Loudspeaker Testing at the Production Line

W. Klippel, S. Irrgang, U. Seidel

KLIPPEL GmbH, Dresden, Germany
info@klippel.de
www.klippel.de

ABSTRACT

Quality control in the mass production of transducers and electro-acoustical systems requires an objective technique for reliable selection of defective units. A new technique is presented for detecting defects which produce almost inaudible symptoms during testing but may degrade sound quality in the final application (e.g. loose particles in the gap). Here the regular distortion which is characteristic for good units is modeled and actively compensated in the measured signal of a device under test to reveal symptoms of irregular defects (*meta-hearing* technology). The paper shows ways to perform high-speed measurements close to the physical limits and how to cope with ambient noise in a production environment. Traditional and more advanced techniques for separating passed and failed units are compared and their integration into process control is discussed. Finally, the paper addresses cost effective implementation in a robust hardware, flexibility to customer's needs, simple handling and other practical requirements.

1. INTRODUCTION

Manufacturer of electro-acoustical transducers such as loudspeakers, headphones, micro-speakers and complete electro-acoustical systems have to test the sound quality of the final products. For a long time the human ear has been considered as the most sensitive way for selecting defect units. Indeed an trained tester may detect defects which produce almost inaudible symptoms for an inexperienced listener.

However, human testing has also some drawbacks: The mass production becomes more and more automated and the cycle times are too short for manual handling and listening. Furthermore, testing hundreds of drivers at high SPL puts high physical and mental load on the operator and on its capabilities to stay focused and maintain full performance. Any fatigue will impair the reliability of the end-of-line test.

Available objective measurement systems try to compete with the human tester. Although significant progress has been made, the superiority of the objective approach is not proven so far. There is still a challenge for more sensitive test equipment which exploits the physical and psycho-acoustical mechanisms in greater detail. In the first part of this paper loudspeaker defects are modeled and conclusions for an optimal signal analysis are derived. In this context a new measurement technique will be presented which is in some aspects more sensitive than a human ear.

The paper also discusses the classification problem e.g. the separation of defect units and the diagnosis of the physical cause which is crucial for process control.

In a second part practical requirements are discussed which are important for performing loudspeaker measurements under production condition. Here the differences to the measurement tools used in R&D are worked out.

Finally, conclusions are derived for designing test equipment which is optimal for automated manufacturing.

2. DETECTION OF LOUDSPEAKER DEFECTS

The main goal in end-of-line testing is the detection of defect units. The more is known about the physics of potential defects, the better the measurement can be adjusted to the particular task. Creating a physical model is the best way to summarize available knowledge.

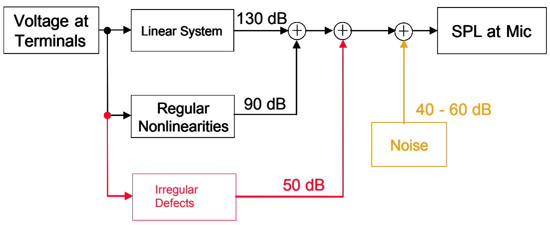

Fig. 1: Modeling the signal flow in a loudspeaker under test.

Applying a test stimulus to a loudspeaker, the sound pressure output p(t) will be not identical with the input u(t). The difference between the time signals may be called signal distortion. As illustrated in Fig. 1 the distortion signal is generated by the following mechanisms:

1. Linear Transfer Characteristic
2. Regular Nonlinearities
3. Irregular Defect
4. Ambient Noise

Defect	Example	Parameter	Symptoms
Linear parameter variation	wrong voice coil length	resistance Re	Amplitude response, Sensitivity, Q-factor
Nonlinear parameter variation	voice coil offset	force factor characteristic Bl(x)	Harmonic distortion, intermodulation distortion
Irregular Defect	loose glue joint	not available	higher-order distortion, crest factor of the distortion signal

Table 1 : Examples for three different loudspeaker defects

2.1. Linear Parameter Defect

The signal distortion found in the small signal domain may be explained by a linear model [1] which uses an amplitude and phase response (or an impulse response) as parameters. At low frequencies the transducer may also be described by an equivalent circuit [2] using a few lumped parameters (R_e, M_{ms}, K_{ms}). The parameters of a prototype or a selected (reference) unit representing the current production may be used as target values. If a parameter of the device under test (DUT) deviates significantly from the target value the DUT may be classified as defective. An example is an increase of the dc-resistance ΔR_e generated by additional windings of the voice coil. This failure will reduce the sensitivity of the loudspeaker and increase the total Q-value of the fundamental resonance as shown in Table 1. Linear parameter variation will only change the amplitude and phase of the fundamental components and can already be detected in the small signal domain.

2.2. Nonlinear Parameter Defect

At higher amplitudes all loudspeakers behave more or less nonlinearly generating signal components which do not exist in the input signal. Most of the dominant nonlinearities are located in motor and suspension which are directly related with the coil-gap configuration and the geometry of the spider and the surround. Equivalent circuits with nonlinear parameters have been developed and allow to predict the distortion at high precision [3]. Some of the loudspeaker nonlinearities are not only imperfection of an ideal loudspeaker design but are intentional made to get a desired large signal behavior. For example progressive stiffness at higher amplitude may protect the voice coil against hitting the back-plate and ensures stability of the motor. A nonlinear motor design leads also to a higher Bl-value at the rest position x=0 and more sensitivity than a linear design. Thus, also prototypes and reference units have "regular" nonlinearities. Comparing the nonlinear parameters of the DUT with the reference parameters may reveal a systematic defect such as a voice coil offset as shown in Table 1.

