AURALIZATION OF SIGNAL DISTORTION IN AUDIO SYSTEMS PART 1: GENERIC MODELING

WOLFGANG KLIPPEL

Klippel GmbH, Germany, www.klippel.de

Auralization techniques are developed for generating a virtual output signal of an audio system where the different kinds of signal distortion are separately enhanced or attenuated to evaluate the impact on sound quality by systematic listening or perceptive modeling. The generation of linear, regular nonlinear and irregular nonlinear distortion components is discussed to select suitable models and measurements for the auralization of each component. New methods are presented for the auralization of irregular distortion generated by defects (e.g. rub & buzz) where no physical models are available. The auralization of signal distortion is a powerful tool for defining the target performance of an audio product in marketing, developing products at optimal performance-cost ratio and for ensuring sufficient quality in manufacturing.

1 INTRODUCTION

Auralization is a modern technique for generating a virtual acoustical output by using measurement or modeling which can be assessed by the human ear or by a perceptive evaluation system. This technique is not limited to sound reproduction and room interaction at low amplitudes where the system behaves linearly [1] but can also be used for evaluating distortion generated by nonlinearities at high amplitudes [2-4]. The auralization of signal distortion is very interesting because the distortion should be just inaudible for most listeners. This new technique is the basis for investigating the following issues:

- How much enhancement or attenuation of the distortion is required to make the distortion just audible?
- What is the impact of the distortion on the perceived sound quality?
- Are there any desired effects of nonlinear distortion?
- How to define the target performance of the audio system according to the final application?
- How can the performance/cost ratio be increased?
- What are the benefits of additional development efforts?
- How to select the best design choice?
- What is the most critical program material in listening tests?
- How to define PASS/FAIL limits in end-of-line testing?

Those issues will be addressed in three parts. The first paper presents techniques for auralizing all kinds of signal distortion including artifacts caused by loudspeaker defects (rub & buzz) where a physical modeling of the generation process is impossible. The following paper develops special auralization schemes for loudspeaker distortion generated by regular nonlinearities related to the design of the transducer. The last paper in this series discusses the practical application of the auralization in marketing, development and manufacturing of audio products.

2 SIGNAL DISTORTION

An audio system (e.g. a loudspeaker) excited by a stimulus $u(t)$ such as a test signal or music generates an output signal (e.g. the sound pressure)

$$\begin{aligned} p(t) &= \alpha u(t-\tau_0) \\ &+ d_{lin}(t) + d_{nlin}(t) + d_{irr}(t) + n(t) \end{aligned} \tag{1}$$

comprising the time delayed and scaled input signal $u(t)$, linear distortions $d_{lin}(t)$, regular nonlinear distortions $d_{nlin}(t)$, irregular nonlinear distortions $d_{irr}(t)$ and noise $n(t)$ as illustrated by a signal flow chart according in Figure 1.

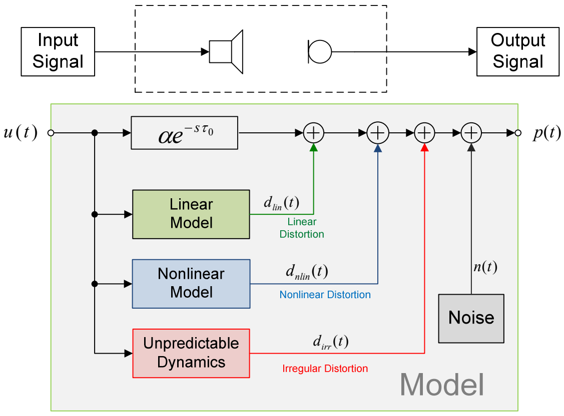

Figure 1: Generation of signal distortion in audio systems

A frequency independent gain factor α and a constant time delay τ_o generated by the audio system or by the sound propagation between source and listening point are not considered as signal distortion. The linear distortion component $d_{lin}(t)$ is generated by the electro-acoustical transduction and the sound propagation in the acoustical environment (e.g. room).

At higher amplitudes the nonlinearities in the transducer generate the nonlinear distortion $d_{nlin}(t)$, which appear as new spectral components in the output signal. However the nonlinearities in the motor and mechanical suspension are considered as regular [5] because they are predictable and directly related with the design of the transducer. Usually a compromise between cost, weight, size and sound quality is required to create a product which satisfies the needs of the user .

The irregular distortions $d_{irr}(t)$ are generated by uncontrolled mechanisms (e.g. rocking mode) or defects caused by the manufacturing process, ageing and other external impacts (e.g. overload, climate) during the later life cycle of the product. Furthermore, loose particles, a rubbing coil and turbulent air flow in enclosure leaks cause artifacts which are random and not predictable. Irregular distortions $d_{irr}(t)$ have a transient waveform comprising high-frequency components increasing the sharpness and roughness of the reproduced sound which is not desired in most applications. The noise component $n(t)$ may be generated by the sensor used to acquire the output signal $p(t)$ or by external noise source in the acoustical environment.

