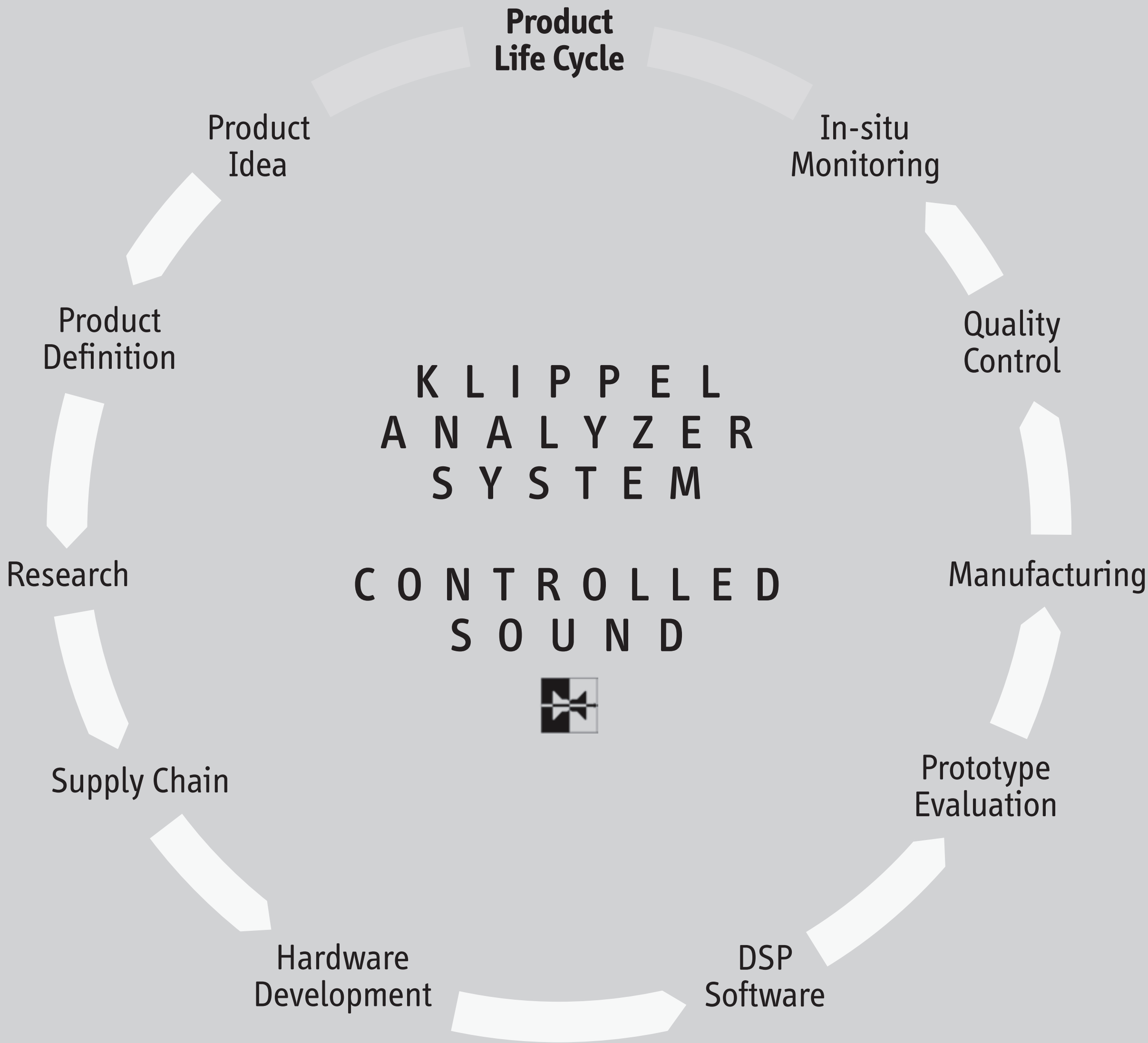
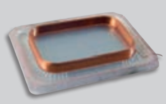
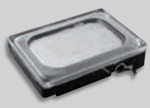
KLIPPEL

EMPOWERING THE AUDIO INDUSTRY




Material


Components


Audio System

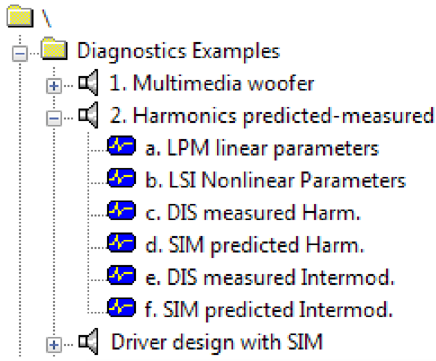

Application

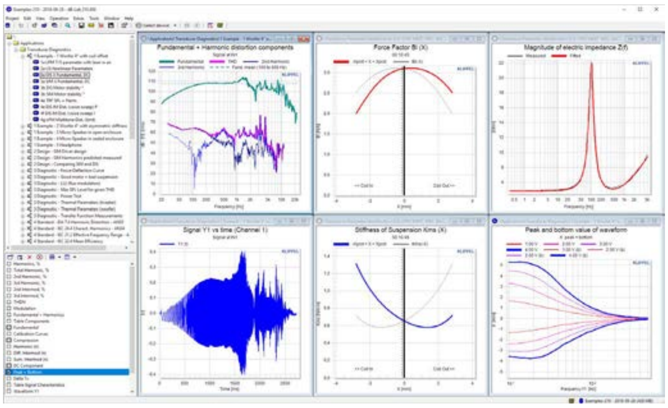
The KLIPPEL Analyzer System empowers engineers around the globe in creating and evaluating audio products over the full product life cycle. This results in products that meet the customers‘ expectations while minimizing manufacturing costs. KLIPPEL‘s vast research on loudspeaker modeling and system identification is the basis for unique measurement and simulation tools as well as innovative DSP algorithms. Together, these form the KLIPPEL Controlled Sound technology (KCS) for adaptive nonlinear control of electro acoustical systems. Discover our latest innovative technologies!

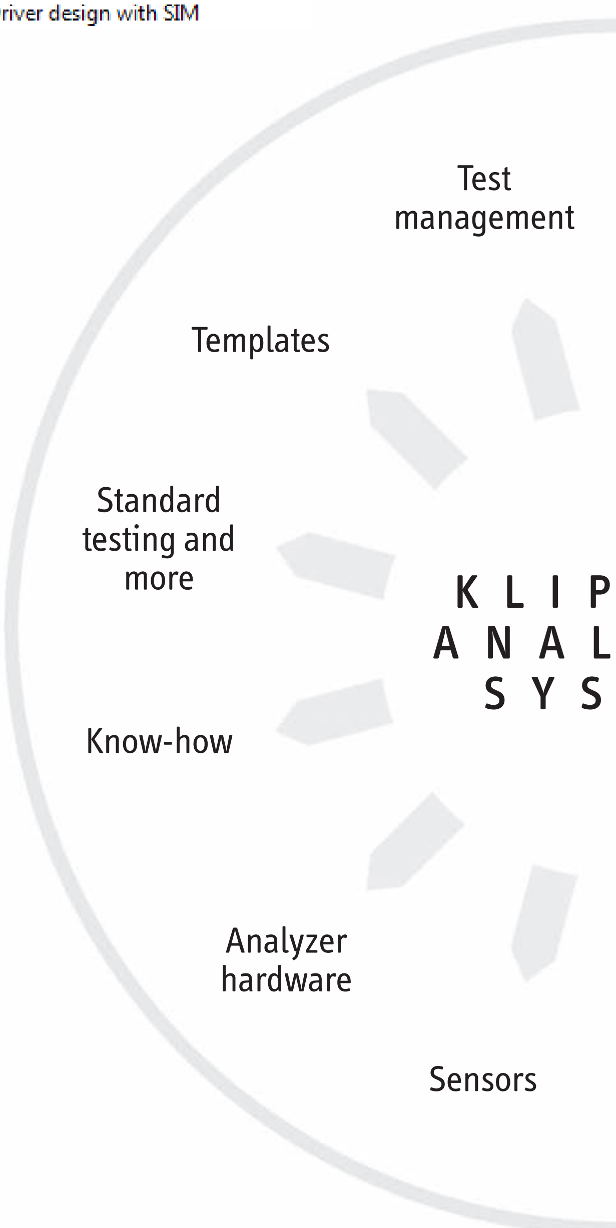
CONTENTS

MORE THAN A MEASUREMENT INSTRUMENT

The **dB-Lab** software is the common working environment for all modules of the KLIPPEL Analyzer System. Measurements, numerical simulations and any kind of post-processing are organized in a common database using a hierarchical structure to handle large and complex projects.
A variety of templates and Application Notes give valuable help for getting started with new methods of loudspeaker design and assessment. Setup parameters and the graphical display of the results can easily be customized to your workflow and stored as templates for future projects.



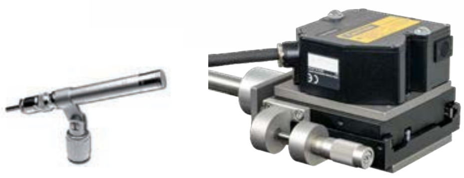




More Practical Knowledge

KLIPPEL not only provides powerful instruments for the physical and perceptive evaluation of the audio device according to international standards, but also know-how for the interpretation of the measurement results, comprehensive loudspeaker diagnostics and root cause analysis. Immerse yourself into our vast collection of application notes, papers and workshops on www.klippel.de.

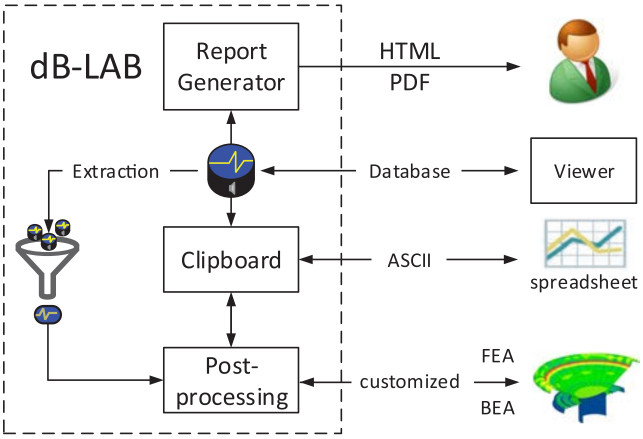
High-speed measurements with outstanding accuracy in a modular system tailored to your needs. For soft parts, transducers, electronic components and complete audio systems.



A wide range of selected microphones and laser sensors provide the optimal signal-to-noise ratio in the required working range and for the particular application.

Master collaborations

dB-Lab supports the exchange of databases between coworkers, suppliers and customers, which simplifies the interpretation of the results. Tools are provided for extracting, collecting, comparing and visualizing data and generating automatic reports in HTML format and customized according to the corporate identity (logo, comments, additional illustrations).



Customization

Import Export

Integration

Automation

Robotics

Unique tools

P E L
Y Z E R
T E M

The module **STAT** simplifies the statistical analysis of large amounts of data and the export to external spreadsheet software. dB-Lab also supports interlinked workflow and data exchange with external numerical simulation software (FEA, BEA).

Robotics at your fingertips

The automation interface and GPIO-pins on the hardware platform provide alternative ways to realize a fast and smooth interaction between external devices and software for lab and end-of-line testing. These also facilitate communications with new KLIPPEL robotic systems used to scan vibrations on mechanical structures and measure sound in 3D space and the magnetic field within the gap.

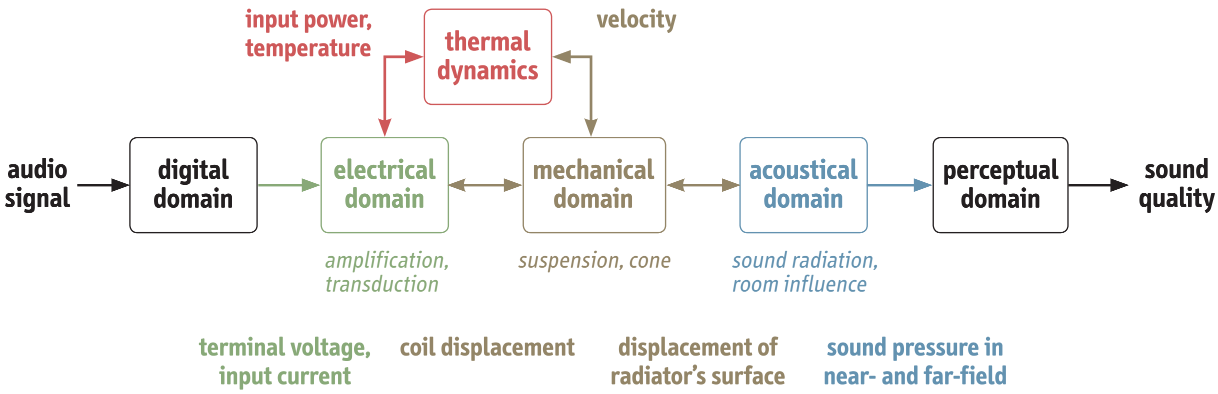
Complete your toolbox with laser stands, jigs for clamping transducers, test boxes, vacuum measurement kits and many other accessories which are important small components that turn the system into a complete solution.

KA

UNLEASH THE POWER OF MODELING

deeper diagnostics - faster development - better design

Modern numerical simulation tools need accurate input information such as geometry, common material properties and effective model parameters measured on a real sample by using system identification techniques (fitting). The model with the identified parameters can be used to predict the acoustical output signal and electrical and mechanical state signals for any audio input signal.



Electrical Measurement

Mechanical Laser Vibrometry

Acoustical Measurement

Electrical Impedance

Parameter Identification

Accumulated Acceleration

Spherical Wave Expansion

Radiation Analysis

Directivity

Linear (T/S) Parameters

Modal Expansion

3D Sound Field

Nonlinearities

Rocking Mode Analysis

Imbalances

SPL in Listening Zone

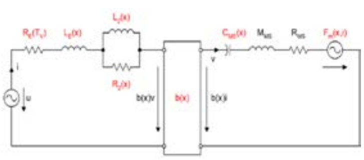
Thermal Parameters

Material Parameters

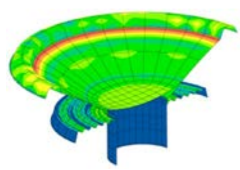
Aging, Fatigue Parameters

Sound Power

Equivalent Network Modeling



Finite Element Modeling
Boundary Element Modeling



Perceptual Modeling

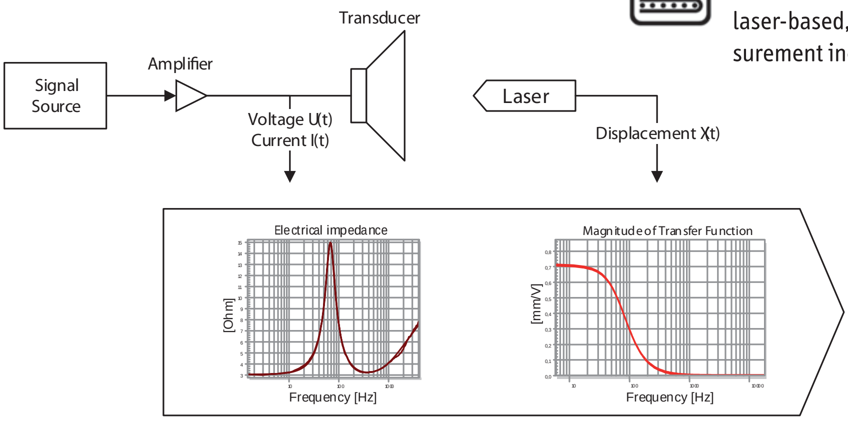
LINEAR PARAMETER MEASUREMENT

BASIC MODELING (LPM, IMP)

- Sensitive technique at low amplitudes
- Alternative identification techniques
- Mechanical creep and lossy inductance
- Easy verification of parameter accuracy
- Multi-tone distortion measurement

The **Linear Parameter Measurement (LPM)** module and the **Impedance Task (IMP)** identify the lumped parameters of the transducer's equivalent circuit including T/S, creep and lossy inductance parameters based on a one-step laser technique, which can also be applied to tweeters, micro-speakers and headphones. **LPM** supports the conventional two-step perturbation techniques (added mass, test enclosure) too. A multi-tone stimulus is used to measure the voltage, current, voice coil displacement and sound pressure at sufficient SNR while operating the transducer in the small signal domain.

IMP is optimized for fast EoL measurements with PASS/FAIL classification. The **TSX** add-on provides laser-based, high-speed lumped parameter measurement including Bl and Mms.



Lumped Parameter

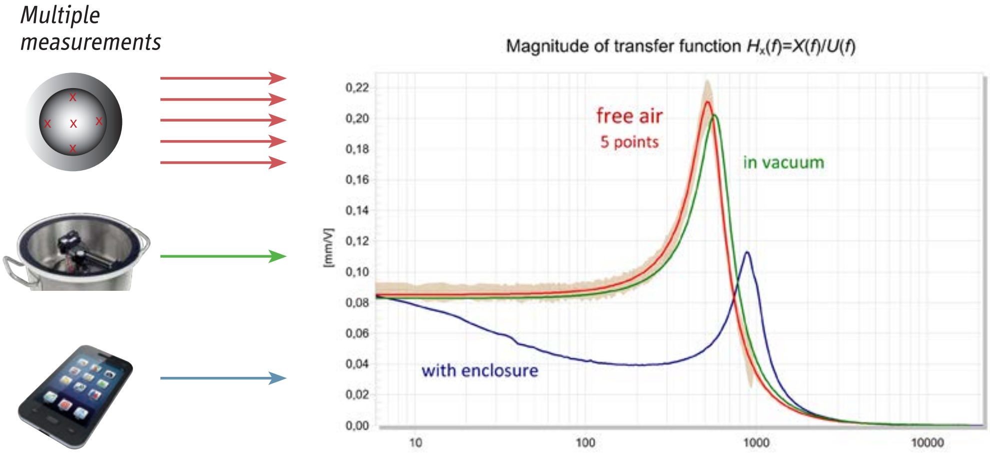
M_{ms}	0.107	g
R_{ms}	0.093	kg/s
C_{ms}	1.18	mm/N
K_{ms}	0.8	N/mm
Bl	0.647	N/A
...	...	...

ADVANCED MODELING (MMT)

- Complete parameter set
- Cope with rocking modes
- Automatic post-processing
- Advanced micro-speaker modeling
- Separation of enclosure and air load
- Pure mechanical parameters

The **Multipoint Measurement Tool (MMT)** module performs post-processing using results of multiple **LPM** measurements to increase the accuracy of the parameter identification. Spatial averaging of the voice coil displacement, measured at multiple points on the diaphragm, compensates for rocking modes and other irregular vibration. Additional measurements performed in vacuum and in the final enclosure show the pure mechanical parameters of the transducer separated from the acoustical elements. Advanced creep models (e.g. Ritter) are supported to model the significant visco-elastic behavior of micro-speakers.

