CURRICULUM VITAE

Prof. Sharon Gannot

August 13, 2023

Personal Data

First Name:	Sharon
Surname:	Gannot
Date of birth:	August 10, 1964
Place of birth:	Hadera, Israel
Nationality:	Israeli
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Education

PhD, 2000	Institute:	Department of Electrical Engineering-Systems, Faculty of Engineering, Tel Aviv University.
	Supervisors:	**David Burshtein and Ehud Weinstein**
	Subject:	Array Processing of Nonstationary Signals with Application to Speech
MSc, 1995	Institute:	Department of Electrical Engineering-Systems, Faculty of Engineering, Tel Aviv University.
	Supervisor:	**Ehud Weinstein**
	Subject:	Algorithms for single microphone speech enhancement
	Distinction:	**Cum Laude**
BSc, 1986	Institute:	Faculty of Engineering, Technion - Israel Institute of Technology.
	Studies:	Electrical Engineering.
	Distinction:	**Summa Cum Laude**.

Academic Appointments

2018-2019	Professor (part-time) at the Technical Faculty of IT and Design Department of Architecture, Design and Media Technology, Aalborg University, Denmark.
2014-	Full Professor at the Faculty of Electrical Engineering, Bar-Ilan University, Ramat-Gan, Israel.
2010–2014	Associate Professor at the Faculty of Electrical Engineering, Bar-Ilan University, Ramat-Gan, Israel.
2006–2010	Senior Lecturer at the School of Electrical Engineering, Bar-Ilan University, Ramat-Gan, Israel.

2004–2006	Lecturer at the School of Electrical Engineering, Bar-Ilan University, Ramat-Gan, Israel.
2004–2011	Adjunct Lecturer at the Faculty of Electrical Engineering, Technion - IIT, Haifa, Israel.
2002–2003	Research Fellow at the Faculty of Electrical Engineering, Technion - IIT, Haifa, Israel.
2001	Post-Doctoral position at the Department of Electrical Engineering (ESAT), Katholieke Universiteit, Leuven, Belgium.
1994–2000	Teaching and research assistant at the Department of Electrical Engineering–Systems, Faculty of Engineering, Tel-Aviv University, Tel-Aviv, Israel.

Other Professional Experience

1994–2015	Consultant to the Israeli Defence forces and the Israeli Ministry of Defense in the area of speech processing.
1987–1993	Israeli Defense Forces. Head of a Research and Development section in the area of Telecommunication, Signal Processing and Speech Processing. Retired with the rank of Major.

Membership in Professional Societies

1. Fellow of the IEEE.
2. Member of the International Speech Communication Association (ISCA).
3. Member of the European Association for Signal Processing (EURASIP).

Research Interests

Statistical signal processing, array processing and machine learning for audio signal processing, specifically speech enhancement, noise reduction, audio source separation and extraction, dereverberation, echo cancellation, and sound source localization and tracking.
Applications range from a single device equipped with single- or multi-microphones to more complex structures as ad hoc networks of multiple devices with audio capabilities and binaural hearing aids and hearables.
In my research, I develop and apply methods from various mathematical disciplines:

1. Data-driven methods, e.g., manifold learning and deep learning, variational auto-encoders.
2. Bayesian, e.g., variational-Bayes, Kalman and Wiener filtering, particle filtering, and non-Bayesian. e.g. recursive and distributed expectation-maximization.
3. Distributed algorithms for wireless ad hoc microphone networks.
4. Performance bounds.

Teaching Experience

Lecturer:

Array Processing	Graduate;	Technical University Aalborg, Denmark; 2018
Array Processing	Graduate;	Technical University Liberec, The Czech Republic; 2018

Array Processing	Graduate;	Friedrich-Alexander-Universität Erlangen-Nürnberg, Germany; 2017
Array Processing	Graduate;	Bar-Ilan University; 2011–
Speech Processing	Graduate;	Bar-Ilan University; 2009–
Statistical Signal Processing I	Undergraduate;	Bar-Ilan University; 2004–
Advanced Lab in Signal Proc.	Undergraduate;	Bar-Ilan University; 2005–
Digital Signal Processing II	Undergraduate;	Bar-Ilan University; 2005–
Speech Processing	Ministry of Defense; 2006	
Signals and Systems	Undergraduate;	Bar-Ilan University; 2004–2007
Linear Systems	Undergraduate;	Bar-Ilan University; 2004
Introduction to Signal Proc.	Undergraduate;	Technion-IIT; 2002–2003
Advanced Stochastic Sig. Pro.	Graduate;	Technion-IIT; 2003
Introduction to Sig. Analysis	Undergraduate;	Tel-Aviv University; 1999–2000

Teaching Assistant:

Introduction to Signal Proc.	Undergraduate;	Tel-Aviv University; 1994–1998
Intro. to Statistical Sig. Proc.	Undergraduate;	Tel-Aviv University; 1994–1998
Random Processes	Graduate;	Tel-Aviv University; 1994
Semiconductor Devices	Undergraduate;	Technion-IIT; 1986

Awards and Distinctions

2022	The **EURASIP Group Technical Achievement Award** for contributions to theory and practice of microphone array signal processing and statistical learning in speech enhancement through extensive activities of his research group.
2021	**IEEE Fellow** for contributions to acoustical modeling and statistical learning in speech enhancement.

Research Awards:

2018	The Rector of Bar-Ilan University **Research Innovation award** for the year 2018.

Teaching Awards:

2014	Bar-Ilan University **Outstanding lecturer award**.
2010	Bar-Ilan University **Outstanding lecturer award**.
2001	Tel-Aviv University, Faculty of Engineering, **Outstanding teacher award**.

Paper Awards:

2022	Y. Hu and S. Gannot, “Closed-form single source direction-of-arrival estimator using first-order relative harmonic coefficients,” in IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP), 2022, **best paper award**.
2020	M. J. Bianco, P. Gerstoft, J. Traer, E. Ozanich, M. A. Roch, S. Gannot, and C.-A. Deledalle, “Machine learning in acoustics: Theory and applications,” The Journal of Acoustical Society of America, Vol. 146, No.5, Nov. 2019. https://doi.org/10.1121/1.5133944, **The Technical Area Pick for Signal Processing of JASA, 2020.**
2018	A. Adler, O. Schwartz, and S. Gannot, ”A weighted multichannel Wiener filter and its decomposition to LCMV beamformer and post-filter for source separation and noise reduction,” in International conference on the science of electrical engineering (ICSEE), Eilat, Israel, Dec. 2018, **best paper award**.

2017	O. Shwartz, A. Plinge, E. Habets, and S. Gannot, "Blind microphone geometry calibration using one reverberant speech event," in IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA), New Paltz, USA, Oct., 2017, **best paper award**.
2016	S. E. Chazan, S. Gannot, and J. Goldberger, "A phoneme-based pre-training approach for deep neural network with application to speech enhancement," in International Workshop on Acoustic Signal Enhancement (IWAENC), Xián, China, Sep. 2016, **best student paper award**.
2015	D. Kounades-Bastian, L. Girin, X. Alameda-Pineda, S. Gannot, and R. P. Horaud, "A variational EM algorithm for the separation of moving sound sources," in IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA), New Paltz, USA, Oct. 2015, **best student paper award**.
2015	E.A.P. Habets, S. Gannot, and I. Cohen, "Late reverberant spectral variance estimation based on a statistical model," IEEE Signal Processing Letters, vol. 16, no. 9, pp. 770-773, Sep. 2009. **The 2014 Signal Processing Society - Signal Processing Letters Best Paper Award**.
2014	Y. Dorfan, G. Hazan, and S. Gannot, "Multiple acoustic sources localization using distributed Expectation-Maximization algorithm," in The 4th Joint Workshop on Hands-free Speech Communication and Microphone Arrays (HSCMA), Nancy, France, May 2014, **best student paper award**.
2012	Markovich-Golan, S. Gannot, and I. Cohen, "A weighted multichannel Wiener filter for multiple sources scenarios," in The IEEE 27th Convention of IEEE Israel (IEEEI), Eilat, Israel, Nov. 2012, **best student paper award**.
2010	L. Ehrenberg, S. Gannot, A. Leshem, and E. Zehavi, "Sensitivity analysis of MVDR and MPDR beamformers," in The 26th Convention of IEEE Israel (IEEEI), Eilat, Israel, Nov. 2010, pp. 416-420, best student paper award.
2010	S. Markovich-Golan, S. Gannot, and I. Cohen, "A reduced bandwidth binaural MVDR beamformer," in The International Workshop on Acoustic Echo and Noise Control (IWAENC), Tel-Aviv, Israel, Aug. 2010, **best student paper award**.
2003	T. Dvorkind and S. Gannot, "Speaker localization exploiting spatial-temporal information," in The International Workshop on Acoustic Echo and Noise Control (IWAENC), Kyoto, Japan, Sep. 2003, pp. 295–298, **distinguished paper**.
2003	S. Gannot and M. Moonen, "On the application of the unscented Kalman filter to speech processing," in The International Workshop on Acoustic Echo and Noise Control (IWAENC), Kyoto, Japan, Sep. 2003, pp. 8–11, **distinguished paper**.