The results of the distortion measurements highly depend on the properties of the stimulus $u(t)$ exciting the audio system under test. Although some measurement techniques (e.g. incoherence) are capable of assessing regular nonlinear distortion while reproducing music or speech most techniques use a special test signal (e.g. sinusoidal chirp) to measure nonlinear symptoms at the highest sensitivity and speed. Furthermore the metric of the characteristics derived from physical data does not correspond with the results of perceptive evaluation of the audio system. The psycho-acoustical processing of the signal in the ear and in upper cognitive layers of the brain determine the audibility of the distortions, their annoyance and the final impact on the perceived sound quality of the audio reproduction.

To overcome the limits of conventional instruments based on physical measurements new kinds of objective evaluation techniques have been developed which consider the transmission of the signal in the peripheral ear, time-frequency decomposition, generation of an excitation pattern and the extraction of features (MOVs) describing loudness, sharpness, roughness and other basic perceptive attributes. A perceptive assessment technique (PEAQ [7]) and auralization techniques [8] have been developed for audio CODECs that are not directly applicable to loudspeakers and other components of the audio system. Systematic listening tests which are a time consuming procedure are required for evaluating the preference or annoyance of particular signal distortion found in audio systems and their impact on the perceived overall sound quality. For example a moving-coil transducer may generate significant intermodulation between a low frequency bass signal and other high-frequency components that are perceived as fluctuations or a disturbing roughness of the sound. Cognitive models summarize those basic perceptional features in a multi-dimensional space using ideal points influenced by experience, training and cultural background of the listener. Some perceptional features (e.g. loudness) are dominant and may mask other features (e.g. spectral colorations) in overall grading. It is known that the perception is an adaptive learning process and some properties (e.g. room influence) which are constant during the test become less important over time.

3 AURALIZATION TECHNIQUES

The auralization scheme generates a virtual output in which each distortion component can be enhanced or attenuated by an arbitrary scaling factor. The individual scaling of the distortion components require a separation process which can be realized in three ways:

- Separation of linear and nonlinear parameters
- Separation of linear and nonlinear subsystems
- Separation of a test and a reference signal

3.1 Separation by Model Parameters

This technique requires a model of the audio system and varies the values of the model parameters as illustrated in Figure 2. The variation of linear parameters P_{lin} such as the impulse response of a FIR filter, the lumped parameters of an equivalent circuit of a transducer or gain, loss factor and resonance frequency of a parametric equalizer can be used to investigate the audibility of peaks and dips in the frequency response. The large signal behavior can be modified by varying the curve shape of the force factor $Bl(x)$, mechanical stiffness $C_{ms}(x)$ and other nonlinear lumped parameters P_{nlin} in transducer modeling or the values of the nonlinear coefficients in polynomial filters using the Volterra-series, neural networks or other generic structures using linear and nonlinear subsystems.

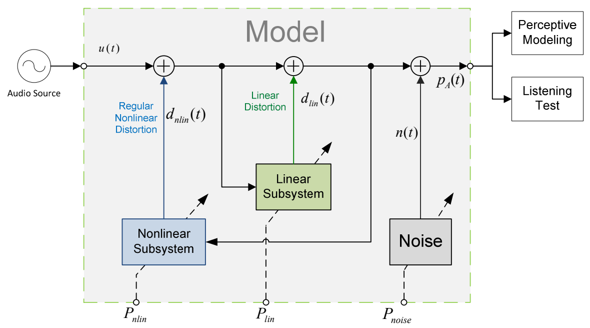

Figure 2: Auralization of signal distortion by varying the parameters of linear and nonlinear systems and a noise generator

3.2 Separation by Subsystems

The signal flow chart as given in Figure 2 is also a good basis for separating linear and nonlinear subsystems which represent the generation of distortion components $d_{lin}(t)$, $d_{nlin}(t)$ and $n(t)$. Scaling the output of each subsystem by gains S_{lin}, S_{nlin} and S_{noise}, respectively, is the basis for an alternative auralization scheme as illustrated in Figure 3.

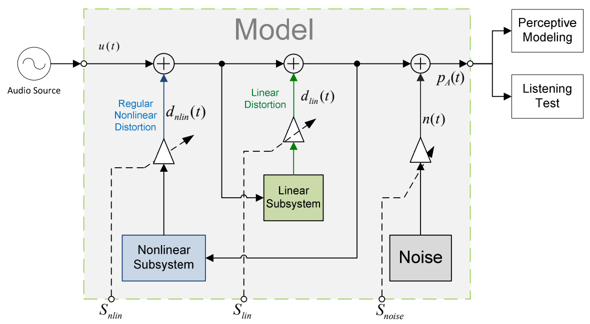

Figure 3: Auralization of signal distortion by scaling the output of linear and nonlinear subsystems and a noise generator

The nonlinear subsystem is part of a feedback loop and the nonlinear distortion component $d_{nlin}(t)$ is added to the input signal $u(t)$. Such a feedback structure corresponds with the differential equation or the poles in a rational transfer function which is the result of transducer modeling at low frequencies. The variation of both the nonlinear parameter P_{nlin} and scaling factor S_{nlin} affects all state variables in the model including the generation of the linear distortion d_{lin} in the linear subsystem. For example the auralization of the distortion generated by a varying mechanical compliance will also change the voice coil displacement x and the depending the nonlinear force factor $Bl(x)$ generating other kinds of distortion.

