In Situ Room Compensation S62

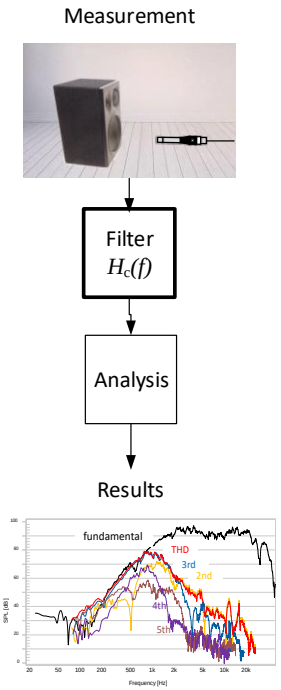
Software Module of the KLIPPEL ANALYZER SYSTEM (Document Revision 1.1)

FEATURES

- Fast measurement in non-anechoic environments
- Simulates standard condition
- Compensation of room reflections
- Far Field data based on near-field testing
- Calibration of measurement rooms and test boxes
- Microphone correction curves

BENEFITS

- Accurate Distortion measurement in any environment (THD, IMD)
- Accurate transient measurement



DESCRIPTION

Most acoustic measurement of audio devices are performed at a single point (e.g. On-Axis in 1m). To determine accurate free field data, these standard tests are usually performed in an anechoic chamber.

Anechoic rooms has a high demand on space and costs. Anyway often these measurement room are insufficiently damped at low frequencies (below 100 Hz), what needs to be corrected.

An alternative is the measurement under so called simulated free field conditions e.g. by windowing the impulse response which isn't applicable for low frequencies (below 500Hz).

The ISC module is using a new approach that corrects the influence of the measurement environment by applying a complex filter to the microphone signal. This enables an accurate acoustic measurement of the fundamental as well as distortion (THD, IMD) with any stimulus (steady state, transient) in any environment (workshop, office).

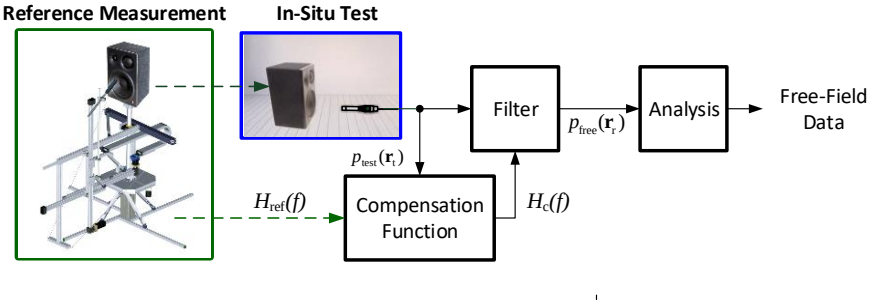
Article number	2520-030

CONTENT

1 Overview

1.1 Principle

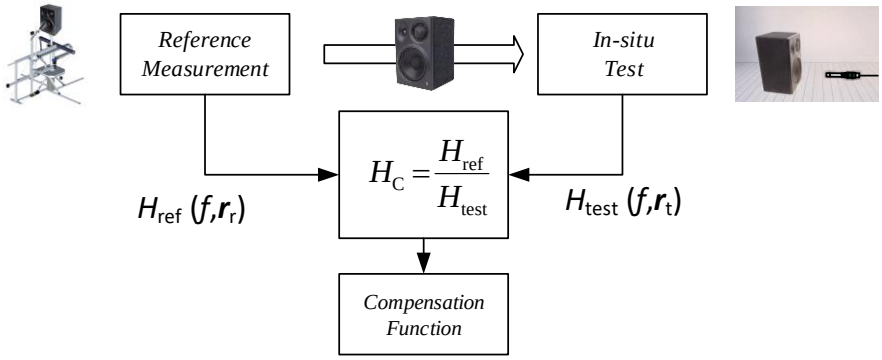
Based on the In Situ Test measurement (e.g. ground floor in workshop) and a Reference measurement (e.g. NFS, anechoic, near field) the ISC module calculates a compensation function.



