KCS Software Library S72

KLIPPEL Controlled Sound Software Library (KCS Technology 2.0)

FEATURES

- Adaptive, nonlinear transducer control
- DSP-Software solution based on current sensing
- Cancellation of nonlinear distortion
- Active transducer overload protection
- More output at higher sound quality
- Desired linear target performance
- Copes with aging, climate, production variance
- Lowers cost, weight and size of the hardware
- Improves performance for AEC, ANC and 3D sound

DESCRIPTION

The KLIPPEL Controlled Sound Technology (KCS) performs equalization, protection, linearization and stabilization of electro-dynamical transducers mounted in enclosures. The KCS software library comprising the adaptive, nonlinear control algorithms can be integrated into customer DSP architectures.

KCS is using current and voltage sensing to monitor the speaker in real-time. Based on this information, speaker parameters and state variables are identified and the internal model is updated. This information can be used for on-line transducer analysis, diagnostics and the development of new generations of smart loudspeaker systems.

The present specification describes the features available with KCS Technology Revision 2.0.

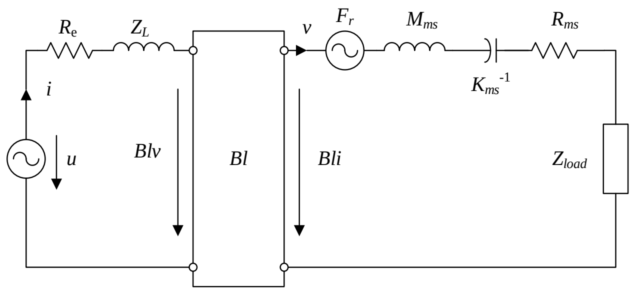
CONTENT

1 Overview

SCOPE	This document explains the concept, features and capabilities of the KCS technology. Not all features are implemented in all KCS software revisions. Separate specifications for specific products with integrated KCS are available.
PROBLEM	The electro-acoustical transducer and the acoustical system is a crucial part in the audio reproduction chain because it contributes significantly to weight, size and cost of the product. Loudspeakers suffer from a low efficiency and produce more heat than sound power output. Nonlinearities in the electrical, mechanical and acoustical domain limit the acoustical output and cause nonlinear distortion as harmonics and intermodulation distortion. The properties of transducers change due to production variances, fatigue, aging and climate influences. These problems reduce the perceived sound quality, the reliability of the product and limit the efficiency of active algorithms for echo and noise cancelation, beam steering and 3D sound reproduction.
IDEA	Digital signal processing can exploit new insights of recent loudspeaker research. By pre-processing the electrical input signal, undesired transducer properties can be compensated and a desired output signal can be "virtually" generated. The most important objectives are: • Generating more output at sufficient sound quality from smaller audio devices • Reliable thermal and mechanical protection of the transducer for any audio input • Improving the efficiency of the transducer to save energy in battery-powered devices and generating less heat • Compensation of production variances and fatigue, aging, climate or external influences over the life time • Higher reliability and robustness of the product in the field • Faster product development and reduced manufacturing cost

1.1 Electro-acoustical Modeling

TRANSDUCER



At low frequencies loudspeakers, headphones, micro-speakers and other actuators can be modeled using an equivalent network comprised of lumped elements.

Nonlinear Elements

The transducer nonlinearities are modeled as lumped parameters which vary with instantaneous state variables such as voice coil displacement x(t), velocity v(t) and input current i(t). The force factor Bl(x), the mechanical stiffness K_{ms}(x), the lossy voice coil inductance represented by the electrical impedance Z_L(x) and the reluctance force F_r(x, i) depend on voice coil displacement x(t). The mechanical resistance R_{ms}(v) is a nonlinear function of velocity v. In larger loudspeaker the electrical impedance Z_L(x, i) varies also with current i(t).

Time-variant Elements

The electrical DC resistance $R_e(T_v)$ depends on the instantaneous voice coil temperature T_v affected by the climate condition and the power dissipated in the transducer. All mechanical elements K_{ms}(t), R_{ms}(t) change with fatigue and aging of the suspension and climate influence. This may also generate an offset x_{off}(t) in the voice coil rest position which shifts the nonlinear curves Bl(x+x_{off}(t)), Kms(x+x_{off}()), Z_L(x+x_{off}(t)) and F_r(x+x_{off}(t)) versus displacement.

Linear Elements

The total moving mass M_{ms} is modelled as a constant parameter which provides linear and time-invariant properties. In most cases the mechanical and acoustical load can be approximated by a linear impedance function Z_{LOAD}(f).

Frequency Dependent Elements

The mechanical load impedance Z_{LOAD}(f) and the electrical impedance Z_L(f) are complex functions of frequency. The stiffness K_{ms}(f) and resistance R_{ms}(f) are modeled as frequency dependent parameters to consider the visco-elastic behavior of the suspension at low frequencies (relaxation and creep).

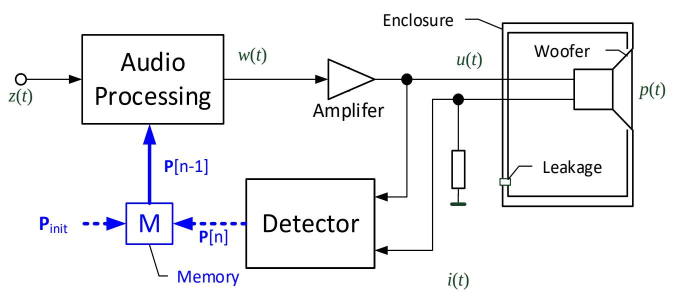
Influence of Production Variances

All parameter values of the lumped element vary over the manufactured units. The most critical parameter variations of the **D**evices **U**nder **C**ontrol (DUC) are caused by the soft parts (diaphragm, spider) affecting the stiffness K_{ms}(DUC) and the offset x_{off}(DUC) of the voice coil rest position. Those production variances can be compensated adaptively by KCS.

ACOUSTICAL LOAD	The load generated by mechanical or acoustical elements coupled to the transducer can be approximated by a load impedance $Z_{load}(f)$. A transducer operated in free air (e.g. dipole loudspeaker) or mounted in an almost sealed enclosure is modeled as a 2nd-order system function having a system resonance f_c and a total quality factor Q_{TC} in KCS. The nonlinearity of the acoustical compliance $C_{AB}(x)$ representing the air enclosed in the box can be merged with the mechanical stiffness $K_{ms}(x)$ if the box is sufficiently sealed. A transducer operated in a cabinet having a larger vent or using a passive radiator is modeled as a 4th-order system function generating a resonance frequency f_b and a quality factor Q_b of the additional resonance. More complex mechanical or acoustical systems like panel-mounted shakers, horn loudspeakers, woofers used in mufflers with active noise cancelation or cavities in the car body can also be approximated by a low order system but this approximation may impair the performance of the system.
MODEL COMPLEXITY	The electro-acoustical model allows prediction and simulation of the nonlinear behavior of the audio device and direct comparison with measured symptoms. A good agreement between model and reality is the basis for an accurate adaptive parameter measurement, good distortion cancelation performance and high protection system reliability. Improving electro-acoustical modeling is an ongoing process. However, more complex models provide not only a higher accuracy but increase also the processing load. Thus, KCS provides some customization of the model complexity to provide the best performance/cost ratio for the particular application.

