

Overview on Discussed Technical Domains

- A **realisation concept** for free field measurement test setup
- **Experience:** Disappointments the author had - and anybody can run into, partly due to not knowing well enough the anechoic chambers qualities and limitations; partly due to the technical chamber concept, which is not sufficient for high precision calibration work.
- **Some theory** to understand the topic directly above more in detail.
- A simple method to get out of a frequency response IR (that got measured including some noise and distortion) **a truly noise- and distortion-less IR** - as required for the stimulus generation. This replacement IR does not include all very fine frequency response ups and downs - just that does not play a role, since during the iteration work this gets corrected. (Free field measured IR are «never» as noise free as ones made in anechoic chambers).
- How to care about the **environmental noise** during the iteration measurements.
- **The problems occurring out of free field measurements**
- **Microphones also have frequency response errors**; so the idea is close, to correct them also by the template for the iteration work. Out of microphone test the conclusion got made, that - supplier dependent - the calibration chart frequency responses are not reliable enough to get used in the template; that one can end up with less errors, by just using the microphone as it is without caring about such details.
- **Some technical details on anechoic chamber design**, so that you can better judge its suitability for precision calibration measurements.
- **An overview on the weak sides and restrictions of free field measurement.**

The author made the experience, that in the end he gets better precision with free field measurements than with the chambers he found and could rent. But FF measurements are not always possible.

Free Field Frequency Response Measurements

This document explains how quality Free Field (FF) Frequency Response IR measurements can get made with moderate investments / efforts.

To make FF measurements is far away from something new; has been done in the past often and has led to very interesting discussions on result reliability. What is new in the process here is, that other than when using an anechoic chamber, the IR from the measurements gets replaced by an IR free of distortions and environmental noise.

If you lose a tooth due to infection and your dentist replaces it by an implantation, then you have new tooth free of illness with same shape and you should not feel any difference. Exact that gets done to the IR in the process described here; from the FF IR only the shape (the 1/3 octave FR) gets reused, its new base is a IR freshly generated by DIRAC using EDIT -> GENERATE -> dirac impulse. By that you have shed of all noise during the measurements and the speakers non-linear distortions - giving the later generated stimulus the best possible MTF qualities.

This document covers also issues regarding noise and other disturbances which occur during the measurements - with practical suggestions.

Deja Vue

The author of this papers went through a *deja vue* experience when looking for anechoic chambers to rent and building up his test configuration: About 25 years ago (or so) he had the job - when the CE certification of products became in Switzerland a must for exportation into EU - the job of managing over 1300 HVAC products in that respect: Screening all products for CE / EMC-compliance, making the triage for OK / rework of the design / or phase-out the products etc. Starting up R&D project for reworking the designs etc.

It these days many Swiss companies where in a similar situation; there was a rush on the few EMC labs to reserve days and month for testing. Then the author needed to decide between small and large anechoic chamber, ones only with electrically conducting foam wedges - and other using ferrite, some between the wedges and the concrete walls of the building.

Similar experiences got made in the last year looking for STI anechoic chamber opportunities:

- Most chambers are small, the larger ones difficult to get into
- Their shapes are partly very different (which plays a role, both for EMC and sound - just the rules are very different)
- Also different absorption technologies got used: EMC / CISPR chamber with or without ferrite; in Anechoic Chambers traditional **wedges** or with the newer **BCA concept / technology** (*broadband compact absorbers*), also using a second layer - a steel sheet in the back of the absorption material for converting low frequency noise energy first into vibration and then into thermal energy.

So to have a free field measuring option was a welcome idea to not get lost in details between chamber types; answer what real FF measurement would change; but also hoping the weather and noise level would be cooperative...

Back on the Road - Again.

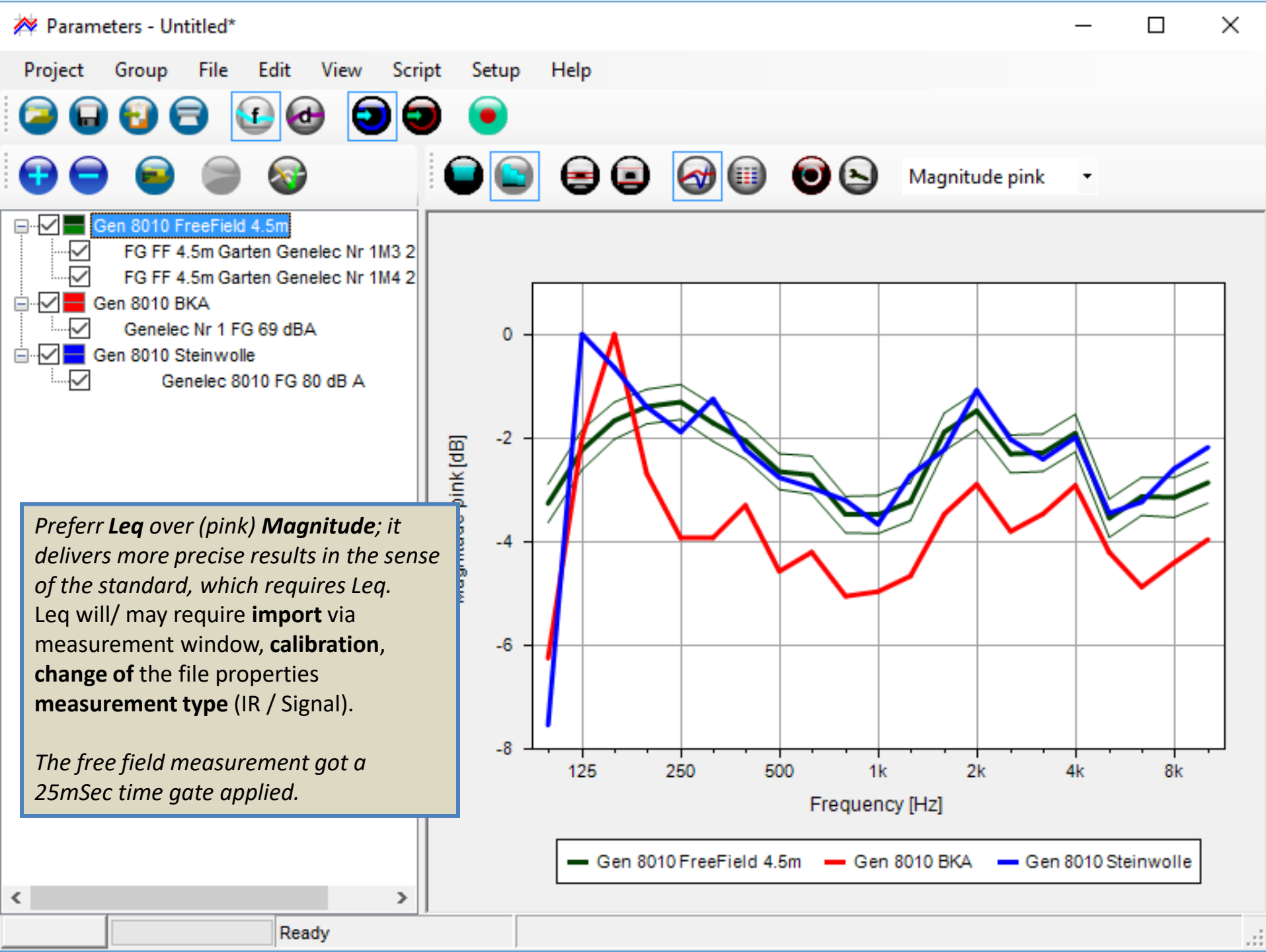
Now with Anechoic Chambers the author went through a similar process of learning as with the EMC chambers - and is still on the way. In those days he spent many days in EMC anechoic chambers trying to better understand the high frequency hocus pocus. One day, when sitting behind the chambers operator, he detected, that the plot drawn on the screen and the display of the antenna signal (an EMC emission test) did not fit; the offset was not constant. The operator answered to the corresponding questions: Ask the boss.

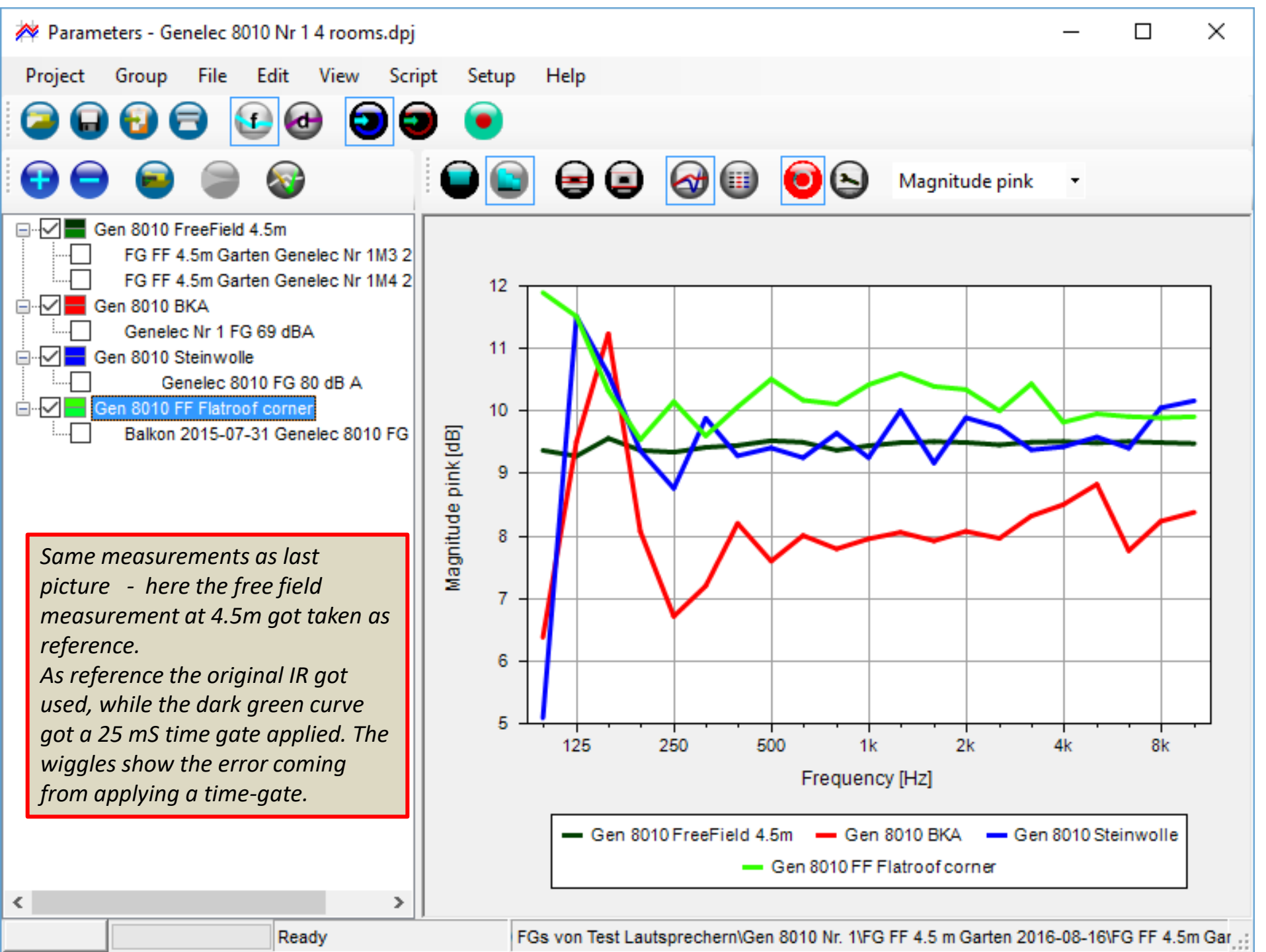
The story was, that the analyser had a internal profile of the chambers uncertainties and added corresponding to the frequency range the chambers uncertainties profile. The bosses answer was: «So you found out - I will need to tell you the whole story». (Today, different processes get used).