2.3. Irregular Defects

Finally there are defects which are not acceptable in a loudspeaker passing the end-of-line test. They may be caused by a wide variety of physical causes and are more difficult to model. Table 1 gives as an example a loose glue joint which generates a buzzing sound. Other examples are a rubbing voice coil, a wire hitting the cone or loose particles in the gap. Usually there is also a nonlinearity involved because the distortion signal contains spectral components which are not in the input *u(t)* and depend highly on the amplitude of the stimulus.

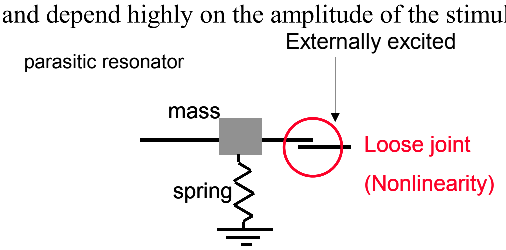

Fig. 2: Mechanical model of a loudspeaker defect (glue problem)

A simple example is a loose glue joint as shown in Fig. 2 which combines the outer edge of the surround with the loudspeaker frame.

The loose part can be modeled by lumped elements (moving mass, spring, losses) forming a resonator with a high Q factor. The loose joint behaves as a nonlinearity which switches the exciting force on and off depending on the state signal (e.g. displacement of the cone). If the joint is open, the mass-spring resonator behaves like an oscillator and starts vibrating at a resonance f_o which is much higher than the exciting frequency f. Applying a sinusoidal excitation signal as shown in Fig. 3 the oscillator generates short bursts at certain time instances spaced periodically (e.g. peak displacement of the cone). The external stimulus initiates and synchronizes the oscillations and provides the energy. Due to the steep characteristic of the nonlinearity and the low losses in the mass-spring-system the coupling is relatively weak.

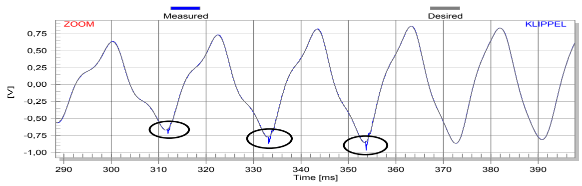

Fig. 3: Impulsive Symptoms of a loudspeaker defect in sound pressure output generated by a sinusoidal stimulus.

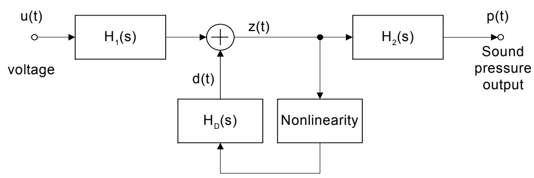

Fig. 4: Signal flow chart for modeling loudspeaker defects

The signal flow chart in Fig. 4 models irregular defects such as voice coil rubbing in the gap, buzzing of loose parts, voice coil hitting the back plate, wire beating the cone and other defects which are deterministic and reproducible. The model comprises three linear subsystems $H_1(s)$, $H_2(s)$ and $H_D(s)$, a nonlinear subsystem $N(z)$ and an adder.

The generation of signal distortion $d(t)$ is modeled by a feedback loop comprising the nonlinearity $N(z)$ and the shaping filter $H_D(s)$. The output of the first linear filter $H_1(s)$ added to the distortion signal $d(t)$ gives the internal state signal $z(t)$ which activates the nonlinear feedback loop. The state signal $z(t)$ may be the voice coil displacement $x(t)$, the velocity of the wire or any other state variable which is directly related with the loudspeaker defect. The linear filter $H_2(s)$ describes the transfer of the internal state signal $z(t)$ to the sound pressure output $p(t)$.

Considering a simple glue problem the linear system $H_D(s)$ represents the spring-mass system represented by a 2nd-order system with a very distinct resonance (high Q-factor). The feed-back loop is the basis for generating free oscillations at f_o with $f_o >> f$. The nonlinearity switches the oscillations on and off and limits the maximal amplitude of the oscillations. The product of the linear systems $H_1(s)$ and $H_2(s)$ in the feed-forward branch corresponds with the overall transfer function of the loudspeaker in the small signal domain.

The only class of defects which can not be modeled by Fig. 4 are defects with a strong accidental nature such as loose particles in the gap.

2.4. Measurement of Parameters

The system function $H(s)=Y(s)/X(s)$ describes the relationship between input and output for any stimulus as long as the loudspeaker behaves linear. Also the lumped parameters of the loudspeaker's equivalent circuit should be independent of the stimulus and the measurement condition. Linear parameters such as R_e, Bl, K_{ms}, M_{ms}, f_s or Q_{ts} are valid in the small signal domain only. At higher amplitudes the "linear parameters" vary significantly with the stimulus and may be interpreted as "effective values" only. Nonlinear characteristics of force factor $Bl(x)$, stiffness $K_{ms}(x)$ and inductance $L(x,i)$ depending on displacement x and current i are required to describe the large signal behavior more adequately. The nonlinear parameters are easy to interpret and reveal a voice coil offset and suspension problems. Currently the nonlinear parameters are measured on randomly selected samples only.

2.5. Measurement of Symptoms

Loudspeaker with irregular defects such as rub & buzz are detected by assessing the large signal behavior. Here the DUT is excited by a special stimulus (e.g. sinusoidal sweep) and the loudspeaker output (sound pressure in the near field) is subject to a signal analysis to find characteristic clues for a defect.

