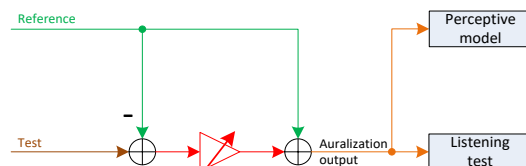


Listening tests are an important utility to define the target performance of a product. The generation of audio files that can be used for those listening tests is an essential preparation step. The examples must be selected with care since they need to be critical to transport the impairment of sound quality under investigation.

The difference auralization is an auralization technique based on decomposition of input signals. By isolating the difference of two signals virtual output signals (auralization output) with enhanced or attenuated distortion may be produced.

These audio files may be used in discussions with decision makers to define the target performance of a product or for market research with statistical valuable listening tests. This application note provides basic guides for using this auralization technique.



CONTENTS:

Background	2
Application: good versus bad prototype with music	5
Application: modeled versus measured response (TRF)	11
Application: QC pass versus fail.....	13
Application: wave versus codec (96 kbits)	13
Application: small versus large signal domain	14
Further reading	20

updated December 19, 2022

Background

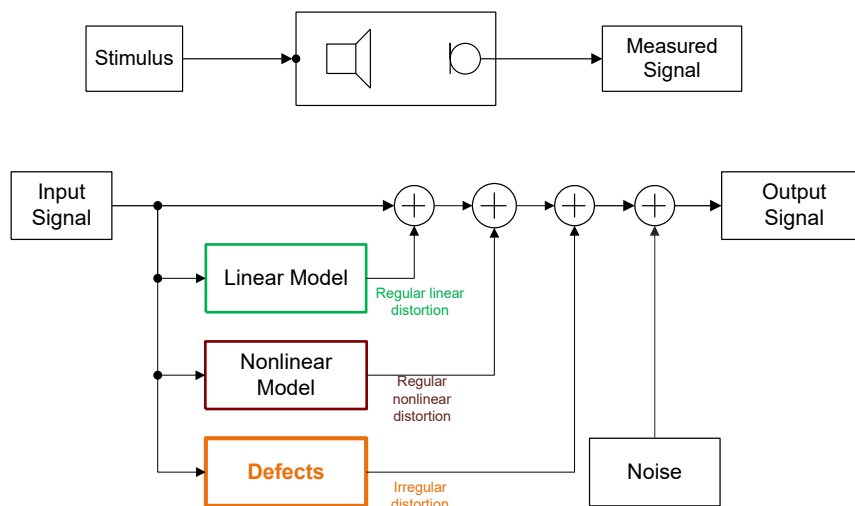
Auralization

Auralization is the generation of virtual audio signals by applying different calculation methods to input signals. Regarding distortion components of audio systems, auralization is the virtual enhancement or attenuation of certain signal components.

For audio products, several auralization techniques are available: **Model-based auralization techniques** simulate different distortion components with a model of the audio product and tab the output of each component. This technique has the drawback that it's not possible to auralize irregular distortion (like Rub&Buzz), since the irregular distortions cannot be modeled. The **difference auralization** is an auralization technique based on decomposition of input signals which is able to auralize irregular distortion components. This application note guides the reader through typical examples for the use of the difference auralization (DIF-AUR).

Distortion components in audio systems

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



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. Another linear distortion is the influence of the listening conditions (room acoustics).

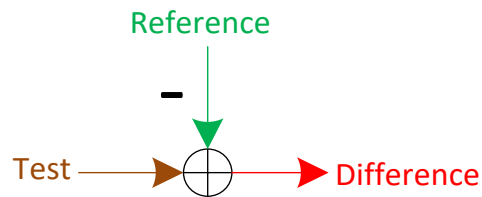
The regular nonlinear distortion can be predicted by lumped parameters of the nonlinear transducer model, the generation of this distortion component is optimized during the design process as well.

The irregular nonlinear distortion may be generated by defects (rub&buzz) in manufacturing or parasitic vibration in the final application 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.

Auralization based on decomposition

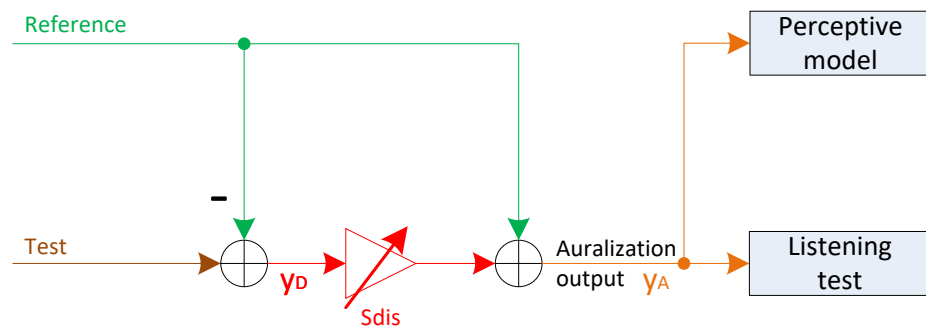
The difference decomposition technique uses two input signals (a reference input signal and a test input signal) to calculate the difference signal.