This interaction can be avoided by tapping the linear signal from the input and the distortion components from the output of the subsystems and synthesizing a virtual output in an external mixing console as illustrated in Figure 4. The scaling of the signals occurs outside of the model and will not affect the internal states and the distortion generation process. If all scaling factors are zero the virtual output $p_A(t)$ equals the undistorted input $u(t)$. Setting all scaling factors to one the virtual output will become identical with the output $p(t)$ of the model. If the linear distortion signal $d_{lin}(t)$ comprises a frequency component which is in anti-phase to the input signal $u(t)$ the amplitude of this particular frequency component will vanish in the auralization output $p_A(t)$ for a particular value of $S_{lin}>0$. For this reason the auralization based on parameter variation is more suitable for the evaluation of linear distortion. Harmonics and intermodulation distortion, noise and irregular distortion which are highly incoherent with the input signal will not generate a noticeable cancellation effect. However, the regular nonlinear distortion $d_{nlin}(t)$ can interfere with the input signal $u(t)$. For example, the mechanical stiffness $K_{ms}(x)$ and the force factor nonlinearity $Bl(x)$ generate also a fundamental components that is in anti-phase with the input $u(t)$ for frequencies below resonance and will decrease the voice coil displacement in the feedback loop of the model. The feed-forward structure of the mixing block may also cause a null in the fundamental component for a particular scaling factor S_{nlin}. Fortunately, this effect occurs usually at very high values of S_{nlin} far above the audibility threshold of the distortion components.

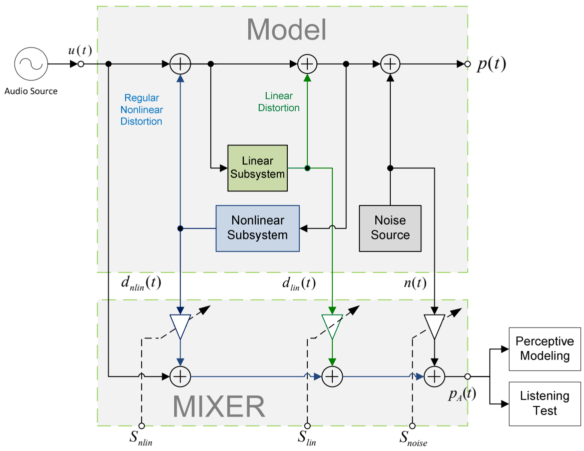

Figure 4: Auralization of signal distortion by tapping the output of the subsystems

The modeling by linear and nonlinear subsystems is very powerful for describing the regular transducer nonlinearities in the electrical and mechanical domain [10]. Those nonlinearities depend on the voice coil displacement, velocity and electrical input current and may be represented by a nonlinear subsystem adding distortion d_{nlin} to the input signal $u(t)$ as illustrated in Figure 5. A following post-filter $H(f)$ describes the linear dynamic of the audio system such as electro-acoustical transduction and sound propagation to the receiving point in the sound field. This model is the basis for measuring the *equivalent input distortion* as described in [10]. If all nonlinearities inherent in the audio system are located in the one-dimensional signal path the nonlinear distortion can be measured at any receiving point in the sound field and transferred by filtering with the inverse transfer function $H(f)^{-1}$ to the signal $d_{nlin}(t)$ at the output of the nonlinear system.

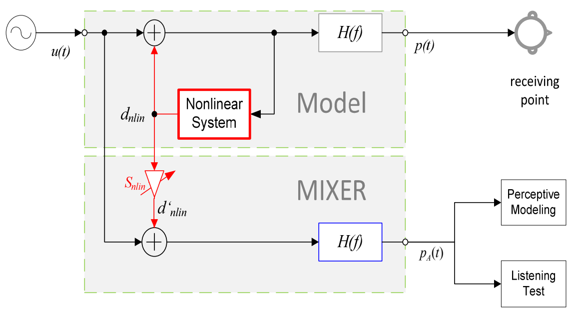

Figure 5: Auralization of regular nonlinear distortion generated by nonlinear differential equation by tapping the input signal u(t) and the equivalent input distortion $d_{dis}(t)$.

