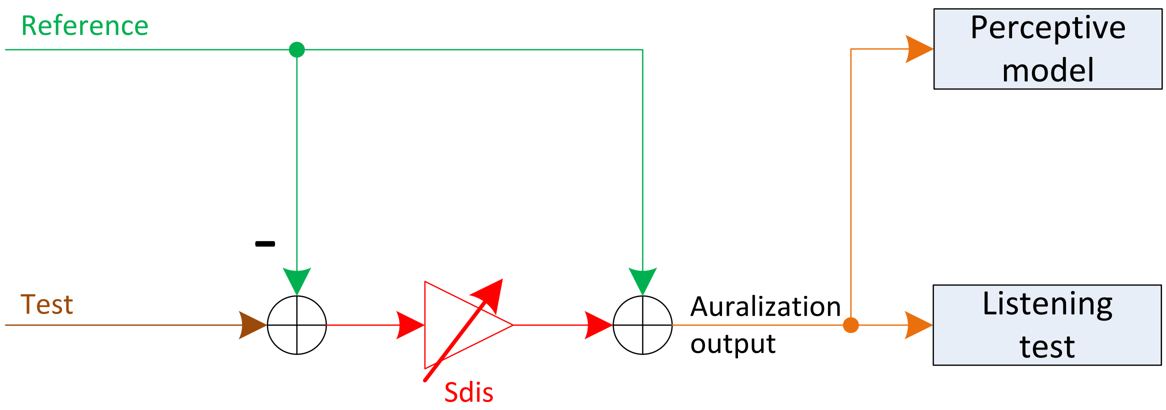
Difference Auralization **S22**

Software Module of the KLIPPEL R&D and QC SYSTEM

FEATURES	BENEFITS
• Automatic alignment of input signals in time • Isolation of difference signal (no model is employed) • Scaling of difference signal to enhance or attenuate distortion • Automatic leveling of auralization output • Export of auralization output to WAVE files • Distortion analysis and frequency domain analysis	• Combines subjective and objective evaluation • Isolates all kinds of regular and irregular distortion (also rub & buzz) • Exported files may be used for listening tests or perceptual simulation • Sensitization of listeners to defect symptom • Communication of sound quality to nontechnical colleagues to define target performance • Determine critical test signals



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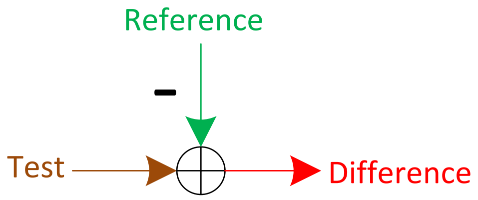
Overview

Introduction

Auralization techniques are used to enhance or attenuate distortion components to communicate symptoms (e.g. of defects) that influence the sound quality.

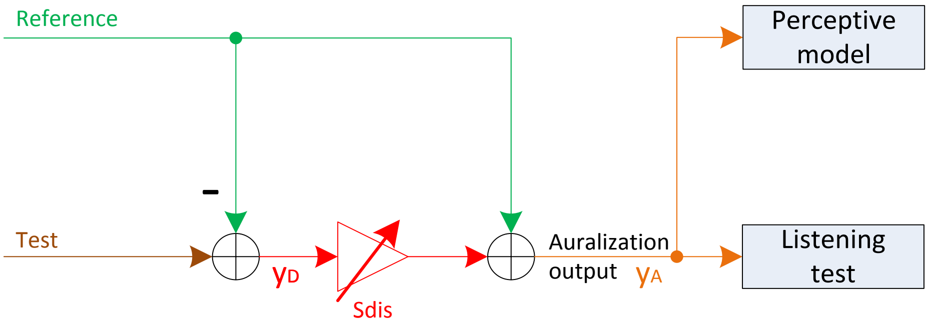
This module uses a difference decomposition technique to isolate the distortion signal. Two input signals (a reference input signal and a test input signal) are used to calculate the difference signal.



The input signals have to be aligned in time and level before the subtraction. The time alignment is performed automatically by the module, thus providing synchronous signals.