The input signals have to be aligned in time and level before the subtraction. The time alignment is performed automatically by the DIF-AUR, 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.



Selection of input signals

The difference signal (the signal components that are regarded as distortion and will be enhanced or attenuated) is defined by the choice of reference and test signal.

The table shows typical choices for the input signals along with the distortion components comprised in the difference signal. Please also refer to the signal flow plan in *Distortion components in audio systems* above.

<i>Difference Signal</i>	<i>Test signal</i>	<i>Reference signal</i>
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)

Application: good versus bad prototype with music

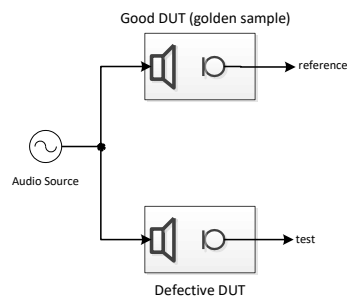
Device under test

The device under test is a loudspeaker product where a good and a bad unit (Rub&Buzz defect) are available.

Two measurements are performed under the same measurement conditions (see below): the response to the stimulus of the good speaker is used as reference input signal, the response of the defective one as test input signal.

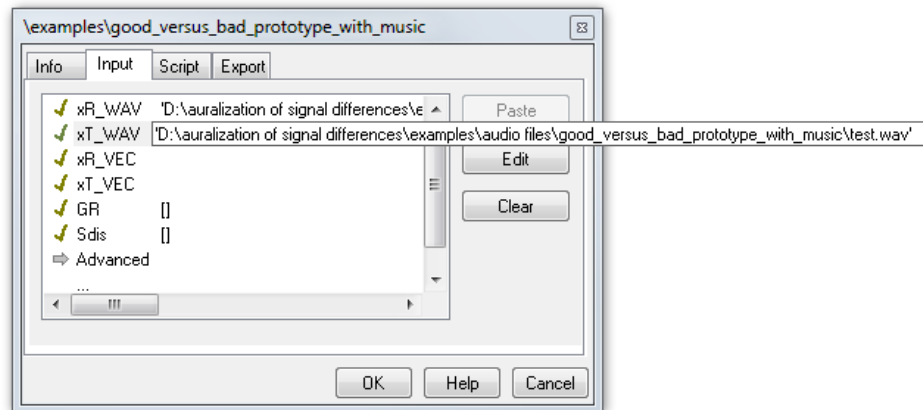
Measurement conditions

Music is used as a stimulus. It's important that the selected stimulus excites the defect in the defective speaker. The input voltage, microphone and speaker positions have to be equal in both measurements. Using microphone and speaker stands and measuring in near field is beneficial for this application.

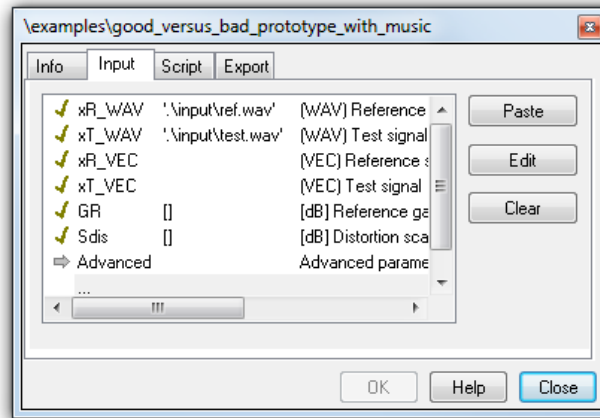


Import of signals in DIF-AUR

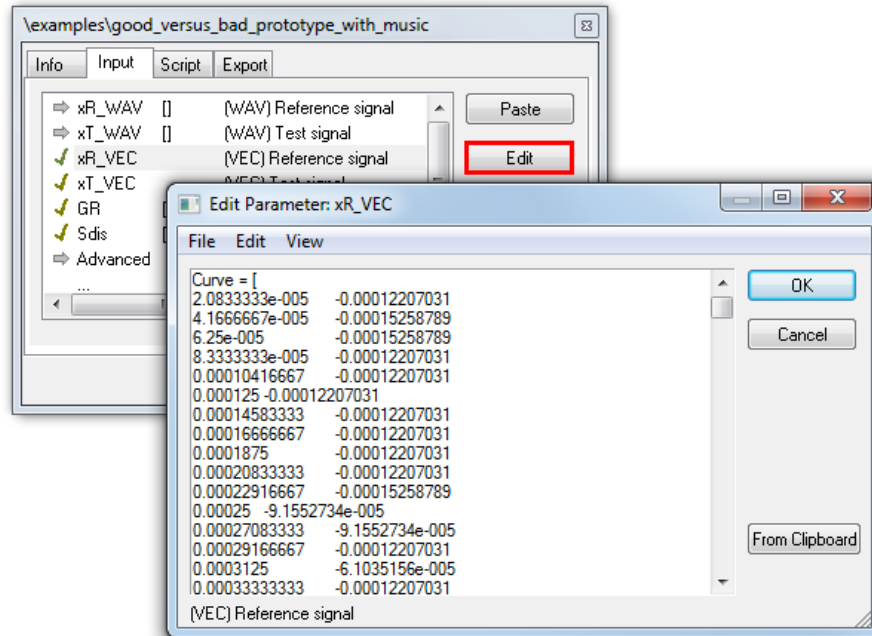
If the recorded responses are available as wave files, the path can be easily copied with “copy as path” by using SHIFT+RIGHT CLICK on the file in Windows Explorer. The path is pasted to the input fields of reference and test WAV input.



