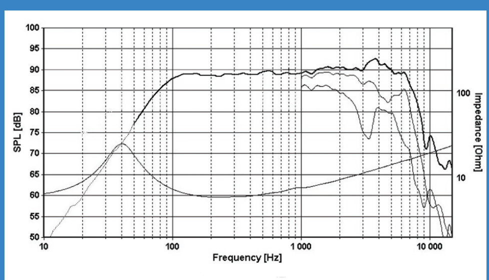
Designing Speakers

Part 8 - Loudspeaker Measurements

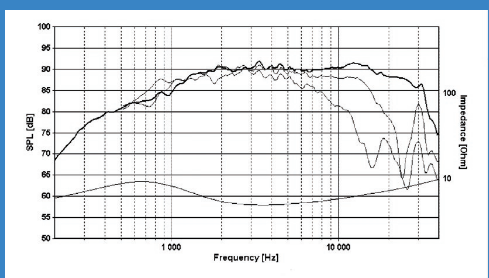
Peter Comeau explains why you need to measure your crossover design, and how to go about it.

If you have ever looked at Noel and Adam's measurements in loudspeaker reviews in this publication you might have wondered exactly what relevance the loudspeaker response has to the subjective comments. Indeed it is not always the speakers that have a 'ruler flat' frequency response that do best.

In fact frequency response is only one arbiter of speaker quality. For the reviewer it is just a check that the speaker designer or manufacturer is not totally incompetent and that there are no 'glaring' faults. The experienced user, like Noel, can spot potential flaws and tonal inaccuracies but, in general, it is not easy to 'see' the character of a speaker from its response. For the designer, however, everything changes. With a little bit of insight you can quickly learn to 'see' potential problems from the individual drive unit response and impedance curves. How can this be?



SEAS H1217 driver frequency response at 0, 30 and 60 degrees off axis, plus impedance



SEAS H1189 tweeter frequency response at 0, 30 and 60 degrees off axis, plus impedance

To start with you can look at the individual 'raw' driver response in your chosen cabinet. If you measure the response over a range of axes, say 30 degrees horizontal and 5 degrees vertical, as well as the 'on axis' response, you can easily start to judge how the speaker is going to 'sound' in the room and how well it will integrate with other units. Combine this with the impedance plot and you'll get a good idea of how simple, or complex, a crossover design is going to be.

WHERE TO START

Don't let any of this preamble put you off! If you haven't understood what I mean by '30 degrees horizontal' don't worry, I am going to lead you through it. We looked, last month, at the basics of crossover design and why you needed to be able to measure the response of the drivers in your cabinet. That doesn't stop you selecting the best drive units for the job at the outset from the manufacturer's data.

What we are looking for are drive units that are going to make our job easier. That way we are more likely to end up with a crossover that is going to be relatively simple, not upset any amplifiers by incorporating crazy impedance phase angles, and is going to be simpler to fine tune when we start listening.

One driver I started work with last month is the SEAS H1217. This has exactly the sort of response we need from a bass/midrange unit for a two-way system. The response on axis, that is directly from in front of the driver, is nice and smooth over a very wide bandwidth. Its working response extends up to 7kHz, which is going to make crossing over at anything up to 3kHz so much easier because the crossover slope beyond 3kHz is going to be fairly linear and so will integrate much easier with a variety of treble units.

But, as we have discussed, on axis response is not everything. To obtain some idea of the true character of this drive unit, look at the frequency response off axis. To take this response the microphone is moved horizontally by, say, 30 degrees. This shows you two things. One is the response that a listener who is sitting away from the optimum listening axis will receive. The other is the frequency range that is going to be reflected from the side walls and floor and ceiling of the room.

It is these reflections that largely determine the 'character' of a speaker in a room, so always check them, especially when you are designing your crossover. So look at the 30 degree response (the thinner line below the main response trace). Here you can see that the response is still smooth up to 3kHz and beyond. The trace below is at 60 degrees and the output here has a dip at 3.5kHz due to interference from the outer parts of the cone, but this shouldn't concern us much at this stage.

