# CURRICULUM VITAE

**Prof. Sharon Gannot**

November 7, 2018

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## Personal Details



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| First Name: | Sharon |
| Surname: | Gannot |
| Date of birth: | August 10, 1964 |
| Place of birth: | Hadera, Israel |
| Nationality: | Israeli |
| Address: | Faculty of Engineering,<br>Bar-Ilan University, Ramat-Gan, 5290002,<br>Israel |
| Telephone: | +972–3–531–7618 (office) |
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| homepage: | `www.eng.biu.ac.il/gannot` |



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## Education

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| PhD, 2000 | Institute: | Department of Electrical Engineering-Systems,<br>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,<br>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,<br>Technion - Israel Institute of Technology. |
| | Studies: | Electrical Engineering. |
| | Distinction: | **Summa Cum Laude**. |

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## Academic Appointments

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| 2017-2018 | Professor (part-time) at the Technical Faculty of IT and Design<br>Department of Architecture, Design and Media Technology,<br>Aalborg University, Denmark. |
| 2014- | Full Professor at the Faculty of Electrical Engineering,<br>Bar-Ilan University, Ramat-Gan, Israel. |
| 2010–2014 | Associate Professor at the Faculty of Electrical Engineering,<br>Bar-Ilan University, Ramat-Gan, Israel. |
| 2006–2010 | Senior Lecturer at the School of Electrical Engineering,<br>Bar-Ilan University, Ramat-Gan, Israel. |

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

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## Other Professional Experience

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| 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<br>in the area of Telecommunication, Signal Processing and Speech Processing.<br>Retired with the rank of Major. |

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## Membership in Professional Societies

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

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## Research Interests

1. Single microphone speech enhancement and source separation.
2. Array processing techniques for source localization and acquisition and noise reduction. Application to speech enhancement, echo cancellation, and dereveberation.
3. Linear and nonlinear optimal filtering. Recursive Bayesian and non-Bayesian estimators with application to speech processing.
4. System identification techniques and adaptive filtering.
5. Speech Processing for Hearing Aid Devices.
6. Distributed algorithms for wireless and ad hoc microphone arrays.
7. Data-driven methods (e.g. manifold learning, deep learning) in speech processing.

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## Teaching Experience

### Lecturer:

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| **Array Processing** | Graduate; | Technical University Aalborg,<br>Denmark; 2018 |
| **Array Processing** | Graduate; | Technical University Liberec,<br>The Czech Republic; 2018 |
| **Array Processing** | Graduate; | Friedrich-Alexander-Universität<br>Erlangen-Nürnberg, Germany; 2017 |
| **Array Processing** | Graduate; | Bar-Ilan University; 2011– |

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| **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:

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| **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

### Research:

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

### Teaching:

2014 Bar-Ilan University Outstanding lecturer award for the year 2014.

2010 Bar-Ilan University Outstanding lecturer award for the year 2010.

2001 Tel-Aviv University, Faculty of Engineering, Outstanding teacher award for the year 2000.

### As a Supervisor:

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), Xin, 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**.

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| 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. 295298, **distinguished paper**. |

### As a Student:

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| 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. 811, **distinguished paper**. |
| 1999 | Excellency scholarship of the students 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 award for Excellency (3 times). |

### As an Industrial Researcher:

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

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## Research Grants

1. **Environment-Aware Data-Driven Acoustic Signal Processing**, *“Kamin” - Israel Innovation Authority*, Dec., 2017. Total amount: 641,840 ILS (BIU 321,840 ILS).

2. **Audio processing algorithms for human-machine interface in mobile and stationary devices**, *“Magneton” - Israel Innovation Authority*, Aug., 2017. Amount: 361,400 ILS.