Relative paths (relative to the database location) are also allowed. In the example the files are located in a folder *input* parallel to the database.



Note: The delivered example uses the vector input (instead of the wave file input) to provide the input signals. The time signals are included in the delivered database:



BTW: The input curves that are defined as vectors are visible in the window *Input Curves*:

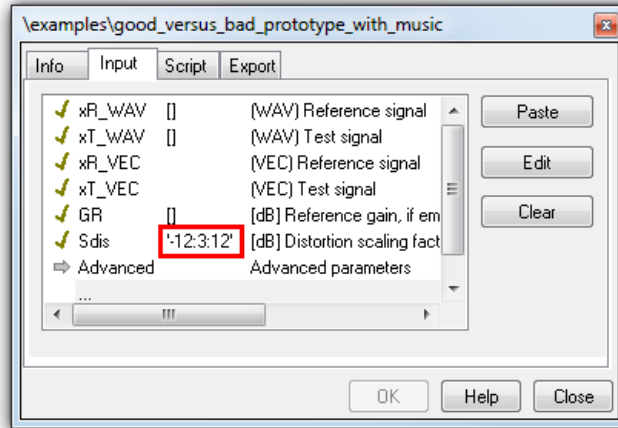


Note: the information contained in vector input is available as wave files after the import: The reference signal is exported separately. The input test signal equals the auralization output @ $S_{dis}=0$ dB.

Parameterization of DIF-AUR

Since the measurement conditions for obtaining the input signals are equal, there is no alignment in level necessary, we'll leave G_R empty, which results in a neutral gain of 0 dB.

The distortion scaling factor S_{dis} is of high interest. If it's leaved empty, the scaling is set to 0 -dB. It's beneficial to produce multiple auralization output signals to get a scaling of the distortion component. The delivered database uses a scaling from -12 dB to +12 dB in 3 dB steps:



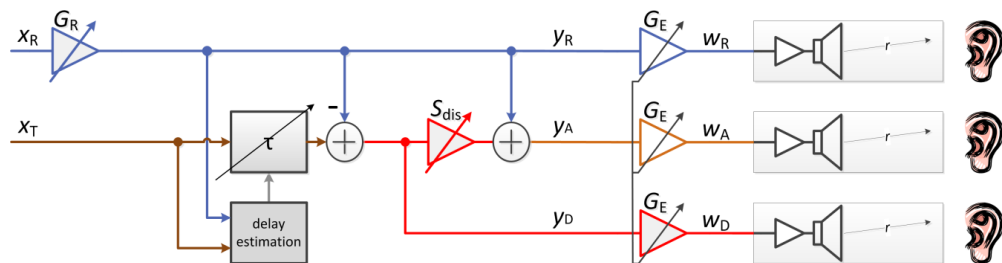
The advanced parameters are not used in this example.

Press the start button for the DIF-AUR to start calculation and export.



Basic signal flow plan

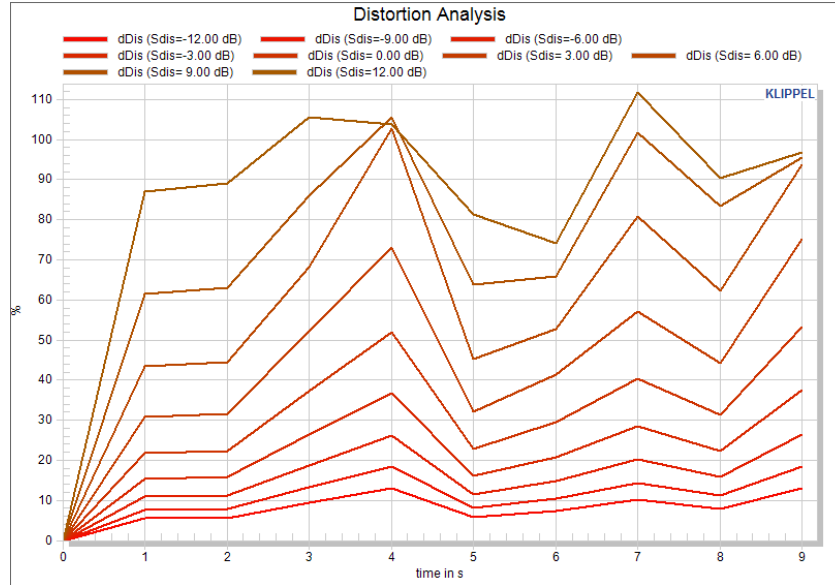
The basic signal flow plan shows important symbols of imported and exported signals and gain stages. Please note that the signal y_A is dependent on S_{dis} . If multiple distortion scaling factors are given, multiple signals $y_{A,S_{dis}}$ exist.



