

PRELIMINARY SPECIFICATION

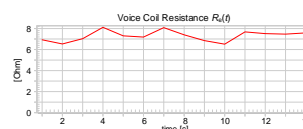
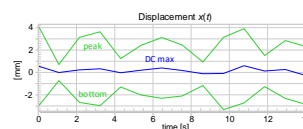
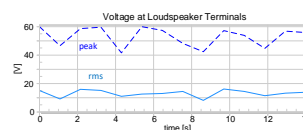
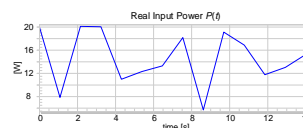
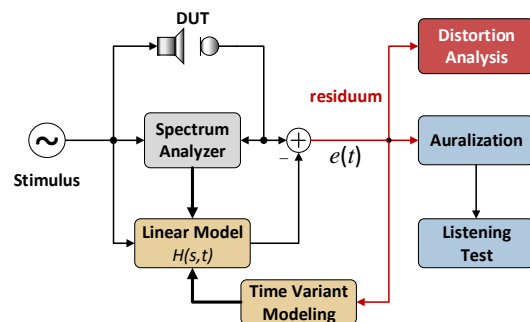
This specification is preliminary and is subject to change.

FEATURES

- Long-term monitoring of state signals
- Time varying transfer function
- Arbitrary stimulus (music, test signal)
- Distortion separation (nonlinear, R&B)
- Amplitude compression
- Voice coil heating
- Listen to isolated distortion

BENEFITS

- Combining measurement and listening
- Applicable to all audio systems
- Comprehensive large signal analysis
- Root cause analysis of loudspeaker defects
- Reveals impulsive distortion from transducer and electrical control



DESCRIPTION

To evaluate the performance of active and passive audio systems under normal operating conditions, the *LAA Live Audio Analyzer* can monitor up to 5 signals simultaneously, and examine both time and frequency data. Using arbitrary audio signals, such as music or speech, as well as dedicated test signals (multi-tone, chirp, single/two tone), critical applications can be identified and analysed. The *LAA* monitors terminal voltage, input current, voice coil displacement, as well as sound pressure output.

Employing adaptive modelling, the time-varying properties (such as heat or visco-elastic changes) can be captured, as well regular and irregular distortion isolated. Thus, triggers using both monitored state signal, as well as distortion measures, help to store only parts of interest. In addition, the impacts and causes of distortion effects can be identified and auralized.

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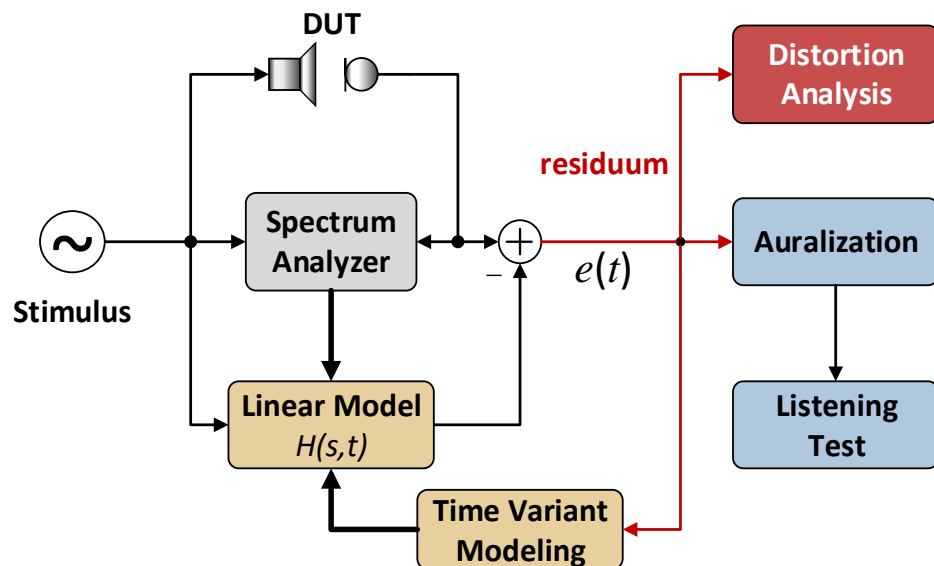
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1 Overview

1.1 Principle



The *Live Audio Analyzer* is a tool dedicated to long-term measurement of active and passive audio systems. It can be used with music, speech or dedicated test signals while monitoring up to five sensor signals simultaneously. For physical and perceptual evaluation of sound quality, the *Live Audio Analyzer* combines modern measurement and listening techniques (auralization).

Signal Analysis

The *Live Audio Analyzer* captures up to 5 sensor signals. Every sensor signal provides various data for analysis:

- high resolution signals,
- statistical data, such as peak, bottom or rms value,
- (long- and short-term) spectra,
- amplitude distribution (probability density function),
- and more.

