Guillaume Fuchs

Professional Experiences

2017-present **Chief Scientist**
4 years Fraunhofer-Institut für Integrierte Schaltungen IIS (Erlangen, Germany)
Immersive spatial audio coding and processing for communication. Multi-channel processing, Ambisonics, Audio object coding and rendering, Binaural rendering.
Project management: Deep Learning approaches for speech coding, synthesis and enhancement. Generative neural networks.

2006-2016 **Senior Scientist**
10 years Fraunhofer-Institut für Integrierte Schaltungen IIS (Erlangen, Germany)
Speech and audio coding, Speech analysis. Contribution in the development of different standards: 3GPP Enhanced Voice Services (EVS), MPEG-H 3D Audio, MPEG-D Unified Speech and Audio Coding (USAC)

2003-2006 **Scientific Collaborator**
3 years VoiceAge (Montréal, Canada)
Music and speech coding for communication, hierarchical coding based on ITU-T G.729 and ITU-T G.722.2. Candidature for ITU-T G.729.1.

Nov. 2001-Dec. 2002 **R&D Engineer**
18 months Canon Research Centre France (Cesson-Sévigné, France)
New quantization techniques for image compression using evolutionary computation.

Feb. – Aug. 2001 **Master internship**
6 months French National Institute for Research in Computer and Control INRIA (Rennes, France)
Joint source-channel coding in image transmission over noisy channel: Unequal Error Protection codes for JPEG200, cycling error-correcting codes for wavelet transforms.

June – Sep. 2000 **Undergraduate internship**
4 months Valeo Électronique (Meung-Sur-Loire, France)
Industrial computer network administration.

Education

2006 **Ph.D. Degree**
University of Sherbrooke (Québec, Canada)
Electrical and Computer Engineering
"Hierarchical audio and speech coding at low bit-rates"

2001 **Master of Science Degree** (DEA)
University of Rennes (France)
Signal Processing and Telecommunications (image option)

2001 **Electrical Engineering Diploma**
INSA (National Institute of Applied Sciences) of Rennes (France)

Languages

English	Fluent in speaking, reading and writing
German	Fluent in speaking, good commands in reading and writing
French	Mother tongue

Teaching Activities

2018 - 2018	**Lecturer** Friedrich-Alexander University of Erlangen – Master course *Speech Coding*
2015 - 2017	**Lecturer** University of Applied Sciences, Hochschule Fulda – Undergraduate course *Theory of information for multimedia signals*
2013 - 2014	**Invited lecturer** Friedrich-Alexander University of Erlangen – Master lecture *Variable bit-rate speech coding*
2005-2006	**Lecturer** University of Sherbrooke – Undergraduate courses and labs *Digital Signal Processing*
March 2003	**Training Workshop Instructor** Centre Sectoriel de Formation en Électronique (Sousse, Tunisia) *Computer Networking, Signal Processing*

Awards

- **Joseph von Fraunhofer prize** for the contribution to the standard 3GPP EVS.
- **Best scientific paper** of the 132nd Audio Engineering Society (AES) Convention.
- **Bonus of scientific excellence** from Fraunhofer Society for the contribution to the standard ISO/MPEG-D USAC.
- **Scholarship of the Canadian Government** for my PhD studies.

Skills

Digital Signal Processing	*Speech and Audio analysis and coding, Speech/Music classification, speech enhancement, time-frequency analysis, filtering, quantization, image processing and compression, channel coding: linear block and convolutional codes*
Acoustic	*Stereo and multichannel processing, Psychoacoustic, perceptual modelling for audio coding*
Machine Learning	*GMMs, SVM, LDA, kmeans, Deep Learning (CNN, RNN, GAN)*
Probability and statistics	*Entropy and Information theory*
Mathematical optimization	*Constrained optimization, Heuristic and stochastic algorithms, Genetic Algorithms*
Programming	***C,** C++**, Matlab, Python** (numpy, scipy, matplotlib, pytorch), shell script, Java, UML*
Project Management	*Project definition, Project planning, conflict management, moderation*
Intellectual Property	*Author and co-author of over **40 US and European patents**.*

Publication list

Articles in Refereed Journals

1. R. Bleidt and al., “**Development of the MPEG-H TV Audio System for ATSC 3.0**”, submitted in IEEE Transactions on Broadcasting, November 2016.
2. M. Neuendorf, M. Multrus, N. Rettelbach, G. Fuchs, J. Robilliard, J. Lecomte, S. Wilde, S. Bayer, S. Disch, C. Helmrich, R. Lefebvre, P. Gournay, B. Bessette, J. Lapierre, K. Kjörling, H. Purnhagen, L. Villemoes, W. Oomen, E. Schuijers, K. Kikuiri, T. Chinen, T. Norimatsu, C. K. Seng, E. Oh, M. Kim, S. Quackenbush, and B. Grill: **“The ISO/MPEG unified speech and audio coding standard – consistent high”**, *Journal J. Audio Eng. Soc*, December 2013, number 1169.

Book Chapters

3. G. Fuchs “**Relaxed Code-Excited Linear Prediction (RCELP)**”, in *Speech Coding: with Code-Excited Linear Prediction*, Editor : Tom Bäckström, Springer, 2017.

4. M. Multrus, M. Neuendorf, J. Lecomte, R. Geiger, G. Fuchs, S. Bayer, J. Robilliard, F. Nagel, S. Wilde, D. Fisher, J. Hilpert, N. Rettelbach, C. Helmrich, S. Disch, R. Geiger and B. Grill: “**MPEG Unified Speech and Audio Coding – Bridging the Gap** ”, in *Microelectronic Systems : Circuits, Systems and Applications*, Editors : Albert Heuberger, Günter Elst, Randolf Hanke, Springer, december 2011.