After this initial calibration of the setup, multiple measurement with different stimuli can be performed. The room reflection are compensated in time domain by applying this compensation filter directly to microphone signal. Thereby, this technique can be used for any stimulus and analysis, acquiring accurate measurements of nonlinear distortion and transient responses.

1.2 Compensation Methods

Complete Compensation with Full Band Reference (FBR)



FEATURES

- compensates for different measurement points $\mathbf{r}_r$ and $\mathbf{r}_t$
- compensation of room influence, position of the measurement points (e.g. near field effects)

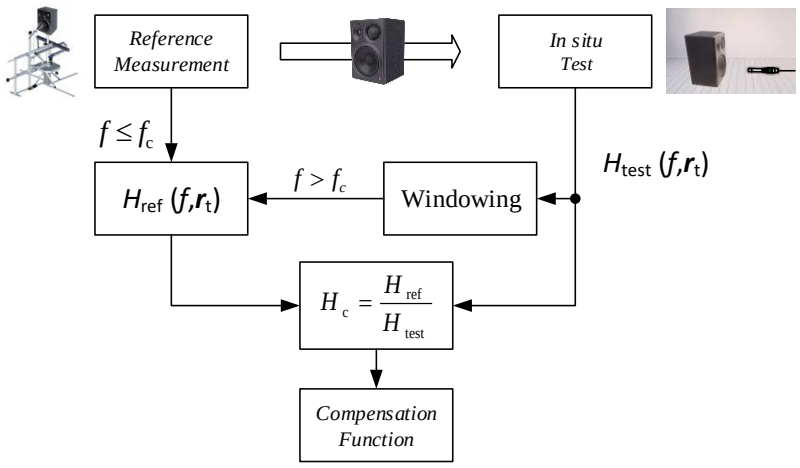
LIMITS

- requires accurate reference response $H_{ref}(f,\mathbf{r}_r)$ with sufficient resolution at all frequencies
- microphone positioning error affects the compensation function $H_c(f)$

APPLICATION

- measurement in small undamped room (e.g. office)
- Far field correction of near field measurements
- Comparison of measurements from different test boxes (e.g. EOL-Test)

Complete Compensation with Low Frequency Reference (LFR)



FEATURES

- reference response $H_{ref}(f,\mathbf{r}_r)$ with sufficient resolution at low frequencies only
- compensation function $H_c(f)$ represents interaction between speaker and room
- microphone positioning error has small influence on compensation function

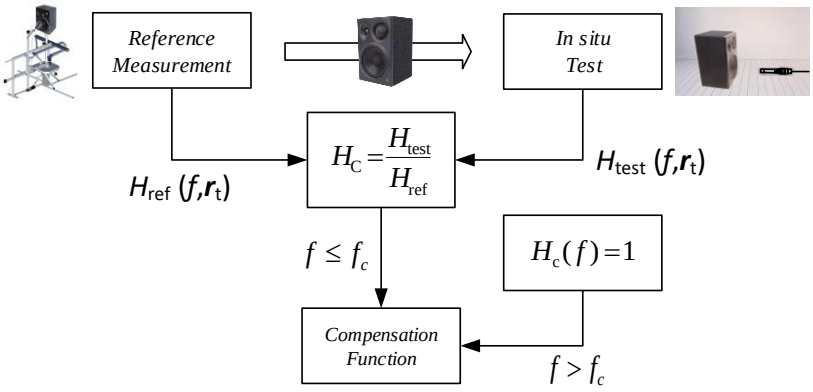
LIMITS

- windowing requires sufficient distance from reflective surfaces
- measurement points $\mathbf{r}_r$ and $\mathbf{r}_t$ shall be identical