Depending on the containing distortion components, the difference signal contains these components that are in the test input signal, but not in the reference input signal.

The difference signal is then scaled with the distortion scaling factor S_{dis} resulting one auralization output signal for each distortion scaling. The signals are exported to WAVE files and may be used in listening tests or a perceptual model.



Requirements

This module is a licensed CAL module. A license and a dongle or distortion analyzer is required.

Input Signals

WAVE data and vector formats are supported as input signals. All kind of audio signal (music, sweep, noise, …) is supported.

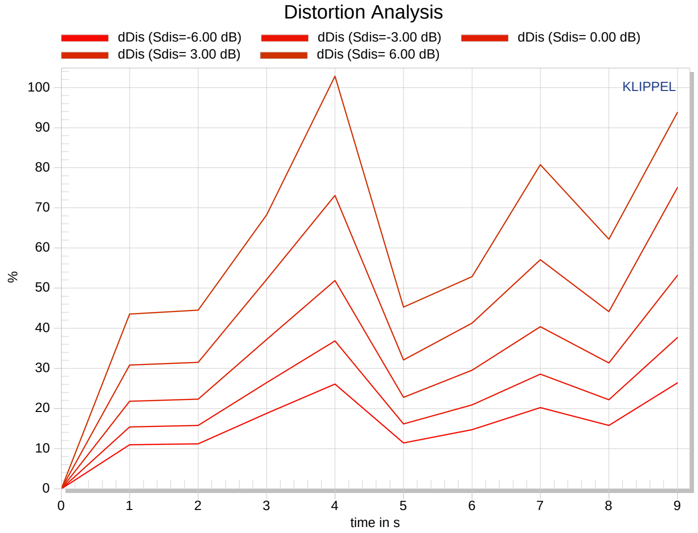
Curve Results

Distortion Analysis

The distortion analysis indicates the amount of contained distortion in the auralization output it's defined as the ratio of scaled difference signal to auralization output within a certain time frame:

$$d_{\mathrm{dis},S_{\mathrm{dis}}} = \frac{10^{\frac{S_{\mathrm{dis}}}{20}} \hat{y}_{\mathrm{D}}(t)}{\hat{y}_{\mathrm{A},S_{\mathrm{dis}}}(t)}$$

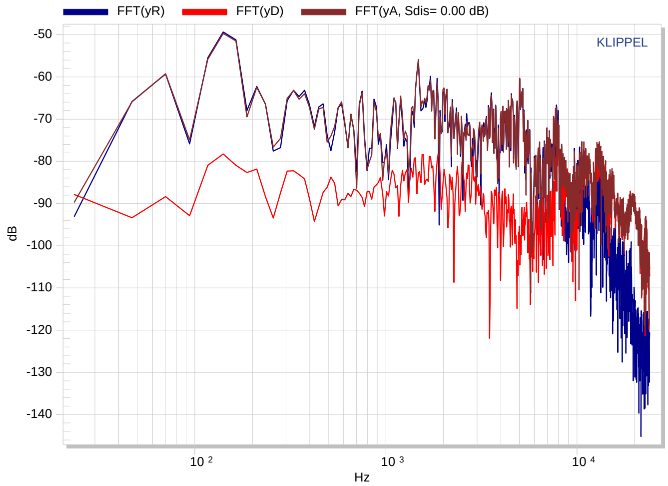
The time frame is set to 1 s to be comparable to the AUR module.



FFT analysis

The module calculates the magnitude spectra of a short term **Fourier** analysis of the auralization output $\hat{y}_{\mathrm{A},S_{\mathrm{Dis}}}(t)$, the difference signal $\hat{y}_{\mathrm{D}}(t)$ and the reference output signal $\hat{y}_{\mathrm{R}}(t)$ at the time defined by the parameter *TimeMarker*.

The FFT analysis is designed for analyzing steady-state signals.