Peer Reviewed Papers in Conference Proceedings

5. A Mustafa, N Pia, G Fuchs, “**StyleMelGAN: An Efficient High-Fidelity Adversarial Vocoder with Temporal Adaptive Normalization**” *ICASSP 2021*
6. Das Sneha, Bäckström Tom and Fuchs Guillaume "**Fundamental Frequency Model for Postfiltering at Low Bitrates in a Transform-Domain Speech and Audio Codec**." *Interspeech 2020*
7. Korse Srikanth, Gupta Kishan and Fuchs Guillaume "**Enhancement of Coded Speech Using a Mask-Based Post-Filter**" *ICASSP 2020*
8. Fuchs Guillaume, Ashour Chamran and Bäckström Tom "**Super-Wideband Spectral Envelope Modeling for Speech Coding**" *Interspeech 2019*
9. Korse Srikanth, Fuchs Guillaume and Bäckström Tom "**GMM-Based Iterative Entropy Coding for Spectral Envelopes of Speech and Audio.**" *ICASSP 2018*
10. Guillaume Fuchs, Anthony Lombard, Emmanuel Ravelli and Martin Dietz: **“A comfort noise addition post-processor for enhancing low bit-rate speech coding in noisy environments”**, *3rd IEEE Global Conference on Signal and Information Processing*, December 2015
11. Guillaume Fuchs: **“A robust speech/music discriminator for switched audio coding”**, *European Signal Processing Conference (EUSIPCO)*, August 2015
12. Takehiro Moriya, Yutaka Kamamoto, Noboru Harada, Tom Bäckström, Christian Helmrich and Guillaume Fuchs: **“Harmonic Model for MDCT based Audio Coding with LPC Envelope”**, *European Signal Processing Conference (EUSIPCO)*, August 2015
13. A. Lombard, S. Wilde, E. Ravelli, S. Dohla, G. Fuchs and M. Dietz: **“Frequency-domain Comfort Noise Generation for Discontinuous Transmission in EVS”**, in *IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP)*,April 2015
14. G. Fuchs, C.R. Helmrich, G. Markovic, M. Neusinger, E. Ravelli and T. Moriya**, “Low delay LPC and MDCT-based audio coding in the EVS codec”**, in *IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP)*, April 2015
15. E. Ravelli, C.R. Helmrich, G. Fuchs and M. Multrus, **“Low-complexity and robust coding mode decision in the EVS coder”**, in *IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP)*, April 2015
16. Guillaume Fuchs, **“Embedded Voronoi codes for successive refinement lattice vector quantization”**, in *IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP)*, 2013
17. M. Neuendorf, M. Multrus, N. Rettelbach, G. Fuchs, J. Robilliard, J. Lecomte, S. Wilde, S. Bayer, S. Disch, C. Helmrich, R. Lefebvre, P. Gournay, B. Bessette, J. Lapierre, K. Kjörling, H. Purnhagen, L. Villemoes, W. Oomen, E. Schuijers, K. Kikuiri, T. Chinen, T. Norimatsu, C. Kok Seng, E. Oh, M. Kim, S. Quackenbush, B. Grill, **“MPEG Unifed Speech and Audio Coding - The ISO/MPEG Standard for High-Efficiency Audio Coding of all Content Types”**, *132nd Audio Engineering Society Convention*, Budapest, Hungary, April 26-29, 2012
18. Guillaume Fuchs, Vignesh Subbaraman, Markus Multrus, “**Efficient context adaptive entropy coding for real-time applications**”, *IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP)*, April 2011

19. Guillaume Fuchs, Markus Multrus, Max Neuendorf and Ralf Geiger, "**MDCT-Based Coder for Highly Adaptive Speech and Audio Coding**", *17th European Signal Processing Conference (EUSIPCO)*, August 2009
20. E. Bernd, S. Disch, S. Bayer, G. Fuchs, R. Geiger, "**A Time-Warped MDCT Approach to Speech Transform Coding**", *AES 126th Convention*, May 2009
21. M. Neuendorf, P. Gournay, M. Multrus, J. Lecomte, B. Bessette, R. Geiger, S. Bayer, G. Fuchs, J. Hilpert, N. Rettelbach, F. Nagel, J. Robilliard, R. Salami, G. Schuller, R. Lefebvre and B. Grill, "**A Novel Scheme for Low Bitrate Unified Speech and Audio Coding – MPEG RM0**", *AES 126th Convention*, May 2009
22. M. Neuendorf, P. Gournay, M. Multrus, J. Lecomte, B. Bessette, R. Geiger, S. Bayer, G. Fuchs, J. Hilpert, N. Rettelbach, R. Salami, G. Schuller, R. Lefebvre and B. Grill: "**Unified Speech and Audio Coding Scheme for High Quality at Low Bitrates**", in *IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP)*, April 2009
23. Guillaume Fuchs and Roch Lefebvre, "**A Scalable CELP/Transform Coder for low bit rate Speech and Audio coding**", *AES 120th Convention*, May 2006
24. Guillaume Fuchs and Roch Lefebvre, "**A New Post-Filtering for Artificially Replicated High-band in Speech Coders**", *in IEEE International Conference on Acoustics, Speech, and Signal Processing (ICASSP)*, May 2006
25. Guillaume Fuchs and Roch Lefbvre, "**A Speech Coder Post-Processor Controlled by Side Information**", *in IEEE International Conference on Acoustics, Speech, and Signal Processing (ICASSP)*, May 2005