APPLICATION

- Measurement non-anechoic rooms (e.g. workshop, office)

Low Frequency Compensation (LFC)



FEATURES

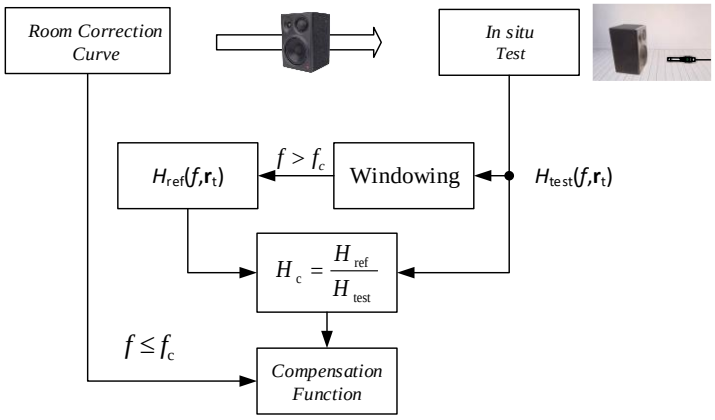
- requires accurate reference response $H_{ref}(f,\mathbf{r}_r)$ with sufficient resolution at low frequencies only (below 1 kHz)
- compensation function $H_c(f)$ is valid for a wide range of speakers
- microphone positioning error has small influence on compensation function

LIMITS

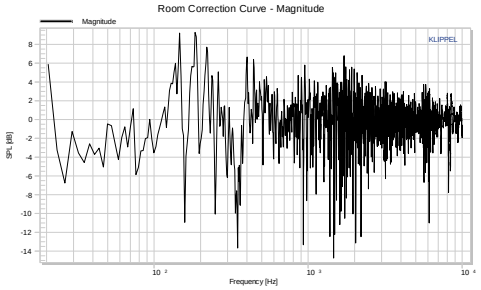
- Requires good acoustic treatment of the measurement room for high frequencies (above 1 kHz)
- measurement points $\mathbf{r}_r$ and $\mathbf{r}_t$ shall be identical

APPLICATION

- Measurement in small (bad) anechoic room
- Calculating a generic room correction curve for a fixed setup

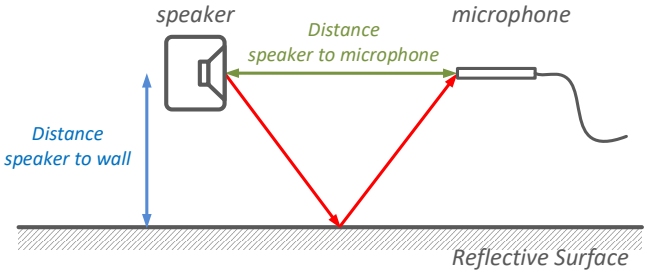
Room Correction Curve (RCC)	 **FEATURES** • no reference measure needed • using a low frequency room correction function • microphone positioning error has small influence **LIMITS** • general reference curve only valid for the same measurement position and similar speaker • measurement points $\mathbf{r}_r$ and $\mathbf{r}_t$ shall be identical • distance correction assuming far field conditions (1/r law) **APPLICATION** • Measurement in small (bad) anechoic room • Calculating a generic room correction curve for a fixed setup • Measurement non-anechoic rooms (e.g. workshop, office)

1.3 Results

Room Correction Curve	Transfer function of complex compensation filter. This filter can be used to correct the setup for further acoustic measurements. (e.g. TRF, TBM, etc.)  Note: This filter is only valid for the current setup and needs to be recalculated when changing the Setup (e.g. changing position of speaker or microphone, measure in a different room)

1.4 Parameter

Compensation Method	Selection of the compensation method according to section 1.2.
In Situ Measurement	The ISC imports data automatically from the selected ***TRF operation***
Reference	Definition of Reference Curve. This curve can be imported via the ***Clipboard*** or by a direct operation link to either a ***TRF operation*** measured in free field or to a ***NFS Visualization*** operation.