2 Software Components of KCS

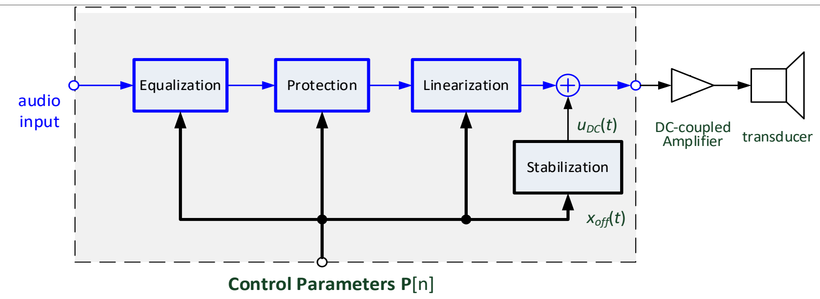
BASIC PRINCIPLE



The objectives defined above cannot be accomplished by using traditional servo control techniques based on negative feedback of a monitored signal but they require more advanced nonlinear control techniques. KCS is based on the following ideas:

- A feed-forward control structure applied to the audio input z(t) generates the loudspeaker input signal w(t) based on a control law derived from physical transducer modeling.
- The control structure uses parameters depending on the linear, nonlinear and thermal properties of the particular loudspeaker type and Device Under Control (DUC). Initial control parameters $\mathbf{P}_{init}$ are stored in a memory (in a host processor) and are used as starting values after booting KCS or when the audio signal does not allow an updating of the parameters.
- A detector provided with voltage u(t) and current i(t) signals sensed at the loudspeaker terminals generates updated control parameters **P**[n].

AUDIO PROCESSING

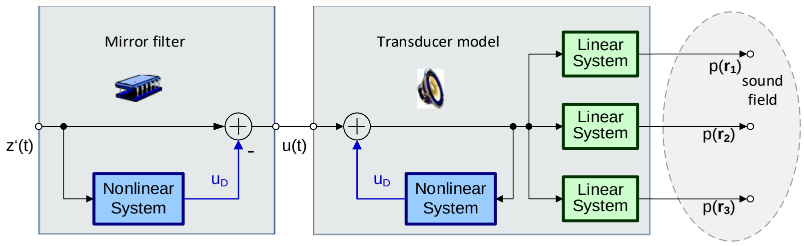


The processing of the audio input signal is performed by a feed-forward structure in the following order:

1. Equalization: Based on the lumped transducer parameters a desired linear overall response is generated by linear filtering (automatic system alignment).
2. Protection: The protection system attenuates the audio signal if a mechanical or thermal overload situation is detected.
3. Linearization: The mirror filter cancels the nonlinear distortion (e.g. THD) in the acoustical output and the DC displacement X_{DC}.
4. Stabilization: The offset x_{off} in the voice coil rest position is compensated by a DC voltage $U_{DC}(t)$ added to the audio signal and transferred via a DC coupled amplifier to the transducer.

2.1 Cancellation of Nonlinear Distortion (Linearization)

TRANSDUCER MODEL

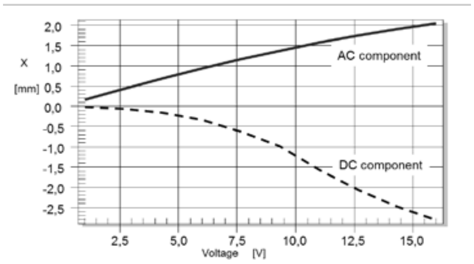


The effect of the dominant nonlinearities such as Bl(x), $Z_L(x,i)$, $K_{ms}(x)$, $R_{ms}(v)$ and other nonlinearities can be summarized in a nonlinear subsystem placed in a feedback loop in the transducer model. The subsystem N generates a nonlinear distortion u_D which adds to the input signal u(t). The total signal at the output of the adder is supplied to a linear system representing electromechanical transduction, radiation and sound propagation to the points $\mathbf{p}(r_i)$ in the sound field. Since the transfer function $\mathbf{H}(f,r_i)$ of the linear system is specific to the listening point r_i, an equalizer applied to the electrical input signal can only realize either the ideal target function at one position or an acceptable compromise over the listening area.

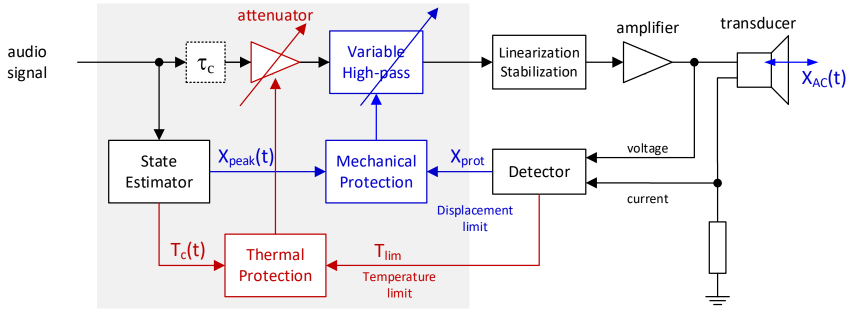
MIRROR FILTER

However, nonlinear distortion generated by the nonlinear system N can be perfectly compensated by synthesizing the same distortion u_D using an identical nonlinear system N and subtracting it from the input z'(t). Both distortion signals cancel out each other in the audio signal path and the undistorted input signal z'(t) is transferred via the different linear systems to the sound pressure output. The inverse nonlinear signal processing comprises the nonlinear system and the adder in a feed-forward structure, which is just the mirror image of the transducer model. This mirror filter exploits the physics of the transducer and causes no additional latency in the transferred signal.

2.2 Stabilization of the Voice Coil Position

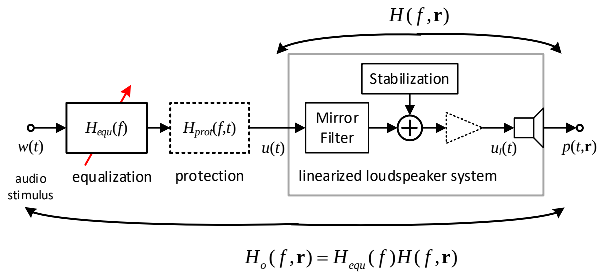
VOICE COIL POSITION	 The absolute voice coil position $X_p(t)= x(t)+X_0+x_{off}(t)$ referenced to the gap geometry is determined based on the following information: • The displacement $x(t)= X_{AC}(t)+X_{DC}(t)$ is generated by the audio signal supplied to the transducer. It consists of a desired and an undesired component. The AC displacement $X_{AC}(t)$ generates the desired sound pressure output p(t). The undesired DC displacement $X_{DC}(t)$ is dynamically generated by a rectification of the audio signal in the loudspeaker nonlinearities and reduces the acoustical output. • The reference rest position X_0 is defined during the nonrecurring measurement of the initial control parameters using the prototype or Golden Reference Device (GRD). • The offset $x_{off}(t)$ is the difference between the actual coil rest position of the particular Device Under Control (DUC) without any input and the reference voice coil rest position X_0. This divergence is caused by production variances, aging, fatigue and external influences such as climate, gravity and load changes. It can be measured by current sensing without any additional sensor.
OFFSET MEASUREMENT	x_{off} is a fast varying parameter that corresponds to a shift in the nonlinear force factor characteristic $Bl(x(t, DUC)+x_{off}(t,DUC))=Bl(x(t),GRD)$ of the Device Under Control (DUC) compared to the Golden Reference Device (GRD) assuming the same motor geometry (coil height, gap depth). This offset can be identified with sufficient accuracy based on the nonlinear distortion found in the current if the force factor nonlinearity shows sufficient decay Bl_{min} < 80% at maximum positive and negative excursion $+X_{max}$ and $-X_{max}$ The measurement of the offset x_{off} will be disabled and the protection system operates the transducer in the *Safe Mode* (see below) if the audio signal does not generate enough displacement to activate the Bl(x)-nonlinearity.
ACTIVE STABILIZATION	The active stabilization system minimizes the offset $x_{off}(t)$ by adding a small DC voltage U_{DC} to the mirror filter output signal, which is transferred to the transducer via a DC-coupled amplifier. This DC voltage U_{DC} shifts the coil to the reference rest position X_0 defined by the initial control parameters measured on the prototype. This ensures maximum peak excursion, high efficiency of the transducer and avoiding high stress and hard limiting in the mechanical suspension. The DC voltage U_{DC} is usually very small because the mechanical suspension is relatively soft at the rest position and the air stiffness in a “sealed” box vanishes at low frequencies due to the barometric vent.