Single value results

The single value results comprise signal characteristics of the following signals (please refer to the signal flow plan for explanation of symbols):

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- *Exported signals*

	Output (Wave)	Symbol	$Level_{RMS}$	$Level_{Peak}$	$Level_A$
	Reference	w_R	-42.6 dB	-20.4 dB	-42.8 dB(A)
	Auralized (S_{dis}=-6 dB)	$w_{A,S_{dis}=-6\ dB}$	-42.7 dB	-20.5 dB	-42.9 dB(A)
	Auralized (S_{dis}=-3 dB)	$w_{A,S_{dis}=-3\ dB}$	-42.7 dB	-20.5 dB	-42.9 dB(A)
	Auralized (S_{dis}=0 dB)	$w_{A,S_{dis}=0\ dB}$	-42.7 dB	-20.6 dB	-42.9 dB(A)
	Auralized (S_{dis}=3 dB)	$w_{A,S_{dis}=3\ dB}$	-42.7 dB	-20.6 dB	-42.9 dB(A)
	Auralized (S_{dis}=6 dB)	$w_{A,S_{dis}=6\ dB}$	-42.8 dB	-20.8 dB	-43.0 dB(A)
	Difference	w_D	-61.1 dB	-33.0 dB	-63.3 dB(A)
*	Calibration	w_C	-14.0 dB	-1.3 dB	-13.9 dB(A)

- *Signals in the pressure domain**

The exported signals are mapped to the pressure domain at the receiving position (using playback gain G_P). Only visible in advanced mode.

	Signal Name	Symbol	SPL_{RMS}	SPL_{Peak}	SPL_A
*	Reference	p_R	51.3 dB	73.6 dB	51.1 dB(A)
*	Auralized (S_{dis}=-6 dB)	$p_{A,S_{dis}=-6\ dB}$	51.3 dB	73.5 dB	51.1 dB(A)
*	Auralized (S_{dis}=-3 dB)	$p_{A,S_{dis}=-3\ dB}$	51.3 dB	73.5 dB	51.1 dB(A)
*	Auralized (S_{dis}=0 dB)	$p_{A,S_{dis}=0\ dB}$	51.3 dB	73.4 dB	51.0 dB(A)
*	Auralized (S_{dis}=3 dB)	$p_{A,S_{dis}=3\ dB}$	51.2 dB	73.3 dB	51.0 dB(A)
*	Auralized (S_{dis}=6 dB)	$p_{A,S_{dis}=6\ dB}$	51.2 dB	73.2 dB	50.9 dB(A)
*	Difference	p_D	32.9 dB	61.0 dB	30.8 dB(A)
*	Calibration	p_C	80.0 dB	92.7 dB	80.1 dB(A)

- *Gain settings*

This table shows an overview of gain settings.

	Name	Symbol	Gain
	Reference Gain	G_R	0 dB
*	Auralization Gain*	G_A	0 dB
*	Level Equalization Gain (S_{dis}=-6 dB)	$G_{L,S_{dis}=-6\ dB}$	-.04 dB
*	Level Equalization Gain (S_{dis}=-3 dB)	$G_{L,S_{dis}=-3\ dB}$	-.07 dB
*	Level Equalization Gain (S_{dis}=0 dB)	$G_{L,S_{dis}=0\ dB}$	-.11 dB
*	Level Equalization Gain (S_{dis}=3 dB)	$G_{L,S_{dis}=3\ dB}$	-.19 dB
*	Level Equalization Gain (S_{dis}=6 dB)	$G_{L,S_{dis}=6\ dB}$	-.32 dB
	Export Gain	G_E	0 dB

- *Input signal information*

Information on the input signals are provided for determining problems due to input headroom and temporal alignment.

Symbol	Signal Name	f_s	Headroom	Length	Delay
x_R	Reference input signal	48000.00 Hz	20.42 dB	9.34 s	-
x_T	Test input signal	48000.00 Hz	20.44 dB	9.34 s	-0.00 s (-10.00 samples)

Items with asterisk (*) are only shown, if an absolute relation is calculated. Please refer to the advanced parameter playback gain G_R.

