Jacob Benesty

List of Publications by Year
in descending order

445 papers	12,372 citations	30070 54 h-index	39675 94 g-index
498 all docs	498 docs citations	498 times ranked	4053 citing authors

#	Article	IF	Citations
1	On the stochastic modeling of the LMS algorithm operating with bilinear forms. , 2022, 122, 103359.		6
2	Cascaded RLS Adaptive Filters Based on a Kronecker Product Decomposition. Electronics (Switzerland), 2022, 11, 409.	3.1	3
3	Kronecker Product Multichannel Linear Filtering for Adaptive Weighted Prediction Error-Based Speech Dereverberation. IEEE/ACM Transactions on Audio Speech and Language Processing, 2022, 30, 1277-1289.	5.8	19
4	Data-Reuse Recursive Least-Squares Algorithms. IEEE Signal Processing Letters, 2022, 29, 752-756.	3.6	9
5	A Variable Step Size Normalized Least-Mean-Square Algorithm Based on Data Reuse. Algorithms, 2022, 15, 111.	2.1	10
6	Efficient Algorithms for Linear System Identification with Particular Symmetric Filters. Applied Sciences (Switzerland), 2022, 12, 4263.	2.5	2
7	LMS and NLMS Algorithms for the Identification of Impulse Responses with Intrinsic Symmetric or Antisymmetric Properties. , 2022, , .		3
8	Study of the Null Directions on The Performance of Differential Beamformers. , 2022, , .		1
9	DNN Based Multiframe Single-Channel Noise Reduction Filters. , 2022, , .		2
10	On Differential Beamforming With Nonuniform Linear Microphone Arrays. IEEE/ACM Transactions on Audio Speech and Language Processing, 2022, 30, 1840-1852.	5.8	2
11	Microphone Array Beamforming With High Flexible Interference Attenuation and Noise Reduction. IEEE/ACM Transactions on Audio Speech and Language Processing, 2022, 30, 1865-1876.	5.8	4
12	Efficient Identification of Acoustic Linear Systems. , 2022, , .		0
13	Multistage approach for steerable differential beamforming with rectangular arrays. Speech Communication, 2022, 142, 61-76.	2.8	8
14	Steering Study of Linear Differential Microphone Arrays. IEEE/ACM Transactions on Audio Speech and Language Processing, 2021, 29, 158-170.	5.8	29
15	On microphone array beamforming and insights into the underlying signal models in the short-time-Fourier-transform domain. Journal of the Acoustical Society of America, 2021, 149, 660-672.	1.1	4
16	Array Beamforming with Linear Difference Equations. Springer Topics in Signal Processing, 2021, , .	0.2	7
17	On the Robustness of the Superdirective Beamformer. IEEE/ACM Transactions on Audio Speech and Language Processing, 2021, 29, 838-849.	5.8	15
18	On the Design of 3D Steerable Beamformers With Uniform Concentric Circular Microphone Arrays. IEEE/ACM Transactions on Audio Speech and Language Processing, 2021, 29, 2764-2778.	5.8	7

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19	Robust Source Separation with Differential Microphone Arrays and Independent Low-Rank Matrix Analysis. , 2021, , .		0
20	Beamforming with Cube Microphone Arrays Via Kronecker Product Decompositions. IEEE/ACM Transactions on Audio Speech and Language Processing, 2021, 29, 1774-1784.	5.8	15
21	On the Design of Differential Kronecker Product Beamformers. IEEE/ACM Transactions on Audio Speech and Language Processing, 2021, 29, 1397-1410.	5.8	15
22	A New Method to Design Steerable First-Order Differential Beamformers. IEEE Signal Processing Letters, 2021, 28, 563-567.	3.6	5
23	Differential Beamforming From the Beampattern Factorization Perspective. IEEE/ACM Transactions on Audio Speech and Language Processing, 2021, 29, 632-643.	5.8	5
24	A Single-Input/Binaural-Output Antiphasic Speech Enhancement Method for Speech Intelligibility Improvement. IEEE Signal Processing Letters, 2021, 28, 1445-1449.	3.6	4
25	Tensor-Based Adaptive Filtering Algorithms. Symmetry, 2021, 13, 481.	2.2	22
26	A Kronecker product CLMS algorithm for adaptive beamforming. , 2021, 111, 102968.		18
27	On the compromise between noise reduction and speech/noise spatial information preservation in binaural speech enhancement. Journal of the Acoustical Society of America, 2021, 149, 3151-3162.	1.1	3
28	A Kalman Filter for Multilinear Forms and Its Connection with Tensorial Adaptive Filters. Sensors, 2021, 21, 3555.	3.8	5
29	Planar Array Geometry Optimization for Region Sound Acquisition. , 2021, , .		2
30	Robust Steerable Differential Beamformers with Null Constraints for Concentric Circular Microphone Arrays. , 2021, , .		1
31	Combined Differential Beamforming With Uniform Linear Microphone Arrays. , 2021, , .		1
32	A Simplified Wiener Beamformer Based on Covariance Matrix Modelling. , 2021, , .		2
33	Robust Recursive Least M-Estimate Adaptive Filter for the Identification of Low-Rank Acoustic Systems. , 2021, , .		3
34	On the Design of Square Differential Microphone Arrays with a Multistage Structure. , 2021, , .		4
35	Identification of Linear and Bilinear Systems: A Unified Study. Electronics (Switzerland), 2021, 10, 1790.	3.1	18
36	A Tensorial Affine Projection Algorithm. , 2021, , .		3