Results

The only results (beside the audio files) of the module are some HTML tables and the distortion analysis providing a general overview.

The distortion analysis displays the amount of contained distortion in the auralization output versus time to estimate the moment of maximum distortion.



The results shown in the HTML output provide information about the output and input signals and allow fast access to important files:

The link [view export directory](#) opens the export directory of this operation.

The table *Signal characteristics for wave export* shows valuable information about the exported wave files.

Signal characteristics for wave export

Output (WAVE)	Symbol	Level _{RMS}	Level _{Peak}	Level _{A-weighted}
Reference	w_R	-27.6 dB	-5.4 dB	-27.9 dB(A)
Auralized ($S_{dis}=-12$ dB)	$w_{A,S_{dis}=-12}$ dB	-27.6 dB	-5.4 dB	-27.9 dB(A)
Auralized ($S_{dis}=-9$ dB)	$w_{A,S_{dis}=-9}$ dB	-27.6 dB	-5.4 dB	-27.9 dB(A)
Auralized ($S_{dis}=-6$ dB)	$w_{A,S_{dis}=-6}$ dB	-27.6 dB	-5.4 dB	-27.9 dB(A)
Auralized ($S_{dis}=-3$ dB)	$w_{A,S_{dis}=-3}$ dB	-27.6 dB	-5.4 dB	-27.9 dB(A)
Auralized ($S_{dis}=0$ dB)	$w_{A,S_{dis}=0}$ dB	-27.6 dB	-5.4 dB	-27.9 dB(A)
Auralized ($S_{dis}=3$ dB)	$w_{A,S_{dis}=3}$ dB	-27.5 dB	-5.4 dB	-27.8 dB(A)
Auralized ($S_{dis}=6$ dB)	$w_{A,S_{dis}=6}$ dB	-27.4 dB	-5.5 dB	-27.8 dB(A)
Auralized ($S_{dis}=9$ dB)	$w_{A,S_{dis}=9}$ dB	-27.2 dB	-5.5 dB	-27.6 dB(A)
Auralized ($S_{dis}=12$ dB)	$w_{A,S_{dis}=12}$ dB	-26.8 dB	-5.5 dB	-27.3 dB(A)
Difference	w_D	-46.1 dB	-18.0 dB	-48.2 dB(A)

The *Gain settings* show our gain stages in the signal flow plan (see above).

Gain settings

Name	Symbol	Gain
Reference Gain	G_R	0.0 dB
Export Gain	G_E	15.0 dB

We see our reference gain G_R was set to 0 dB (because it was left empty) and the export gain G_E was set automatically to 15 dB to obtain wave files with optimal headroom.

The *Input signal characteristics* provide some information about the input signals. Here we have data sampled at 48 kHz. The automatic delay detection detected no delay.

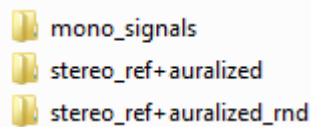
Input signal characteristics

Input	Symbol	f_s	Peak	RMS	Length	Delay
Reference input signal	x_R	48000 Hz	-20.4 dB	-42.6 dB	9.34 s	-
Test input signal	x_T	48000 Hz	-20.4 dB	-42.6 dB	9.34 s	0.000000 s (0 samples)

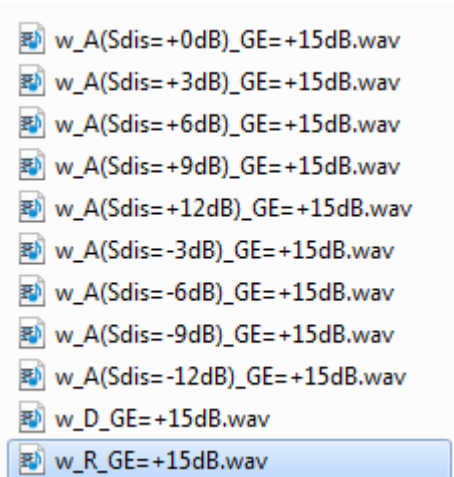
Output signals

After the execution is finished, you may notice a new folder parallel to the opened database. It contains calculation data and audio files. The folder hierarchy relates to the database name and the operation path inside the database.

Three different export configurations are used for the audio files:

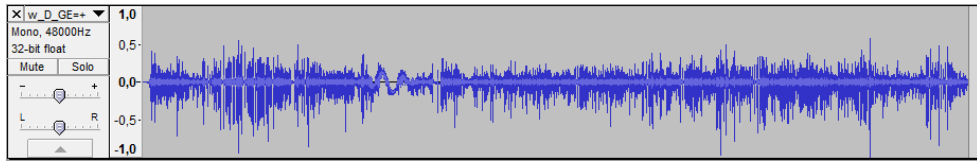


If you click on the links, an explorer window will open with the relevant file selected.



