References

[1] 3GPP TR 26.976. AMR-WB speech codec performance characterisation, December 2001.

[2] 3GPP TS 26.171. AMR wideband speech codec; general description, March 2001.

[3] 3GPP TS 26.190. AMR wideband speech codec; transcoding functions, December 2001.

[4] R.M. Aarts. *On the Design and Psychophysical Assessment of Loudspeaker Systems*. PhD thesis, Delft University of Technology, 1995.

[5] R.M. Aarts. Low-complexity tracking and estimation of frequency and amplitude of sinusoids. *Digital Signal Processing*, **14**(4):372–378, July 2004.

[6] R.M. Aarts and R.T. Dekkers. A real-time speech-music discriminator. *J. Audio Eng. Soc.*, **47**(9):720–725, 1999.

[7] R.M. Aarts, H. Greten, and P. Swarte. A special form of noise reduction. In *Proceedings of 21st AES Conference*, St. Petersburg, Russia. Audio Engineering Society, 2002.

[8] R.M. Aarts, R. Irwan, and A.J.E.M. Janssen. Efficient tracking of the cross-correlation coefficient. *IEEE Trans. Speech Audio Process.*, **10**(6):391–402, 2002.

[9] R.M. Aarts and A.J.E.M. Janssen. Approximation of the struve function H1 occurring in impedance calculations. *J. Acoust. Soc. Am.*, **113**(5):2635–2637, 2003.

[10] R.M. Aarts, E. Larsen, and O. Ouweltjes. A unified approach to low- and high-frequency bandwidth extension. In *115th AES Convention*, New York. Audio Engineering Society, 2003.

[11] M. Abe and Y. Yoshida. More natural sounding voice quality over the telephone! *NTT Rev.*, **7**(3):104–109, 1995.

[12] M. Abramowitz and I.A. Stegun. *Handbook of Mathematical Functions*. Dover, 1972.

[13] A.J. Abrantes, J.S. Marques, and I.M. Trancoso. Hybrid sinusoidal modeling of speech without voicing decision. In *Proceedings of EUROSPEECH*, volume 1, pages 231–234, Genova, Italy, September 1991.

[14] R.M. Adelson. Frequency estimation from few measurements. *Digital Signal Process.*, **7**:47–54, 1997.

[15] R.M. Adelson. Rapid power-line frequency monitoring. *Digital Signal Process.*, **12**:1–11, 2002.

Audio Bandwidth Extension: Application of Psychoacoustics, Signal Processing and Loudspeaker Design. E. Larsen and R. M. Aarts

ISBN 0-470-85864-8

[16] A.M. Abdelatty Ali and J. Van der Spiegel. Acoustic-phonetic features for the automatic classification of fricatives. *J. Acoust. Soc. Am.*, **109**(5):2217–2235, 2001.

[17] A.M. Abdelatty Ali, J. Van der Spiegel, and P. Mueller. An acoustic-phonetic feature-based system for the automatic recognition of fricative consonants. In *Proceedings of ICASSP*, volume 2, pages 961–964, Seattle, WA, May 1998.

[18] J.B. Allen. Short-term spectral analysis, synthesis, modification by discrete Fourier transform. *IEEE Trans. Speech Audio Process.*, **25**:235–238, 1977.

[19] J.B. Allen and S.T. Neely. Modeling the relation between the intensity just-noticeable difference and loudness for pure tones and wideband noise. *J. Acoust. Soc. Am.*, **102**(6):3628–3646, 1997.

[20] American Standards Association. *Standard Acoustical Terminology (S1.1-1994 (ASA 111-1994))*. ASA, New York, 1994.

[21] G.B. Arfken and H.J. Weber. *Mathematical Methods for Physicists*. John Wiley & Sons, 4th edition, 1995.

[22] B.S. Atal and L.R. Rabiner. A pattern recognition approach to voiced-unvoiced-silence classification with applications to speech recognition. *IEEE Trans. Acoust. Speech Signal Process.*, **ASSP-24**(3):201–212, 1976.

[23] C. Avendano, H. Hermansky, and E.A. Wan. Beyond nyquist: towards the recovery of broad-bandwidth speech from narrow-bandwidth speech. In *Proceedings of EUROSPEECH*, volume 1, pages 165–168, Madrid, Spain, September 1995.

[24] L.R. Bahl, P.F. Brown, P.V. de Souza, and R.L. Mercer. Maximum mutual information estimation of hidden markov model parameters for speech recognition. In *Proceedings of ICASSP*, pages 49–52, Tokyo, Japan, April 1986.

[25] I. Barrodale and C. Phillips. An improved algorithm for discrete Chebyshev linear approximation. In *Proceedings of 4th Manitoba Conference on Numerical Mathematics*, pages 177–190. University of Manitoba, Canada, 1974.

[26] D.W. Batteau. The role of the pinna in human localization. *Proc. R. Soc. London B*, **168**:158–180, 1967.

[27] A.G. Bell. Telegraphy (Filed Morning of 14 February 1876), US Patent 174,465, Issued March 7, 1876.

[28] L.L. Beranek. *Acoustics*. McGraw-Hill, New York, 1954. (Reprinted by ASA 1986).

[29] A.J. Berkhout. Least-squares inverse filtering and wavelet deconvolution. *Geophysics*, **42**(7):1369–1383, 1977.