Similarly the impedance plot (the very bottom trace, which is plotted against the index on the right hand side of the graph) shows only a small perturbation around 800-1kHz. This indicates that the cone breakup (the points at which the cone stops behaving as a true piston and only the inner sections of the cone are producing high frequency output) is nice and smooth. This augers well for a clear, low coloration, midrange performance.

Applying the same criteria to a treble unit we can look at a dome unit from the same stable, the SEAS H1189. Again what we are looking for here is a smooth response to way below the crossover frequency, and no peaks in the upper treble area. H1189 extends smoothly beyond 20kHz and has a low fundamental resonance at 550Hz.

Once again the off-axis responses tell us something more. At 30 degrees the response holds up very well to beyond 12kHz, which will help maintain the 'character' of our final speaker design and deliver good quality treble to listeners sitting off the 'sweet spot'. The peak at 30kHz indicates that this is the primary dome breakup mode – well beyond the limit of audibility thankfully.

The impedance plot shows the well damped fundamental resonance (this unit has ferrofluid damping and a rear chamber). Overall this unit should work nicely with a second order or third order crossover, we hope.

MEASURE FOR MEASURE

Of course you are free to select your own drivers from the extensive range that is out there – just bear in mind the criteria I have outlined when doing so. As far as cone and dome materials are concerned it is easier to avoid the exotic unless you have plenty of experience dealing with their occasional difficulties. For example ultra stiff cones, such as aluminium, will have high frequency resonances which you will have to suppress with a notch filter. With a bit of experience you could incorporate this into your crossover, but it is a problem you could do without when just starting on the speaker design road if you ask me.

So let us get the measurements underway. As I said last month there are now several systems that you can use to generate a frequency response from your speakers. The old method was to take the speaker into an anechoic (without reflections) chamber and sweep a slowly increasing frequency range into the speaker and plot the output from a calibrated microphone on a graph paper.

The modern method is to feed a pulse into the speaker which contains every frequency we want to measure and to use a computer to analyse the output of the calibrated microphone. Now there are several advantages to this method. The major one for the home designer is that we no longer specifically require an anechoic chamber. Because we are just measuring everything from a pulse we can select only the pulse and ignore any room reflections which will arrive later at the microphone.

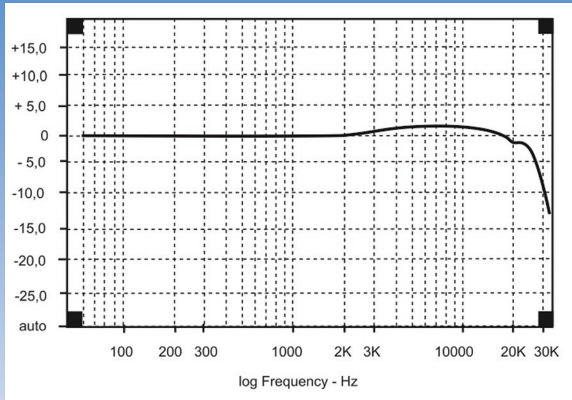
We mentioned last month that the great grand daddy of MLS is the MLSSA program. This has become an industry standard for speaker designers but is sadly outdated by its DOS interface and requirement for a full length ISA card which you can't put into modern computers. Another frequently used system that features an MLS option is CLIO, and this is the system that Noel and Adam use for loudspeaker measurements in the magazine. As an all-in-one solution, complete with calibrated microphone, CLIO is very versatile and will get you up and running quickly, but it is not cheap.

So to get you going I suggest a 'home made' system that you can use with any computer. Proper, calibrated, measurement microphones are expensive, but for our purposes we don't need them. All we need is something that has a flat frequency response. Now you might think that all high quality mics are like that, but they are not. Mics often have a 'presence boost' in the midband to highlight vocals, and rarely have an extended response at bass or treble ends of the spectrum.