Mono signals contain the individual signals in separate files. The stereo signals contain one auralization output signal along with the reference signal. A fixed and random channel configuration is available.

Check the difference signal; it should not contain significant musical components, just the isolated defect. Open the file in a wave editor (e.g. [Audacity](#)) and increase the amplitude if necessary (e.g. Edit→Select→All, Effect → Normalize). Listen to the isolated defect.



If no musical components are audible, the isolation of the irregular defect was ok. Please note that the quality of isolation depends on the quality of the input signals (and their equality of measurement conditions).

The isolation with the delivered data probably works fine at your computer. If you use your own signals and the isolation fails, please check

- The detected delay: does it correspond to the input wave files? To double-check, open the files in a wave editor and zoom in on a distinct peak available in both signals.
- The level alignment: do the amplitudes correspond to each other?

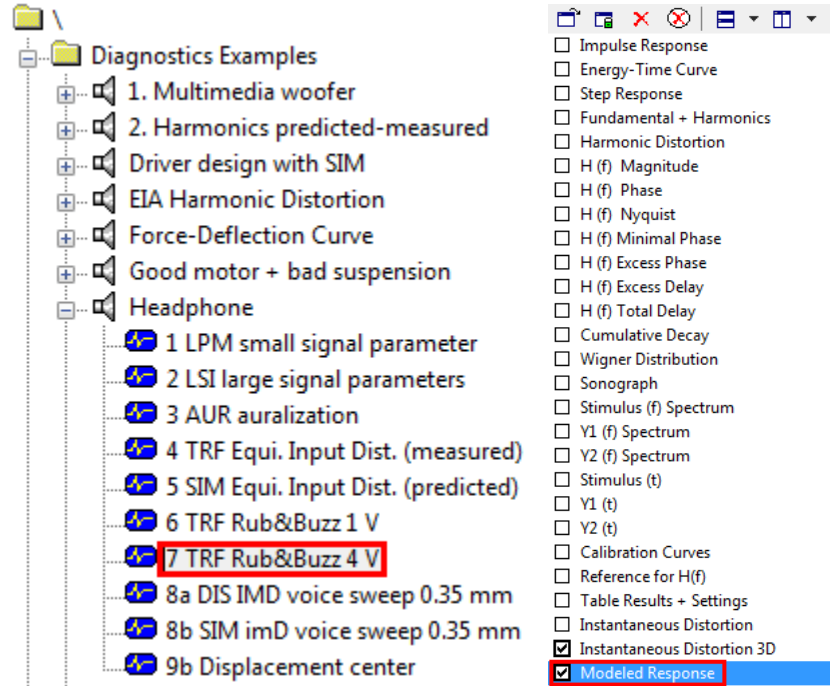
Application: modeled versus measured response (TRF)

Device under test

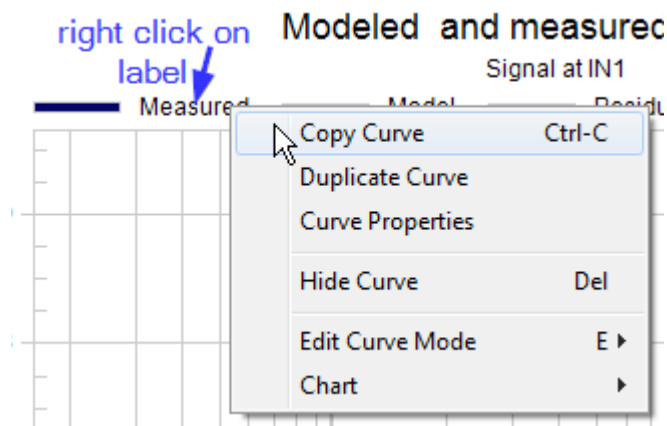
A speaker measured with Klippel RnD TRF module showing Rub&Buzz symptoms. Only one measurement is necessary. The modeled response is used as reference input signal, the measured response is used as test input signal.

The delivered example uses time signals taken from the RnD database delivered along with every RnD installation. The operation's path is `\Diagnostics Examples\Headphone\7 TRF Rub&Buzz 4 V`.

The corresponding window is *Modeled Response*. You can zoom in and compare the modeled and measured response visually.



If you want to use your own TRF data, just copy the modeled and measured response curves and paste them to the DIF-AUR module.