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37	On the Stochastic Modeling of the NLMS Algorithm Operating with Bilinear forms. , 2021, , .		0
38	An Insightful Overview of the Wiener Filter for System Identification. Applied Sciences (Switzerland), 2021, 11, 7774.	2.5	3
39	Adaptive line enhancer for nonstationary harmonic noise reduction. Computer Speech and Language, 2021, 70, 101245.	4.3	0
40	A New Class of Differential Beamformers. IEEE/ACM Transactions on Audio Speech and Language Processing, 2021, 29, 594-606.	5.8	3
41	Time Difference of Arrival Estimation Based on a Kronecker Product Decomposition. IEEE Signal Processing Letters, 2021, 28, 51-55.	3.6	20
42	On a Particular Family of Differential Beamformers With Cardioid-Like and No-Null Patterns. IEEE Signal Processing Letters, 2021, 28, 140-144.	3.6	6
43	Beamforming with First-Order Linear Difference Equations. Springer Topics in Signal Processing, 2021, , 23-65.	0.2	1
44	An Iterative Multichannel Wiener Filter Based on a Kronecker Product Decomposition. , 2021, , .		3
45	Robust Dereverberation With Kronecker Product Based Multichannel Linear Prediction. IEEE Signal Processing Letters, 2021, 28, 101-105.	3.6	23
46	Binaural Heterophasic Superdirective Beamforming. Sensors, 2021, 21, 74.	3.8	3
47	Quadratic Beamforming for Magnitude Estimation. , 2021, , .		1
48	Robust Differential Beamforming with Rectangular Arrays. , 2021, , .		6
49	Improved Affine Projection Algorithm for the Identification of Multilinear Forms. , 2021, , .		0
50	On the Identification of Symmetric and Antisymmetric Impulse Responses. , 2021, , .		3
51	An efficient Kalman filter for the identification of low-rank systems. Signal Processing, 2020, 166, 107239.	3.7	19
52	Design of Planar Differential Microphone Arrays With Fractional Orders. IEEE/ACM Transactions on Audio Speech and Language Processing, 2020, 28, 116-130.	5.8	28
53	A class of multichannel sparse linear prediction algorithms for time delay estimation of speech sources. Signal Processing, 2020, 169, 107395.	3.7	5
54	Kronecker Product Beamforming with Multiple Differential Microphone Arrays. , 2020, , .		13