One relatively low cost mic that will suffice for our purposes is the Behringer ECM8000. You can pick this up for £35 from a variety of sellers, such as Digital Village (www.dv247.com) for example. This is a condenser type microphone which is ruler flat from 20Hz to 3kHz, has a mild 1dB lift to 20kHz and comes complete with a mic stand adapter.

Being a condenser type it does require power (15v to 48v) but this

Behringer's ECM8000 Microphone - an ideal microphone for a budget measurement system.



Behringer ECM8000 microphone frequency response

can be supplied by any microphone preamp offering what the PRO boys call 'phantom power'. Again there's a variety of mic mixers that will give you this, but I've found an all-in-one soundcard and mic preamp that interfaces very nicely with the ECM8000.

SOUND CARD

For MLS measurements you need to be able to use your computer to generate the MLS pulse and receive the input from the microphone at the same time. So your computer has to have a soundcard that is 'full duplex', in other words it can handle input and output simultaneously. If your soundcard doesn't do that then I have a solution for you.

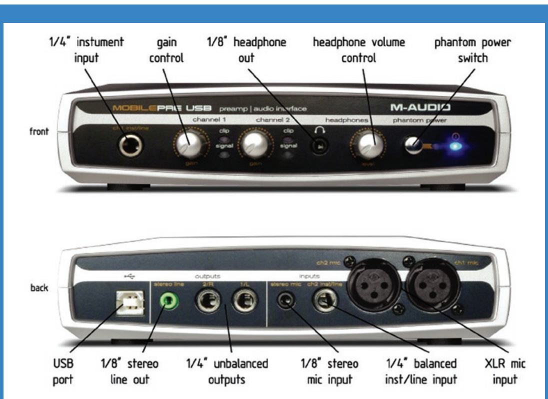
The M-Audio MobilePre USB (around £89) is an all-in-one mic preamp and soundcard that plugs straight into your USB port, so it can be used with a laptop too. It provides phantom power on its XLR mic inputs, so an XLR cable will connect straight to the ECM8000 and away you go!

Now all we need is some software to generate the MLS pulse and a method of analysing the output from the microphone. There are a few systems available to download:

(1) WinMLS. Easy to use and you can try it free for 30 days – but you need the Level 4 PRO version to generate an MLS output, so not much use for the amateur!
http://www.winmls.com

(2) ARTA. Complete, and complex, program developed by Ivo Mateljan. A demo is available and the program is £69 to purchase if you like it.
http://www.shareit.com/programs.html?productid=300069215

(3) Speaker Workshop. Clunky interface but it is free! Good support via the forum, and you'll need it until you've mastered its strange system.
http://www.speakerworkshop.com/SW/Download.htm



M-Audio MobilePre USB microphone preamplifier/soundcard has a good array of features at a budget price

The Speaker Workshop forum also has a discussion about using the Panasonic electret microphone for measurement. If you are DIY capable this is an excellent way of cutting the microphone costs right down as there is a preamp circuit detailed for this too.

Of course you will also need to connect the soundcard output to your amplifier. A good soundcard will produce a line level output that you can connect to the Aux input of your amplifier in order to drive the speaker. You want to avoid a soundcard that only has a speaker output as its internal amplifier can often be very noisy.

Whichever software you choose you will find that it uses one channel of its input for measurement and one for reference. The reference signal is an attenuated voltage fed from the output of your amplifier. The purpose of this is to make sure that any noise or frequency response errors in your soundcard and amplifier do not affect the measurement. By comparing the output of your amplifier (the reference signal) to the output of the microphone the software can dial out any errors caused by the soundcard or amplifier output.

This type of self calibration is essential if you are to see a clean signal. Although the software is working digitally, the test signal is converted to analogue through your soundcard output, amplifier and speakers as well as your microphone and mic amplifier. Any errors in the chain will screw up your measurements and give you a false reading. Both ARTA and Speaker Workshop show you how to build a suitable attenuator and wire up the inputs to obtain the reference signal.