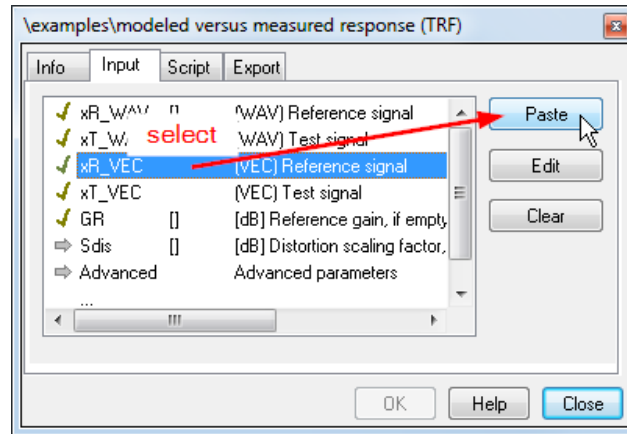


Measurement conditions

The TRF uses a sweep as stimulus. It's important that the modeling identifies the Rub&Buzz as residual correctly.

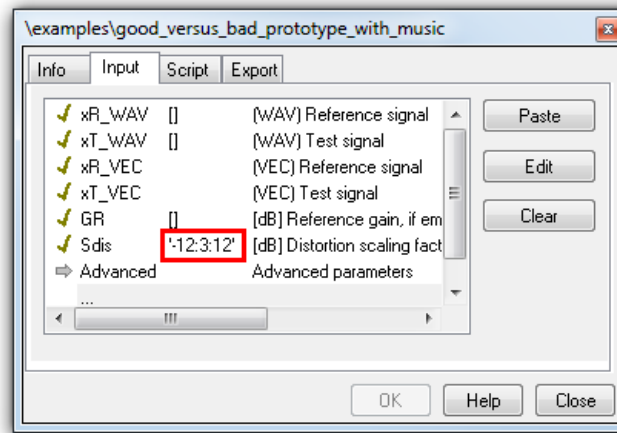
Import of signals in DIF-AUR

Copy the modeled and measured responses as described above and paste the vectors in xR_VEC and xT_VEC.



Parameterization of DIF-AUR

For the example we're using a distortion scaling from -12 dB to +12 dB in 3 dB again.



Press the start button to produce the output signals with the DIF-AUR.



Results and output signals

The results are the same as the previous example - the output signals obviously are different.

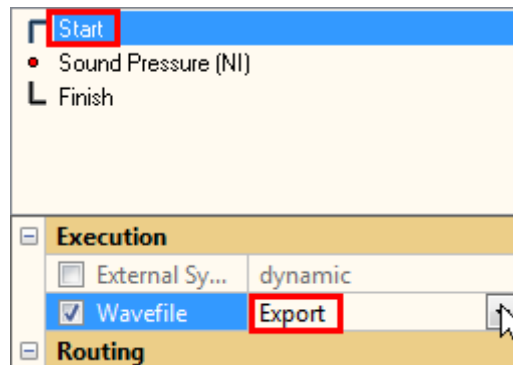
Application: QC pass versus fail

Obtaining the input signals

Comparing QC responses of good and bad DUTs may be beneficial to identify problems. The operation conditions of different DUTs are very similar in QC measurements.

Please note that the measured response in a QC test box does not necessarily represent the final application area.

Having a QC test, the measured responses may be exported with the WAVE export feature (available for QC v4 and above):



The resulting wave files may be imported into DIF-AUR directly.

This way it's possible to auralize all irregular defects: loops particles, rubbing, air leakage...

Description

The example uses vector input of a passed and failed unit of the QC test. Loose particles are simulated with grains of salt.

Please note that writing WAVE files during production check is not recommended. If the WAVE export is not needed, it should be deactivated due to performance reasons. Of course it's possible to auralize all kind of irregular nonlinear distortion for debugging or training purposes.

Application: wave versus codec (96 kbits)

Audio codecs

Lossy audio codecs (like mp3, ogg vorbis, ...) remove data from an audio signal that is regarded as irrelevant data under psycho-acoustical aspects. Hence the stored data and the necessary bandwidth for transferring files is minimized.

The audibility of the codec's impact on the audio signal strongly depends on the applied codec algorithm and on the bitrate used for the data reduction.

Description

This example isolates compression artifacts from an audio codec (mp3). The output data reveals the isolated artifacts.

The approach may be used to identify audio codec artifacts and finding out how they need to be scaled in order to become audible. Is it necessary to reduce the impact because the codec's impact is audible without scaling ($S_{dis} = 0$ dB), or is there some "safety headroom" because the distortion needs to be enhanced in order to become audible?

Loudness equalization ensures equally scaled which is important to obtain meaningful results from a comparison in a listening test. In this example, the level alignment was set to "level", resulting in wave files with the same average level. An equalization to the same loudness using a perceptual model is also available in the DIF-AUR.

Application: small versus large signal domain

Problem

Sometimes it is not possible to position the microphone exactly. For automobiles comparing good versus bad products is not very handy. If exactly equal conditions (including room acoustics) cannot be guaranteed, one will not obtain good reference and test input signals for the difference decomposition technique.