By specifying two sensor signals as input $x(t)$ and output $y(t)$, a linear transfer function $H(s, t) = Y(s, t)/X(s, t)$ is determined. Different actions can be triggered if signal and distortion characteristics exceed a predefined limit, such as storing signal blocks or abort the measurement.

<p>Modelling</p>	<p>The audio system is modelled as a black box, represented by a transfer function between any monitored signal. It considers the time-variant, linear transfer function $H(s,t)=Y(s)/X(s)$ with an input signal $x(t)$ and an output signal $y(t)$. The measured input signal $x(t)$ and the output signal $y(t)$ may be any selection from the stimulus, internal state signals (e.g. displacement), or line signals such as sound pressure output.</p> <p>Using modeling, the long-term monitoring of the transfer function is improved and can cope with time-variant effects, such as voice coil heating, visco-elastic effects, as well as aging. Based on the linear modeled response, distortion and noise can be separated from the measured signal. Additional triggers using peak and mean distortion information help to store only high-resolution sensor data if necessary, thus helping to run sleek long-term tests without high memory consumption.</p> <p>Stored signals (measured, modeled and residual (distortion + noise)) can be exported to a waveform format.</p>
<p>1.2 Results</p>	
<p>Time Signals</p>	<p>Measured waveform of the input signal $x(t,n)$ and output signal $y(t,n)$ and statistical characteristics (positive peak, negative bottom, DC component and rms value, crest factor, probability density function) of the input and output signal in the analysis block $n=0,1,2,\dots, N$.</p> <p>Variation of the voice coil resistance $R_e(t)$ and increase of corresponding of voice coil temperature $\Delta T_v(t) = T_v(t) - T_v(t = 0)$ versus time t if voltage and current can be monitored at the transducer terminals.</p>
<p>Spectra</p>	<p>Complex transfer function $H(s,n)=Y(s,n)/X(s,n)$ between two signals $x(t)$ and $y(t)$.</p> <p>Incoherence function $IC(\omega,n)$, revealing the ratio of the spectral power in the output signal $y(t)$ which is not (linearly) correlated the transfer function $H(s)$ with the input signal $x(t)$.</p> <p>Difference signal $d(t)$ reveals the instantaneous noise and distortion generated by the DUT for any input signal (e.g. music), separated from the undistorted output signal $y'(t)$.</p>
<p>Auralization</p>	<p>Auralization and export of the measured, modeled and residual waveforms.</p>
<p>Ringbuffer</p>	<p>The signals can be stored</p> <ul style="list-style-type: none"> • for every time step, • using a ring-buffer (saving up to 10 min of data), • saving the last measurement block only.

1.3 Input Parameter	
Monitored State Signals and data acquisition	
Supported signals	<ul style="list-style-type: none"> • 2x Line Input / Microphone (via XLR & Laser Card) • 1x Voltage / 1x Current (via Speaker Card) • 1x Laser (via Laser Card)
Scaling signals	<ul style="list-style-type: none"> • Desired level or voltage of internally generated signal (including amplifier check) • Input headroom expansion (-12 dB, 0 dB, +12 dB)
Capture length	The Live Audio Analyzer allows continuous capturing of signals with a capture length $T_C = 1\text{ s}, \dots, 30\text{ s}$.
Stimulus	
Supported stimulus	<ul style="list-style-type: none"> • Internal generator (white / pink noise) • Wave file (playback via KA3) • External route through IN1/IN2 • No output
Stimulus Generator	Using a parameter stimulus set ¹ (for one output channel) for <ul style="list-style-type: none"> • White and pink noise
Pilot tone	A pilot tone for measuring the time-variant voice coil DC resistance $R_e(t)$ can be specified. <ul style="list-style-type: none"> • If the stimulus is played directly via KA3, the pilot tone is automatically determined and mixed to the source material. The pilot tone is applied with variable gain control, frequency can be user specified if necessary. • If the stimulus is provided via external routing, a tracking frequency (or frequency band) can be specified, which is used to determine the voice coil DC resistance.
Processing	
Frequency Resolution	Two processing modes are available, defining the temporal T_{DFT} and frequency resolution Δf with $\Delta f = 1/T_{DFT}$: <ul style="list-style-type: none"> • Fast (200 ms) • Slow (1000 ms)


2 Applications

2.1 Long-term Monitoring (R&D e.g. Maximum Output Evaluation)	
	<p>The LAA can monitor up to 5 sensor signals over a given or infinite measurement duration. By using modelling techniques to learn and update the linear system, long-term durability tests with minimum storage space may be defined.</p> <p>The user can set up automatic tests to find critical stimuli for a given active or passive system, or running accelerated aging and durability tests in a lab environment to identify the performance of the speaker while retaining most data.</p>
2.2 Live Analysis of Speaker States	
	The user is able to capture up to 5 signals (as stated in 1.3 Input Parameter), inspect time signal and spectrum as well as the transfer function between every signal. In addition to the noise signals already available, special test signals can be defined by using wav-files, and applied to the DUT. The LAA also supports external excitation (e.g. via sound card),