[30] A.J. Berkhout, D. de Vries, and M.M. Boone. A new method to acquire impulse responses in concert halls. *J. Acoust. Soc. Am.*, **68**(1):179–183, 1980.

[31] B. Bessette, R. Lefebvre, M. Jelínek, J. Rotola-Pukkila, H. Mikkola, and K. Järvinen. The adaptive multirate wideband speech codec (AMR-WB). *IEEE Trans. Speech Audio Process.*, **10**(8):620–636, 2002.

[32] F.A. Bilsen. *On the Interaction of a Sound with its Repititions*. PhD thesis, Delft University of Technology, 1968.

[33] F.A. Bilsen and R.J. Ritsma. Some parameters influencing the perceptibility of pitch. *J. Acoust. Soc. Am.*, **47**(2)(Part 2):469–475, 1970.

[34] J. Blauert. *Spatial Hearing. The Psychophysics of Human Sound Localization*. MIT Press, 2nd edition, 1984.

[35] P. Boersma and D. Weenink. http://www.fon.hum.uva.nl/praat/, University of Amsterdam, The Netherlands, Retrieved July 2003.

[36] J. Borwick, editor. *Loudspeaker and Headphone Handbook*. Butterworths, London, 1988.

[37] M. Bosi and R.E. Goldberg. *Introduction to Digital Audio Coding and Standards*. Kluwer Academic Publishers, 2003.

[38] A.S. Bregman. *Auditory Scene Analysis*. MIT Press, 1990.

[39] F.H. Brittain. The loudness of continuous spectrum noise and its application to loudness measurements. *J. Acoust. Soc. Am.*, **11**:113–117, 1939.

[40] A.W. Bronkhorst. Localization of real and virtual sound sources. *J. Acoust. Soc. Am.*, **98**:2542–2553, 1995.

[41] O. Brosze, K.O. Schmidt, and A. Schmoldt. Der Gewinn an Verständlichkeit beim "Fernsehsprechen". *Nachrichtentech. Z. (NTZ)*, **15**(7):349–352, 1962 (in German).

[42] D. Byrne, H. Dillon, A.S. Khanh Tran *et al.*. An international comparison of long-term average speech spectra. *J. Acoust. Soc. Am.*, **96**(4):2108–2120, 1994.

[43] J.P. Campbell and T.E. Tremain. Voiced/unvoiced classification of speech with applications to the U.S. government LPC-10E algorithm. In *Proceedings of ICASSP*, pages 473–476, Tokyo, Japan, April 1986.

[44] H. Carl. *Untersuchung Verschiedener Methoden der Sprachkodierung und eine Anwendung zur Bandbreitenvergrößerung von Schmalband-Sprachsignalen*. PhD thesis, Ruhr-Universität Bochum, Bochum, Germany, 1994 (in German).

[45] H. Carl and U. Heute. Bandwidth enhancement of narrow-band speech signals. In *Proceedings of EUSIPCO*, volume 2, pages 1178–1181, Edinburgh, Scotland, September 1994.

[46] J.D. Carroll and P. Arabie. Multidimensional scaling. *Annu. Rev. Psychol.*, **31**:607–649, 1980.

[47] C.-F. Chan and W.-K. Hui. Wideband re-synthesis of narrowband CELP coded speech using multiband excitation model. In *Proceedings of ICSLP*, volume 1, pages 322–325, Philadelphia, PA, October 1996.

[48] C.-F. Chan and W.-K. Hui. Quality enhancement of narrowband CELP-coded speech via wideband harmonic re-synthesis. In *Proceedings of ICASSP*, volume 2, pages 1187–1190, Munich, Germany, April 1997.

[49] Y.M. Cheng, D. O'Shaughnessy, and P. Mermelstein. Statistical recovery of wideband speech from narrowband speech. In *Proceedings of ICSLP*, pages 1577–1580, Edmonton, Canada, 1992.

[50] S. Chennoukh, A. Gerrits, G. Miet, and R. Sluijter. Speech enhancement via frequency bandwidth extension using line spectral frequencies. In *Proceedings of ICASSP*, volume 1, pages 665–668, Salt Lake City, UT, May 2001.

[51] D. Clark. High-resolution subjective testing using a double-blind comparator. *J. Audio Eng. Soc.*, **30**(5):330–338, 1982.

[52] T.M. Cover and J.A. Thomas. *Elements of Information Theory*. Wiley Series in Telecommunications, 1991.

[53] R.E. Crochiere and L.R. Rabiner. *Multirate Signal Processing*. Prentice Hall, Englewood Cliffs, NJ, 1983.

[54] M. G. Croll. Sound-quality improvement of broadcast telephone calls. Technical Report 1972/26, The British Broadcasting Corporation (BBC), 1972.

[55] C. Cuttriss and J. Redding. Telephone (Filed 28 November 1877), US Patent 2 42 816, Issued June 14, 1881.

[56] H.A. David. *The Method of Paired Comparisons*. Ch. Griffin, 2nd edition, London, 1988.

[57] S.B. Davis and P. Mermelstein. Comparison of parametric representations for monosyllabic word recognition in continuously spoken sentences. *IEEE Trans. Acoust. Speech Signal Process.*, **ASSP-28**(4):357–366, 1980.

