Schuyler Quackenbush, PhD

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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

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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 07932

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 07974

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. M. Neuendorf, et al., “ISO/MPEG Unified Speech and Audio Coding Standard – Consistent High Quality for all Content Types at all Bit Rates,” JAES vol. 61, issue 12, pp. 956-77, Dec 2013.
2. Quackenbush, S., “MPEG Unified Speech and Audio Coding,” IEEE Multimedia Magazine, vol. 20, no. 2, Apr-Jun 2013, pp. 72-78
3. Quackenbush, S. and Herre, J., “MPEG Surround,” IEEE Multimedia Magazine, vol. 12, issue 4, Oct.-Dec. 2005, pp.18-23.
4. 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.
5. 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.
6. Synder, J. H., Quackenbush, S. R., Melchner, M. J and Kapilow, D. A., “Tools for real-time signal-processing research,” IEEE Comm. Mag., vol. 31, no. 11, Nov. 1993, pp. 64-74.
7. Cox, R. V., Gay, S. L., Seshadri, N., Shoham, Y., Quackenbush, S. and Jayant, N. S., “New directions in sub-band coding,” IEEE Jour. Selected Areas in Communications, Special Issue on Voice Coding for Communications, vol. 6, no. 2, Feb. 1988, pp. 391-409.

Books, Book Chapters, and International Standards

1. Quackenbush, S., “Chapter 7: MPEG Audio Compression Advances,” “Chapter 8: MPEG Audio Compression Future,” and Baroncini, V. and Quackenbush, S., “Chapter 13: MPEG Video/Audio Quality Evaluation,” all in *The MPEG Representation of Digital Media*, Chiariglione, L. (Ed.), Springer, New York, 2012.
2. Quackenbush, S. R. and Noll, P., “Part 1 Chapter 1, MPEG Digital Audio Coding Standards,” in *The Digital Signal Processing Handbook,* 2nd Edition, Madisetti, V. K. (Ed.), CRC Press, Boca Raton, FL, 2010.
3. Sinha, D., Johnston, J. D., Dorward, S. and Quackenbush, S. R., “The perceptual audio coder (PAC)” in *The Digital Signal Processing Handbook*, 2nd Edition, Madisetti, V. K. (Ed.), CRC Press, Boca Raton, FL, 2010.
4. Quackenbush, S. R. and Wylie, F., “Digital Audio Compression Technologies” in *National Association of Broadcasters Engineering Handbook*, 10th Edition, Williams, E. A. (Ed.), Focal Press, Burlington, MA, 2007.
5. Johnston, J. D., Quackenbush, S., Herre, J. and Grill, B, “Review of MPEG-4 General Audio Coding,” in *Multimedia Systems, Standards, and Networks*, Puri, A. and Chen, T. (Ed), Marcel Dekker, New York, 2000.
6. Johnston, J. D., Quackenbush, S., Davidson, G., Brandenburg, K. and Herre, J., “MPEG Audio Coding,” in *Wavelet, Subband and Block Transforms in Communications and Multimedia,* Akansu, A. N.and Medley, M. J. (Ed.), Kluwer, Dortrecht, The Netherlands, 1999.
7. Sinha, D., Johnston, J. D., Dorward, S. and Quackenbush, S. R., “The perceptual audio coder (PAC)” in *The Digital Signal Processing Handbook*, Madisetti, V. K. and Douglas, B. W. (Ed.), CRC Press, IEEE Press, 1998, pp. 42-1 to 42-18, Chapter 42.
8. Herre, J., Johnston, J. D., Brandenburg, K., Quackenbush, S. et al., “Generic coding of moving pictures and associated audio: Advanced Audio Coding,” ISO/IEC JTC1/SC29/WG11 MPEG International Standard ISO 13818-7, 1997.
9. Johnston, J. D., Sinha, D., Dorward, S. and Quackenbush, S., “AT&T perceptual audio coder (PAC)” in *Collected Papers on Digital Audio Bit-Rate Reduction*, Gilchrist, N. and Grewin, C. (Ed.), Audio Engineering Society, 1996.

10. Quackenbush, S. R., Barnwell, T. P. III and Clements, M. A., *Objective Measures of Speech Quality*, Prentice-Hall, New York, NY, 1988.

