Schuyler Quackenbush, PhD

Expert in Digital Audio Technology

Audio Research Labs
336 Park Avenue, Suite 200
Scotch Plains, NJ 07090
Phone: +1 908 490 0700
Fax: +1 908 842 9151
Email: srq@audioresearchlabs.com
Web: www.audioresearchlabs.com

Professional Profile:

- **Visionary** – can predict technology trends based on deep understanding of state-of-the-art technology and current market needs.
- **Entrepreneurial** – can pair market needs with technology capability to create new products and services.
- **Scientifically rigorous** – brings the rigorous scientific method of both academic research and Bell Lab's research review process to all aspects of technology development and review.
- **Ethical** – honesty and accuracy is of paramount importance in all aspects of technology investigation, development and review. Fair, honest and open dealings are a requirement for all client relationships.
- **A Mediator** – has the ability to quickly forge consensus within groups so as to develop shared values and goals.

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

Current Professional Responsibilities:

2002 - **Founder and President**
Audio Research Labs, LLC
336 Park Avenue, Scotch Plains, NJ, 07090 (908) 490-0700
www.audioresearchlabs.com

I am the principal consultant at Audio Research Labs, which is a media technology consulting company. I am expert in speech and audio signal processing, audio signal compression, real-time implementation of algorithms. At ARL, I have done extensive patent-related work, both as a testifying and non-testifying expert witness, done engineering consulting to both large companies and start-ups, executed all aspects of numerous formal subjective quality tests and participated in MPEG and other

standardization activities. In addition, ARL has developed and is selling products for subjective audio evaluation and for multi-channel audio mixing.

Expert Witness – I have acted as testifying expert in the following cases:

- 2010 – *Major European national laboratory v. major international telecommunications equipment manufacturer*. Quackenbush acted as fact witness, wrote initial and rebuttal report and testified in front of the arbitrator. Arbitration results were decided early in 2011 and the side Quackenbush represented prevailed in all issues.
- 2007 – *Witness Systems, Inc. v. NICE Systems, Ltd. and NICE Systems, Inc.* (District: N.D. Ga.; Judge: Batten, J.). Expert for Fish and Richardson, council for Witness Systems (Nick Setty lead attorney) in patent infringement claim involving speech analytics patent - word spotting, talk over analysis and emotion detection - tried to verdict in 2008; jury returned verdict of infringement in favor of Witness Systems and awarded damages. Quackenbush wrote more than 125 pages of expert report and had more than 6 hours of testimony over the course of a 5-day trial. Council felt that Quackenbush testimony was critical for case outcome.
- 2006 – *Lucent Tech. v. Microsoft Corp.* (No. 08-1485, Southern District of California) Quackenbush acted as fact witness, wrote an expert report and gave expert testimony in a 6-hour deposition.

I have acted as non-testifying expert for numerous law firms where my responsibilities ranged from valuing patents, researching patent prior art as a means of establishing invalidity and creating patent claim charts.

Engineering Services

Development – I have consulted to both large companies and start-ups, developing audio processing systems or audio compression systems for the client. Typically, all applications are required to run on the target platform (client or server) in real time.

Subjective Quality Evaluation – I have participated in all aspects of numerous formal subjective quality tests, including test design, test administration and test data analysis and report generation. These tests range from presentation via stereo headphones to ITU-T 5.1 channel loudspeaker configurations and from listening in a sound booth to listening in a major motion picture sound mixing theatre.

Standardization - I have participated in International Standards Organization (ISO) Motion Pictures Expert Group (MPEG) standardization activities since 1995 and served as Chair of the Audio Subgroup of the since 1998. In addition, I have participated in 3GPP, ATSC and SCTE standardization work.

ARL Products - At Audio Research Labs I specified, developed and marketed two products. They are STEP (Subjective Training and Evaluation Program), which is an PC-based application facilitates subjective quality evaluation of multi-channel audio material and ARL Sound Stage, which is a Digidesign Pro Tools TDM plug-in for mixing of mult-channel audio material from “dry” sound sources using perceptual and spatial audio principles.

1998 - **Chair, MPEG Audio Subgroup**, ISO/IEC Standards Organization

As Chair of the International Standards Organization Motion Picture Experts Group (ISO/MPEG) Audio subgroup, I am responsible for setting and executing the agenda for current work and developing a vision for future work. The Audio subgroup consists of more than 50 audio experts and specific responsibilities including delegating tasks to and managing task completion by the group, forging consensus on group decisions and

reporting on the group's work to the MPEG plenary. 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)

Professional Experience:

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 was 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, where 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 collaboration. High-bandwidth server hubs at three geographic locations provided more than 250 simultaneous interactive "jam rooms" with 4 "jammers" per room, each of whom experienced less than 50 ms round-trip audio latency. Any given "jam session" could be broadcast live to up to up to 1500 listeners. Audio jam "archives" could be edited into songs that could be downloaded or posted on a user's home page.

For Lightspeed Audio Labs, real-time means LIVE!

2000 - 2002 **Acting Supervisor, Speech and Audio Coding Group**, AT&T Laboratories
Florham Park, NJ
Responsible for directing and mentoring the work of technical staff members.