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

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

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

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

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

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

9. **Distributed Microphone Arrays for Personal Devices**, *Intel* research grant, 2014-2017. Amount: 350,000USD.
10. **Supervised Speaker Tracking Using Diffusion Kernel Combined with Extended Particle Filter**, *Ministry of Science* research grant for women is science, 2014. Amount: 20,000ILS.
11. **Distributed Microphone Arrays**, *MAFAAT* research grant, 2014. Amount: 150,000ILS for first year, 175,000ILS for second year.
12. **3-D Audio Rendering**, *Orbit* research grant, 2014. Amount: 48,000ILS.
13. **Two Microphone Noise Reduction in Adverse Conditions**, *Cardo* research grant, 2013. Amount: 225,000ILS.
14. **Binaural Speech Dereverberation**, *German-Israeli Foundation (GIF)*, 2013-2015, Jointly with E.A.P. Habets and S. Doclo. Amount: 198,500€.
15. **Keyboard Noise Reduction**, *Waves* research grant, 2012. Amount: 165,000ILS.
16. **Robust ASR in Reverberant Environment**, *Samsung* research grant, 2012-2013. Amount: 390,000ILS.
17. **Robust ASR in Car Environment**, *General Motors* research grant, 2012-2013. Amount: 290,000ILS.
18. **Beamforming Using Flat Microphones**, *MAFAAT* grant (Israel Ministry of Defense) 2011-2013. Amount: 62,500ILS for first year, 62,500ILS for second year .
19. **Differential Microphone Arrays**, *Rubidium* grant for undergraduate project, 2011. Amount: 15,000ILS.
20. **Hearing Aids**, *Qualcomm* research grant, 2011. Amount: 140,000$.
21. **Speech Separation Using Single- and Multi-Microphone processing**, *Magneton* grant, Aug., 2010, together with *Nice Systems Ltd.* Amount: 473,000ILS for first year. 409,000ILS for second year.
22. **Speech processing using Microflowns**, *MAFAAT* grant (Israel Ministry of Defense), Feb., 2010. Amount: 250,000ILS for first year, 220,000ILS for second year, 150,000ILS for third year.
23. **Echo Cancellation for Improving Speech Recognizers**, *Magneton* grant, Aug. 2008, together with *Nice Systems*. Amount: 500,000ILS for first year. 500,000ILS for second year.
24. **Speaker Localization in Noisy and Reverberant Environment**, *MAFAAT* grant (Israel Ministry of Defense), Jul., 2008. Together with Israel Cohen. Amount: 280,000ILS for first year. 150,000ILS for second year. 150,000ILS for third year.
25. **Acoustic Array Signal Processing in Adverse Environments**, the state-of-Israel *Higher-Education Council post-doctoral* grant for Emanuël A.P. Habets (host laboratory).
26. Funded undergraduate project (Hanan Ashwega and Nir Russo), **Speech Source Localization in Noisy and Reverberant Environment using the Particle Filter**, 2007. Amount: 30,000ILS.
27. Funded undergraduate project (Arieh Jerichover and Ariel Bierendorf), **Residual Echo Cancellation**, 2007. Amount: 30,000ILS.
28. **Blind Speaker separation in Adverse Conditions**, *MAFAAT* grant (Israel Ministry of Defense), Dec., 2005, together with Israel Cohen. Amount: 275,000ILS for first year, 285,000ILS for second year, 295,000ILS for third year.

29. **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$.

30. **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: 160,000ILS for first year; 300,000ILS for second year; 300,000ILS for third year; 200,000ILS for fourth year, 225,000ILS for fifth year, 150,000ILS for sixth year.

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

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## Professional Activities

### Conference Chairing:

1. Technical Co-Chair, **the international conference on Latent Variable Analysis and Signal Separation, LVA/ICA**, University of Surrey, Guildford, UK, July 2018.
2. General Co-Chair, **the 29th IEEE Israel conference - International Symposium on Speech and Audio Processing**, Nov. 2016, Eilat, Israel.
3. General Co-Chair, **the IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA)**, Mohonk Mountain House, New Paltz, NY, October 2013.
4. General Co-Chair, **the International Workshop on Acoustic Echo and Noise Control (IWAENC)**, Aug.-Sep. 2010, Tel Aviv, Israel.
5. Chair, **the International workshop LVA/ICA - Audio Day**, Bar-Ilan University, Mar. 2012.
6. Co-chair, **the Bar-Ilan Workshop on Signal Processing**, Bar-Ilan University, Israel, Jan., 2011.

### Conference Organization:

1. Area chair, the European Signal Processing Conference (EUSIPCO), Nice, France, Aug. 2015.
2. Technical committee member, **Reverb Challenge**, organized by IEEE Signal Processing Society, **Audio and Acoustics Signal Processing Technical committee**, May, 2014.
3. Area chair, the European Signal Processing Conference (EUSIPCO), Marrakech, Morocco, Sep. 2013.
4. 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.
5. 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 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.

2. 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.
3. 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.
4. 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.
5. Organizer, **special session on Advances in Multi-Microphone Speech Separation and Noise Reduction**, the European Signal Processing Conference (EUSIPCO), Marrakech, Morocco, Sep. 2013.
6. Organizer and Chair, **special session on microphone array processing**, the 27th IEEE Israel conference, Dec., 2012.
7. Organizer and Chair **special session on speech enhancement**, the 25th IEEE Israel conference, Dec., 2012.