Anechoic chamber customers - to the authors experience so far - do not get informed about the chambers uncertainties, if they do not ask - and not all chambers have made and passed measurements as according to ISO 3745 - which is for calibration work not really demanding enough.

On the next slide you see the frequency response of copy Nr. 1 of the Genelec 8010 speakers:

- The blue curve was measured in a 2.8 x 2.8 x 2.8 m anechoic chamber having a well done stone wool absorption of undefined depth, guessed to be about 50 cm deep.
- The red curve got measured in an modern BCA technology chamber, with about 50% **more volume** and slightly less height. Here something went wrong - and that is the content of the following slides
- The dark green curve got measured with the free field process and equipment described in this article.
- The pale green curve (second picture only) comes from a FF measurement made on a flat roof corner looking out into the green.





Every Anechoic Chamber has limits - these you need to dodge.

The next picture shows the setup used in a modern BCA chamber. Since many speakers got tested in varying situations, a setup got chosen, where the large door could get fully opened, without the risk of it touching the spread legs of the stand / tripod.

Brown as arrows the microphone <> speaker setup is shown. The author was not aware, the speaker was already outside of the “virtual globe of class 1 precision ” range, as shown in red. This globe has its broadest area on half the heights of the room. But with the microphones lower down, the area gets smaller - illustrated by the blue circle. Further in the lower half of the room the class 1 area shrinks, if the industrial grade floor grid does not get covered by absorption material.

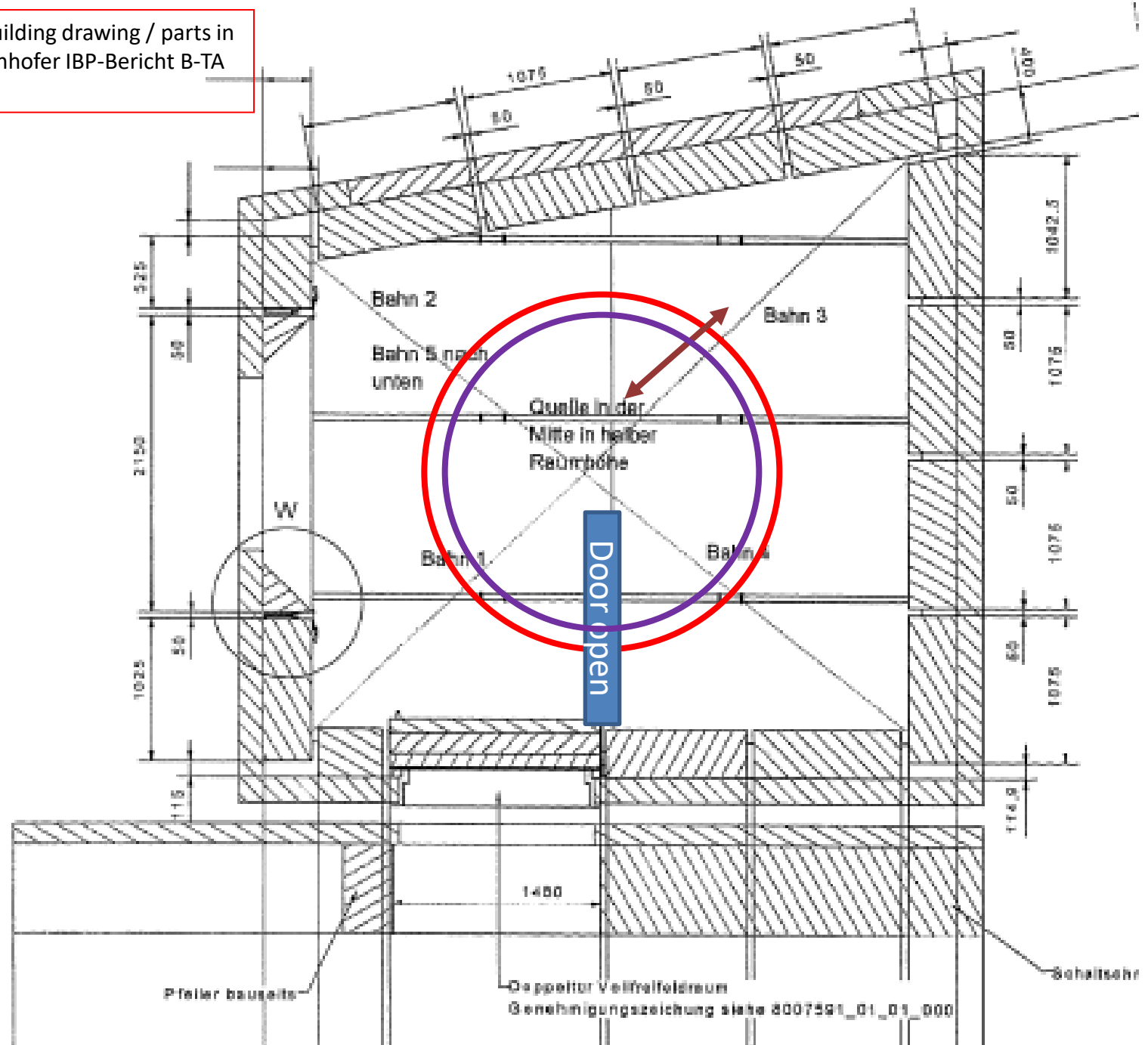
BCA have a excellent low end and mid range absorption curve - except at about 160 Hz it drops according to specs to about 0.8; the problem area of the measurements here. Further BCA's absorption rate drops beyond about 2000 Hz.

The anechoic chamber got tested by the Fraunhofer Institute for ISO 3745 compliance, and the chamber got found to be class 1 within a «virtual globe» of 1.3 meters, with the source in the middle of the chamber. The report shows also, that - especially with low frequencies, outside the red circle the error goes up in extreme situations to 5 dB.

Many these things the author was then, when these measurements got made, not enough aware of.

Thanks to have done also the Free Field measurements, the author got confronted with the situation, that the measurement he made in the anechoic chamber, are by far not good enough for STI speaker calibration.

Source of building drawing / parts in grey: «Fraunhofer IBP-Bericht B-TA 4/2004»



Every Anechoic Chamber has limits - these you need to dodge (III).

To get precise measurements in an anechoic chamber, several aspects must be cared about:

- Ask for all information available of the chamber. If no class 1 compliance test got done, make up your mind, if this is the right place to go.
- If the anechoic chamber has large walking grid areas, ask for information on ready available absorption material to be laid in the area between microphone and loudspeaker on to the walking grid - and maybe elsewhere.
- Maybe reduce the distance between speaker and microphone to 50 cm. By that you introduce new error sources; but can reduced error impacts of the room. The certification data could be helpful for this decision.
- Plan the layout for your measurements; check the compliance report for the area you want to use if the tolerance values are small enough: ISO 3745 has larger tolerances than STI allows:
 - Below 630 Hz and above 6300 Hz: ± 1.5 dB; otherwise 1 dB. From the STI frequency response tolerance of 1 dB you need to reserve a part for weather sensitivity / drift and aging.
- If possible, make a free field test as described here, to get more certainty that the result should be fine.

Planning the outdoor measurement:

- The critical issue is a good place to find. Far away from village centres sounds nice, but one needs electricity and one must be far away from roads when there is continuous traffic, because when one car follows another on a road, the sound source is of line type which carries over very long distances, since its sound level decays only with $1/d$ (d = distance), while with low traffic the decay is of second order ($1/d^2$).
- But if you don't find a place with enough SNR, e.g. due to very unsteady noise profile, on-going short noise peaks like a scooter speeding up, then quality FF measurements are not possible. That the noise is stable is of high importance; the noise measured phase in DIRAC's intermittent mode is several seconds later, and is then no reliable information. Only your ear (or a SLM where you sit) can say, if the noise during the signal and noise measurement phase changed or was stable. Do several measurements to be sure, the situation is stable - and sit at a place, when your ear is closer / opener to the environmental noise than the microphone - e.g. by being 10 or more meters nearer to the noise than the microphone, where the sound out of the loudspeaker will not cover the noise coming to your ears (SLM) receives.
- Houses have a positive effect: They keep low noise like from traffic fairly well away, if houses surround a small garden area large enough and free of noise of the people living there.
- Time of day plays a huge role - and the day of the week also. And one may not want to disturb people sleeping in the houses with open windows. What I found out, is that planes can have a positive aspect: Between 06:00 and about 07:30 the planes on the way to Zurich airport some 35 km away have a periodic noise free time, when the earlier plane has more or less landed, but the next must still wait some minutes. In such a time the noise one makes measuring the speaker is lower than the noise of the plane before and later - so the measurement signals won't disturb people nearby.
- Use 50 cm distance between microphone and speaker for better SNR (2-way speakers rather more).

Measuring the Frequency Response - use the special version of the iteration template

The frequency response gets measured in a microphone stand of at least about 4.5m heights with no reflecting surfaces nearer that 5 meters. If using a garden, make sure not tree leaves come into this area. The measurement its self uses Pink MLS in the intermittent mode; the Leq results gets copied form the PARAMETER dialog (PROJECT -> COPY table) to the «FR of speaker» TAB of the specialized template – putting the cursor in the yellow field (A3) and pasting the data (here dummy data to show how the noise gets taken into account):

Initial Frequency Response of loudspeaker

<My Speaker name date>

Leq [dB]	Speaker	Noise	Result
xxx			
Frequency [Hz]	Ch.1 Avg	Ch.2 Avg	dB
25	15.00	10	13.35
31.5	20.00	15	18.35
40	25.00	15	24.54
50	30.00	20	29.54
63	35.00	20	34.86
80	35.00	20	34.86
100	40.00	30	39.54
125	40.00	35	38.35
160	45.00	35	44.54
200	50.00	35	49.86
250	50.00	35	49.86
315	50.00	35	49.86
400	50.00	35	49.86
500	50.00	40	49.54
630	50.00	45	48.35
800	50.00	45	48.35
1000	50.00	45	48.35
1250	50.00	35	49.86
1600	50.00	30	49.96
2000	50.00	25	49.99
2500	50.00	20	50.00
3150	50.00	20	50.00
4000	50.00	20	50.00
5000	50.00	20	50.00
6300	55.00	20	55.00
8000	57.00	20	57.00
10000	59.00	20	59.00
12500	51.00	20	51.00
16000	45.00	20	44.99
20000	30.00	20	29.54

Measurement must have made using in intermittent mode and Pink MLS of the signal length the stimulus shall have !