Some defects (e.g. loose particle) produce symptoms which have extremely low energy but are still detectable by a trained ear. Fig. 1 also shows the sound pressure level of the different signal components measured in the near field of a woofer. Whereas the fundamental component may exceed 130 dB, the symptoms of irregular defects are about 50 dB close the ambient noise level of about 40 – 60 dB. The regular nonlinearities generate symptoms at much higher SPL which can mask the symptoms of the irregular defects. Thus, the measurement of total harmonic distortion (THD) is a good indicator for motor and suspension problems but less useful for detecting rub and buzz problems.

2.5.1. Critical Stimulus

For the measurement of linear defects and regular nonlinearities a wide variety of test signals may be used. Sufficient excitation can be provided by an ordinary audio signal (music) or (artificial) speech preferred for drivers used in hearing aids and telecommunication. Calculation of power spectra of the input and output signal and the cross spectrum gives the amplitude response. From the same data an additional measure may be calculated which describes the deviation of the DUT from a linear system (incoherence). This technique can not distinguish between nonlinear distortion and ambient noise corrupting the measurement.

A synthetic stimulus with audio-like properties is the multi-tone complex which sounds like an organ tone. Usually the stimulus is periodic and has the same length as the FFT used for signal analysis. Thus, there is no windowing required and fundamental components can be separated from the distortion components with high resolution. An additional measurement without excitation may be used to separate noise and distortion.

Testing with a multi-tone stimulus is very fast and gives a very good signal to noise ratio (SNR) because the excitation can just be focused on selected frequencies which are usually logarithmically distributed over the frequency range of interest.
Neither an ordinary audio signal, a multi-tone complex nor any other signal with high spectral and temporal complexity is a critical stimulus for detecting irregular defects. The most sensitive stimulus for human and automatic testing is a sinusoidal test signal (stationary tone or sweep) for reasons which are discussed in greater detail in the next chapter.

2.6. State Variables

Monitoring of relevant state variables is crucial for the detection of loudspeaker defects. Measurement of voltage and current at the loudspeaker terminals shows the electrical input impedance and is the basis for calculating linear and nonlinear loudspeaker parameters. The measurement of the voice coil displacement would be required for a direct measurement of the mechanical parameters (*Bl, M_{ms}*) in absolute units (*N/A* and *gram,* respectively). However, deviations of the mechanical parameters can also be detected by derived parameters (resonance frequency, *Q*-factor, sensitivity) which can more conveniently be measured in the electrical or acoustical domain. Most of the irregular defects are generated in the mechanical domain and are not detectable by monitoring electrical signals. For the detection of irregular defects the sound pressure signal measured in the near field provides symptoms at the best signal to noise ratio.

2.7. Symptoms of Irregular Defects

An appropriate analysis of the sound pressure signal is required to separate meaningful symptoms of the irregular defects from fundamental component and the regular distortion. Different techniques have been developed so far and are now discussed in greater detail:

2.7.1. Frequency Domain Analysis

Exciting a rubbing voice coil by a sinusoidal stimulus may give a sound pressure spectrum similar to the schematic representation in Fig. 5.

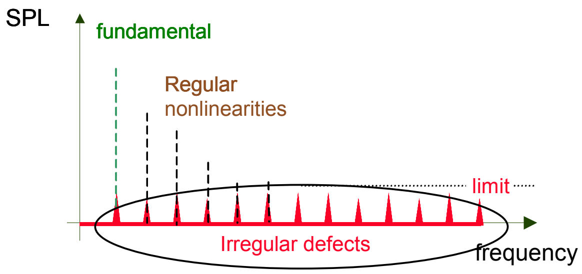

Fig. 5: Separation of irregular distortion in the frequency domain

The fundamental component gives the highest SPL level followed by harmonic distortion components decreasing in amplitude with rising order. The 2nd- or 3rd-order harmonic components are usually the highest components indicating an asymmetrical or symmetrical curve shape of the regular nonlinearities located in motor or suspension. Above 7th … 20th order the amplitude of the higher-order harmonics stagnates at a level of –60 … - 90 dB below the fundamental. Using a FFT analysis which is synchronous to the excitation tone a second interesting observation can be made: Whereas the distortion generated by the regular nonlinearities are located exactly at multiples nf_0 of the fundamental frequency f_0, the distortion are distributed (smeared) around nf_0 (for n > 10). Both observations may be explained by the defect model developed before. The irregular defects behaves like an oscillator which is loosely coupled to fundamental and only "synchronized" by the stimulus. The oscillator generates short bursts or dirac impulses on a regular basis (e.g. at a certain position of the coil) which give constant contribution to all higher-order harmonics. The contribution to the low-order harmonics (n < 10) is completely masked by the contribution from the regular nonlinearities which may be up to 40 dB higher. Since the regular distortion vanishes rapidly with rising order, symptoms of the irregular defects might become dominant at 7th but usually at a much higher-order. Unfortunately each higher-order harmonic has an extremely small amplitude which is close to the measurement noise. Due to the low signal-to-noise ratio the amplitude of the higher-order harmonics is a very bad basis for the pass/fail classification and makes the adjustment of the limits difficult (as illustrated in Fig. 5). More advanced techniques have been developed to exploit the available symptoms and increase the sensitivity of the measurement equipment.

2.7.2. Time domain Analysis

The signal-to-noise ratio of symptoms can not be improved by summing up the amplitude values of the higher-order harmonics. However, the phase of the higher-order harmonics gives valuable information about the waveform of the distortion signal *d(t)*. Clearly, the phase information of 200 separate harmonic components is neither interpretable by an engineer nor applicable to an automatic pass/fail classification. A more elegant solution is to continue the analysis in the time domain [4] as illustrated in Fig. 6.