Multiple measurements



TS-parameter (+air load)

M_{ms} = 0.107 g
K_{ms} = 0.8 N/mm
R_{ms} = 0.093 kg/s

Pure mechanical parameter:

M_{md} = 0.091 g
K_{md} = 0.63 N/mm
R_{md} = 0.043 kg/s

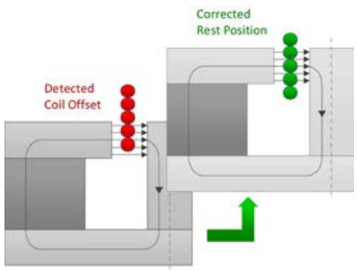
NONLINEARITIES – UNCOVERING POTENTIAL FOR MORE OUTPUT

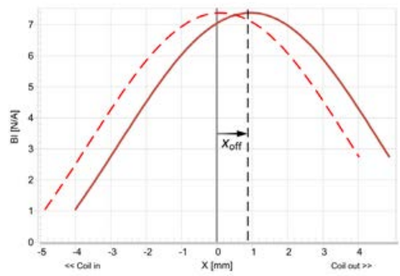
LARGE SIGNAL IDENTIFICATION (LSI)

patented KLIPPEL technology

- Linear and nonlinear parameters
- Voice coil rest position
- Stiffness asymmetry of suspension
- State (temperature, displacement, ...)
- Mechanical and thermal protection
- Physical cause of nonlinear distortion

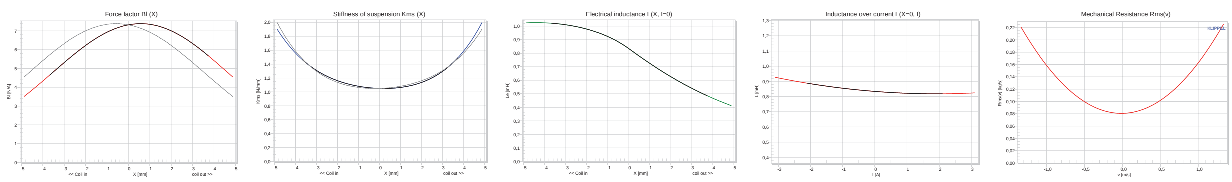
The **Large Signal Identification (LSI)** module dynamically measures the linear and nonlinear parameters of electro-dynamic transducers by monitoring and current at the transducer terminals according to IEC standard 62458. The transducer can be operated under normal working conditions (free air, sealed or vented enclosure) and is excited by an audio-like signal (noise). Different software modules are available for micro-speakers and woofers depending on the resonance frequency fs and the enclosure.





Force factor Bl(x) shows an offset x_{off} in the voice coil rest position

The force factor Bl(x), inductance L_e(x,i), stiffness K_{ms}(x) and mechanical resistance R_{ms}(v) are nonlinear functions of voice coil displacement x, input current i and velocity v. The nonlinearities determine the performance in the large signal domain, generate signal distortion (THD, IMD), limit the maximal output (SPL) and may cause unstable behavior (DC displacement). The nonlinear curves have a close link to the practical design and are easy to interpret (e.g. voice coil offset).



Force Factor Bl(x) | Stiffness K_{ms}(x) | Inductance L_e(x) | Inductance L(i) | Mech. Resistance R_{ms}(v)

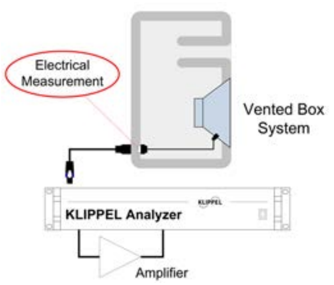
MOTOR + SUSPENSION CHECK (MSC)

patented KLIPPEL technology

- Voice coil offset in mm
- Stiffness asymmetry in %
- Early defect detection
- Root cause information
- Maximize output and reliability

The **Motor + Suspension Check (QC-MSC)** is a unique tool for high-speed identification of non-linear driver parameters on the production line. Based on patented KLIPPEL technology, effective linear and nonlinear parameters (Bl(x) and K_{ms}(x)) are identified to provide meaningful single value parameters like voice coil offset or suspension asymmetry. This provides valuable information for taking immediate action to ensure peak performance and fix causes of loudspeaker distortion.

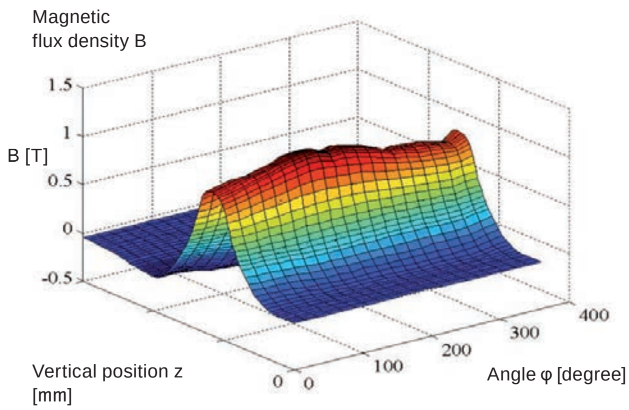
The driver under test may be operated in free air as well as in sealed or vented enclosures, giving additional enclosure parameters (f_b, Q_b). Neither microphones nor mechanical sensors (lasers) are required as all information is provided by the electrical input current, making the measurement immune to ambient noise.



MAGNETIC B-FIELD SCANNER (BFS)

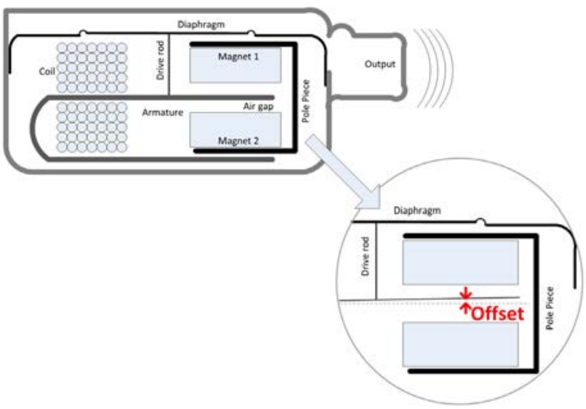
- Manual and automatic scanning
- Accurate positioning
- Hall sensor fits in small gaps
- Find magnetization problems
- Force factor Bl(x) prediction

The **B-Field Scanner (BFS)** measures the static flux density B(φ, z) in the gap versus angle φ and height z by using a Hall sensor and a mechanical scanning technique. This technique provides a prediction of the static force factor Bl(x) and finds irregularities in the magnetic field caused by design or problems in the assembling and magnetization process. Axial-symmetrical magnet field geometry is required to suppress rocking modes which cause voice coil rubbing and impulsive distortion.



BALANCED ARMATURE CHECK (BAC)

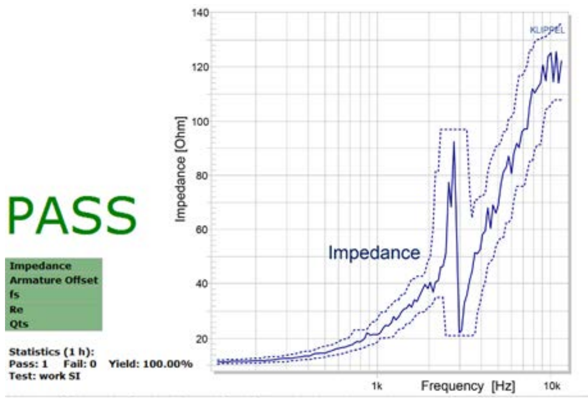
patented KLIPPEL technology



- Armature offset in µm
- Patented large signal identification
- High speed test: 0.5 – 2 s
- Linear parameters: R_e, f_s, Q_{ts}, L_e
- Peak excursion of armature (without laser)
- Immune to production noise

The performance of balanced armature transducers depends on the rest position of the armature, which should be properly centered in the magnetic gap. Any offset generates excessive nonlinear distortion and reduces the peak excursion and acoustical output. Measuring armature offset of the final transducer is hardly possible using mechanical or optical sensors.

The **Balanced Armature Check (BAC)** module is a unique measurement tool dedicated to testing electro-magnetic transducers at the end of the production line. The armature offset in µm and effective linear parameters are determined from a purely electrical, high-speed measurement. This gives valuable diagnostics information for minimizing rejection rate.



[Task output: Balanced Armature Check]

Armature Offset: 44.27 µm

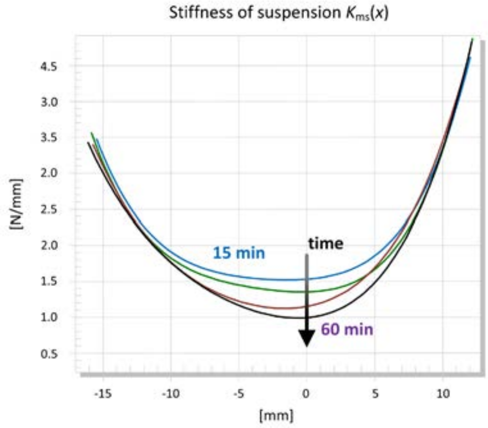
Name	Value	Min Limit	Max Limit	Unit	Description
Armature Offset	44.27	36.05	52.63	µm	armature offset
fs	2688.8	2286.0	3092.9	Hz	resonance frequency (approximated from 2nd order fitting)
Re	11.18	10.62	11.74	Ohm	electrical coil resistance at DC
Qts	1.80	1.53	2.07		total Q-factor (approximated from 2nd order fitting)

Name	Value	Unit	Description
Xpeak	162.10	µm	positive peak displacement during measurement
Xbottom	-150.04	µm	negative peak displacement during measurement

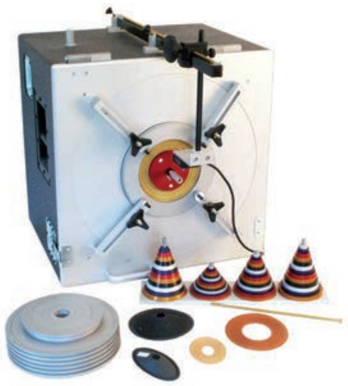
SUSPENSION PART MEASUREMENT (SPM, MSPM, LST)

patented KLIPPEL technology

- Nondestructive, dynamic testing
- Linear and nonlinear characteristics
- Long-term fatigue test
- Evaluate component quality
- Sustain high quality of soft parts
- Benefit from simple and cost-effective testing



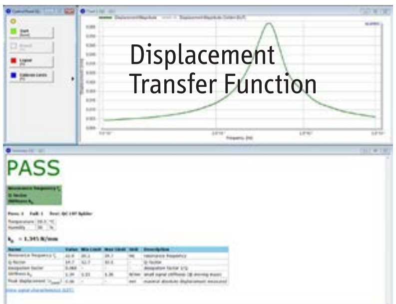
The **Suspension Part Measurement (SPM, MSPM)** features a dynamic identification of the small and large signal parameters of soft parts according to IEC standard 62459. The module SPM is dedicated to spiders, surrounds, cones and passive radiators (drones), while the **Micro Suspension Part Measurement (MSPM)** is dedicated to diaphragms used in micro-speakers, headphones, tweeters and microphones. The suspension part is pneumatically excited to dynamically measure the stiffness. Test benches and a clamping system comprising sets of rings, cones and cups are provided for nondestructive measurement. The measurements reveal the linear and nonlinear stiffness parameters, visco-elastic effects, and the dependency on ambient conditions (temperature and humidity).



Suspension Part Measurement (SPM)

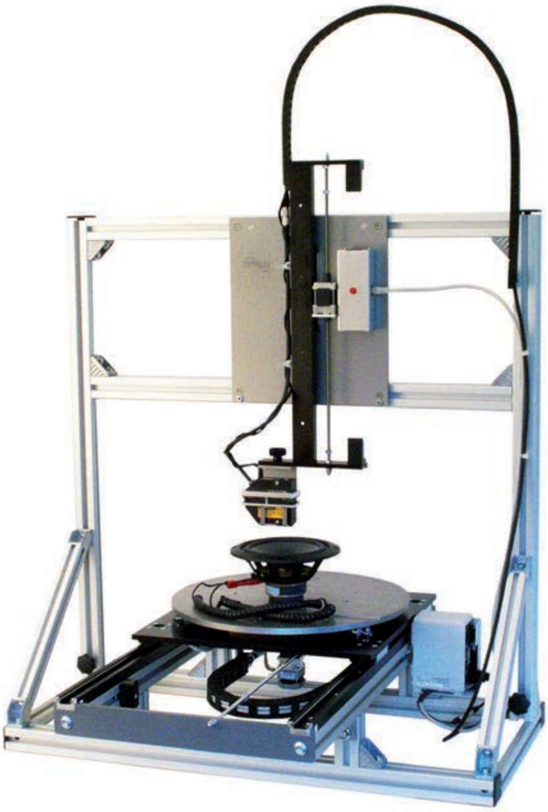
Micro-speaker Suspension Part Measurement (MSPM)

The **Linear Suspension Test (LST)** is dedicated to the quality control of moving parts. Linear mechanical parameters like resonance frequency and stiffness **(LST Lite)** or relative mass and stiffness deviation (**LST Pro** for passive radiators) are determined from the displacement transfer function using the same methods as the **(M)SPM**. The high test speed and simple clamping lead to short test cycle times, allowing for 100 % testing. The **LST** may be operated with all available test benches to cover part diameters from a few millimeters to more than 18''.



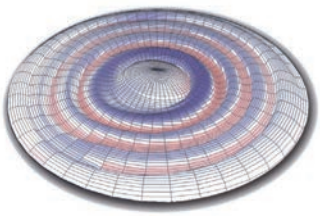
SCANNING VIBROMETER (SCN)

- Assess geometry and mechanical vibration
- Perform modal analysis
- Evaluate rocking and circumferential modes
- Predict acoustical output and directivity
- Separate mechanical and acoustical problems
- Verify mechanical simulation (FEM)
- Measure effective radiation area (S_d)

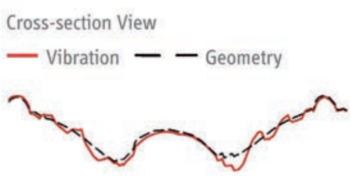


The **Scanning Vibrometer (SCN)** performs a non-contact measurement of the mechanical vibration of cones, diaphragms, panels and enclosures over the whole audio band (< 25 kHz). One rotational and two linear actuators (φ, r, z) position a laser displacement sensor on a user-defined grid. Additionally, you get the geometric data, which can be exported to other FEA/BEA applications. The vibration data can be analyzed within the **SCN** software. Modern techniques of image processing are used for enhancing relevant information, suppressing noise and animating the vibration as a stroboscopic video.

Engineering Poster



3D-Animation



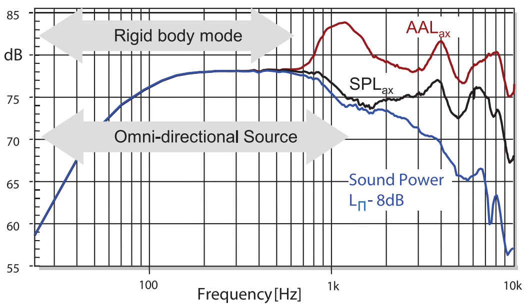
Engineering Poster

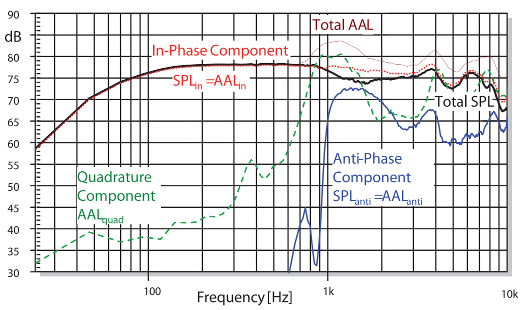
Vibration analysis

The accumulated acceleration level (AAL) describes the total vibration of the radiator (potential SPL), which is compared with the acoustical output (actual SPL) and is the basis for a modal analysis. The modal information can be used to verify the results of numerical simulation and to measure material parameters (e.g. loss factor) required for FEA. The energy of rocking modes, which may cause voice coil rubbing and impulsive distortion, can be evaluated. Significant differences between AAL and SPL reveal acoustical cancellation problems.

Radiation analysis

The acoustical output (SPL) can be calculated by using the geometry and vibration of the scanned radiator. The total vibration is decomposed into an in-phase, anti-phase and quadrature component generating a constructive, destructive and no contribution to the radiated sound, respectively. This information allows mechanical and acoustical problems to be separated and gives valuable indications for changing the design with respect to material and geometry.

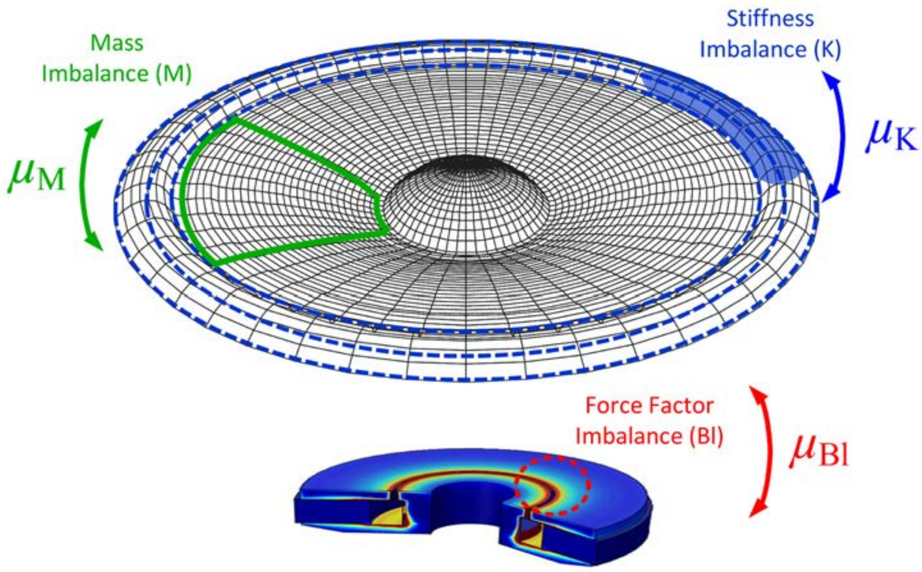




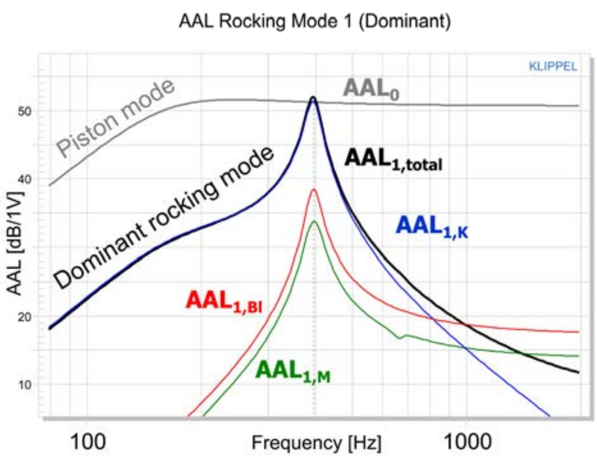
ROCKING MODE ANALYSIS (RMA)

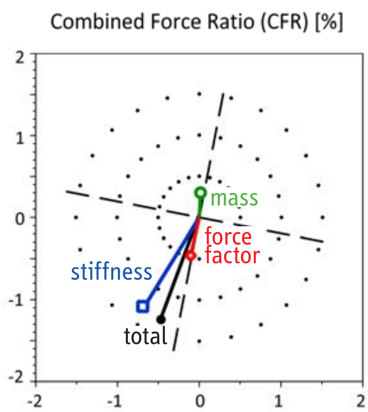
- Cope with voice coil rubbing
- Based on laser vibrometry
- Shows physical root cause
- Imbalances in mass, stiffness, Bl
- Provides location of imbalances

The **Rocking Mode Analysis (RMA)** uses vibration data measured by laser vibrometry (e.g. **SCN**) to separate the undesired rocking modes from the desired piston mode. **RMA** identifies the imbalances in the distribution of mass, stiffness and force factor in the electro-mechanical system. Those imbalances generate tilting moments μ_M μ_K μ_{Bl}, exciting the coil to high rocking amplitudes at the modal resonance.



The **RMA** quantifies the contribution of each root cause to the total rocking motion and reveals the position of the center of imbalance for mass, stiffness and excitation force on the diaphragm. With this information available, weak points in the design can be found and problems in manufacturing solved. The risk of voice coil rubbing and impulsive distortion in the output signal is effectively reduced with great benefits, especially for headphone and micro-speaker applications.



