AES SWISS SECTION NEWSLETTER

95th Issue

INFORMATION ON NEXT MEETING

Measurement of the speech intelligibility of sound systems

(Joint meeting with SGA/SSA)

Thursday, 6th of May 2004, 17h30 at EMPA-Akademie,
Überlandstrasse 129, Dübendorf

SPEAKERS: Kurt Eggenschwiler, Vergleichmessungen
David Norman, STIPA

ORGANIZER: Walter Köller

LANGUAGE: German

The common meeting between the sister-societies that are the AES and the SSA will have for theme : measurement of the speech intelligibility of sound systems. Kurt Eggenschwiler will present the surprising comparative measurement results and will analyze the sources of mistake. David Norman will speak to us of the new measurement of the speech intelligibility: STIPA. Demonstrations of some specific facilities for these measurements will also take place.

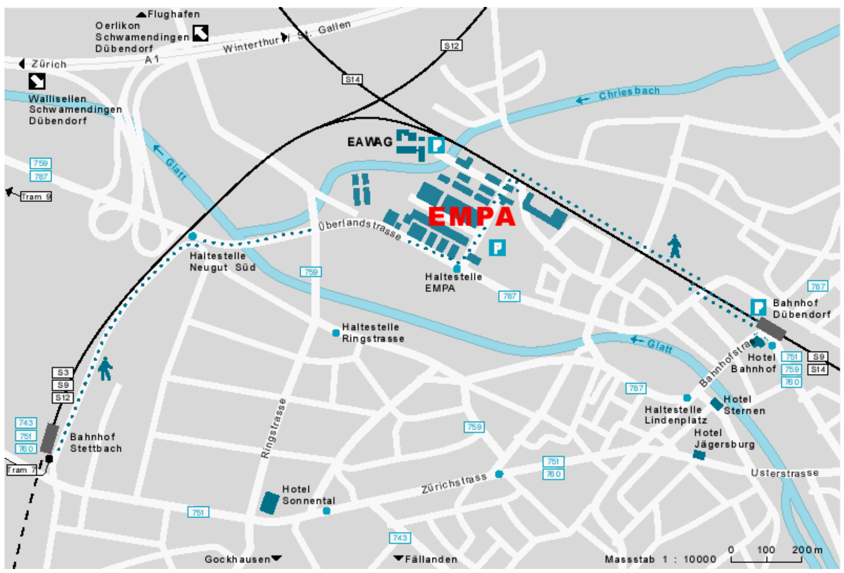
Intercomparison measurements of room acoustical parameters and measures for speech intelligibility in a room with a sound system

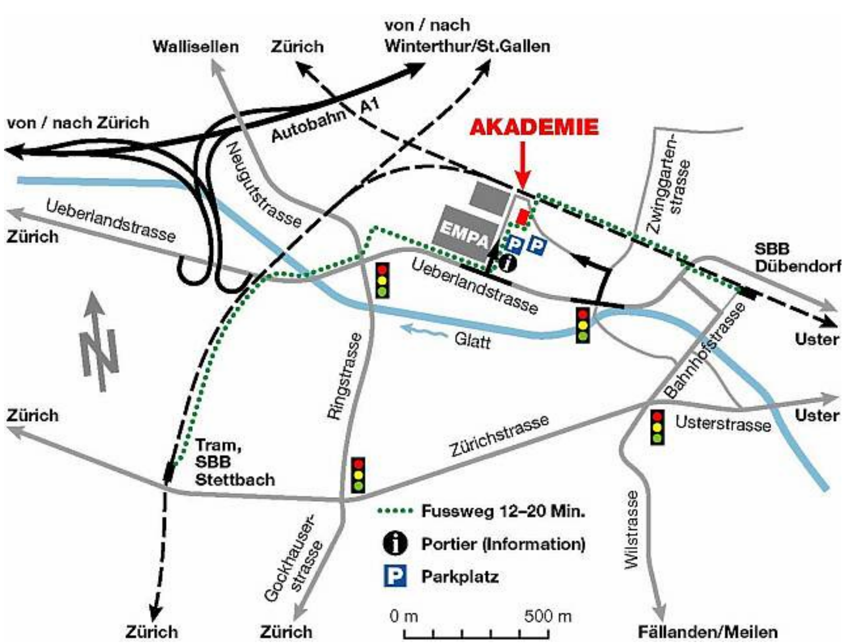
Intercomparison measurements of room acoustical parameters and speech intelligibility measures were performed in a 1500 m^3 room containing a sound system. The parameters included the reverberation times T20 and T30 averaged over the entire room, the speech intelligibility indices STI, STI(mod), STI(male), STI(female), RASTI and %Alcons, the frequency response in third-octave bands and room acoustical parameters at various positions in accordance with ISO 3382. As far as can be ascertained, two factors appear to be responsible for the spread in results. On the one hand, the noise disturbances, at times quite strong, seemed to pose difficulties for the participants. Frequently, too little attention was paid to achieving an adequate signal-to-nose ratio. Furthermore, for some measurements, the results of certain participants varied greatly from those of the others, partly as a result of incorrect settings of the instruments. In the measurement analysis, the standard deviations were compared to the perception thresholds. When the data from a few conspicuous participants was eliminated, the spread in data was often reduced considerably, the standard deviations then often lying in the region of the perception thresholds.