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55	A Multichannel Recursive Least-Squares Algorithm Based on a Kronecker Product Decomposition. , 2020, , .		4
56	Heterophasic Binaural Differential Beamforming for Speech Intelligibility Improvement. IEEE Transactions on Vehicular Technology, 2020, 69, 13497-13509.	6.3	7
57	An Iterative Wiener Filter for the Identification of Multilinear Forms. , 2020, , .		6
58	Continuously steerable differential beamformers with null constraints for circular microphone arrays. Journal of the Acoustical Society of America, 2020, 148, 1248-1258.	1.1	17
59	Steerable differential beamformers with planar microphone arrays. Eurasip Journal on Audio, Speech, and Music Processing, 2020, 2020, .	2.1	8
60	Robust and steerable kronecker product differential beamforming With rectangular microphone arrays. , 2020, , .		19
61	Beamforming With Small-Spacing Microphone Arrays Using Constrained/Generalized LASSO. IEEE Signal Processing Letters, 2020, 27, 356-360.	3.6	8
62	Robust Frequency-Domain Recursive Least M-Estimate Adaptive Filter For Acoustic System Identification. , 2020, , .		1
63	A Recursive Least-Squares Algorithm for the Identification of Trilinear Forms. Algorithms, 2020, 13, 135.	2.1	11
64	A Simple Theory and New Method of Differential Beamforming With Uniform Linear Microphone Arrays. IEEE/ACM Transactions on Audio Speech and Language Processing, 2020, 28, 1079-1093.	5.8	50
65	Differential Beamforming on Graphs. IEEE/ACM Transactions on Audio Speech and Language Processing, 2020, 28, 901-913.	5.8	16
66	Joint Sparse Concentric Array Design for Frequency and Rotationally Invariant Beampattern. IEEE/ACM Transactions on Audio Speech and Language Processing, 2020, 28, 1143-1158.	5.8	12
67	An Improved Solution to the Frequency-Invariant Beamforming with Concentric Circular Microphone Arrays. , 2020, , .		1
68	LMS Algorithms for Multilinear Forms. , 2020, , .		5
69	Quadratic approach for single-channel noise reduction. Eurasip Journal on Audio, Speech, and Music Processing, 2020, 2020, .	2.1	2
70	Adaptive and hybrid Kronecker product beamforming for far-field speech signals. Speech Communication, 2020, 120, 42-52.	2.8	5
71	On the Performance of LMS-Based Algorithms for the Identification of Low-Rank Systems. , 2020, , .		0
72	A Regularized RLS Algorithm for the Identification of Third-Order Tensors. , 2020, , .		1

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73	A Proportionate Affine Projection Algorithm for the Identification of Sparse Bilinear Forms. , 2019, , .		3
74	On the Design of Flexible Kronecker Product Beamformers with Linear Microphone Arrays. , 2019, , .		14
75	An Iterative Wiener Filter for the Identification of Trilinear Forms. , 2019, , .		2
76	An Optimized Differential Step-Size LMS Algorithm. Algorithms, 2019, 12, 147.	2.1	4
77	Nonlinear Kronecker product filtering for multichannel noise reduction. Speech Communication, 2019, 114, 49-59.	2.8	9
78	Differential Kronecker Product Beamforming. IEEE/ACM Transactions on Audio Speech and Language Processing, 2019, 27, 892-902.	5.8	29
79	System Identification Based on Tensor Decompositions: A Trilinear Approach. Symmetry, 2019, 11, 556.	2.2	17
80	On the Design of Target Beampatterns for Differential Microphone Arrays. IEEE/ACM Transactions on Audio Speech and Language Processing, 2019, 27, 1295-1307.	5.8	12
81	Properties and Limits of the Minimum-norm Differential Beamformers with Circular Microphone Arrays. , 2019, , .		7
82	Design of Optimal Linear Differential Microphone Arrays Based Array Geometry Optimization. , 2019, , .		3
83	A Recursive Least-squares Algorithm Based on the Nearest Kronecker Product Decomposition. , 2019, , .		10
84	Approach with Nonuniform Linear Arrays. Springer Topics in Signal Processing, 2019, , 113-145.	0.2	0
85	Spatiotemporal Signal Enhancement. Springer Topics in Signal Processing, 2019, , 169-186.	0.2	0
86	Recursive Least-Squares Algorithms for the Identification of Low-Rank Systems. IEEE/ACM Transactions on Audio Speech and Language Processing, 2019, 27, 903-918.	5.8	61
87	Array Processing. Springer Topics in Signal Processing, 2019, , .	0.2	47
88	On Robust and High Directive Beamforming With Small-Spacing Microphone Arrays for Scattered Sources. IEEE/ACM Transactions on Audio Speech and Language Processing, 2019, 27, 842-852.	5.8	15
89	On the design of time-domain differential microphone arrays. Applied Acoustics, 2019, 148, 212-222.	3.3	11
90	Incoherent Synthesis of Sparse Arrays for Frequency-Invariant Beamforming. IEEE/ACM Transactions on Audio Speech and Language Processing, 2019, 27, 482-495.	5.8	18