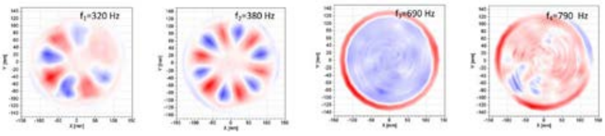


Example above: An imbalance in the stiffness distribution on the surround area is the main cause of the rocking mode. The high quality factor of the modal resonance at 390 Hz generates a rocking level that exceeds the AAL of the desired piston mode. The direction and strength of each imbalance are provided in data tables and polar plots.

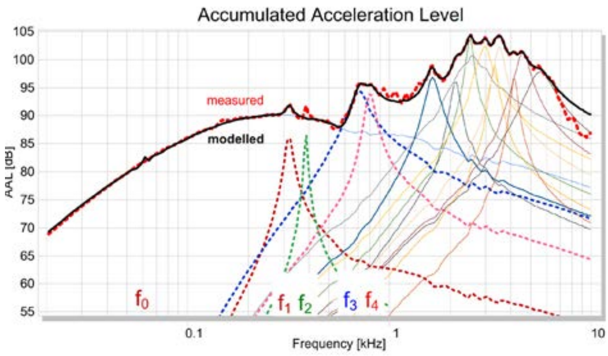
HIGHER MODAL ANALYSIS (HMA)

- Study vibration-modes separately
- Analyzes simulated and measured data
- Validates numerical simulation (FEA)
- Extracts modal parameters
- Reveals causes of cone distortion
- Visual animation of mode shapes

Each mode generates a characteristic distribution of the displacement on the radiating surface which is independent of frequency.



The **Higher Modal Analysis (HMA)** tool describes the mechanical system (cone, suspension, coil) over the full audio band by decomposing the total vibration into modes – each of which can be fully described by a vibration pattern (mode shape) and a frequency response similar to the fundamental resonance of the piston mode. The modal resonance frequencies f_n and quality factors Q_n correspond to the geometry of the structure and the properties of the material used. The **HMA** provides a validity check for finite element analysis (FEA) by comparing the simulated data with measured data provided by laser vibrometry. Undesired modes that are generating low acoustical output and high nonlinear distortion (THD, Rub & Buzz) are also revealed.

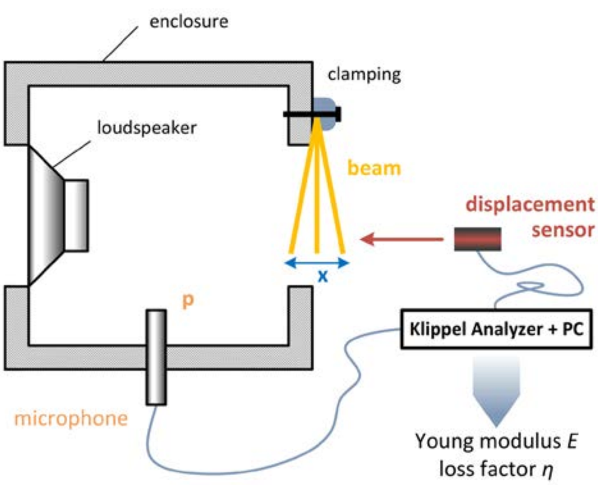


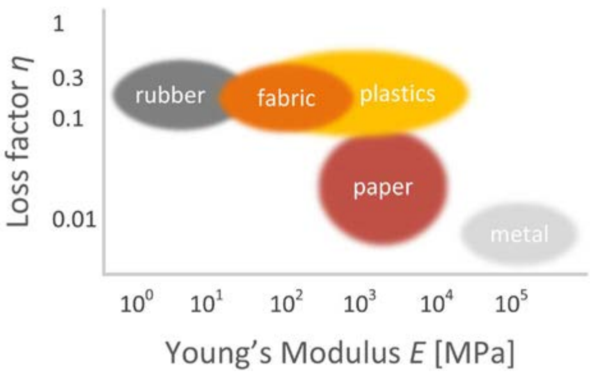
Each mode has a characteristic frequency response which can be compared with the contribution of the other modes to the total vibration in the Accumulated Acceleration Level (AAL).

MATERIAL PARAMETER MEASUREMENT (MPM)

- Dynamic measurement technique
- Simple and robust testing
- Applicable to loudspeaker materials
- Simplifies specification of parts
- Checks consistency of products

The Young's E modulus and the loss factor η of raw material used in diaphragms, cones, suspension and other parts are dynamically measured by a beam technique (ASTM E 756-93), which is modified to also be capable of measuring soft materials such as thin foils of plastic, rubber and any kind of paper and impregnated fabric. 1 cm strips taken from flat samples are clamped on one side and pneumatically excited by a loudspeaker. This robust technique is easy to use and provides reliable information, which simplifies the communication between loudspeaker designers and suppliers.





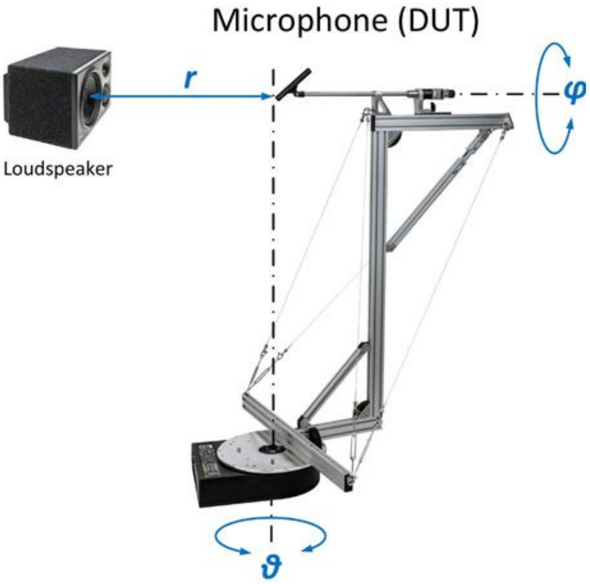
M

POLAR FAR-FIELD MEASUREMENT (POL)

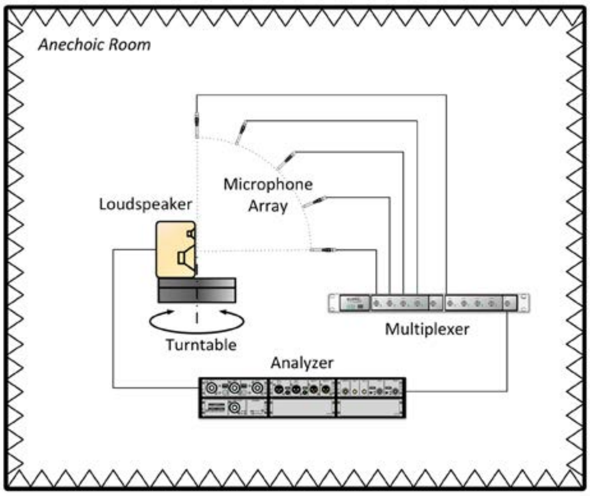
patented KLIPPEL technology

- Directivity of loudspeakers and microphones
- Measurement under anechoic far-field condition
- Automatic control of multiple turntables
- Multiplexing of transducers and sensors
- Angular interpolation based on wave modeling
- 2D and 3D display modes (polar, balloon, ...)
- Open export interface (ASCII, EASE, CLF, VACS)

The **Polar Field Measurement (POL)** module offers a fully automated measurement of the directivity of loudspeakers and microphones under free-field condition in polar coordinates φ, ϑ according to IEC standard 60268-21. The measurement distance r should be constant and larger than the dimension d of the device (r >> d) and the wavelength (r >> λ) to measure directivity in the far-field. The module supports various kinds of turntables and the KLIPPEL multiplexers for switching transducers and microphones.

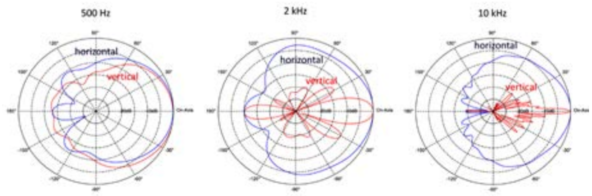


Measurement of 3D microphone directivity with turntable and rotational motor

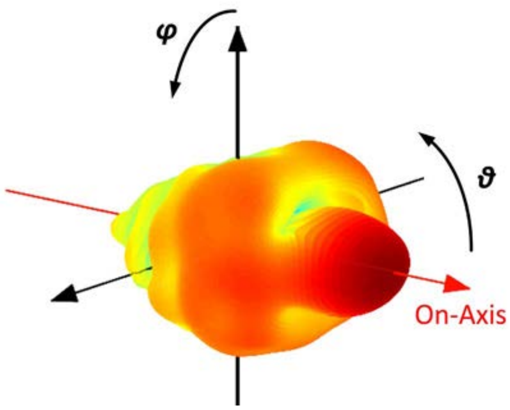


Conventional 3D measurement of loudspeaker directivity with microphone array and turntable

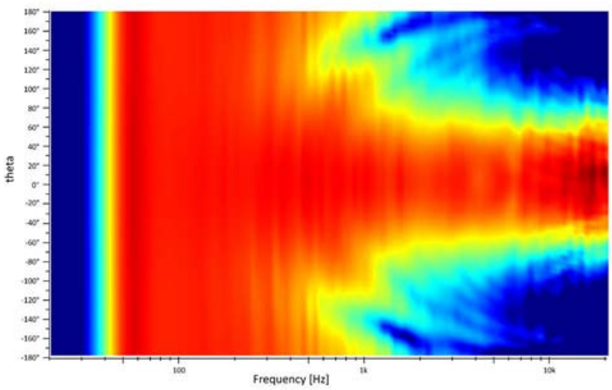
The **POL** module visualizes the directivity by polar, balloon and contour plots and calculates the directivity index, sound power and other characteristics. Spherical wave expansion can be applied to the measured data to check the accuracy of the measurement and to improve the interpolation between the measurement points.



Polar plot showing horizontal and vertical SPL at three frequencies



Balloon plot of SPL versus angles ϑ and φ at 2 kHz



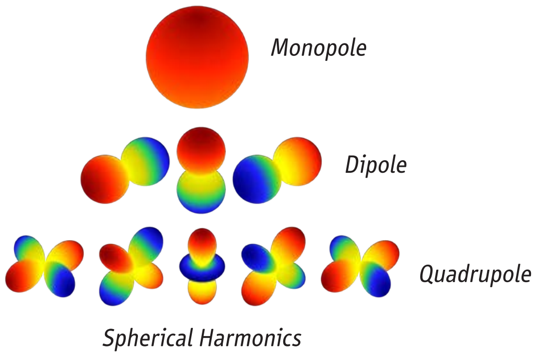
Contour plot of SPL versus angle ϑ and frequency f

HOLOGRAPHIC NEAR-FIELD MEASUREMENT (NFS)

patented KLIPPEL technology

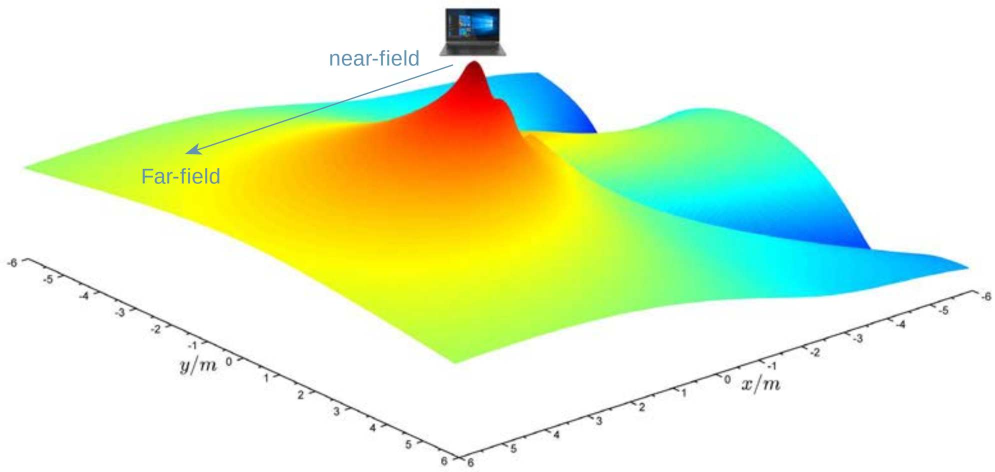
- Applicable to all audio devices
- 3D near and far field data
- Accurate amplitude and phase response
- Self-check for validity and symmetry
- Half-space and full-space measurement
- Light and portable scanning equipment
- Multiplexing of transducers in line array

The **Near-Field Scanner 3D (NFS)** uses a moving micro-phone to scan the sound pressure in the near-field of a compact sound source such as a loudspeaker system or a transducer mounted in a baffle. The device under test (< 500 kg) does not move during the scanning process. Therefore the reflections in the non-anechoic environment are consistent and can be monitored with our unique analysis software, which uses acoustical holography and field separation techniques according to IEC standard 60268-21 to extract the direct sound and to reduce room reflections.



Spherical Wave Expansion

The sound field generated by the source is reconstructed by a weighted sum of Spherical Harmonics and Hankel-functions, which are solutions of the wave equation.
The weighting coefficients in this expansion represent the unique information found in the near-field scan while significantly reducing the amount of data.



Far-Field Directivity

The near-field data, measured at a high SNR, is the basis for predicting the direct sound at farther distances. This avoids the diffraction problems present in classical far-field measurements (non-homogeneous media).

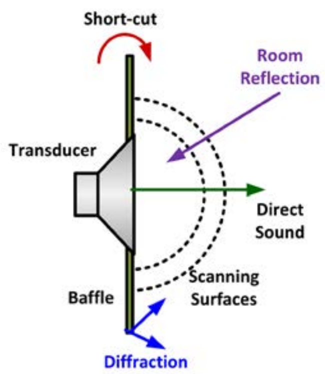
Near-Field Analysis

The wave expansion provides the sound pressure at any point outside the scanning surface, which is required for assessing studio monitors, tablets and other personal audio devices where the near-field properties are important.

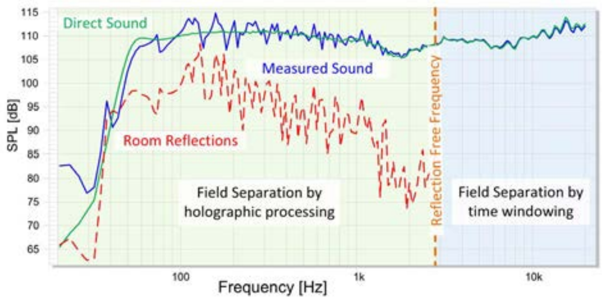
DIRECT SOUND SEPARATION (DSS)

patented KLIPPEL technology

- Separates direct sound from reflections
- Copes with baffle size, clamping, robotics
- More accurate than anechoic rooms
- Generates accurate reference data
- Mobile, cost effective solution



Separation of the direct sound from undesired sound waves generated by the room, baffle and rear side of the speaker



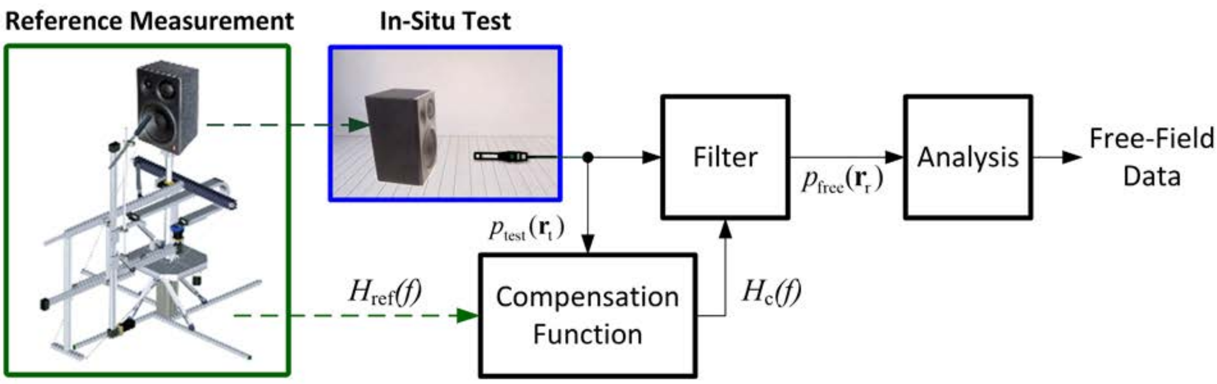
Standard acoustical measurements require a full-space or half-space anechoic environment. Even expensive room treatment cannot completely suppress standing room modes at low frequencies where windowing of the impulse response also fails to provide simulated free-field data.
The **Direct Sound Separation (DSS)** module extracts the direct sound with high accuracy by holographic processing of the near-field data scanned on two surfaces enclosing the loudspeaker according to IEC standard 60268-21.

IN-SITU ROOM COMPENSATION (ISC)

patented KLIPPEL technology

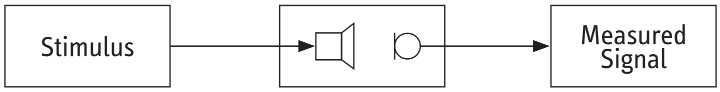
- Fast in-situ measurements
- Simulates standard condition
- Compensates room reflections
- Far-field data based on near-field testing
- Calibrates setups (boxes, rooms)
- Provides accurate transient behavior

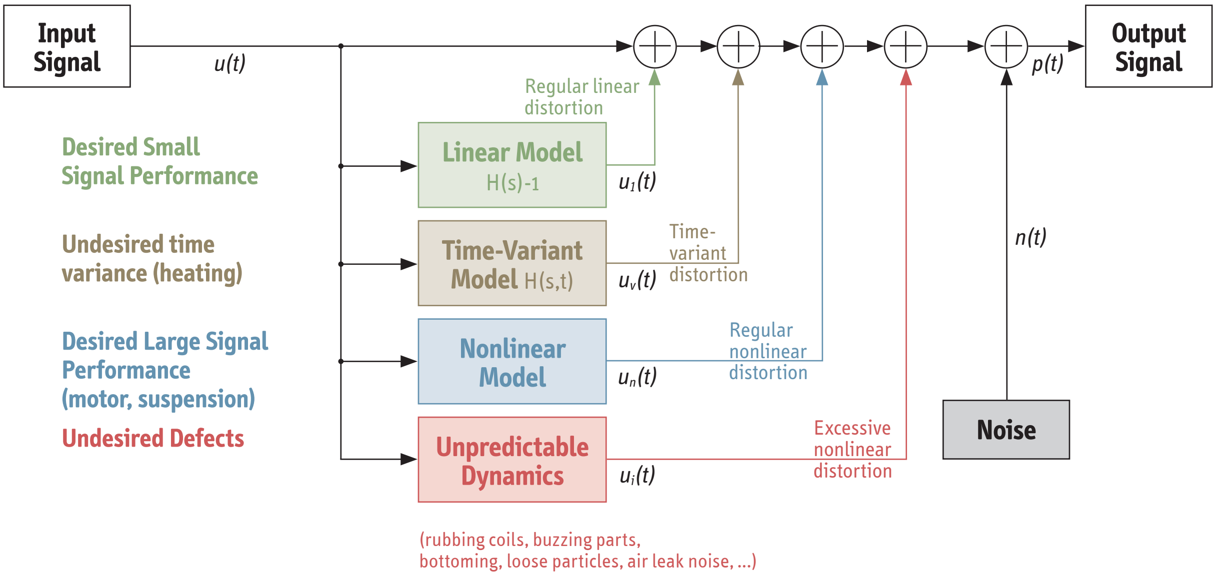
Most standard acoustical measurements shall be performed on a single reference point (e.g. 1 m distance on-axis) without assessing the full directivity of the loudspeaker. The **In-situ Room Compensation (ISC)** module copes with the imperfections of the acoustical environment (room, positioning, test box). It automatically generates a complex compensation function $H_c(f)$ that is used in a filter to transform the microphone signal $p_{test}(r_t)$ measured at a convenient position (e.g. near-field) into a simulated free-field signal $p_{free}(r_f)$ at the requested reference point r_r (e.g. in the far-field) according to IEC standard 60268-21.



