Schuyler Quackenbush, PhD

Audio Research Labs
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Summary:

- Over 30 years of academic and professional experience in speech and audio signal processing, speech and audio signal compression technology, standards for speech and audio compression, and software implementation on real-time platforms and in client/server architectures.
- Chairman, MPEG Audio Subgroup, ISO/IEC Standards Organization
- Member of Technical Staff at Bell Laboratories for 16 years.
- Founder of two start-ups: Audio Research Labs and Lightspeed Audio Labs
- Holder of 24 Patents
- Experienced patent litigation expert witness

Areas of Expertise:

- Speech and audio digital signal processing (DSP)
- Speech and audio signal compression
- Speech and audio compression standards: MPEG MP3, MPEG AAC, MPEG HE-AAC, MPEG Surround, MPEG Spatial Audio Object Coding (SAOC), MPEG Unified Speech and Audio Coding (USAC); 3GPP aacPlus, Enhanced aacPlus, AMR, AMR-WB, AMR-WB+; ETSI GSM.
- Measurement of speech and audio signal quality: Mean Opinion Scores, ITU-T and ITU-R quality standards.
- Operating Systems: Linux, MS Windows (hardware drivers and GUI)
- Programming Languages: C and C++, Matlab, shell, various assembly languages
- Web Languages: HTML, PHP, MySQL

Education:

1985   Ph.D. in Electrical Engineering; Thesis: "Objective Measures of Speech Quality"
                    Georgia Institute of Technology, Atlanta, GA
1980   MS in Electrical Engineering; specializing in Signal Processing
                    Georgia Institute of Technology, Atlanta, GA
1975   BSE in Electrical Engineering, with honors
                    Princeton University, Princeton, NJ

Professional Experience:

2002 – Present **Founder and CEO**
                Audio Research Labs, LLC
                336 Park Avenue, Scotch Plains, NJ, 07090

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I am the principal consultant at Audio Research Labs (ARL), which is a media technology consulting company. While at ARL, I have done patent expert witness work, patent litigation support, patent valuation, engineering consulting, and participated in standardization activities. ARL has developed and is selling products for subjective audio evaluation and for multi-channel audio mixing.

2013 – Present **Adjunct Professor**
New York University Steinhardt School
New York, NY
I teach the graduate-level course GE-2632 "Introduction to Perceptual Audio Coding." The course gives an introduction to the elements from which digital audio codecs are built: the human auditory system, hearing acuity, modeling of noise masking in human hearing & sound localization in space; filter banks, transforms, predictors, quantization & coding. These principles are illustrated via an investigation of several MPEG audio coding architectures: MPEG-1 Layer III (MP3), MPEG-4 Advanced Audio Coding (AAC), MPEG-4 High-Efficiency Advanced Audio Coding (HE-AAC), MPEG Surround, and MPEG Unified Speech & Audio Coding (USAC).

1998 – Present **Chair, MPEG Audio Subgroup**
ISO/IEC Standards Organization
Geneva, Switzerland

As Chair of the International Standards Organization, Moving Picture Experts Group (ISO/MPEG) Audio subgroup, I am responsible for recommending areas for possible standardization, setting and executing the agenda for current work and developing a vision for future work. The Audio subgroup consists of approximately 50 audio experts, and my responsibilities include delegating tasks to and managing task completion by the group, forging consensus on group decisions, and reporting on the group's work in the MPEG plenary sessions. Notable accomplishments of the group during my tenure were standardizing the following technology
- High-Efficiency Advanced Audio Coding (HE-AAC)
- Enhanced Low Delay Advanced Audio Coding (AAC-ELD)
- MPEG Surround
- Spatial Audio Object Coding (SAOC)
- Unified Speech and Audio Coding (USAC)

2006 – 2009 **Founder and VP of Audio Technology**
Lightspeed Audio Labs, Inc.
106 Apple Street, Suite 221, Tinton Falls, NJ 07724
(732) 450-1444

Lightspeed Audio Labs is about changing the way people create, listen, and share audio content on and over the Internet. Its technology platform provides a virtual studio and venue for musicians, jammers, and fans alike to participate in the

music making process, in which unique musical content is shared with others in real-time.

I was responsible for designing, developing, and testing all aspects of the Lightspeed client/server architecture for real-time audio streaming, recording, mixing and playback. High-bandwidth servers at three geographic locations provided more than 250 simultaneous virtual "jam rooms" in which "jammers" could collaborate with other musicians in the virtual room via low-latency streaming audio links (with less than 50 ms round-trip latency). Any given "jam session" could be streamed live to as many as 1500 listeners. Audio jam "archives" could be edited into songs that could be downloaded or posted on a user's home page. One part of the client user interface was a web browser with pages coded with HTML and PHP languages and using a MSQL database for user state information. A second part of the user interface was a helper application that connected to an streaming audio application server and used UDP for robust real-time performance.

2000 – 2002	**Acting Supervisor** **Speech and Audio Coding Group** **AT&T Laboratories** 180 Park Avenue, Florham Park, NJ

I supervised technical staff in the Speech and Audio Coding Department. Responsibilities included: mentoring technical staff, setting research goals, conducting performance evaluation reviews and reporting evaluations to management.

In addition to supervisory responsibilities, I continued the research work discussed under the AT&T Bell Laboratories heading, immediately below.

1996 – 2000	**Principal Technical Staff Member** **Speech and Audio Research Department** **AT&T Laboratories** 180 Park Avenue, Florham Park, NJ

My responsibilities at AT&T Laboratories continued un-interrupted from those at AT&T Bell Laboratories, hence they are discussed under the AT&T Bell Laboratories heading, immediately below.