2.3 Overload Protection

TARGETS	Small loudspeakers operated with excessive low frequency equalization and powerful amplification require an active protection system with the following features: • Prevention of critical overloads which can generate excessive distortion or cause permanent damage to the transducer. • Maximization of acoustical output while attenuating the signal components contributing to the overload. • Avoidance of artifacts and audible distortion when the protection system becomes active and attenuates the input signal. • Automatic identification of the permissible limits of the working range. • Consideration of transducer instabilities, heating time constants, ambient temperature and long-term parameter variations. • Minimal latency of the protection system
PRINCIPLE	 These features cannot be fulfilled by just limiting the maximal voltage or electrical input power supplied to the loudspeaker terminals. KCS generates the following state information that are related with a mechanical and thermal overload in the transducer: • anticipated voice coil peak displacement $x_{peak}(t)$ of the linearized and stabilized transducer • increase of voice coil temperatures $T_v(t)$ in Kelvin compared to the cold reference speaker If the peak displacement $x_{peak}(t)$ exceeds the mechanical limit x_{prot}, the low-frequency signal components in the audio signal are attenuated by a high-pass filter with variable cut-off frequency. If the voice coil temperature $T_v(t)$ exceeds the thermal limit T_{prot} the complete audio signal will be attenuated.
ANTICIPATED PEAK DISPLACEMENT	KCS technology calculates the anticipated peak displacement $x_{peak}(t)$ by two steps: 1. The voice coil displacement $x_{AC}(t)$ of the loudspeaker is calculated by using a linear filter in the State Estimator supplied with the audio signal and updated parameters **P**[n]. This modelled displacement $x_{AC}(t)$ corresponds with the real displacement of the transducer because the linearization and stabilization is cancelling the DC displacement x_{DC} and the offset x_{off} in the rest position. 2. The peak value $x_{peak}(t)$ of the following voice coil displacement $x_{AC}(t)$ is anticipated by considering the kinetic and potential energy of the mass-spring-system. That way the audio signal can be attenuated half a period earlier (> 25 ms for a 20 Hz stimulus) before an overload occurs. This improves the accuracy of the protection system and significantly reduces audible artifacts at higher frequencies. Since this technique does not require any delay of the audio path (τ_C=0) this feature is important for zero latency applications. However, in applications where latency is not crucial, a small delay τ_C < 5ms can slow down the activation of the protection system. This makes the activation of the mechanical protection system almost inaudible even in critical situations where a sudden impulse without protection would otherwise generate a high displacement overload (> 12 dB).

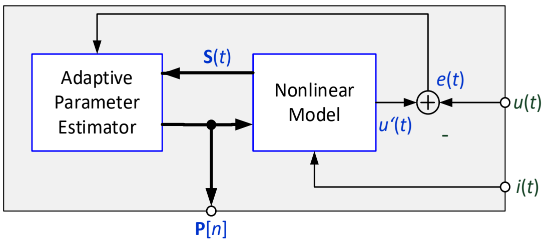
DISPLACE-MENT LIMIT X_{PROT}	The mechanical protection system attenuates the low frequency audio components if the anticipated voice coil peak displacement will exceed the limit value $$X_{prot} = X_{max}\, R_{Xmax}(1\text{-}M_x(t))$$ which depends on the following characteristics: • The maximum displacement X_{max} of the transducer is automatically detected by the initial identification based on mechanical protection criteria. • A user-defined reduction factor R_{Xmax} can reduce X_{max} to consider an external safety margin in the protection limit X_{prot}. The default value of R_{Xmax} is 100% in order to generate the maximum displacement resulting in the best bass performance. However, if the transducer produces some unexpected rub & buzz problems at maximum excursion during the practical evaluation, a reduction factor R_{Xmax} < 100% can be used to reduce the displacement limit X_{prot} and provide a fast software solution without changing the hardware components at the end of the product development. • The internal safety margin $M_{x,M}(t)$ depends on the instantaneous operation mode of KCS. In *Normal Mode* the safety margin is set to $M_{x,M}(t)=0$ to generate the maximum displacement. If KCS switches to *Safe Mode* a small safety margin $M_{x,M}(t)>0$ is activated to cope with parameter uncertainties because the adaptive process is not converged. The different operation modes are described in chapter 2.5 in greater detail.
VOICE COIL TEMPERATURE	The increase of the voice coil temperature $T_c(t)$ is determined by a combination of predictive thermal modeling and electrical measurement of the voice coil resistance $R_E(t)$ based on current sensing. The reference temperature is determined by the ambient temperature during initial parameter identification.

2.4 Automatic Equalization

TASK	 The mirror filter and active coil stabilization generates a linear and time-invariant transfer function $H(f,\mathbf{r})$ between mirror filter input u(t) and sound pressure output p(t,r). Thus, a time-invariant linear filter $H_{equ}(f)$ applied to the audio input w(t) can equalize the overall system to a desired target response $H_o(f,\mathbf{r})$ defined by the customer. After KCS has identified the lumped parameters of the electro-acoustical model this information can be used to provide an automatic alignment of the transfer function at low frequencies. The user can specify a generalized high-pass filter type (e.g. Butterworth) and some meaningful characteristics (order, cut-off frequency, Q-factor) to customize the response to the particular application.

OPTIMAL TARGET RESPONSE	The equalizer replaces the poles of the electro-acoustical system by desired poles and virtually shifts the resonance frequency f_s and quality factor Q_{TS} of the transducer and generates the desired target response $H_o(f,\mathbf{r})$. In reality the extension of the bass response is limited by the bass boost generated in the input voltage and mechanical load in the transducer. The protection system between equalizer and mirror filter will automatically attenuate the low frequency components to avoid a damage or overload of the transducer. KSC can easily support a modification of the target response $H_o(f,\mathbf{r})$ without repeating the initial identification of the electro-acoustical system. It is also possible that KCS automatically modifies the target response $H_o(f,\mathbf{r})$ during in-situ operation when the detector identifies a significant change of the transducer parameters due to aging of the device and external influences. For example, the maximum displacement of a cold woofer operated during winter time (-18° C) is significantly reduced by the temporary increase of the stiffness in a rubber surround. Until air conditioning warms up the suspension system it is recommended to use a target response $H_o(f,\mathbf{r})$ with a higher cut-off frequency.