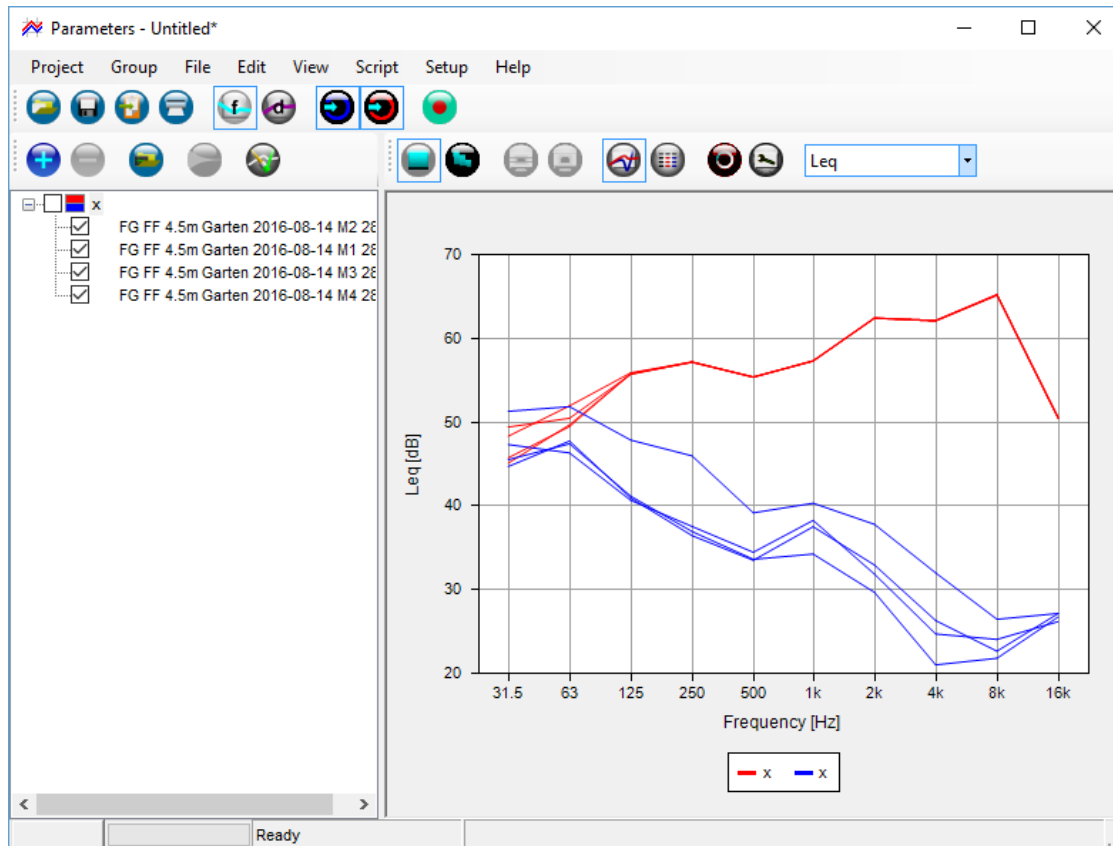


Important: The frequency response must get measured at the signal level the speaker shall gett used for measurements.

Some Results (I):

The picture below shows the result using a Fostex 6301B loudspeaker; a product often used for STI measurements. As you see: When the SNR breaks away (< 125 Hz), the frequency response result gets negatively disturb in the Leq level by the noise. To compensate for this, the template has a special version as shown in the last slide.

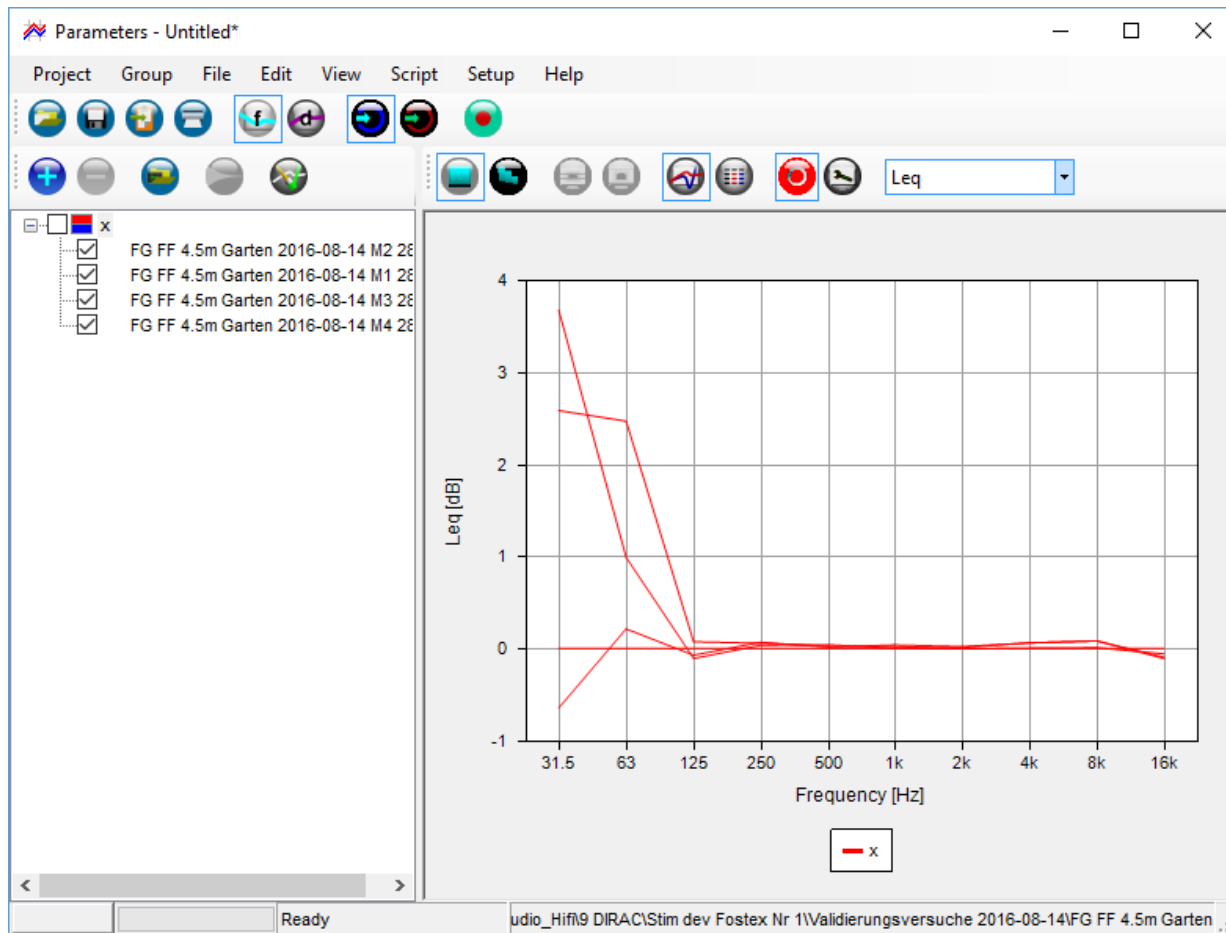
As one sees, the noise fluctuations are quite strong; 3 of the 4 measurements can get used; the result was already clear by ear before this chart got made.



So the quietest of the 3 speaker IR's was used to generate the replacement FR IR (the «dentist IR»).

Some Results (II):

The PARAMETER window of DIRAC has many possibilities for interesting reports; below you see very in detail how much the noise varied the frequency response results. Changing the signal level used for measuring the frequency response should also get check in such a way; the author suggest to use pink MLS frequency response measurements at 70 dBA.



Here one measurement looks the best. But what did you note / register during the signal measurement phase by ear or SLM ? Only if both impressions are inline, use the measurement; otherwise discard it. The best looking was used as reference. Reduce graph low end range to 125 Hz for better resolution.

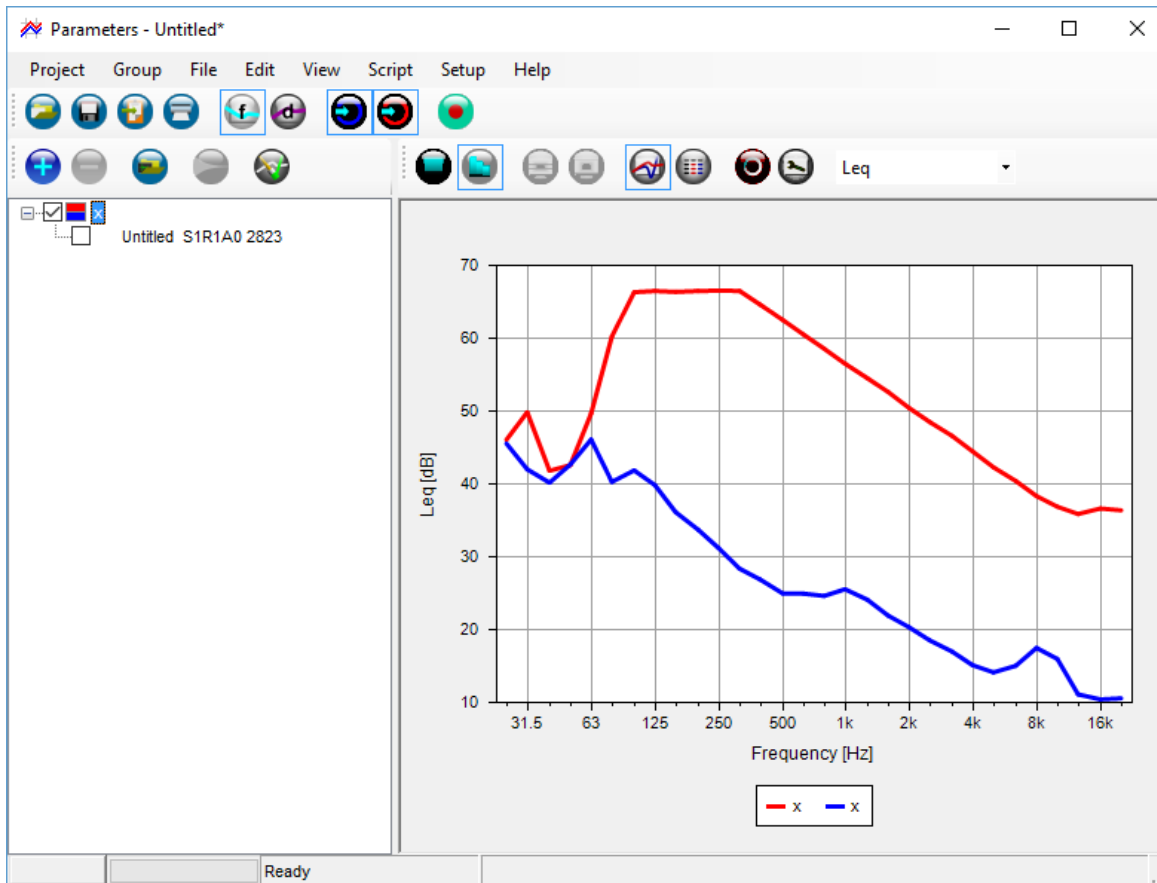
After the measurement - the replacement IR

- When several measurements have been made, look at them using the PARAMETER DIALOG is 1/3 octave resolution / 2-channel mode to see the noise.
- Use **Leq** values, not Magnitude
- Check that the SNR is always above 15 dB. Dismiss poorer measurements
- Form the rest of the measurements use the one with the least noise
- Use PROJECT -> COPY TABLE of the PARAMETER window to get the 1/3 octave FR-
- Open a new DIRAC window
- Generate an dirac impulse using EDIT -> GENERATE -> dirac impulse.
- Open SFD and put from the PARAMETER DIALOG the COPY TABLE value into a empty Excel sheet and the select the frequency and level data (without the heading info) into the SFD shape table. Press ENTER and then OK. Don't forget the ENTER before the o.k.
- Run SFD
- Normalize the result you just got
- Store it under a self-explaining name with date.
- Now you have your replacement IR - noise free, distortion-free; the only disadvantage is, that within the 1/3 octave bands the FR is not corrected. From side of the standard that is no problem; but is the minimally poorer result as DIRAC's standard approach - which should always get used, when being in quiet anechoic chamber.