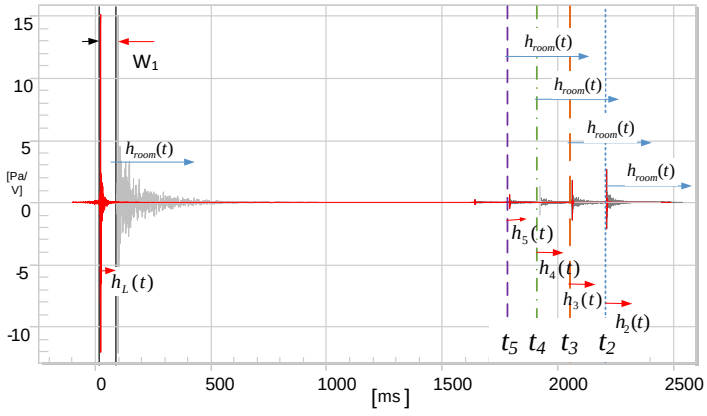
Time Windowing (LFR only)	For the Complete Compensation with a low frequency reference curve (LFR) the ISC uses time windowing to extract the direct sound from the In Situ measurement at high frequencies. The length of this window is calculated automatically based on the measurement distance and the distance to the nearest room boundary. 
Transfer Function Adjustments	When comparing a reference measurement with an In Situ measurement, there are usually small mismatches in the mechanical alignment of the measurement setup. These little mismatches like Gain, Delay, Polarity are checked automatically and can be adjusted to ensure valid compensation filters.

2 Applications

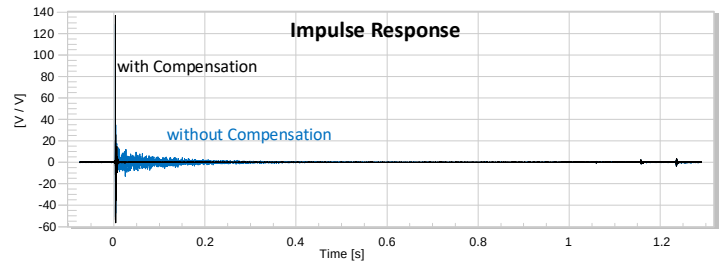
2.1 Harmonic Distortion measurement with chirp

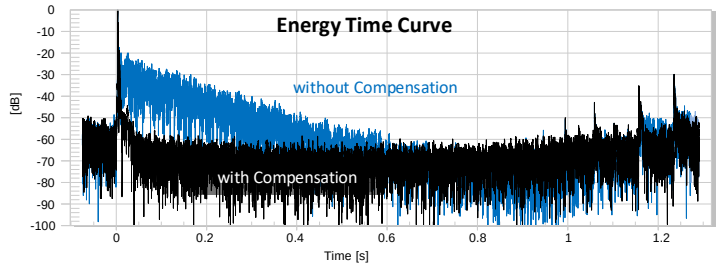
When measuring the impulse response of an audio device with a sweep, the harmonic distortion of the nonlinear system can be separated in the time domain by windowing the impulse response. This particularity is caused by the frequency time mapping of the chirp.

Anyway, this separation technique requires short impulse responses. If the acoustic measurement is performed in a non-anechoic room the decay of the room can be easily longer than 100 ms. This makes a separation of the harmonics distortion almost impossible, because for example the decay of the 3rd harmonics can ring into the 2nd harmonics.