This model is a suitable basis for synthesizing a virtual auralization output $p_A(t)$ by applying a simple post-filtering with $H(f)$ to the scaled equivalent input distortion $d'_{nlin}=S_{nlin}d_{nlin}$ added to the linear input $u(t)$.
This and the previous discussed auralization schemes are relatively simple but very powerful because they are directly derived from the physical modeling of the audio system.
If a reliable theory on the particular distortion mechanism is not available a generic model has to be applied. A good candidate is the polynomial filter comprising a linear, quadratic and higher-order homogenous power subsystem according to the Volterra series as illustrated in Figure 6.
The linear system $H_1(f)$ can be realized by a FIR filter comprising a delay line and a linear combiner that scales the delayed input signal and generates the linear distortion signal $d_{lin}(t)$. The quadratic subfilter $H_2(f_1,f_2)$ uses the linear combiner for scaling and adding the products of two signals in all combinations of the time delay. The nth-order subfilter $H_n(f_1,f_2,\ldots f_n)$ multiplies n signals with each other which are tapped at the delay line. The regular nonlinear distortion d_{nlin} corresponds with the sum of the outputs of all nonlinear subsystems.
The predicted output signal

$$p'(t) = \alpha u(t-\tau_0) + d_{lin}(t) + d_{nlin}(t) \qquad (2)$$

comprising the time delayed and scaled input signal and the modeled linear and regular nonlinear distortion. The error signal

$$e(t) = p(t) - p'(t) \approx d_{irr}(t) + n(t) \qquad (3)$$

is the difference between measured and estimated sound pressure output and reflects the irregular distortion, noise and imperfections of the model.

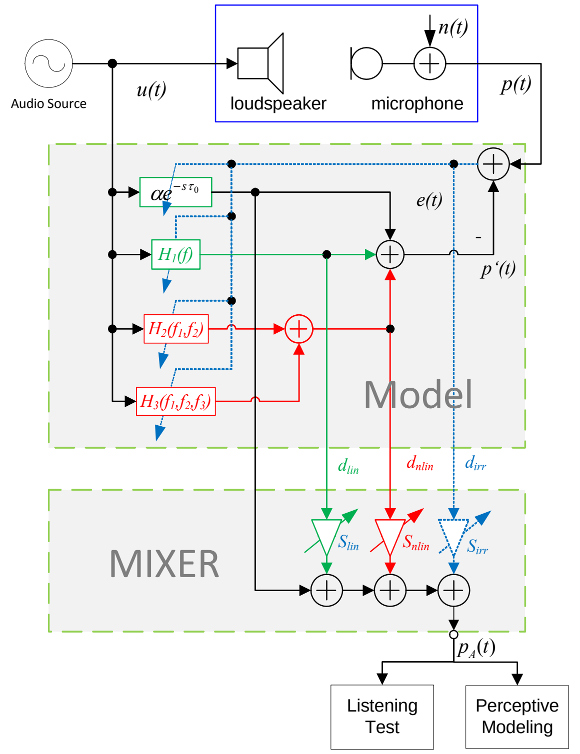

Figure 6: Auralization of regular nonlinear signal distortion by tapping the output of adaptive polynomial subsystems

The gain α, time delay τ and the weights in the linear combiners in the polynomial filter are the free parameters which can be determined by an adaptive algorithms (e.g. LMS) using the error signal *e(t)*. When the free parameters have been converged to the optimal estimates and the error signal *e(t)* becomes minimal the the linear distortion $d_{lin}(t)$ and the regular nonlinear distortion can be used for synthesizing a virtual auralization output $p_A(t)$. If the output *p(t)* is measured at high signal-to-noise-ratio and the order of the polynomial filter is sufficient to describe the regular nonlinear behavior of the audio system the error signal *e(t)* will reflect the irregular distortion $d_{irr}(t)$ of the audio system. In this case the error signal *e(t)* will be scaled and added to the auralization output.

3.3 Separation by a Reference System

Using the error signal *e(t)* according Eq. (3) for generating a virtual auralization output p_A belongs to a further auralization scheme that uses the difference between a test signal $x_T(t)=p(t)$ and a reference signal $x_R(t)=p'(t)$ as illustrated in Figure 6. Only this approach is capable of scaling irregular distortion $d_{irr}(t)$ in the auralization output $p_A(t)$. There is no need for finding a reliable model for loudspeaker defects (e.g. coil rubbing) or complex mechanical structures (e.g. buzzing panel in a car) or random processes (e.g. loose particles) which are not predictable. All of the modeling will be restricted to a reference system which has a linear or nonlinear transfer characteristic to describe the regular and deterministic behavior of the device under test.

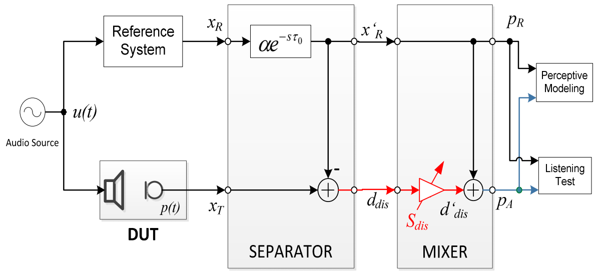

Figure 7: Auralization of signal distortion by differential decomposition by using the output of a reference system and the output of the device under test (DUT).