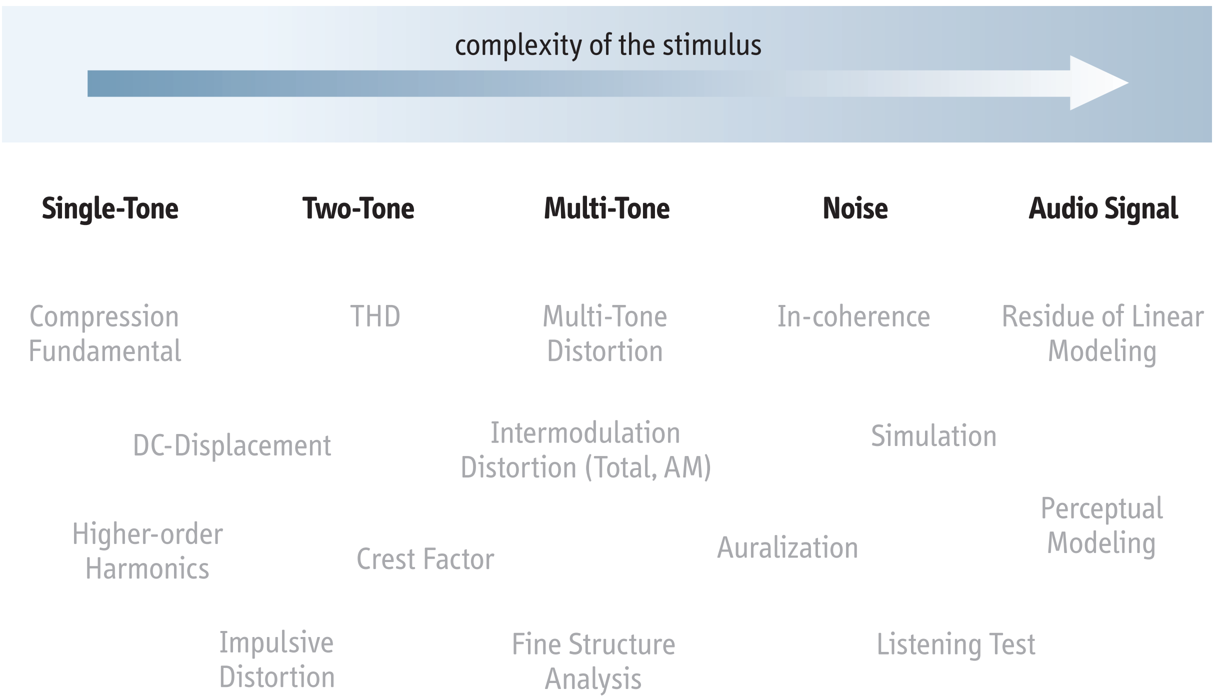
The inverse filtering applied prior to the signal analysis ensures accurate measurement of nonlinear distortion and transient behavior (burst testing). The **ISC** module uses reference data H_{ref} provided by near-field scanning or by conventional measurements performed under acceptable conditions.

OVERCOMING SIGNAL DISTORTION





An electro-acoustical transducer or any real audio device cannot generate an output signal that is identical with the input signal. These deviations, called signal distortion, are generated by five fundamental subsystems that differ with respect to linearity, time variance and predictability. Besides random noise, most of the signal distortion depends on the instantaneous properties of the stimulus. Some artificial test signals have a low complexity in order to only excite a selected frequency range and simplify the analysis of the output signal as well as the interpretation of the results. However, a comprehensive evaluation of the audio system requires more complex stimuli (e.g. music), which are used in the target application.

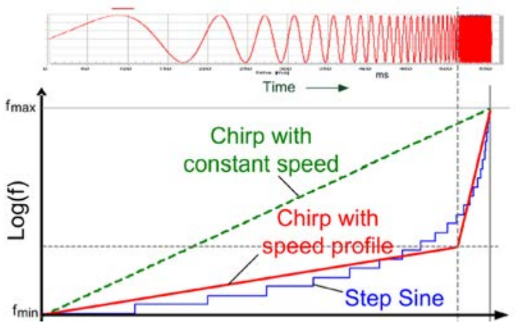


FAST CHIRP MEASUREMENTS (TRF, SPL)

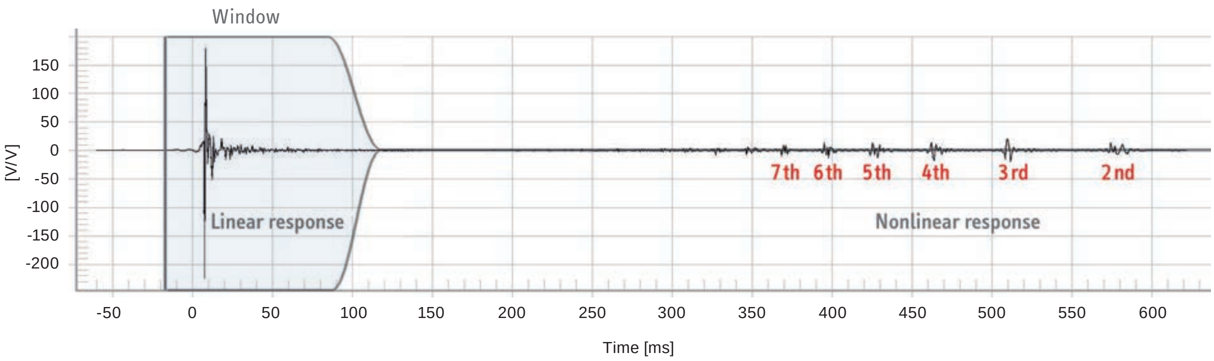
- Fast testing at high SNR
- Frequency response at high resolution
- THD and harmonic distortion components
- Impulsive distortion (Rub & Buzz)
- Perfect for EoL testing

The **Transfer Function Measurement (TRF)** and the **Sound Pressure (QC-SPL)** task are universal measurement modules for all kinds of electrical, mechanical and acoustical signals in the time and frequency domain. The excitation signal is a logarithmic chirp with adjustable sweep speed and amplitude profile to measure the frequency responses of the fundamental and various signal distortions at a desired resolution, bandwidth and SNR according IEC standard 60268-21.

The **QC-SPL** provides a chirp with increasing sweep speed, which allows ultra-fast testing at end-of-line with a dense excitation spectrum, giving higher frequency resolution than a stepped sine sweep.

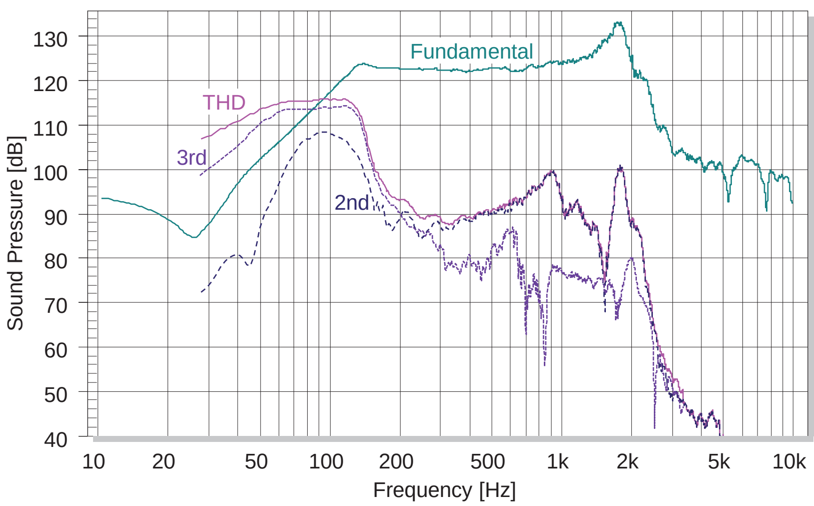


Various analysis techniques (e.g. windowing) are provided to separate the direct sound part from early reflections, room modes and harmonics in the impulse responses. The mean group delay is detected automatically from the maximum of the energy-time curve.



Transfer Function

Fourier transformation of the windowed impulse response provides the amplitude and phase of the transfer response in the frequency domain. Hilbert transformation is used to separate the minimum phase from excess phase.



Harmonic Distortion

The properties of the sinusoidal chirp signal allow for the calculation of the frequency responses of the 2nd, 3rd and higher-order harmonics (HOHD) and the total harmonic distortion (THD).

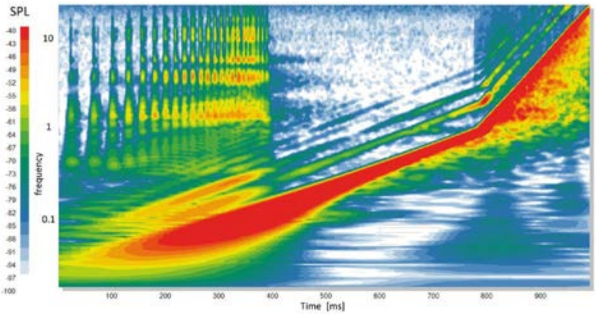
D

SENSITIVE RUB & BUZZ ANALYSIS (TRF, TFA, PLAY, SPL, MHT)

patented KLIPPEL technology

- Analysis in time and frequency domain
- Higher-order harmonics
- Fine-structure of impulsive distortion
- Detection of Rub & Buzz, air leakage, loose particles ...
- Root cause analysis
- Evaluation of maximum output

Distortion analysis in time-frequency domain (spectrogram)

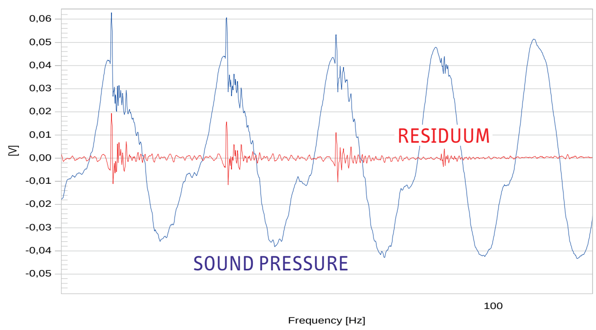


High-pass filtering, 3D spectrogram limits **(QC-SPL)** and the meta-hearing technology **(QC-MHT)** are required to separate the irregular distortion (residuum) from the fundamental and other regular distortion in the time domain, giving the *Impulsive Distortion* according to IEC 60268-21. The filtered time signal exploits the phase and amplitude information of all high-frequency components. The **Player (PLAY)** module allows the investigation of the fine structure of the impulsive distortion by the human ear in a lower frequency band by downsampling.

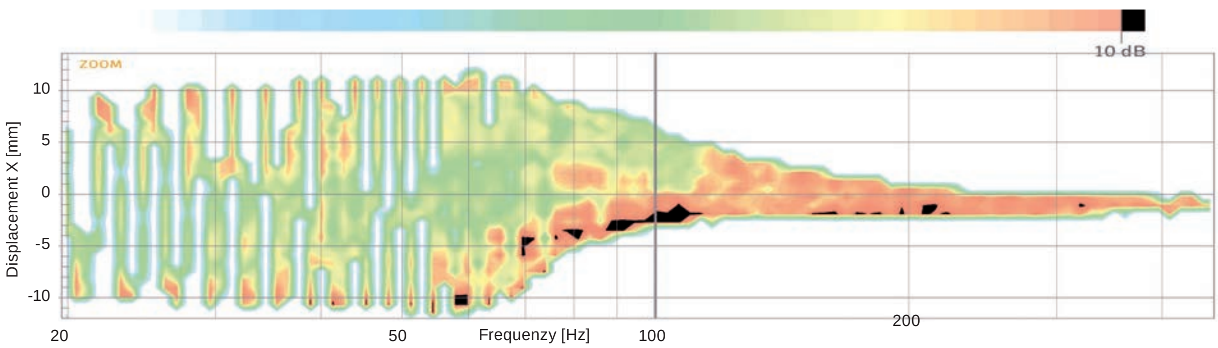
The **Transfer Function (TRF) and QC-SPL** modules offer outstanding sensitivity for all kinds of distortion caused by undesired dynamics (e.g. rocking modes), manufacturing problems and defects (e.g. Rub & Buzz). Irregular distortion components might have low energy but could become worse over time and clearly audible in the final product. Therefore, design and manufacturing need tools to ensure robust products and to avoid shipping defective units.

The **Time-Frequency Analysis (TFA)** is a powerful tool for investigating the impulse response and nonlinear distortion generated by an audio system. The spectrogram reveals the generated nonlinear signal components that are at much higher frequencies than the fundamental excitation frequency of the chirp stimulus. The RMS value of the accumulated higher-order harmonics with order n > 10, according to IEC 50268-21, is a good indicator for defects generating deterministic properties (e.g. bottoming). However, defects with random properties (e.g. loose particles) generate distortion components that are not concentrated at the harmonics but are distributed over all frequencies in the analyzed frequency band.

Distortion analysis in time domain



Instantaneous crest of higher-order distortion (ICHD)



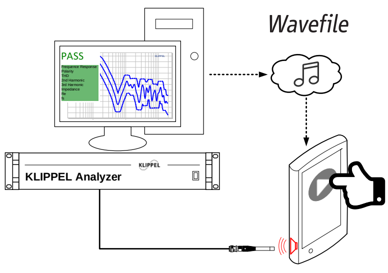
The impulsive distortion has a higher crest factor than other regular nonlinear distortion and noise. The diagram above shows the instantaneous crest factor of the impulsive distortion (coded as color) versus frequency and voice coil displacement. Around the resonance frequency, the crest factor of the impulsive distortion exceeds the critical value of 10 dB and generates black spots at the inner turning point, which is a symptom of voice coil rubbing.

WIRELESS, DIGITAL & OPEN-LOOP TESTING (SYN, TRF, …)

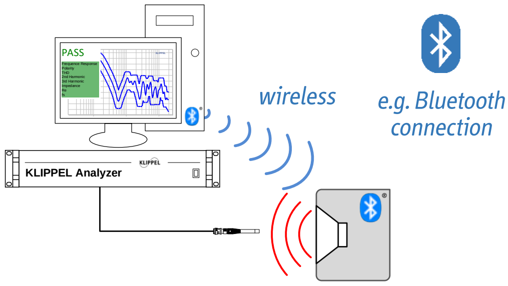
- Synchronize any audio device
- Sample rate tolerance
- Transmission error detection
- Trigger test with unique watermark
- Export and import wave files

The **External Synchronization (SYN)** uses a unique acoustical watermark stimulus to trigger measurements, which is especially useful for EoL applications.

Testing audio devices with digital or wireless interfaces (e.g. Bluetooth® devices) generates challenges like unknown or varying delay, sample rate conflicts, dropouts and other transmission errors. The KLIPPEL Analyzer System provides a reliable, flexible and easy-to-use solution even for sensitive applications with strict requirements on phase accuracy (e.g. in **Near Field Scanner**).
Stand-alone devices such as tablets and smart speakers require the stimulus to be transferred to the device or to the cloud and played back autonomously. The software provides the stimulus signal as an audio file and awaits the playback. Additionally, audio file import provides the off-line analysis or recorded responses (e.g. microphone testing).



Open-loop testing



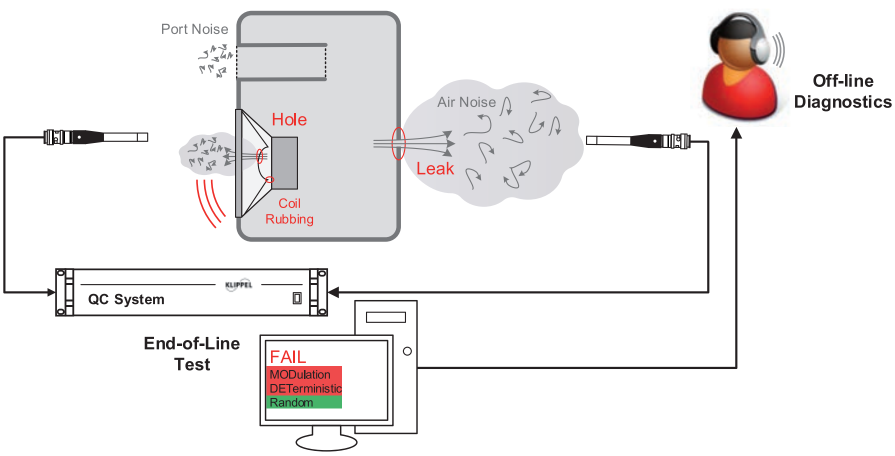
Closed-loop testing

AIR LEAK DETECTION (ALD) & STETHOSCOPE (ALS)

patented KLIPPEL technology

- Detect small air leaks
- Ambient noise immune (auto repeat)
- End-of-line application (ALD)
- Off-line diagnostics (ALS)
- Localize and auralize defects

The **Air Leak Detection** module (**QC-ALD**) applies special audio signal processing to isolate pulsating flow noise generated by air leakage in loudspeaker drivers and enclosures as well as ports and orifices. Limits can be applied to the resulting absolute and relative distortion levels in order to qualify and quantify air leakage and many other defects (rubbing, buzzing, loose particles).
This technology can be applied as an automatic end-of-line test which seamlessly integrates in the test sequence with single tone excitation or as part of the standard chirp measurement (**QC-SPL**).
Sharing the same processing kernel, the **Air Leak Stethoscope** (**QC-ALS**) is a powerful off-line diagnostics tool to auralize and locate defects interactively.

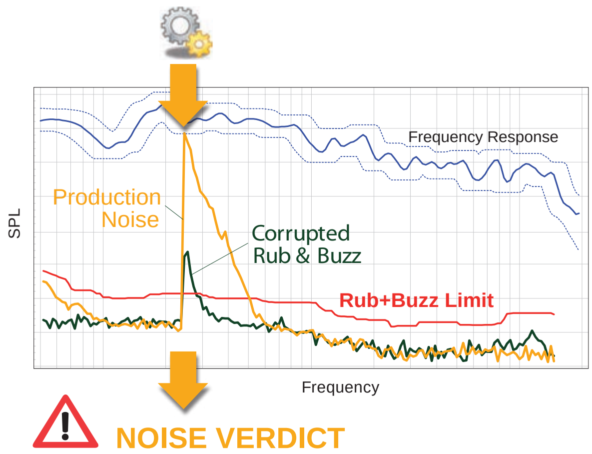


COPE WITH PRODUCTION NOISE

PRODUCTION NOISE DETECTION (SPL)

patented KLIPPEL technology

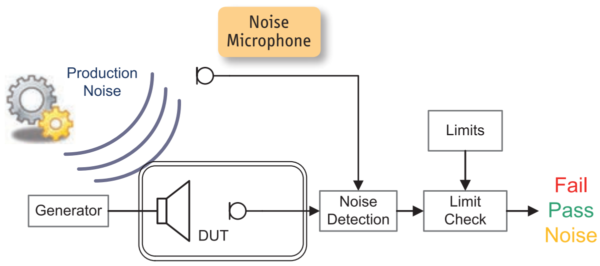
- Cope with noise in production or lab
- Find corrupted measurements
- Reduce false rejects
- Improve yield rate in production
- Solves test box limitations



Detecting defects with the highest sensitivity requires robustness against any external noise from production (EoL) or in a lab environment (RnD). The level of impulsive disturbances may be much higher than defect symptoms. Even well-designed test enclosures do not attenuate ambient noise by more than 40 dB, which is insufficient to prevent noise corruption.

The unique *Production Noise Detection* of the **QC-SPL** task uses a dedicated ambient microphone with identical signal processing as the test microphone. The two signals are correlated with respect to time and level (incl. box attenuation) to distinguish actual defects from external corruption.