Exported audio files

The difference auralization exports the following signals to one channel WAVE files:

- *Calibration signal*

  The calibration signal is a noise signal with one octave bandwidth centered at 1 kHz. The wave file is used to calibrate the audio playback system in order to realize the sound pressure levels of the signals in pressure domain. Please refer to the section calibration of playback system.

- *Reference signal*

  This represents the reference signal used for the auralization. It represents the best sound quality without any distortion ($S_{\mathrm{dis}} = -\infty\ \mathrm{dB}$) and may be used as the reference signal in listening tests or perceptual models.

- *Auralized signals*

  The auralization output represents the signals with differently scaled distortions. For each defined distortion scaling factor S_{dis} an auralized signal is calculated.

- *Difference signal*

  The difference signal is exported to provide the possibility for manually checking the isolated difference.

The mono signals are available as file link in the signal characteristics table of the exported signals.

In addition to the mono files, stereo files are generated that consists of reference and auralized signals in separate channels. Files in the folder *stereo_ref+auralized* have a fixed channel assignment: the reference signal is the 1st channel, the auralized signals for different distortion scaling factors are in the 2nd channel. Files in the folder *stereo_ref+auralized_random* have a random channel assignment and may be used for listening tests directly. The file *solutions_distorted_channel.txt* contains the channel assignment for the distorted signal.

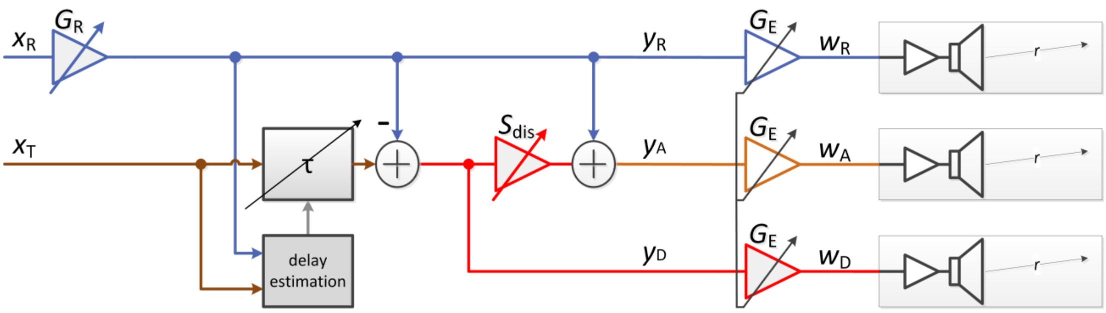
Basic Signal Flow Plan

The basic signal flow plan shows all major input and output signals for operating the DIF-AUR. The minimal set of input parameters allows a fast execution of the auralization:

- Reference and test input signals (vector or wave file)
- G_R – if necessary, set to 0 dB if not defined
- S_{dis} – defines the scaling of isolated distortion, set to 0 dB if not defined

Available, yet optional parameters are:

- Delay, determined automatically by maximum of cross correlation
- G_E – determined automatically to ensure efficient headroom for exported wave files



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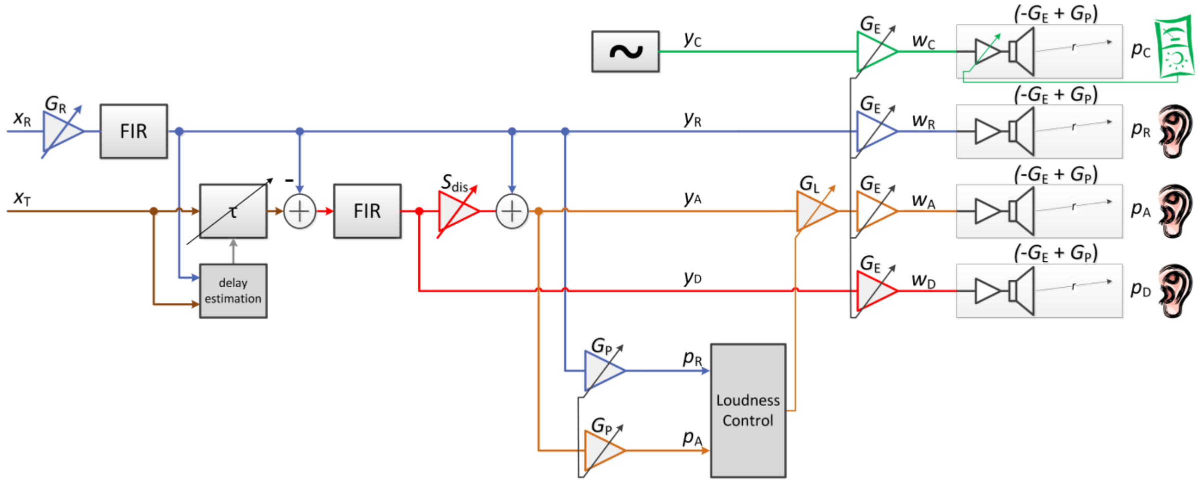
Advanced Signal Flow Plan