Some Results (III):

Do at the same time the iteration process of finally develop your stimulus. You should get results / pictures as below.

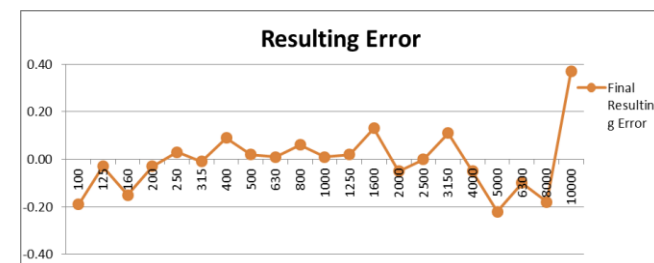
If you are sure, that the second channel information of DIRAC, the noise represents also the noise during the signal phase of the intermittent measurement, you can assume the results you get are of same quality as in a very good anechoic chamber.



Depending of the situation of your test build up; you may need to use a time-gate of about 25 mS with every measurement.

If this is not required thanks to very clean signal quality, leave it away; the time gate has a small negative impact of the low end of the 125 Hz band.

Use the template information for residual error (below). Cross check with the PARAMETER view.

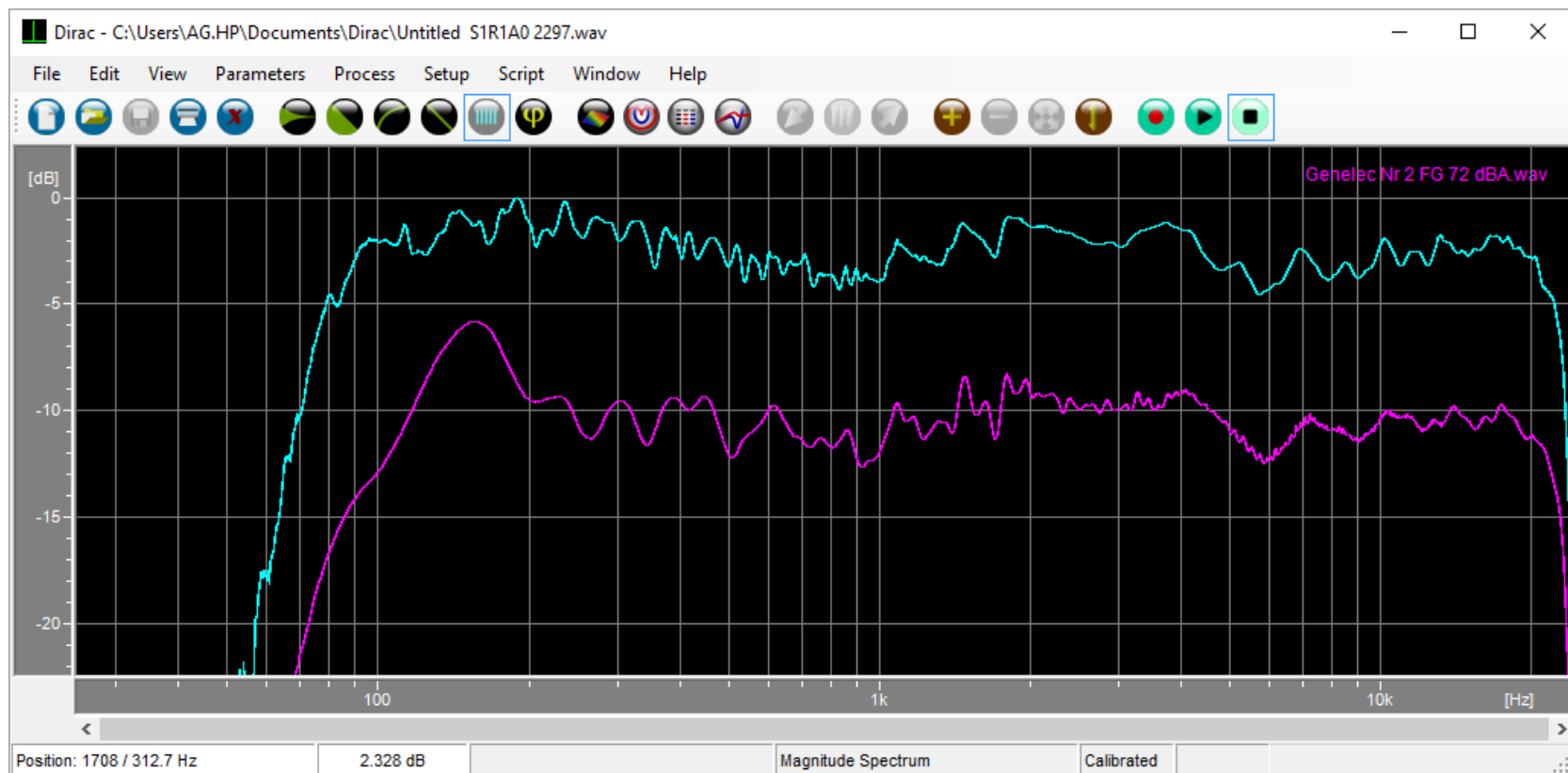


Some Results (IV) - and lessons learned:

Here something went wrong - as explained in the first part of this document in detail. This is a measurement in a rather smaller sized anechoic chamber using BCA technology. That requires precise setup planning to make sure, the test setup is within the “beauty spot” of the chamber. Further asking, if absorption material is available for covering the heavy duty walking grids.

With small anechoic chamber, reducing the distance between speaker and microphone may be required (-> 0.5m)

So having a second view - in this case a FF view - is important to make sure no such error happens.



The Dark Side of Free Field Frequency Response Measurements - Use Limitations

Free field frequency response measurements have also negative aspects. Some have been handled already; here an overview:

- Noise from the environment
- Low availability: Quite a lot of days no measurements are possible due to rain, snow, too low temperature, wind or too high humidity. With bad luck, it can go days or weeks until again a measurement time slot gets possible.
- For the time of day, where noise is low, the weather can be far from ideal - and invers: At night noise is mostly lower, but temperature and humidity can be far away from the climate in inside rooms, chambers, class rooms etc.
- The weather can not get defined or varied as in a lab; and therefore also a setup at reference weather conditions is not possible.
- Sun can play a negative role by heating up devices, driving them into weather dependent behaviour.
- Wind can have an impact on DIRAC's distance measurement results since wind carries the sound additionally. That can change the distance measurement of the IR and also the signal level slightly. With wind, repeated measurements are required to check, that wind strength is still acceptable.

These are relevant restrictions; and it is not possible to check methodically a device for weather immunity.

The Dark Side of Free Field Frequency Response Measurements - Weather impact

Weather has an impact onto the measurement results:

- Wind can make the measurement unusable.
- Low or high temperature and / or humidity can have an impacts of the frequency response and signal output level. Temperature and humidity change the physical qualities of membrane, magnet force, resonance frequencies, Q and others.
- In general one must count, that the weather influence is dependent on the quality level of the speaker
 - the poorer the speaker, the stronger the weather impact is.

It is well possible, that a poor speaker will drop out of the ± 1 dB tolerance of ISO 60268-16 due to weather impact - while a speaker of higher quality will be more stable in its results.

The author of this document got during his test also the impression, that broadband speakers are more weather dependent than 2-way speaker systems. Broadband speakers have more parameters to optimize - making a weather-independent design more difficult.

By the way: Temperature changes do not only come from the weather; running several tests with Lombard level (70 dB(A)) heats up the speaker's coil and magnet somewhat - what can change the signal shape slightly.

The Dark Side of *Free Field and Anechoic chamber* Frequency Response Measurements

None of the solutions is without a price to pay:

- The best solution is a large chamber with optimized wedges. But the hourly rates can be very high - and such a chamber may be far away.
- Free field measurements are a low cost option; just for many operational processes the availability due to weather, noise etc. is not sufficient. Not only the measurements may suffer; the management of the low end cut (for optimal deconvolution during the measurement phase) can get difficult due to too poor low-end SNR.
- What can be very frustrating is: Only after building up the test set and having made some measurements it gets clear, if the measurements will be a success. Sometimes the weather is good at the beginning, and changes later. Therefore it makes sense to have a modular approach, where blocks of measurements deliver usable results - even if out a part of the time useful measurements were possible.
- Small anechoic chambers with simple absorption solutions **may** be sufficient even for calibration purposes - just one needs to learn where the restrictions are and how to dodge them. And maybe after investing quite some time it only gets clear; that the selected chamber is not sufficient for the purpose.

The Dark Side of *Free Field* Frequency Response Measurements

Don't get discouraged too quickly, when staring with the frequency response measurements. SNR is only 10 dB at 100 Hz ? No problem; the frequency response of the IR won't be all too precise - just: Who worries ?

The frequency response IR MEASUREMENT HAS NO OTHER JOB TO DO, THAN TO HELP TO KEEP THE NUMBER OF ITERATION LOOPS REASONABLE. Its precision has no impact whatever on the final result; using the Replacement IR - the «dentist IR» - we fully decouple the stimulus quality from the frequency response IR quality. We could start with a Dirac impulse instead of the dentist IR - just we would end up with (likely much) more boring iteration work.

The spectral precisions comes from the iteration measurements - and there we are in a much better position, since environmental noise is mainly high in low frequencies, so the SNR problem is much reduced - and shifts to the higher end of the frequency range.

The reason for the much less critical low end SNR is the change from a small speaker frequency response with a weak low end - to the male spectra with its high signal level up to about 1 kHz.

Measurement precision - more details

We have covered quite some aspects; time for an overview:

- EN 60268-16 allows us ± 1 dB on 1/3 octave level
- EN 60268-16 does not cover this issue, but tolerances for anechoic chambers are ± 1 dB, increasing to ± 1.5 dB at the frequency range ends; as given by an established standard (ISO 3745).
- Class 1 sound level equipment standards allow tolerances of similar size, frequency dependent.

So overall about 3 dB error accumulation is in the worst case possible; and our goal should be to get overall sum as small as possible. While the standards tolerance is given, and free field measurements can have very small errors, if the area is wind free and quiet; and climate impact is no issue. So what stays is the microphone.