[58] E. de Boer. *On the 'Residue' in Hearing*. PhD thesis, University of Amsterdam, 1956.

[59] A.P. Dempster, N.M. Laird, and D.B. Rubin. Maximum likelihood from incomplete data via the EM algorithm. *J. R. Stat. Soc. Ser. B*, **39**(1):1–38, 1977.

[60] M. Dietrich. Performance and implementation of a robust ADPCM algorithm for wideband speech coding with 64 kbit/s. In *Proceedings of International Zürich Seminar on Digital Communications*, Zürich, Switzerland, March 1984.

[61] M. Dietz, L. Liljeryd, K. Kjorling, and O. Kunz. Spectral band replication, a novel approach in audio coding. In *Proceedings of AES 112th Convention*, Paper 5553, Munich, Germany. Audio Engineering Society, 2002.

[62] A. Ehret, M. Dietz, and K. Kjorling. State-of-the-art audio coding for broadcasting and mobile applications. In *Proceedings of AES 114th Convention*, Paper 5834, Amsterdam, The Netherlands. Audio Engineering Society, 2003.

[63] G. Ekman. Dimensions of color vision. *J. Psychol.*, **38**:467–474, 1954.

[64] N. Enbom and W.B. Kleijn. Bandwidth expansion of speech based on vector quantization of the mel frequency cepstral coefficients. In *IEEE Speech Coding Workshop*, pages 171–173, Porvoo, Finland, September 1999.

[65] J. Epps. *Wideband Extension of Narrowband Speech for Enhancement and Coding*. PhD thesis, School of Electrical Engineering and Telecommunications, The University of New South Wales, 2000.

[66] J. Epps and W.H. Holmes. A new technique for wideband enhancement of coded narrowband speech. In *IEEE Speech Coding Workshop*, pages 174–176, Porvoo, Finland, September 1999.

[67] C. Erdmann, P. Vary, K. Fischer, W. Xu, M. Marke, T. Fingscheidt, I. Varga, M. Kaindl, C. Quinquis, B. Koevesi, and D. Massaloux. A candidate proposal for a 3GPP adaptive multi-rate wideband speech codec. In *Proceedings of ICASSP*, volume 2, pages 757–760, Salt Lake City, UT, May 2001.

[68] ETSI Rec. GSM 03.50. Digital cellular telecommunication system (phase 2+); transmission planning aspects of the speech service in the GSM public land mobile (PLMN) system. Version 8.1.1, 2000.

[69] ETSI Rec. GSM 06.10. GSM full rate speech transcoding. Version 3.2.0, February 1992.

[70] L.D. Fielder and E.M. Benjamin. Subwoofer performance for accurate reproduction of music. *J. Audio Eng. Soc.*, **36**(6):443–456, 1988.

[71] J.L. Flanagan. *Speech Analysis, Synthesis and Perception*. Springer-Verlag, 2nd edition, Berlin, Heidelberg, New York, 1972.

[72] H. Fletcher. Auditory patterns. *Rev. Mod. Phys.*, **12**:47–65, 1940.

[73] H. Fletcher and W.A. Munson. Loudness, its definition, measurement and calculation. *J. Acoust. Soc. Am.*, **5**:82–108, 1933.

[74] W. Flügge. *Viscoelasticity*. Blaisdell Publishing Company, 1967.

[75] F.J.M. Frankort. *Vibration and Sound Radiation of Loudspeaker Cones*. PhD thesis, Delft University of Technology, 1975.

[76] N.R. French and J.C. Steinberg. Factors governing the intelligibility of speech sounds. *J. Acoust. Soc. Am.*, **19**:90–119, 1947.

[77] J.A. Fuemmeler and R.C. Hardie. Techniques for the regeneration of wideband speech from narrowband speech. In *IEEE Workshop on Nonlinear Signal and Image Proceedings*, Baltimore, MD, June 2001.

[78] J.A. Fuemmeler, R.C. Hardie, and W.R. Gardner. Techniques for the regeneration of wideband speech from narrowband speech. *EURASIP J. Appl. Signal Process.*, **2001**(4):266–274, 2001.

[79] K. Fukunaga. *Introduction to Statistical Pattern Recognition*. Morgan Kaufmann, Academic Press, 2nd edition, San Francisco, San Diego, 1990.

[80] S. Furui. *Digital Speech Processing, Synthesis and Recognition*. Marcel Dekker, 1989.

[81] K.R. Gabriel. The biplot graphical display of matrices with application to principal component analysis. *Biometrika*, **58**:453–467, 1971.

[82] A. Gabrielsson and B. Lindström. Perceived sound quality of high-fidelity loudspeakers. *J. Audio Eng. Soc.*, **33**(1/2):33, 1985.

[83] W.S. Gan, S.M. Kuo, and C.W. Toh. Virtual bass for home entertainment, multimedia PC, game station and portable audio systems. *IEEE Trans. Cons. Electron.*, **47**(4):787–793, 2001.

[84] M.R. Gander. Fifty years of loudspeaker developments as viewed through the perspective of the audio engineering society. *J. Audio Eng. Soc.*, **46**(1/2):43–58, 1998.

[85] E. Geddes and L. Lee. *Audio Transducers*. 2002.

[86] C.D. Geisler. *From Sound to Synapse: Physiology of the Mammalian Ear*. Oxford University Press, 1998.