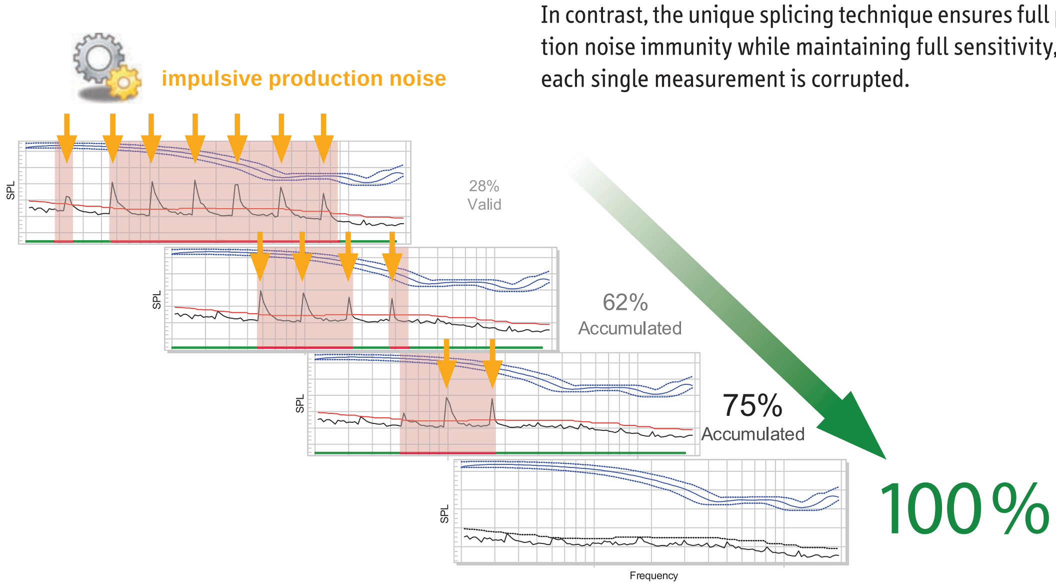
The NOISE verdict clearly indicates a corrupted result.



PRODUCTION NOISE IMMUNITY (PNI)

patented KLIPPEL technology

- Repeat and accumulate valid data
- Highest test speed in noisy environment
- Distinguish defects from noise
- Efficient solution with minimum hardware



The **Production Noise Immunity (QC-PNI)** module is based on the *Production Noise Detection* but provides additional benefits for end-of-line testing. Corrupted measurements are repeated automatically while valid parts of each measurement are stored and merged together, eventually providing complete valid results.

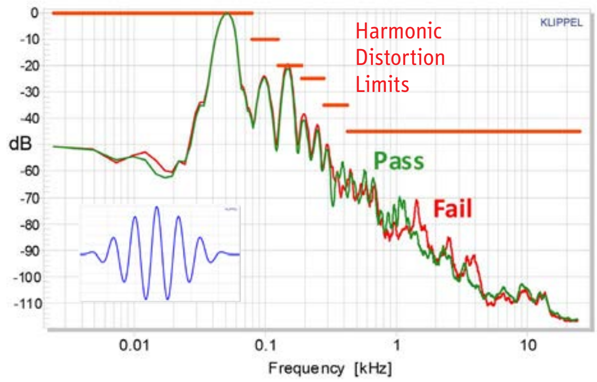
Ordinary techniques like a simple repeat approach, where the signal is repeated until one completely undisturbed test finishes, or averaging, which impairs sensitivity, does not solve the problem.

In contrast, the unique splicing technique ensures full production noise immunity while maintaining full sensitivity, even if each single measurement is corrupted.

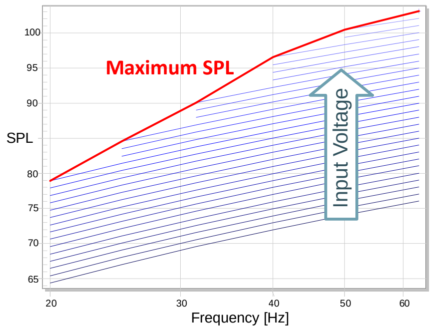
MAXIMUM SPL OUTPUT

TONE BURST MEASUREMENT (TBM)

- Maximum short-term SPL (CTA/ANSI/IEC)
- Sinusoidal transient stimulus (burst)
- Minimum heating of the device
- Harmonic distortion
- Simulated anechoic far-field condition
- Compensation of room influence & distance



Spectrum of the reproduced burst signal

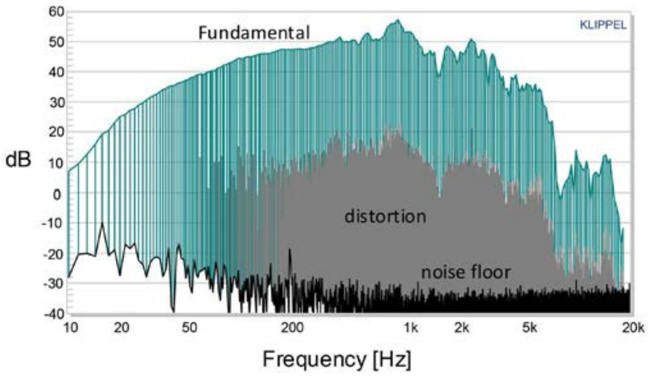


Maximum SPL versus frequency limited by harmonic distortion limits

The **Tone Burst Measurement (TBM)** module uses a transient sinusoidal burst to measure the peak SPL and harmonic distortion versus frequency and amplitude according to Standard ANSI/CTA-2010 and ANSI/CTA-2034. During the test, the input voltage is automatically increased until the harmonic distortion exceeds a critical threshold, avoiding damage to the device under test. The short tone burst in the **TBM** generates a minimum of voice coil heating in the transducer but still activates nonlinearities inherent in the system. A second state variable (displacement, voltage, current) can be measured simultaneously.

MULTI-TONE MEASUREMENT (MTON)

- Maximum continuous SPL (CTA/ANSI/IEC)
- Shaping represents typical audio stimulus
- Amplitude compression (thermal & nonlinear)
- Harmonic & intermodulation distortion
- Passive and active systems (e.g. Bluetooth)
- Compensation of room influence & distance



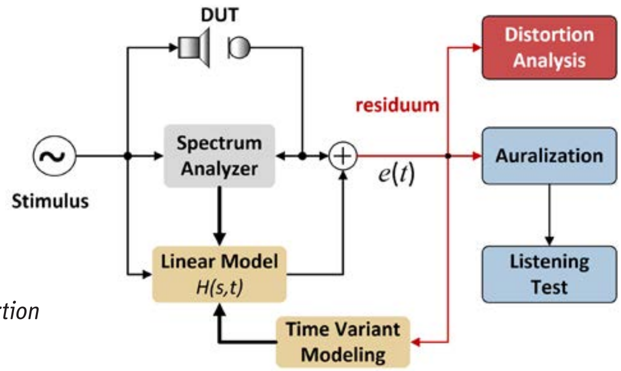
The **Multi-Tone Measurement (MTON)** module uses a multi-tone complex as a broadband, pseudo-random stimulus with a spectrum and crest factor similar to a typical audio signal. The compression of the fundamental and nonlinear distortion components are measured automatically and compared with permissible values to find the maximum output, which is limited both by heating and nonlinearities in the active or passive audio system. The **MTON** complements the **TBM** measurement by defining a meaningful max SPL value according to standards like ANSI/CTA-2010/2034 and IEC 60268-21. Both tests, **TBM** and **MTON**, can be performed in the near-field of the loudspeaker operated in a non-anechoic environment by performing an inverse filtering of the microphone signal based on the compensation function provided by the **ISC** module. This generates accurate data corresponding to standard conditions.

LIVE AUDIO ANALYZER (LAA)

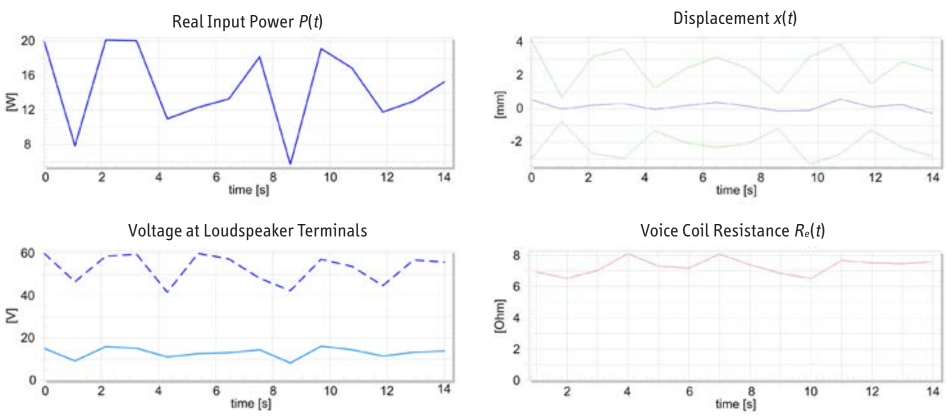
patented KLIPPEL technology

- Long-term monitoring
- Active and passive systems
- Assesses heating and aging
- Distortion measurement with music
- Physical and perceptual evaluation

The **Live Audio Analyzer (LAA)** evaluates the performance of active and passive audio systems under normal working conditions, reproducing an arbitrary audio signal (speech, music) or dedicated test stimuli (multi-tone, chirp, single/two tone). Multiple sensors are supported for monitoring the instantaneous state of the transducer such as terminal voltage, input current, voice coil displacement and sound pressure output. Adaptive modeling is used to capture the time-varying properties caused by heat, visco-elastic changes and aging.

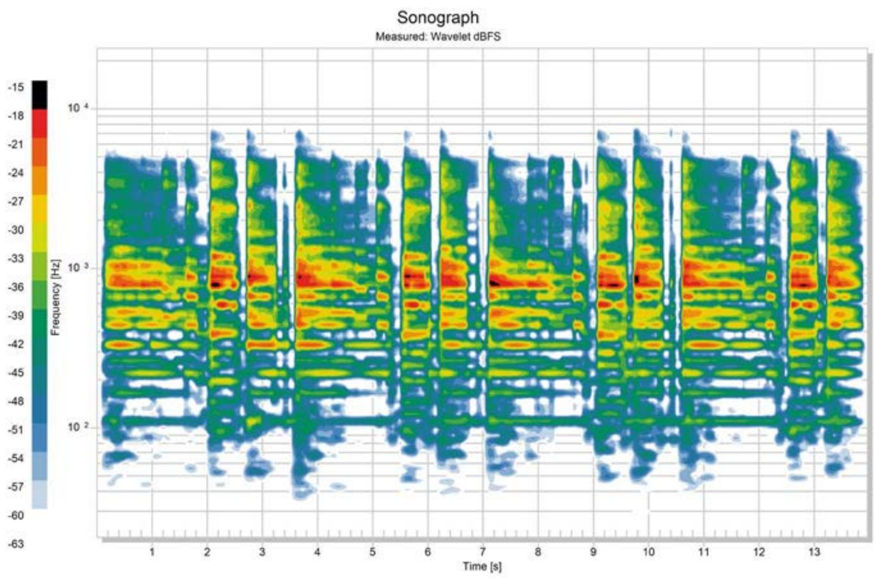


Separating the signal distortion by adaptive modeling



State signals of the transducer under test versus time

The error signal generated in the linear modeling (residuum) reveals the distortion generated by nonlinearities and defects (Rub & Buzz). The residuum and linear model output allow the signal distortion to be auralized and systematic listening tests to be performed to evaluate audibility and impact on perceived sound quality.



D

POWER TEST SOLUTIONS (LAA, MTON, LPM, LSI, KCS)

D

- Long-term testing of transducers and systems
- Analog, digital, wireless and audio file
- Stimuli: standard and user (e.g. music)
- Measure coil temperature, displacement
- Maximal amplitude limits (P_{max}, X_{max})
- Analyze root causes of damages
- Monitor climate influence, load changes
- Assess durability and reliability

The KLIPPEL Analyzer System and Controlled Sound Technology (**KCS**) provide powerful and flexible solutions for long-term testing of the performance and reliability of all kinds of audio devices. An internal generator provides a variety of test signals complying with IEC, EIA and other standards including noise with amplitude profile (stepping, ON/OFF cycle). The stimuli can be supplied to the device under test via the KA3 hardware platform using an internal or external power amplifier, via digital streaming or as audio files from a playlist (M3U). Measuring current and terminal voltage in KA3, smart amplifiers or **KCS** applications is the preferred way to monitor parameters (e.g. fs, coil offset, Kms(x)) and state variables for transducers. A single microphone is sufficient for testing multiple active systems in one room by using watermarked stimuli. A range of statistical analysis tools will evaluate your data and can provide automatic defect classification and root cause analysis.

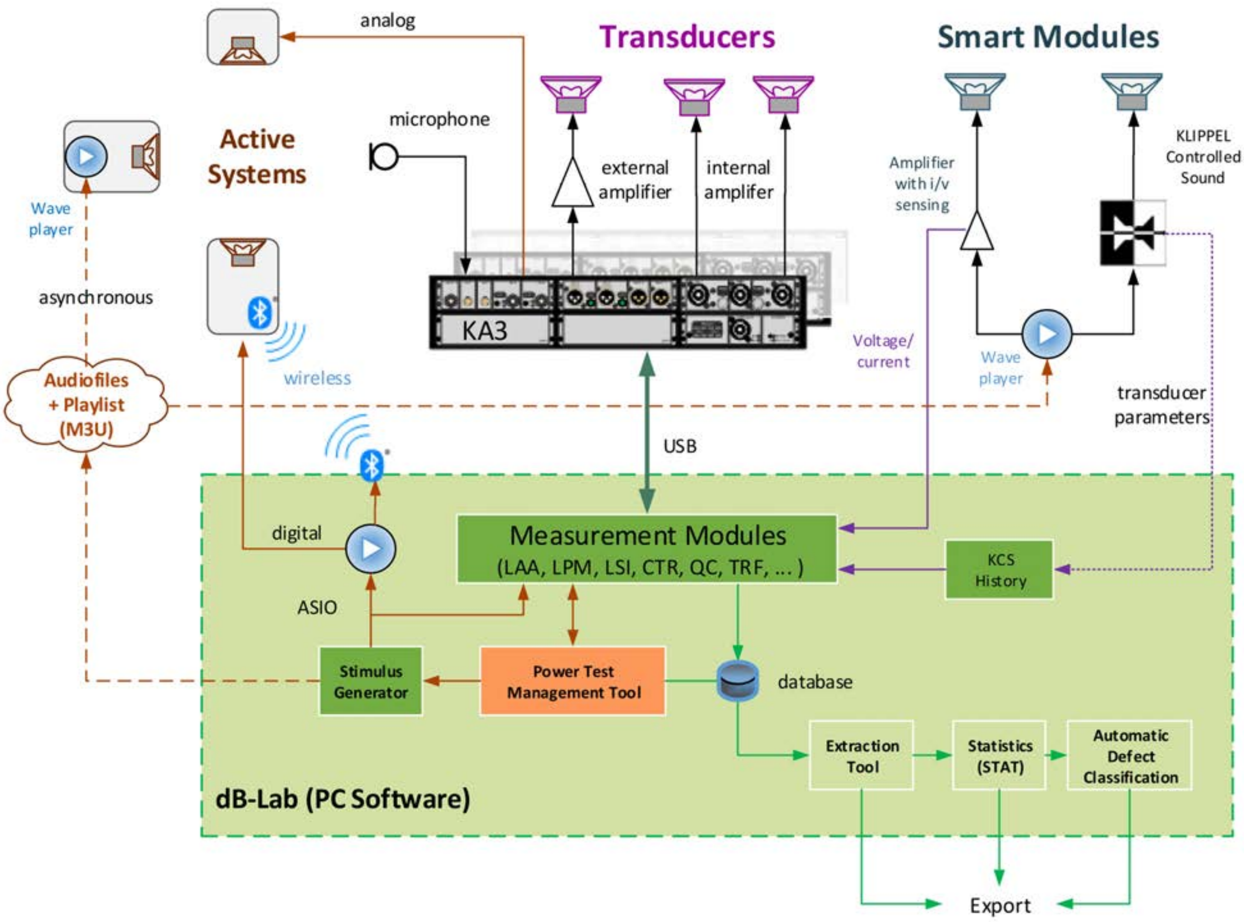
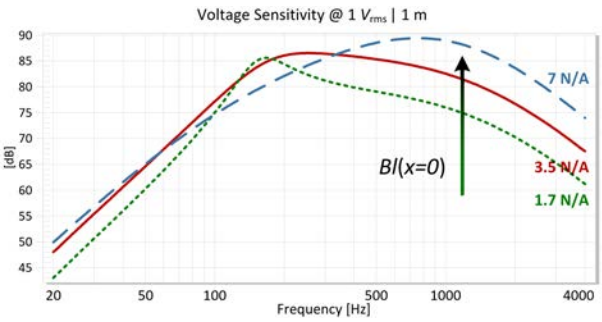


Illustration of potential solutions for long-term testing of active and passive audio systems and components.

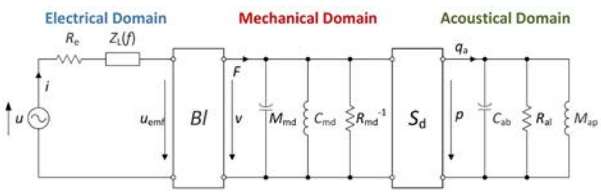
LINEAR SIMULATION (LSIM)

- Fast lumped parameter simulation
- Linear model valid at small amplitudes
- SPL response, transfer functions, impedances
- Any stimulus (music, test signal)
- Efficiency and voltage sensitivity
- Common enclosure types and complex loads
- Acoustical parameters from geometrical input

The **Linear Simulation (LSIM)** module describes the small signal performance of transducers and complete passive and active systems at small amplitudes. It is based on lumped parameter modeling (T/S) with frequency-dependent impedances and transfer functions that consider lossy inductance, creep, modal vibration, mechanical and acoustical loads, room influence and electrical filters (crossover). This module simplifies the design of headphones, loudspeakers with common enclosure types, passive radiators and more complex loads. By considering the spectral properties of the stimulus, the **LSIM** module calculates the overall efficiency and the voltage requirement of the amplifier.



Investigating the influence of the force factor Bl on the voltage sensitivity of the loudspeaker system.

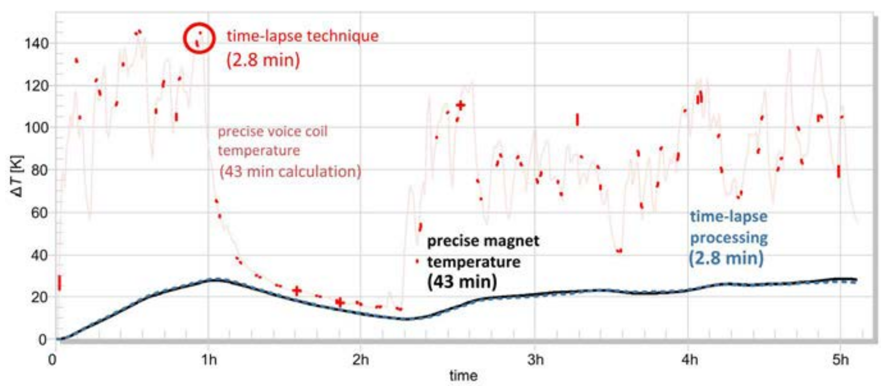


Equivalent circuit based on lumped parameter modeling

THERMAL SIMULATION (SIM2, SIM-AUR)

- Predicts voice coil, gap and magnet temperature
- Based on thermal parameters
- Any stimulus (music, test signal)
- Analyze heat flow
- Optimize air convection cooling
- Faster than real-time

The large signal simulation modules **Simulation (SIM2)** and **Simulation-Auralization (SIM-AUR)** calculate the heat flow and mean temperature of the voice coil, pole plates and magnet/frame structure based on thermal parameters. While the **SIM2** describes the transducer under steady-state conditions where the heating and cooling process is in the thermal equilibrium, the **SIM-AUR** module reveals the thermal dynamics for any input signal (music) at full temporal resolution. Both modules consider the nonlinear air convection cooling.