### Technical Committees:

1. IEEE Signal Processing Society, chair **Audio and Acoustics Signal Processing Technical committee**, since 1.1.2017.
2. IEEE Signal Processing Society, vice chair **Audio and Acoustics Signal Processing Technical committee**, since 1.1.2016.
3. IEEE Signal Processing Society, member of the **Audio and Acoustics Signal Processing Technical committee**, since 1.1.2010.
4. IEEE Signal Processing Society, **Audio and Acoustics Signal Processing Technical committee**, Chair of EDICS subcommittee, Oct. 2011–Jun. 2013.
5. EURASIP, member of **Audio, Speech and Music (ASMSP) Special Area Team (SAT)**, since Aug. 2015, **organizer of new member elections**, Dec. 2017.
6. Member of the Technical and Steering committee, **the International Workshop on Acoustic Echo and Noise Control (IWAENC)**, since 2005.
7. Member of the evaluation committee of the **Exact Sciences and Technology branch** of the Israel Science Foundation (ISF), 2007.

### Associate Editor:

1. Moderator, **Arxiv, Electrical Engineering and Systems Science - Audio and Speech Processing**, 2016–
2. Senior area chair, **IEEE Transactions on Audio, Speech and Language Processing**, 2013–2017.
3. Associate editor **IEEE Transactions on Audio, Speech and Language Processing**, 2009–2013.
4. Associate editor **EURASIP journal on Advances in Signal Processing**, 2004–2012.
5. Associate editor, **Springer Handbook of Speech Processing and Speech Communication**, 2006.

## Guest Editor of Special Issues in Journals:

1. 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, February, 2018.

2. 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.

3. 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.

4. 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.

5. 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.

6. 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.

7. 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 GlobalSip.

Journals — IEEE Transactions on Signal Processing; IEEE 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. 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.
2. Member of the jury of PhD defense of Ante Juki´ on **Sparse Multi-channel Linear Prediction for Blind Speech Dereverberation**, University of Oldenburg, Germany, Oct. 2017.
3. PhD Reviewer, **Localization and Tracking of Acoustic Sources in Room Environment**, Wu Kai, Nanyang Technological University, Singapore, Feb. 2017.

4. PhD Reviewer, **Robust Multichannel Microphone Beamforming**, Craig Anderson, Victoria University of Wellington), New-Zealand, Jan. 2016.
5. Member of the jury of PhD defense of Yuan Zeng on **Distributed Speech Enhancement in Wireless Acoustic Sensor Networks**, TUDelft, The Netherlands, Jun. 2015.
6. Member of the jury of PhD defense of Joseph Szurley on **Distributed Signal Processing Algorithms for AcousticSensor Networks**, K.U.Leuven, Belgium, May 2015.
7. 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.
8. PhD Reviewer, **Multichannel Equalization Applied to Speech Dereverberatio**, Rajan Sonhana Rashobh, Nanyang Technological University, Singapore, Aug. 2014.
9. 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.
10. 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.
11. PhD Reviewer, **Adaptive System Identification and Equalization Algorithms for Acoustic Echo Cancellation and Speech Dereverberation**, Liao Lei, Nanyang Technological University, Singapore, Mar. 2013.
12. Habilitation reviewer, **Nonlinear, Time-Varying, and Blind Acoustic System Identification**, Gerlad Enzner, Ruhr University, Bochum, Germany, May 2012.
13. PhD Reviewer, **Acoustical Time-Reversal Signal Processing: New Developments and Applications**, Nguyen Dinh Quy, Nanyang Technological University, Singapore, Sep. 2010.
14. 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.
15. Referee, PhD thesis, **Speech enhancement using microphone arrays**, S. Y. Low, Curtin University, Australia, Sep. 2005.
16. 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, Lior Fritz, Mar. 2018.
2. MSc, Izhak Zimmermann, Music Department, Bar-Ilan, Sep. 2017.
3. MSc, Ori Katz, Technion, Aug. 2017.
4. PhD, Vladimir Tourbabin, Ben-Gurion, Apr. 2017.
5. MSc Oren Rosen, Technion, Jan. 2017.
6. PhD propsal, Tal Schnizer, Tecjnion, Dec. 2016.
7. MSc, Yoav Biederman, Ben-Gurion, Dec. 2016.
8. MSc, Kfir Aberman, Technion, Dec. 2016.
9. PhD defense, Yaakov Bucris, Nov. 2016.