[87] A. Gersho and R.M. Gray. *Vector Quantization and Signal Compression*. Kluwer Academic Publishers, Boston, Dordrecht, London, 1992.

[88] G.A. Gescheider. Psychological scaling. *Annu. Rev. Psychol.*, **39**:169–200, 1988.

[89] B.R. Glasberg and B.C.J. Moore. Derivation of auditory filter shapes from notched-noise data. *Hear. Res.*, **47**:103–138, 1990.

[90] B.R. Glasberg and B.C.J. Moore. A model of loudness applicable to time-varying sounds. *J. Audio Eng. Soc.*, **50**(5):331–342, 2002.

[91] J.L. Goldstein. Auditory nonlinearity. *J. Acoust. Soc. Am.*, **41**(3):676–689, 1967.

[92] J.L. Goldstein. An optimum processor theory for the central formation of the pitch of complex tones. *J. Acoust. Soc. Am.*, **54**(6):1496–1516, 1973.

[93] G.H. Golub and C.F. van Loan. *Matrix Computations*. Johns Hopkins University Press, 1989.

[94] P.S. Gopalakrishnan, D. Kanevsky, A. Nádas, and D. Nahamoo. A generalization of the Baum algorithm to rational objective functions. In *Proceedings of ICASSP*, volume 1, pages 631–634, Glasgow, Scotland, May 1989.

[95] A.H. Gray and J.D. Markel. Distance measures for speech processing. *IEEE Trans. Acoust. Speech Signal Process.*, **24**(5):380–391, 1976.

[96] R.M. Gray, A. Buzo, A.H. Gray, and Y. Matsuyama. Distortion measures for speech processing. *IEEE Trans. Acoust. Speech Signal Process.*, **ASSP-28**(4):367–376, 1980.

[97] R.M. Gray and D.L. Neuhoff. Quantization. *IEEE Trans. Inf. Theory*, **44**(6):2325–2383, 1998.

[98] M. Greenspan. Piston radiator: some extensions of the theory. *J. Acoust. Soc. Am.*, **65**(3):608–621, 1979.

[99] R.A. Greiner and J. Eggers. The spectral amplitude distribution of selected compact discs. *J. Audio Eng. Soc.*, **37**(4):246–275, 1989.

[100] D.W. Griffin and J.S. Lim. Multiband excitation vocoder. *IEEE Trans. Acoust. Speech Signal Process.*, **36**(8):1223–1235, 1988.

[101] A. Gröschel, M. Schug, M. Beer, and F. Henn. Enhancing audio coding efficiency of MPEG layer-2 with spectral band replication for digitalradio (DAB) in a backwards compatible way. In *Proceedings of AES 114th Convention, Amsterdam*. Audio Engineering Society, 2003.

[102] N. Guttman and S. Pruzansky. Lower limits of pitch and musical pitch. *J. Speech Hear. Res.*, **5**(3):207–214, 1962.

[103] R. Hagen. Spectral quantization of cepstral coefficients. In *Proceedings of ICASSP*, volume 1, pages 509–512, Adelaide, Australia, April 1994.

[104] A. Härmä, M. Karjalainen, L. Savioja, V. Välimäki, U.K. Laine, and J. Huopaniemi. Frequency-warped signal processing for audio applications. *J. Audio Eng. Soc.*, **48**:1011–1031, 2000.

[105] W.M. Hartmann. The effect of amplitude envelope on the pitch of sine wave tones. *J. Acoust. Soc. Am.*, **63**:1105–1113, 1978.

[106] P. Hedelin and J. Skoglund. Vector quantization based on Gaussian mixture models. *IEEE Trans. Speech Audio Process.*, **8**(4):385–401, 2000.

[107] D.A. Heide and G.S. Kang. Speech enhancement for bandlimited speech. In *Proceedings of ICASSP*, volume 1, pages 393–396, Seattle, WA, May 1998.

[108] H. Helmholtz. *Die Lehre von den Tonempfindungen [The Theory of Tone Perception]*. Vieweg, 1954. English translations of 4th German edition of 1877.

[109] W. Hess. *Pitch Determination of Speech Signals*. Springer, Berlin, 1983.

[110] V. Hohmann. Frequency analysis and synthesis using a gammatone filterbank. *Acta Acoust.*, **88**:433–442, 2002.

[111] D. Homm, T. Ziegler, R. Weidner, and R. Bohm. Bandwidth extension of audio signals by spectral band replication. In *Proceedings of 1st IEEE Benelux Workshop on MPCA, Louvain, Belgium*. IEEE, 2002.

[112] D. Homm, T. Ziegler, R. Weidner, and R. Bohm. Implementation of a DRM audio cecoder (aacPlus) on ARM architecture. In *Proceedings of AES 114th Convention, Paper 5833, Amsterdam, The Netherlands*. Audio Engineering Society, 2003.

[113] A.J.M. Houtsma and J.L. Goldstein. The central origin of the pitch of complex tones: evidence from musical interval recognition. *J. Acoust. Soc. Am.*, **51**(2)(Part 2):520–529, 1972.

[114] F.V. Hunt. *Electroacoustics*. John Wiley & Sons, 1954.