Note that this reference signal cannot take account of your microphone performance. In particular, in order to obtain accurate sound pressure level readings you will need to enter the manufacturer's calibration figures into the software. This only matters if you want to see the true sensitivity of your speakers. For crossover design we don't need that level of accuracy.

FROM A DISTANCE

So, armed with our microphone and software let's look at what we can do. I'm going to use MLSSA here simply because I use it all the time, but ARTA provides a straightforward interface with easy to create graphs that will give you exactly the same results.

First off we need to measure our drive unit in an enclosure for both frequency response and impedance. The standard distance for measuring speakers is at 1m distance between the front baffle of the speaker and the front of the microphone. This is to allow for a fair degree of integration from the drive units, to integrate the time of arrival of the individual wavefronts from the treble unit and bass unit, although it would be closer to the reality of most listener's typical positions if speaker measurements were taken at 2m.

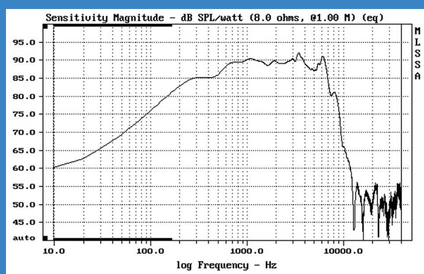
But the problem with increasing the microphone distance is that you bring the time of arrival of the MLS pulse from the speaker and reflections from nearby boundaries closer together making it more difficult to 'gate' the measurement window. So I would recommend that you stick to around 0.5m or less when measuring in a room. This will give you accurate measurements of the individual drive units, and is usually good enough for working out the starting point of a crossover.

For frequency response we want to look at the on-axis response, that is directly in front of the speaker at the height that your ears are at when you are seated. With a speaker design with treble unit at the top this usually means on the treble unit axis. For a tall speaker you could be pointing the mic mid-way between the drive units. The point of choosing this axis is to find a place which will allow you to align your crossover design to give an even response. Once you've chosen your axis of measurement stick exactly to it for all future on-axis measurements.

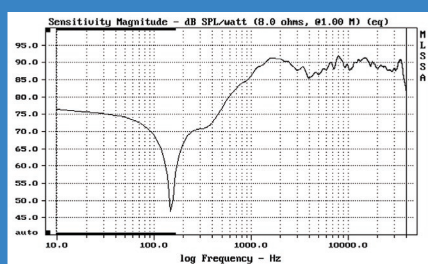
So, find your ideal on-axis position and run a frequency response. You will almost certainly see a deviation from the drive unit manufacturer's response. For example, as outlined last month, you'll see the baffle step for the bass driver. Also, presuming that your mic is pointing at the treble unit, the bass unit treble output will fall off a little earlier than expected. This is because you are actually measuring the bass unit off its central axis and its high frequency output will be curtailed.

Turning to the treble unit you'll see, here, the effects of the baffle edges and any impediment to the baffle itself. Because a dome treble unit sprays high frequencies in all directions any surface edge, ridge or hollow on the baffle will cause an early reflection which can interfere with the main output of the treble unit. This is why speaker grilles are generally considered a 'bad thing' when listening to speakers. Interestingly, the biggest hollow on the baffle is the bass unit cone itself, and its surround and chassis cause reflections very close to the treble unit. This is one good reason to recess the chassis of the bass unit into the front baffle.

So you will probably see a dip in the treble unit response which, again, makes it look rather less smooth than the manufacturer's spec. At this stage I wouldn't worry about these deviations. You could spend days trying to cure them and often the



With H1217 in an enclosure with a 20cm wide baffle the baffle step becomes clearly visible. Ignore the response below 200Hz as the gating is not wide enough to show any accuracy down here, as shown by the black bars.



H1189 in the enclosure doesn't look much like the manufacturer's spec, does it? The dip between 3 – 6kHz is partly due to the proximity of the cabinet edges and also due to the bass unit cone below it. Again ignore the trace below 200Hz.

