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Technical support advisory 051018b

Calibration hints for the calibration wizard

The purpose of this document is to provide supplemental information to the 'PDC installer's manual' on calibration wizards and templates.

Requirements:-

- Version 2.0 DEQX Calibration software

Start the DEQX Calibration software, open an existing project, and use the 'Calibrate Speakers' wizard to open an existing calibration template or create a new one.

SPEAKER CORRECTION LIMIT WINDOWS

In the calibration page there are a few different limit windows; the 'anechoic' window (red), the freq-boost-cut 'limit' window (blue) and the phase coherence limits (top left of the page).

As you are probably aware, the 'anechoic' window is used for truncating the speaker measurement so room effects do not corrupt your speaker measurement (i.e. exclude room reflections). The next tab to the right (Smoothed measurements) shows the truncated measurement WITH the effects of smoothing (situated at the top of the page).

Smoothing is used to suppress minor reflections and irregularities in the speaker measurement. If you have ever seen or used 1/3 octave spectrum analyzers you will know that each band automatically averages (they're analog); the DEQX being digital has incredible accuracy and therefore can show us way more info than we really need. I personally find it useful to also think of the smoothing also as a sort of 'analog' control.

The anechoic truncation and smoothing interplay with each other quite a bit, though the anechoic truncation can make the biggest difference. It's for this reason; I prefer to set my smoothing to be 100%, and then experiment with different anechoic truncation lengths (if required, more on this later). I have never used smoothing values below 80% to create correction filters, ever, even when measuring speakers in an anechoic chamber! I do however find it very useful to change the smoothing to many different settings (0-100%) to 'visually' see what the speaker is doing; the point being that when you create speaker correction filters, I recommend smoothing is above 80%.

After changing the anechoic truncation length (Anechoic tab), you can view the smoothed measurements tab. If you look closely, you will see the response curve change; this is an excellent way to see the effects of different anechoic truncations on your measurement.

The longer the anechoic window time is, the more accurate the measurement will be at the lower octaves; but of course you don't want to include room reflections so this is where you must take a little care.

As you probably read in the manual, ideally you want to measure the speaker in the middle of the room to maximize the time between the initial signal emitted from the speaker, and the reflected signal from the room; this is to give us the longest anechoic window possible (and therefore more lower frequency information) within our environment.

If you were to set the anechoic windowing at 2ms▲ after the speaker impulse, you would notice that the smoothed measurement would probably roll off after about 500Hz▲; the window is simply not long enough to accurately process the lower frequencies, so the software does a gentle roll off at the bottom end.



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Sometimes, some reflections can be included in the speaker measurement (i.e. not truncated out) with minimal effect (compared to reflections being truncated out). When this happens we may get a nice little bonus; more accuracy at the low end. This why sometimes "...I prefer to set my smoothing to be 100%, and then experiment with different anechoic truncation lengths."

Some speakers may be too large and bulky to move to the centre of the room for measurement. In this case read the technical support advisory 'Dealing with measurement reflections', and set the smoothing to 100%. To get some idea of what the low end of our speaker is doing, set the anechoic window to be 20ms after the speaker impulse and look at the smoothed measurement. You will see significant low frequency information, albeit with the detrimental effects of the room reflections. Visually note the response (you could even do a 'print screen' and paste it into MS Paint) and mentally average 'out' the effects of the reflections ('brain' smoothing at 1000%). Return to the anechoic window and set it to exclude all reflections, return to the smoothed measurement and compare it against the previous measurement. You have just viewed the two extremes, and by experimenting with different anechoic truncation lengths you should be able to get a response that includes a reflection(s), extending low frequency information while having a minimal effect on the true response of the speaker.

Once you have decided on your preferred anechoic truncation position, it's time to choose the correction range in the 'Limit' tab. Simply move the blue limits box to encapsulate your speakers response curve. Anything within the blue box will be corrected FLAT.

You do not have to choose the number of filters to use or what the frequency bands should be, the limit page does away with all of that; you simply specify the area you wish to correct and let the software worry about the rest. The PDC 2.6 as 4000 filter taps PER OUTPUT (up to 12000 filter taps per left/right channel), so you shouldn't worry about it running out of resources!

Remember, your truncated anechoic measurement normally defines the lowest frequency to correct down to. There is not much point in trying to correct down to 20Hz when the measurement is only accurate down to 250Hz. Don't worry too much if you can't set the left frequency limit down to the 'ideal' 20Hz, the low frequencies of the speaker interact significantly with the listening environment so when we do the room correction we will correct the bottom end *at the same time!*

With the frequency range and cut/boost limits defined it's time to create your correction filter, and then do a 'verification measurement'. This will upload your filter to the PDC and then measure the corrected frequency response (the microphone should not have moved and be in the same position as the original measurement).

The nice thing about this 'verification' measurement is it is NOT an anechoic measurement. i.e. You get full resolution at lower frequencies however room reflections will cause the measurement to not appear as flat as it really is. Assuming you are happy with the correction filter it is time to upload it to the PDC, move your speakers to the listening position and then do a room measurement(s) and correction. This is all documented rather well in the manual.



Frequency (f) = 1/ Period (t)

This formula is used as an indication of the relationship between truncation window and frequency; it is not intended as a defining limit indicating that speaker correction should not be attempted below this frequency.