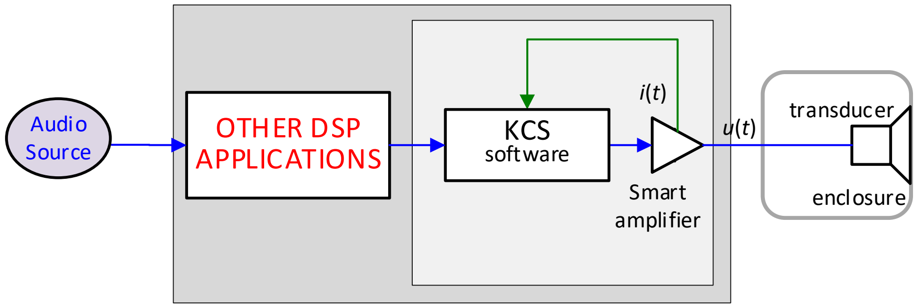
2.5 Parameter Detector

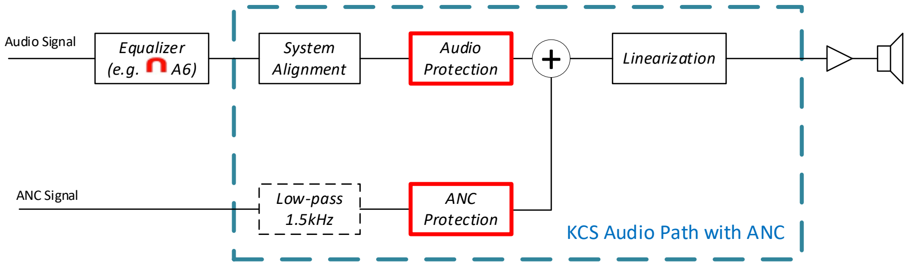
TASKS	The parameter detector • measures transducer and system parameters **P**[n] which depend on the particular device under control (DUC) and vary over time • adaptively models the internal state of the transducer (e.g. temperature and displacement) • requires sensing of the transducer input current (voltage is optional) • copes with any properties of the instantaneous audio signal • detects hardware defects, external influences and other unpredictable conditions and activates special operation modes • improves the reliability of the product and extends the product life • provides valuable diagnostic information for improving future system designs (e.g. fatigue of the suspension)
PRINCIPLE	 Optimum values of the model's free parameters **P**[n-1] are determined for the particular transducer under control based on monitored voltage and current at the transducer terminals. A nonlinear transducer model predicts the terminal voltage u'(t) using the measured input current i(t). An error signal e(t)= u'(t)-u(t) which is the deviation between predicted and measured voltage as well as state variables **S**(t) generated in the model are used in the adaptive parameter estimator to generate a new update of the parameter vector **P**[n]. The adaptive parameter estimation based on the electrical signals at the transducer terminals has several advantages: • Sensors for voltage and current are robust and inexpensive. • No additional sensor wires are required. • Monitoring via loudspeaker cables works over long distances. • Immune against ambient noise.

PERSISTENT EXCITATION	The adaptive parameter detector stays operative for any audio signal and artificial test stimuli as well. In some special cases if the input signal provides non-persistent excitation of the transducer (e.g. single tone input), the updating of slowly varying parameters has to be temporarily paused to avoid a parameter bias.
REQUIRED INPUT INFORMATION	The following information should be provided at the beginning of adaptive learning: • Initial control parameters measured on the golden reference device (e.g. prototype) of the loudspeaker type. • Allowed deviation of the updated parameters from the initial control parameters.
NON-ELECTRICAL OUTPUT	The adaptive modeling based on voltage and current provides electrical information (such as electrical resistance $R_e(t)$ and inductance $L_e(x)$) and valuable information about the mechanical and acoustical system as well. The latter is achieved by exploiting the back EMF generated by the nonlinear force factor Bl(x) and coil velocity v. The mechanical information can be expressed as relative quantities like • Relative displacement $x(t)/X_{max}$ in percent of the maximum displacement X_{max}, • System resonance frequency f_c in Hz and quality factors Q_{TC}, • Relative shapes of the mechanical and acoustical nonlinearities, • Relative deviation of the DUC parameters compared to the GRD parameters. The PC software *KCS Monitor* can transform all relative quantities into absolute quantities using mechanical and acoustical units based on one calibration information (e.g. force factor Bl(x=0) or moving mass M_{ms}).
MALFUNCTION DETECTION	The detector can identify a significant load change, hardware problems or any other malfunctions by checking: 1. the error signal e(t) at each sample → update rate 0.1 ms 2. the state variables x(t), $T_v(t)$, $P_e(t)$, u(t), i(t) → update rate 1-1000ms 3. the transducer parameter $x_{off}(t)$ → 100-5000ms Ref. to 1.: An increase in the error signal shows a significant disagreement between the model and real speaker at the sample rate of the detector. A high update rate is important for detecting a mismatch between the DUC and the initial parameters while starting KCS and activating **Emergency Mode** to prevent an overload of the speaker. Emergency mode is also required if the error signal caused impulsive spikes generated by a loose wire connection or a temporal short circuit of the windings at the pole tips caused by coil rubbing. Ref. to 2.: Based on the measured input current i(t) and the control parameters P[n] the model permanently generates the voice coil displacement x(t) and the coil temperature $T_v(t)$ and electrical input power $P_e(t)$ which describes the load supplied by the audio signal to the transducer. This information is important for activating the mechanical and thermal protection system. Ref. to 3.: The identification of the voice coil offset $x_{off}(t)$ and other linear or nonlinear parameters is an iterative process with a learning time which depends on the stimulus exciting the transducer. After powering up KCS or when the learning cannot be completed the detector activates the **Safe Mode** temporally. After convergence of the deviation between the updated parameters and the initial parameters can be used to assess the fatigue of the suspension and to detect a loudspeaker defect.

OPERATION MODE	The progress of the parameter updating and the deviation of the updated parameters decides about the KCS operation mode. There are 4 defined modes: **Normal Mode**: If the updated parameters and the monitored states of the particular DUC are within permissible tolerances of the GRD, the protection system uses minimum safety margins giving maximum acoustical output. **Safe Mode:** The maximum output of the loudspeaker is reduced by using a larger safety margin (e.g. $M_{x,M}$>0%) as long as the adaptive detector provides no valid parameter and state information of the DUC. This generally happens after powering up KCS or when the audio signal provides no or poor excitation of the loudspeaker. **Emergency Mode**: The audio signal will be significantly attenuated to avoid a damage of the DUC (e.g. operation with wrong initial parameters). **Customer-defined Mode**: This mode considers the particular conditions (e.g. climate or safety requirements in the target application. For example, the reliability of a loud-speaker used in a pedestrian warning system can be increased by defining a *Retirement Mode* where a lower mechanical protection limit (e.g. x_{prot}=70%*X_{max}) slows down the aging and fatigue in the mechanical suspension.

3 Interaction with Other Sound Processing

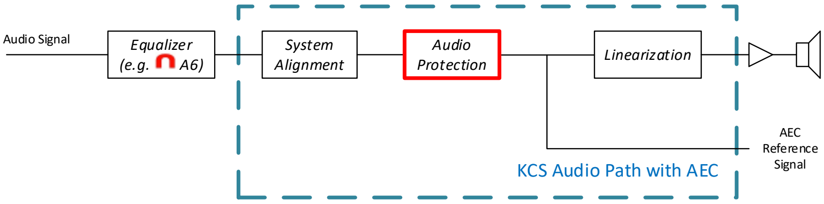
OVERVIEW	 Combining the linearization, stabilization and protection capabilities of KCS with other DSP applications in the audio signal chain that are considering the electro-acoustical trans-ducer as a (linear) black-box system offers substantial benefit. KCS enhances the perfor-mance and simplifies the system design for applications like: • General Sound Enhancement (e. g. crossover, equalizer, compressor, immersive 3D sound decoder such MPEG-H, Ambisonic, …) • Acoustic Echo Cancellation (e. g. in audio-conferencing, hands-free telephony, voice interfacing, wake-word detection, …) • Active Noise Cancellation (e. g. in headphones, ear protectors, engine sound at-tenuation, road noise attenuation, active mufflers, …) • Active Beam Shaping using loudspeaker arrays • Acoustic Contrast Control for rendering different audio content in multiple listing zones