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

35. 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.
36. MSc thesis on **Maximum A-Posteriori Probability Multiple Pitch Tracking Using the Harmonic Model**, Amitai Koretz, en-Gurion University, Nov. 2009.
37. MSc thesis on **New Methods for Speech Recognition**, Roee Lahav, Ben-Gurion University, Dec. 2010.
38. MSc thesis on **Speaker Recognition in Reverberant Environment**, Itai Peer, Ben-Gurion University, Aug. 2009.
39. MSc thesis on **Adaptive Stereo Acoustic Echo Cancelation in reverberant environments**, Amos Schreibman, the Technion–IIT, Oct. 2009.
40. MSc thesis on **Time Difference of Arrival Estimation in Multi-path Environmen**, Ity Erlich Tel-Aviv University, Mar. 2009.
41. MSc thesis on **Packet Loss Concealment for Voice Applications**, Yishai Gil, Ben-Gurion University, Feb. 2009.
42. MSc thesis on **Time-Varying Perceptual Linear Prediction for Speech Application**, Oron Gamliel, Ben-Gurion University, Feb. 2009.
43. PhD Dissertation on **System Identification in the Short-Time Fourier Transform Domain**, Yekutiel Avargel, the Technion, Nov. 2008.
44. 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.
45. MSc thesis on **Direct Localization of Cyclo-Stationary Sources**, Alit Mendelsson-Reuven, Tel-Aviv University, Jan., 2008.
46. MSc thesis on **MIMO-AR blind source separation for GMM-distributed and finite alphabet signals**, Tirza Ruttenberg at Ben-Gurion University, Jan. 2008.
47. 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.
48. MSc thesis on **Support Vector Machine Training for Improved Hidden Markov Modeling**, Alba Sloin, Tel-Aviv University, Oct. 2006.
49. MSc thesis on **MIMO Decision Directed Channel Estimation for Dynamic Channels**, Ory Eger, Tel-Aviv University, May 2006.
50. Member of PhD candidate committee, **Signal Localization**, Alon Amar, Tel-Aviv University, Sep. 2005
51. MSc thesis on **Bootstrap Kalman Filter: A Hybrid Scheme for Bilinear State-Space Models**, Yuval Yosef Domb, Tel-Aviv University, 2005.
52. MSc thesis on **Perceptual Time-Varying Modelling of Speech Signals for ASR and Compression Application**, Ben-Gurion University, 2005.
53. MSc thesis on **Time-Frequency Representation for Speech Recognition**, Ben-Gurion University, 2005.
54. MSc thesis on **Time-Frequency Representation of Nonstationary Signals Using NAR Autocorrelation**, Ben-Gurion University, 2005.
55. MSc thesis on **On the Resolution of Overlapping Echoes of a Random Signal**, Tel-Aviv University, 2005.

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

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## Tutorials and Keynote Addresses

1. Keynote address, **Multi-Microphone Speaker Localization and Tracking on Manifolds**, the 13th ITG Conference on Speech Communication, Oldenburg, Germany, October 10-12, 2018.
2. 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.
3. 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.
4. 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.
5. Keynote address, **Multi-Microphone Speech Enhancement Using LCMV Beamformers**, the International Workshop on Acoustic Signal Enhancement (IWAENC), Aachen, Germany, Sep. 2012.
6. 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.
7. 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.
8. Tutorial (together 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.

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## Invited Talks

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.

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| 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. |

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| 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

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| 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. |
| Since 2013 | 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:

Post-Doctorate

1. Elior Hadad, **Speaker Localization and Separation**.
   Start: Apr. 2017.
2. Shmulik Markovich-Golan, **Algorithms for Speech Processing**.
   Start: Mar. 2013.

PhD

1. Renana Opochinsky (Kleinman). Co-supervisor Gal Chechik, Start: Apr. 2018.
2. Yaron Laufer, **Baeysian Methods in Speech Processing**, Start: Oct. 2017.
3. Shlomi Chazan, **Deep Learning Methods for Speech Enhancement**.
   Co-supervisor: Jacob Goldberger, Start: Oct. 2015.
4. Bracha Laufer, **Manifold Learning Methods for Speech Processing** (direct track).
   Co-supervisor: Ronen Talmon, Technion, Start: Oct. 2013.

MSc

1. Hodaya Hammer. Start Apr. 2018.
2. Koby Visberg, **Speaker Tracking**. Start: Oct. 2017.
3. Nilli Cohen, **Speech Dereverberation**. Start: Oct. 2017.
4. Ori Ernest, **Speech Enhancement using Generative Adversarial Network**. Start: Oct. 2016.
5. Maya Veisman, **Simultaneous Room Geometry Inference and Speaker Localization**.
   Start: May 2016.

6. Yosef Soussana, **Bayesian Speaker Localization**.
   Start: Feb. 2017.
7. Ariel Malek, **Beamforming for Automatic Speech Recognition**.
   Start: Mar. 2013.

FORMER:

Post-Doctorate

1. David Levin, **Distributed Algorithms for Microphone Arrays**. May 2016–April 2017.
2. Emanuël A.P. Habets, **Speech Processing**. Feb. 2007–Jan. 2009.