Student Awards:

1999	**Excellence scholarship** of the student's Dean in Tel-Aviv University, in memory of Okrein Eliezer and Marko Shaul.
1997	1997-8 **Intel-Dean award** for Excellency.
1996	**Wolf prize and scholarship** for PhD students.
1983–1985	President of the Technion annual **Excellence Award** (3 times).

As an Industrial Researcher:

1995	The Israeli Defense Forces (IDF), General Staff, Head of Intelligence Branch, **Creativity award** in memory of Col. Uzi Yairi.

Research Grants

1. **Mask-based and All-DNN Beamformers**, *Meta Reality Lab*, 2022, 183,750 US$.

2. **Audience: Audio-Visual Analysis and Separation**, *The Council for Higher Education (CHE), The Planning and Budgeting Committee (PBC), Data Science Program*, 2022-2024, with Ethan Fetaya and Jacob Goldberger, 2,934,000 ILS.

3. **Speech Enhancement with Moving Arrays**, *Facebook Reality Lab*, 2021, 175,625 US$.

4. **Audio-Visual Speaker Separation**, *General Motors*, 2021-2022, with Ethan Fetaya, 350,500 ILS.

5. **Audio-visual referring expressions**, *Bar-Ilan University, Data Science Institute*, Mar. 2020, with Gal Chechik, 86,000 ILS.

6. **Audio processing algorithms in adverse conditions**, *"Magneton" - Israel Innovation Authority, with CEVA Ltd.*, 2020-2022, 700,000 ILS.

7. **Socially Pertinent Robots in Gerontological Healthcare - SPRING**, *H2020 consortium*, 2020-2023. Total amount 8.4M € for eight groups, 1.04M € for BIU. https://spring-h2020.eu/

8. **Combined Neural Interface and Deep Learning Methods for Multi-Microphone Assisted Listening and Selective Attention Devices**, *Ministry of Science*, 2020-2022, with Elana Zion-Golumbic Jacob Goldberger. Total amount 1,798,000 ILS, 600,000 to S. Gannot.

9. **Environment-Aware Data-Driven Acoustic Signal Processing**, *"Kamin" - Israel Innovation Authority*, Dec., 2017. 1st year: Total amount 641,840 ILS (BIU 321,840 ILS); 2nd year: Total amount 659,040 ILS (BIU 329,040 ILS).

10. **Institutional Equipment**, *Israeli Science Foundation (ISF)*, 2018, with Gal Chechik and Yoav Goldberg, BIU, 924,000 ILS.

11. **Audio processing algorithms for human-machine interface in mobile and stationary devices**, *"Magneton" - Israel Innovation Authority, with CEVA Ltd.*, Aug. 2017. 1st year: 361,400 ILS; 2nd: 365,600 ILS.

12. **Single- and Multi-microphone Deep Learning Methods for Improved Speech Enhancement**, *Intel* grant, Dec. 2016, with Jacob Goldberger, BIU. Amount: 50,000USD.

13. **Advanced Deep Learning Methods for Improved Speech Enhancement**, *Starkey Hearing Aids* grant, Oct., 2016, with Jacob Goldberger, BIU. Amount: 60,000USD.

14. **Acoustic environment detection**, *MAFAAT* grant (Israel Ministry of Defense), May 2016, with Ronen Talmon, Technion. Amount: 100,000 ILS.

15. **Single microphone speech enhancement using deep neural networks**, *MAFAAT* grant (Israel Ministry of Defense), Feb. 2016, with Jacob Goldberger, BIU. Amount: 1st year 100,000 ILS; 2nd year 100,000 ILS.

16. **Distributed speaker localization and separation**, *MAFAAT* grant (Israel Ministry of Defense), Feb. 2016-2017. Amount: 1st year 100,000 ILS; 2nd year 100,000 ILS.

17. **Acoustic Scene Aware Speech Enhancement for Binaural Hearing Aids (ATHENA)**, *Joint Lower-Saxony Israeli* research grant, 2015-2018. Amount: 298,200€ (124,900€ for BIU).

18. **Distributed Microphone Arrays for Personal Devices**, *Intel* research grant, 2014-2017. Amount: 350,000 US$.

19. **Distributed Microphone Arrays for Personal Devices**, *Intel* research grant, 2015 (2nd year). Amount: 100,000 US$.
20. **Distributed Microphone Arrays for Personal Devices**, *Intel* research grant, 2014. Amount: 100,000 US$.
21. **Supervised Speaker Tracking Using Diffusion Kernel Combined with Extended Particle Filter**, *Ministry of Science* research grant for women in science, 2014. Amount: 20,000 ILS.
22. **Distributed Microphone Arrays**, *MAFAAT* research grant, 2014. Amount: 150,000 ILS for the first year and 175,000 ILS for the second year.
23. **3-D Audio Rendering**, *Orbit* research grant, 2014. Amount: 48,000 ILS.
24. **Two Microphone Noise Reduction in Adverse Conditions**, *Cardo* research grant, 2013. Amount: 225,000 ILS.
25. **Binaural Speech Dereverberation**, *German-Israeli Foundation (GIF)*, 2013-2015, with E.A.P. Habets and S. Doclo. Amount: 198,500 €.
26. **Keyboard Noise Reduction**, *Waves* research grant, 2012. Amount: 165,000 ILS.
27. **Robust ASR in Reverberant Environment**, *Samsung* research grant, 2012-2013. Amount: 390,000 ILS.
28. **Robust ASR in Car Environment**, *General Motors* research grant, 2012-2013. Amount: 290,000 ILS.
29. **Beamforming Using Flat Microphones**, *MAFAAT* grant (Israel Ministry of Defense) 2011-2013. Amount: 62,500 ILS for the first year and 62,500 ILS for the second year.
30. **Differential Microphone Arrays**, *Rubidium* grant for undergraduate project, 2011. Amount: 15,000 ILS.
31. **Hearing Aids**, *Qualcomm* research grant, 2011. Amount: 140,000$.
32. **Speech Separation Using Single- and Multi-Microphone processing**, *Magneton* grant, Aug., 2010, together with *Nice Systems Ltd.* Amount: 473,000 ILS for the first year. 409,000 ILS for the second year.
33. **Speech processing using Microflowns**, *MAFAAT* grant (Israel Ministry of Defense), Feb. 2010. Amount: 250,000 ILS for first year, 220,000 ILS for second year, 150,000 ILS for third year.
34. **Echo Cancellation for Improving Speech Recognizers**, *Magneton* grant, Aug. 2008, together with *Nice Systems*. Amount: 500,000 ILS for the first year. 500,000 ILS for the second year.
35. **Speaker Localization in Noisy and Reverberant Environment**, *MAFAAT* grant (Israel Ministry of Defense), Jul. 2008. Together with Israel Cohen. Amount: 280,000 ILS for the first year, 150,000 ILS for the second year, 150,000 ILS for the third year.
36. **Acoustic Array Signal Processing in Adverse Environments**, the state-of-Israel *Higher-Education Council post-doctoral* grant for Emanuël A.P. Habets (host laboratory).
37. Funded undergraduate project (Hanan Ashwega and Nir Russo), **Speech Source Localization in Noisy and Reverberant Environment using the Particle Filter**, 2007. Amount: 30,000 ILS.
38. Funded undergraduate project (Arieh Jerichover and Ariel Bierendorf), **Residual Echo Cancellation**, 2007. Amount: 30,000 ILS.