Although there are other forms and manifestations of this idea the third auralization scheme can be described generally as

$$p_A(t) = \alpha x_R(t-\tau_0) + S_{dis}(x_T(t) - \alpha x_R(t-\tau_0)) \qquad (4)$$

which corresponds with the signal flow chart depicted in Figure 7. The choice of the reference system determines which distortion signal d_{dis} will be generated in the following separator synchronizes and adjust the amplitude of the reference signal to the test signal before the difference signal is calculated. The following mixer scales the distortion component d_{dis} by an user defined scaling factor S_{dis} and generates the auralization output $p_A(t)$. The mixer may also generate a reference signal output $p_R(t)$ which is useful for predictive modeling and for AB-comparison in listening tests.

Table 1 gives an overview on further manifestations of the differential decomposition technique to auralize individual distortion components or combination of those by selecting particular test and reference signals.

Auralized Distortion	Test signal x_T	Reference signal x_R	Example
Linear + Regular nonlinear + Irregular nonlinear	Output of the DUT	Stimulus at input of the DUT	Figure 8
Linear + Regular nonlinear	Output of a nonlinear model	Stimulus at input of the DUT	Figure 9
Regular nonlinear + Irregular nonlinear	Output of the DUT operated at high amplitudes	Output of a linear model	Figure 10
	Output of the DUT operated at high amplitudes	Output of the DUT operated at small amplitudes	Figure 11
Linear	Output of the DUT operated at small amplitudes	Stimulus at input of the DUT	Figure 8
Regular nonlinear	Output of a nonlinear model of the DUT	Output of a linear model	Figure 12
Irregular nonlinear	Output of the DUT	Output of the nonlinear model	Figure 13
	Output of the DUT	Output of a *Golden Reference Unit*	Figure 14

Table 1: Modes of Operation using Differential Decomposition

3.3.1 Total Distortion

In order to separate the sum of all distortion components $d_{lin}(t)$, $d_{nlin}(t)$ and $d_{irr}(t)$ in Eq. (1) from the delayed and scaled input signal $\alpha(u(t-\tau_o)$ in the test signal $x_T(t)=p(t)$, the reference signal $x_R(t)= u(t)$ uses the stimulus $u(t)$ as illustrated in Figure 8.

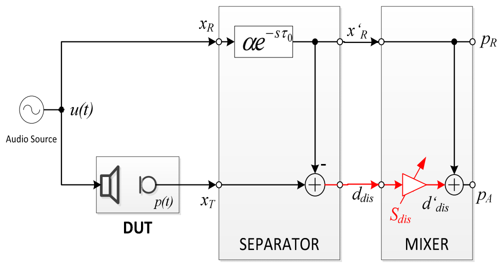

Figure 8: Auralization of signal distortion by differential decomposition using the stimulus $u(t)$ as reference signal and the output $p(t)$ of the device under test (DUT) mixing signal components separated by modeling

However, this form has been developed for CODECs by Faiten [8] but has not much practical value for loudspeakers or complete audio systems because the linear distortion $d_{lin}(t)$ will mask the other nonlinear distortion components and may generate a null at some fundamental signal components as discussed before.

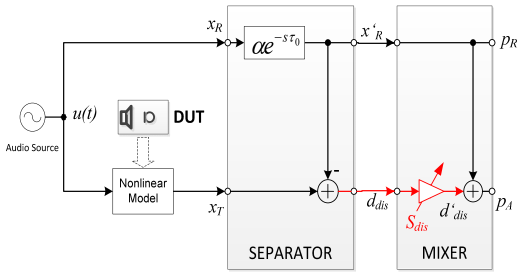

Figure 9: Auralization of linear and regular nonlinear distortion by using the output of a nonlinear model of the device under test (DUT) as the test signal $x_T(t)$

3.3.2 Linear and Regular Nonlinear Distortion

Using instead of a real DUT a model which is capable of describing the linear and regular nonlinear distortion then this form of the differential decomposition corresponds with the auralization scheme in Figure 4.

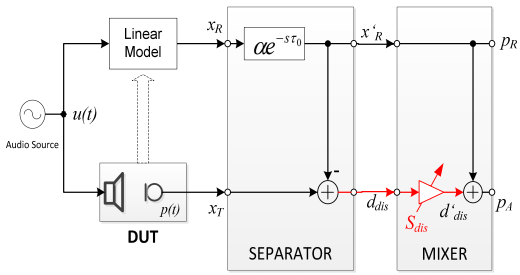

Figure 10: Auralization of regular and irregular nonlinear distortion by using a linear model of the device under test (DUT).

3.3.3 *Regular and Irregular Nonlinear Distortion*

In order to auralize regular nonlinear and irregular distortion components, $d_{nlin}(t)$ and $d_{irr}(t)$, respectively, the reference system should provide the linear distortion $d_{lin}(t)$. Figure 10 shows a first realization by using a linear model of the DUT that convolutes the stimulus signal $u(t)$ with the impulse response of the system under test. This impulse response should be measured at low amplitudes where the regular and irregular nonlinear distortions are negligible.