LOW LATENCY PROTECTION	KCS supports ultra-short processing latency (<0.1msec) for applications like active noise control (ANC) where processing latency has to be minimal in order to reach the desired cancellation effect. The overload protection system of KCS typically adds 1-2msec latency on the audio signal to avoid any audible artifacts in the audio reproduction. For applications with shorter latency demands this delay can be reduced to 0ms while still assuring reliable overload protection. One option to process a minimal-latency signal like in ANC is illustrated below:  The low latency signal is provided to KCS via a separate input channel to avoid unnecessary digital filters (EQ / Alignment) adding group delay to the signal. A dedicated overload protection system for this separate input path can be applied with ultra-short latency (0.1ms) while the audio protection system has a larger, configurable latency to minimize audible artifacts. The accurate loudspeaker modeling available inside KCS allows different budgeting strategies to balance the available displacement working range between the different input channels. One budgeting option is to reserve a certain percentage of the movement range for ANC and only use the remaining free working range for audio reproduction. Another option would be to attenuate the ANC signal in case sufficient audio signal is playing. An optional low-pass can be added to suppress higher frequencies in the ANC input. As KCS realizes a linear, time-invariant behavior of the loudspeaker, it is possible and advisable to consider the available displacement working range also inside the external ANC algorithm itself. The instantaneous model displacement can be either fed back from KCS to any other DSP processing block or KCS can also provide filter parameters enabling the precise prediction of the model displacement. That is beneficial for system tuning i.e. to avoid triggering any overload protection activation inside KCS if desired.

SPECIAL REFERENCE SIGNALS

By linearization and adaptive parameter identification KCS can always assure a linear time-invariant transfer function between audio input and any point in the sound field. That is especially beneficial for algorithms relying on a linear system behavior like active echo cancellation (AEC).

These algorithms usually model that transfer function with an adaptive linear model. A dedicated reference signal for these algorithms can be provided by KCS to improve the performance of the echo cancellation.



The ideal reference signal is forked just behind the *Audio Protection* block to avoid disturbing the algorithms by transient changes created in case the protection system is active. The *Linearization* allows for using AEC technology based on linear filters, offering fast conversion rates while time-invariance of the system supports stable conversion.

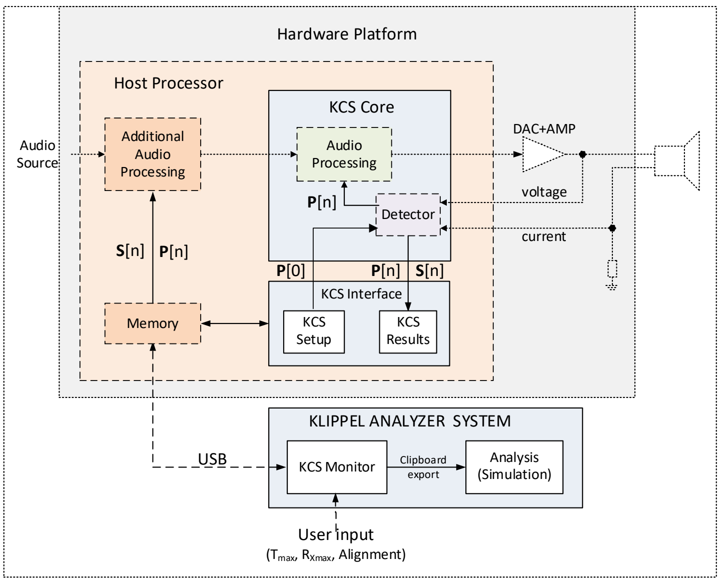
OUTPUT LIMITER

In case the final audio output signal would exceed the peak voltage capabilities of the amplifier, the generation of excessive clipping distortion should be avoided by a limiter. For that purpose, a configurable look-ahead limiter can be applied to attenuate the transient signal peaks. At the same time, the clipping information can be used to activate the protection system for preventing coming signal clipping in order to maximize the audio quality.

It is also possible to provide the limiter activation information as reference signal to other external DSP processing for an optimized overall system reaction on such an event.

4 Software Interfaces

OVERVIEW

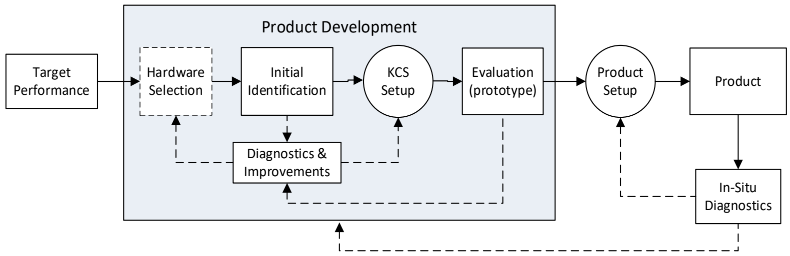


The diagram above shows a Host DSP Processor providing the audio frame for the KCS Core and additional audio processing and the KLIPPEL ANALYZER SYSTEM for operating KCS and loudspeaker diagnostics.

KCS CORE	The KCS Core consists of the audio signal processing and the detector for adaptive identification of the speaker parameters.
KCS AUDIO FRAME	The KCS Audio frame enables the KCS Core to control the transducer and comprises • Audio signal routing • Power amplification (preferred dc coupled) • Current sensing and optional voltage measurement • Storing the KCS setup parameters **P**[0] in a memory • External Interaction (Communication, Pins, …) • Additional processing of the audio signal
ADDITIONAL AUDIO PROCESSING	The KCS generates a virtual transducer with a defined linear transfer response $H_o(f,\mathbf{r})$. This provides an ideal condition for additional processing of the audio input signal for applications like crossovers, sound enhancement, beam steering or echo cancelation. The tuning of the sound enhancement is significantly simplified by showing the instantaneous state and protecting the transducer, hence KCS does not need an expert for parameter adjustment. The tuning is optimal for any DUC and valid over the full product life. The KCS results **S**[n] and **P**[n] can also be used in additional audio processing for example an adaptive sound enhancement which considers the impact of climate and aging to realize a desired balance between different transducers.
KCS INTERFACE	The KCS interface comprises the KCS Setup and a number of KCS Result data sets. The KCS setup **P**[0] consists of initial transducer parameters, margins and other control parameters required for startup. The KCS Results provide access to instantaneous parameters **P**[n] and identified state variables **S**[n] (e.g. displacement) for diagnostics. The data can be inspected via the *KCS Monitor* in *Klippel dB-Lab*.
KCS MONITOR	The software module *KCS Monitor* running as a PC software module in the KLIPPEL Analyzer System is a convenient front-end for a system or transducer engineer to customize KCS Setup parameters **P**[0] and view the KCS Results uploaded permanently or provided by an off-line history file. The state variables **S**[n] and parameter variations **P**[n] are displayed over time. This information is crucial for endurance testing and to learn more about the root cause of aging and loudspeaker defects. *KCS Monitor* generates Thiele/Small (T/S) parameters and nonlinear parameters in the same format as other KLIPPEL software modules (e.g. LARGE SIGNAL IDENTIFICATION (LSI)).
SIMULATION	The transducer and system parameters generated by *KCS Monitor* can be exported via the clipboard to the R&D module Linear Simulation (LSIM) and LARGE SIGNAL SIMULATION (SIM) where the small and large signal behavior (maximal output, distortion, compression...) can be predicted for a sinusoidal stimulus. A curve editor may be used to modify the nonlinear parameters and to investigate alternative design choices.
AURALIZATION	The transducer and system parameters provided by *KCS Monitor* can also be used in the software module SIM-AUR for predicting the sound power output and the internal states of the transducer for any stimulus (e.g. music). The module will also measure the contribution of each nonlinearity to the total distortion. In addition, the SIM-AUR module can be used to enhance or attenuate selected distortion components in the reproduced stimulus and generate a virtual loudspeaker evaluated in listening tests (Auralization).