1986 – 1996	**Member of Technical Staff** **Signal Processing Research Department** **AT&T Bell Laboratories** 550 Mountain Avenue, Murray Hill, NJ

At Bell Labs (and subsequently at AT&T Labs) I was an expert in audio coding and real-time signal processing, and I developed a considerable expertise in speech and image signal processing and system engineering. I gained a wealth of experience in managing groups of technical experts, in the context of the International Standards Organization (ISO) and related industrial groups.

My principal research projects and responsibilities were:

Chair, MPEG 4 Audio Patent Holders Group

I organized the first meeting of the MPEG 4 Audio Patent Holders Group and was appointed chair by the group members. My responsibilities were to organize meetings, set agendas responsive to the group's needs, and ensure that work delegated to members was completed on schedule.

Chair, MPEG 4 Industry Forum (M4IF) Audio Patent Licensing Group

As chair of this group, I was responsible for formulating a process for identifying the essential patents for practice of the audio portion of the MPEG-4 standard. This task involved identifying expert legal councel, identifying expert technical consultants, and gaining consensus for my plan amongst the prospective patent-holding companies.

Error mitigation for streaming audio signals on 3G Cellular and IP channels

I developed algorithms and corresponding real-time implementations for a novel method of mitigating errors in an MPEG-2 Advanced Audio Coding (AAC) compressed data stream. Subjective quality assessments indicate that this method is always preferred to strategies such as mute or repeat, and in special cases is indistinguishable from the clear channel signal.

AT&T's "A2B" music over the Internet initiative

I was responsible for transferring the AAC technology to AT&T's business of secure sales of music over the Internet. This involved the legal and business aspects of patent licensing and the engineering aspects of bitstream packetization and encryption in a system using compressed rates of 16 kbps for music preview and 96 kbps for music sales.

MPEG-2 Advanced Audio Coding (AAC) International Standard

I was AT&T's principal delegate to the MPEG audio subgroup and was responsible for coordinating the activities of myself and two other audio coding researchers who contributed to the AAC standard. Our team had to work closely with international audio experts to meet the monthly or even weekly milestones as part of the very aggressive MPEG schedule over the course of the 26-month work plan. Due largely to my efforts, AAC contains virtually all of AT&T's audio coding technology, which in large part enabled it to achieve transparent coding of 5-channel audio at 64 kbps/channel. I wrote a significant portion of the software for the AAC encoder and virtually all of the decoder.

US Digital Audio Broadcast standard

AT&T participated in a US standardization effort for digital audio broadcast, sponsored by the National Association of Broadcasters (NAB) and the Electronics Industry Association (EIA). I was responsible for all aspects of the design of the audio encoder and decoder: system engineering, including timing, clock recovery and error robustness; hardware design, including processor specification and custom interface circuits; and software design, including real-time performance. In this effort I led a team of four engineers over a period of 18 months. The resulting real-time audio encoder and decoder achieved compact-disk quality at a channel bit rate of 160 kbps. The entire DAB system had numerous successful trials broadcasting in the FM band.

Streaming media

I developed and implemented a client/server music player using the AT&T audio technology. This used an OpenGL graphical user interface and UNIX socket-based client/server communication. I developed and implemented a streaming client/server architecture for audio and image data that communicated via ISDN.

Reducing various algorithms to practice

I designed and/or refined algorithms, wrote the software and built the hardware for several prototype image, speech, and audio codecs based on DSP chips. These include a wide-band 16 kbps speech coder, a high-quality still image coder and AT&T's first 128 kbps stereo audio coder.

1978 – 1979   **Hardware Design Engineer**
                Diagnostic/Retrieval Systems, Inc.
                Oakdale, NJ
                I was a member of a team that designed and built a ship-based SONAR system. My responsibility was analog signal input, A/D conversion and band-pass filtering of the signal prior to signal frequency content analysis.

1975 – 1978   **Test Engineer**
                Loral Electronics
                Yonkers, NJ
                I was a member of a team that designed and built an aircraft-based RADAR jamming system. My responsibilities were to design and build custom test equipment to exercise and ensure the correct operation of aspects of the overall system.

Professional Memberships:

Audio Engineering Society (AES)
                1997-2006   Member,
                2006-Present   Fellow
International Electrical and Electronics Engineers (IEEE)
                1979-2001   Member,
                2001-Present   Senior Member

Programming Skills:

- Operating systems: Unix, MS Windows. Have written hardware drivers and GUI for each.
- Programming languages: C and C++, Matlab, shell, awk, various DSP assembly languages.
- Web languages: PHP, MySql

Publications:

Journal Papers

1. Quackenbush, S. and Herre, J., "MPEG Surround," IEEE Multimedia Magazine, vol. 12, issue 4, Oct.-Dec. 2005, pp.18-23.
2. Quackenbush, S. and Lindsay, A. "Overview of MPEG-7 Audio," IEEE Transactions on Circuits and Systems for Video Technology, pp. 725-9, vol. 11, no. 6, June 2001.
3. M. Bosi, K. Brandenburg, S. Quackenbush, L. Fielder, K. Akagiri, H. Fuchs, M. Diets, J. Herre, G. Davidson and Y. Oikawa, "ISO/IEC MPEG-2 Advanced Audio Coding," Journal of the Audio Engineering Society, 45-10, Oct. 1997, pp. 789-814.

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