1996 - 2000 **Principal Technical Staff Member**
Speech and Audio Research Department, AT&T Laboratories
Specific projects and duties outlined below.
Florham Park, NJ

1986 - 1996 **Member of Technical Staff**
Signal Processing Research Department, AT&T Bell Laboratories
Specific projects and duties outlined below.
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 insure 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 council, 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 the 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 had responsibility 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 rate of 160 kbps. The entire DAB system had numerous successful trials broadcasting in the FM band.

- 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

1975 - 1978 Test Engineer
Loral Electronics, Yonkers, NJ

Professional Memberships:

Fellow, Audio Engineering Society (AES)
Senior Member, International Electrical and Electronics Engineers (IEEE)

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, Volume 12, Issue 4, Oct.-Dec. 2005 Page(s):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.
4. 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.
5. 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: 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 Book*, Chiariglione, L. (Ed.), Springer, New York, 2011.
2. 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.
3. 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.
4. Johnston, J. D., Quackenbush, S., Davidson, G., Brandenburg, K. and Herre, J., "Wavelet, subband and block transforms" in *Communications and Multimedia*, Akansu, A. N.and Medley, M. J. (Ed.), Kluwer, April 1998.
5. 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.
6. Herre, J., Johnston, J. D., Brandenburg, K., Quackenbush, S. and 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.
7. 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.
8. 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. Quackenbush, S. and Lefebvre, R., "Performance of MPEG Unified Speech and Audio Coding," AES 131st Convention, October 20-23, New York, USA.
2. 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), Sep. 29 – Oct 1, 2011, Pohang, Republic of Korea.
3. 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.
4. Quackenbush, S.R, Driessen, P.F.; "Error Mitigation in MPEG-4 Audio Packet Communication Systems," 115th AES Conventon, Sep. 2003, Preprint 5981.

5. Shyh-shiaw Kuo; Johnston, J.D.; Turin, W.; Quackenbush, S.R.; "Covert audio watermarking using perceptually tuned signal independent multiband phase modulation," IEEE International Conf. Acoustics, Speech and Signal Proc. (ICASSP '02). IEEE International Conference on, Volume 2, 2002 Page(s):1753 - 1756
6. Lacy, J., Quackenbush, S. R., Reigman, 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, Wash., May 1998.
7. Quackenbush, S., "Coding of Natural Audio in MPEG-4," IEEE International Conf. Acoustics, Speech and Signal Proc. (ICASSP 98), Seattle, Wash., May 1998.
8. Quackenbush, S.R.; Johnston, J.D.; "Noiseless coding of quantized spectral components in MPEG-2 Advanced Audio Coding," IEEE 1997Workshop Applications of Signal Processing to Audio and Acoustics, 1997 on 19-22 Oct. 1997 Page(s):4 pp.
9. 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.
10. Quackenbush, S. R. and Parikh, V. N., "Using C++ for real-time signal processing," Proc. 1995 IEEE ICSPAT, Boston, Oct. 1995.
11. 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.
12. Quackenbush, S. R., "A 7kHz bandwidth 32 kpbs speech coder for ISDN," Proc. 1991 ICASSP '91, Toronto, Canada, May 1991.
13. Quackenbush, S. R., "Hardward implementation of a color image decoder for remote database access," Proc. ICASSP '90, Albuquerque, NM, April 1990, pp. 985-088.
14. Quackenbush, S. R., Ordentlich, E. and Snyder, J., "Hardward implementation of a 128 kbps monophonic audio coder," 1989 IEEE ASSP Workshop on Appl. Sig. Proc. Audio & Acoust., New Paltz, NY, Oct. 1989.
15. Quackenbush, S. R., "Hardware implementation of a 16 Kbps subband coder using vector quantization," Proc. IEEE ICASSP, New York, NY, Apr 1988, pp. 386-389.
16. 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.
17. 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.

Patents:

1. 8,041,038, System and method for decompressing and making publically available received media content
2. 7,802,101, System and method of retrieving a watermark within a signal
3. 7,725,808, System and method for representing compressed information
4. 7,529,941, System and method of retrieving a watermark within a signal
5. 7,492,902, Custom character-coding compression for encoding and watermarking media content
6. 7,451,319, System and method of watermarking a signal
7. 7,353,447, System and method for representing compressed information
8. 7,146,503, System and method of watermarking signal
9. 7,131,007, System and method of retrieving a watermark within a signal
10. 7,076,426, Advance TTS for facial animation
11. 7,042,933, System and methods for transmitting data
12. 6,885,749, Scrambling a compression-coded signal
13. 6,850,559, System and methods for transmitting data
14. 6,760,443, Custom character-coding compression for encoding and watermarking media content
15. 6,718,507, System and method for representing compressed information
16. 6,704,576, Method and system for communicating multimedia content in a unicast, multicast, simulcast or broadcast environment
17. 6,493,457, Electronic watermarking in the compressed domain utilizing perceptual coding

18. 6,341,165, Coding and decoding of audio signals by using intensity stereo and prediction processes
19. 6,266,419, Custom character-coding compression for encoding and watermarking media content
20. 5,825,976, Device and method for efficient utilization of allocated transmission medium bandwidth
21. 5,463,641, Tailored error protection
22. Canadian Patent 2,260,222, Coding and Decoding of Audio Signals by Using Intensity Stereo and Prediction