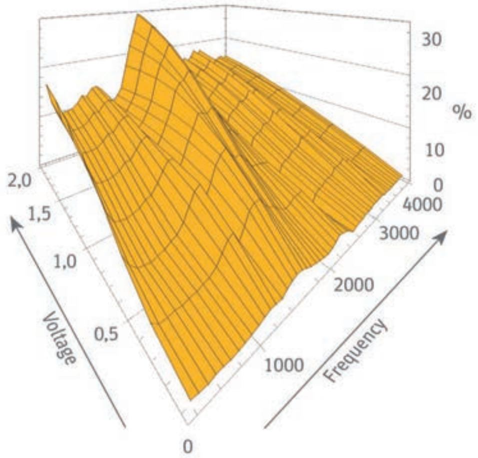
Fast simulation of voice coil and magnet temperatures while reproducing 5 h of common music material

The mass of the iron parts, magnet and frame generate thermal time constants that exceed minutes or even 1 hour in large loudspeakers. While the **SIM-AUR** can calculate all states at full sample frequency (48 kHz) faster than real-time, the thermal identification can be sped up by a factor of 15 using a time-lapse technique to approximate the same maximum temperatures.

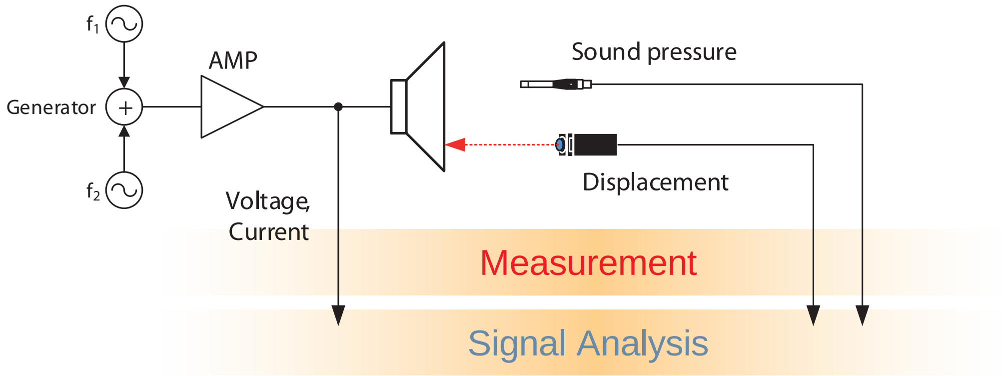
3D DISTORTION MEASUREMENT (DIS)

- Steady-state measurement
- Analysis of large signal behavior
- Harmonic distortion, intermodulation
- Dependency on amplitude and frequency
- Voice coil displacement (peak, RMS, DC)
- Reveals amplitude compression
- Active transducer protection

Third-order intermodulation distortion in percent

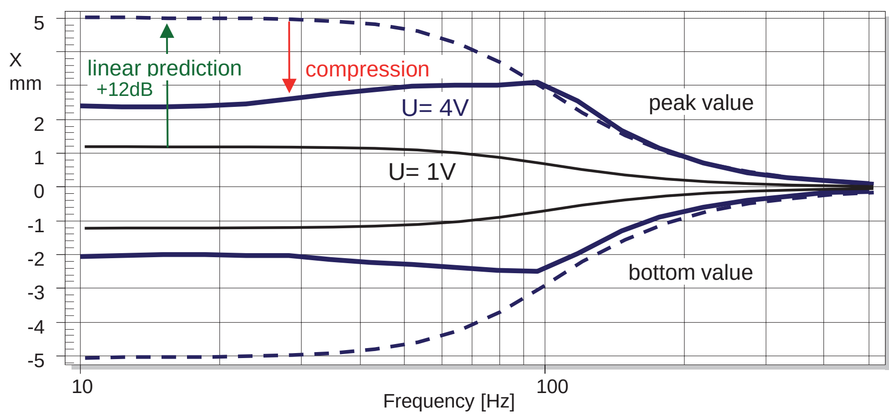


The **3D Distortion Measurement (DIS)** module performs a series of measurements using a single or two-tone stimulus varied in amplitude and frequency to check the large signal behavior of audio systems according to IEC standard 60268-21. Voltage at the terminals may be configured to adjust automatically between user-selected levels. The voice coil temperature is monitored by impedance measurements. The measurement is interrupted if excessive mechanical and/or thermal overload will damage the transducer. The FFT is synchronous to the stimulus length, giving maximal spectral resolution and eliminating windowing. The results are the steady-state amplitude responses of the DC component, fundamentals, harmonic and intermodulation components. The amplitude variation of the stimulus reveals the thermal and nonlinear compression of the spectral components.



Harmonics DC-component Fundamental
Intermodulation Compression Peak/Bottom RMS-Value

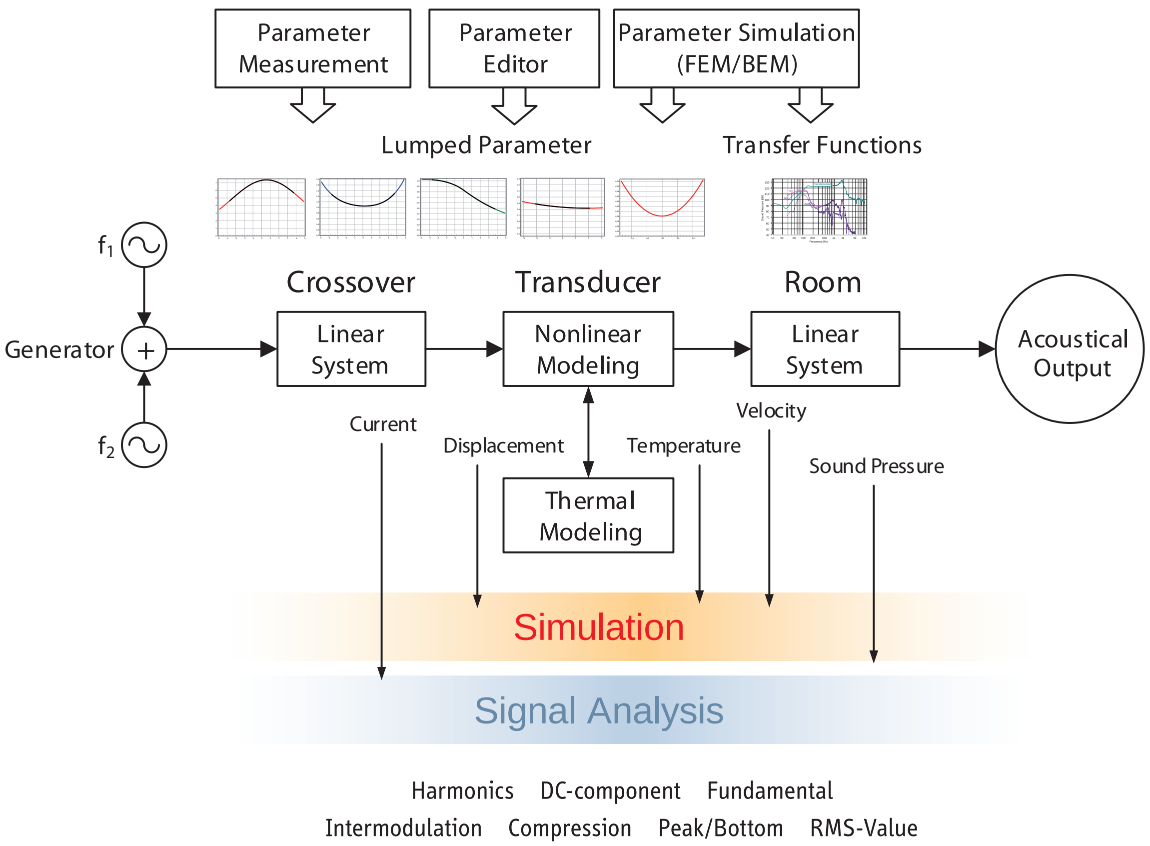
The increase of the input voltage by 12 dB generates an amplitude compression in the measured peak and bottom displacement (increase < 12 dB).



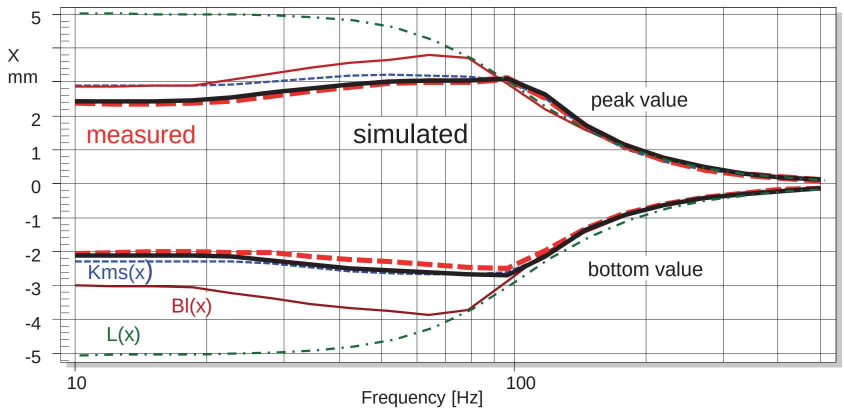
SIMULATION (SIM2, SIM-AUR)

- Large signal modeling
- Based on lumped parameters
- Active system design (e.g. enclosure)
- Results comparable with DIS module
- Evaluation of design choices
- Save cost in prototyping
- Analyze effect of each nonlinearity

The nonlinear simulation modules **Simulation (SIM2)** and **Simulation-Auralization (SIM-AUR)** are based on linear, non-linear and thermal parameters, which can be provided by measurements using the LSI and LPM module or calculated by other external simulations such as FEA and BEA. A parameter editor provided in the **SIM2** module allows nonlinearities in virtual loudspeaker to be synthesized and design choices to be evaluated in order to understand the relationship between physical causes and distortion.



The **Large Signal Simulation (SIM2)** is the twin module of the **3D Distortion Measurement (DIS)**. It uses the same generator and signal analysis to provide results in a format that supports a direct comparison between simulation and measurement.



The nonlinear model can describe the compression of the peak and bottom displacement as measured by the DIS module. By activating only one nonlinear parameter while assuming all others as linear, the contribution of each separated nonlinearity can be assessed. The nonlinear stiffness $K_{ms}(x)$ is the dominant cause of the compression in the example speaker shown here.

D

COMBINE PERCEPTIVE AND PHYSICAL EVALUATION

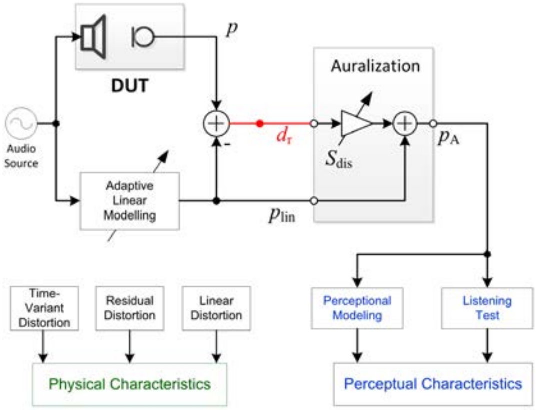
D

- Auralization of signal distortion
- Arbitrary stimulus (music, test signals)
- Scale distortion in virtual output
- Assess distortion ratio in audio signals
- Define target performance

Auralization is a new technique that combines physical modeling and measurement with systematic listening to assess the impact of signal distortion on the perceived sound quality. This new technique can be applied to any test signal or audio stimulus and synthesizes an output signal comprising distortion components that are either artificially attenuated or enhanced by a user-defined scaling factor. To generate a virtual audio output in a mixing device, the distortion components are separated from the linear signal by transducer modeling or measurements.

DIFFERENCE AURALIZATION (DIF-AUR)

patented KLIPPEL technology



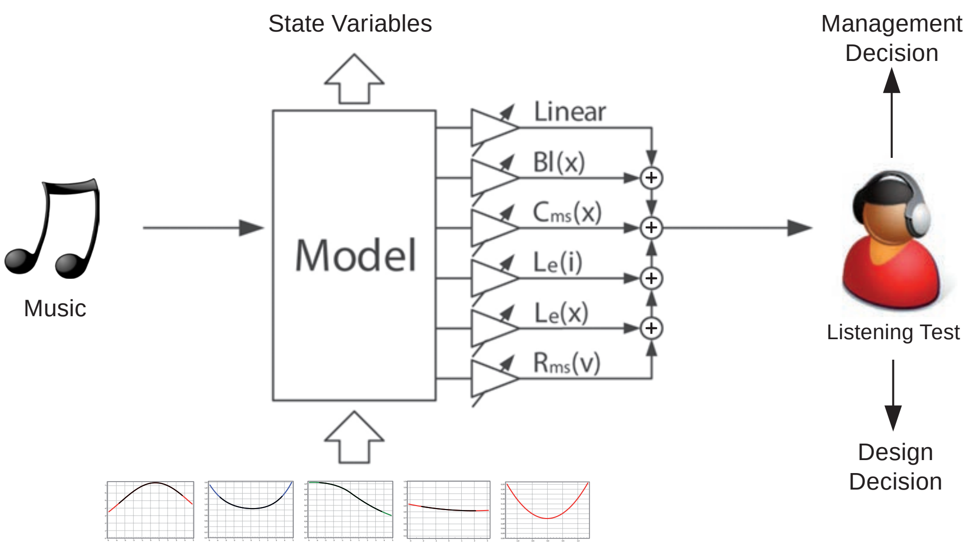
For auralizing irregular distortion symptoms (e.g. generated by rubbing voice coil, buzzing car doors, loose particles and other defects), the **Difference Auralization (DIF-AUR)** requires two wave files. The module can either use a test measurement versus a reference measurement or it can model the linear transfer function by using the test measurement versus the stimulus. The output wave files with varying levels of distortion may be used to perform listening tests or as an input for perceptual modeling.

MODEL-BASED AURALIZATION TECHNIQUES (SIM-AUR)

patented KLIPPEL technology

The model-based auralization technique **Simulation-Auralization (SIM-AUR)** is based on electrical, mechanical and acoustical parameters. It is perfect for the evaluation of motor, suspension and enclosure design choices.

Temperature Cone Velocity Distortion Displacement Sound Pressure

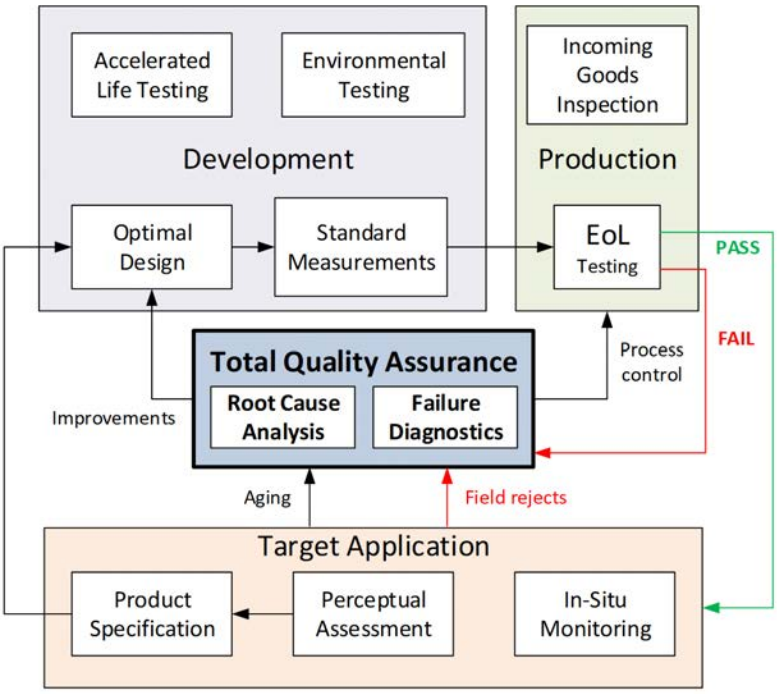


ASSURING QUALITY

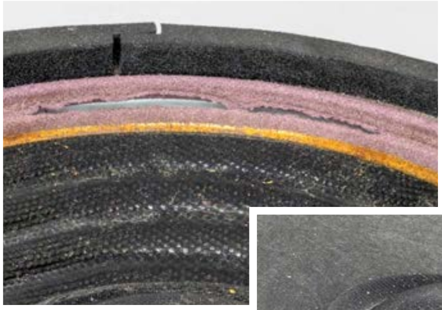
- Best performance-cost ratio
- Lean manufacturing (5M‘s)
- Maximum yield rate
- Autonomous self-testing
- In-situ monitoring
- Failure analysis
- Risk management
- Reliable products
- Customer satisfaction

The KLIPPEL Analyzer System supports quality control over the complete product life cycle, including a clear specification of the target performance, an optimal design process, a consistent replication of the approved prototype in manufacturing and self-monitoring of the smart audio product in the field. Powerful measurement techniques combined with effective data management provide traceability over the supply chain to identify the root causes of failures. Characteristics compliant with international standards ensure comparability of the results. Statistical analysis extracts the relevant information and simplifies the interpretation of the results.

QA



Defective surround



Loose wire

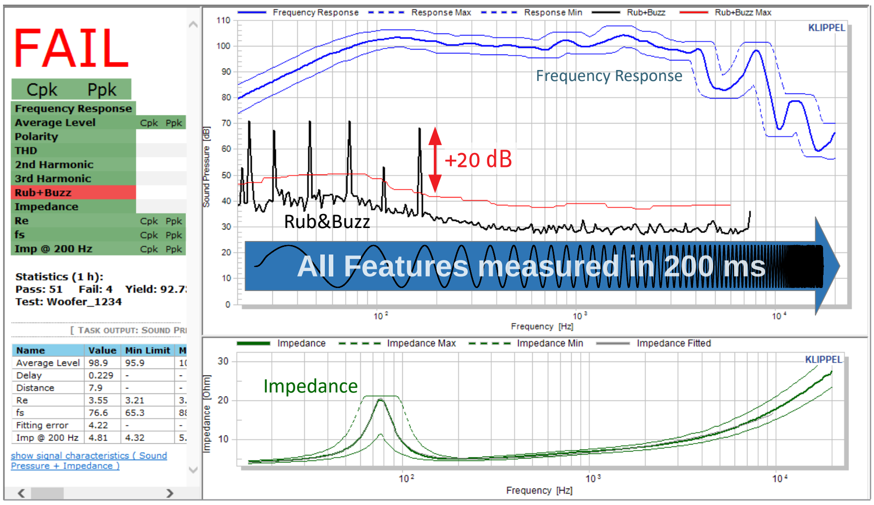
Leaky dust cap

Burnt voice coil

TEST SPEED AND SENSITIVITY

- Ultra-fast testing (> 200 ms)
- Chirp with level and sweep speed profile
- Reliable detection of failures
- More sensitive than a trained human tester
- Robust in a production environment
- Copes with ambient noise

The KLIPPEL Analyzer System provides unique QC measurements optimized for end-of-line (EoL) testing that provide maximum sensitivity for defects while requiring minimum measurement time. These fast measurements use special stimuli such as a continuous sine chirp with a flexible sweep speed, which is slowed down at frequencies where the loudspeaker needs a longer excitation. Voice coil rubbing, loose particles, air leakage and other irregularities, which generate impulsive distortion that is hardly audible in a noisy production environment, can be reliably detected. Defects such as these become worse over time and will finally be detected by the end user if not sorted out at EoL.