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91	Acoustic Source Localization Based on Geometric Projection in Reverberant and Noisy Environments. IEEE Journal on Selected Topics in Signal Processing, 2019, 13, 143-155.	10.8	21
92	Adaptive filtering for the identification of bilinear forms. , 2018, 75, 153-167.		45
93	Frequency-Domain Design of Asymmetric Circular Differential Microphone Arrays. IEEE/ACM Transactions on Audio Speech and Language Processing, 2018, 26, 760-773.	5.8	24
94	Beamforming based on null-steering with small spacing linear microphone arrays. Journal of the Acoustical Society of America, 2018, 143, 2651-2665.	1.1	8
95	Adaptive Beamforming. Springer Briefs in Electrical and Computer Engineering, 2018, , 103-117.	0.5	0
96	Single-Channel Speech Enhancement in the STFT Domain. Springer Briefs in Electrical and Computer Engineering, 2018, , 37-57.	0.5	2
97	Multichannel Speech Enhancement in the STFT Domain. Springer Briefs in Electrical and Computer Engineering, 2018, , 79-101.	0.5	1
98	A Connection Between the Kalman Filter and an Optimized LMS Algorithm for Bilinear Forms. Algorithms, 2018, 11, 211.	2.1	7
99	On Speech Enhancement Using Microphone Arrays in the Presence of Co-Directional Interference. , 2018, , .		4
100	Asymmetric Supercardioid Beamforming Using Circular Microphone Arrays. , 2018, , .		1
101	Low-Complexity RLS Algorithms for the Identification of Bilinear Forms. , 2018, , .		2
102	Microphone array beamforming based on maximization of the front-to-back ratio. Journal of the Acoustical Society of America, 2018, 144, 3450-3464.	1.1	6
103	Differential Beamformers Derived from Approximate Performance Measures. , 2018, , .		1
104	On the Design of Optimal Linear Microphone Array Geometries. , 2018, , .		4
105	Dereverberation with Differential Microphone Arrays and the Weighted-Prediction-Error Method. , 2018, , .		12
106	On the Design of Robust Steerable Frequency-Invariant Beampatterns with Concentric Circular Microphone Arrays. , 2018, , .		11
107	Regularized Recursive Least-Squares Algorithms for the Identification of Bilinear Forms. , 2018, , .		0
108	An Optimized LMS Algorithm for Bilinear Forms. , 2018, , .		1

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109	Efficient recursive least-squares algorithms for the identification of bilinear forms. , 2018, 83, 280-296.		27
110	A Single-Channel Noise Reduction Filtering/Smoothing Technique in the Time Domain. , 2018, , .		3
111	Identification of Bilinear Forms with the Kalman Filter. , 2018, , .		10
112	A flexible high directivity beamformer with spherical microphone arrays. Journal of the Acoustical Society of America, 2018, 143, 3024-3035.	1.1	29
113	Noise Robust Frequency-Domain Adaptive Blind Multichannel Identification With $ell _p$-Norm Constraint. IEEE/ACM Transactions on Audio Speech and Language Processing, 2018, 26, 1608-1619.	5.8	5
114	A Proportionate NLMS Algorithm for the Identification of Sparse Bilinear Forms. , 2018, , .		3
115	Insights Into Frequency-Invariant Beamforming With Concentric Circular Microphone Arrays. IEEE/ACM Transactions on Audio Speech and Language Processing, 2018, 26, 2305-2318.	5.8	69
116	On the design of differential beamformers with arbitrary planar microphone array geometry. Journal of the Acoustical Society of America, 2018, 144, EL66-EL70.	1.1	27
117	Linear System Identification Based on a Kronecker Product Decomposition. IEEE/ACM Transactions on Audio Speech and Language Processing, 2018, 26, 1793-1808.	5.8	80
118	Speech Enhancement Via Correlation Coefficients. Springer Briefs in Electrical and Computer Engineering, 2018, , 45-64.	0.5	0
119	On the Output SNR in Speech Enhancement and Beamforming. Springer Briefs in Electrical and Computer Engineering, 2018, , 65-81.	0.5	0
120	Speech Enhancement from the Fullband Output SNR Perspective. Springer Briefs in Electrical and Computer Engineering, 2018, , 83-104.	0.5	0
121	Robust variable-regularized RLS algorithms. , 2017, , .		12
122	On single-channel noise reduction with rank-deficient noise correlation matrix. Applied Acoustics, 2017, 126, 26-35.	3.3	6
123	On the Design of Frequency-Invariant Beampatterns With Uniform Circular Microphone Arrays. IEEE/ACM Transactions on Audio Speech and Language Processing, 2017, 25, 1140-1153.	5.8	106
124	Design of robust concentric circular differential microphone arrays. Journal of the Acoustical Society of America, 2017, 141, 3236-3249.	1.1	44
125	On the Identification of Bilinear Forms With the Wiener Filter. IEEE Signal Processing Letters, 2017, 24, 653-657.	3.6	78
126	Study of widely linear multichannel wiener filter for binaural noise reduction. , 2017, , .		2