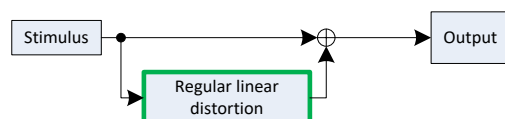
The sound quality is not only influenced by the transducer's quality, but also by vibrating parts that are excited by air or structure borne sound. This parasitic vibration shall be auralized in this example. To excite the symptoms (parasitic vibrations) but to avoid masking the symptoms, the music is only played through the subwoofer channel of the automobile in the delivered example.

Approach

A solution to this problem is to perform the two measurements with the same device under test - with the same stimulus and the exact same microphone position, but with different stimulus levels.

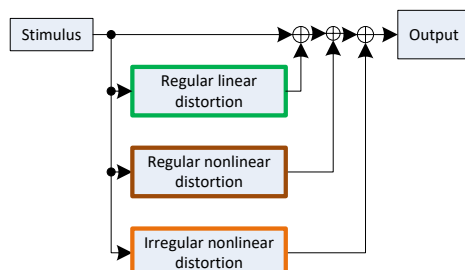
Reference signal

One measurement is performed with a low stimulus level. The loudspeakers operate in small signal domain. Regular and irregular nonlinear distortions from the loudspeakers are not excited. Parasitic vibrations (which are also assigned to the group of irregular nonlinear distortion) are neither excited. The response is used as reference input signal for the difference auralization.



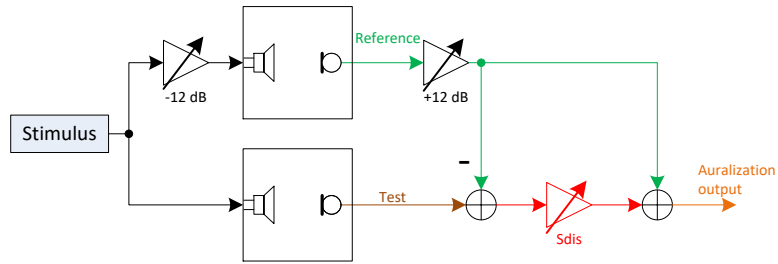
Test signal

The second measurement is realized with a high stimulus level. Exciting regular nonlinear distortions of the loudspeaker and irregular nonlinear distortion (e.g. parasitic vibrations of door panels). This large signal domain response is used as test input signal for the difference auralization.



Difference signal

Comparing the two signals requires a level alignment before the subtraction. In this example the small signal domain measurement is performed with 12 dB attenuation of the full amplitude that is used for the large signal domain measurement. The gain G_R is used to compensate for the level difference of the reference input signal.



This way parasitic vibration is isolated. The regular nonlinear distortion component is also contained in the difference signal but is neglected in this case (see *Filtering*).

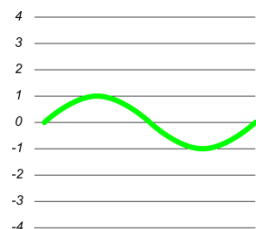
The noise component is also neglected.

Filtering

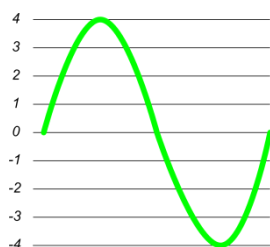
Due to the different stimulus amplitudes regular nonlinear distortion components are also included in the difference signal. This includes the amplitude compression of the fundamental, which has an impact on the auralized signals. The compression is mainly caused by the suspension and present at low frequencies.

Watch the impact of amplitude compression step by step:

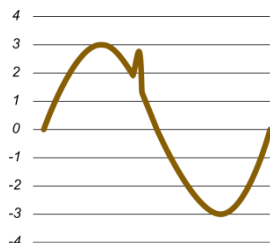
1. You get the reference signal with the small signal domain measurement (minus 12 dB). The signal (for illustration only a sine period) is not distorted.



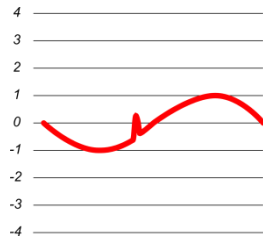
2. The reference signal is scaled by 12 dB to compensate for the small signal domain measurement.



3. The test signal is recorded in large signal domain (no -12 dB scaling before the measurement). You get a higher amplitude than in the small signal domain measurement and you get some irregular nonlinear distortion (the peak) which are excited in large signal domain. But another thing happens: the loudspeaker has regular nonlinearities (e.g. loss of force factor and increase of stiffness) which limit the displacement and the fundamental component.

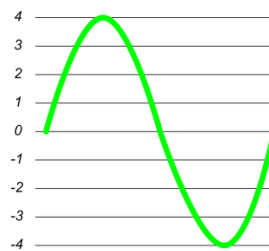


4. Let's calculate the difference signal. We get the peak, which is very good since this is the thing we're interested in. Also we get a sinusoidal component, which is only due to amplitude compression. Since this is a natural self-protection of the loudspeaker we're not so interested in getting that component in the auralization output.