5 KCS in Product Development

OVERVIEW



The KCS technology accelerates the development of active audio products due to the close connection between hardware and software design, simulation, measurement, diagnostics and on-line evaluation. An active loudspeaker system typically consists of a Digital Signal Processor (DSP), a power amplifier and a transducer mounted in an enclosure.

5.1 Design Considerations

TARGET PERFORMANCE

While considering the production constraints, costs and other requirements of the particular application, the development process starts with the definition of the target performance:

- Geometrical dimensions, weight, cost
- Maximal sound pressure level SPL(**r**) at observation point **r** generated by a typical program material
- Overall target response $H_o(f,\mathbf{r})$ defined by alignment type and cut-off frequency of the high-pass response
- Crossover frequency

HARDWARE SELECTION

A simulation tool (e.g. Klippel LSIM, SIM) can be used to determine the optimum values of the lumped parameter model which are required to fulfill the target performance.

Transducer Characteristics

- Nominal electrical input impedance
- Efficiency and voltage sensitivity of the transducer in the pass-band according to IEC 60268-22 (CDV 2019)
- Stiffness $K_{ms}(x=0)$ of mechanical suspension at the rest position
- Maximum power handling capacity P_{real}
- Effective radiation area S_d
- Target peak displacement X_{target}

Acoustical and Mechanical Load

- Air volume of the enclosure V_B
- Resonance frequency of port or passive radiator (determines the lower cut-off frequency of the overall system)
- Geometrical dimensions of the port or passive radiator

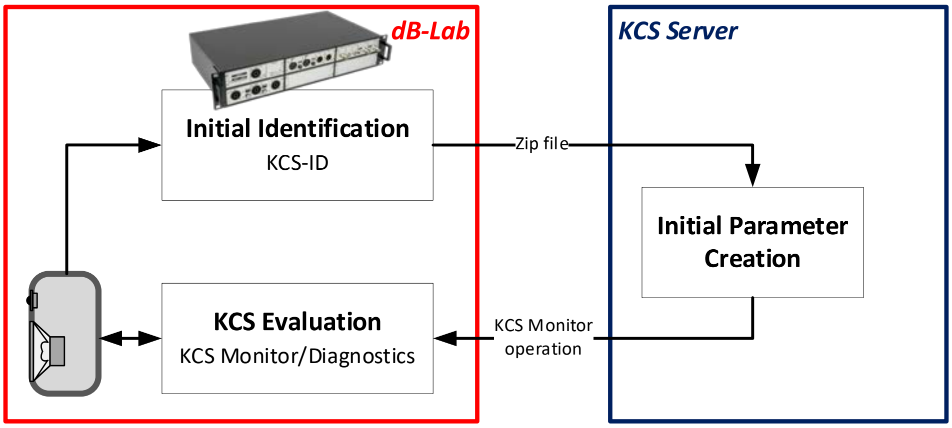
Amplifier Characteristics

- Long-term output power capabilities
- Maximal peak values of voltage and current
- Frequency response (amplitude and phase)

Based on all of the above characteristics, the hardware components are selected and a prototype of the passive loudspeaker system is assembled.

TARGET PEAK DISPLACEMENT X_{TARGET}	The target peak displacement X_{target} should fulfill the following conditions in order to maximize the efficiency of the transducer (*Green Speaker Design* paradigm) and to ensure sufficient reliability of the product. • $x_{target} \approx x_{Bl}$, where x_{Bl} is the force factor limited displacement at a minimum force factor ratio Bl_{min}=50% according to IEC 62458 giving the largest motor efficiency factor Bl^2/R_e, • $x_{target} < x_c$, where x_c is the compliance limited displacement at a minimum compliance ratio C_{min}=50% according to IEC 62458 giving the largest efficiency at low frequencies and • $x_{target} < x_{mech}$, where x_{mech} is the maximum excursion neither generating a damage according to IEC 62458 nor impulsive distortion indicating irregular behavior (hard limiting, rub & buzz) according to IEC 60268-21.

5.2 Initial Identification

OVERVIEW	 The identification of speaker parameters is done using the software module *KCS-ID Parameter Identification*. The signals recorded by the KCS-ID are sent to Klippel's *KCS Server*. This server automatically creates initial KCS data for the particular KCS hardware platform and a *KCS Monitor* operation is provided on the KCS Server's web interface. The *KCS Monitor* operation can connect to any supported KCS hardware platform and a control session can be started.
KCS-ID	The linear, nonlinear and thermal properties of the passive audio system (transducer and enclosure) are measured by the initial identification module *KCS-ID* using the KLIPPEL ANALYZER 3 (KA3). This short measurement uses electrical sensors for voltage and current as well as a microphone to measure the sound pressure in the near field. Optionally, the voice coil displacement is measured if the loudspeaker membrane is accessible by an optical laser sensor. During this measurement the amplitude of the stimulus will be increased to determine the maximum peak displacement X_{max} where at least one of the following conditions is fulfilled: • Peak displacement x_{peak} equals user-defined target value x_{target} • Maximum impulsive distortion ratio according to IEC 60268-21 to consider irregular distortion (rub & buzz) of the transducer is reached
KCS SERVER	The monitored signals collected during the KCS-ID measurement are sent to the KCS Server where the linear, nonlinear and thermal parameters of the loudspeaker model are calculated, the plausibility of the data is checked and the maximum peak displacement X_{max} according IEC standard IEC 62458:2010 limiting the working range $-X_{max} < x < X_{max}$ is generated. Finally, the initial control parameters **P**[0] and other information about the passive audio system are provided on the KCS Server web interface via a KCS Monitor operation.

5.3 Online KCS Operation

TOOLS	The *KCS Monitor* and *KCS Monitor Pro* are software modules of the Klippel framework dB-Lab. They are used for viewing transducer parameters and states, modifying the KCS setup and for online monitoring. The *KCS Monitor Pro* provides a full set of features and supports all KCS hardware platforms while the regular *KCS Monitor* provides a cleaner and easier user interface as it supports less hardware platforms. Its features are sufficient for most users.

VIEWING TRANSDUCER PARAMETERS	The *KCS Monitor* displays the linear T/S parameters and the nonlinearities of the prototype identified in the maximum working range $-X_{max} < x < X_{max}$. The T/S parameters reveal the efficiency and the voltage sensitivity in the passband. The nonlinear curves of $C_{ms}(x)$ and Bl(x) reveal the minimum compliance ratio C_{min} and minimum force factor ratio Bl_{min}. The force factor Bl(x) should be the dominant nonlinearity limiting X_{max} in the transducer following the *Green Speaker Design* paradigm where maximum efficiency reduces the electrical input power P_{real} and the increase of the voice coil temperature ΔT_{v}.
MODIFICATION OF THE CONTROL SETUP	The user interface can be used to modify the limits of the thermal and mechanical protection system, the target values of the system alignment and other parameters of the control setup.
ON-LINE MONITORING	After starting the *KCS Monitor* operation, the default or modified control setup **P**[0] is loaded into the detector and the KCS audio processing is started. The updated control parameters and peak values of the state variables are sampled and collected in the KCS results **P**[n] and **S**[n] and can be transferred back to the *KCS Monitor* for Analysis. The transducer itself is part of the measurement chain while any stimulus (e.g. music) is played and listening tests can be performed to evaluate the perceptual quality of the audio product.