39. **Blind Speaker separation in Adverse Conditions**, *MAFAAT* grant (Israel Ministry of Defense), Dec., 2005, together with Israel Cohen. Amount: 275,000 ILS for first year, 285,000 ILS for second year, 295,000 ILS for third year.

40. **Optical Realization of Viterbi Decoding Algorithm and Trellis Diagrams via All-Optical Solid Free Space Switches**, *Bar-Ilan* grant for research with industrial potential, Sep., 2004. together with Zeev Zalevsky. Amount: 55,000$.

41. **Multi-Input-Multi-Output (MIMO) Communication Systems: Channel Models, Equalizers and Noise Suppression**, *MAGNET Consortium*, Ministry of Industry and Commerce, State of Israel, 2004, together with Ephraim Zehavi and Amir Leshem. Amount: 160,000 ILS for the first year; 300,000 ILS for the second year; 300,000 ILS for the third year; 200,000 ILS for the fourth year; 225,000 ILS for the fifth year; 150,000 ILS for the sixth year.

42. *Philips* grant for **post-doctoral position** in K.U.Leuven, Belgium 2001. Principal investigators: Prof. Marc Moonen, Sharon Gannot and Dr. Koen Eneman. Amount: 25,000€.

Professional Activities

Conference Chairing:

1. General Co-Chair, **Interspeech**, to be held in Summer 2024, Jerusalem, Israel.
2. Co-organizer, **Data Science and Learning Workshop: Unraveling the Brain**, A satellite workshop of ICASSP, Jun. 4, 2023, Rhodes Island, Greece.
3. General Co-Chair, **the 29th IEEE Israel conference - International Symposium on Speech and Audio Processing**, Nov. 2016, Eilat, Israel.
4. General Co-Chair, **the IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA)**, Mohonk Mountain House, New Paltz, NY, October 2013.
5. General Co-Chair, **the International Workshop on Acoustic Echo and Noise Control (IWAENC)**, Aug.-Sep. 2010, Tel Aviv, Israel.
6. Chair, **the International workshop LVA/ICA - Audio Day**, Bar-Ilan University, Mar. 2012.
7. Co-chair, **the Bar-Ilan Workshop on Signal Processing**, Bar-Ilan University, Israel, Jan., 2011.

Conference Organization:

1. Special session co-chair, **the European Signal Processing Conference (EUSIPCO)**, Amsterdam, The Netherlands, Aug. 2020.
2. Area chair, **the IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA)**, New-Paltz, NY, USA, Oct. 2019.
3. Technical Co-Chair, **the international conference on Latent Variable Analysis and Signal Separation, LVA/ICA**, University of Surrey, Guildford, UK, July 2018.
4. Area chair, **the European Signal Processing Conference (EUSIPCO)**, Nice, France, Aug. 2015.
5. Technical committee member, **Reverb Challenge**, organized by IEEE Signal Processing Society, **Audio and Acoustics Signal Processing Technical committee**, May, 2014.
6. Area chair, the European Signal Processing Conference (EUSIPCO), Marrakech, Morocco, Sep. 2013.

7. Member of the Technical Committee and Coordinator of the Best Student Paper Award, **the 25th, 26th, 27th, 28th 29th IEEE Israel conference**, 2008–2016, Eilat, Israel.
8. Member of the Technical committee of the **3rd European DSP Education and Research Symposium (EDERS)**, Jun. 2008, Tel Aviv, Israel.

Special Sessions:

1. Organizer (together with Emanuël A.P. Habets), **Acoustic Signal Processing Using Machine Learning**, IEEE International Workshop on Machine Learning for Signal Processing (MLSP), Aalto University, Espoo, Finland, Sep. 2020.
2. Organizer (together with Walter Kellermann), **Signal Processing and Machine Learning Methods for Acoustic Sensor Networks**, IEEE International Workshop on Computational Advances in Multi-Sensor Adaptive Processing (CAMSAP), Guadeloupe, West Indies, Dec. 2019.
3. Organizer (together with Herbert Buchner), **Acoustic scene analysis and tracking for time-varying reverberant environments**, IEEE International Conference on Audio and Acoustic Signal Processing (ICASSP), Brighton, UK, May 2019.
4. Organizer (together with Peter Willett), **Speaker localization in dynamic real-life environments**, IEEE International Conference on Audio and Acoustic Signal Processing (ICASSP), New-Orleans, LA, USA, Mar. 2017.
5. Organizer (together with Antoine Deleforge), **special session on Learning-based Sound Source Localization and Spatial Information Retrieval**, IEEE International Conference on Audio and Acoustic Signal Processing (ICASSP), Shanghai, China, Mar. 2016.
6. Organizer (together with Zbyněk Koldovký), **special session on Estimation and Modeling of Relative Transfer Functions between Microphones in Noisy Environments**, the European Signal Processing Conference (EUSIPCO), Nice, France, Aug. 2015.
7. Organizer (together with Afsaneh Asaei), **special session on Sparse and Low-rank Modelling for Acoustic Signal Processing**, LVA-ICA Workshop, Liberec, The Czech Republic, Aug. 2015.
8. Organizer, **special session on Advances in Multi-Microphone Speech Separation and Noise Reduction**, the European Signal Processing Conference (EUSIPCO), Marrakech, Morocco, Sep. 2013.
9. Organizer and Chair, **special session on microphone array processing**, the 27th IEEE Israel conference, Dec., 2012.
10. Organizer and Chair **special session on speech enhancement**, the 25th IEEE Israel conference, Dec., 2012.

Technical Committees:

1. IEEE Signal Processing Society, member of **Data Science Initiative**, since 1.5.2019; Chair 2022-2023.
2. IEEE Signal Processing Society, member of **Education Center Editorial Board**, since 1.1.2022.
3. IEEE Signal Processing Society, member of **Education Board Committee**, 1.1.2020-31.12.2023.
4. Latent Variable Analysis and Signal Separation (LVA/ICA), member of the **Steering Committee**, since Oct. 2019;
5. IEEE Signal Processing Society, member of **Conference Board Committee**, since 1.1.2019; member of the **Conferences Board Executive Subcommittee**, since 1.1.2020.

6. IEEE Signal Processing Society, **Audio and Acoustics Signal Processing Technical Committee**: Member 2010-2015; Chair of EDICS subcommittee 2011-2013; Vice-Chair 2016; Chair 2017-2018; Past Chair 2019.

7. European Acoustics Association (EAA), member of the **Audio Signal Processing** Technical Committee, since 1.1.2018.

8. EURASIP, member of **Audio, Speech and Music (ASMSP) Special Area Team (SAT)**, 2016-2021, **organizer of new member and chair elections**, 2017 and 2018.

9. Member of the Technical and Steering committee, **the International Workshop on Acoustic Echo and Noise Control (IWAENC)**, since 2005.

Associate Editor:

1. Member of the Senior Editorial Board of IEEE Signal processing Magazine, 2020–

2. Moderator, **Arxiv, Electrical Engineering and Systems Science - Audio and Speech Processing**, 2016–2021

3. Senior area chair, **IEEE Transactions on Audio, Speech and Language Processing**, 2013–2017, 2021–

4. Associate editor **IEEE Transactions on Audio, Speech and Language Processing**, 2009–2013.

5. Associate editor **EURASIP journal on Advances in Signal Processing**, 2004–2012.

6. Associate editor, **Springer Handbook of Speech Processing and Speech Communication**, 2006.

Guest Editor of Special Issues in Journals:

1. Sharon Gannot, Walter Kellermann, Zbyněk Koldovský, Shoko Araki and Gaël Special Issue on “Model-based and Data-Driven Audio Signal Processing,” Call open (due Jan. 2025).

2. Sharon Gannot, Walter Kellermann, Peter Willett and Martin Haardt, Special Issue on “Acoustic source localization and tracking in dynamic real-life scenes,” Journal of Selected Topics in Signal Processing, Mar. 2019.

3. Keisuke Kinoshita, Sharon Gannot, Armin Sehr, Emanuël Habets, Walter Kellermann, and Reinhold Haeb-Umbach, Special Issue on “Silencing the echoes – Processing of Reverberant Speech,” EURASIP Journal on Advances in Signal Processing, Dec., 2015.