The author has long had the idea, to add a column to the iteration template, to also can compensate the frequency response of the microphone. 3 aspects have been so far holding him back, to do that:

- It would require, same as at the low end frequency response knee region around 88 Hz has, a higher resolution than 1/3 octave steps - likely 1/12 octave steps.
- The template would need to get further tabs, recalculation the microphone calibration data to 1/3 octave steps for all other frequency response parts, where no higher resolution is required. Questions are open, what frequency response region must really get «stretched».
- All supplier come along with different format for their data, requiring supplier specific tabs.
- And - see next page - the open question on the reliability of supplier calibration data.

Measurement precision - more details (II)

Microphone suppliers deliver an individual frequency response chart to each “class 1 microphone”**. That can be a small, poorly readable piece of paper, or - as with B&K - a Windows application (called Microphone Viewer) that comes along with the data, allowing to export it, simulate angle and wind screens influences etc. etc. Most supplier deliver on demand the chart as Excel file; there is no standardized data format.

The author got early the impression, that some supplier data is questionable. He made several free field tests (In anechoic chamber, in rooms using a 5 mS gate, so the chart are only above 1 kHz; below all mics are flat enough). He compared and optimized the test, optimized the speaker choice, microphone stand arm and holder setup and got at the end a result, with less uncertainties than the errors, that got found. Although the data is not precise (maybe +/- 0.3 dB); a conclusion is possible - and it is rather hot.

4 free field “class 1 or 0” microphone from leading suppliers*** got measured, all free field type on ½ inch size; one of them a 40 kHz / 15mV/pascal type. 4 chart got made, each assuming that one of the suppliers data is correct - and how would then the other 3 microphone come out. (Next slide top one example is shown).

2 of such scenarios are highly NOT probable, because microphone frequency responses must have a relation to the membrane resonance, the back-plate damping effect and the pressure increase at higher frequencies by the protection grid (German: Staudruckkurve). From the 2 other scenarios, one is more probable; so it got used as reference, *assuming* that this microphones calibration data is correct.

** There is no such thing as a class 1 microphone, formally only class 1 Sound Level Meters (SLM) exist. But microphones must comply to class 1 SLM requirements; so the microphones inherits (the) requirements. Such microphones are «eichfähig», authorities accept them for calibration (Required for use to can deliver legally accepted measurement results).

*** All microphones class 1 or class 0; all with repolarization. Microphone measurements are tricky; the influence of the microphone stand much higher than what one would expect.

Measurement precision - more details (III)

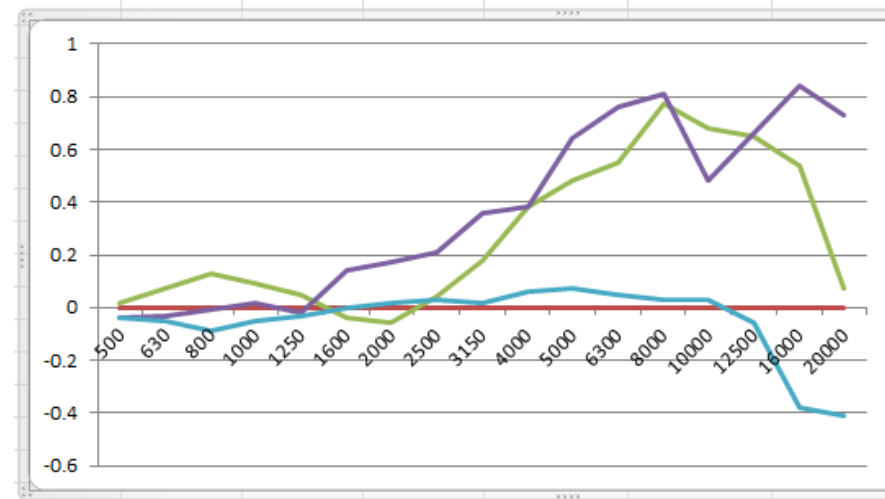
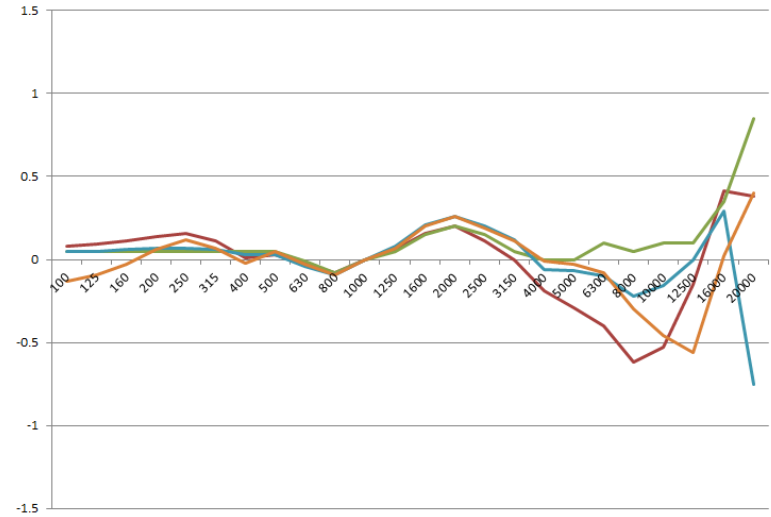
On the right side one of the 4 scenarios. It does not look much like a realistic representation of the frequency response shaping elements of a microphone.

The lower picture shows the **deviations of the measured to the frequency response as given in the calibration charts** - assuming the chosen reference microphone has a correct chart / response.

«Hey, there must be 2 philosophies, how to define a frequency response» was the first thought. But microphone tests like the one made by SP for PCB, do not show any such tendencies with electrostatic actuator measurements. Looking at the free field correction curves of similar looking design bring up questions, why they vary so much while looking the same (see next slide).

So there are several questions open; which slowed down the drive, to put a microphone frequency response column into the iteration template. The unexplainable differences in the lower chart are massive, and would use up nearly the whole speaker equalization tolerance of EN 60268-16.

Open stays also the question, if that was just poor luck with the microphone copies that got used. Measurements on further copies would be required to clarify this issue. Just there are no indicators, that these 2 microphones have a defect.



Here the microphone with the most realistic curve got taken as reference; its errors assumed to be zero. How well the blue curve / microphone fits up to 12.5 kHz is impressive, also how the 2 other curves show to each other a similar, but principally different shape.

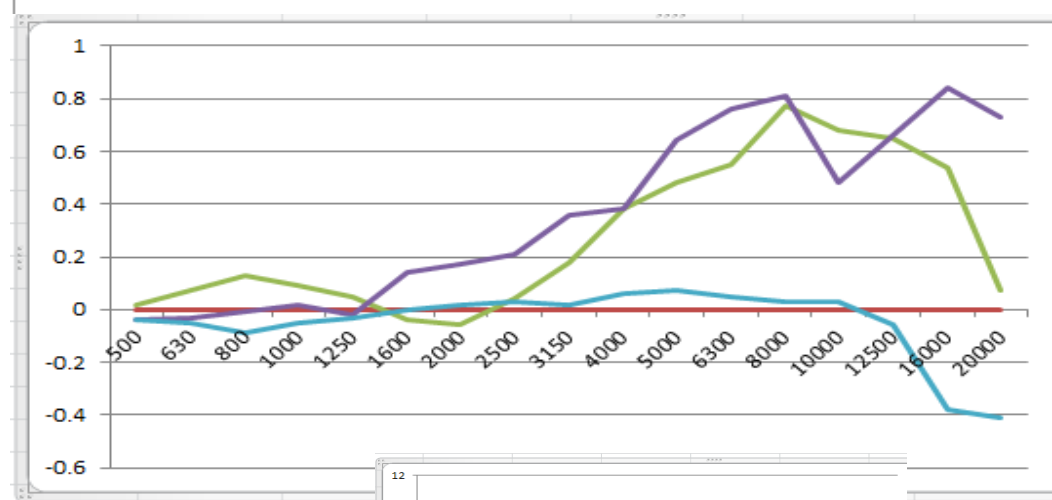
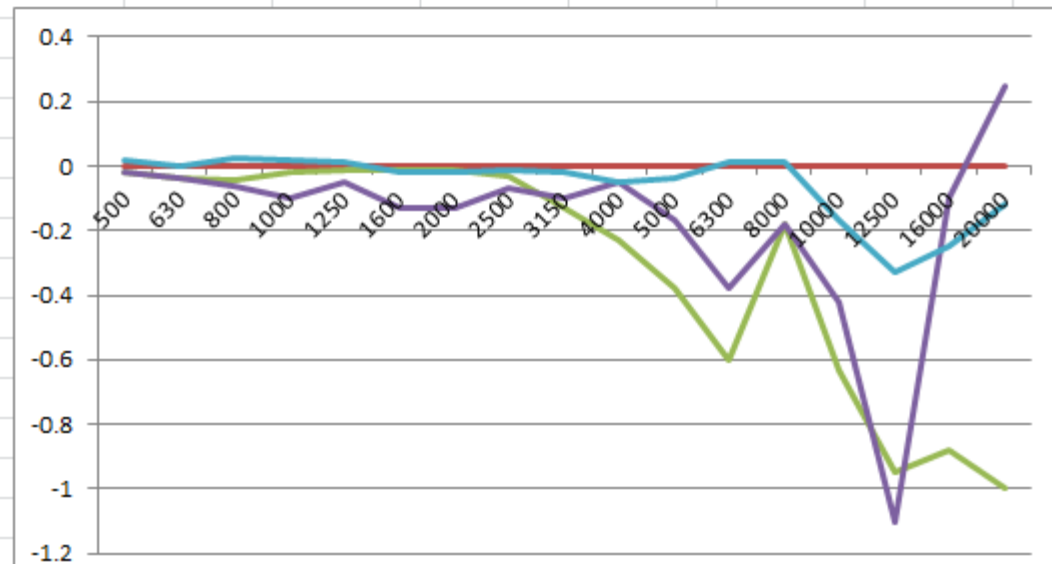
Measurement precision - more details (IV)

Here a more precise look at the free field correction curves as supplied from the microphone producers. ("Staudruckkurve").

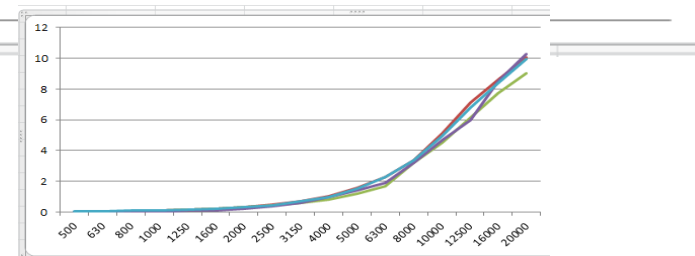
3 of the devices grids look very similar; one is somewhat different. All have precisely the same diameter - as given by the standard. All seem to have the same size of diaphragm diameter, the surface the wave hits on finally. Unclear is, how much variations are between grid inner surface and the diaphragm; but the distance is small and any resonance would be outside of the audible range.

As one can see, the upper curve showing the differences between the microphones free field corrections curves, with the same microphone as reference and the same colours for the devices, shows a picture, where the frequency response deviation are roughly the invers profile of the free field correction curve differences.

"Normal" 1/2 " free field measurement microphones (WS - Working Standard types) have typically about +/- 0.2 dB uncertainty (K=2; 95%) at 10 kHz using electrostatic actuator calibration, increasing to +/- 0.5 at 20kHz; half coming from the protection grid. With lower the frequency, less. (see B&K Tech Review 2001 /1 for details)



Lowest picture: The free field correction curves - not the differences.



