Chengshi Zheng

List of Publications by Year
in descending order

60 papers	688 citations	759233 12 h-index	713466 21 g-index
62 all docs	62 docs citations	62 times ranked	255 citing authors

#	Article	IF	Citations
1	Glance and gaze: A collaborative learning framework for single-channel speech enhancement. Applied Acoustics, 2022, 187, 108499.	3.3	57
2	Measurement and modeling of the mechanical impedance of human mastoid and condyle. Journal of the Acoustical Society of America, 2022, 151, 1434-1448.	1.1	6
3	Noise-robust blind reverberation time estimation using noise-aware timeâ€"frequency masking. Measurement: Journal of the International Measurement Confederation, 2022, 192, 110901.	5.0	4
4	A separation and interaction framework for causal multi-channel speech enhancement. , 2022, 126, 103519.		4
5	Embedding and Beamforming: All-Neural Causal Beamformer for Multichannel Speech Enhancement. , 2022, , .		17
6	Joint Magnitude Estimation and Phase Recovery Using Cycle-In-Cycle GAN for Non-Parallel Speech Enhancement. , 2022, , .		6
7	Dual-Branch Attention-In-Attention Transformer for Single-Channel Speech Enhancement. , 2022, , .		28
8	A Neural Beamspace-Domain Filter for Real-Time Multi-Channel Speech Enhancement. Symmetry, 2022, 14, 1081.	2.2	3
9	Low-latency monaural speech enhancement with deep filter-bank equalizer. Journal of the Acoustical Society of America, 2022, 151, 3291-3304.	1.1	4
10	Filtering and Refining: A Collaborative-Style Framework for Single-Channel Speech Enhancement. IEEE/ACM Transactions on Audio Speech and Language Processing, 2022, 30, 2156-2172.	5.8	6
11	Taylor, Can You Hear Me Now? A Taylor-Unfolding Framework for Monaural Speech Enhancement. , 2022, , .		7
12	On the importance of power compression and phase estimation in monaural speech dereverberation. JASA Express Letters, 2021, 1, .	1.1	49
13	Two Heads are Better Than One: A Two-Stage Complex Spectral Mapping Approach for Monaural Speech Enhancement. IEEE/ACM Transactions on Audio Speech and Language Processing, 2021, 29, 1829-1843.	5.8	73
14	Low-complexity artificial noise suppression methods for deep learning-based speech enhancement algorithms. Eurasip Journal on Audio, Speech, and Music Processing, 2021, 2021, .	2.1	6
15	ICASSP 2021 Acoustic Echo Cancellation Challenge: Integrated Adaptive Echo Cancellation with Time Alignment and Deep Learning-Based Residual Echo Plus Noise Suppression. , 2021, , .		11
16	Investigation of an MAA Test With Virtual Sound Synthesis. Frontiers in Psychology, 2021, 12, 656052.	2.1	2
17	Finite data performance analysis of one-bit MVDR and phase-only MVDR. Signal Processing, 2021, 183, 108018.	3.7	4
18	ICASSP 2021 Deep Noise Suppression Challenge: Decoupling Magnitude and Phase Optimization with a Two-Stage Deep Network. , 2021, , .		29

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19	An optimization framework for designing robust cascade biquad feedback controllers on active noise cancellation headphones. Applied Acoustics, 2021, 179, 108081.	3.3	13
20	Distributed node-specific block-diagonal LCMV beamforming in wireless acoustic sensor networks. Signal Processing, 2021, 185, 108085.	3.7	4
21	Deep learning-based stereophonic acoustic echo suppression without decorrelation. Journal of the Acoustical Society of America, 2021, 150, 816-829.	1.1	9
22	A two-stage complex network using cycle-consistent generative adversarial networks for speech enhancement. Speech Communication, 2021, 134, 42-54.	2.8	9
23	A Low-Complexity Volterra Filtered-Error LMS Algorithm with a Kronecker Product Decomposition. Applied Sciences (Switzerland), 2021, 11, 9637.	2.5	1
24	A temporal-spectral generative adversarial network based end-to-end packet loss concealment for wideband speech transmission. Journal of the Acoustical Society of America, 2021, 150, 2577-2588.	1.1	12
25	Learning to Inference with Early Exit in the Progressive Speech Enhancement. , 2021, , .		3
26	Wideband sparse Bayesian learning for off-grid binaural sound source localization. Signal Processing, 2020, 166, 107250.	3.7	9
27	The effect of pinna filtering in binaural transfer functions on externalization in a reverberant environment. Applied Acoustics, 2020, 164, 107257.	3.3	3
28	A Supervised Speech Enhancement Approach with Residual Noise Control for Voice Communication. Applied Sciences (Switzerland), 2020, 10, 2894.	2.5	7
29	Speech enhancement using progressive learning-based convolutional recurrent neural network. Applied Acoustics, 2020, 166, 107347.	3.3	48
30	Joint estimation of binaural distance and azimuth by exploiting deep neural networks. Journal of the Acoustical Society of America, 2020, 147, 2625-2635.	1.1	6
31	Evaluation of headphone phase equalization on sound reproduction. Applied Acoustics, 2019, 156, 208-216.	3.3	2
32	Guided spectrogram filtering for speech dereverberation. Applied Acoustics, 2018, 134, 154-159.	3.3	7
33	Statistical Analysis of the Multichannel Wiener Filter Using a Bivariate Normal Distribution for Sample Covariance Matrices. IEEE/ACM Transactions on Audio Speech and Language Processing, 2018, 26, 951-966.	5.8	12
34	A perceptually motivated LP residual estimator in noisy and reverberant environments. Speech Communication, 2018, 96, 129-141.	2.8	6
35	An efficient and robust speech dereverberation method using spherical microphone array. , 2018, , .		0
36	Robust Adaptive Beamforming Using Noise Reduction Preprocessing-Based Fully Automatic Diagonal Loading and Steering Vector Estimation. IEEE Access, 2017, 5, 12974-12987.	4.2	27