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127	A minimum variance partially distortionless response filter for single-channel noise reduction. , 2017, , .		1
128	Distributed max-SINR speech enhancement with ad hoc microphone arrays. , 2017, , .		9
129	Robust multichannel TDOA estimation for speaker localization using the impulsive characteristics of speech spectrum. , 2017, , .		7
130	Study of the frequency-domain multichannel noise reduction problem with the householder transformation. , 2017, , .		1
131	Analysis of an LMS algorithm for bilinear forms. , 2017, , .		3
132	An NLMS algorithm for the identification of bilinear forms. , 2017, , .		5
133	An adaptive solution for nonlinear system identification. , 2017, , .		0
134	Asymmetric beampatterns with circular differential microphone arrays. , 2017, , .		2
135	An RLS algorithm for the identification of bilinear forms. , 2017, , .		2
136	A single-channel noise cancelation filter in the short-time-fourier-transform domain. , 2016, , .		3
137	Constrained Wiener gains and filters for single-channel and multichannel noise reduction. , 2016, , .		0
138	Single-channel noise reduction in the STFT domain from the fullband output SNR perspective. , 2016, , .		3
139	A family of optimized LMS-based algorithms for system identification. , 2016, , .		4
140	FPGA implementation of an optimized NLMS algorithm. , 2016, , .		2
141	Subspace superdirective beamformers based on joint diagonalization. , 2016, , .		14
142	Design of robust differential microphone arrays with the Jacobiâ€“Anger expansion. Applied Acoustics, 2016, 110, 194-206.	3.3	26
143	Reduced-Order Robust Superdirective Beamforming With Uniform Linear Microphone Arrays. IEEE/ACM Transactions on Audio Speech and Language Processing, 2016, 24, 1548-1559.	5.8	15
144	Fundamentals of Differential Beamforming. Springer Briefs in Electrical and Computer Engineering, 2016, , .	0.5	44

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145	Problem Formulation. Springer Briefs in Electrical and Computer Engineering, 2016, , 13-26.	0.5	0
146	Performance Measures Revisited. Springer Briefs in Electrical and Computer Engineering, 2016, , 41-50.	0.5	0
147	Conventional Optimization. Springer Briefs in Electrical and Computer Engineering, 2016, , 51-79.	0.5	0
148	Beampattern Design. Springer Briefs in Electrical and Computer Engineering, 2016, , 81-110.	0.5	0
149	Joint Optimization. Springer Briefs in Electrical and Computer Engineering, 2016, , 111-120.	0.5	0
150	A partitioned approach to signal separation with microphone ad hoc arrays. , 2016, , .		2
151	Variable span filters for speech enhancement. , 2016, , .		1
152	On the numerical properties of an optimized NLMS algorithm. , 2016, , .		1
153	First-order differential microphone arrays from a time-domain broadband perspective. , 2016, , .		12
154	Subspace superdirective beamforming with uniform circular microphone arrays. , 2016, , .		7
155	Multichannel time delay estimation for acoustic source localization via robust adaptive blind system identification. , 2016, , .		2
156	A multiframe parametric wiener filter for acoustic echo suppression. , 2016, , .		4
157	A tunable beamformer for robust superdirective beamforming. , 2016, , .		3
158	Robust superdirective beamformer with optimal regularization. , 2016, , .		2
159	Superdirective Beamforming Based on the Krylov Matrix. IEEE/ACM Transactions on Audio Speech and Language Processing, 2016, 24, 2531-2543.	5.8	27
160	Ad hoc microphone array beamforming using the primal-dual method of multipliers. , 2016, , .		7
161	On time delay estimation based on multichannel spatiotemporal sparse linear prediction. , 2016, , .		3
162	Single-channel noise reduction via semi-orthogonal transformations and reduced-rank filtering. Speech Communication, 2016, 78, 73-83.	2.8	6