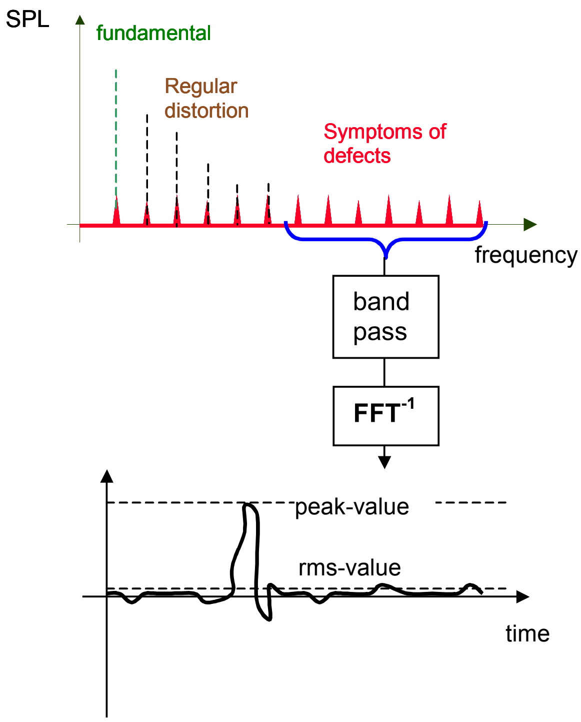

Fig. 6: Separation of symptoms of the loudspeaker defect by band-pass filtering and time domain analysis

This can be realized by exploiting all of the higher-order harmonics in amplitude and phase while setting all of the other components to zero and performing an inverse FFT. Alternatively, a high-pass filter or band-pass at fixed frequencies or with tracking capabilities may be used. More important is the following analysis in the time domain. Calculating the *rms*-value with a relatively high time constant (more than the period of the fundamental) would give a mean value which does not exploit the fine structure of distortion waveform. An interesting alternative is the peak value of the distortion signal which might be much higher than the rms-signal if the irregular defect generates short bursts, clicks or other transients. The crest factor of the higher-order harmonics which is the ratio between peak and bottom value is a very simple but informative measure which summarizes the phase information. Noise, the regular nonlinearities and other stationary defects (e.g. permanent rubbing of the voice coil in the gap) generates a low crest factor below 10 dB. However, most irregular defects generate more transient distortion with a much higher crest factor [5].

2.7.3. Masking of Irregular Defects

The distortion of the regular nonlinearities highly depend on the amplitude of the stimulus during testing. Doubling the excitation voltage will increase the low-order distortion by more than 6 dB as illustrated in Fig. 7.

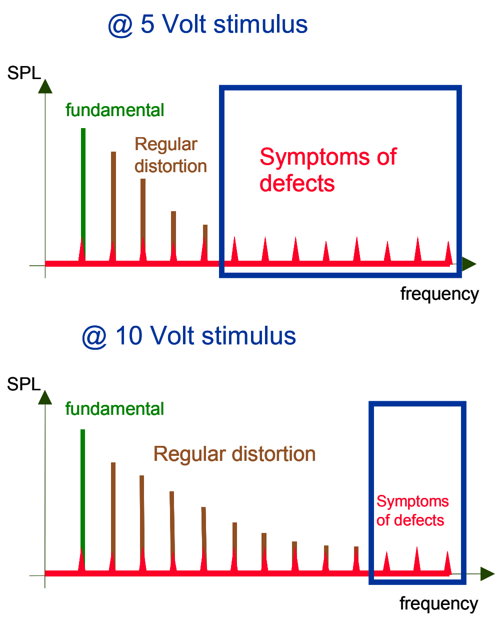

Fig. 7: Symptoms of irregular defects are masked by regular nonlinearities at higher amplitudes

After initiating irregular defects the generated symptoms are almost independent of the excitation level. Thus the regular nonlinearities become more and more dominant and mask the symptoms of the irregular defects with rising level. The masking effect explains why both a human ear and a conventional testing system may find reliable symptoms at medium amplitudes but can not detect the defect at a higher voltage which might be required by the specification.

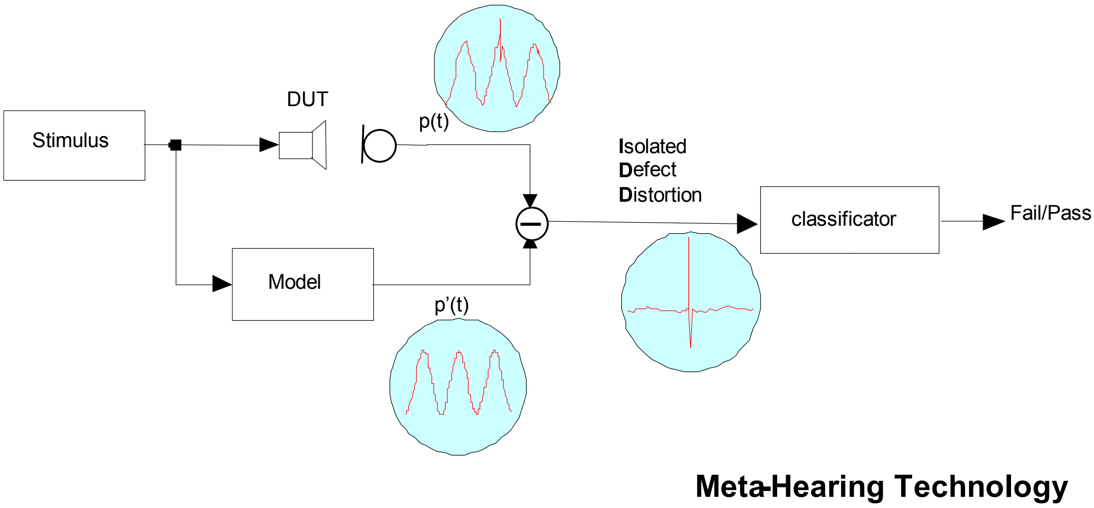

Fig. 8 Detection irregular defects (e.g. rub and buzz) by active compensation of regular distortion