[115] C. Huygens. En envoiant le problème d'Alhazen en France ... '. *Oevres Complètes*, Vol. 10. Société Hollandaises de Sciences, Haarlem, The Netherlands, 1905 (Originally published 1693).

[116] A. Illényi and P. Korpássy. Correlation between loudness and quality of stereophonic loudspeakers. *Acoustica*, **49**(4):334–336, 1981.

[117] International Standard ISO 226-1987(E), Acoustics–normal equal-loudness level contours, 1987.

[118] International Standard ISO 7029-1984(E), Acoustics–Threshold of hearing by air conduction as a function of age and sex for otologically normal persons, 1984.

[119] B. Iser and G. Schmidt. Neural networks versus codebooks in an application for bandwidth extension of speech signals. In *Proceedings of EUROSPEECH*, pages 565–568, Geneva, Switzerland, September 2003.

[120] F. Itakura. Line spectrum representation of linear predictor coefficients of speech signals. *J. Acoust. Soc. Am.*, **57**(Suppl. 1):35, 1975 (89th Meeting of the Acoustical Society of America).

[121] ITU-T Rec. G.132. Attenuation performance. In Blue Book, vol. Fascicle III.1 (General Characteristics of International Telephone Connections and Circuits), 1988.

[122] ITU-T Rec. G.151. General performance objectives applicable to all modern international circuits and national extension circuits. In Blue Book, vol. Fascicle III.1 (General Characteristics of International Telephone Connections and Circuits), 1988.

[123] ITU-T Rec. G.711. Pulse code modulation (PCM) of voice frequencies, 1972.

[124] ITU-T Rec. G.712. Performance characteristics of PCM channels between 4-wire interfaces at voice frequencies. In Blue Book, vol. Fascicle III.4 (General Aspects of Digital Transmission Systems; Terminal Equipments), 1988.

[125] ITU-T Rec. G.722. 7 khz audio coding within 64 kbit/s. In Blue Book, vol. Fascicle III.4 (General Aspects of Digital Transmission Systems; Terminal Equipments), 1988.

[126] ITU-T Rec. G.722.1. Coding at 24 and 32 kbit/s for hands-free operation in systems with low frame loss, September 1999.

[127] ITU-T Rec. G.722.2. Wideband coding of speech at around 16 kbit/s usign adaptive multi-rate wideband (amr-wb), July 2003.

[128] P. Jax. *Enhancement of Bandlimited Speech Signals: Algorithms and Theoretical Bounds*. PhD thesis, Aachen University (RWTH), Aachen, Germany, 2002.

[129] P. Jax and P. Vary. Wideband extension of telephone speech using a hidden Markov model. In *IEEE Speech Coding Workshop*, pages 133–135, Delavan, WI, September 2000.

[130] P. Jax and P. Vary. Enhancement of band-limited speech signals. In *Proceedings of Aachen Symposium on Signal Theory*, pages 331–336, Aachen, Germany, September 2001.

[131] P. Jax and P. Vary. An upper bound on the quality of artificial bandwidth extension of narrowband speech signals. In *Proceedings of ICASSP*, volume 1, pages 237–240, Orlando, FL, May 2002.

[132] P. Jax and P. Vary. Artificial bandwidth extension of speech signals using MMSE estimation based on a hidden Markov model. In *Proceedings of ICASSP*, volume 1, pages 680–683, Hong Kong SAR, China, April 2003.

[133] P. Jax and P. Vary. On artificial bandwidth extension of telephone speech. *Signal Process.*, **83**(8):1707–1719, 2003.

[134] F. Jay, editor. *IEEE Standard Dictionary of Electrical and Electronics Terms*. IEEE, 2nd edition, 1978.

[135] A.J. Jerri. The shannon sampling theorem – its various extensions and applications: a tutorial review. *Proc. IEEE*, **65**:1565–1596, 1977.

[136] J. D. Johnston. Transform coding of audio signals using perceptual noise criteria. *IEEE J. Select. Areas Commun.*, **6**(2):314–323, 1988.

[137] E. Joliveau, J. Smith, and J. Wolfe. Tuning of vocal tract resonances by sopranos. *Nature*, **427**:116, 2004.

[138] E.I. Jury. *Theory and Application of the z-Transform Method*. John Wiley & Sons, New York, 1964.

[139] M. Kahrs and K. Brandenburg. *Applications of Digital Signal Processing to Audio and Acoustics*. Kluwer Academic Publishers, 1998.

[140] A.J.M. Kaizer. *On the Design of Broadband Electrodynamical Loudspeakers and Multiway Loudspeaker Systems*. PhD thesis, Eindhoven University of Technology, 1986.

[141] B. Kedem. *Time Series Analysis by Higher Order Crossings*. IEEE Press, New York, 1994.

[142] L.E. Kinsler, A.R. Frey, A.B. Coppens, and J.V. Sanders. *Fundamentals of Acoustics*. John Wiley & Sons, 1982.

[143] W. Klippel. Dynamic measurement and interpretation of the nonlinear parameters of electrodynamic loudspeakers. *J. Audio Eng. Soc.*, **38**(12):944–955, 1990.

[144] W. Klippel. The nonlinear large-signal behavior of electrodynamic loudspeakers at low frequencies. *J. Audio Eng. Soc.*, **40**(6):483–496, 1992.