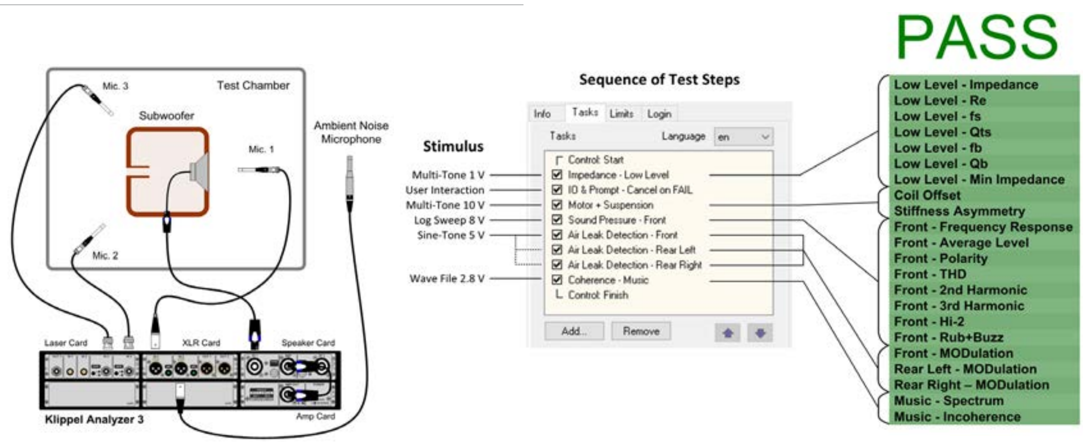


Electrical and acoustical testing of a woofer within 200 ms while providing high senstivity for loudspeaker defects (one grain of salt generates 20 dB above limits).

FLEXIBILITY

- Comprehensive testing of passive and active devices
- Multiple audio connections (Bluetooth, HDMI, analog)
- Unlimited sequence of KLIPPEL modules
- Repeat corrupted task automatically
- One stimulus – multiple analyses
- Basis for 100 % testing

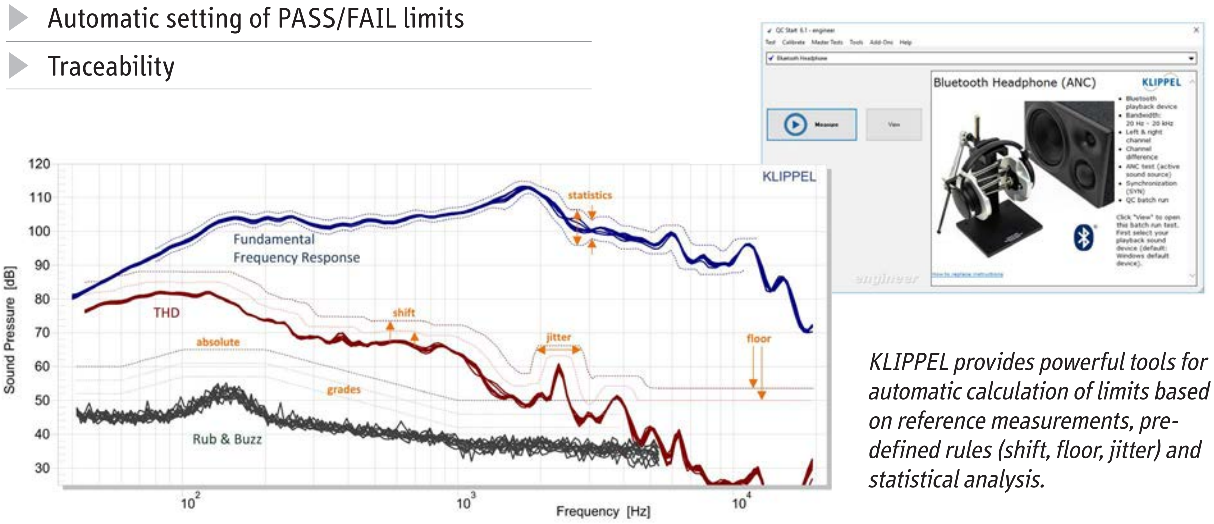
The KLIPPEL Analyzer System combines the advantages of a comprehensive measurement suite with the flexibility of a modular system. It provides optimal solutions for testing any electro-acoustical transducer as well as passive and active audio systems with analogue, digital or wireless input. Multiple measurement tasks can be interlaced in one QC operation to realize ultra-fast testing. QC operations and any other measurement module can be linked to an unlimited test sequence for more complex measurements, such as those required for type approval tests or detailed diagnostics.



EASY TO USE

- Train new operators in minutes
- Foolproof operation
- Intuitive, multi-lingual user interface
- Quick start with application templates
- Automatic setting of PASS/FAIL limits
- Traceability

QC Start is an efficient framework software exclusively designed for EoL testing to simplify test and data management. It offers powerful features while the setup and complexity is minimized. With the support of templates and a simple user interface, an operator is trained within minutes. Full traceability of any changes, foolproof operation using barcodes and visual illustrations ease and assure efficient QC operation.

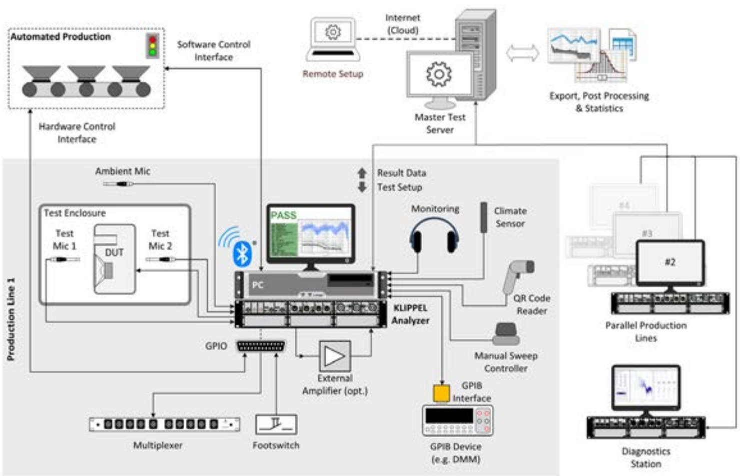


KLIPPEL provides powerful tools for automatic calculation of limits based on reference measurements, predefined rules (shift, floor, jitter) and statistical analysis.

SMOOTH INTEGRATION

- Robust hardware
- Simple interfaces
- Ready for automated environment
- Low time variance for fixed cycles
- Synchronize multiple test stations
- Statistical process control and diagnostics
- Remote setup and monitoring

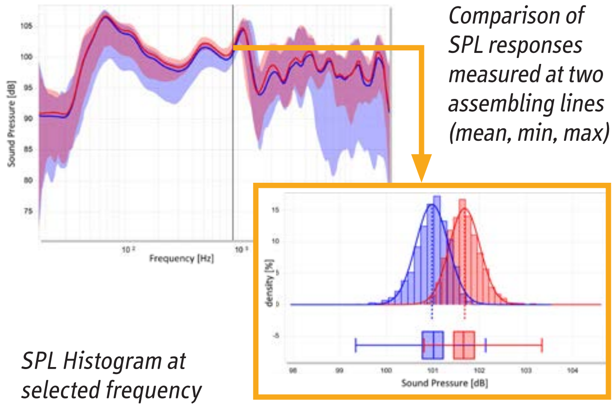
The KLIPPEL Analyzer components integrate perfectly into automated production environments for 100 % end-of-line testing. A variety of open hard- and software interfaces are available for data access, remote control and monitoring status information. 3rd party test equipment can be integrated and controlled (e.g. via *GPIB*) as well as wireless interfaces configured and used (e.g. automatic Bluetooth pairing). Even a cloud based wave file analysis is supported and satisfies todays requirements for distributed and lean manufacturing.
The comprehensive test data format in a single file database allows result data, test settings and limits to be easily exchanged between supplier and customer.



PROFIT FROM STATISTICS

STATISTICAL ANALYSIS (STAT)

- Extract essential information
- Pool-based organization of results
- Outlier and PASS/FAIL detection
- Visualization of differences and variances
- Detection of golden units
- Point & click for limit setting
- Text export of all calculation results



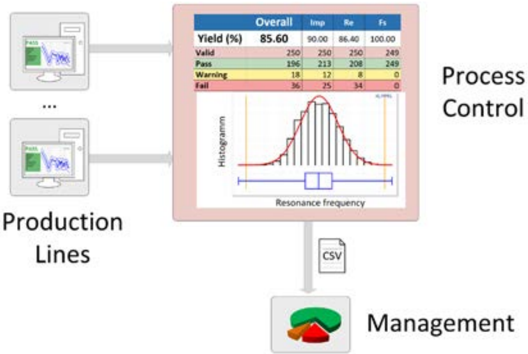
Comparison of SPL responses measured at two assembling lines (mean, min, max)

SPL Histogram at selected frequency

The **Statistical Analysis (STAT)** module is the basis for comparing measurement results of multiple test objects and statistical analysis. Visualizing the min/max range and the variance of a data pool becomes an easy task. Test objects may be organized into different pools and compared visually or through calculations. Limits may be entered interactively or via parameters. They can be exported or used for sorting the test objects into pools. The detection of golden units finds those test objects that fit best to the defined reference results.

YIELD & SINGLE VALUE STATISTICS (YST)

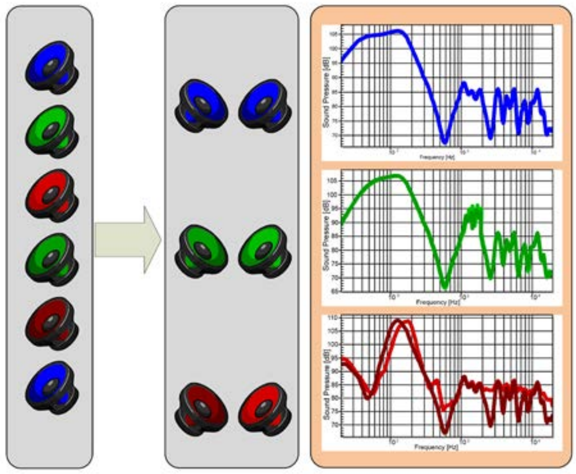
- Fast, comprehensive, simple
- One-click yield calculation
- Analysis of single values (fs, Re, ...)
- Boxplots, distribution, outliers
- Time course analysis
- Automated reports and exports



The **Yield & Single Value Statistics (YST)** module provides an overview of production with one or multiple lines by calculating the yield and single value statistics. The clear tabular and graphical representations are the basis for comprehensive automated statistical reports. The customizable integration into the QC Software allows fast access with only one mouse click. Optional CSV export provides an open interface to 3rd party statistical software.

MATCH SPEAKER (MSP)

- Cope with production variances
- Maximize quality by sorting
- Automatic matching
- Post-processing of test results

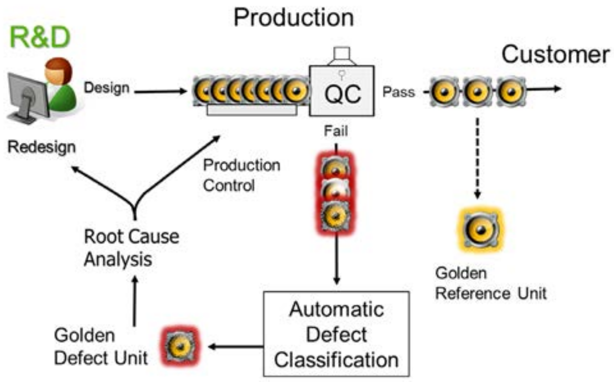


Unsorted DUTs *Optimal pairs* *Frequency responses*

The **Match Speaker (MSP)** Tool automatically selects optimal pairs of tested speakers to form high-quality stereo systems (e.g. for audiometry, high-end). Different pairing algorithms are available in order to find the best matching pairs or the maximal amount of pairs. Weighting functions and deviation limits provide a customizable solution to yield the best audio quality from production.

LEARNING FROM PRODUCTION

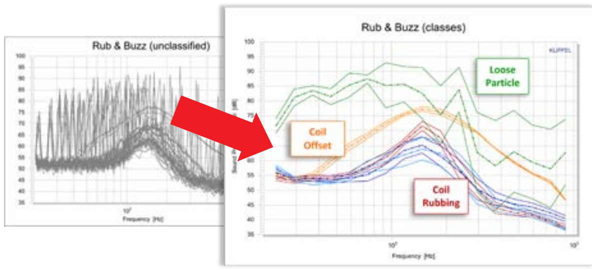
- Accumulate knowledge from EoL-testing
- Increase transparency of the process
- Trigger actions quickly for process control
- Increase yield rate and product reliability
- Simplify feedback to R&D
- Find weak points in product design
- Get new ideas for future products



100% testing of all audio devices at the end of the production line provides valuable information for tuning the manufacturing process and for developing new powerful products.

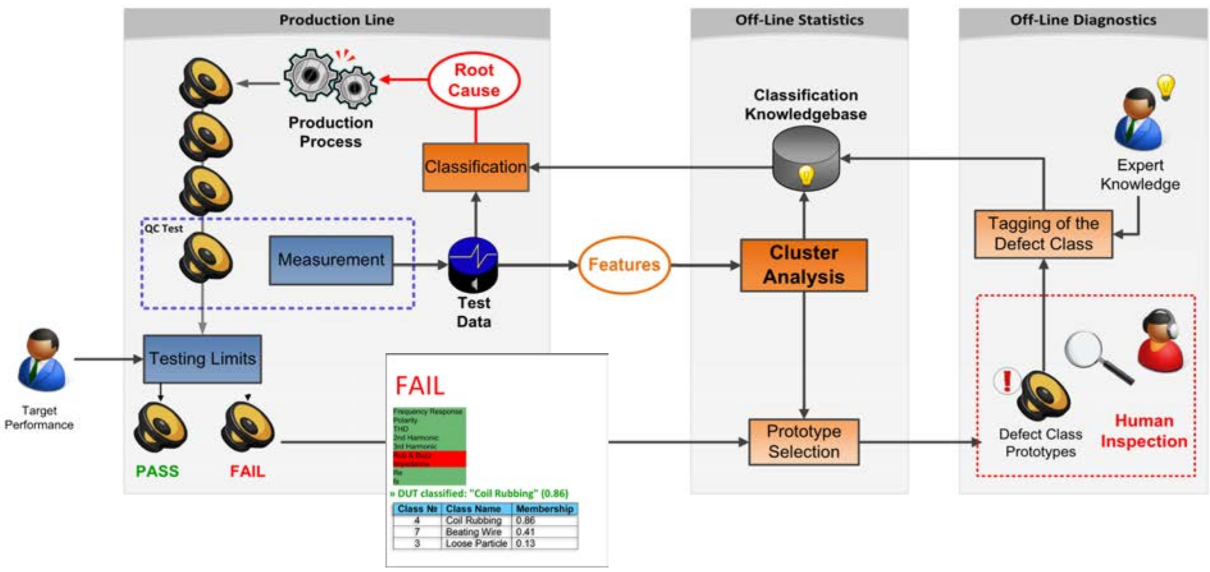
AUTOMATIC DEFECT CLASSIFICATION (ADC)

- Find patterns in your test data
- Root cause analysis
- Meaningful on-line diagnostics
- Accumulate knowledge from manufacturing
- Feedback from production to R&D



The **Automatic Defect Classification (ADC)** is an advanced statistical tool for root cause analysis and classification of KLIPPEL test data. Cluster analysis is applied to large data sets (curve and single value data) in order to unveil systematic patterns of distinct classes. For each detected class, the tool assigns the most representative DUTs („golden“ defect prototypes) as well as characteristic features (parameters or characteristic frequency band), making the class unique. This provides valuable information for linking the objective classes to the physical root cause of typical production problems („Tagging“) by manual inspection or expert knowledge.

The condensed class and diagnostics information is accumulated in your company's internal knowledge bases. By applying this information, new test data may be automatically classified off-line or directly at the production line. In addition to the conventional Pass/Fail verdicts, the resulting classification offers instantaneous root cause diagnostics without human interaction for immediate feedback to the production process and R&D department.



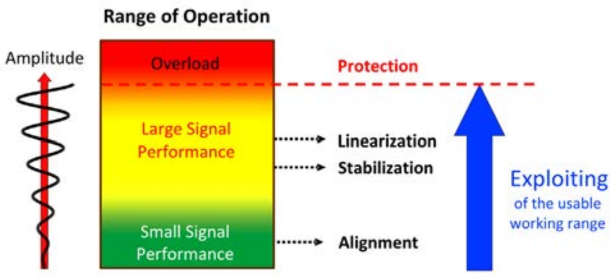
KLIPPEL CONTROLLED SOUND (KCS)

patented KLIPPEL technology

KCS

A NEW PARADIGM

- More sound pressure output
- Active protection against overload
- Cancellation of nonlinear distortion
- Desired linear target performance
- Copes with aging, climate, production variance
- Lowers cost, weight and size

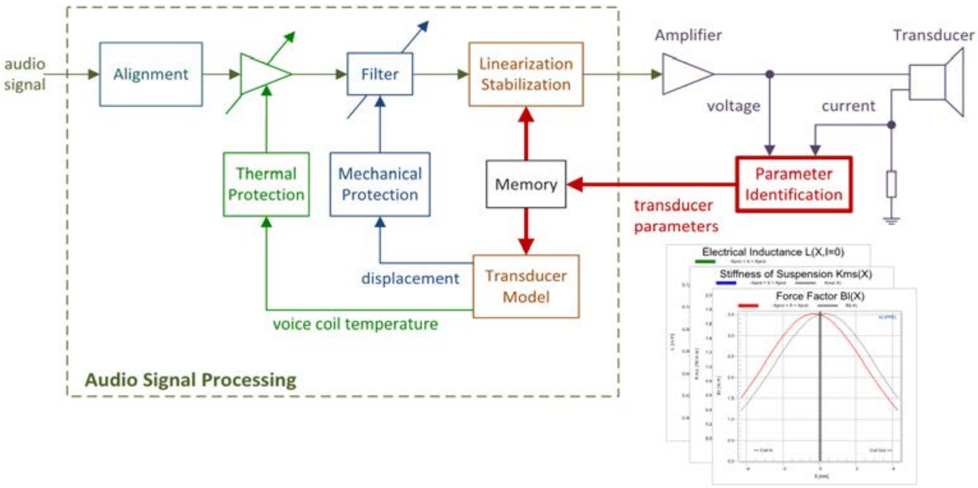


Loudspeakers are highly nonlinear and time-variant systems. Signal distortion, heating, aging, climate and other external influences limit the maximum level and the quality of the reproduced sound. The adaptive nonlinear control system **KCS** can cope with these undesired effects and generate the desired linear behavior over the entire working range.
The adaptive control structure is based on electro-acoustical modeling and combines real-time monitoring of the transducer parameters with active protection against thermal and mechanical overload, nonlinear distortion cancellation, system alignment and stabilization of the voice coil position.
These features lead to an extension of the usable working range to increase bass and sound pressure level or allow transducers to be made smaller, lighter and more cost effective. Additionally, transducer design can focus on increased efficiency by reducing parameter linearity to create a new generation of *Green Speakers* producing more acoustical output and less heat by requiring less energy.

SELF-LEARNING SYSTEM

- Adaptive software solution
- Based on a nonlinear physical model
- Automatic parameter identification
- On-line learning with any audio signal
- Uses the transducer itself as sensor

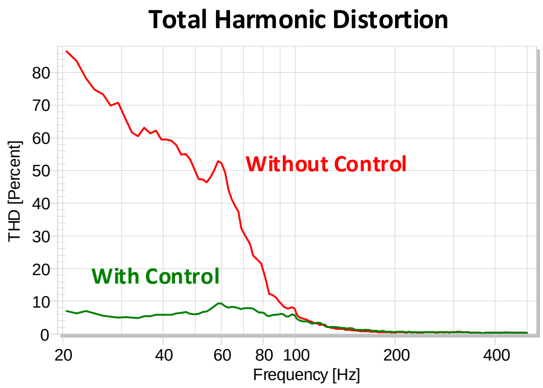
KCS uses the transducer itself as the sensor to identify the instantaneous transducer parameters by monitoring voltage and current at the speaker terminals. The nonlinearities indicate the usable working range of the transducer, eliminating time-consuming tuning by a human expert.