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37	Stereophonic channel decorrelation using a binaural masking model. Applied Acoustics, 2016, 110, 128-136.	3.3	1
38	Analysis of Additional Stable Gain by Frequency Shifting for Acoustic Feedback Suppression using Statistical Room Acoustics. IEEE Signal Processing Letters, 2016, 23, 159-163.	3.6	3
39	Bandwidth extension for speech acquired by laser Doppler vibrometer with an auxiliary microphone. , 2015, , .		4
40	Active Headrest with Robust Performance against Head Movement. Journal of Low Frequency Noise Vibration and Active Control, 2015, 34, 233-250.	2.9	19
41	Binaural coherent-to-diffuse-ratio estimation for dereverberation using an ITD model. , 2015, , .		8
42	Speech quality evaluation of a sparse coding shrinkage noise reduction algorithm with normal hearing and hearing impaired listeners. Hearing Research, 2015, 327, 175-185.	2.0	13
43	Equalization of loudspeaker response using balanced model truncation. Journal of the Acoustical Society of America, 2015, 137, EL241-EL247.	1.1	6
44	On Generalized Auto-Spectral Coherence Function and Its Applications to Signal Detection. IEEE Signal Processing Letters, 2014, 21, 559-563.	3.6	11
45	Evaluation of the sparse coding shrinkage noise reduction algorithm in normal hearing and hearing impaired listeners. Hearing Research, 2014, 310, 36-47.	2.0	3
46	A Constrained MMSE LP Residual Estimator for Speech Dereverberation in Noisy Environments. IEEE Signal Processing Letters, 2014, 21, 1462-1466.	3.6	10
47	Twoâ€stage optimisation algorithm for adaptive IIR notch filter. Electronics Letters, 2014, 50, 985-987.	1.0	2
48	A statistical analysis of power-level-difference-based dual-channel post-filter estimator. Applied Acoustics, 2014, 83, 40-46.	3.3	0
49	A modified power-level-difference-based noise reduction for dual-microphone headsets. , 2013, , .		1
50	A Statistical Analysis of Two-Channel Post-Filter Estimators in Isotropic Noise Fields. IEEE Transactions on Audio Speech and Language Processing, 2013, 21, 336-342.	3.2	10
51	A cepstrum-based preprocessing and postprocessing for speech enhancement in adverse environments. Applied Acoustics, 2013, 74, 1458-1462.	3.3	16
52	Spectral subtraction based on two-stage spectral estimation and modified cepstrum thresholding. Applied Acoustics, 2013, 74, 450-458.	3.3	13
53	Detection of multiple sinusoids in unknown colored noise using truncated cepstrum thresholding and local signal-to-noise-ratio. Applied Acoustics, 2012, 73, 809-816.	3.3	9
54	On second-order statistics of log-periodogram and cepstral coefficients for processes with mixed spectra. Signal Processing, 2012, 92, 2560-2565.	3.7	4

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55	Robustness analysis of time-domain and frequency-domain adaptive null-forming schemes. , 2011, , .		1
56	Optimal smoothing for microphone array post-filtering under a combined deterministic-stochastic hybrid model. Journal of Electronics, 2011, 28, 524-530.	0.2	0
57	Two-channel post-filtering based on adaptive smoothing and noise properties. , 2011, , .		10
58	Acoustical Vehicle Detection Based on Bispectral Entropy. IEEE Signal Processing Letters, 2009, 16, 378-381.	3.6	5
59	On the relationship of non-parametric methods for coherence function estimation. Signal Processing, 2008, 88, 2863-2867.	3.7	26
60	A Recursive Network with Dynamic Attention for Monaural Speech Enhancement. , 0, , .		19