PhD

1. Dani Cherkassky, **Microphone Array Processing: Theoretical Study**.
   Start: Oct. 2013. Submitted Oct. 2018.
2. Yuval Dorfan, **Distributed Localization and Tracking of Acoustic Sources**.
   Start: Oct. 2013. Approved Sep. 2018.
3. Boaz Schwartz, **Derverberation Methods for Binaural Hearing** (direct track).
   Co-supervisor: E.A.P. Habets (FAU, Erlangen-Nuremberg, Germany).
   Start: Oct. 2010, Approved, May 2018.
4. Ofer Schwartz, **Multi-microphone Derverberation Algorithms**.
   Co-supervisor: E.A.P. Habets (FAU, Erlangen-Nuremberg, Germany).
   Start: Oct. 2012, Approved, Mar. 2018.
5. David Levin, **Speech processing using Acoustic Vector Sensors** (direct track).
   Co-supervisor: E.A.P. Habets (FAU, Erlangen-Nuremberg, Germany). Approved, Mar. 2018.
6. Elior Hadad, **Speech Processing for Hearing Aids**. Graduated: Jun. 2016.
7. Shmulik Markovich-Golan, **Speech Processing using Distributed Microphone Networks**.
   Co-supervisor: Israel Cohen (Technion-IIT). Graduated: Aug. 2013.
8. Ronen Talmon, **Supervised Speech Processing Based on Geometric Analysis**.
   Co-supervisor: Israel Cohen (Technion–IIT). Graduation: July, 2011.
9. 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

1. Anna Baranov, **Microphone Array Processing on Multiple Devices**, Start: Feb. 2014. Graduation: Mar. 2018.
2. Aviel Adler, Project, **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.
3. Tamar Marom-Shalev, **Wireless Acoustic Sensor Networks: Combined Acoustic Echo Cancellation and Adaptive Beamforming**. Graduation: Nov. 2017.
4. 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.

5. Shlomi Chazan, **Deep Learning Methods for Speech Enhancement**. Co-supervisor: Jacob Goldberger, Graduation: Oct. 2015.
6. Boaz Castro, **Speech Dereverberation using Subspace Methods**. Graduation: Oct. 2015.
7. Yochay Yeminy, **Single Microphone Speech Separation**. Co-supervisors: Yossi Keller. Graduation: Oct., 2011.
8. Ofer Schwartz, **Concurrent Speech Localization**. Graduation: Oct. 2012.
9. Livnat Ehrenberg, **Performance Bounds on MIMO tracking systems**. Co-supervisor: Amir Leshem. Graduation: Aug. 2011.
10. Avinoam Levi, **Speaker Localization using Particle filters**. Graduation: May 2011.
11. Shmulik Markovich, **Multichannel Eigenspace Beamforming in a Reverberant Noisy Environment with Multiple Interfering Speech Signals**. Co-supervisor: Israel Cohen (Technion–IIT). Graduation: Oct. 2008.
12. 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.
13. Tsvi G. Dvorkind, **Speaker Localization in a Reverberant and Noisy Environment**. Graduation **Summa Cum Laude**, Nov. 2003.

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## Undergraduate Projects

### Current:

Bar-Ilan Uni.

1. Guy Jisfan and Daniel Shalev, **Emotion recognition using artificial neural networks**, 2018.
2. Roy Tendlerr and Or Gershon, **Bandwidth extension using artificial neural networks**, 2018.
3. Dan Segev, **Music separation using NMF**, 2018.

### Former:

Bar-Ilan Uni.

1. Hagai Teitelbaum and Assaf Hallela, **Speech enhancement using artificial neural networks**, 2017.
2. Gilad Rat and Hodia Hamer, **Speech enhancement using neural networks and EM algorithm**, 2017.
3. Ido Berkovich and Matan Leibovich, **Acoustic Simultaneous Localization and Mapping**, 2017.
4. Haggay Margalit, Matan Berl, **Multichannel DSP system for noise reduction**, 2017.
5. Liat Neviei, Daphna Ernest, **Real-time binaural speech dereverberation**, 2016.
6. Avi Attal, Guy Zager, **Real-time ad hoc microphone array for speech enhancement**, 2016.
7. Zahi Elbaz, Uri Ernest, **Speaker tracking in noisy and reveberant environment**, 2016.
8. Assaf Avinoam, Avihai Arbel, **Nested GSC for joint noise reduction and dereverberation**, 2016.