2.7.4. Active Compensation of Regular Distortion

The masking effect can be removed because the nonlinearities of the motor and suspension are "regular" in the sense that their symptoms are reproducible and predictable by a physical model. Fig. 8 illustrates the "active" isolation of the symptoms by using a model which is also fed by the same stimulus as the DUT. The model estimates the sound pressure output *p'(t)* of a "good" unit considering also the regular nonlinearities in the motor and suspension. The model may be updated permanently the properties of "good" units and corrects adaptively for minor parameter variation and changed ambient conditions. The predicted output *p'(t)* subtracted from the measured output *p(t)* give the isolated defect distortion which is subject of a regular pass/fail classification. This technique (meta-hearing technology [6] which is comparable to active noise cancellation can detect symptoms which are inaudible during testing but indicate a defect which is not acceptable in final application (e.g. loose particles).

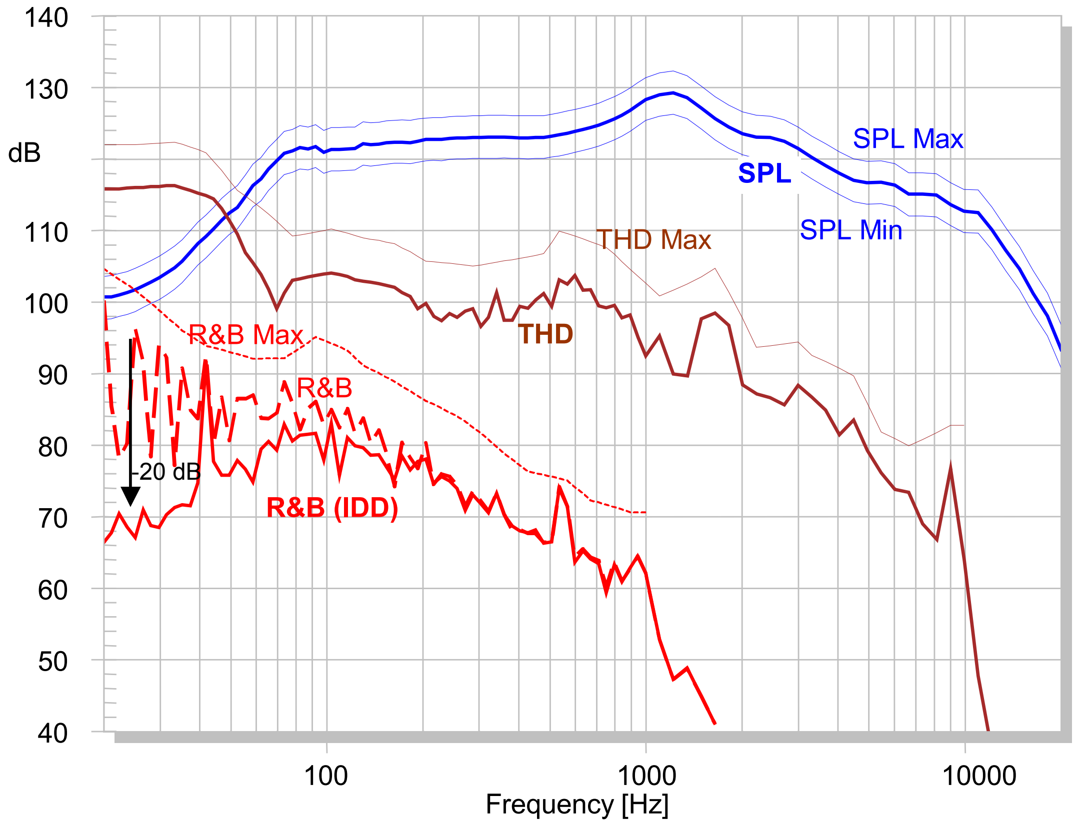

Fig. 9: Detecting one grain of fine salt on a cone

The high sensitivity of the new technology can be illustrated on one grain of fine salt put on the cone of a loudspeaker operated in horizontal position. Applying a sinusoidal tone sweeping through the audio band downwards in 200 ms the piece of salt is bouncing on the cone accidentally like a loose particle in the gap.

Fig. 9 shows the results of such a measurement. The amplitude response of the fundamental (SPL) appears at about 130 dB. The total harmonic distortion (THD) are also high at low frequencies where the voice coil displacement is significant. The peak value of the higher-order distortion (R&B) can not reveal the impulsive distortion of the bouncing salt grain and stays below the limit value (R&B Max).

However, the isolated defect distortion (IDD), where the regular distortion is actively compensated by more than 20 dB, shows clearly the symptom of the loose particle at 40 Hz. This makes it possible to reduce the limit (R&B MAX) an to detect the defect reliably.

