VOICE COIL
THE PERIODICAL FOR THE LOUDSPEAKER INDUSTRY

Article prepared for www.audioXpress.com

Voice Coil Interviews Dr. Wolfgang Klippel

By Steve Mowry

PHOTO 1: The staff of Klippel GmbH from left to right: Michael Mende, Aaron Heuschmidt, Uta Klippel, Peter Hauptmann, Stefan Irrgang, Ulf Seidel, Wolfgang Klippel, Jan Günther, Joachim Schlechter, Annette

Wolfgang Klippel was born in Halle, Germany, in 1957. He studied electrical engineering at the University of Technology in Dresden, Germany, from which he received a degree in the field of speech recognition in 1982. After graduating, Wolfgang joined the loudspeaker research group of VEB Nachrichtenelektronik Leipzig, where he was engaged in the research of transducer modeling, acoustic measurement, and psychoacoustics. In 1987 he received a doctor's degree in technical acoustics. His thesis was on the "Multidimensional Relationship Between Subjective Listening Impression and Objective Loudspeaker Parameters."

He continued research on the audibility of nonlinear loudspeaker distortion and started modeling the nonlinear mechanisms in low frequency transducers and horn loudspeakers. In 1992 he summarized the results of his research in a thesis, "The Nonlinear Transfer Characteristic of Electroacoustic Transducer," submitted for a Certificate of Habilitation. In 1993 he received the AES Publications Award for his outstanding paper, "The Mirror Filter—A New Basis for Reducing Nonlinear Distortion and Equalizing Response in Woofer Systems."

After spending a postdoctoral year with the prestigious Audio Research Group in Waterloo, Canada, and working at Harman International, in Northridge, Calif., he moved back to Dresden in 1995, where he became an independent consulting engineer. In 1997 he founded Klippel GmbH (www.klippel.de), an innovative company that produces novel control and measurement systems for transducers and loudspeakers. Wolfgang Klippel is currently an AES Fellow and participates actively in the AES Standards Committee SC-04-03. He is currently Vice President of ALMA International Europe. In 2007, he became a professor of electroacoustics at the University of Technology in Dresden.

SM: Could you tell us about Klippel GmbH's mission and what role the individual staff expertise plays?

WK: The mission of the Klippel GmbH is just engineering in the very old sense. We enjoy developing new ideas up to products in the field of control and measurement dedicated to loudspeakers and other transducers. I like the combination of traditional electroacoustics, signal processing, and the daily business of a small-size company. Our work comprises not only the (rare) euphoric moments of having a new idea, but also the long tedious way of its validation (you may call it research) and finally making a product out of it. The last step is most challenging and satisfying because the communication with providers and customers is always stimulating.

Our ultimate goal is the benefit for the user of our tools. If we do not find the right answer here our small company would not exist very long. This puts our feet on earth and we like it. This pragmatic attitude coupled with a flavor of adventure is the common motive for everybody who has joined our company. Besides the fact that everybody contributes something depending on his technical, social, or organizational skills, a team of 11 people is like a small boat where everybody has to find his place. There is enough free room for personal development and for taking responsibility.

SM: It seems that almost overnight the Klippel DA System

has become the industry standard for transducer measurement and evaluation. Would you please comment on that?
WK: For most of our customers our measurement system is indeed an indispensable tool for loudspeaker development and manufacturing. We see this each time a hardware unit gets its regular calibration in Dresden and we have to provide a replacement unit temporarily. The reason is that distortion measurements provide only symptoms of the nonlinearity, but large signal parameters reveal much more directly the physical causes which limit the maximal output of a loudspeaker. This is crucial for developing new drive units and systems having smaller size, less weight and cost but superior quality than traditional units.

Actually, essential pioneering work in this field has been performed by A. Kaizer, who developed a good large signal model for loudspeakers, and David Clark, who made the first available system (DUMAX) for measuring Bl(x) and Kms(x) curves. Our contribution is the full dynamic identification of the large signal parameters following the steps of Morten Knudsen in the early nineties. A powerful large signal model and an easy way for measuring a set of few meaningful parameters also changed the design process. Now harmonic and intermodulation distortion, thermal and nonlinear compression of the output can be predicted by simulation and auralization techniques.