Documenting your calibration work

The author is here in a early stage. At the moment a template gets used, which includes the following elements:

- The speakers frequency response with 60 dB(A) of MLS (PALE BLUE)
- The Leq spectra of the stimulus signal, as used in the measurement window when measuring STI. (PALE GREEN)
- The resulting frequency response out of both upper elements; which can well be not in the ± 1 dB tolerance of EN 60268-16 due to nonlinearities. (GREY)
- Further the effective frequency response of the speaker when running the male stimulus. This curve can be shape by distortion in the sense, that the frequency response is slightly tilted to higher or lower frequencies.
- The chart showing both frequency responses; the better they match, the more linear the speaker is.

With these two curves in the graph one gets a «at one glance» information on the speakers capability to stand up to the heavily base loaded stimulus.

Below included (only in the PPT version if this document, otherwise handed on separately) the document at its early stage it has. Next slide shows the template, low right the results from a re-test got included. That is one of the ideas how to use this document: To track the 2 frequency responses as criteria for the speakers health.



Microsoft Excel
7-2003-Arbeitsblatt

Calculation of the STI Stimulus equalization error

Loudspeaker	Genelec 8010 Nr 2	Date	16.08.2016
FreqResp IR			
Path	C:\Data\Akustik_Audio_Hifi\9 DIRAC\Stim dev 8010 Nr 2 60268-16\Validierungsversuche 2016-08-17		
Stimulus	Stimulus Genelec Nr 2 step 3 -2.31 dB 2016-08-16.wav		
Path	C:\Data\Akustik_Audio_Hifi\9 DIRAC\ Aktuelle Stimuli		

FreqResponse of IR or replacement IR	Stimulus spectra	STI Male	Calculated Freq Resp error	Effective Frequency Response after calibration	Effective FR after cal
Measured with Pink MLS @ 60 dBA	Stimulus for 60 dBA	ISO	Assuming linear speaker	Measured with Stimulus	

Magnitude pink [dB]	Magnitude [dB]	Leq [dB]
---------------------	----------------	----------

Fostex Nr 1 FreeField 4.5m	Fostex Nr 1 FreeField 4.5m			xxx	Offset
----------------------------	----------------------------	--	--	-----	--------

Frequency [Hz] Ch.1 Avg		Frequency [Hz] Ch.1 Stimulus		Genelec Nr 2 step 3 -2.3		Frequency [Hz] Ch.1 Avg		
100	-1.91	100	0	10	-11.91	100	54.53	-11.97
125	-0.82	125	-1.14	10	-11.96	125	54.62	-11.88
160	-0.21	160	-1.59	10	-11.8	160	54.6	-11.9
200	0	200	-1.66	10	-11.66	200	54.63	-11.87
250	-0.04	250	-1.7	10	-11.74	250	54.66	-11.84
315	-0.38	315	-1.44	10	-11.82	315	54.57	-11.93
400	-0.82	400	-3.14	8	-11.96	400	54.67	-11.83
500	-1.27	500	-4.66	6	-11.93	500	54.59	-11.91
630	-1.49	630	-6.58	4	-12.07	630	54.65	-11.85
800	-2.08	800	-7.85	2	-11.93	800	54.64	-11.86
1000	-2.05	1000	-9.71	0	-11.76	1000	54.64	-11.86
1250	-1.46	1250	-12.39	-2	-11.85	1250	54.64	-11.86
1600	-0.57	1600	-15.49	-4	-12.06	1600	54.68	-11.82
2000	-0.14	2000	-18.03	-6	-12.17	2000	54.68	-11.82
2500	-0.52	2500	-19.42	-8	-11.94	2500	54.63	-11.87
3150	-0.77	3150	-21.27	-10	-12.04	3150	54.66	-11.84
4000	-1.05	4000	-23.33	-12	-12.38	4000	54.72	-11.78
5000	-2.37	5000	-23.55	-14	-11.92	5000	54.62	-11.88
6300	-2.33	6300	-25.57	-16	-11.9	6300	54.7	-11.8
8000	-2.06	8000	-28.18	-18	-12.24	8000	54.7	-11.8
10000	-1.99	10000	-30.38	-20	-12.37	10000	54.68	-11.82

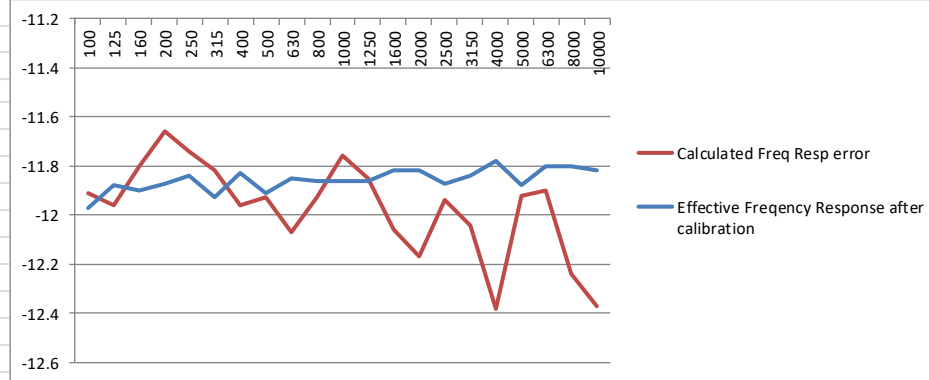
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The **effective frequency response** gets measured by running the stimulus, measuring the result using SKIP DECONVOLUTION and putting the result in relation to the ISO 60268-16 male spectra; using DIRAC's PROPERTIES dialog with reference on a male spectra (in 1/3 octave steps).

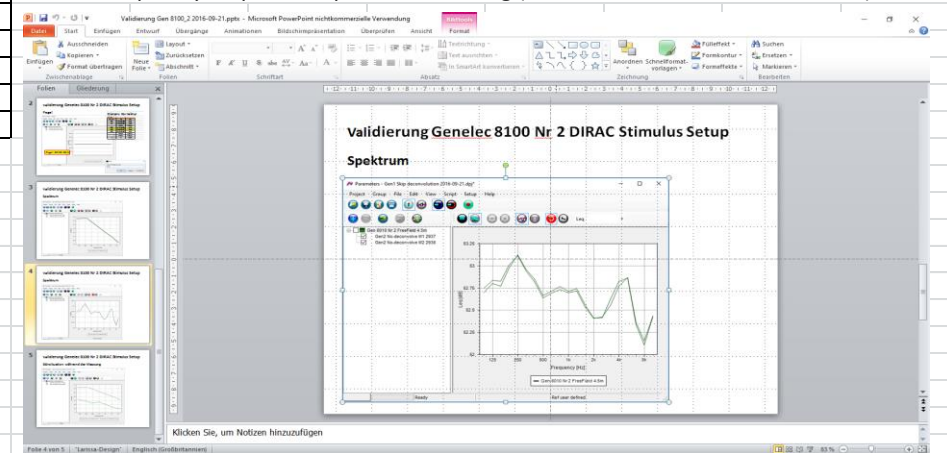
The standard wise measured (with 60 dB of Pink MLS) frequency response is displayed in red; The effective frequency response (as active when running the stimulus) is displayed in blue. The consistence of both is a strong indicator on the speakers (Non-) linearity.

Signal level in the chart below is not calibrated; it is adjusted to optical favor (use violett offset field).

All measurements in Free Field or Anechoic Chamber.



Below: Frequency response at periodic retesting (a month later - different weather)



Some Theory:

Some Theory (I):

The most important aspect to know is, that 2 physical qualities play a large role in anechoic chambers:

- *The surface (as geometric structure) of the absorption material*
- *The type of absorption material (e.g. mineral wool, foam)*
- *Additional elements like steel sheet for dedicated absorption of low frequencies - compound solutions*

The job of the first point is to make sure, any reflection do not go back into the room, but get sent down into the sound dungeon. 99% broad band absorption is required; 20 cm absorption material will typically by far not reach 99% absorption in all relevant frequency bands (at the lower frequency end).

The main benefit of compound steel sheet absorbers is much less demand in volume and the cost compared to wedge solution - for low frequencies - the trick of some EMC chamber (Ferrite & wedges).

Broadband Compact Absorbers (BCA, BKA) *in this document talking generically, not of product brand s!*

This is a relatively new trend, its benefit is strongly based on the steel sheet absorber technology, similar to low end steel sheet sound absorption used e.g. for diesel or cooling machine rooms. The main benefits are:

- For good low end absorption only about 20 cm material depth is required; that in contrast with maybe over a meter with mineral wool wedges. The low end absorber sheet eliminated the low frequencies. Ready to use flat surface compound products are available using BKA in the brand name.
- Due to the flat surface, such products may have a perforated metal sheet housing, easing the operational use of the chamber a lot.

In small rooms the BCA technology brings dramatic improvement in size efficiency. If you have e.g. a 5 x 6 x 4 bare / concrete room, with BCA a much larger **net room size** will “survive” after installing the sound absorption, than with conventional wedge absorbers.

Some Theory (II):

Traditional wedge absorbers

The traditional wedge absorbers are based on his core idea: When the sound wave hits the surface with a flat angle, not all sound will get absorbed, a part will be reflected. The wedge shape controls the reflection, so that most energy of the reflected wave gets send deeply into the wedge «landscape», where further absorption takes place.

But traditional wedge absorbers (without an underlying steel sheet) must fulfil the wave-length rules - and must therefore be very deep to can cope with very low frequencies. If a chamber should be capable to measure correctly HIFI woofers, wedge depth of several meters would be requires. So a wedge-only solution is mainly practical in (very) large anechoic chambers - or when the low end is not important.

Going to physics: Diffuse field / reverberation radius (German: Hallradius)

Maybe it sound a bit nutty, to talk about the border between direct and diffuse sound in an anechoic chamber (AC) - since the walls should “swallow” all sound waves touching them; but is relevant, because beyond that limit the class 1 precision measurement quality is rather a hope.

Taking a room 3.5 x 4.5 x 3 m with over 99% absorptions (as required by a DIN standard for AC's), we would have overall absorption surface of 80 m². This leads according to the formula $Rh = \text{square root}(\text{Absorption surface} / 50)$ to about 1.3 m as radius. The red curve, in slide 5 of this slide set, was measured in a room with the size discussed here. The microphone was maybe about 30 cm outside of the direct sound field border - and that was not detected during the measurements.

Using data from other room sizes, shows similar good results for the formula. And what also gets clear is, that size counts: In a small anechoic chamber one can NOT expect at 1 m distance (as required for STI calibration work) really precise STI frequency response equalization measurements.

Some Theory (III):

Absorbers

The **traditional wedge concept** uses mineral wool which shows at 4 kHz and above - especially when in wedge shape, mostly no deterioration in absorption. Absorption data above 4 kHz is not easy available - but no indicators got found, that in the 8 kHz STI frequency band problems could occur.