5.4 Evaluation

OVERVIEW	The *KCS Monitor* already reveals information corresponding to the lumped parameter model of the transducer by sensing the electrical input current only. A comprehensive evaluation of the prototype requires an acoustical measurement of the output signal using a microphone placed close to the diaphragm. If the diaphragm is also accessible to an optical laser sensor the KLIPPEL Analyzer Hardware KA3 can be used to measure the voice coil displacement and to compare it with the predicted signal provided by the *KCS Monitor*.
IRREGULAR DISTORTION (RUB&BUZZ)	Only an acoustical measurement can reveal critical impulsive distortions generated by undesired behaviors of mostly mechanical components (e.g. rubbing coils, loose particles) but also loose electrical connections and airborne noise from leaky boxes. There are many causes for irregular behavior occurring in many forms which cannot accurately be modeled and predicted during the design process. This complexity and the impulsive nature of the acoustical symptoms are the main reasons that those undesired distortions cannot be actively compensated by digital signal processing. The hardware design has to provide sufficient robustness of the product to ensure consistent manufacturing and reliability in the target application.
AMPLITUDE FREQUENCY RESPONSE	Measurement of the linear transfer response between the digital input and the sound pressure output in the near field can be used to check the automatic system alignment and KCS protection system in an overload situation. This measurement is also useful to tune an additional equalizer and other components in the sound enhancement.

CANCELATION OF REGULAR DISTORTION	The performance of the active linearization provided by KCS can be checked by measuring the harmonic distortion (e.g. THD), accomplished by using the same sinusoidal chirp as required for the measurement of the higher-order distortion and impulsive distortion according IEC 60268-21. It is recommended to assess the equivalent input harmonic distortion EHID according to IEC 60268-21. Those distortions are directly comparable with the cancelation distortion synthesized in the mirror filter. However, the harmonic distortion measurement based on a sinusoidal stimulus is not a comprehensive test of the nonlinear distortion generated by an audio device because no intermodulation distortion generated by complex audio stimuli (music, speech) are considered. A comprehensive test can be performed by using a multi-tone stimulus in accordance with IEC 60268-21. This stimulus has a similar spectral distribution as typical program material which allows for separating all kinds of nonlinear distortion at the frequencies not excited by the sparse multi-tone stimulus. The reduction of the nonlinear multi-tone distortion is typically between 6-20 dB depending on the frequency components provided by the stimulus and other conditions such as • particularities of the audio product not considered in KCS modeling • production variances (deviation from the prototype) • persistent excitation of the transducer limiting the learning speed of the adaptive parameter identification • KCS features implemented in the particular software version

5.5 Improving the Design

OVERVIEW	The information provided by *KCS Monitor* based on current sensing, systematic listening with typical program material (e.g. double-blind A/B test) as well as comprehensive evaluation using external measurement instruments provide valuable clues for improving the audio device. This can be done by software or hardware tuning.
SOFTWARE REMEDIES	A small modification of the KCS Setup parameters provides a fast solution for many hardware problems. One example: If the critical evaluation of the prototype using a Golden Reference transducer reveals significant impulsive distortion at higher amplitudes this can be fixed by reducing the reduction factor (R_{Xmax} < 100%) in the mechanical protection system and using a lower protection limit (X_{prot}<X_{max}), accepting a compromise in the target performance (reduced maximum bass output) at the same time.
HARDWARE REMEDIES	If the software limits are maxed out major design changes must be done by replacing or redesigning hardware components. Usually this requires more time and may delay the product development.

5.6 Production

PRODUCT KCS SETUP	It is strongly recommended to send the final KCS Setup and test results from the evaluation process applied the prototype to the KCS support team where an expert checks data for proper operation of the KCS and all hardware components (transducer, enclosure, amplifier, sensor, KCS software revision). An **Approved KCS Setup** together with a measurement report will send back to the customer.

MANUFACTURING	The **Approved KCS Setup** will be stored in each unit of the same product type and used as starting point when the controller is powered up in the final product. This simplifies the manufacturing of the active system but requires that the transducer, amplifier and other hardware components match the prototype within defined tolerances. It is also possible to select the most suitable from a number of approved KCS Setups in order to cope with larger acceptable tolerances. During normal long-term operation of the audio product under varying ambient conditions (e.g. climate, load, gravity...) the time varying properties of the transducer are available as KCS Results.
FIELD DIAGNOSTICS	The KCS Results data collected over a large number of units in the target application can be statistically analyzed to assess production variances, long-term stability and robustness of the product. The data can also be used to generate an initial parameter set which is more representative. Furthermore, the diagnostic information provides valuable feedback for future designs.

6 Performance Limits

These limits are generally applicable for the current **KCS Technology Version 2.0** and may be further restricted by amplifier and other hardware limitations. Separate Specifications for the respective product implementations of KCS are available.

6.1 Requirements (Transducer + Enclosure)

Parameters of the Prototype	Symbol	Min	Typ	Max	Unit
Type	electro-dynamical transducer				
DC voice coil resistance	R_e	1	4-8	100	Ω
Transducer resonance frequency	f_s	20		1000	Hz
Transducer total loss factor	Q_{TS}	0.5		6	
Voice coil impedance ratio[1]	$Z_L(f_L)/R_e$	0		8	
Minimum nonlinear force factor ratio[2,3]	Bl_{min}	30	50	100	%
Minimum nonlinear compliance ratio[2]	C_{min}	25		100	%

[1]describing the effect of the lossy voice coil inductance at frequency f_L= 6kHz

[2]Bl_{min} describes the lowest value of Bl(x)/Bl(x=0) in the working rang $-X_{max} < x < X_{target}$ using the maximum peak displacement X_{max} according IEC 62458

[3]The force factor nonlinearity Bl(x) is required for the identification of voice coil offset x_{off}. A small voice coil overhang exploiting the fringe field gives maximum motor efficiency and best condition for offset measurement at low displacement.

PERMISSABLE PARAMETER VARIATION (TRANSDUCER)[1]	Symbol	Min	Typ	Max	Unit
Voice coil offset[2]	x_{off}/X_{max}	-0.25	0	0.25	
Voice coil resistance[3]	R_e/R_e'	0.95	1	1.05	
Resonance frequency[4]	f_s/f_s'	0.5	1	2	
Total loss factor	Q_{TS}/Q_{TS}'	0.5	1	2	
Voice coil impedance	$Z_L(f_L)/ Z_L(f_L)'$	0.95	1	1.05	
Moving mass	M_{MS}/M_{MS}'	0.95	1	1.05	
Force factor maximum	Bl_{max}/Bl_{max}'	0.95	1	1.05	
Stiffness at rest position[4]	K_{MS}/K_{MS}'	0.25	1	4	
Voice coil height	h_{coil}/h_{coil}'	0.97	1	1.03	

[1] The permissible limits are applied to the ratio between selected parameters (X_{off},R_e, f_s, Q_{TS},$Z_L(f_L)$, M_{MS}, Bl_{max}, K_{MS}, h_{coil}) of the unit considering production variances, fatigue and aging during product life and climate influence and the parameters (X_{off}', R_e', f_s', Q_{TS}', $Z_L(f_L)'$, M_{MS}', Bl_{max}', K_{MS}', h_{coil}') of the reference transducer (prototype) measured under standard condition.

[2] Although the variances of the coil offset X_{off} can be compensated by the active stabilization provided by KCS, the required DC voltage limits the maximum parameter variation.