¹ May also be provided by another module

	or pure monitoring of the sensor signals. Optional saving of all captured data enables the possibility to perform processing on every stored block.
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3 Requirements

3.1 Hardware	
Analyzer	<p><i>Klippel Analyzer 3</i> is the hardware platform for the measurement modules performing the generation, acquisition and digital signal processing in real time.</p> 
Amplifier	<i>[optional]</i> Depending on the capture method, a power amplifier (either extern or Amp Card) is required.
Microphone	<i>[optional]</i> A microphone can be used to record the sound pressure during the measurement.
Laser Displacement Sensor	<i>[optional]</i> A high precision laser displacement sensor may be used to capture the membrane movement.
Computer	A personal computer is required for performing the measurement.
3.2 Software	
dB-Lab (>210.560)	Project Management Software of the KLIPPEL R&D SYSTEM. Requires at least version 210.560.

4 Limitations

4.1 Data Acquisition	
Capturing	<ul style="list-style-type: none"> • Capture length is fixed at maximum 30 s. • After every capture block a short interruption in capturing is present. • Adding full streaming support is one of the features for upcoming versions. • Capturing and saving all signals generates a high amount of data. Make sure that you have enough free disc space before and use the optional triggers.
Measurements on Device	The measurement is limited to one LAA instance per device only.
Trigger	Triggering the save of capture blocks depending on a measure is currently not supported.
Pilot tone	
For External Routings	For external routings, an external program (e.g. Audacity) may be used to mix the pilot tone with music and send to sound card or save as wave-file.
4.2 Processing	
Impedance	Impedance processing is identical with transfer function (using same resolution).
Spectrum Averaging	Averaging options are shared between impedance processing and transfer function.
Time-variant Modeling	Time-variant modeling is only available for one signal combination.

5 Output

5.1 State Monitoring and Signal Analysis	
Time Signals	<ul style="list-style-type: none"> • Time signals with full / reduced resolution for one capture block length T_c • Statistical values of captured signals <ul style="list-style-type: none"> ○ Peak / Bottom ○ RMS ○ Mean SPL (microphone only) ○ x_{DC} and $x_{DC,max}$ (displacement only)
Spectrum	<ul style="list-style-type: none"> • Long-term spectrum of any captured input / stimulus with a given resolution Δf, using exponential (τ) and linear ($N * T_{DFT}$) averaging • Short-term spectrum depending on capture block length T_c <p>Spectrum can be displayed in full resolution (linear), or displayed using preferred frequencies as defined in ISO 3, ISO 266 and IEC 61260 (R series; logarithmic).</p>
Transfer Function	<p>Transfer function of any captured input / stimulus (as defined in property page)</p> <ul style="list-style-type: none"> • Short-term transfer function (settings coupled with short-term spectrum) • Long-term transfer function (settings coupled with long-term spectrum) <p>Spectrum can be displayed in full resolution (linear), or displayed using preferred frequencies as defined in ISO 3, ISO 266 and IEC 61260 (R series; logarithmic).</p> <p>The signals of the transfer function are chosen at the beginning of the measurement and cannot be changed afterward.</p>
Impulse Response	Impulse response using the same settings as the long-term transfer function.
Energy Time Curve	Energy time curve using the same settings as the long-term transfer function.
Incoherence	<p>Incoherence using the same settings as the long-term transfer function.</p> <ul style="list-style-type: none"> • only available for external signal and random noise, not for multi tone • independent of capture block length T_c (incoherence is calculated successively over the complete measurement)
Voice coil Resistance $R_e(t)$	Voice coil resistance $R_e(t)$ at DC, using the spectral time constant as provided by the frequency resolution of the short-term spectrum T_{DFT} . The resistance is derived from the values at a (automatically) selected frequency f_{pilot} (pilot tone).
Voice coil Difference Temperature $\Delta T_v(t)$	Temperature derived from the voice coil resistance $R_e(t)$ at DC.
Impedance	Electrical impedance using the settings of the short-term spectrum
Power	Real input power $P(t)$ derived at the frequency f_{pilot} using the time constant T_{DFT} of the short-term spectrum.
5.2 Time-variant System Modeling	
Reference	Based on initial measurement with persistent excitation (@ multiple start levels [0 dB, -3 dB]) for dense white noise
Optimal $H(f)$	<p>Optimal Estimation of transfer function (coping with noise and poor excitation)</p> <ul style="list-style-type: none"> • reference transfer function $H_0(f)$, • momentary transfer function $H(f,n)$ • long-term transfer function $H(f)$
Distortion	Distortion metrics applied to residuum (peak, rms) can be expressed relatively (dB or percent) referred to fundamental or MEAN spectrum over f
Auralization	Auralization and export of signal distortion (residuum, linear model output, measured)

6 References

6.1 Related Modules	DIF-AUR, SIM-AUR
6.2 Manuals	LAA Manual

Find explanations for symbols at:

<http://www.klippel.de/know-how/literature.html>

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