FINE-TUNING THE MLS MEASUREMENT

To measure without an anechoic chamber we need to be able to separate out the pulse received from the speaker and the reflections from the room. Graph 1 shows the first 20mS of sound received by the microphone. You can clearly see the initial pulse from the speaker, then the output settles down to a smooth (silent) period, followed by the microphone picking up the small reflections from the room boundaries numbered 1 to 5. The first, no.1, reflection is that from the floor. This is because the floor is the closest boundary to the drive units. After that the reflections will arrive from the next nearest of the walls or ceiling and, finally, from the wall behind the speaker or the microphone.

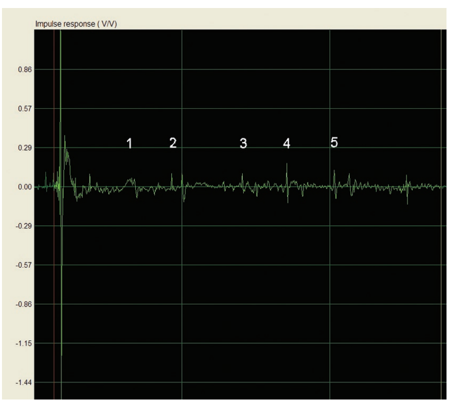
Now, just by moving the cursor on the time display, we can tell the computer only to select the part of the pulse which falls between the cursor marks. We call this 'gating'. The cursor marks form a 'gate' where the open area between the cursor marks is the period we want to measure.

Although the pulse, called a Maximum Length Sequence (MLS), contains every frequency we want to measure, the system is only accurate if the time period enclosed by the gate is long enough to capture the time taken for the lowest frequency to form a half wave. The reason for this is that the MLS, although looking to us like a pulse, is actually considered periodic by the receiving software. So, unless the gate is long enough, there will be errors in the low frequency measurement.

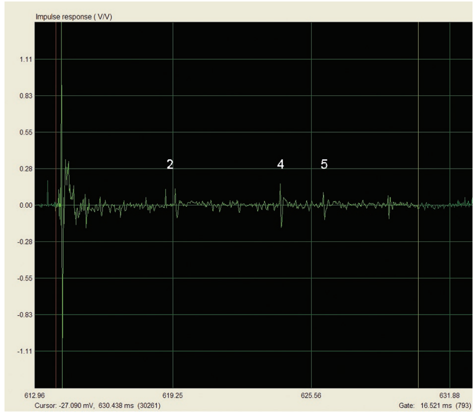
In fact we don't have to worry about this too much. Opening the gate too wide will bring in the room reflections, so don't do that. Keep the measurement gated to remove the reflections and put up with a curtailed LF response instead. The bass end of the graph doesn't really tell you what the speaker is doing in the room anyway and, as we want to concentrate on the crossover area, we really only need to look at the midrange and treble performance.

The more seasoned MLS user will mathematically splice a measurement taken close to the drive unit to one taken at 1m. This effectively gives a good indication of the bass performance in free air as the close mic technique (usually around 1cm from the driver) avoids all reflections. It doesn't tell you how the speaker works in a room, though, so the amateur designer is better off using his ears. This close mic technique can be used successfully for subwoofer design though, so it is worth mentioning.

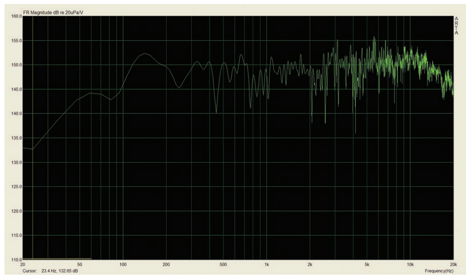
There is one trick that you can use, however, to open the gate wider. Position the mic and speaker at opposite ends of a sofa, place lots of soft cushions or pillows on it, and you should 'lose' the floor reflection. That will be as close as you can get to 'anechoic' in your living room!