4. Hervé Bourlard, Afsaneh Asaei, Tara N. Sainath and Sharon Gannot, Special Issue on “Advances in Sparse Modeling and Low-rank Modeling for Speech Processing,” ELSEVIER Speech Communication, Dec., 2015.

5. Alexander Bertrand, Simon Doclo, Sharon Gannot, Nobutaka Ono, Toon van Waterschoot, Special issue on “Wireless Acoustic Sensor Networks & Ad Hoc Microphone Arrays,” ELSEVIER Signal Processing, February, 2014.

6. S. Nordholm, T. Abhayapala, S. Doclo, S. Gannot, P. Naylor, and K. Tashev, Special issue on “Microphone Array Speech Processing,” EURASIP Journal on Advances in Signal Processing, vol. 2010, Jul. 2010.

7. P. Loizou, I. Cohen, S. Gannot, and K. Paliwal, Special issue on “Speech Enhancement,” ELSEVIER Speech Communication, vol. 49, no. 7-8, pp. 527–529, 2007.

8. S. Gannot, J. Benesty, J. Bitzer, I. Cohen, S. Doclo, R. Martin, and S. Nordholm, Special issue on "Advances in Multimicrophone Speech Processing," EURASIP Journal on Applied Signal Processing, vol. 12, p. 1, Apr. 2006.

Reviewer for Journals and Conferences:

Conferences — IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP); European Signal Processing Conference (EUSIPCO); IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA); International Workshop on Acoustic Echo and Noise Control (IWAENC); Interspeech; IEEE Workshop on Statistical Signal Processing (SSP); IEEE Workshop on Sensor Array and Multichannel Signal Processing; IEEE Workshop on Machine Learning for Signal Processing; IEEE International Workshop on Computational Advances in Multi-Sensor Adaptive Processing (CAMSAP); IEEE GlobalSip.

Journals — Proceedings of the IEEE; IEEE Transactions on Signal Processing; IEEE/ACM Transactions on Speech and Audio and Language Processing; IEEE Signal Processing Letters; IEEE Signal Processing Magazine; Journal of Acoustical Society of America; EURASIP Journal of Advances on Signal Processing; ELSEVIER Signal Processing; ELSEVIER Speech Communication; IEEE Transactions on Circuits & Systems II; IEEE Transactions on Neural Networks; IEEE Transactions on Systems, Man, and Cybernetic; IEE Proc. Vision, Image & Signal Processing.

PhD and MSc Reviewer:

International

1. PhD Reviewer, **Spatial dissection of a soundfield using spherical harmonic decomposition**, Abdullah Fahim, Australian National University, Australia, Nov. 2019.
2. Member of the jury of PhD defense of Andreas Koutrouvelis on **Multi-Microphone Noise Reduction for Hearing Assistive Devices**, TU Delft, The Netherlands, Dec. 2018.
3. Member of the jury of PhD defense of Simon Leglaive on **Mixing Models For Multichannel Audio Source Separation In Reverberant Environments**, Telecom PaeisTech, Paris, France, Dec. 2017.
4. Member of the jury of PhD defense of Ante Jukić on **Sparse Multi-channel Linear Prediction for Blind Speech Dereverberation**, University of Oldenburg, Germany, Oct. 2017.
5. PhD Reviewer, **Localization and Tracking of Acoustic Sources in Room Environment**, Wu Kai, Nanyang Technological University, Singapore, Feb. 2017.
6. PhD Reviewer, **Robust Multichannel Microphone Beamforming**, Craig Anderson, Victoria University of Wellington), New-Zealand, Jan. 2016.
7. Member of the jury of PhD defense of Yuan Zeng on **Distributed Speech Enhancement in Wireless Acoustic Sensor Networks**, TUDelft, The Netherlands, Jun. 2015.
8. Member of the jury of PhD defense of Joseph Szurley on **Distributed Signal Processing Algorithms for AcousticSensor Networks**, K.U.Leuven, Belgium, May 2015.
9. Member of the jury of PhD defense of Klaus Josef Reindl, **Multichannel Acoustic Signal Extraction for Reverberant Environments**, Friedrich-Alexander-Universität Erlangen-Nürnberg, Germany, Mar. 2015.
10. PhD Reviewer, **Multichannel Equalization Applied to Speech Dereverberatio**, Rajan Sonhana Rashobh, Nanyang Technological University, Singapore, Aug. 2014.

11. Member of the jury of PhD defense of Karim Helwani on **Adaptive identification of acoustic multichannel systems using sparse representations**, T.U. Berlin, Germany, Feb., 2014.
12. Member of the jury of PhD defense of Jose Manuel Gil-Cacho on **Adaptive filtering algorithms for acoustic echo cancellation and acoustic feedback control in speech communication applications**, K.U.Leuven, Belgium, Dec., 2013.
13. PhD Reviewer, **Adaptive System Identification and Equalization Algorithms for Acoustic Echo Cancellation and Speech Dereverberation**, Liao Lei, Nanyang Technological University, Singapore, Mar. 2013.
14. Habilitation reviewer, **Nonlinear, Time-Varying, and Blind Acoustic System Identification**, Gerlad Enzner, Ruhr University, Bochum, Germany, May 2012.
15. PhD Reviewer, **Acoustical Time-Reversal Signal Processing: New Developments and Applications**, Nguyen Dinh Quy, Nanyang Technological University, Singapore, Sep. 2010.
16. Member of the jury of PhD defense of Alexander Bertrand on **Signal Processing Algorithms for Wireless Acoustic Sensor Networks**, K.U.Leuven, Belgium, May, 2011.
17. Referee, PhD thesis, **Speech enhancement using microphone arrays**, S. Y. Low, Curtin University, Australia, Sep. 2005.
18. Member of the jury of PhD defense of Koen Eneman on **'Subband and Frequency-Domain Adaptive Filtering Techniques for Speech Enhancement in Hands–Free Communication**, K.U.Leuven, Belgium, 2002.

Israel

1. MSc Hodaya Halevi, Mar. 2021.
2. MSc Nadav Rachimi, Jan. 2021.
3. MSc Nadav Yazdi, Jan. 2021.
4. PhD Proposal Marina Eini, Jan. 2021.
5. MSc Carmi Shimon, Feb. 2020.
6. MSc Ben Fishman, Mar. 2020.
7. PhD Yaakov Bucris, Nov. 2019.
8. MSc Gilad Vered, Sep. 2019.
9. MSc Ido Shabtai, Aug. 2019.
10. MSc Ido Binyamini, Aug. 2019.
11. MSc David Cohen, June 2018.
12. MSc, Lior Fritz, Mar. 2018.
13. MSc, Izhak Zimmermann, Music Department, Bar-Ilan, Sep. 2017.
14. MSc, Ori Katz, Technion, Aug. 2017.
15. PhD, Vladimir Tourbabin, Ben-Gurion, Apr. 2017.
16. MSc Oren Rosen, Technion, Jan. 2017.
17. PhD proposal, Tal Schnizer, Technion, Dec. 2016.
18. MSc, Yoav Biederman, Ben-Gurion, Dec. 2016.
19. MSc, Kfir Aberman, Technion, Dec. 2016.
20. PhD defense, Yaakov Bucris, Nov. 2016.
21. MSc, Reuven Barkon, Technion, Mar. 2016.