Conference Papers

1. Jameel, J, et al., “ECMA-407: New Approaches to 3D Audio Content Data Rate Reduction with RVC-CAL,” AES 137st Convention, Oct. 2014.
2. M. Neuendorf, et al., “MPEG Unified Speech and Audio Coding – The ISO/MPEG Standard for High-Efficiency Audio Coding of all Content Types,” AES 132st Convention, Apr. 26-29, 2012, Budapest, Hungary.
3. Quackenbush, S. and Lefebvre, R., “Performance of MPEG Unified Speech and Audio Coding,” AES 131st Convention, Oct. 20-23, New York, USA.
4. Quackenbush, S. “MPEG Unified Speech and Audio Coding,” AES 43rd Conference on Audio for Wirelessly Networked Personal Devices, POSCO International Center, Pohang University of Science and Technology (POSTECH), Sept. 29 – Oct. 1, 2011, Pohang, Republic of Korea.
5. Quackenbush, S. and Gross, A., “Analysis of Subjective Data from the MPEG Unified Speech and Audio Coding Call for Proposals,” AES 38th International Conference on Sound Quality Evaluation, June 13-15, 2010, Pitea, Sweden.
6. Quackenbush, S.R and Driessen, P.F., “Error Mitigation in MPEG-4 Audio Packet Communication Systems,” 115th AES Convention, Sept. 2003, Preprint 5981.
7. Kuo, S., Johnston, J. D., Turin, W. and Quackenbush, S.R.; “Covert audio watermarking using perceptually tuned signal independent multiband phase modulation,” IEEE 2002 International Conf. Acoustics, Speech and Signal Proc. (ICASSP '02), vol. 2, pp. 1753-6.
8. Lacy, J., Quackenbush, S. R., Reibman and Snyder, J. H., “Intellectual property protection systems and digital watermarking,” Optics Express, Vol. 3. No. 12, Dec. 1998, pp. 478-84.
9. Lacy, J., Quackenbush, S. R., Reibman, A. R., Shur, D. and Snyder, J. H., “On combining watermarking with perceptual coding,” IEEE International Conf. Acoustics, Speech and Signal Proc. (ICASSP ‘98), Seattle, WA., May 1998.
10. Quackenbush, S. R., “Coding of Natural Audio in MPEG-4,” IEEE International Conf. Acoustics, Speech and Signal Proc. (ICASSP 98), Seattle, Wash., May 1998.
11. Quackenbush, S.R. and Johnston, J.D., “Noiseless coding of quantized spectral components in MPEG-2 Advanced Audio Coding,” IEEE 1997 Workshop Applications of Signal Processing to Audio and Acoustics, Oct. 19-22, 1997.
12. 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,” 101st Convention of the Audio Engineering Society, Oct. 1996, Preprint 4382.
13. Quackenbush, S. R. and Parikh, V. N., “Using C++ for real-time signal processing,” Proc. 1995 IEEE ICSPAT, Boston, Oct. 1995.
14. Quackenbush, S. R., “A CD-quality audio and color still image multi-media platform using the DSP32C,” Proc. IEEE Workshop Audio & Acoustics, Oct. 1991, Mohonk House.
15. Quackenbush, S. R., “A 7kHz bandwidth 32 kpbs speech coder for ISDN,” Proc. 1991 ICASSP '91, Toronto, Canada, May 1991.
16. Quackenbush, S. R., “Hardware implementation of a color image decoder for remote database access,” Proc. ICASSP '90, Albuquerque, NM, Apr. 1990, pp. 985-8.
17. Quackenbush, S. R., Ordentlich, E. and Snyder, J., “Hardware implementation of a 128 kbps monophonic audio coder,” 1989 IEEE ASSP Workshop on Appl. Sig. Proc. Audio & Acoust., New Paltz, NY, Oct. 1989.
18. Quackenbush, S. R., “Hardware implementation of a 16 kbps subband coder using vector quantization,” Proc. IEEE ICASSP, New York, NY, Apr 1988, pp. 386-9.

19. Barnwell, T.P. III, Quackenbush, S. R., “Objective estimation of perceptually specific subjective qualities,” 1985 IEEE ICASSP, 1985, pp. 419-22.
20. Quackenbush, S. R., Barnwell, T.P. III, “The estimation and evaluation of pointwise nonlinearities for improving the performance of objective speech quality measures,” 1983 IEEE ICASSP, vol. 8, Apr. 1983, pp. 547-50.
21. Barnwell, T.P. III, Quackenbush, S. R., “An analysis of objectively computable measures for speech quality testing,” 1982 IEEE ICASSP, vol. 7, May 1982, pp. 996-9.

Patents:

1. 9,160,495, System and methods for transmitting data
2. 8,428,185, System and methods for transmitting data
3. 8,095,794, System and method of watermarking a signal
4. 8,041,038, System and method for decompressing and making publically available received media content
5. 7,802,101, System and method of retrieving a watermark within a signal
6. 7,725,808, System and method for representing compressed information
7. 7,529,941, System and method of retrieving a watermark within a signal
8. 7,492,902, Custom character-coding compression for encoding and watermarking media content
9. 7,451,319, System and method of watermarking a signal
10. 7,353,447, System and method for representing compressed information
11. 7,146,503, System and method of watermarking signal
12. 7,131,007, System and method of retrieving a watermark within a signal
13. 7,076,426, Advance TTS for facial animation
14. 7,042,933, System and methods for transmitting data
15. 6,885,749, Scrambling a compression-coded signal
16. 6,850,559, System and methods for transmitting data
17. 6,760,443, Custom character-coding compression for encoding and watermarking media content
18. 6,718,507, System and method for representing compressed information
19. 6,704,576, Method and system for communicating multimedia content in a unicast, multicast, simulcast or broadcast environment
20. 6,493,457, Electronic watermarking in the compressed domain utilizing perceptual coding
21. 6,341,165, Coding and decoding of audio signals by using intensity stereo and prediction processes
22. 6,266,419, Custom character-coding compression for encoding and watermarking media content
23. 5,825,976, Device and method for efficient utilization of allocated transmission medium bandwidth
24. 5,463,641, Tailored error protection
25. Canadian Patent 2,260,222, Coding and Decoding of Audio Signals by Using Intensity Stereo and Prediction