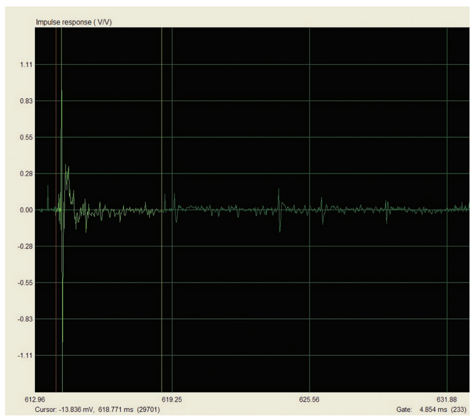
Graph 1 - This is the impulse response of a speaker in a room. The initial pulse is the speaker output, followed by boundary reflections 1 to 5.



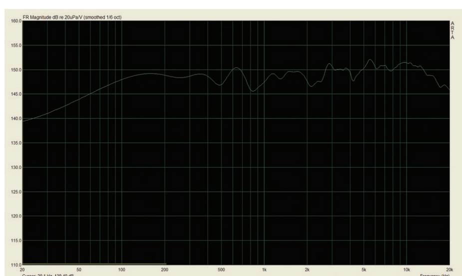
Graph 2 - The little trick of placing microphone and speaker either end of a sofa removes the ground reflections and subdues some of the wall reflections.



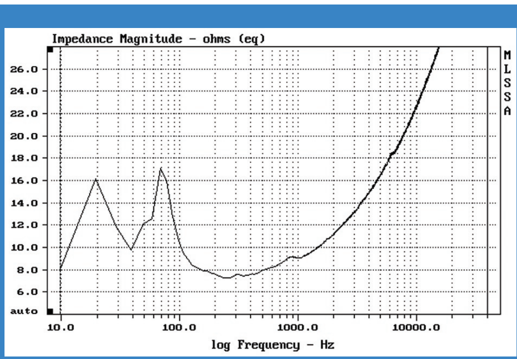
Graph 3 - With the time gate open fairly wide these remaining reflections still cause a very jagged looking frequency response.



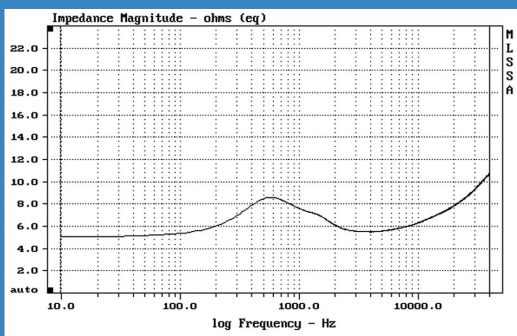
Graph 4 - We use the cursor markers to limit (gate) the analysis and 'lose' the reflections.



Graph 5 - This gives us a usable frequency response but with an inaccurate bass content (shown by the yellow bar at the base of the graph).



The impedance plot of H1217 shows the typical double hump of a bass reflex tuned to 40Hz. Ignore the triangular look of the peaks which is because the number of points plotted down here doesn't provide high accuracy. The wrinkle in the trace at 300Hz indicates an enclosure resonance (well damped). Another wrinkle at 800 Hz is part of the drive unit characteristic. We know this because it can be seen when the impedance is measured outside the enclosure.



Impedance of the H1189 treble unit shows the broad, well damped fundamental resonance of the unit due to ferrofluid damping and the rear enclosure used on this unit.

cure is worse, acoustically, than the problem.

THE IMPEDANCE QUESTION

Now let's turn to impedance. These traces should be fairly straightforward and, with a little bit of knowledge, will help you see what is going on inside the cabinet. Each trace will have a hump at the bottom end of the drive unit's bandwidth which shows the fundamental system resonance Fs. For a bass unit in a sealed box you will see one major peak. For a reflex system you will see two major peaks, and here it is the valley between the peaks which shows the box tuning frequency. A transmission line should look a little bit like a reflex box but with the bottom peak so low it may well be off the end of your graph. We'll go into these in more detail when we get to the actual design stage.