9. Chen Tsfaty, Nir Chen, **Android implementation for single microphone DNN-based speech enhancement**, 2016.
10. Reut Toker, Arie Kalmanovich, **Android implementation for speech enhancement algorithm**, 2016.
11. Natan David, **Acoustic scene analysis**, 2016.
12. Eliezer Hershkovich, Nati Frankel, **Android implementation for dual-microphone speech enhancement**, 2015.
13. Avishai Armon, **Statistically-based speech dereverberation**, 2015.
14. Uri Smolyan, Eldad Haimm, **Ad hoc unsynchronized microphone arrays for speech enhancement**, 2015.
15. Barak Ozeri, **Derverberation using Kalman Filter**, 2014.
16. Eran Hadad, Dima Fishman, **3-D Audio Rendering**, 2014.
17. Shlomi Chazan and David Cohen, **Differential Microphone Arrays for Dereverberation**, 2013.
18. Michael Sharvit and Liron Gerby, **Binaural Speech Enhancement for Hearing Aids**, 2013.
19. Barak Ozeri and Roee Ovadia, **Concurrent Speaker Localization**, 2013.
20. Yafit Feldman and Idit Dagan, **Multi Sensor system for Neural Activity Analysis**, Co-supervisor Orit Shefi, 2013.
21. Shay Yederman and Ehud Dagan, **Analog Front-end for Muscle Activity Analysis**, Co-supervisor Eli Kolberg, 2013.
22. Idan Chen and Yosi Buchnik, **Speech Enhancement–Android Implementation**, 2013.
23. Bracha Goldstein, **Localization based on Diffusion Maps**, 2013.
24. Chaim Livschis, Yechiel Klein, **Binaural Algorithms for Speech Enhancement**, 2012.
25. Ariel Malek, Aviaad Chashuel, **Stereophonic Echo Cancellation**, 2012.
26. Amichai Polishuk, Oded Kaminsky, **Distributed Linearly Constrained Beamformer**, 2012.
27. Yatir Tuati, Ishai Alouch, **Speech Seperation Utilizing Sparsity**, 2012.
28. Tomer Meged, Shachar Zigdon, **Real-Time Multi-Microphone Speech Enhancement**, 2012.
29. Ariel Livschis, David Zuker **Adaptive Differential Microphone Array**, 2012.
30. Israel Danziger, **Motion Based Music Instrument**, 2012.
31. Elad Moskovich, Yoni Klein, **Differential Microphones**, 2011.
32. Doron Gluzer, Zvi Sankeivich **Multiple Constraint Beamformer**, 2011.
33. Shlomo Bugdari, Ohad Sharabi **Focused Beamformer**, 2011.
34. Yossi Ben Simon, **Real-Time Speech Enhancement**, 2011.
35. Zuriel Hadad, Yehuda wolker, **Derevereberation using Linear Prediction**, 2011.
36. Shosi Frank, Tal Friedman, **Multichannel Neuro-analysis**, 2011.
37. Inbar Gesner, Yair Antmann, **Multi-Channel Wiener Filter with Localization Cues for Noise Reduction in Binaural Hearing Aids**, 2010.
38. Boaz Schwartz, Idan Bakish, **Distributed Adaptive Node-Specific MMSE Signal Estimation In Sensor Networks**, 2010.

39. Ofir Pinchas, Tali Warshavski, **Real-Time Implementation of a Small Dual-Channel Microphone Array for Speech Enhancement**, 2010.
40. Dvir Avzirat, Ofer Busani, **'Frequency-domain adaptive Kalman filter for acoustic echo control in hands-free telephones**, 2010.
41. Yochay Yeminy , **"Speech Enhancement via dimensionality reduction and GMM"**, Co-supervisor Yosi Keller, 2010 .
42. Zach David, Rotem Mor , **Development multi-channel system for analyzing neural activity**, Co-supervisor Orit Shefi, 2010.
43. Zur Lev and Shooky Zadok, **A robust adaptive beamformer for microphone arrays with a blocking matrix using constrained adaptive filters**, 2009.
44. Aviaad Bienenstok and Yedidya Weiss, **Nonlinear residual echo suppression using a power filter model of the acoustic echo path**, 2009.
45. Aviaad Rossmann and Yonatan Moshkovich, **A Nonparametric VSS NLMS Algorithm**, 2009.
46. Hila Barel and Shachar Bar-Tikva, **Multichannel Eigenspace Beamforming in a Reverberant Noisy Environment with Multiple Interfering Speech Signals**, 2009.
47. Nir Admaty and Eizak Rackach, **Speech Enhancement Based on the Subspace Method**, 2009.
48. Ofer Schwarz and Avishai Friedman, **Information, Prosody, and Modeling with Emphasis on Tonal Features of Speech**, 2009.
49. Shooky Hadad and Eran Mershein, **Voice Morphing**, 2009.
50. David Levin and Evyatar Wiesel, **Speech dereverberation using Room Impulse Response Modeling**, 2008.
51. Benni Robinov, B.Sc. , **Image Method for Frequency Domain Room Impulse Response Simulator**, 2008.
52. Eyal Heller and Ami Buchabza, **Double Talk Detectors for Robust Echo Cancellation**, 2008.
53. Avi Dayan, **Speaker TDOA Estimation using Multiple-Microphones"**, Bar-Ilan University, 2008.
54. Dmitry Kochubeevsky and Tomer Zildman, **Single Channel Speaker Separation using the MIXMAX Model**, 2008.
55. Tomer Levi, **GSVD-based Speech Enhancement Algorithm with Microphone Arrays**, 2007.
56. Shachaf Melman and Amichai Meiri, **Convolutive Blind Source Separation**, 2007.
57. Hanan Ashwega and Nir Russo, **Speech Source Localization in Noisy and Reverberant Environment using the Particle Kalman Filter**, 2007.
58. Arieh Jerichover and Ariel Bierendorf, **Residual Echo Cancellation**, 2007.
59. Amit Strauss and Ofer Margalit, **Speech Distortion Weighted Multichannel Wiener Filtering Techniques for Noise Reduction**, 2006.
60. Yosef Fryszer and Rabin Cohen-Tov, **"Blind Source Separation using the JADE algorithm"**, 2006.
61. Michael Yarezky, **Psychoacoustic Research of Auditory Biofeedback**, 2006.