BCA concept compound are well known for their impressive low end absorption capability out of small depth. But there are also aspects one should care about:

- According to specs, materials with a flat surface have an absorption loss between about 125 (or even 100) Hz and 200 Hz; dropping to about 0.85. This leads according to statistical acoustics (for the chamber as in the last slide) to a sound level increase of up to 1 dB - so for equalizing STI boxes we could have a problem in the 125...160 Hz band.
- Beyond 4000 Hz; all BCA data sheets the author has seen so far, show no absorption specifications above 4 kHz; but a significant absorption drop at 4 kHz - as common with flat absorption surfaces. In this range also the perforated steel protection cassette around the absorption material can have a slight negative influence.

Walking construction

The industry grade steel grid walking construction gives sound waves a certain amount of reflection surface - especially when the sound wave comes along as a horizontal wave. The author has seen anechoic chamber solutions which (therefore ?) have no walking grid cover over the whole floor surface - or a floor that has been designed to be demountable - with many walking grid modules mostly taken away. Steel wire mesh floors give waves less reflection surface - through its fine structure the influence is limited mainly to very high and much less critical frequencies.

If you have access to an ISO 3745 class 1 compliance report - check for remarks requiring additional absorption material to cover the walking grid (and maybe other room elements).

Some Theory (IV):

Free field Measurements

Free Field measurements have none of these problems, but an other: Disturbing noise and weather impact. By definition, measuring outside / in quiet free field delivers correct data, since Free Field is per definition the sound propagation so much above ground, that no negative influence from the ground is here.

That requires a high construction; but the higher, the more solid and the thicker the construction of the speaker / microphone stand must be - leading to reflections. Further: It must be at a quiet place; and wind may to disturb the sound wave (meteo effects). So measurements between buildings can be specially good; the houses protect from wind and keep noise away.

Above about 4.5m heights, more robust mechanics than a simple microphone stand gets required - and no standard products are easy available. That's why the author ended with a 4.5 m high construction; 4.5m is enough for STI with its 90 Hz low end range. The requirement, to be so much over ground that no negative impacts results, gets finally achieved by applying a time gate to the IR measurements. (In no direction, within 4.5m radius, any reflecting surfaces are allowed).

4.5m heights corresponds to a chamber of 9 x 9 x 9 m or more; a 730 m³ (or more) sound lab ! Keeping that in your mind, may give you the required motivation to look for a quiet place, and to measure at «impossible» day times. But a Free Field measurement lab, always ready to use at working hours - that will be mostly an illusion.

So far, using DIRACS pre-averaging capability did not get used for getting better SNR, nor tested. It has the potential for relevant improvements; just the longer the measurement goes, the longer the operator is «blind» for noise upcommings. So the use depends on your specific noise situation. The author plans to do test on this topic next year, when weather allows.

Conclusions out of Some Theory (V):

When to rent an anechoic chamber, when to make a Free Field measurement

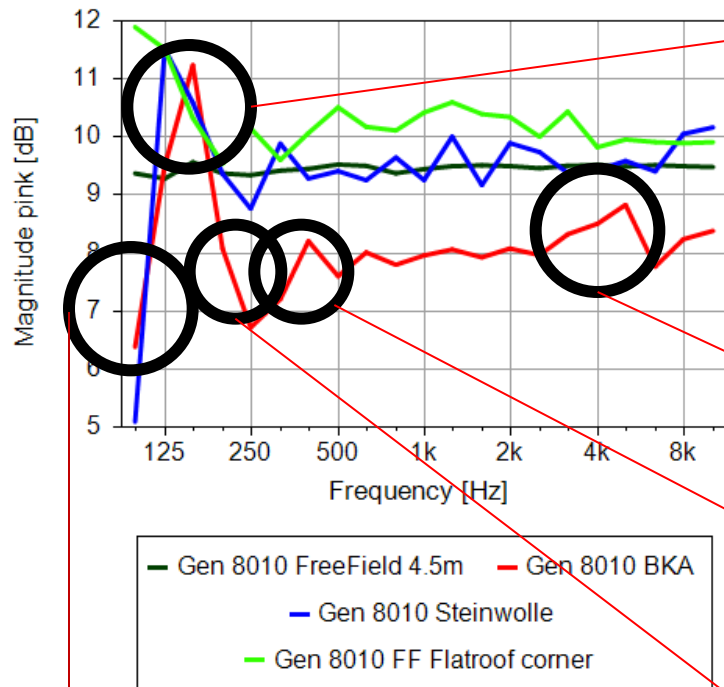
Gather data about the chamber:

- Ask for the results, if any ISO 3745 compliance tests were done.
- Look at the report details, see if for a 1m measurement all frequency ranges would be within 0.5 dB tolerance. Out of the numbers you should get also some information, if the frequency response will be much better at 0.5m distance instead of 1m distance (microphone <> speaker).
- If the floor is made of industrial grade walking grid, ask what for «hand out» absorption element are ready for use. Ask the operator, where to place them on the grid and elsewhere if required.
- Ask if data of the absorption material is available; there (1/3 octave bands) where a drop in the α value exists, calculate the level impact (formula impact = $10 \cdot \log(\alpha)$). Look at both, high end and low end of the frequency range. If the data ends already at 4 kHz, then the chamber was likely never thought for frequency response calibration tests.
- Make a calculation of the radius of the boarder between direct and diffuse sound: *Hr = square root out of the overall sum of absorption material surface, divided by 50.*

With all this data, you should get a good impression, if it is the right chamber to go for.

After your decision - in case of a chamber: Make measurement at 1 m and 50 cm and compare the results. Regarding signal level: The difference must not be exactly 6 dB; since a 10 cm source is no point source; the 6 dB/ double distance decade is only correct for point sources.

Conclusions out of Theory (VI) - *Probable explanation for the red curve in slide 5*



- 1 dB of the 3 dB overshoot relative to the 1 kHz level is fully explainable by the drop in the absorption chart in that frequency region. *But where do the other 2 dB come from ?* Maybe by the walking grid; this would need to get confirmed by experiments.
- The increase at 4 kHz can get explained by an absorption drop at 4 kHz in the absorption specs chart.
- The drop at 250..300 Hz could well be the result of reflection on the walking grid in the area between loudspeaker and microphone.
 - frequency and geometry would fit.
- The drop at about 125..200 Hz is typical for designs with a flat surface / missing wedges.
- The drop of the red curve **below 125 Hz** can get explained by low end calculation models / formulas. (different formulas exist; see also B&K Technical Review 1968-2).

Wedges, BKA, VPR and other (I) ?

We are in the situation, that for calibrating a STI speaker, we need higher precision, than for sound power measurements, as defined in ISO 3745 with its +/- 1 resp. 1.5 dB tolerance.

Since 60268-16 requires a speaker frequency and signal level calibration quality within +/- 1 dB; anechoic chambers according to ISO 3745 are not a priori o.k. Larger ones may be sufficient; especially when keeping in the centre of the chamber. The author believes, that more than about 0.3 dB error is too much.

Looking for anechoic chamber quality data leads mostly to absorption charts, where all curves are “sticking” up at the high end of the scale, all near to 1.00; with differences barely visible. While these graphs are nice for propaganda; they do not really help us to evaluate chambers:

99.00% (Energy) absorption = 10% signal pressure reflection

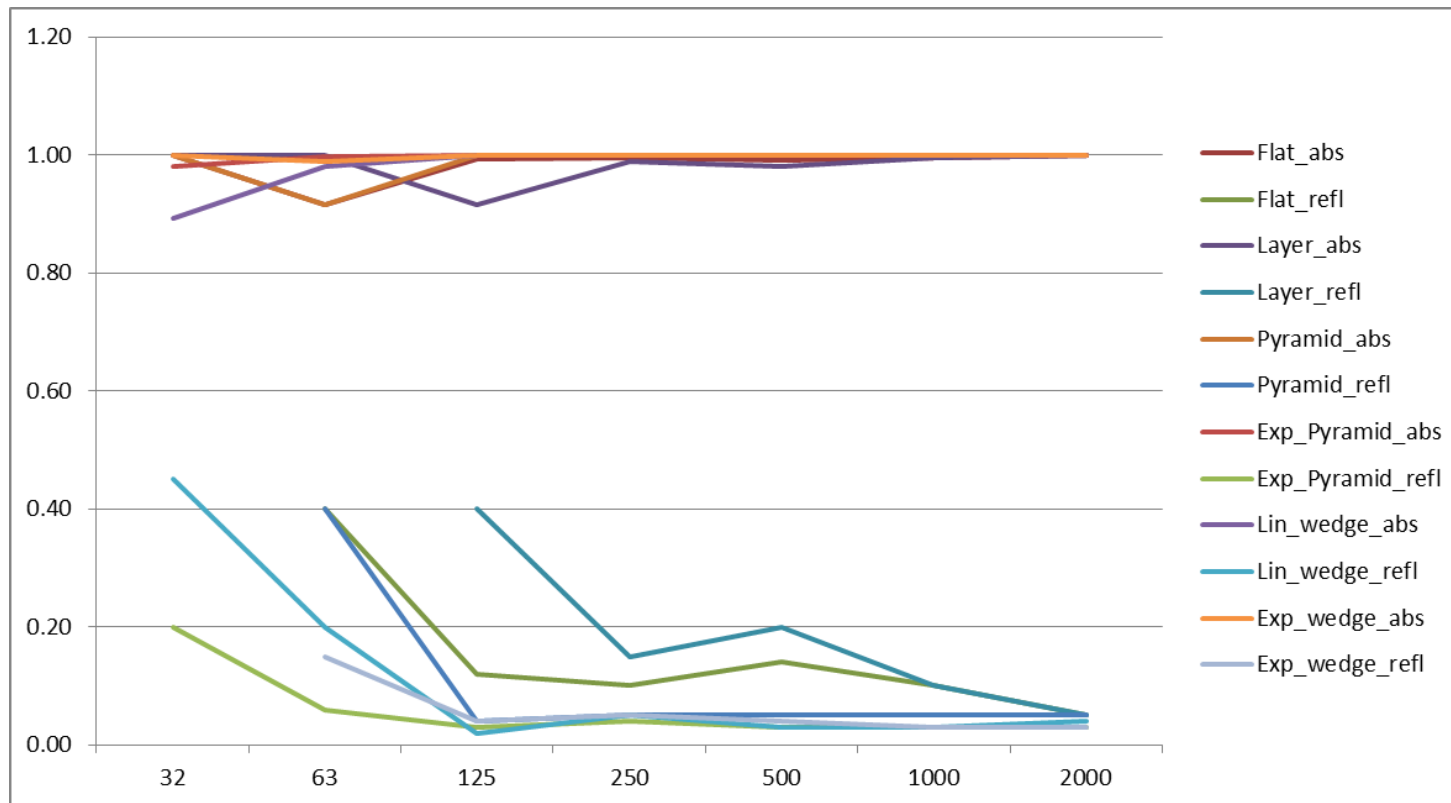
Having **this relation** in mind, it makes much more sense to check out the sound **reflected** by walls / absorption materials; further where the signal reflections goes to. Here the value of wedges comes into the game, since they do not (like flat surfaces) just send the reflected signal back into the room (where it disturbs); a large part of the signal gets send down into the depth of the wedges landscape, where in multiple reflection it gets absorbed.

How much energy gets absorbed by the wedges, depends on their shape. Of the classical shape, flat in the front and getting linearly thicker in direction on the wall, modified versions exists with different profiles having about 5% SPL reflection. Flat surface absorbers have about 15...20% (in the 100..200 Hz Region), this without the influence of the perforated steel cassette, which modern flat surface absorbers are often build into. Such cassettes may lead to additional reflections, at high frequencies.