The height and breadth of these peaks will depend on the amount of damping included in the system. This is important. If the peak is low, smooth and broad (low Q) you may have overdamped the system and have too 'lean' a bass performance.

If the peak is high, narrow and sharp (high Q) the system is probably underdamped and will sound boomy or resonant.

If you see any major deviations in the impedance trace higher up the frequency scale then these indicate other problems. What you have to remember is that the impedance trace is taken from the back EMF of the drive unit – the voltage fed back from the drive unit to your computer. So the drive unit is acting as a microphone itself.

A bass cone, for example, will pick up any resonances inside the enclosure, and these will reflect as deviations in the impedance trace. So, for a floorstanding cabinet, you will probably see a small peak in the impedance trace between 100 – 250Hz. This will be the effect of the primary standing wave set up from the bottom to the top of the cabinet.

Higher up in frequency you may see another peak in the trace. This may be the reflection from the back to the front of the cabinet. If the sound from the back of the cone hits the cabinet right behind the bass unit it will bounce back onto the cone if you didn't put any damping material in this area.

You can also look for wiggles in the trace which are due to panel resonances in your enclosure. If you have an unbraced box these can be quite severe and you'll need to apply panel damping compound to the interior of any resonant panels (you can hear these just as well by tapping the enclosure panels with your knuckles).

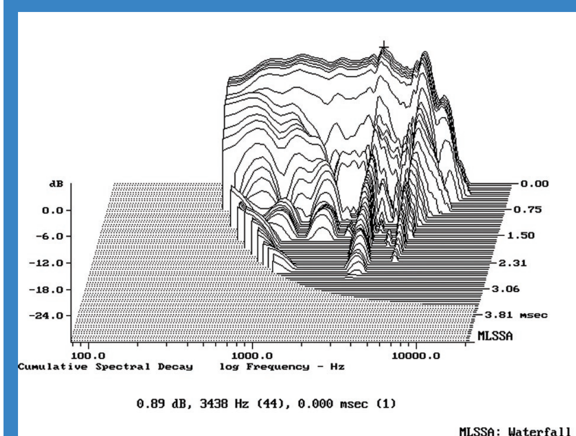
For a treble unit you'll probably just see its Fs. If the treble unit has ferrofluid in its voice coil gap then this fundamental resonance will be broad and smooth indicating the effect of the ferrofluid damping. But have a look and see if there are any wiggles in the trace at high frequencies. These can indicate resonances in, or behind, the dome. Not much you can do about these, of course, so they are only of interest as to the 'quality' of the drive unit design.

So, now that you've made a start on your measurements and, hopefully, are feeling more confident in your use of the software, what's next? Well next month you will be happy to know that we are starting our first loudspeaker design project and, as the easiest way to teach is by example, I'll show you how to run through a speaker design from start to finish.

You can get a clearer picture of all these resonances if you run a 'waterfall' graph. Some of the software I have recommended provide this facility, otherwise called a Delayed Resonance display. What the software actually does is to run an FFT analysis on different parts of the output of the speaker over time. If you like it does its own 'gating' and looks at the speaker output after the pulse has finished as well as the pulse itself.

So what you see in the waterfall graph is the initial frequency response of the unit, displayed at the back, and then response traces taken at increasing time intervals stacked up in front of it. A perfect speaker would only have output at the initial pulse but, of course, all speakers hang on to some energy (drive units as well as cabinets) and this is output after the pulse has finished.

These 'delayed resonances' are shown as ripples in the response which spread forward as the time from the initial pulse increases. Now it is going to be difficult for you to separate out the resonances from the enclosure and the resonances from the room. You will need to play around with the way you use the gate markers in order to avoid room resonances creeping into the time window for the waterfall trace and this takes a lot of experiment and experience to make any sense of what is really going on in the speaker itself. But do have a go – just don't get worried by the results at this stage.



'Waterfall' delayed resonance graph of H1217 in its enclosure. The ridges at 3500Hz and 6000Hz correspond to peaks in the frequency response graph and indicate minor resonances in the drive unit.