22. PhD, Hadas Benisty, Technion, Jan. 2016
23. MSc, Eliav Benesty, Technion, Nov. 2015.
24. MSc, Tamir Tapuchi, Ben-Gurion, Nov. 2014.
25. PhD, Yotam Peled, Ben-Gurion, Feb. 2014.
26. MSc thesis on **Localization of Speakers in a Highly Reverberant Enclosures with a Spherical Microphone Array**, Or Nadiri, Ben-Gurion University, Dec. 2013.
27. MSc thesis on **Multisensory speech enhancement in noisy environments using bone-conducted and air-conducted microphones**, Mingzi Li, the Technion–IIT, Nov. 2013.
28. MSc thesis on **Multimodal audio inpainting**, Yuval Bahat, the Technion–IIT, Nov. 2013.
29. PhD propsal, David Alon, Ben-Gurion, Jan. 2013.
30. MSc thesis on **MMSE-based speech enhancement using the harmonic model**, Yair Stark, Ben-Gurion University, Sep. 2012.
31. MSc thesis on **Direction-of-Arrival estimation of reflections by spherical array processing of room impulse responses**, Nejem Huleihel, Ben-Gurion University, Sep. 2012.
32. MSc thesis on **Packet lost concealment based on the GAPES Algorithm**, Yacov(Yaki) Umflat, Ben-Gurion University, Sep. 2012.
33. MSc thesis on **Information Theoretic Pairwise Clustering**, Avishay Friedman, Bar-Ilan University, Sep. 2012.
34. PhD thesis on **Speech Perception**, Eitan Globerson, Bar-Ilan University, Feb. 2012.
35. MSc thesis on **Speech Bandwidth Extension**, Itai Katsir, the Technion–IIT, Feb. 2012.
36. MSc thesis on **A specral approach to inter-carrier interference mitigation in OFDM systems**, Avi Septimus, Bar-Ilan University, Feb. 2012.
37. MSc thesis on **Speech Diarization and Verification**, Oren Tadmor, Ben-Gurion University, Apr. 2012.
38. MSc thesis on **Microphone Arrays–Design Criteria**, Vladimir Tourbabin, Ben-Gurion University, Nov. 2011.
39. MSc thesis on **Localization of Radio Sources**, Noy Cohen, Tel-Aviv University, Nov. 2011.
40. MSc thesis on **Beamforming for a Spherical-Aperture Microphone**, Morag Agmon, Ben-Gurion University, Dec. 2010.
41. MSc thesis on **Speech Recognition**, Roee Lahav, Ben-Gurion University, Dec. 2010.
42. MSc thesis on **'Bayesian Focusing Methods in Beamforming**, Yaakov Buchris, the Technion–IIT, Sep. 2010.
43. MSc thesis on **Dominant Source Detection**, Ilana Volfin, the Technion, Nov. 2011.
44. PhD thesis on **Approximation and Reconstruction problems under Affine Constraints**, Gur Benjamin Solomon, Tel-Aviv University, Apr. 2010.
45. PhD defense on **Speaker Recognition Systems of Reduced Complexity**, Avi Matsa at Tel-Aviv University, July. 2011.

46. MSc thesis on **Smoothing Techniques for High-Resolution Direction-of-Arrival Estimation of Coherent Signals Using Spherical Arrays**, Dmitry Khaykin, Ben-Gurion University, Nov. 2009.
47. MSc thesis on **Maximum A-Posteriori Probability Multiple Pitch Tracking Using the Harmonic Model**, Amitai Koretz, en-Gurion University, Nov. 2009.
48. MSc thesis on **New Methods for Speech Recognition**, Roee Lahav, Ben-Gurion University, Dec. 2010.
49. MSc thesis on **Speaker Recognition in Reverberant Environment**, Itai Peer, Ben-Gurion University, Aug. 2009.
50. MSc thesis on **Adaptive Stereo Acoustic Echo Cancelation in reverberant environments**, Amos Schreibman, the Technion–IIT, Oct. 2009.
51. MSc thesis on **Time Difference of Arrival Estimation in Multi-path Environmen**, Ity Erlich Tel-Aviv University, Mar. 2009.
52. MSc thesis on **Packet Loss Concealment for Voice Applications**, Yishai Gil, Ben-Gurion University, Feb. 2009.
53. MSc thesis on **Time-Varying Perceptual Linear Prediction for Speech Application**, Oron Gamliel, Ben-Gurion University, Feb. 2009.
54. PhD Dissertation on **System Identification in the Short-Time Fourier Transform Domain**, Yekutiel Avargel, the Technion, Nov. 2008.
55. MSc thesis on **Improvements and Generalization of the Support Vector Machine Re-Scoring Algorithm of Continuous Hidden Markov Models**, Amir Alfandary, Tel-Aviv University, Jan. 2008.
56. MSc thesis on **Direct Localization of Cyclo-Stationary Sources**, Alit Mendelsson-Reuven, Tel-Aviv University, Jan., 2008.
57. MSc thesis on **MIMO-AR blind source separation for GMM-distributed and finite alphabet signals**, Tirza Ruttenberg at Ben-Gurion University, Jan. 2008.
58. MSc thesis on **Analysis of Lombard Effect's influence on Automatic Speaker Verification Systems and Methods of Compensation**, Roman Goldenberg, Ben-Gurion University, Oct. 2005.
59. MSc thesis on **Support Vector Machine Training for Improved Hidden Markov Modeling**, Alba Sloin, Tel-Aviv University, Oct. 2006.
60. MSc thesis on **MIMO Decision Directed Channel Estimation for Dynamic Channels**, Ory Eger, Tel-Aviv University, May 2006.
61. Member of PhD candidate committee, **Signal Localization**, Alon Amar, Tel-Aviv University, Sep. 2005
62. MSc thesis on **Bootstrap Kalman Filter: A Hybrid Scheme for Bilinear State-Space Models**, Yuval Yosef Domb, Tel-Aviv University, 2005.
63. MSc thesis on **Perceptual Time-Varying Modelling of Speech Signals for ASR and Compression Application**, Ben-Gurion University, 2005.
64. MSc thesis on **Time-Frequency Representation for Speech Recognition**, Ben-Gurion University, 2005.
65. MSc thesis on **Time-Frequency Representation of Nonstationary Signals Using NAR Autocorrelation**, Ben-Gurion University, 2005.
66. MSc thesis on **On the Resolution of Overlapping Echoes of a Random Signal**, Tel-Aviv University, 2005.

67. MSc thesis on **A Fast Converging Scheme for Echo Cancellation**, Tel-Aviv University, 2004.
68. MSc thesis on **Asymptotically Optimal Blind Separation of Parametric Gaussian Sources"**, Tel-Aviv University, 2004.
69. MSc thesis on **Automatic Modulation Classification of MPSK Modulated Signals in Fading Channels**, Tel-Aviv University, 2002.
70. MSc thesis on **Phoneme-Based Speaker Verification with Selection of Adaptation and Scoring Modes of Gaussian Mixture Models**, Tel-Aviv University, 2002.
71. MSc thesis on **Driving Speakers by Filter Bank Generated Equalizing Signal"**, the Technion–IIT, 2002.
72. MSc thesis on **Asymptotically Optimal Blind Separation of Parametric Gaussian Sources**, Tel-Aviv University, 2004.

Tutorials and Keynote Addresses

1. Keynote address, **Multi-Microphone Speaker Localization on Manifolds**, The 2022 IEEE International Conference on Signal Processing, Communications and Computing (ICSPCC 2022), Xi'an, Shaanxi, China (Virtual), 25 October 2022.
2. Tutorial, **Distributed Speech Processing Algorithms for Ad Hoc Microphone Arrays and Wireless Acoustic Sensor Networks**, Seasonal School on "Distributed Signal Processing and Optimization," Imperial College London, UK, 19-23 September 2022.
3. Talk, **Acoustic Vector Sensors: DOA Estimation, Beamforming and Applications**, China Computer Federation, January 6, 2021.
4. Tutorial (together with Bracha Laufer-Goldshtein and Ronen Talmon), **Multi-Microphone Speaker Localization and Tracking on Manifolds**, the European Signal Processing Conference (EUSIPCO), A Coruña, Spain, Sep. 2019.
5. Keynote address **Multi-Microphone Speaker Localization on Manifolds**, the Audio Analysis Workshop, Aalborg, Denmark, Aug. 2018.
6. Keynote address, **Multi-Microphone Speaker Localization and Tracking on Manifolds**, the 13th ITG Conference on Speech Communication, Oldenburg, Germany, October 10-12, 2018.
7. Keynote address, **Multi-Microphone Speaker Localization on Manifolds: Achievements and Challenges**, the International Conference on Latent Variable Analysis and Independent Component Analysis LVA/ICA, Grenoble France, Februray, 2018.
8. Tutorial (together with Dr. Alexander Bertrand), **Introduction to Distributed Speech Enhancement Algorithms for Ad Hoc Microphone Arrays & Wireless Acoustic Sensor Networks**, European Signal Processing Conference (EUSIPCO), Marrakesh, Morocco, Sep. 2013.
9. Tutorial (together with Prof. E.A.P. Habets), **Linear and Parametric Microphone Array Processing**, the International Conference on Acoustics, Speech and Signal Processing (ICASSP), Vancouver, Canada, May 2013.
10. Keynote address, **Multi-Microphone Speech Enhancement Using LCMV Beamformers**, the International Workshop on Acoustic Signal Enhancement (IWAENC), Aachen, Germany, Sep. 2012.