Additional advanced parameters allow the fine-tuning of the exported wave files.

Overview:

- FIR filters: The available FIR filters allow manipulating the
  - **reference input signal** by loading an impulse response (VEC or WAV) and convolving the original input signal.
  - difference signal by applying a band-pass filter and reducing the signal components of the difference signal to the desired frequency range.
- Playback gain G_P: allows the specification of an absolute pressure definition (please also refer to the section *Calibration of playback equipment*) by defining a playback gain and thus the sound pressure level in the listening experiment.
  - It may be specified as a relative gain ({number}) or
  - as the target sound pressure level of the reference output signal.

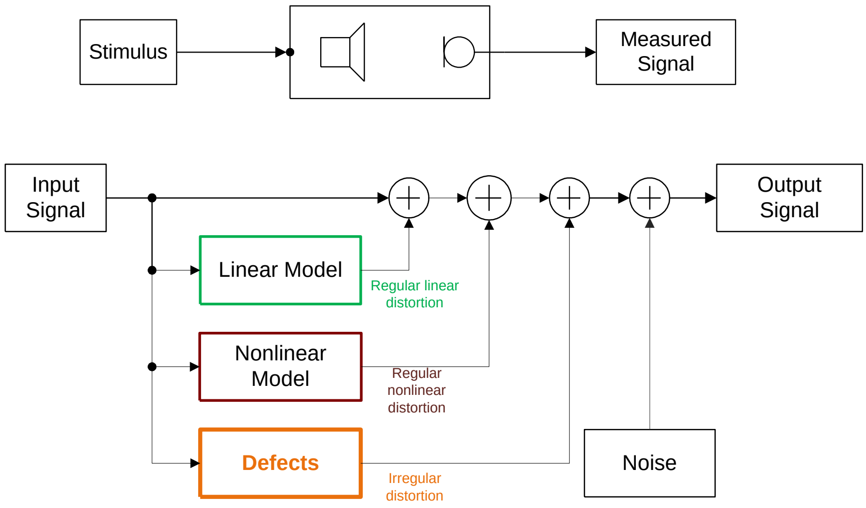
  An absolute pressure definition is necessary to apply the psycho-acoustical model for the level alignment and to calculate the sound pressure levels at the receiving position.
- G_L – allows the level alignment of the exported wave files. It may be defined as a gain (in dB) or set to 'level' or 'loud'. Please note that 'loud' requires an absolute pressure reference. The module applies the given gain to all auralization output signals.
  - {number} applies the gain in dB to all auralization signals.
  - 'level' applies an individual gain to all auralization signals (multiple signals, if multiple distortion scaling factors S_{dis} are defined) to realize wave files with the same level as the reference output signal.
  - 'loud' applies a psycho-acoustical model to calculate the necessary gain for each auralized signal to be perceived as loud as the reference output signal.



Distortion components

Signal flow plan

The distortion components of audio products may be modeled with the following signal flow plan.



Modeling distortion components

The regular linear distortion can be predicted by lumped or distributed parameters of the linear transducer model. The linear distortion generation is optimized during the design process.

The regular nonlinear distortion can be predicted by lumped parameters of the nonlinear transducer model, the generation of this distortion component is optimized during the design process as well.