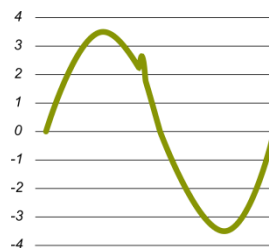


5. Now let's calculate the auralization output:

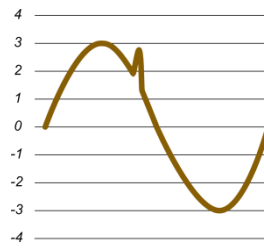
- a. For $S_{dis} = -\infty$ dB, the auralization output equals the scaled reference input signal.



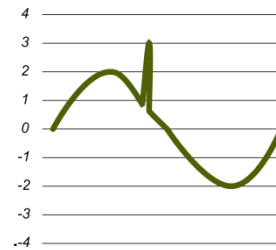
- b. For S_{dis} in the range between $-\infty$.. 0 dB, only a part of the difference signal is added. The peak is scaled from zero (reference) to the size of the test signal. The amplitude of the sinusoidal component is compressed from reference signal to test signal.



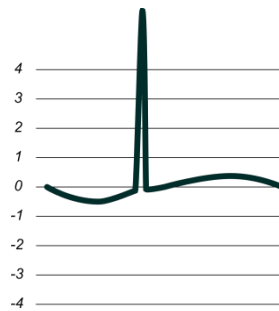
c. For $S_{dis}=0$ dB, the auralization output equals the input test signal.



d. For $S_{dis}>0$ dB, the peak is enhanced, but the sine component is reduced even further ...



e. Eventually, the sinusoidal component flips over the zero axis.

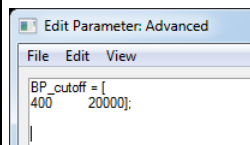


What are the consequences?

- The impact of amplitude compression is scaled with all other components.
- For S_{dis} between $-\infty$ and 0 dB the effect is scaled from not present (reference) to reality (test). Since the amplitude compression is present at low frequency, this reflects in a loss of base components.
- For higher values of S_{dis} the low frequency components are reduced even further until they are inverted and the auralization output obtains more bass again.
- In most cases the user is interested in irregular nonlinear distortion, not the effect of amplitude compression. Hence, a filtering of desired components of the difference signal is recommended.

One symptom of this effect is a decreasing (orange) level followed by an increasing (red) level of the auralized signals.

Output (WAVE)	Symbol	Level _{RMS}	Level _{Peak}
Reference	w_R	-34.3 dB	-22.5 dB
Auralized ($S_{dis}=-24$ dB)	$w_{A,S_{dis}=-24}$ dB	-34.5 dB	-22.8 dB
Auralized ($S_{dis}=-21$ dB)	$w_{A,S_{dis}=-21}$ dB	-34.6 dB	-22.9 dB
Auralized ($S_{dis}=-18$ dB)	$w_{A,S_{dis}=-18}$ dB	-34.7 dB	-23.1 dB
Auralized ($S_{dis}=-15$ dB)	$w_{A,S_{dis}=-15}$ dB	-34.9 dB	-23.3 dB
Auralized (-35.1 dB	-23.6 dB
Auralized (decreasing level	-35.4 dB	-24.1 dB
Auralized ($S_{dis}=-6$ dB)	$w_{A,S_{dis}=-6}$ dB	-35.9 dB	-24.9 dB
Auralized ($S_{dis}=-3$ dB)	$w_{A,S_{dis}=-3}$ dB	-36.5 dB	-26.0 dB
Auralized ($S_{dis}=0$ dB)	$w_{A,S_{dis}=0}$ dB	-37.4 dB	-26.8 dB
Auralized ($S_{dis}=3$ dB)	$w_{A,S_{dis}=3}$ dB	-38.3 dB	-27.2 dB
Auralized ($S_{dis}=6$ dB)	$w_{A,S_{dis}=6}$ dB	-38.4 dB	-26.0 dB
Auralized ($S_{dis}=9$ dB)	$w_{A,S_{dis}=9}$ dB	-36.5 dB	-23.3 dB
Auralized (increasing level	-32.9 dB	-19.4 dB
Auralized (-29.0 dB	-14.9 dB
Auralized ($S_{dis}=18$ dB)	$w_{A,S_{dis}=18}$ dB	-25.2 dB	-11.0 dB
Auralized ($S_{dis}=21$ dB)	$w_{A,S_{dis}=21}$ dB	-21.7 dB	-7.4 dB
Auralized ($S_{dis}=24$ dB)	$w_{A,S_{dis}=24}$ dB	-18.3 dB	-4.0 dB
Difference	w_D	-41.3 dB	-27.0 dB



The delivered example uses a band-pass with cut-off frequencies 400 Hz and 20 kHz.

Further reading

- Specification S22 Difference Auralization
- [Paper Combining Subjective and Objective Assessment of Loudspeaker Distortion](#)
- [AUR: Model-Based Auralization](#)

updated December 19, 2022



Klippel GmbH
Mendelssohnallee 30
01309 Dresden, Germany

www.klippel.de
info@klippel.de

TEL: +49-351-251 35 35
FAX: +49-351-251 34 31