11. Tutorial (together with Prof. Israel Cohen, Prof, E.A.P. Habets and Prof. Ronen Talmon), **Speech Enhancement for Acoustic Communication using Multiple Microphones and Diffusion Maps**, European Signal Processing Conference (EUSIPCO), Bucharest, Romania, Aug. 2012.
12. Tutorial (together with Prof. Israel Cohen and Prof. Ronen Talmon), **Speech Modeling and Enhancement Using Diffusion Maps**, the International Conference on Acoustics, Speech and Signal Processing (ICASSP), Kyoto, Japan, May 2012.
13. Tutorial (with Prof. Israel Cohen), **Speech Modeling and Enhancement in Nonstationary Noise Environment**, the IASTED International Conference on Signal and Image Processing and Applications, Crete, June 2011.

Invited Talks

Oct. 2022	Keynote Talk, **Multi-Microphone Speaker Localization on Manifolds**, The 11th IEEE International Conference on Signal Processing, Communications and computing (IEEE ICSPCC).
Sep. 2022	Keynote Talk, **Distributed Speech Processing Algorithms for Ad Hoc Microphone Arrays and Wireless Acoustic Sensor Networks**, SOUNDS Seasonal School, Imperial College, London, UK.
Jan. 2021	Keynote Talk, **Acoustic Vector Sensors: DOA Estimation, Beamforming and Applications**, China Computer Federation.
Apr. 2020	**Multi-Microphone Speaker Localization on Manifolds**, CS Colloquium, Hebrew University Jerusalem, Israel.
Jan. 2019	**Speech Dereverberation using EM Algorithm and Kalman Filtering**, Aalborg University, Denmark.
Sep. 2018.	**Multi-Microphone Speaker Localization on Manifolds**, Riken AIP, Japan.
Sep. 2018	**Speech Dereverberation using EM Algorithm and Kalman Filtering**, Univeristy of Toulouse, France.
Jul. 2018	**Multi-Microphone Speaker Localization on Manifolds**, Imperial College, London.
Dec. 2017	**Speech Dereverberation using EM Algorithm and Kalman Filtering**, Telecom ParisTech, Paris, France.
Oct. 2017	**Speech Enhancement using a Deep Mixture of Experts**, Jones Hopkins, Baltimore, MD, USA.
Sep. 2017	**Multi-Microphone Speaker Localization on Manifolds**, INRIA, Rôhne-Alpes, Grenoble, France.
Jul. 2017	**Multi-Microphone Speaker Localization on Manifolds**, Friedrich-Alexander-Universität, Erlangen-Nürnberg, Germany.
Feb. 2016	**Expectation-Maximization (EM) Framework for Multiple Speaker Localization and Tracking**, INRIA, Rôhne-Alpes, Grenoble, France.
Oct. 2015	**Multi-Microphone Speech Enhancement: Theory & Applications** , University of Maryland, Baltimore County (UMBC), MD, USA.
Jun. 2015	**Expectation-Maximization (EM) Framework for Multiple Speaker Localization and Tracking**, KULeuven, Belgium.
Jun. 2015	**Expectation-Maximization (EM) Framework for Multiple Speaker Localization and Tracking**, TU Delft, The Netherlands.
Jun. 2014	**Expectation-Maximization (EM) Framework for Multiple Speaker Localization and Tracking**, The 3rd Annual Underwater Acoustics Symposium, Tel-Aviv University.
Apr. 2014	**Multiple Speaker Localization and Tracking**, Paderborn Univesity, Germany.
Apr. 2014	**Microphone Array Processing**, TU Dortmund, Germany.

Apr. 2014	**Microphone Array Processing**, International Audio Labs, Fraunhufer Institute, Erlangen, Germany.
Feb. 2014	**Microphone Array Processing**, INRIA, Rôhne-Alpes, Grenoble, France.
Jan. 2013	**Multi-Microphone Speech Enhancement Centralized and Distributed Beamformers**, TU Berlin, Germany.
Jan. 2013	**Sounds of Silence? Speech Enhancement with Microphone Arrays**, Electrical Engineering Colloquium, the Technion–IIT.
Sep. 2012	**Multi-Microphone Speech Enhancement Using LCMV Beamformers**, International Audio Labs, Fraunhufer Institute, Erlangen, Germany.
April 2012	**Multichannel Eigenspace Beamforming in a Reverberant Noisy Environment with Multiple Interfering Speech Signals**, NTT Labs, Kyoto, Japan.
Oct. 2011	**Multichannel Eigenspace Beamforming in a Reverberant Noisy Environment with Multiple Interfering Speech Signals**, RWTH Aachen, Germany.
Oct. 2011	**Multichannel Eigenspace Beamforming in a Reverberant Noisy Environment with Multiple Interfering Speech Signals**, Speech Processing workshop, University of Oldenburg, Germany.
May 2011	**Multichannel Eigenspace Beamforming in a Reverberant Noisy Environment with Multiple Interfering Speech Signals**, K.U.Leuven, Belgium.
Sep. 2009	**Multichannel Eigenspace Beamforming in a Reverberant Noisy Environment with Multiple Interfering Speech Signals**, Imperial College, London, UK.
May 2012	**Multiple Hypothesis Speaker Localization Using Particle Filters**, Tel-Aviv University Seminar.
Jun. 2011	**A Reduced Bandwidth Binaural MVDR Beamformer**, The 10th anniversary Tel-Aviv University DSP labs.
Jan. 2011	**Beamforming Algorithms for Speech Enhancement and Speaker Separation**, Israeli Signal Processing Workshop, Bar-Ilan University.
Oct. 2008	**Adaptive Beamforming and Post-filtering**, Hewlett-Packard Research Laboratories, California, USA.
Feb. 2008	**Multi-Microphone Speech Dereverberation using Eigen-decomposition**, the Technion–IIT DSP Seminar.
Jun. 2007	**Speaker Localization and Tracking**, Katholieke Universiteit, Leuven, Belgium.
Feb. 2007	**Speaker Localization Using the Unscented Kalman Filter**, Workshop on "Speech Enhancement and Multichannel Audio Processing", CCIT, the Technion–IIT.
May 2004	**On the Application of the Unscented Kalman Filter to Speech Processing**, Tel-Aviv University Seminar.
Jun. 2003	**Speech Enhancement Based on the General Transfer Function GSC and Postfiltering**, IBM Israel seminar on "Speech processing", Haifa, Israel.
Jun. 2001	**Noise Reduction and Dereverberation algorithms for Speech Communications and Voice-Controlled Systems**, the IEEE Benelux Signal Processing Chapter workshop on "Adaptive Signal Processing Systems", K.U.Leuven, Belgium,.
May, 1997	**Signal Separation**, IEEE Israel workshop on "Topics in Signal Processing with Application in Bio-Medicine", Tel-Aviv University, Israel.
Sep. 1994	**Single microphone speech enhancement**, MIT, Cambridge, Massachusetts, USA.

Short-Term Research Visits

Sep.-Oct., 2014	Visiting professor, Signal Processing Research Department, Starkey Hearing Technology.
Mar.-May, 2014	Visiting professor, International Audio labs, Fraunhofer IIS and Friedrich-Alexander University, Erlangen-Nuremberg, Germany.
2013-2019	Multiple short-term stays at INRIA Grenoble, Rhône-Alps, France.

Jul.-Aug., 2013	Tan Chin Tuan visiting professor, Nanyang Technological University, Singapore.
Apr., 2013	Institute of Information Technology and Electronics, Liberec, the Czech Republic. Funded by Project ESF CZ.1.07/2.2.00/28.0050.
Jan. 2013	TU Berlin, Germany.
Sep., 2012	International Audio labs, Fraunhofer IIS and Friedrich-Alexander University, Erlangen-Nuremberg, Germany.
Sep., 2009	Imperial College, London, UK.