The irregular nonlinear distortion are generated by defects (rub&buzz) in manufacturing and can usually not be modeled or predicted.

Noise is caused by external factors, e.g. environmental noise, production noise, noise in a typical application (tire and air noise for automobiles). This component is independent of the stimulus.

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Auralization of distortion components

Traditional auralization techniques are able to auralize distortion components with models of the transducer. Irregular nonlinear distortions cannot be modeled due to their random characteristics. The difference auralization may auralize all components (including irregular nonlinear distortion), since it does not employ a model.

The definition which distortion components are isolated lies solely with the choice of input signals. All components that are included in the test input signal, but not in the reference input signal are defined as distortion and reflected in the difference signal. The following table provides an overview of possible choices.

Difference Signal	*Test signal*	*Reference signal*
Regular Linear Distortion	Transducer output at small amplitudes (amplitude adjusted to listening level)	Stimulus (time delay and amplitude adjusted to test signal)
Regular Nonlinear Distortion	Total output (linear + distortion) of the AUR module (digital model in DA using nonlinear parameters)	Linear output of the AUR module (digital model in DA using nonlinear parameters)
Irregular Nonlinear Distortion	Transducer output at high amplitudes	Total output (linear + distortion) of the AUR module (amplitude and time delay adjusted)
Regular Linear + Regular Nonlinear Distortion	Total output (linear + distortion) of the AUR module (digital model in DA using nonlinear parameters)	Stimulus (time delay and amplitude adjusted to test signal)
Regular + Irregular Nonlinear Distortion	Transducer output at high amplitudes	Transducer output at small amplitudes
All Distortion **(Regular Linear +Regular Nonlinear + Irregular)**	Transducer output at high amplitudes	Stimulus (time delay and amplitude adjusted to test signal)

Please refer to the section application for selected examples in greater detail.

Note that the signals that are used for subtraction are aligned automatically in time. The level must be aligned manually in order to obtain a physically sensible subtraction (e.g. comparing transcuder output at high and small amplitudes).

Calibration of playback equipment

Background

All signals to be exported are defined as pressure signals. To meet the required headroom for the WAVE export (signals must be in range -1..+1), the export gain can be applied.

If an absolute pressure reference is given, the wave files may be calibrated to the displayed sound pressure levels.

All signals are exported with the same export gain, thus keeping the exported signals relatively aligned. If different output signals of other DIF-AUR runs shall be used, a common export gain is beneficial.

Calibration	The exported calibration signal is played back in loop mode by the sound reproduction system of the listening experiment. The resulting sound pressure level at the receiving positions can be measured with a SPL meter. By providing a steady-state calibration signal, the measurement result is stable. The gain of the sound equipment is adjusted until the resulting SPL equals the defined sound pressure level of the calibration signal, thus compensating for the export gain G_E and realizing the necessary playback gain G_P. When the playback equipment is calibrated all signals (that where exported with the same export gain) result in the calculated sound pressure level.

Input Parameters

XR_WAV	Path of reference input signal WAVE file. Dominant against vector input.
XT_WAV	Path of test input signal WAVE file. Dominant against vector input.
XR_VEC	Copied vector ([abscissa, ordinate]) from another Klippel Module, that represents the reference input signal. Recessive against wave file input.
XT_VEC	Copied vector ([abscissa, ordinate]) from another Klippel Module, that represents the test input signal. Recessive against wave file input.
GR	[dB] Reference gain, if empty G_R = 0 dB are applied

Sdis	[dB] Distortion scaling factor, if empty S_{dis} = 0 dB.
	
	The distortion scaling factor can be given as a single number or as a matrix in Scilab format.
	