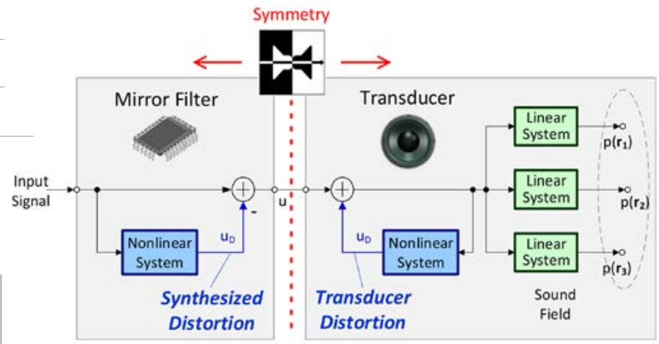
While playing music in on-line mode, **KCS** constantly monitors voltage and current at the speaker terminals to continuously adapt the internal model with time-varying transducer properties, such as variances of mechanical stiffness, voice coil temperature and voice coil position. Based on the identified parameters, the nonlinear transducer model estimates precise state information such as voice coil displacement, which is required by the protection systems.

DISTORTION COMPENSATION

- Linear and nonlinear distortions are reduced
- Constant transducer behavior over lifetime
- Based on a nonlinear physical speaker model
- Improved sound quality
- Enhanced echo and noise cancellation



Nonlinear and time-variant transducer parameters will cause nonlinear and linear distortion in the transducer's output signal. **KCS** uses a nonlinear filter structure, which is a mirror image of the determined transducer model.

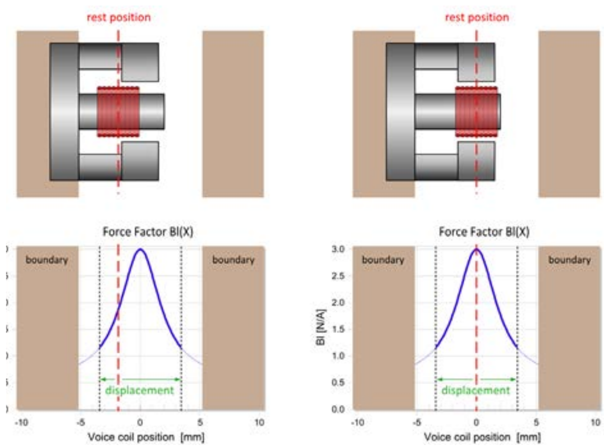


The harmonic and intermodulation distortions synthesized in the mirror filter are subtracted from the input signal before it is fed to the transducer. Thus, the distortions generated by the transducer are compensated, and a linear relationship between the input signal and sound pressure output is established.

ACTIVE STABILIZATION OF VOICE COIL POSITION

- Measurement of voice coil position
- No mechanical sensor required
- Active compensation of coil offset
- Copes with production variance and aging
- Maximum peak-to-peak displacement
- More bass from smaller speakers

For achieving maximum bass level, the peak-to-peak displacement must be maximized. This requires the voice coil being centered between the boundaries. However, the voice coil position is not stable because it depends on soft parts, which show high production variances and will change over time due to temperature, aging and other external influences like air pressure. In addition, transducer nonlinearities can cause dynamic voice coil position shifts due to instable behavior.

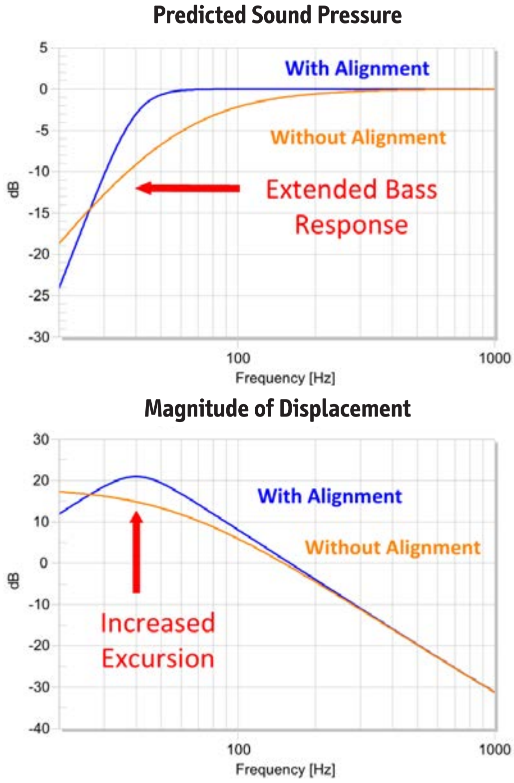


KCS detects the absolute position of the coil without a mechanical sensor by monitoring the input current and identifying an offset in the nonlinear curves. The detected offset can be actively compensated by supplying an appropriate DC voltage to the transducer via a DC-coupled amplifier. This ensures maximum positive and negative voice coil swing, giving maximum bass generated at high efficiency over the lifetime of the speaker.

SYSTEM ALIGNMENT

- Extended bass response
- Optimal transducer-enclosure alignment
- Decoupled enclosure and transducer design

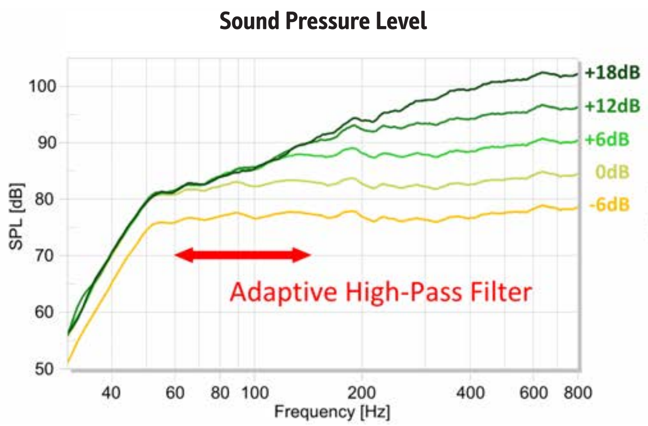
KCS ensures a constant linear transfer behavior between audio input and sound pressure output. By exploiting the information about the transducer and the coupled mechanical and acoustical system (box, vent, passive radiator), the **KCS** automatically equalizes the overall transfer function to a desired alignment (e.g. *Butterworth*) by applying a pre-filter to the input signal.
Matching the transducer to a given enclosure is no longer required as cut-off frequency, Q-factors and other alignment parameters can be adjusted in the software.

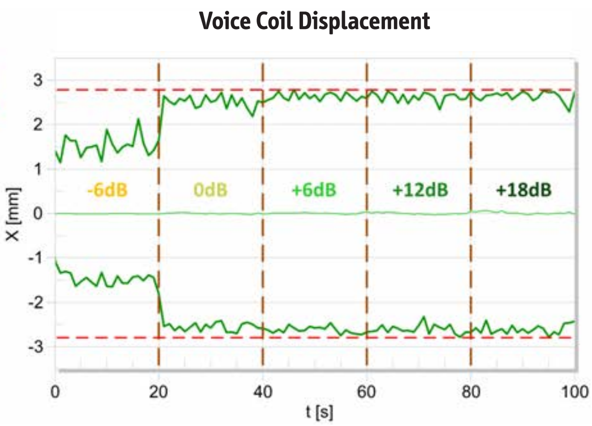


RELIABLE PROTECTION

- Reliable mechanical and thermal protection
- Exploits the entire voice coil swing
- Minimum artifacts
- Zero latency possible

Electro-mechanical transducers need active protection against mechanical overload at high excursion and against thermal overload at high input power to avoid excessive audible distortion or even destruction. The nonlinear and thermal modeling combined with the permanent parameter identification of **KCS** provide a very accurate displacement and voice coil temperature estimation. Thus, the protection system can anticipate critical situations and attenuate signal components to prevent overload.

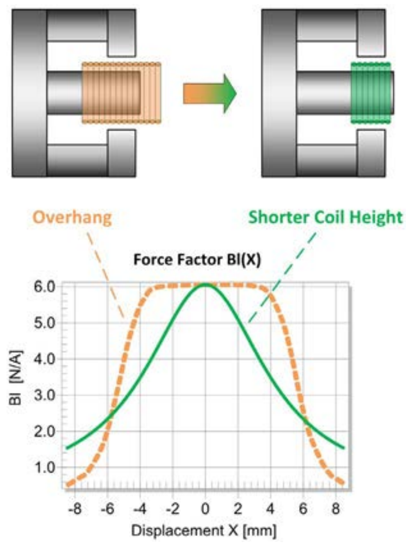




While the thermal protection reduces the level of the entire input signal to reduce the electrical power, the mechanical protection system only attenuates low frequencies where the voice coil excursion is high. Hence the maximum allowed excursion is fully utilized because the audio level can still be increased while only the excursion is restrained. This technique allows the transducer to be reliably protected without latency and avoids artifacts generated by compressors and limiters that impair the perceived sound quality.

GREEN SPEAKER DESIGN

- Increased efficiency and voltage sensitivity
- More bass from smaller speakers
- More SPL output with less heat
- Longer battery life



The unique features provided by **KCS** allow a change of paradigm in passive transducer and system design. Increasing efficiency and voltage sensitivity of the transducer has the highest priority for using available resources such as energy, size, weight, material, manufacturing effort and cost. This leads to *Green Speaker* Design aiming at more output while needing less energy.

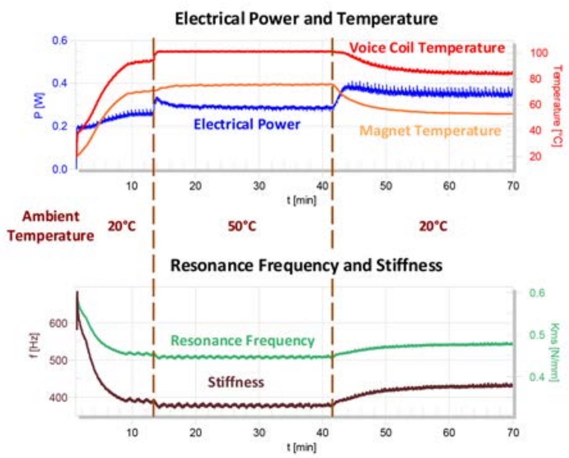
Pass-band efficiency: $$\eta_0 = \frac{P_a}{P_e} = \frac{(Bl)^2}{R_e M_{ms}^2} \frac{\rho_0 S_d^2}{2\pi c}$$

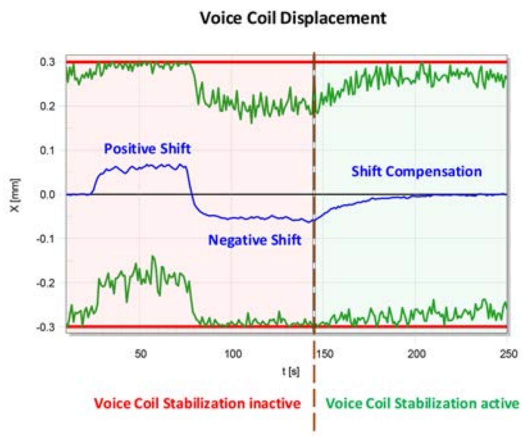
Many design choices dedicated to improving efficiency, such as using very soft suspensions or very nonlinear motors, were not applicable in the past due to the high risk of destruction and increased distortion.
These issues can be solved with adaptive nonlinear control as the increased signal distortion is compensated and the protection system prevents any overload.
For instance, the voice coil height can be significantly reduced without changing the pole plate, magnet and other transducer parts. This significantly increases the efficiency and voltage sensitivity because the resistance R_e and the moving mass M_{ms} are reduced.

ON-LINE DIAGNOSTICS

- Measurement with any audio signal
- In-situ monitoring over lifetime
- Comprehensive information
- Feedback to the design process

KCS extracts valuable information about the instantaneous properties of the transducer in the target application from the voltage and current signals. The parameter and state information reveal the influence of climate, acoustical load and the progress of the natural aging process.





Furthermore, the parameters can give early indications of defects that may eventually lead to a complete breakdown. The diagnostic information provided by **KCS** can be used to safely operate the transducer at reduced amplitudes until the defective transducer is replaced.

ACCESSORIES

- Round off the system
- Out-of-the box solutions
- Get tools approved by KLIPPEL
- Minimize trouble and save time
- Simplify customization

The accuracy of measurements highly depends on the quality of the accessories. KLIPPEL always evaluates microphones, lasers, amplifiers and other third-party products to ensure sufficient performance at affordable prices. Clamping jigs, electrical part, kits and other special tools that are not commonly available are manufactured by KLIPPEL to provide a complete solution, which simplifies getting started.

ELECTRICAL TOOLS

Multiplexers

- Testing multi-channel systems
- Manual and digital control
- BNC version with microphone supply
- XLR In and Out versions for line signals
- Neutrik speakON® version for high power speaker

Speaker Cables

- 4-wire Kelvin connection with separate force and sense wires for accurate electrical measurements
- Neutrik speakON® connector
- Convenient one-hand terminal clip
- Applicable to long distances

Amplifiers

- Various types to cover wide range of applications
- Low impedance stable
- Extended bandwidth for measurement requirements
- Optimal for large signal identification
- Cost-effective solution for power tests

Manual Sweep Controller

- Controller for *QC Manual Sweep*
- Intuitive control of the frequency and voltage (toggle modes)
- Ergonomic handling
- Coarse and fine control of speed

Professional Speakers

- Reference studio monitors
- Sound source for testing micro phones, sound attenuation (ANC) and voice-controlled devices
- Active, DSP controlled, overload protection
- Balanced inputs
- Sturdy aluminum housing
- Protected volume setting
- Flexible mounting options

QR/Bar Code Scanner

- Automatic test selection, execution, and serial number logging in QC
- Logged data can be easily analyzed
- Ergonomic handling

GPIB Controller

- GPIB-USB interface to control third party insruments
- Extend testing capabilities
- Acquire external measurement data

SENSORS AND ACTUATORS

Microphones

- Accurate condenser microphones (CLASS A)
- Cost-effective electret (CLASS B)
- IEPE and phantom power supply
- Types are available for high SPL, high sensitivity, and wide bandwidth

Laser Displacement Sensors

- Cost effective triangulation principle
- Cone vibration (DC to 30 kHz)
- High sensitivity and linearity
- Applicable to all transducers
- Measures distance, geometry
- Easy calibration

Temperature & Humidity Sensors

- Robust for end-of-line testing
- Requires no calibration
- USB interface
- Automatic logging in QC data reveals climate influence

Mouth Simulators

- ITU-T recommendation P51
- Jigs available CCIT P51 and IEEE269
- Internal amplifier
- 100 dB at mouth reference point
- Frequency range 100 Hz ... 16 kHz

Artificial Ears

- Headsets and earphone testing
- IEC 60318-1,2 & 4
- ITU-T Rec. P57 (08/96)
- ISO Standards 4869-4 (60711)
- And other standards

Head & Torso Simulators

- KEMAR representing average human
- Meeting standards by ISO, IEC and ANSI
- Testing ear- & headphones
- Headset & Handset
- 100 % backwards compatibility

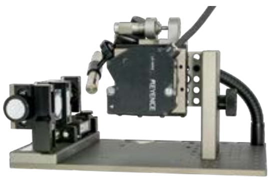
MECHANICAL TOOLS

Transducer Stands

- Applicable to all transducers
- Up to 24 inch woofer
- One-hand operation
- Non-magnetic material
- Microphone and laser
- Laser calibration tool

Turntables

- Precise, rotational control
- Supported by KLIPPEL POL-Module
- Heavy duty, rugged construction
- Operation in any position and orientation
- Programmable velocity, acceleration



Micro-Speaker Fixture

- Dedicated clamping for micro-speakers, tweeters, headphones, etc.
- Non-magnetic material
- Microphone and laser
- Laser calibration tool

Vacuum Measurement Kit

- Non-magnetic material
- Vacuum pump (abs. min. 15 mBar)
- Tap, pressure manometer, tubes
- Electrical transducer connection
- Clear cover for laser scanning

Headset Positioning Systems

- Mobile phones & conventional handsets
- In-situ measurements with head and torso
- Positioning in three planes
- Adjustable force (pinna leakage pressure)
- Supporting ITU-T Recommendations

Headphone Test Fixture

- Various types for different applications (headsets, earphones, hearing aids, hearing-protectors, ear plugs, ear muffs, etc.)
- Equipped with microphones and optional ear and mouth simulators

MEASUREMENT GUIDE

CHARACTERISTICS	MODULES KLIPPEL CONTROLLED SOUND (KCS) *optimized for EoL-testing*	PARTS	TRANSDUCER	SYSTEM
amplitude and phase response (e.g. sensitivity)	TRF, DIS, LPM, *SPL*		X	X
polarity, time delay, latency	TRF, *SPL*, KCS		X	X
3D direct sound output (near- and far-field)	NFS, DSS, POL		X	X
far-field directivity (polar and balloon plots)	POL, NFS, SCN, TRF, DSS		X	X
radiated sound power	DSS, SCN, LPM		X	X
efficiency, voltage sensitivity	LPM, LSIM, LSI, KCS		X	X
maximum SPL_{max} (IEC 60268-21, CTA, ...)	MTON, TBM, LAA, TRF		X	X
electrical input power, voltage, current	LSI, LAA, LSIM, KCS		X	X
electrical impedance	LPM, TRF, LSIM, *IMP*, *MSC*, *BAC*, KCS		X	X
linear lumped parameters (e.g. T/S Parameter)	LPM, MMT, LSI, LSIM, SIM-AUR, SPM, SCN, *IMP*, *TSX*, *MSC*, *BAC*, *LST*, KCS	X	X	X
visco-elastic behavior (creep)	LPM, MMT, LSI, SPM, *IMP+TSX*, KCS	X	X	X
instability (e.g. dynamic DC displacement)	DIS, SIM, TRF, LSI, SIM-AUR, *SPL+DCX*, KCS		X	X
vibration of cone, diaphragm, enclosure	SCN, TRF		X	X
modal analysis (rocking modes, imbalances)	SCN, HMA, RMA	X	X	X
material parameters (E modulus, loss factor)	MPM, HMA	X	X	
nonlinear lumped parameters $Bl(x)$, $K_{ms}(x)$, $L_e(x)$, $L_e(i)$), $R_{ms}(v)$	LSI, SPM, MSPM, SIM, SIM-AUR, BFS, *MSC*, *BAC*, KCS	X	X	X
voice coil offset, stiffness asymmetry	LSI, SIM, SPM, *MSC*, *BAC*, *LST*, KCS	X	X	X
magnetic induction (static B-field)	BFS	X	X	
transient analysis (impulse response, wavelet, decay spectrum)	TRF, TFA, *SPL*		X	X
harmonic distortion (higher-order, THD, THDN)	TRF, DIS, SIM, *SPL*		X	X
HI-2 (weighted harmonics)	DIS, *SPL*		X	X
intermodulation distortion (AM, Doppler)	DIS, SIM		X	X
multi-tone distortion	MTON, LPM, *QC-System*		X	X
incoherence	LAA, *QC-System*		X	X
distortion contribution ($Bl(x)$, $K_{ms}(x)$, $L_e(x)$, $L_e(i)$)	LSI, SIM, SIM-AUR, KCS		X	X
impulsive distortion (Rub & Buzz, air leakage, loose particle, noise)	TRF, DIF-AUR, LAA, *SPL*, *MHT*, *ALD*, *ALS*		X	X
listening, blind AB-test, distortion auralization	SIM-AUR, DIF-AUR, LAA, PLAY, *SPL*, *ALS*		X	X
in-situ monitoring in target application (any audio signal)	LAA, *SPL*, ALS, KCS		X	X
power testing (destructive, accelerated life & environmental)	SPM, LAA, KCS	X	X	X
aging, fatigue of suspension parts	LSI, LAA, SPM, KCS	X	X	X
thermal parameters	LSI, SIM, SIM-AUR, KCS		X	X
temperature and power flow (voice coil, magnet, ...)	LSI, LAA, SIM, SIM-AUR, KCS		X	X
amplitude compression (thermal & nonlinear)	DIS, LSI, SIM,TRF, KCS		X	X
voice coil position and maximum displacement (X_{max})	LSI, DIS, SIM, LAA, *MSC*, *BAC*, *SPL*, KCS		X	X
quality assurance (statistics, root cause analysis, process control)	*STAT*, *YST*, *MSP*, *ADC*, KCS	X	X	X

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