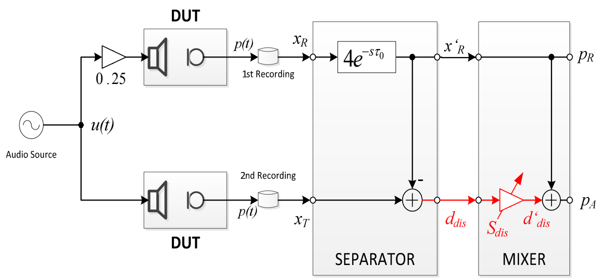

Figure 11: Auralization of regular and irregular nonlinear distortion by using the result of a small signal measurement of the device under test (DUT) as a reference signal $x_R(t)$

Figure 11 shows an interesting alternative form which uses the DUT itself as the reference system and dispenses with a linear model. While exciting the DUT by a deterministic stimulus $u(t)$ at sufficiently low amplitude where the DUT behaves almost linear the output p(t) is recorded and used as the reference signal $x_R(t)$. A second recording is required to capture the large signal behavior of the DUT by using the same stimulus $u(t)$ without attenuation. The attenuation factor (e.g. S_u= 0.25) used in the 1st recording will be compensated by an inverse gain factor (e.g. $\alpha=S_u^{-1}=4$) in the separator. Clearly the DUT and the measurement condition (e.g. loudspeaker position in the room) should be identical and two measurements performed immediately one after the other to reduce the influence of time varying properties.

3.3.4 *Linear Distortion*

Although the variation of the linear parameters as illustrated in Figure 2 is a more useful auralization scheme than the differential decomposition the setup of Figure 9 where the stimulus is used as reference signal is also applicable for auralizing the linear distortion. Since the nonlinear distortion $d_{nlin}(t)$ and $d_{irr}(t)$ are highly depending on the excitation level of the DUT a sufficiently low excitation level will ensure that the test signal will only contain linear distortion d_{lin}.

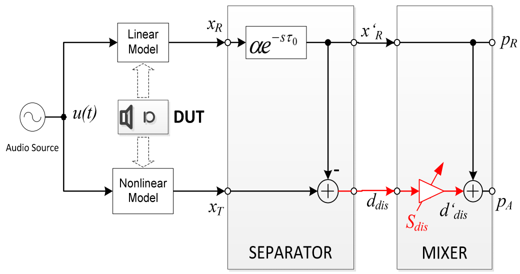

Figure 12: Auralization of regular nonlinear distortion by using the output of a linear model as a reference signal x_R and the output of a nonlinear model of the device under test (DUT) as a test signal $x_T(t)$

3.3.5 *Regular Nonlinear Distortion*

The differential decomposition may also be used to separate the regular nonlinear distortion caused by nonlinearities in the transducer or other parts of the audio system as illustrated in Figure 12. However, the generation of the reference and test signals requires a linear and a nonlinear model, respectively. The separation by subsystems as discussed in connection with Figure 4 is a more elegant auralization scheme.

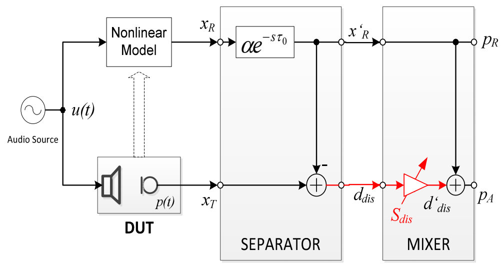

Figure 13: Auralization of irregular nonlinear distortion by using a nonlinear model of the device under test (DUT).

3.3.6 *Assessment of Irregular Nonlinear Distortion*

The differential decomposition is the only way to auralize the irregular nonlinear distortion $d_{irr}(t)$ separately from other distortion components. A nonlinear model of the DUT is used as the reference system generating a reference signal comprising the linear and regular nonlinear distortion components d_{lin} and d_{nlin}, respectively, as illustrated in Figure 13.

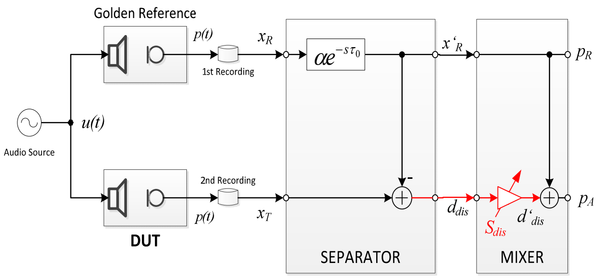

Figure 14: Auralization of irregular nonlinear distortion by using a nonlinear model of the device under test (DUT).