[145] M.H. Knudsen and J.G. Jensen. Low-freqency loudspeaker models that include suspension creep. *J. Audio Eng. Soc.*, **41**(1/2):3–18, 1993.

[146] A. Kohlrausch and A.J.M. Houtsma. Pitch related to spectral edges of broadband signals. *Philos. Trans. R. Soc. London B*, **336**:81–88, 1992.

[147] U. Kornagel. Spectral widening of the excitation signal for telephone-band speech enhancement. In *Proceedings of IWAENC*, pages 215–218, Darmstadt, Germany, September 2001.

[148] W. Krebber. *Sprachübertragungsqualität von Fernsprech-Handapparaten*. PhD thesis, RWTH Aachen, 1995 (in German).

[149] K.R. Krishnamachari, R.E. Yantorno, J.M. Lovekin, D.S. Benincasa, and S.J. Wenndt. Use of local kurtosis measure for spotting usable speech segments in co-channel speech. In *Proceedings of ICASSP*, volume 1, pages 649–652, Salt Lake City, UT, May 2001.

[150] P. Kroon, E.F. Depettere, and R.J. Sluyter. Regular-pulse excitation – a novel approach to effective and efficient multipulse coding of speech. *IEEE Trans. Acoust. Speech. Signal Process.*, **34**(5):1054–1063, 1986.

[151] J.B. Kruskal and M. Wish. *Multidimensional Scaling, Quantitative Applications in the Social Sciences*. Sage Publishers, 10th edition, 1983.

[152] J.B. Kruskal, F.W. Young, and J.B. Seery. *How to Use the KYST2A, A Very Flexible Program to do Multidimensional Scaling and Unfolding*. Bell Telephone Laboratories, 1978.

[153] O. Kunz. SBR Explained: White Paper, December 2003, http://www.codingtechnologies.com.

[154] H.J. Landau and H.O. Pollack. Prolate spheroidal wave functions, Fourier analysis and uncertainty, II. *Bell Syst. Tech. J.*, **40**:65–84, 1961.

[155] H.J. Landau and H.O. Pollack. Prolate spheroidal wave functions, Fourier analysis and uncertainty, III. *Bell Syst. Tech. J.*, **41**:1295–1336, 1962.

[156] E. Larsen and R.M. Aarts. Reproducing low-pitched signals through small loudspeakers. *J. Audio Eng. Soc.*, **50**(3):147–164, 2002.

[157] E. Larsen, R.M. Aarts, and M. Danessis. Efficient high-frequency bandwidth extension of music and speech. In *proceedings of 112th AES Convention*, Munich, Germany. Audio Engineering Society, 2002.

[158] N. Le Goff, R.M. Aarts, and A.G. Kohlrausch. Thresholds for hearing mistuning of the fundamental component in a complex sound. In *Proceedings of the 18th International Congress on Acoustics (ICA2004)*, Paper Mo. P3.21, p. I-865, Kyoto, Japan, 2004.

[159] W.J.M. Levelt, J.P. van de Geer, and R. Plomp. Triadic comparisons of musical intervals. *Br. J. Math. Stat. Psychol.*, **19** (Part 2):163–179, 1966.

[160] B.G. Levi. Acoustic experiments shows why it's so hard to make out the heroine's words at the opera. *Phys. Today*, **57**(3):23–25, March 2004.

[161] J.C.R. Licklider. Auditory frequency analysis. In C. Cherry, editor, *Information Theory*. Academic Press, New York, NY, 1956.

[162] Y. Linde, A. Buzo, and R.M. Gray. An algorithm for vector quantizer design. *IEEE Trans. Commun.*, **28**(1):84–95, 1980.

[163] S.P. Lloyd. Least squares quantization in pcm. *IEEE Trans. Inf. Theory*, **IT-28**(2):129–137, 1982.

[164] E.M. Long and R.J. Wickersham. Method and apparatus for operating a loudspeaker below resonant frequency. US patent 4,481,662, 1984. Filing year 1982.

[165] X. Maitre. 7 kHz audio coding within 64 kbit/s. *IEEE J. Select. Areas Commun.*, **6**(2):283–298, 1988.

[166] J. Makhoul. Linear prediction: a tutorial review. *Proc. IEEE*, **63**:561–580, 1975.

[167] J. Makhoul and M. Berouti. High-frequency regeneration in speech coding systems. In *Proceedings of ICASSP*, pages 428–431, Washington, DC, April 1979.

[168] J.D. Markel and A. H. Gray. *Linear Prediction of Speech*. Springer-Verlag, Berlin, Heidelberg, New York, 1976.

[169] S.L. Marple. Computing the discrete-time "analytic" signal via FFT. *IEEE Trans. Inf. Theory*, **47**(9):2600–2603, 1999.

[170] R.J. McAulay and T.F. Quatieri. Sinusoidal coding. In W. Bastiaan Kleijn and K.K. Paliwal, editors, *Speech Coding and Synthesis*, chapter 4, pages 121–173. Elsevier, 1995.

[171] A. McCree, T. Unno, A. Anandakumar, A. Bernard, and E. Paksoy. An embedded adaptive multi-rate wideband speech coder. In *Proceedings of ICASSP*, volume 2, pages 761–764, Salt Lake City, UT, May 2001.

[172] N.W. McLachlan. *Loudspeakers*. Oxford at the Clarendon Press, 1934.