62. Michael Bezman and Michael Laptanikov, **Speech Source Localization in Noisy and Reverberant Environment using the Unscented Kalman Filter**, 2006.
63. Shay Dekel and Tal Gorgi, **Joint Noise Reduction and Echo Cancellation for Speech Communication Application**, 2005.
64. Nir Laufer and Eyal Reich, **An Integrated Real-Time Beamforming and Postfiltering System for Non-Stationary Noise Environments**, 2005.
65. Ofer Limon and Israel Grunwald, **Speech Source Localization in Noisy and Reverberant Environment**, 2005.
66. Livnat Erenberg and Yariv Erenberg, **Multi-Microphone Speech Enhancement-Algorithms Comparison and Assessment**, 2005.
67. Asaf Danino, **Speech Recognition Front End-Algorithms Comparison and Assessment**, 2005.
68. Adva HaLachmi and Meital Nachum, **Blind Source Separation**, 2005.
69. Bloomenfeld Hadas, **Voice Activity Detector-Algorithms Comparison and Assessment**, 2005.
70. Eyal Pdael and Ariel Perez, **Single Mic. Speech Enhancement using Kalman Filter-Implementation in SPDemo**, 2005.
71. Ravid Solomon and Yoni Beck, **Signal Enhancement Using Beamforming and Non-Stationarity with application to Speech-Implementation in SPDemo**, 2005.
72. Hagay Toledano and Itay Kahana, **Speech Morphing**, 2005.

Technion-IIT

1. Dorit Baras ,**Implementation of the MixMax algorithm for Single Microphone Speech Enhancement**, 2003.
2. Shira Nemirovsky, **Improvments of the MUSIC and ESPRIT algorithms**, 2003.

Tel-Aviv Uni.

1. Eran Groso, **Acoustic Echo Cancellation**, 1997.
2. Anelia Baruch, B.Sc , **On Improving the performance of the LPC-10 decoder**, 1996.
3. Uri Sharony and Oren Bahat, **Iterative-Batch Kalman Filter-Based Speech Enhancement Algorithms**, 1995.
4. Alex Margolis and Michael Pevzner, **Sequential Kalman Filter-Based Speech Enhancement Algorithms**, 1995.

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## Publications

### Edited Books:


[1] E. Vincent, T. Virtanen, and S. Gannot, Eds., *Audio Source Separation and Speech Enhancement*. Wiley, Sep. 2018.

[2] Y. Deville, S. Gannot, R. Mason, M. D. Plumbley, and D. Ward, "Proceedings of the 14th international conference on latent variable analysis and signal separation (LVA/ICA)," Guildford, United Kingdom, July 2–5 2018.

[3] I. Cohen, J. Benesty, and S. Gannot, Eds., *Speech processing in modern communication: Challenges and perspectives*, ser. Topics in signal processing. Springer, 2010.

### Theses:


[1] S. Gannot, "Array processing of nonstationary signals with application to speech," Ph.D. dissertation, Tel-Aviv University, 2000.

[2] S. Gannot, "Algorithms for single microphone speech enhancement," Master's thesis, Tel-Aviv University, Apr. 1995.


### Manuals:


[1] S. Gannot, V. Avrin, B. Schwartz, D. Levin, and P. Tandeitnik, *Advaned lab in signal processing based on TMS6713 and Simulink*, 2nd ed., Faculty of Engineering, Bar-Ilan University, Feb. 2013, in Hebrew.


### Book Chapters:


[1] L. Girin, S. Gannot, and X. Li, *Multi-modal Behavior Analysis in the Wild: Advances and Challenges.* Elsevier, ch. Audio source separation into the wild, under edition.

[2] E. Vincent, S. Gannot, and T. Virtanen, *Audio Source Separation and Speech Enhancement.* Wiley, Sep. 2018, ch. Introduction.

[3] T. Virtanen, E. Vincent, and S. Gannot, *Audio Source Separation and Speech Enhancement.* Wiley, Sep. 2018, ch. Time-frequency processing Spectral properties.