2.8. Pass/Fail Decision

The main target in quality control is to separate "good" and "bad" units. This is a classification problem which can be solved by defining limit values. Unfortunately, not all of the limit values can be defined independent of the production process such as the permissible voice coil resistance. Other limits have to be determined under the special measurement conditions found at the end-of-line. This is not only the acoustical environment which is less ideal than an anechoic chamber, but also the short production cycle which gives not enough time to wait for a steady-state response.

2.8.1. Calculation of Limits

Typically a set of faultless units is measured under production line conditions. Prototypes may be included to make comparisons with R&D measurements possible. The mean value over all reference units is taken and the limits are calculated by applying a (frequency dependent) shift to the mean. The shift can be either specified by the user or set to be proportional to the standard deviation of the data at the particular frequency.

2.8.2. Adaptive Adjustment of Limits

Usually constant limit values are used for the classification. However, variable limits which change in a clearly defined range have the benefit to consider normal parameter variation found in good units. For example the break-up modes of the cone and cancellation effects generate very distinct resonance peaks and narrow dips at particular frequency which vary from DUT to DUT. While the general shape of the amplitude is characteristic the particular position of the resonances is not. To use also narrow limits values for those varying parameters the classificator has to calculate (shift and stretch) the limits adaptively.

2.8.3. Calibration of Limits

One major problem in loudspeaker testing is that loudspeaker's properties vary with the ambient conditions. The resonance frequency might change by more than 10 percent during the normal variation of temperature and humidity at the production site. The problem can be solved by readjusting the limits as ambient conditions change. This procedure can be simplified by using "golden" DUT which represents the "mean" of the reference units in a lest-square sense. If the ambient conditions change a measurement of the golden DUT is performed. The data is compared to the original data of the DUT and the limits are adjusted according to the deviation.

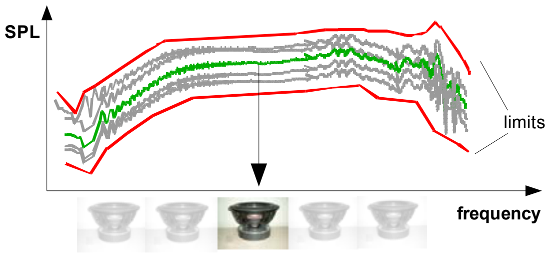

Fig. 10: Selecting representative units (golden DUTs) for the calibration of the limits

2.8.4. Grading

Sometimes not only a pass/fail decision is required but rather a classification of the units into several quality levels (grades). Multiple sets of limits can be used where each set corresponds to a particular grade.

2.8.5. Diagnosis

An experienced human tester can not only hear that a particular DUT sounds differently than a reference unit but can also provide clues on the physical or technical cause of the problem. This information is very important especially if the defect has a systematic cause somewhere in the production process.
The linear and nonlinear parameters are easy to interpret and the link to the physical causes is obvious as shown in Table 1 on the example of a voice coil offset and additional windings.

The relationship between the symptoms of irregular defects and the physical causes is much more complex. An automatic classification of the causes (rubbing coil, missing glue at the surround, …) remains a challenge for an automatic test system.

2.9. Process Control

A QC system should also monitor the process stability and shall provide warnings before systematic problems in manufacturing cause defect units. This can be achieved by statistical indices such as:

<u>Process Capability Index (*Cpk*):</u>

$$CpK = \min\left(\frac{L_{up}-E_c}{3\sigma_c}, \frac{E_c-L_{low}}{3\sigma_c}\right)$$

using

L_{up} upper specified limit

L_{low} lower specified limit

C number of DUTs investigated

E_C expected value of test results X_i within the last C measured DUTs

$$E_C = \frac{1}{c}\sum_{i=1}^{c} X_i$$

σ_c Standard deviation of test results X_i within the last C measured DUTs

$$\sigma_C = \sqrt{E_C\left[\left(X - E_C(X)\right)^2\right]}$$

Cpk is defined as the distance between the average of a fixed number *c* of samples (short term) and the closest limit divided by half of the process width. The process width is usually six times the standard variation.

Process Performance Index (*Ppk*):

$$PpK = \min\left(\frac{L_{up} - E_P}{3\sigma_P}, \frac{E_P - L_{low}}{3\sigma_P}\right)$$

using

P number of DUTs in total production

E_P expected value of all test results X_i

$$E_P = \frac{1}{P}\sum_{i=1}^{P} X_i$$

σ_P Standard deviation of all test results X_i

$$\sigma_P = \sqrt{E_P\left[\left(X - E_P(X)\right)^2\right]}$$

Ppk is the overall process performance similar defined as *Cpk* but based on the total process variation of a batch (all samples used) where *Cpk* only takes into account a fixed number of samples.

To analyze trends within production only passed tests should be used to assess the variance of the production process not disturbed by failed stray units. Analyzing *Cpk* provides early information about shifts in production even before a limit is exceeded. Thus readjustment of the production and preventing failed units from being manufactures becomes possible as illustrated in Fig. 11.

Correlating these indices with monitored temperature and humidity will provide clues to find the causes for the process variation.

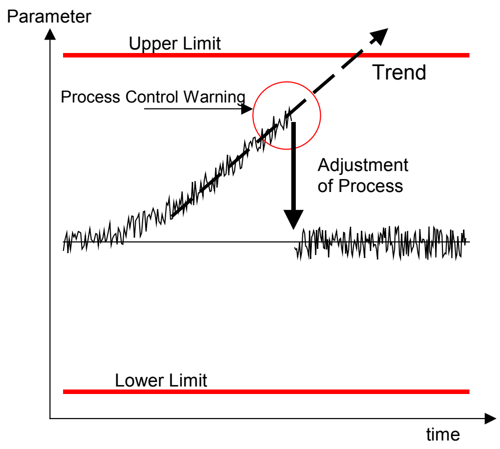

Fig. 11: Applying process statistic to find trends and readjust the production process

3. PRACTICAL REQUIREMENTS

3.1. High speed performance

Modern assembly lines have very short production cycles. So measurements must be performed at physical limits.