SM: I see the DA as a powerful verification and measurement tool that is most useful when applied at the end of the transducer development process. A transducer (DUT) is required. The transducer component finite and boundary element and numerical design simulations are typically performed at the very beginning of the new product development process. Sure the DA could be used in an iterative design process, build–test–build–test. . . much like other measurement systems have over the years, but that's typically an ineffective and/or inefficient design methodology. All we have to do is to look at what other industries are doing. It's impossible to add value to a product by testing alone. The information obtained from testing can be used to verify and/or help correct the material properties and/or simulation models in a closed-loop design and development process that begins in the virtual domain on the desktop. Could you please comment on this?
WK: Yes, loudspeaker development is always a combination of predictive work on virtual targets and the verification of the real prototype by measurements. In the past most of the design process has been accomplished by using lumped parameter models and analytical approximations, but nowadays powerful numerical simulation tools (FEA, BEA) are available for solving distributed problems such as break-up modes and radiation. We have been perceived as being active only on the measurement side, but this is not true. We developed tools for integrating numerical analysis with measurements and subjective testing.

For example, we satisfied the need of measuring the Young's E-modulus and loss factor of any material used in loudspeaker design and scanning the geometry of loudspeaker parts at high precision which are required as FEA input. There is also a tendency that the loudspeaker designer uses special, separate tools that are just optimal for designing motor, cone, and suspension and the acoustical system. The results of each tool are usually linear transfer functions (e.g., sound and impedance response) and parameters (e.g., Re, Mms, and a Bl(x) curve) that are independent on the instantaneous signal.

For putting the different linear and nonlinear parts together to a loudspeaker system, we developed a Large Signal Simulation module (SIM) that calculates the sound pressure output, distortion, amplitude compression, coil temperature, and even motor stability for different kinds of test signals. The SIM module can also be used at the beginning of the design process, e.g., for defining the large signal parameters which give the desired target SPL at a permissible level of distortion. A curve editor for the nonlinear parameters makes it simple to change virtually the voice coil height and overhang, use and position shorting rings, and other parameters of the suspension and enclosure system.

SM: Could you describe your controller chip?
WK: For many years I had a dream of a control system dedicated to loudspeakers and other dynamic transducers. As you know, loudspeaker history is full of such ideas using motional feedback, current-driven amplifiers, negative impedance sources, and other things. I am working on a feed-forward control that can be realized as a low-cost software solution implemented in DSP where anti-nonlinear distortion components are synthesized to compensate for the distortion generated by the transducer.

We are still on the algorithmic level but have started thinking more and more about a cost-effective implementation. General-purpose DSPs are just too expensive to compete with paper, steel, and magnet and the processing power is actually not required. We think that our approach can be better realized by simple fixed-point DSPs produced in high quantities for general motor control in white goods, automotive, and other applications.

SM: Is your controller an application of your mirror filter?
WK: Yes, the mirror filter is the feed-forward control part where the audio signal is processed. However, there is another part, which performs automatic learning and updating of the large signal parameters while reproducing an audio signal. There is also full mechanical and thermal protection of the loudspeaker, which can be only realized by using the instantaneous state information (temperature and displacement of the coil) available in the controller. All those parts are already used in our Large Signal Identification (LSI) and the Power Testing module (PWT) within our R&D measurement system.
SM: What are some of the limitations of your controller?
WK: Our approach is based on physical modeling of the transducer. It works only for electrodynamical transducers coupled with an acoustical system having a few number of resonances like a closed or vented box. We can reduce the distortion generated by motor and suspension nonlinearities but not the nonlinearities in the cone and in the following

multi-dimensional transfer path. Fortunately, the dominant nonlinearities are located in the one-dimensional path close to loudspeaker terminals. Thus we have better chances to compensate the dominant loudspeaker nonlinearities than equalizing the amplitude response at multiple points in the sound field.

SM: When will the controller be available for system implementation?

WK: I think it will take some more years; however, this is a short time compared with the 20 years that I have been working on this subject already. The technology is completely new and should be smoothly embedded in the audio chain. The system designer needs a simple-to-use black box system that behaves robustly under all conditions and costs almost nothing.

The main benefit will be generated if the system designer couples high efficient, lightweight drive units (which are nonlinear!) with class-D amplifiers and nonlinear control. Passive driver design will get new degrees of freedom and will focus on efficiency, size, weight, cost, and radiation into the 3D space—things that can never be realized by one-dimensional signal processing. Finally, the integration into a modern loudspeaker system is the most important challenge and requires new ways of communication between driver, system, and DSP design.