Wedges and others - Illustration of charts.

The picture below shows some materials with their **reflection** (bottom) and **absorption** (top) **values**. For precision measurements, as calibrating the frequency response of a STI speaker, an absorption rate of about 0.99 (or more, especially in small rooms) may be required - chamber size dependent (the larger the room, the lower it may be for 1 m speaker <> microphone distances).

With the pressure reflection graph one can see nicely, that **only** the blue-grey curve, representing an exponential shaped wedge, fulfils the 99% / 10% rule from about 80 Hz upwards. The absorption graph does not give sufficient information.



The low end capabilities of an anechoic chamber result out of the overlap of the absorption capabilities and the room capabilities (-> modes). The more restrictive element will define the lowest frequency with still acceptable negative impact on the signal level (as in ISO 3745 test)

Wedges and other (III) ?

So there are 2 issues to judge, when choosing an anechoic chamber for your measurements:

- Modern designs with inlay steel sheet absorbers give much better low end absorptions out of small material depth (e.g. 20 cm) than similar thick wedges.
- But to minimize (broadband wise) surface reflection, a wedges-based surface is required.

Principally one could combine both; but internet research of the author did not show up any solutions other the so called ASA solution (-> Fraunhofer Gesellschaft / Insitute in Germany).

The reason, why combinations are rare in the market, did not get clear. Impedance adaption problems ? Or licence / license fee reasons ?

Summary:

To estimate the quality of an anechoic chamber is not easy, even when test data is available. Best is a additional check by an Free Field measurements; further the following aspects can be helpful for first estimations:

- Large chambers are better than small one (larger direct field area)
- Small chambers (about or below e.g. 4 x 5 m) will (normally) not go below 100 Hz
- Well made wedges (of sufficient size) do not send the reflection back into the room, as flat surfaces.
- The combination of wedge surface on absorbers with a steel sheet have the potential for delivering best precision
- A free field measurement can bring in additional clarity; and can replace anechoic chamber measurements.

Highly absorbing anechoic chamber exist since a long time; the Murray Hill chamber built in 1940 had a absorption rate of 99.995% above 200 Hz. The wedges were 4.5 feet long with a base of 1 x 1 foot. With this, it should be clear, that absorption charts from maybe 30 to 100% absorption are not much help, when modern, popular chambers have above 99%; up to 99.995% (or more).

Going to the Meta Level

Maybe you get the impression, the information has some «dispersion»; then that has a reason. The demands on anechoic chambers can vary a lot; as example two major customer interests:

- Car industry, where the DUT are large and the working style is very operational; where standards define the required precision level; where low cost is a premium demand. Often then rooms with a hard floor get implemented, where ISO 3745 allows larger tolerances.
- Research, where best possible precision and highest flexibility is required.

In the first case, mostly sound level / power measurements are main jobs and their demands define the design of the chamber. In the second case, it is much more open, what and how shall get measured. Further the DUTs in the first situation are mostly larger.

Such expectations shape the available solutions on the market. The attitude of chamber suppliers for the second group seems to be: «The absorption must be always beyond 99.00%; there where it drops below that value - when going to low frequencies - the cut off frequency of the chamber is».

For calibration purpose the second type of chamber will typically be the way you want to go. When searching / evaluating chambers, technical schools and institutes are possible places - but their chambers are not «automatically» designed for the second group of customers - although higher schools are working on research.

Building a Test Setup for Free Field Measurements

Building a Test Setup for Free Field Measurements

- Here we are in a hands-on chapter
- The author hopes to hear from readers with some better ideas...
- This is a pragmatic solution, costing some hours for planning and ordering the parts, and maybe a day same for setting it up and doing some tests. Material costs about 400 \$.-; depending side aspects as if your microphone holder you have already can get used.
- Be aware, a microphone stand design has an impact of the result by reflection of the sound on the stand. The more is common between the test stand, and the stand used for STI field measurements, the better the stimulus precision will be.
- Please be aware, the best results you will (likely) get is in summer during morning hours, when the temperature is not far off from room temperature, and no sun shines on the speaker and the wind is low. So your solution and organisation should support this.
- To be sure, that no noise and wind played had disturbing effect in the seconds, when the test signal covers all other environmental changes, make several measurements each time - and carefully look for any differences.
- The author did not find any reasons, why the measurement quality should be second to large high quality anechoic chambers, when the weather conditions - as mentioned above - are fulfilled.
- Have a good look at the results when deciding, if you need to use a time gate or can leave it away. Especially look at the low end noise level and jaggedness.
- In winter month free field measurements will often be not possible - due to the (unknown) weather sensitivity of the small speakers. So improvisation is required to do calibration work in not so good chambers: Try to understand the chambers impact of the spectra and change the templates goal values (column «B» in the iteration template) to compensate for chamber errors. In small or mediocre chambers the low end spectra will first increase (due to missing absorption) and the drop (-> modes).



Up in the sky:

- Make sure, that in at least 5 m distances absolutely nothing is in the way that could reflect the sound.
- Sound level should be below about 35 dBA for a typical «village sound mix» spectra.
- The adapter 24010-314-55 (see next page) need a fine foam (German: Moosgummi) piece between the stand and the adapter to make sure, the adapter does not slip on the stand by vibrations, wind, handling.
- After making the replacement IR («the dentist IR») go into the iteration process as described in blog 7 using the template there.
- When the iteration is done, repeat it so see you have stable results - otherwise the noise or wind impact is too high. Ups and downs of 0.1...0.2 dB are fine; they can show slight tilting to higher or lower frequencies.

The loudspeaker gets held to the wood table by a screw through the table (and tripod adapter) into the speaker; using the factory made thread for a wall mounting adapter made by Fostex.

Manfrotto tripod adapter
(screwed to the board)

Reduziergewinde maybe
215, 217, 218, 219 required

Steel band, as used to fix water
pipes to the ceiling in cellars.

Steel tube 10mm; 1 mm
thick, about 1.2m long.

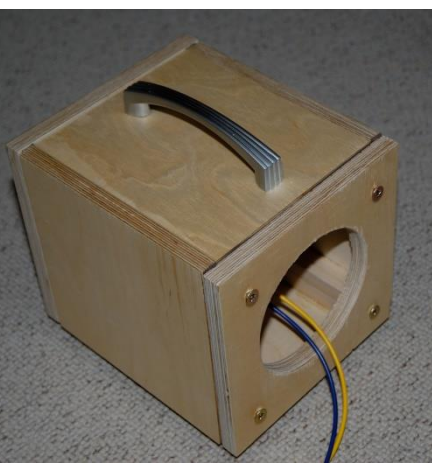
Adapter 24010-314-55

24050-311-55

Mic stand K+M SNR 20811-409-55

Last but not least - security (I):

- Make sure, that the speaker sits perfectly secure on the microphone stand as shown in the last 2 pictures.
- Imagine, that while you push up the speaker up to 4.5m heights, the speaker hold construction would break and the 2 kg heavy device lands on your head - good luck !
- The speaker must can hold securely when the stand is held horizontally and when heavy bumps happen in this position.
- The picture left shows the solution used by the author so far; a Manfrotto camera adapter plate shrewd with 3 M5 screws, nuts and washer through the strong housing. The 3/8 inch microphone thread is precisely in the centre of the box. (This is a box under construction; it shall take up a Tangband W4-1879 chassis; the most powerful speaker the author found for STI, still fully compliant to the standard).
- The same approach can be used for speakers like Fostex 6301; make sure the surface gets the required treatment (sand paper, chemically) as given by the glue supplier, before gluing the adapter plate to the speaker. The glue must be of top strength (Araldit etc).
- ***Before using an adapter, remove any anti-slippage rubber coating. Make sure - when using glue - there is enough contact surface with the speaker housing; underlay metal sheets with glue on both sides if required. If you find an other adapter model / brand with better suited shape, then that should be your choice.***



Originally the plate was fixed by 5 wood screws; 3 got replaced by M5 screws as described on the right..

Security (II) - Long planed Improvement for the Fostex 6301B Speaker:

The wood table solution as in slide 28 was not practical; made handling complicated. To change this was an old plan; but only now with a rainy weekend it got reality.

At the same time the speaker got mounted up-side-down, to get closer to a correct vertical directivity: The «sound opening» on a human head is on the lower half; with the new Fostex solution this is also true for the speaker.



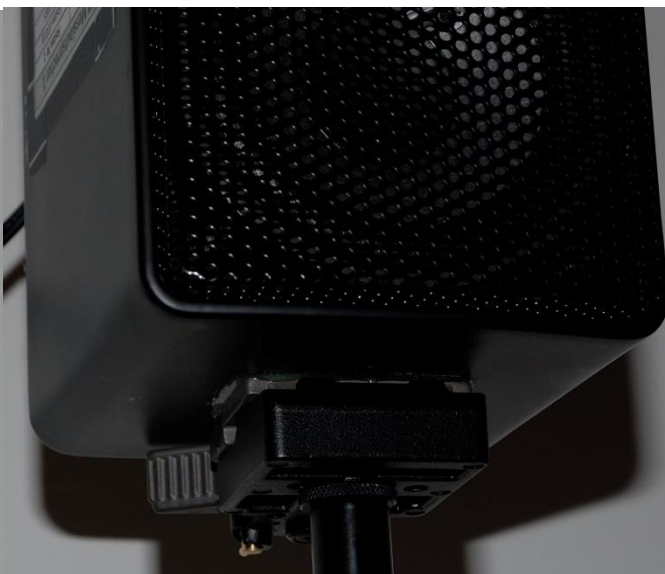
Powerful 2-component glue is required; treat the surfaces as described in the glue data sheet (chemically, sanding) for perfect stability.

UPDATE - just received a snap-in adapter:

In a professional photo shop they suggested a snap-in adapter. It brings in huge advantages: Since the free field microphone stand is over 2 meter high; mounting the speaker overhead wise is difficult, if the speaker has a long mains cable. To snap in is much easier.

Was a bit critical about how well the glue would hold the speaker and adapter together, but it seems to hold well enough: The new adapter is much smaller.

The snap-in adapters rubber coating gets taken away, into the now open, shallow part a small copper gets piece / plate, cut to fit, with glue on both sides, get put in. Head down the sandwich gets glued to the speaker - with its surface prepared for gluing.



The adapter part on the stand fits for photo and microphone stands and has a knob to make sure, the "clutch" does not get opened by accident.



Here you see both parts of the **Manfrotto adapter type 323**.

The yellow colour of the copper plate can well be detected in the hole in the middle of the adapter part glued to the speaker.

Finale

Below you find a special version of the iteration template mentioned in AE log Nr. 6. It has modified “FR of speaker” tab, which uses DIRAC’s intermittent mode and subtracts the environmental noise levels out of the Leq signal level. It should get used when the SNR is between 10 and 20 dB; if lower the Free Field measurements gets questionable; if higher than 20 dB you don’t need to care about this calculation.

The embedded object is only available in the PowerPoint version of this file. Elsewise included somewhere in the blog.

If you find errors, made differing experiences, have ideas or remarks to make, feedback is welcome to alastair.gurtner@sunrise.ch

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