Mamoru IWAKI, Ph.D.

Associate Professor
Program: Electrical and Information Engineering
Area: Human Science and Assistive Technology

Professional Expertise

His professional expertise encompasses speech and human audio information processing, digital signal processing, approximation and interpolation theory. He and his group have developed researches from engineering, psychological and physiological points of view. Signal processing technologies for audio, visual and biosignals are major topics in his laboratory. Psychological characteristics measurements are also main topics for developing useful assistive technology.

Research Fields of Interest

Speech and human audio

- Speech quality measurement and improvement, Speech analysis-synthesis system, Speech, speaker and emotion recognitions, Sound source segregation and integration, Auditory model, Psychoacoustics, Psychological and physical effect of hearing task on listeners, Mutual influence of visual and audio information, Information hiding for sound signals, Sound data compression

Digital signal processing

- Filter design, Sound source separation

Signal representation and analysis

- Spline functions, Natural observation theory

Education

1994: Doctor. Eng., University of Tsukuba
1991: Master Eng., University of Tsukuba
1989: Bachelor Eng., University of Tsukuba

Professional Societies and Activities

1. IEEE
2. The Institute of Electronics, Information and Communication Engineers (IEICE)
3. The Acoustical Society Japan (ASJ)
4. The Japanese Society for Wellbeing Science and Assistive Technology (JSWSAT)

Major Publications Papers

SPEECH AND HUMAN AUDIO

[1] "A Method of High Bit-rate Data Hiding in Music using Spline Interpolation", Proc. IEEE Int'l Conf. Intelligent Information Hiding and Multimedia Signal Processing, IIHMSP-2006-IS01-003, pp.11-14, 2006.

[2] "Frequency Dependency of Illusory Continuity in Binaural Hearing with Respect to Interaural Time Difference", JASA, vol.120, iss.5, pp.3082-3082, 2006.

[3] "Binaural integration of neural activity patterns using a coincidence gate", British Soc. Audiology, Short Papers Meeting on Experimental Studies of Hearing and Deafness, vol.53, pp.105 – 106, 2006.

[4] "An Internet-Based Cycle Ergometer Health Promotion System for Providing Personally Fitted Exercise", IEICE Trans. Information and Systems, vol. E88-D, no.8, pp.1985-1992, 2005.

[5] "Speaker Individuality in Excitation Pattern of Single Vowel", Proc. 18th Symp. Biol. and Physiol. Eng., 3P-2, pp.363-364, 2003.
[6] "Illusory Continuity of Intermittent Pure Tone in Binaural Listening and Its Dependency on Interaural Time Difference", Eurospeech, pp.2065-2068, 2003.
[7] "An Effect of Amplitude Modulation on Perceptual Segregation of Tone Sequences", ICSLP, vol.1, pp.657-660, 2002.
[8] "Auditory Perception of Amplitude Modulated Sinusoid Using A Pure Tone and Band-limited Noises as Modulation Signals", ICSLP, pp.610-613, 2000.
[9] "A Vocal Tract Area Ratio Estimation From Spectral Parameter Extracted by STRAIGHT", ICSLP, pp.596-599, 2000.
[10] "Fundamental Frequency Fluctuation in Continuous Vowel Utterance and its Perception", ICSLP, vol.4, no.27, pp.1519-1522, 1998.
[11] "Spectral Sequence Compensation Based on Continuity of Spectral Sequence", ICSLP, vol.4, no.28, pp.1407-1410, 1998.