SM: Why do you think transducer and loudspeaker manufacturers typically do not publish the large signal parameters?

WK: We have to distinguish between the manufacturer of drive units and complete loudspeaker systems. I don't see a need for telling the consumer or any other end-user about large signal parameters of the drive unit used in an active loudspeaker system together with a digital input, a digital crossover, a dedicated amplifier, and an enclosure. The end-user is more interested in the output performance that can be described by a maximal SPL at permissible distortion level.

By the way, a traditional measurement of harmonic distortion would not be sufficient, but we must consider the (amplitude) intermodulation distortion as well. The electrical input power of an active loudspeaker system becomes also less important because it tells the consumer more about power saving in an environmental sense than about acoustical output performance. The acoustical output power of the loudspeaker system gives only small numbers that are not very impressive for marketing and sales.

The situation is completely different for the relationship between driver and system manufacturer. Large signal parameters based on linear, nonlinear, and thermal modeling are independent of the stimulus (test signal, music) and make the communication more effective. For example, a driver without aluminum or copper rings may produce a few percent of harmonic distortion but 10 times more intermodulation distortion products. The nonlinear curves of the parameters L(x) and L(i) directly reveal the cause of the distortion and give further indication for optimizing size and position of the shorting rings.

The measurements of symptoms and output performance are time-consuming and depend on the particular measurement conditions. This is also true for assessing the thermal behavior and power handling which highly depends on the spectral properties of the stimulus. A few thermal parameters can describe forced air convection cooling and pole heating by eddy currents and allow prediction of coil and magnet temperature for any input.

The concise specification of complete loudspeaker systems and drive units is a subject of new projects running in the AES standard committee SC-04-03 in connection with ALMA International.

SM: Do the large signal parameters represent nonlinearities that are audible at low frequencies, or are they masked by psychoacoustic phenomenon and require some special listening protocol to quantify perception?

WK: Audibility of nonlinear distortion generated by loudspeaker systems is a complicated issue. We must be careful with generalizations and simplifications. The audibility highly depends on properties of the stimulus, the linear and nonlinear parameters of the transducer, and also on the hearing capabilities and training of the listener.

For example, we can easily detect a few percent distortion in a signal having a sparse spectrum like a two-tone signal, multi-tone complex, or an organ tone, but we accept more than 10% distortion in a complex stimulus like rock'n'roll music. Psychoacoustic mechanisms such as spectral masking give some explanation for this. We also know that the temporal properties of the waveform and the envelope are important. For example, our ear is much more sensitive to amplitude modulation caused by L(x)-variation than for phase modulation caused by the Doppler effect where the amplitude spectra are similar.

The matter becomes even more complex if we discuss the impact on perceived sound quality considering attributes such as preference, disturbance, and naturalness. The distortion related to motor nonlinearities, Bl(x), L(x) and L(i), contains a lot of intermodulation components, which cover the whole audio band and impair the sound quality significantly. Contrary to this, distortion generated by a nonlinear suspension in a subwoofer is partly acceptable and makes the bass louder and more aggressive. However, if the resonance frequency is higher than 100Hz, the higher-order harmonics cannot replace the effect of the fundamental, and we then perceive the distortion as not very natural.

Practically speaking, subjective listening tests are also required to assess the loudspeaker's large signal performance. Contrary to linear distortion (which is generated in the same way also in the small signal domain), nonlinear distortions are directly related to maximal output, efficiency, size, weight, and cost of the loudspeaker. We should keep it just as low as required by our target user group. Clearly in a convenience product much higher values are acceptable than in a high-quality home theater. The new auralization technique developed in our company makes it possible to combine subjective and objective testing and to empirically tune the loudspeaker just for the target application.

SM: Is the Klippel DA et al. being used at any universities as a research tool?
WK: Yes, it is used in education and research at the Technical University of Denmark, National Chiao-Tung University in Taiwan, the Jusang College in Korea, and other universities all over the world.

We support this by giving a special discount to schools because we think that the combination of modeling and measurement is an exciting new field, especially for young students of acoustics and audio. Our measurement system also provides and incorporates SCILAB, which is a high-level language very similar and almost compatible to MATLAB. SCILAB is free and can be used everywhere at the campus and in the industry. This language and easy access to our measurement modules via the COM interface will inspire and help young students to develop their own ideas.