[173] J. Merhaut. *Theory of Electroacoustics*. McGraw-Hill, 1981.

[174] G. Miet, A. Gerrits, and J.C. Valière. Low-band extension of telephone-band speech. In *Proceedings of ICASSP*, volume 3, pages 1851–1854, Istanbul, Turkey, June 2000.

[175] T.K. Moon. The expectation-maximization algorithm. *IEEE Signal Process. Mag.*, **13**(6):47–60, 1996.

[176] B.C.J. Moore. Frequency difference limens for short-duration tones. *J. Acoust. Soc. Am.*, **54**:610–619, 1973.

[177] B.C.J. Moore. *Handbook of Perception and Cognition: Hearing*. Academic Press, 1995.

[178] B.C.J. Moore. *An Introduction to the Psychology of Hearing*. Academic Press, 5th edition, 2003.

[179] B.C.J. Moore, R.W. Peters, and B.R. Glasberg. Thresholds for the detection of inharmonicity in complex tones. *J. Acoust. Soc. Am.*, **77**:1861–1867, 1985.

[180] P.M. Morse and K.U. Ingard. *Theoretical Acoustics*. McGraw-Hill, 1968.

[181] E. Moulines and W. Verhelst. Time-domain and frequency-domain techniques for prosodic modification of speech. In W. Bastiaan Kleijn and K.K. Paliwal, editors, *Speech Coding and Synthesis*, chapter 15, pages 519–555. Elsevier, 1995.

[182] *NAG Library Manual*. Numerical Algorithm Group Ltd, Oxford, 1999. Mark 19, Chapter E02.

[183] Y. Nakatoh, M. Tsushima, and T. Norimatsu. Generation of broadband speech from narrowband speech using piecewise linear mapping. In *Proceedings of EUROSPEECH*, volume 3, pages 1643–1646, Rhodos, Greece, September 1997.

[184] S.T. Neely and J.B. Allen. Invertibility of a room impulse response. *J. Acoust. Soc. Am.*, **66**(1):165–169, 1979.

[185] M. Nilsson, S.V. Andersen, and W.B. Kleijn. On the mutual information between frequency bands in speech. In *Proceedings of ICASSP*, volume 3, pages 1327–1330, Istanbul, Turkey, June 2000.

[186] M. Nilsson, H. Gustafsson, S.V. Andersen, and W.B. Kleijn. Gaussian mixture model based mutual information estimation between frequency bands in speech. In *Proceedings of ICASSP*, volume 1, pages 525–528, Orlando, FL, May 2002.

[187] M. Nilsson and W.B. Kleijn. Avoiding over-estimation in bandwidth extension of telephony speech. In *Proceedings of ICASSP*, volume 2, pages 869–872, Salt Lake City, UT, May 2001.

[188] F. Nordén, T. Eriksson, and P. Hedelin. An information theoretic perspective on the speech spectrum process. In *IEEE Speech Coding Workshop*, pages 93–95, Delavan, WI, September 2000.

[189] G. Oetken and W. Schüßler. On the design of digital filters for interpolation. *Arch. Elektron. Übertragungstech. (AEÜ), Electron. Commun.*, **27**(11):471–476, 1973.

[190] G.S. Ohm. Über die definition des tones, nebst daran geknüpfter theorie der Sirene and ähnlicher tonbildender Vorrichtungen [On the definition of the tone and the related theory of the siren and similar tone-producing devices]. *Ann. Phys. Chem.*, **59**:513–565, 1843.

[191] J.P. Olive, A. Greenwood, and J. Coleman. *Acoustics of American English Speech*. Springer-Verlag, 1993.

[192] H.F. Olson. *Acoustical Engineering*. Van Nostrand, 1957.

[193] H.F. Olson. Analysis of the effects of nonlinear elements upon the performance of a back-enclosed, direct radiator loudspeaker mechanism. *J. Audio Eng. Soc.*, **10**(2):156–162, 1962.

[194] A.V. Oppenheim and R.W. Schafer. *Discrete-Time Signal Processing*. Prentice Hall, Englewood Cliffs, NJ, 1989.

[195] A.J. Oxenham and C.A. Shera. Estimates of human cochlear tuning at low levels using forward and simultaneous masking. *J. Assoc. Res. Otolaryngol.*, **4**(4):541–554, 2003.

[196] K.K. Paliwal. Interpolation properties of linear prediction parametric representations. In *Proceedings of EUROSPEECH*, volume 2, pages 1029–1032, Madrid, Spain, September 1995.

[197] K.K. Paliwal and W.B. Kleijn. Quantization of LPC parameters. In W. Bastiaan Kleijn and K.K. Paliwal, editors, *Speech Coding and Synthesis*, chapter 12, pages 433–466. Elsevier, 1995.

[198] W.J. Palm III. *Modeling, Analysis, and Control of Dynamic Systems*. John Wiley & Sons, 2000.

[199] A. Papoulis. *Probability, Random Variables, and Stochastic Processes*. McGraw-Hill, 3rd edition, New York, 1991.

[200] K.-Y. Park and H.S. Kim. Narrowband to wideband conversion of speech using GMM-based transformation. In *Proceedings of ICASSP*, volume 3, pages 1847–1850, Istanbul, Turkey, June 2000.