Figure 14 shows an alternative solution dispensing with nonlinear modeling of the DUT. The reference signal $x_R(t)$ is generated by a *Golden Reference Unit* with desired or accepted properties as the device under test without any defects. Such Golden Reference Units are selected in manufacturing and used to simplify the setting of PASS/FAIL limits in the context of end-of-line testing. If Golden Reference Units are not available it is also possible to use as a reference system a full functional DUT which has passed the test prior to the defective unit. Both units show usually similar linear and regular nonlinear distortion because the critical soft parts (e.g. spider, surround and diaphragm) came from the same batch and the settings (e.g. glue dispenser) are identical.

4 IMPLEMENTATION

Practical aspects of auralization technique will be discussed on the example of differential decomposition but is also applicable to the other methods.

Figure 15 shows in greater details the implementation of the separator and mixer block. It is convenient to provide the test and reference signal x_T and x_R, respectively, as a digital recording such as wavefiles.

A perfect synchronization and an alignment of the amplitude of reference signal and test signal is a basic requirement of this technique, otherwise excessive distortion, which are not meaningful, are generated in the output d_{dis} of the separator.

This objective can be achieved by calculating the correlation of the input signals x'_R and x_T by blockwise FFT processing in the *Separator*.

If the test signal is longer than the reference signal then the test signal is cut to the length of the reference signal. Zero padding will be applied to the test signal if it is shorter than the reference signal. Both rules ensure a constant length of the auralization output which is important for listening tests.

The time delay τ can be found by searching for the maximum in the correlation function. If the linear distortion are not subject of the auralization then an adaptive linear FIR filter F_R can be used for removing any signal component in the distortion signal d_{dis} which is coherent with the reference input x_R.

In the following *Mixer* the separated distortion signal d_{dis} will be transferred in a modified distortion signal d'_{dis} by applying a user defined scaling S_{dis} and optional linear filtering to shape the spectrum of the distortion with the transfer function $H_{dis}(s)$. A high-pass characteristic with a cut-off frequency at the resonance frequency of the electro-acoustical transducer is useful to suppress the nulling of the fundamental components at high values of S_{dis} which is related to “unnatural” enhancement of the compression effect. This high-pass filter is also useful for removing residual components in the distortion signal caused by imperfect modeling in the reference filter or gain adjustment and synchronization in the separator.

The mixer may contain an optimal generator adding a noise signal $n(t)$ to the reference signal x_R to simulate a steady-state background noise as found in automotive applications which masks the distortion components. Two gain controllers are provided at the output of the mixer to adjust the sound pressure level of the auralization output $p_A(t)$ and reference output $p_R(t)$ according to the original measurement condition or target application. This output scaling has influence on the audibility of irregular distortion because the random and impulsive waveform generates a wide band spectrum of distortion component close to the hearing threshold.

The system DM determines the peak values of the scaled distortion signal $d'_{dis}(t)$ and virtual auralization signal $y_A(t)$ within a time frame (e.g. 1s) and calculates the relative distortion measure

$$d_{rel}(t) = \frac{\hat{d}'_{dis}(t)}{\hat{y}_A(t)} 100\% \qquad (4)$$

for any time t of the auralization output signal.

Furthermore the enhancement of distortion component by a high value of S_{dis} will increase the loudness of the auralization output $p_A(t)$ compared to the reference signal $p_R(t)$ required in a forced A/B listening test. Therefore, the mixer provides tools to calculate the individual loudness of both output signals and to compensate differences by using separate gain values G_A and G_R.

There is also a need to support the export of the output to any reproduction system (stereo, headphone, multi-system). The export gain G_E ensures that the output signals $p_A(t)$ and $p_R(t)$ are stored in wavefiles w_R and w_R at high SNR without any clipping. A test tone $c(t)$ at defined calibration level L_c is recorded in a separated wavefile w_c with the same gain G_E and is used for calibration of the reproduction system prior to the listening test.