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163	Design of Directivity Patterns with a Unique Null of Maximum Multiplicity. IEEE/ACM Transactions on Audio Speech and Language Processing, 2016, 24, 226-235.	5.8	12
164	Signal Enhancement with Variable Span Linear Filters. Springer Topics in Signal Processing, 2016, , .	0.2	20
165	A Framework for Speech Enhancement With Ad Hoc Microphone Arrays. IEEE/ACM Transactions on Audio Speech and Language Processing, 2016, 24, 1038-1051.	5.8	31
166	Noise Reduction with Optimal Variable Span Linear Filters. IEEE/ACM Transactions on Audio Speech and Language Processing, 2016, 24, 631-644.	5.8	38
167	An optimized NLMS algorithm for system identification. Signal Processing, 2016, 118, 115-121.	3.7	73
168	Multichannel Signal Enhancement in the STFT Domain. Springer Topics in Signal Processing, 2016, , 115-147.	0.2	0
169	General Concept with Filtering Matrices. Springer Topics in Signal Processing, 2016, , 25-39.	0.2	0
170	Multichannel Signal Enhancement in the Time Domain. Springer Topics in Signal Processing, 2016, , 75-113.	0.2	0
171	General Concept with Filtering Vectors. Springer Topics in Signal Processing, 2016, , 7-24.	0.2	0
172	Single-Channel Signal Enhancement in the STFT Domain. Springer Topics in Signal Processing, 2016, , 41-74.	0.2	1
173	Binaural Signal Enhancement in the Time Domain. Springer Topics in Signal Processing, 2016, , 149-163.	0.2	0
174	An optimized proportionate adaptive algorithm for sparse system identification. , 2015, , .		8
175	Design of robust differential microphone arrays with orthogonal polynomials. Journal of the Acoustical Society of America, 2015, 138, 1079-1089.	1.1	22
176	Direction-of-arrival estimation of passive acoustic sources in reverberant environments based on the Householder transformation. Journal of the Acoustical Society of America, 2015, 138, 3053-3060.	1.1	15
177	An overview on optimized NLMS algorithms for acoustic echo cancellation. Eurasip Journal on Advances in Signal Processing, 2015, 2015, .	1.7	34
178	A low complexity reweighted proportionate affine projection algorithm with memory and row action projection. Eurasip Journal on Advances in Signal Processing, 2015, 2015, .	1.7	2
179	Combined Beamformers for Robust Broadband Regularized Superdirective Beamforming. IEEE/ACM Transactions on Audio Speech and Language Processing, 2015, , 1-1.	5.8	32
180	Pseudo-coherence-based MVDR beamformer for speech enhancement with ad hoc microphone arrays. , 2015, , .		14

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181	Investigation of a parametric gain approach to single-channel speech enhancement. , 2015, , .		2
182	An optimized affine projection algorithm for acoustic echo cancellation. , 2015, , .		4
183	Design of Circular Differential Microphone Arrays. Springer Topics in Signal Processing, 2015, , .	0.2	83
184	Study of nonuniform linear differential microphone arrays with the minimum-norm filter. Applied Acoustics, 2015, 98, 62-69.	3.3	18
185	A multistage minimum variance distortionless response beamformer for noise reduction. Journal of the Acoustical Society of America, 2015, 137, 1377-1388.	1.1	9
186	On the performance of an optimized NLMS algorithm. , 2015, , .		0
187	Optimal single-channel noise reduction filtering matrices from the pearson correlation coefficient perspective. , 2015, , .		2
188	Optimal design of directivity patterns for endfire linear microphone arrays. , 2015, , .		8
189	An optimized NLMS algorithm for acoustic echo cancellation. , 2015, , .		1
190	Theoretical Analysis of Differential Microphone Array Beamforming and an Improved Solution. IEEE/ACM Transactions on Audio Speech and Language Processing, 2015, 23, 2093-2105.	5.8	50
191	On a multichannel maximum SNR filter for noise reduction in the STFT domain. , 2015, , .		0
192	Binaural Noise Reduction in the Time Domain. Springer Briefs in Electrical and Computer Engineering, 2015, , 51-65.	0.5	0
193	Design of Second-Order Circular Differential Arrays. Springer Topics in Signal Processing, 2015, , 53-80.	0.2	1
194	Superdirective Beamforming with Circular Arrays. Springer Topics in Signal Processing, 2015, , 91-111.	0.2	0
195	Design of Circular Differential Arrays with the Jacobi-Anger Expansion. Springer Topics in Signal Processing, 2015, , 143-164.	0.2	0
196	Minimum-Norm Solution for Robust Circular Differential Arrays. Springer Topics in Signal Processing, 2015, , 113-142.	0.2	0
197	Single-Channel Speech Enhancement in the Time Domain. , 2014, , 47-63.		0
198	General Concept with the Diagonalization of the Speech Correlation Matrix. , 2014, , 11-27.		0