Graduate Students

Current:

PhD

1. Aviad Eisenberg. **DNN-based method for Speaker Separation and Extraction**, Start: Oct. 2021.
2. Yochai Yemini, **Deep Learning Methods for Audio-Visual Scene Analysis**, Co-supervisor Ethan Fetaya, Start: Oct. 2019.
3. Renana Opochinsky (Kleinman). **Deep Learning Methods for Human-Robotic Interaction**, Start: Apr. 2018.
4. Yarden Menashri. **A Comparison of Available Methods and a Construction of a New Model for Quantifying Physiological Group Synchrony**, Main supervisor Ilanit Gordon, Start: Aug. 2019.

MSc

1. Efraim Yanir. **Audio Processing using Diffusion Processes**, joint supervision with David Burshtein (Tel-Aviv University), start Jun. 2023.
2. Rina Weller. **Audio Processing**, start Mar. 2023.
3. Sagi Dela Torre. **Analysis of Room Acoustics**, start Oct. 2022.
4. Ohad Cohen. **Audio-Visual Emotion Recognition**, Co-supervisor: Gal Chechik, start Oct. 2022.
5. Idan Cohen. **Acoustic Scene Localization and Mapping**, Co-supervisor: Ofir Lindenbaum, start Oct. 2021.
6. Daniel Levi. **Graphical Convolutional Neural Networks for Speech Processing**, start Oct. 2021.
7. Adi Cohen. **All Neural Network Microphone Array Processing**, start Oct. 2021.
8. Amit Eliav. **Audio-Visual Acoustic Activity Detection**, start Oct. 2021.
9. Mordehay Moradi. **Single microphone speaker separation using DNN**, Start Jan. 2021.
10. Boris Rubenchik. **Low-latency single microphone speaker separation using DNN**, Start Oct. 2020.
11. Amit Sofer. **Graph neural networks for robust beamforming**, Start Oct. 2020.
12. Roi Gueta. **Binaural speech enhancement controlled by EEG signals**, Start Oct. 2020.
13. Oren Shmaryahu. **On beam-patterns in reverberant environments**, Start Apr. 2020.
14. Ayal Schwartz. **Deep Learning Methods for Beamforming**, Start Oct. 2019.
15. Yosef Soussana, **Bayesian Speaker Localization**. Start: Feb. 2017.

Former:

Post-Doctorate

1. Shlomi Chazan, **Deep Learning Methods for Speech Enhancement**, Oct. 2021–Sep. 2023.
2. Yonggang Hu, **Speech Processing Using Spherical Microphone Arrays**, Oct. 2022–Mar. 2023.
3. Elior Hadad, **Speaker Localization and Separation**. Apr. 2017–Oct. 2019.
4. Shmulik Markovich-Golan, **Algorithms for Speech Processing**. Mar. 2013–Oct. 2020.
5. David Levin, **Distributed Algorithms for Microphone Arrays**. May 2016–April 2017.
6. Emanuël A.P. Habets, **Speech Processing**. Feb. 2007–Jan. 2009.

PhD

1. Shlomi Chazan, **Deep Learning Methods for Speech Enhancement**, Co-supervisor: Jacob Goldberger, Oct. 2015-Mar. 2021.
2. Yaron Laufer, **Bayesian Methods in Speech Processing**, Oct. 2017–Oct. 2020.
3. Bracha Laufer, **Manifold Learning Methods for Speech Processing** (direct track), Co-supervisor: Ronen Talmon, Technion, Oct. 2017–Oct. 2020.
4. Dani Cherkassky, **Robust Speech Processing using Ad-Hoc Microphone Arrays**.
Graduation: Feb. 2019.
5. Yuval Dorfan, **Distributed Localization and Tracking of Acoustic Sources**.
Graduation: Sep. 2018.
6. Boaz Schwartz, **Derverberation Methods for Binaural Hearing** (direct track).
Co-supervisor: E.A.P. Habets (FAU, Erlangen-Nuremberg, Germany).
Graduation: May 2018.
7. Ofer Schwartz, **Multi-microphone Derverberation Algorithms**.
Co-supervisor: E.A.P. Habets (FAU, Erlangen-Nuremberg, Germany).
Graduation: Mar. 2018.
8. David Levin, **Speech processing using Acoustic Vector Sensors** (direct track).
Co-supervisor: E.A.P. Habets (FAU, Erlangen-Nuremberg, Germany). Graduation: Mar. 2018.
9. Elior Hadad, **Speech Processing for Hearing Aids**. Graduation: Jun. 2016.
10. Shmulik Markovich-Golan, **Speech Processing using Distributed Microphone Networks**.
Co-supervisor: Israel Cohen (Technion-IIT). Graduation: Aug. 2013.
11. Ronen Talmon, **Supervised Speech Processing Based on Geometric Analysis**.
Co-supervisor: Israel Cohen (Technion–IIT). Graduation: July, 2011.
12. Emanuël A.P. Habets, **Single- and Multi-Microphone Speech Dereverberation using Spectral Enhancement**.
Supervisor: J.W.M. Bergmans (T.U. Eindhoven), co-supervisor: P.C.W. Sommen (T.U. Eindhoven). Graduation Jun., 2007.

MSc (Thesis)

1. Dalia Sherman. **Emotion recognition from audio signals**, Graduation: Apr. 2023.
2. Avital Bross. **Multi-Speaker Localization and Tracking based on Manifold Learning and Clustering**, Graduation: Nov. 2021.

3. Aviad Eisenberg. **Blind Acoustic Source Separation Algorithms Based On Statistical Models**, Graduation Jun. 2021.
4. Nili Cohen, **An Expectation-Maximization for Speech Separation, Echo Cancellation and Dereverberation**, Graduation: Jun. 2021.
5. Koby Wiseberg, **Simultaneous tracking and separation of multiple sources using factor graph model**, Graduation: Oct. 2020.
6. Hodaya Hammer. **Deep Learning Methods for Speech Enhancement and Speaker Localization**, Graduation: Sep. 2020.
7. Maya Veisman, **Simultaneous Room Geometry Inference and Speaker Localization**. Graduation: Jan. 2020.
8. Ori Ernest, **Speech Enhancement using Generative Adversarial Network**. Graduation: May 2019.
9. Natan David, **Room Classification from Reverberant Speech using Relative Transfer Function**. Graduation: May 2019.
10. Anna Barnov, **Microphone Array Processing on Multiple Devices**, Start: Feb. 2014. Graduation: Mar. 2018.
11. Tamar Marom-Shalev, **Wireless Acoustic Sensor Networks: Combined Acoustic Echo Cancellation and Adaptive Beamforming**. Graduation: Nov. 2017.
12. Yossi Daniel (Open University, Israel), **Speech Localization using Microphone Arrays through Sparse Recovery Approach**,
Co-supervisor: Hagit Messer (Tel-Aviv University, Israel). Graduation: Aug. 2016.
13. Shlomi Chazan, **Deep Learning Methods for Speech Enhancement**.
Co-supervisor: Jacob Goldberger, Graduation: Oct. 2015.
14. Boaz Castro, **Speech Dereverberation using Subspace Methods**. Graduation: Oct. 2015.
15. Yochay Yeminy, **Single Microphone Speech Separation**.
Co-supervisors: Yossi Keller. Graduation: Oct., 2011.
16. Ofer Schwartz, **Concurrent Speech Localization**.
Graduation: Oct. 2012.
17. Livnat Ehrenberg, **Performance Bounds on MIMO tracking systems**.
Co-supervisor: Amir Leshem. Graduation: Aug. 2011.
18. Avinoam Levi, **Speaker Localization using Particle filters**.
Graduation: May 2011.
19. Shmulik Markovich, **Multichannel Eigenspace Beamforming in a Reverberant Noisy Environment with Multiple Interfering Speech Signals**.
Co-supervisor: Israel Cohen (Technion–IIT). Graduation: Oct. 2008.
20. Gal Reuven, **Dual Transfer Function Generalized Sidelobe Canceller and Application to Joint Noise Reduction and Echo Cancellation**.
Co-supervisor: Israel Cohen (Technion–IIT). Graduation: Jan. 2006.
21. Tsvi G. Dvorkind, **Speaker Localization in a Reverberant and Noisy Environment**.
Graduation **Summa Cum Laude**, Nov. 2003.

