This would explain why a listener group, like Floyd's, who put spaciuosness very high in the "preference scale" would find every early reflections "preferable", even if this same

thing degrades "imaging". It may well be that these two things will always have to be traded off and that there will never be a "one size fits all". Its hard to say.
Geddes

Question: Localization vs spatiousness

Spaciousness is a primarily quality of the room (influenced by speaker directivity), while localization is primarily a quality of the speaker (influenced by the rooms early reflections). One cannot correct the room with the speaker so the only choice is to design the speaker for good localization and then design the room for good spaciousness. When properly done both qualities can be achieved, but only if BOTH designs are done correctly.