SM: Can you describe the ideal transducer/loudspeaker/system, but within today's technology limits?
WK: I believe that the electrodynamical transducer principle will still be used for some time because it is now the best compromise in sound quality, robustness, cost, and ease of manufacturing. Later on we might cultivate an alternative principle that is mainly required to realize a distributed sound system comprising not only two, five, or ten loudspeakers, but maybe hundreds of individually controlled sound sources. This would give us the possibility of reproducing the directivity pattern of the original source more precisely and to compensate for undesired properties of the room. If the number of loudspeakers increases, the size of each unit should be significantly smaller, with a digital input and easy power supply. I think that our loudspeaker industry is already on this path, of course, within today's technology limits.

SM: I know there will soon be available a cone scanning module for the Klippel DA. You presented an interesting paper on this topic at the 121st AES Convention in San Francisco in October. Can you describe the module and your acquisition and decomposition of asymmetric and axisymmtric modes?
WK: Two things in "loudspeakers" always attracted me. One is behavior at high amplitudes (loud), the other is the conversion into mechanical vibration and the radiation into the 3D space (speaker). The prediction and measurement of cone vibration is actually a fundamental key in loudspeaker design.

A variety of simulation tools based on finite element analysis provides this data that also must be verified by mechanical measurement. Available Doppler interferometers are cost-intensive, and the measurement of velocity gives no precise geometry of the measurement target (cone). However, both kinds of data (vibration and geometry) are required to predict the sound pressure output and to investigate the relationship between mechanical and acoustical domain.

We developed an interesting alternative, which is *not* Doppler interferometry, but uses a displacement laser sensor based on a triangulation technique that provides the precise geometry and vibration data up to 25kHz. In addition to the cost-effective scanning hardware, we also developed a powerful software analysis tool for visualization, animation, and sound pressure prediction. A new thing is the decomposition technique in which we separate different modes (asymmetric and axisymmetric) and the vibration components that produce the sound (in-phase), reduce the sound (anti-phase), and produce no sound at all (quadrature component). This technique simplifies the interpretation and gives indications for further improvements.

SM: Do you plan to teach any classes at the Technical University in Dresden?
WK: Yes, I am preparing a lesson on "active control of sound and vibration," addressing advanced topics in signal processing and acoustics. It is a new challenge. I like not only the subject but also the opportunity to work with young people.

SM: What can I expect to see from Klippel GmbH in the near future?
WK: We are asked by loudspeaker manufacturers to measure also large signal parameters during end-of-line testing. Our LSI module takes a few minutes to present the nonlinear curves and can only be used for quality check on randomly selected units. Currently we are developing a motor and suspension check that provides the voice coil offset in mm, asymmetry of Kms(x) in percent, and other useful numbers within 1s. This information may be valuable for 100% testing, checking suspension parts, and controlling the production process.

SM: Is there anything that you would like to add to our discussion?
WK: I mentioned already the importance of the communication between driver, system, and DSP design and the role of a meaningful specification. I would like to encourage all engineers working with loudspeakers to participate in the standard projects running in standard committees in AES, IEC, and ALMA International.
SM: Well, thank you for your time and your dedication to improving transducers, loudspeakers, and audio systems.

Interviewer's Closing Comments

Wolfgang Klippel is a bright, energetic, and diligent researcher, educator, and entrepreneur who has helped to raise the technical standards of loudspeaker engineering. His work on the nonlinear parameters is some of the most significant to come along in the last ten years. I consider his technical paper, "Loudspeaker Nonlinearities—Causes, Parameters and Symptoms," to be a classic and a must read (www.klippel.de/download/Nonlin/klippel,%20Loudspeaker%20 nonlinearities%20-%20causes%20and%20symptoms.pdf). Transducer and loudspeaker engineers should have the Klippel "Loudspeaker Nonlinearities" poster on their office wall. (www.klippel.de/download/Nonlin/Klippel_nonlinearity_poster.jpg).

Additionally, Wolfgang Klippel's commitment to presenting papers and holding technical information seminars throughout the world while remaining active in the AES and ALMA is a nice way of giving something back to the loudspeaker industry. ***VC***

Steve Mowry, president of SM Audio Engineering, has a BS, Business Administration, from Bryant College, and a BS and MS, Electrical Engineering, from URI with highest distinction. Steve has worked in R&D at BOSE, TC Sounds, EASTTECH, and P.Audio. Steve is currently an independent consultant/lecturer in project management/transducer and system design. His website is www.s-m-audio.com.