DIGITAL SIGNAL PROCESSING

[12] "Accurate Estimation of Minimum Filter Length for Optimum FIR Digital Filters", IEEE Trans. Circuits and Systems II, Oct-47, pp.1008-1016, 2000.
[13] "A Fundamental Study on Instantaneous Characteristics of Natural Observation Transform of Normal Type", IEICE Trans., volJ81-A, no.11, pp.1575-1582, 1998.
[14] "A New Estimation Formula For Minimum Filter Length of Optimum FIR Digital Filters", Proc. Int'l Conf. Signal Processing, vol.1, pp.89-92, 1998.
[15] "A Simple Design Method of FIR Digital Filters Based on Linear Phase Decomposition", Proc. IASTED Int'l Conf. Signal and Image Processing, pp.693-697, 1998.
[16] "Uncertainty in Natural Observation Theory of Normal Type and an Optimum Value of Time Constant Parameter", IEICE Trans., vol. J81-A, no.4, pp.743-750, 1998.
[17] "A New Estimation Formula for Minimum Filter Length of Optimum FIR Low-Pass Digital Filters", Proc. IEEE Int'l Conf. Information, Communication and Signal Processing, vol.3, 3B1.4, pp.1303-1307, 1997.
[18] "On Uncertainty Principle in Natural Observation Transform of Normal Type", IEICE Trans., vol. J80-A, no.1, pp.228-236, 1997.
[19] "On Uncertainty Principle in Natural Observation Transform of Neighboring Type", IEICE Trans., vol.J79-A, no.11, pp.1886-1893, 1996.
[20] "Consideration on Derivation Principles of Natural Observation Method of Discrete Type", IEICE Trans., vol.J79-A, no.11, pp.1894-1898, 1996.
[21] "Description of FIR digital filters in the form of parallel connection of linear phase FIRs", Electronics Letters, Nov-32, pp.979-980, 1996.
[22] "Natural Observation Method for Discrete-Time Waveforms", IEICE Trans., vol.J79-A, no.3, pp.728-735, 1996.
[23] "Fundamental Theory of Natural Observation Method with Complete Reconstruction Property by Finite Sum -Natural Observation Theory of Normal Type-", IEICE Trans., vol.J79-A, no.1, pp.77-87, 1996.

SIGNAL REPRESENTATION AND ANALYSIS

[24] "Sampling Theorem for Spline Signal Space of Arbitrary Degree", IEICE Trans., vol.E77-A, no.5, pp.810-817, 1994.
[25] "Method of Designing Windows Based on Spline Approximation", Int'l J Systems Science, Aug.-24, pp.1539-1549, 1993.
[26] "A Fast Polynomial Interpolation for Remez Exchange Method", Proc. IEEE Pacific Rim Conf. Communications, Computers and Signal Processing, vol.2, pp.411-414, 1993.
[27] "A Series of Discrete-time Models of A Continuous-time System Based on Fluency Signal Approximation", Proc. IEEE Pacific Rim Conf. Communications, Computers and Signal Processing, vol.2, pp.493-496, 1993.
[28] "A Design Method of Windows by Spline Functions Based on Weighted Equiripple Approximation", IEEJ Trans, vol.113-C, no.4, pp.254-260, 1993.
[29] "Polynomial Interpolation for Remez Exchange Method", Electronics Letters, vol.28, no.20, pp.1900-1902, 1992.
[30] "Contribution of Ultrasound to Timbre", Proc. Int'l Symposium on Musical Acoustics, pp.127-130, 1992.
[31] "Periodic Sampling Basis and Its Biorthonormal Basis for the Signal Spaces of Piecewise Polynomials", IEICE Trans, vol.J75-A, no.6, pp.1003-1012, 1992.
[32] "On Pseudo High Vision Systems using Time-Varying Two Variate Splines", Proc. Int'l Symposium on Biotelemetry, pp.263-267, 1990.
[33] "Dual Sampling Theorem for Fluency Functions", Proc Symp. Applied Functional Analysis - Information Theory and Related Topics -, pp.11-19, 1990.
[34] "Biorthonormal Expansion in Signal Spaces Composed of Spline Functions", ICASSP, vol.2, pp.1119-1122, 1989.
[35] "A Method of Improving Image Quality for Medical Video Hardcopy", ICASSP, vol.3, pp.1500-1503, 1989.