	Examples:
	• `[3]` $S_{\mathrm{dis}} = 3\ \mathrm{dB}$
	• `[-6,6]` two distortion scaling factors given: $S_{\mathrm{dis},1} = -6\ \mathrm{dB}$, $S_{\mathrm{dis},2} = 6\ \mathrm{dB}$
	• `[-18:3:18]` defines 13 distortion scaling factors with step size 3 dB:
	    ○ $S_{\mathrm{dis},1} = -18\ \mathrm{dB}$
	    ○ $S_{\mathrm{dis},2} = -15\ \mathrm{dB}$
	    ○ …
	    ○ $S_{\mathrm{dis},7} = 0\ \mathrm{dB}$
	    ○ …
	    ○ $S_{\mathrm{dis},12} = 15\ \mathrm{dB}$
	    ○ $S_{\mathrm{dis},13} = 18\ \mathrm{dB}$

Advanced	Advanced parameters allowing access to details of the DIF-AUR • Loop Loops the input signals with the given number, reasonable FFT sizes of input signals are assumed. If not defined Loop = 1 (no repetition). Example: `Loop = 10`

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- GP

  [dB or @dB_{SPL}] playback gain defines the absolute relation of the relative signals (*y*) to the pressure domain. If not defined, the absolute relation is not available. A relative playback gain should represent the gain factor of the sound reproduction system used for listening tests.

  Format:

  - Any number in dB defines a relative playback gain. If GP = 0 dB, a RMS value of 1 for the relative signals *y* results in 94 dB_{SPL} in the pressure domain.
  - @{SPL_{REF}} defines the sound pressure level of the reference pressure signal. The relative playback gain is calculated automatically to yield a reference pressure signal p_R with the average sound pressure level of {SPL_{REF}}. The calculated relative playback gain is applied to all signals mapped to the pressure domain.

  Examples:

  - `10` applies a relative playback gain of 10 dB. A relative signal *y* with RMS value 1, results in a pressure signal *p* with 104 dB_{SPL}
  - `@85` calculates a relative playback gain to yield a reference pressure signal with sound pressure level 85 dB_{SPL}. The gain is applied to all signals mapped to the pressure domain.

- GL*

  [dB or keyword] equalization gain to equalize the level of auralized signals to the reference output signal. If not defined G_L = 0 dB. Only available if an absolute relation is defined (see GP).

  Format

  - Any number in dB applies the specified gain
  - ‘LEVEL’ automatic calculation of equalization gain to obtain the same sound pressure level as the reference output signal
  - ‘LOUD’ automatic calculation of equalization gain to obtain the same loudness as the reference output signal using a perceptual model

- Lpcal*

  [dB_{SPL}] sound pressure level of calibration signal, if not defined, $L_{p,cal}$ = 80 dB. Only available if an absolute relation is defined (see GP).

- GE

  [dB or keyword] export gain used to map the relative signals in the wave file range. If not defined, G_E = ‘AUTO’.

  Format

  - Any number in dB applies the specified gain
  - ‘AUTO’ automatically selects a gain that all signals (reference output, auralization output, calibration output and difference signal) can be exported within the +/- 1 range of the wave file. The stepsize of the gain is 6 dB.

- BP_cutoff

  [Hz] two-column matrix defining cut-off frequencies for band-bass filter of difference signal

  Example

  [200 20000] defines a pass band for the FIR filter for the difference signal from 200 Hz to 20 kHz.

- Delay

  [s] Definition of known delay (test after reference) and deactivation of automatic delay estimation

- CorrThres

  [%] Definition of minimal correlation coefficient for valid delay detection, if not defined, the threshold is set to 75%

- MaxDelay

  [s] Defines a maximum delay for the automatic delay estimation. If not defined, MaxDelay is set to 5 seconds.

- IR_WAV

  File path specifying the impulse response for convolving the input reference signal before any other processing. Absolute or relative paths (related to database location) are allowed. Dominant against vector definition of impulse response. If not defined, no convolution is performed.

  Example: `IR_WAV = "C:\data\ir_room.wav"`

- IR_VEC

  Vector specifying the impulse response for convolving the input reference signal before any other processing. Recessive against wave definition of impulse response. If not defined, no convolution is performed.

  Example:

  ```
  IR_VEC = [
  0    0
  0.000028333  -0.001234
  0.000041667  0.01234
  ...
  ];
  ```

- ExportDir

  Export directory as absolute or relative (relative to database's location) path

- TimeMarker

  [s] time marker for FFT analysis

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