[3] Although the variances of the resistance R_e are compensated by KCS, the variance of the voice coil resistance affects the measured the voice coil temperature ΔT_v and the activation of the thermal protection system.

[4] Although the variances resonance frequency f_S and stiffness K_{MS} are compensated by KCS, a large displacement of the harder suspension at low temperatures may generate a significant stress in the material.

PARAMETERS (ENCLOSURE, PASSIVE RADIATOR)[1]	Symbol	Min	Typ	Max	Unit
2nd-Order mechanical-acoustical system	driver operated in free air, in sealed enclosure, side-fire with rear chamber				
Production variance air volume in Enclosure[2]	V_{AB}/V_{AB}'	0.5	1	10	
4th-Order mechanical-acoustical system	vented box, passive radiator, horn loaded				

Production variance of acoustical resonance (port, passive radiator in box)	f_p/f_p'	0.95	1	1.05	

[1] The permissible limits are applied to the ratio between selected parameters (V_{AB}, f_p) of the unit considering production variances, climate influence and a defect (e.g. broken enclosure) during product life and the parameters (V_{AB}', f_p') of the reference transducer (prototype) measured under standard condition.

[2] Although the adaptive nonlinear control compensates for the variation of the air stiffness related to the enclosure volume but may activate the mechanical protection system if a reduced air stiffness generates high displacement.

6.2 KCS Core

GENERAL					
Supported DSP Platforms	ARM Cortex-M, Sharc, Tensilica (other platforms upon request)				
Supported Audio Frameworks	DSP Concepts Audio Weaver				
ROM memory (instructions + RO-data)[1]	*ROM*	128			kByte
RAM memory (per channel)[1]	*RAM*	32			kByte
MIPS[1] 48kHz audio + 6kHz sensor processing	*MIPS*		150		Core cycles per second
Constraints	• Initial Identification requires Klippel HW • Based on an Approved KCS Setup • Currently only mono channel implementation available				
AUDIO PROCESSING					
Audio Sample Rate	Fs_{audio}	6		48	kHz
Minimal time delay	τ_{min}	$1/Fs_{audio}$			ms
Total time delay (= $\tau_{min} + \tau_{prot}$) [2]	τ_{total}	$1/Fs_{audio}$	2.6	5.1	ms

[1] dependent on platform, audio framework and KCS configuration

[2] Although KCS also supports zero latency operation, it allows to apply an additional time delay τ_{prot} in the mechanical protection system to reduce the deceleration of the coil movement in an overload case

MECHANICAL PROTECTION	**Symbol**	**Min**	**Typ**	**Max**	**Unit**
Monitored state variable	Absolute voice coil position $X_p()$ with respect to the rest position defined by reference speaker (prototype)				
Principle of monitoring	• Modeling based on the digital audio input signal • Compensation of offset X_{off} in voice coil rest position[2] • Compensating DC-displacement caused by transducer nonlinearities				
Look ahead peak detection	Anticipation based on energy balance between potential and kinetic energy (required for zero latency)				
Attenuation element	high-pass with variable cut-off frequency and Q-factor				
Active deceleration of the coil	is required in zero latency applications				
Optional time delay	τ_{prot}	0	2.5	5	ms
Maximum overshoot over permissible working range[3] (KCS without latency τ_{prot}=0 ms)	O_{act}		0	10	%
Harmonic Distortion (steady state)[4]	THD_{SS}		< 0.5	1	%
Harmonic Distortion (attack slope @ τ_{prot} =1ms)[4]	THD_{AS}		1	5	%

[1] In a zero-latency application the moving coil has to be slowed down actively by a deceleration signal to cope with kinetic energy stored in the coil. The active deceleration signal can significantly be reduced if a small delay τ_{prot} is applied to the audio signal, which makes the activation of the protection system in an overload situation inaudible.

[2] The monitoring of the absolute voice coil position is based on the measurement of the offset X_{off} in the voice coil rest position. This measurement requires sufficient peak displacement X_{peak} within an interval (1s) generating a decay of the relative force factor ratio $Bl(X_{peak})/Bl(x=0)< 0.8$ and $Bl(-X_{peak})/Bl(x=0)< 0.8$. If the stimulus generates not enough displacement the measurement of the offset X_{off} is stalled and the internal safety margin $M_{x,M}$ is temporarily increased.

[3] The absolute voice coil position $X_p(t)$ referred to the rest position X_0 as measured on the reference unit (prototype) shall be in the permissible operation range $X_0-X_{max} \leq X_p(t) \leq X_0+X_{max}$ as identified by the initial identification. The overshoot in percent is the maximum difference of $|X_p(t)-X_0|$ divided by the maximum peak displacement X_{max}.

[4] Test with sinusoidal burst at f_c where displacement without protection would be at $10*X_{max}$

THERMAL PROTECTION					
Monitored state variable	Voice coil temperature T_{tv}				
Principle of monitoring	• short term variation is predicted by dissipated input power and thermal modeling • long term variation is calculated based on measured R_e (current sensing)				
Error in voice coil temperature[1]	E_{Tv}		10	30	Kelvin
Dynamic overshoot over thermal protection limit T_{lim}[2]	T_v-T_{lim}		0	10	Kelvin

[1] The error E_{Tv} does not includes the production variance of the voice coil resistance R_e

[2] Test with sinusoidal burst at $2*f_c$ where temperature without protection would be at $4*T_{lim}$

LINEARIZATION					
Nonlinearities considered in active distortion cancelation	displacement varying nonlinearities ($Bl(x)$, $K_{ms}(x)$, $L_e(x)$) velocity varying nonlinearity ($R_{ms}(v)$)				
Reduction of harmonic distortion[1]	$\Delta L_{THD}(f_h)$		> 12		dB
Reduction of multi-tone distortion[2]	$\Delta MTD(f_m)$		> 6		dB
Reduction of DC-displacement[3]	$\Delta L_x(f_x)$		> 12		dB

[1] The reduction $\Delta L_{THD}(f_h)$ is the difference between the total harmonic distortion $L_{THD}(f_h)$ in dB according to IEC 60268-21 measured in the acoustical output of the loudspeaker with and without KCS using a sinusoidal stimulus at an excitation frequency f_h at which KCS generates the largest compensation signal.

[2] A sparse multi-tone stimulus simulating the typical program material (e.g. music) is used to measure the distortion spectrum MTD(f) comprising intermodulation and harmonic distortion components according to IEC 60268-21. The reduction $\Delta MTD(f_m)$ is the difference between the distortion spectrum $MTD(f_m)$ measured in the acoustical output of the loudspeaker with and without KCS at a spectral frequency f_m at which KCS generates the largest spectral component of the compensation signal.

[3] The reduction of $\Delta L_x(f_x)$ is the difference in the DC component in the voice coil displacement of the loudspeaker with and without KCS measured according to IEC 60268-22 (draft) by using a sinusoidal stimulus at frequency f_x at which KCS generates the largest DC compensation signal.

7 Patents

GERMANY	102007005070; 1020120202717; 102014005381.4; 19714199; 4111884.7; 4336608.2; 43340407; 4332804.0
USA	8,078,433; 14/436,222; 14/683,351; 6,058,195; 5,438,625; 6005952; 5.577.126; 5815585; 5,528,695
CHINA	ZL200810092055.4; 201380054458.9; 201510172626.5; 981062849
JAPAN	5364271; 2972708
EUROPE	13786635.6; 0508392A2
TAIWAN	102137485
INDIA	844/MUMNP/2015
GB	2324888
HONG KONG	1020403

Find explanations for symbols at:
http://www.klippel.de/know-how/literature.html
Last updated: September 21, 2021