MSc (Project)

1. Zahi Elbaz, **Bayesian Methods for Speech Enhancement in dynamic and Reverberant Enviroments**, Co-supervisor: Yaron Laufer, Graduation: Jun. 2020.
2. Itai Druker, **A Bayesian Framework for Blind Adaptive Beamforming**, Co-supervisor: Yaron Laufer, Graduation: Apr. 2020.

3. Aviel Adler, **A Weighted Multichannel Wiener Filter and its Decomposition to LCMV Beamformer and Postfilter for Source Separation and Noise Reduction**. Co-supervisor: Ofer Schwartz, Graduation: Nov. 2017.

Undergraduate Projects

Current:

Bar-Ilan Uni.

1. Noam Bar-On, **Single microphone speech enhancement with multi-output DNN**, 2021.
2. Amit Ben-Simon, **Blind audio source separation based on statistical model**, 2021.
3. Aviv Doron and Oren Chizkya, **Online blind audio source separation**, 2021.
4. Tomer Ron and Asaf Skital, **Voice activity detector based on LRT**, 2021.
5. Daniel Levi and Adi Cohen, **Reverberation time estimation using DNN**, 2021.
6. Maor Yevadiev and Omer Pinchas, **Wind noise suppression using spectral methods**, 2021.
7. Seffi Reshef, **cGMM-based TF clustering for LCMV beamforming**, 2021.
8. Hila Yehezkel and Avi Goldfisher, **Speaker separation using TasNet**, 2021.
9. Avigail Kolobov and Gal Bental, **Deep ranking-based sound-source localization and tracking**, 2021.
10. Yuval Saraf and Sara Eliasev, **Audio selection using DNN**, 2021.
11. Amit Eliav and Aaron Taub, **Singing voice and music synchronization**, 2021.

Former:

Bar-Ilan Uni.

1. Amit Sofer, **Robust beamforming on manifolds**, 2020.
2. Afek Steinberg and Ofek Ofir, **Blind source separation in noisy environment using model-based EM algorithm**, 2020.
3. Orel Shoshani and Yanir Edry, **Deep Ranking-Based Sound Source Localization**, 2020.
4. Amit Pinchas and Yogev Klein, **Speech Separation with Utterance-Level Permutation Invariant Training of Deep Recurrent Neural Networks**, 2020.
5. Idan Uri and Mor Zecharia, **Intelligibility improvement in very low SNR**, 2020.
6. Ravid Avraham and Liroi Dokhanian, **Directed loudspeaker array**, 2020.
7. Orel Navi and Kfir Cohen, **Muting a spatial area using loudspeaker array**, 2020.
8. Benaya Levy and Mordechai Muradi, **Learning to Separate Object Sounds by Watching Unlabeled Video**, 2020.
9. Noam Korengut and Sara Ernest, **Video-based stereo spatialization from mono recording**, 2020.
10. Yevgeny Ivanov and Shani Dagan, **Fabrication and Characterization of Graphene Microphone Arrays**, with Prof. Doron Naveh, 2019.
11. Shoval Rahamim and Sharon Meged, **DNN-based Audio Center**, 2019.

12. Meir Rosenblat and Hanan Aharonov, **Speaker Separation**, 2019.
13. Yaakov Rahimy and Azaria Mashraky, **Wind Noise Reduction**, 2019.
14. Ayal Schwartz and Adi Wassermann, **Source separation using microphone array and DNN**, 2019.
15. Liron Steinberg and Dalia Sherman, **Source separation and localization using model-based EM algorithm**, 2019.
16. Dvir Hazut and David Radushiyzky, **CNN-based Dereberberation**, 2019.
17. Nevo Geslevich and Tomer Levanon, **Audio Center**, 2019.
18. Guy Jisfan and Daniel Shalev, **Emotion recognition using artificial neural networks**, 2018.
19. Roy Tendlerr and Or Gershon, **Bandwidth extension using artificial neural networks**, 2018.
20. Dan Segev, **Music separation using NMF**, 2018.
21. Hagai Teitelbaum and Assaf Hallela, **Speech enhancement using artificial neural networks**, 2017.
22. Gilad Rat and Hodya Hamer, **Speech enhancement using neural networks and EM algorithm**, 2017.
23. Ido Berkovich and Matan Leibovich, **Acoustic Simultaneous Localization and Mapping**, 2017.
24. Haggay Margalit, Matan Berl, **Multichannel DSP system for noise reduction**, 2017.
25. Liat Neviei, Daphna Ernest, **Real-time binaural speech dereverberation**, 2016.
26. Avi Attal, Guy Zager, **Real-time ad hoc microphone array for speech enhancement**, 2016.
27. Zahi Elbaz, Uri Ernest, **Speaker tracking in noisy and reveberant environment**, 2016.
28. Assaf Avinoam, Avihai Arbel, **Nested GSC for joint noise reduction and dereverberation**, 2016.
29. Chen Tsfaty, Nir Chen, **Android implementation for single microphone DNN-based speech enhancement**, 2016.
30. Reut Toker, Arie Kalmanovich, **Android implementation for speech enhancement algorithm**, 2016.
31. Natan David, **Acoustic scene analysis**, 2016.
32. Eliezer Hershkovich, Nati Frankel, **Android implementation for dual-microphone speech enhancement**, 2015.
33. Avishai Armon, **Statistically-based speech dereverberation**, 2015.
34. Uri Smolyan, Eldad Haimm, **Ad hoc unsynchronized microphone arrays for speech enhancement**, 2015.
35. Barak Ozeri, **Derverberation using Kalman Filter**, 2014.
36. Eran Hadad, Dima Fishman, **3-D Audio Rendering**, 2014.
37. Shlomi Chazan and David Cohen, **Differential Microphone Arrays for Dereverberation**, 2013.
38. Michael Sharvit and Liron Gerby, **Binaural Speech Enhancement for Hearing Aids**, 2013.
39. Barak Ozeri and Roee Ovadia, **Concurrent Speaker Localization**, 2013.

40. Yafit Feldman and Idit Dagan, **Multi Sensor system for Neural Activity Analysis**, Co-supervisor Orit Shefi, 2013.
41. Shay Yederman and Ehud Dagan, **Analog Front-end for Muscle Activity Analysis**, Co-supervisor Eli Kolberg, 2013.
42. Idan Chen and Yosi Buchnik, **Speech Enhancement–Android Implementation**, 2013.
43. Bracha Goldstein, **Localization based on Diffusion Maps**, 2013.
44. Chaim Livschis, Yechiel Klein, **Binaural Algorithms for Speech Enhancement**, 2012.
45. Ariel Malek, Aviaad Chashuel, **Stereophonic Echo Cancellation**, 2012.
46. Amichai Polishuk, Oded Kaminsky, **Distributed Linearly Constrained Beamformer**, 2012.
47. Yatir Tuati, Ishai Alouch, **Speech Seperation Utilizing Sparsity**, 2012.
48. Tomer Meged, Shachar Zigdon, **Real-Time Multi-Microphone Speech Enhancement**, 2012.
49. Ariel Livschis, David Zuker **Adaptive Differential Microphone Array**, 2012.
50. Israel Danziger, **Motion Based Music Instrument**, 2012.
51. Elad Moskovich, Yoni Klein, **Differential Microphones**, 2011.
52. Doron Gluzer, Zvi Sankeivich **Multiple Constraint Beamformer**, 2011.
53. Shlomo Bugdari, Ohad Sharabi **Focused Beamformer**, 2011.
54. Yossi Ben Simon, **Real-Time Speech Enhancement**, 2011.
55. Zuriel Hadad, Yehuda wolker, **Derevereberation using Linear Prediction**, 2011.
56. Shosi Frank, Tal Friedman, **Multichannel N €-analysis**, 2011.
57. Inbar Gesner, Yair Antmann, **Multi-Channel Wiener Filter with Localization Cues for Noise Reduction in Binaural Hearing Aids**, 2010.
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[1] A. Schwartz, E. Hadad, S. Gannot, and S. E. Chazan, "Array configuration mismatch in deep DOA estimation: Towards robust training," in *IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA)*, New Paltz, USA, Oct. 2023.

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