3.1.1. Ultra Short Stimuli

The time limitations require stimuli that excite the DUT in an optimal way. Often there is not enough time to wait for a steady-state response but it is crucial to obtain at least a sufficient signal-to-noise ratio. A multi-tone stimulus has the advantage of concentrating the energy at selected frequencies. Unfortunately, a sparse multi-tone spectrum can not ensure an excitation of irregular defects having a narrow band resonance. A sinusoidal sweep with continuously changing frequency excites all frequencies and avoids this problem. The high Q-value of a irregular defect requires also an certain excitation time. Since the total measurement time is restricted, a chirp signal which ahs a lower sweep speed in the critical frequency range and a higher sweep speed at others gives the best excitation for a limited test time. Furthermore the amplitude of the sweep can be varied in order to make the stimulus more critical.

3.1.2. Parallel Signal Analysis

The test can be speeded up by starting the signal analysis while the DUT is still excited. This way the time needed for post-processing can be significantly reduced.

3.1.3. Multi-Channel Measurement

The time required to place the units on the test bench contributes considerably to the overall test time. With a two channel system almost the whole cycle time can be used for excitation. (see Fig. 12). While a first DUT is measured at one channel a second DUT is connected to the other channel. After completing the first measurement the system should automatically detect the second DUT and start the next measurement immediately.

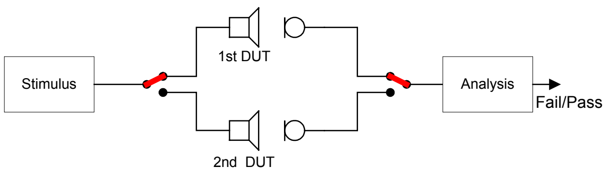

Fig. 12: Speeding up the measurements by using a multi-channel measurement chain

3.2. Production Environment

Robust measurement techniques, well controlled conditions and reliable as well as accurate hardware are crucial for repeatable and reproducible testing.

3.2.1. Acoustical Requirements

The measurement conditions have a similar influence on the final result as the measurement techniques discussed in section 2, Furthermore, the more sensitive the measurement is the more consistent the test conditions have to be.

Box Enclosure: Loudspeaker drivers are small enough to be tested in closed cavities. This reduces the production noise at the test microphone and ensures reproducible acoustical conditions. Reflections and damping are almost identical for all tests which is difficult to guarantee in free air.

Near-field measurement: To get a high SNR the acoustical output should be measured in the near-field of the DUT. This is mandatory for testing in free air.

3.2.2. Production noise immunity

The noise level occurring under production conditions can easily mask the symptoms of a defect DUT. In some cases the noise is so high that even the box enclosure and an additional isolated test cabin give not sufficient suppression. Thus a technique is required to detect measurements corrupted by noise:

A second microphone may be used to measure the ambient sound pressure and to predict the noise at the location of the DUT. Due to the 3D-complexity of the ambient sound field it appears not to be feasible to compensate the ambient noise influence at the microphone position similar to an active noise compensation. With respect to robustness a corrupted measurement should be repeated automatically.

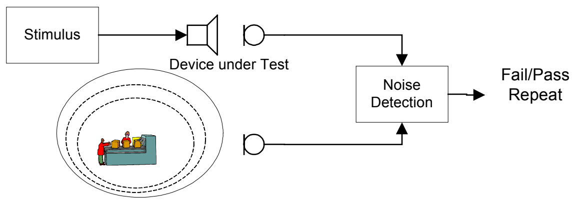

Fig. 13: Automatic repeat of a measurement corrupted by ambient noise

3.3. Hardware

There are a couple of practical demands on the measurement hardware:

3.3.1. EMC-Requirements

For reliable operation under production conditions immunity against electromagnetic disturbances (caused by robotics, motors and magnetizers) is required. Thus a dedicated hardware which integrates all hardware components in one device is preferable.

3.3.2. Sensors

End of line testing requires usually four sensors: One test microphone for measuring high SPL at the DUT, a second microphone for ambient noise, and voltage and current at the loudspeaker terminals.

The usage of galvanically decoupled current and voltage sensors avoids ground loops and gives the robustness required under production condition.

The test microphone should have a sufficient frequency response, handle the high sound pressure occurring in the near field and a good signal to noise ratio.

3.3.3. Power Amplifiers

The measurement of micro speakers and high power PA drivers can not be accomplished with the same amplifier. Thus an external power amplifier gives more flexibility to satisfy different applications.

3.3.4. Sensors for Climate Condition

Monitoring temperature and humidity is necessary to initiate a recalibration of the limits in time. Automatic recording of climate condition should complete the measured data.

3.3.5. Digital Control Interface

The measurement system should have a flexible interface in order to be integrated into an automatic assembling line.

3.4. Flexibility

Quality control in mass production depends highly on the particular product, expectations of the customer, organization of the manufacturing process, etc. Thus, the measurement system has to be flexible to meet the individual needs. On the other hand customization should be accomplished with minimal effort in a short time. The measurement system should provide simple tools giving the functionality while hiding unnecessary complexity. Most customizations should be accomplished by changing the setup without any programming. Beyond that a high-level programming language should be available.