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199	Multichannel Speech Enhancement in the Time Domain. , 2014, , 65-78.		0
200	On the noisereduction performance of the MVDR beamformer innoisy and reverberant environments. , 2014, , .		6
201	On the design and implementation of linear differential microphone arrays. Journal of the Acoustical Society of America, 2014, 136, 3097-3113.	1.1	86
202	Joint Spatio-Temporal Filtering Methods for DOA and Fundamental Frequency Estimation. IEEE/ACM Transactions on Audio Speech and Language Processing, 2014, , 1-1.	5.8	8
203	A reduced-rank approach to single-channel noise reduction. , 2014, , .		2
204	Robust blind identification of room acoustic channels in symmetric alpha-stable distributed noise environments. Journal of the Acoustical Society of America, 2014, 136, 693-704.	1.1	16
205	User determined superdirective beamforming. , 2014, , .		1
206	A Family of Maximum SNR Filters for Noise Reduction. IEEE/ACM Transactions on Audio Speech and Language Processing, 2014, 22, 2034-2047.	5.8	23
207	A Kalman filter with individual control factors for echo cancellation. , 2014, , .		7
208	Examples of optimal noise reduction filters derived from the squared Pearson correlation coefficient. , 2014, , .		3
209	A brief overview of speech enhancement with linear filtering. Eurasip Journal on Advances in Signal Processing, 2014, 2014, .	1.7	3
210	A practical variable forgetting factor recursive least-squares algorithm. , 2014, , .		20
211	Single-channel noise reduction using unified joint diagonalization and optimal filtering. Eurasip Journal on Advances in Signal Processing, 2014, 2014, .	1.7	11
212	A Kalman filter for stereophonic acoustic echo cancellation. , 2014, , .		1
213	Noise reduction in the time domain using joint diagonalization. , 2014, , .		3
214	Widely linear general Kalman filter for stereophonic acoustic echo cancellation. Signal Processing, 2014, 94, 570-575.	3.7	20
215	Performance Study of the MVDR Beamformer as a Function of the Source Incidence Angle. IEEE/ACM Transactions on Audio Speech and Language Processing, 2014, 22, 67-79.	5.8	66
216	A practical solution for the regularization of the affine projection algorithm. , 2014, , .		3

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217	Design of Robust Differential Microphone Arrays. IEEE/ACM Transactions on Audio Speech and Language Processing, 2014, 22, 1455-1466.	5.8	46
218	A Bayesian Framework for Blind Adaptive Beamforming. IEEE Transactions on Signal Processing, 2014, 62, 2370-2384.	5.3	12
219	Multichannel Noise Reduction in the Karhunen-LoÃ¨ve Expansion Domain. IEEE/ACM Transactions on Audio Speech and Language Processing, 2014, 22, 923-936.	5.8	5
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222	Evaluation of the Time-Domain Speech Enhancement Filters. , 2014, , 107-131.		0
223	A minimum variance distortionless response filter based on the bifrequency spectrum for single-channel noise reduction. , 2014, 33, 169-179.		14
224	A multichannel widely linearwiener filter for binaural noise reduction in the short-time-fourier-transform domain. , 2014, , .		3
225	General Concept with the Joint Diagonalization of the Speech and Noise Correlation Matrices. , 2014, , 29-46.		0
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