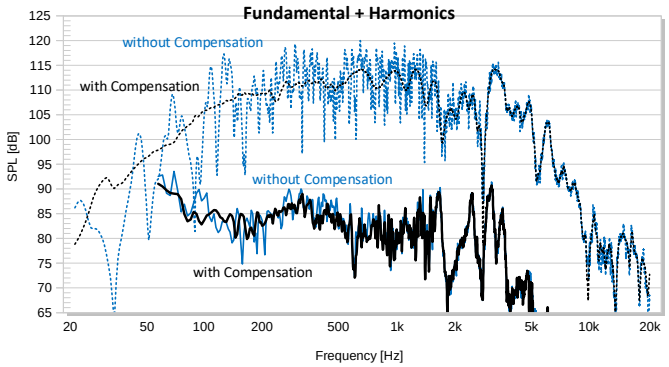


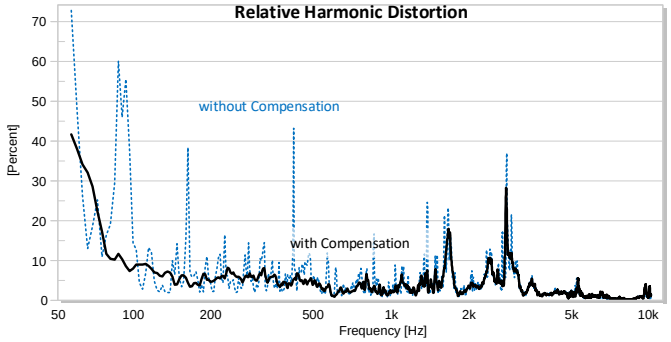
Applying to the microphone signal a filter, which compensates for the room influence reduces the ringing of the impulse response strongly.





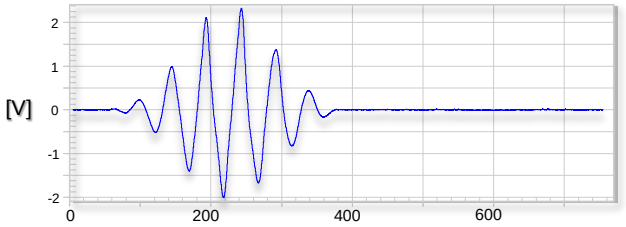
Thus, the Farina processing can be applied and accurate harmonic distortion can be determined.





2.2 Burst Test (Maximum Peak SPL- ANSI/CEA2010)

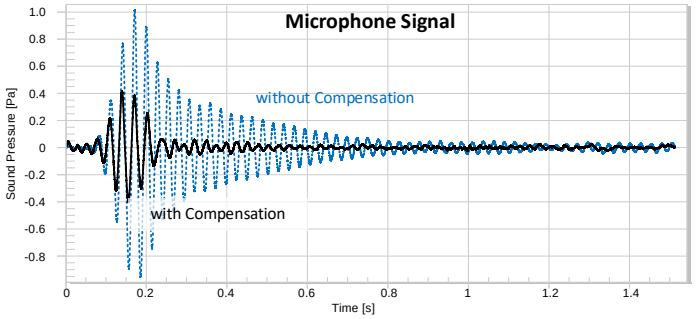
The target of the ANSI/CEA 2010A burst is the measurement of the maximum peak SPL of a speaker. The burst is a short time (transient) excitation that uses a windowed sinusoidal tone with 6.5 periods.



The standard specifies a frequency range from 20 Hz to 63 Hz and the peak needs to be detected in the time signal of the measurement microphone. These requirement makes a compensation filter for this standard measurement indispensable.

The following example shows a burst measurement at a room resonance. As shown in the picture the short burst is enough to excite a room resonance which rings almost 1s.