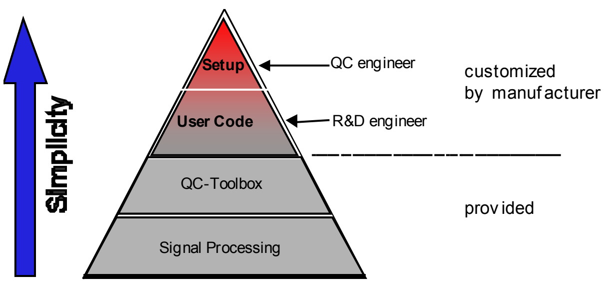

Fig. 14: Hiding information and complexity to make customization simple

3.5. Simple use

The users of a QC system can not be considered as a homogeneous group. They have to solve particular tasks, have different qualifications and use the measurement system in a different way. The operation of the system should be simple for all of them. This will shorten the learning curve, simplify the handling and avoid human errors in the operation of the system.

To make the user interface as intuitive as possible it is natural to present a separate interface for each user group. The interface should be minimal for the operator e.g. a button to start the measurement and a pass/fail display. The QC engineer needs a much more complex interface in order to modify the measurement setup and to adjust the limits.

Another way to make the user interface more intuitive is to adapt it to the special application. Only those setup parameters that are required for the current application should be visible. This can be realized by using a programmable user interface. The user interface for each application could be defined in a script which can be easily customized.

QC staff may speak different languages. The system should support multiple languages in order to shorten training times. This is especially important for the user interface used by the operator.

3.6. Communication across Borders

In a globalized economy production sites may be located on one end of the world and the engineering experts on the other. The measurement have to support the data exchange considering the following demands

- distribute sealed setups
- view measured data on other PCs
- store data in tamper-proof files
- access the QC system remotely.

4. CONCLUSION

The discussion of the different aspects of the end-of-line testing showed that the requirements in production are different than demands in R&D.

The short production cycles make it more and more impossible to wait for a steady-state response of the transducer. Spectral resolution, the SNR of the monitored signal and probability for activating irregular defects becomes worse while reducing the measurement time. Thus the selection of an optimal stimulus and the signal analysis applied to derive meaningful data requires other tools as used in loudspeaker design.

The majority of measurements made in R&D are performed to assess the performance of the DUT and to compare it with design choices and competitive units. In this case the interpretation on an “absolute scale” based on general standards is important while the detection of defective units under production conditions can be realized with relative data due to the non-ideal measurement conditions.

Furthermore, the measurement of symptoms of irregular defects which are almost inaudible plays a minor role in R&D. On the contrary, the measurement system in production has to select a defective unit even if the symptoms are small. For example loose particle in the gap may produce only low distortion if the driver is operated in horizontal position on the assembling line but might impair the sound quality significantly in the final application. Thus the sensitivity of an objective measurement instrument should aim beyond the human ear.

While there are obvious reasons for performing the routine work by an automatic test system the human operator is set free for new tasks. His expertise is required to detect the physical cause of a potential defect and to recognize systematic problems as soon as possible. The separation of systematic and accidental defects can be supported by statistical analysis performed by the measurement system. Exploiting predicted trends the operator becomes proactive and takes actions in process control before defective units have been produced. The fast measurement of the nonlinear parameters *(Bl(x), Kms(x))* and the detection of a voice coil offset and other defects of the suspension is still a challenge.

The user group of a R&D tool is much more homogeneous than the QC staff. A measurement platform intended for loudspeaker design may provide more information than necessary to solve the particular task. It shows the performance from different perspectives and generates a creative environment which inspires the engineer to go beyond the known limits. A measurement tool with high complexity is less useful in manufacturing. Here a conservative tool which hides irrelevant information is preferable. Thus a flexible user-interface is required to meet the different expectations of an operator, QC engineer or programmer. In addition every manufacturer needs a highly customized measurement system to meet the internal conditions on the production site and to satisfy the special needs of his customer.

Only an affordable system will be used on the assembling line. The cost comprises not only the price of the system but also costs for training, support and maintenance. The costs can be reduced by starting with a minimal configuration while having the option to upgrade the system when the requirements grow. This gives a high benefit-cost ratio on a long term basis.

Summarizing loudspeaker testing at the assembling line requires a special tool which is highly dedicated to this task.

5. REFERENCES

[1] L. Beranek, "Acoustics”, Published by the Acoustical Society of America through the American Institute of Physics, Inc., New York, 1993.

[2] R. H. Small, „Direct-Radiator Loudspeaker System Analysis,“ J. Audio Eng. Soc., vol. 20, pp. 383 – 395 (1972 June).

[3] W. Klippel, „Loudspeaker Nonlinearities – Causes, Parameters, Symptoms”, presented at the 119th Convention of the Audio Engineering Society, 2005 October 7–10, New York, USA, preprint #6584.

[4] W. Klippel, „Measurement of Impulsive Distortion, Rub and Buzz and other Disturbances”, preprint # 5734 presented at the 114th Convention of the Audio Engineering Society, 2003 March 22–25, Amsterdam, The Netherlands.

[5] W. Klippel, S. Irrgang, „Rub & Buzz Detection without Golden Unit“, Application Note #22 of the KLIPPEL ANALYZER SYSTEM, www.klippel.de, Dresden 2003

[6] W. Klippel, U. Seidel "Signal Distortion Measurement and Assessment System and Method,” U.S.-Patent #10/376,080 (2003), Chinese Patent application #03108708,6