[4] E. Vincent, S. Gannot, and T. Virtanen, *Audio Source Separation and Speech Enhancement.* Wiley, Sep. 2018, ch. Acoustics Spatial properties.

[5] E. Vincent, S. Gannot, and T. Virtanen, *Audio Source Separation and Speech Enhancement.* Wiley, Sep. 2018, ch. Perspectives.

[6] S. Markovich-Golan, W. Kellermann, and S. Gannot, *Audio Source Separation and Speech Enhancement.* Wiley, Sep. 2018, ch. Spatial filtering.

[7] S. Markovich-Golan, W. Kellermann, and S. Gannot, *Audio Source Separation and Speech Enhancement.* Wiley, Sep. 2018, ch. Multichannel parameter estimation.

[8] S. Doclo, S. Gannot, D. Marquardt, and E. Hadad, *Audio Source Separation and Speech Enhancement.* Wiley, Sep. 2018, ch. Binaural speech processing with application to hearing devices.

[9] S. Markovich-Golan, S. Gannot, and I. Cohen, *Audio Source Separation.* Springer, 2018, ch. Multimicrophone MMSE-based speech source separation.

[10] K. Kinoshita, M. Delcroix, S. Gannot, E. A. P. Habets, R. Haeb-Umbach, W. Kellermann, V. Leutnant, R. Maas, T. Nakatani, B. Raj, A. Sehr, and T. Yoshioka, *New Era for Robust Speech Recognition: Exploiting Deep Learning.* Springer, 2017, ch. REVERB challenge: A benchmark task for reverberation-robust ASR techniques.

[11] R. Talmon, I. Cohen, and S. Gannot, *Speech Processing in Modern Communication.* Springer, 2010, ch. Identification of the Relative Transfer Function between Sensors in the Short-Time Fourier Transform Domain, pp. 33–47.

[12] E. Habets, J. Benesty, S. Gannot, and I. Cohen, *Speech Processing in Modern Communication.* Springer, 2010, ch. The MVDR beamformer for speech enhancement, pp. 225–254.

[13] S. Markovich, S. Gannot, and I. Cohen, *Speech Processing in Modern Communication.* Springer, 2010, ch. Extraction of Desired Speech Signals in Multiple-Speaker Reverberant Noisy Environments, pp. 255–279.

[14] S. Doclo, S. Gannot, M. Moonen, and A. Spriet, *Handbook on Array Processing and Sensor Networks*. Wiley-IEEE Press, 2010, ch. Acoustic beamforming for hearing aid applications.

[15] S. Gannot, *Speech Dereverberation*. Springer, 2010, ch. Multi-microphone Speech Dereverberation Using Eigen-decomposition, pp. 129–156.

[16] E. Habets, S. Gannot, and I. Cohen, *Topics in Speech and Audio Processing in Adverse Environments*. Springer, 2008, ch. Dereverberation and Residual Echo Suppression in Noisy Environments, pp. 185–227.

[17] I. Cohen and S. Gannot, *Springer Handbook of Speech Processing and Speech Communication*. New York: Springer-Verlag, 2007, ch. Spectral enhancement methods.

[18] S. Gannot and I. Cohen, *Springer Handbook of Speech Processing and Speech Communication*. New York: Springer-Verlag, 2007, ch. Adaptive beamforming and postfiltering.

[19] S. Gannot and A. Yeredor, *Springer Handbook of Speech Processing and Speech Communication*. New York: Springer-Verlag, 2007, ch. The Kalman filter.

[20] S. Gannot, *Speech Enhancement*. Springer, 2005, ch. Application of the Kalman Filter in the Estimate-Maximize (EM) Framework, pp. 161–198.

### Journal Publications - Published and Accepted:

[1] B. Schwartz, S. Gannot, E. A. Habets, and Y. Noam, “Recursive maximum likelihood algorithm for dependent observations,” *IEEE Transactions on Signal Processing*, Mar. 2018.

[2] X. Li, S. Gannot, L. Girin, and R. Horaud, “Multichannel identification and non-negative equalization for dereverberation and noise reduction based on convolutive transfer function,” *IEEE/ACM Transactions on Audio, Speech, and Language Processing*, Sep. 2018, accepted for publication with required minor revisions.

[3] Y. Laufer and S. Gannot, “A bayesian hierarchical model for speech enhancement with time-varying audio channel,” *IEEE/ACM Transactions on Audio, Speech, and Language Processing*, Sep. 2018, accepted for publication with required minor revisions.

[4] B. Laufer-Goldshtein, R. Talmon, and S. Gannot, “Source counting and separation based on simplex analysis,” *IEEE Transactions on Signal Processing*, Sep. 2018, accepted for publication; arXiv preprint arXiv:1802.09221.

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