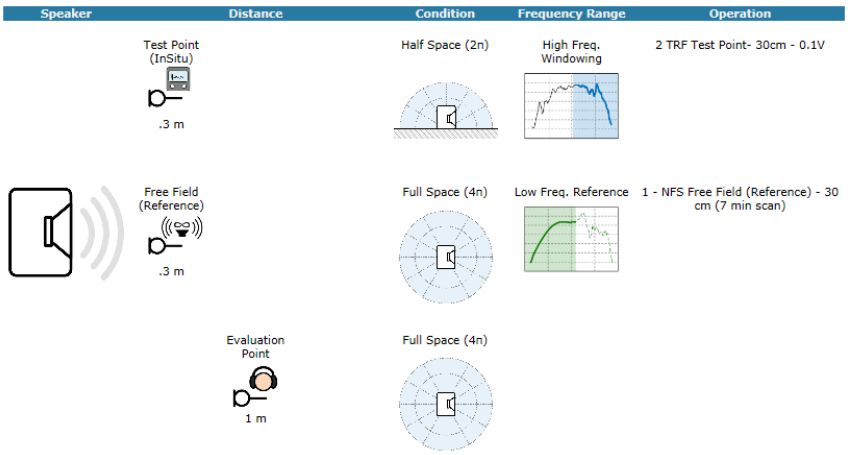
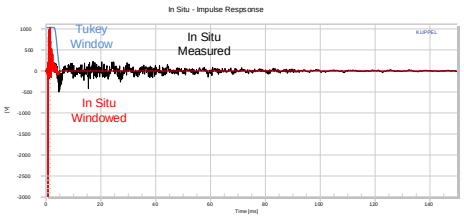
When applying the compensation filter the resonance is compensated and the free field sound pressure output of the speaker can be determined.

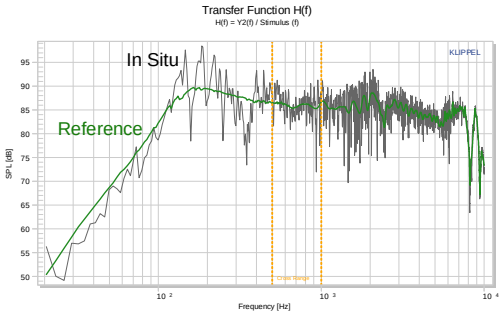


3 Requirements

3.1 Hardware		
Analyzer	Klippel Analyzer 3 (KA3) or Distortion Analyzer 2 (DA2) Hardware platform of Klippel R&D System.	
Amplifier [optional]	External amplifier to drive the device under test. Note: The KA3 has an internal amplifier card, which can be used as well.	
Microphone	Measurement microphone (XLR or BNC) to capture the sound pressure of device under test.	
3.2 Software		
dB-Lab (>210.560)	Project Management Software of the KLIPPEL R&D SYSTEM. Minimum version 210.560	
Transfer Function Module (TRF)	The Transfer function (TRF) is a dedicated PC software module for measurement of the transfer behavior of a loudspeaker.	
3.3 Optional Software Modules		
Near Field Scanner 3D (NFS)	The Near Field Scanner (NFS) can be used to determine a very accurate free field reference curve measured in any non-anechoic environment.	

4 Output

Summary Window	The summary window lists all input parameters of the module, as well as the automatic adjustments and mismatches between the In Situ and Reference measurement. In addition warning and error will be propagated here. 
Impulse Response	Impulse response of the imported In Situ Measurement with the applied automatic time window for high frequencies. (only for LFR method) 

Transfer Function	Transfer function of the In Situ and Reference measurements. These transfer function are used to calculate the compensation filter. 

5 References

5.1 Related Modules	[1] *Transfer function (TRF)*, Specification S7, 2016 Klippel GmbH [2] *Near Field Scanner 3D (NFS)*, Specification C8, 2016 Klippel GmbH [3] *Tone Burst Measurement (TBM),* Specification S44, 2019 Klippel GmbH
5.2 Publications	[4] C. Bellmann, W. Klippel: *Fast Loudspeaker Measurement in Non-Anechoic Environment*, presented at the 143rd Convention of the AES in New York, NY, USA, 2017 October 18-21.

Find explanations for symbols at:
http://www.klippel.de/know-how/literature.html
Last updated: June 03, 2021
Designs and specifications are subject to change without notice due to modifications or improvements.