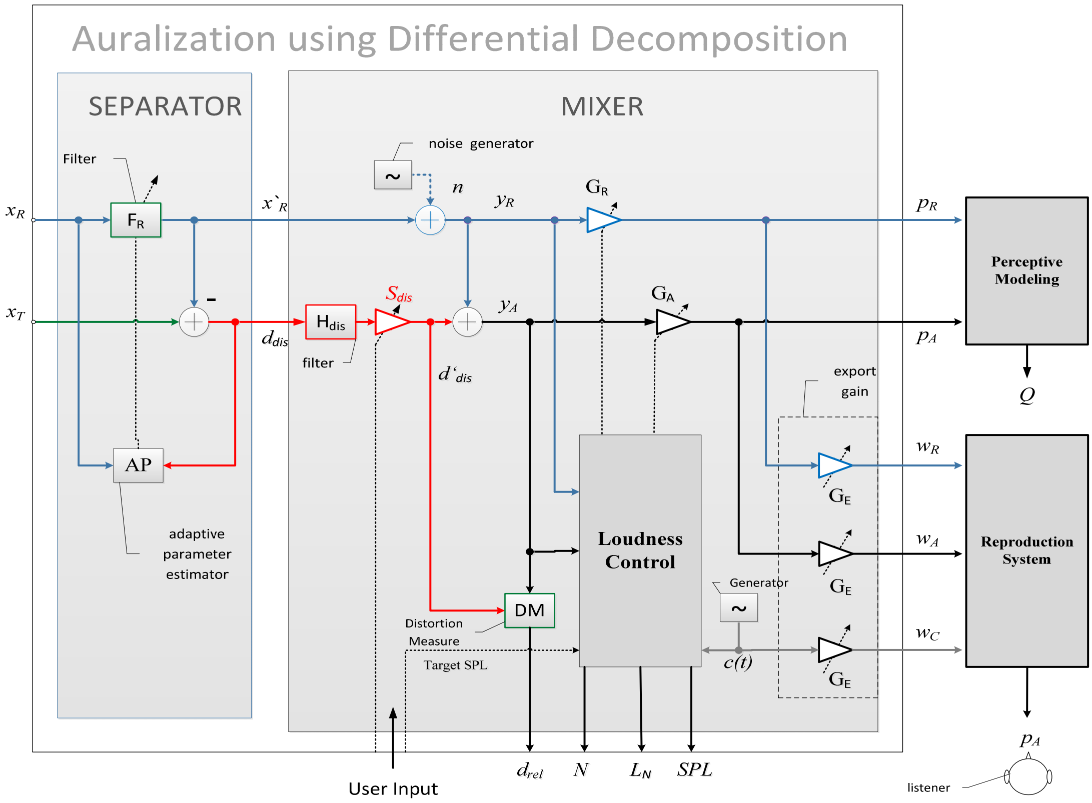

Figure 15: Implementation of the auralization scheme based on differential decomposition

5 CONCLUSIONS

Auralization of signal distortion in audio systems requires different techniques to satisfy the particularities of linear, regular nonlinear and irregular distortion. The variation of model parameter such as the gain, resonance frequency and loss factor in a parametric equalizer is the most useful technique for auralizing linear distortion avoiding any artifacts generated by other techniques.

The auralization of regular nonlinear distortion requires a technique which is capable of providing a large enhancement to make even small distortion audible. Increasing the values of the nonlinear parameters is a less suitable technique because the feedback will cause significant changes in the internal state variables and generate a virtual system which behaves completely different than the original DUT and may become unstable. Those problems can be avoided by tapping the distortion at nonlinear subsystems and using an additional mixer with a feed-forward structure.

The differential decomposition is the only auralization scheme which can cope with irregular distortion $d_{irr}(t)$ which cannot be modeled but require a measurement of the output signal $p(t)$ of the DUT. To separate the irregular distortion from the other distortion components a reference system is used to generate the linear und regular nonlinear behavior of the DUT which is deterministic, reproducible and predictable.

6 ACKNOWLEDGEMENT

The author would like to thank Marian Liebig for his ideas and suggestions while implementing and evaluating the differential decomposition technique.

7 REFERENCES

[1] Michael Vorländer: Auralization - Fundamentals of Acoustics, Modelling, Simulation, Algorithms and Acoustic Virtual Reality Springer, Berlin 2008. ISBN 978-3-540-48829-3

[2] W. Klippel, "Speaker Auralization – Subjective Evaluation of Nonlinear Distortion" presented at the 110th Convention of AES, 2001 May 12–15 Amsterdam,

[3] M. S. Rodrıguez, "Modeling And Real-Time Auralization of Electrodynamic Loudspeaker Non-Linearities," presented at the ICASSP 2004 of the IEEE

[4] Farina, et. al., "Real-time Auralization Employing a Not Non-Linear, Not Time-Invariant Convolver" in his paper presented at 123rd Convention of the Audio 2007, October 5-8, NY.

[5] W. Klippel, R. Werner, "Objective and Subjective Assessment of Regular Loudspeaker Distortion" International Symposium of Electro-Acoustic Technology ISEAT, Shenzhen China, November 2011.

[6] W. Klippel, R. Werner "Objective and Subjective Assessment of Rub & Buzz and other Irregular Loudspeaker Distortion" International Symposium of Electro-Acoustic Technology ISEAT, Shenzhen China, November 2011.

[7] Thiede, et. al. , "PEAQ - The ITU Standard for Objective Measurement of Perceived Audio Quality," J. Audio Eng. Soc. Vol. 48, No 1/2 , Jan/Feb, 2000, p. 2-29

[8] B. Feiten, "Measuring the Coding Margin of Perceptual Codecs with the Difference Signal, presented at the 102nd Convention 19197 March 22-25, Munich, Germany, preprint 4417.

[9] W. Klippel, Tutorial: "Loudspeaker Nonlinearities - Causes, Parameters, Symptoms" J. Audio Eng. Society 54, No. 10 pp. 907-939 (Oct. 2006).

[10] W. Klippel, "Equivalent Input Distortion," J. Audio Eng. Society 52, No. 9 pp. 931-947 (Sept. 2004).