Biographies:

Kurt Eggenschwiler studied electrical engineering at the Swiss Federal Institute of Technology (ETH) at Zurich. Since 1985 he has worked at the Laboratory of Acoustics at Swiss Federal Laboratories for Materials Testing and Research EMPA, Dübendorf, today as the head of Laboratory of Acoustics. His main interests are room acoustics and environmental acoustics in research, development and consulting. Since1995 Kurt Eggenschwiler has been a lecturer for architectural acoustics and later for noise control at the Swiss Federal Institute of Technology (ETH). He is member of the Audio Engineering Society and of Swiss Acoustical Society.

David Norman was born in 1947 in Ipswich, England and Studied Physics at Royal College of Science London gaining BSc, and ARCS titles. He started working in professional audio at the end of the `60s in a small London company J. Richardson Electronics Ltd. He was then invited to work in Switzerland and designed the complete electronics for a recording studio in Biel/Bienne Switzerland. After this he worked for a while designing electronics for hi-fi applications at Lenco Switzerland and then joined Electro-Voice SA Switzerland (later Mark IV Audio) where, for 16 years, he was the European Technical Manager. During this time he was responsible for the design of many sound systems in stadiums such as Hallenstadion Zurich, Ice Stadium Allmend Bern, Sport Hall St. Jakob Basle and Hardturm Football Stadium, Zurich, as well as other applications such as the Film Festival, Locarno and was responsible for the sound for 10 years at the Jazz Festival, Montreux.
Since 1996 he has been running his own consulting company working on many sound systems in and around Switzerland. He is a member of the: Audio Engineering Society and the Swiss Acoustical Society. He has been chosen to represent Switzerland on the IEC committee dealing with inductive loop systems.





REPORT ON PREVIOUS MEETING

DSD vs PCM – Principles and comparison

Wednesday, 10th of March 2004 at Hochschule des Künste Bern, 3000 Bern

SPEAKERS: Claude Cellier, Merging Technologies
John Goldstraw, Metropolis, London

REPORTERS: Attila Karamustafaoglu

Around 60 participants, invited by the Swiss AES Section and J+C Intersonic joined the evening part of the event starting at 1700. After a short introduction Claude Cellier started with his presentation, which was addressed to the theoretical and the technical aspects of DSD in comparison to conventional PCM. First a theoretical introduction was made. He explained, that DSD is an audio format, which has 1 bit resolution and is normally sampled with 2.8MHz sampling rate. The sampling rate way beyond the audible spectrum allows to shift the compared to PCM high quantization noise to the inaudible bands and provide a signal which is capable of the representation of higher spectral components as even the high definition PCM formats as 96kHz or 192kHz. In analogy to analogue it was shown that when a DSD signal is representing a “Dirac Impulse”, the pre ringing does not occur which might be audible with PCM. Next he explained that DSD is mainly a delivery format and not a processing format. On the simple example of a gain change, it was shown, that a 1-bit sampled sequence couldn’t easily be gain-changed. So for almost every process, the format has to be converted to PCM, which results in a small loss of quality due to the noise shaping algorithms. As in analogue, there is therefore generation degradation between every processing step. For Merging’s adoption of DSD into their Pyramix workstation, they have defined therefore an intermediate format called DXD, which stands for “Digital Extreme Definition”. Not being a DSD format anymore, it utilises 32bit PCM with 352.8 kHz or 8 times the base sampling rate of 44.1. This format should allow the user to work within the workstation without giving up too much of the prospected quality of DSD. As most people knew already, DSD is the audio format of the high definition layer of the SACD driven by Sony and Philips.

At the end of the presentation, Claude Cellier gave an outlook to the future of DSD, letting even higher data rates to be expected, also for DXD. After this, John Goldstraw from Metropolis Studios in London went more into the practical aspects of DSD. He explained his first experiences and problems with this format, which transports also signal information outside the audible range. One of it being DC he told from a session where the speakers began to overheat due to the DC offset and broke down. Then he explained that actually there are seven factories being able to produce DSD SACDs and at the beginning there was no common standard for the master data. This leads in one case to a SACD production where the audio track was shrunk to several seconds instead of around one hour. But he pointed out that after some learning time, as there is with almost every technology, they are now able to produce DSD as well as higher rate PCM and have Sadie workstations using SuperMAC to interconnect. He pointed out that their philosophy is to provide everything the customer wants. There was even one example where one customer wanted to make a production where some tracks were in DSD and some in PCM – a real challenge for a sound engineer. As a big plus in the market battle between SACD and DVD-A he mentioned the simple fact that every SACD has a standard CD layer and is therefore playable in a standard CD player. Some customers even don’t know that they are buying SACDs since they don’t differ in price compared to CDs.

After these very interesting talks the meeting was closed and many discussions continued at the following dinner in the restaurant nearby.