[201] P.J. Patrick. *Enhancement of Bandlimited Speech Signals*. PhD thesis, Loughborough University of Technology, 1983.

[202] R.D. Patterson. http://www.mrc-cbu.cam.ac.uk/cnbh/aimmanual/. AIM Manual, Retrieved September 2003.

[203] R.D. Patterson, M.H. Allerhand, and C. Giguère. Time-domain modeling of peripheral auditory processing: a modular architecture and a software platform. *J. Acoust. Soc. Am.*, **98**(4):1890–1894, 1998.

[204] R.D. Patterson, J. Nimmo-Smith, J. Holdsworth, and P. Rice. An efficient auditory filterbank based on the gammatone function. In *Paper Presented at a Meeting of the IOC Speech Group on Auditory Modelling at RSRE*, December 1987.

[205] E. Paulus and E. Zwicker. Programme zur automatischen Bestimmung der Lautheit aus Tertzpegeln oder Frequenzgruppenpegeln. *Acoustica*, **27**(5):253–266, 1972.

[206] J. Paulus. *Codierung Breitbandiger Sprachsignale Bei Niedriger Datenrate*. PhD thesis, RWTH Aachen, 1997 (in German).

[207] J.W. Paulus. Variable rate wideband speech coding using perceptually motivated thresholds. In *IEEE Speech Coding Workshop*, pages 35–36, Annapolis, MD, September 1995.

[208] A.D. Pierce. *Acoustics, An Introduction to Its Physical Principles and Applications*. ASA, 1989.

[209] R. Plomp. Pitch of complex tones. *J. Acoust. Soc. Am.*, **41**:1526–1533, 1967.

[210] R. Plomp and H.J.M. Steeneken. Effect of phase on the timbre of complex tones. *J. Acoust. Soc. Am.*, **46**:409–421, 1969.

[211] F.J. Pompei. The use of airborne ultrasonics for generating audible sound beams. *J. Audio Eng. Soc.*, **47**(9):726–731, 1999.

[212] D. Povey and P.C. Woodland. Improved discriminative training techniques for large vocabulary continuous speech recognition. In *Proceedings of ICASSP*, volume 1, pages 45–48, Salt Lake City, UT, May 2001.

[213] S.R. Powell and P.M. Chau. A technique for realizing linear phase IIR filters. *IEEE Trans. Signal Process.*, **39**(11):2425–2435, 1991.

[214] G. Quinn and E.J. Hannan. *The Estimation and Tracking of Frequencies*. Cambridge University Press, Cambridge, 2001.

[215] L.R. Rabiner. A tutorial on hidden Markov models and selected applications in speech recognition. *Proc. IEEE*, **77**(2):257–286, 1989.

[216] L.R. Rabiner and B.-H. Juang. *Fundamentals of Speech Recognition*. Prentice Hall International, 1993.

[217] L.R. Rabiner and R.W. Schafer. *Digital Processing of Speech Signals*. Prentice Hall, 1978.

[218] L.R. Rabiner and B. Gold. *Theory and Application of Digital Signal Processing*. Prentice Hall, Englewood Cliffs, NJ, 1975.

[219] J.W.S. Rayleigh. *The Theory of Sound*, Vol. 2. Dover, 1945.

[220] D.G. Raza and C.F. Chan. Quality enhancement of CELP coded speech by using an MFCC based Gaussian mixture model. In *Proceedings of EUROSPEECH*, pages 541–544, Geneva, Switzerland, September 2003.

[221] O. Read and W.L. Welch. *From Tin Foil to Stereo: Evolution of the Phonograph*. Howard Sams and Bobbs-Merill, New York, IN, 1959.

[222] D.A. Reynolds and R.C. Rose. Robust text-independent speaker identification using Gaussian mixture speaker models. *IEEE Trans. Speech Audio Process.*, **3**(1):72–83, 1995.

[223] C.W. Rice and E.W. Kellog. Notes on the development of a new type of hornless loud speaker. *Trans. Am. Inst. Electron. Eng.*, **44**:982–991, 1925.

[224] L.F. Richardson and J.S. Ross. Loudness and telephone current. *J. Gen. Psychol.*, 121–164, 1916.

[225] R.J. Ritsma. Existence region of the tonal residue. I. *J. Acoust. Soc. Am.*, **34**(9):1224–1229, 1962.

[226] R.J. Ritsma. Existence region of the tonal residue. II. *J. Acoust. Soc. Am.*, **35**(8):1241–1245, 1963.

[227] R.J. Ritsma. Frequencies dominant in the perception of the pitch of complex sounds. *J. Acoust. Soc. Am.*, **42**:191–198, 1967.

[228] R. Roberts and C. Mullis. *Digital Signal Processing*. Addison-Wesley, 1987.

[229] E.E. Roskam. The method of triads for nonmetric multidimensional scaling. *Ned. Tijdschr. Psychol. Grensgeb.*, **25**:404–417, 1970.

[230] R. Russell. http://home.earthlink.net/ rogerr7/ionovac.htm. About the Ionophone Loudspeaker, Retrieved April 2004.

[231] B. Scharf. In E.C. Carterette editor, *Handbook of Perception*, Vol. IV. Academic Press, New York, 1978.

[232] B. Scharf and A.J.M. Houtsma. In K.R. Boff, L. Kaufman, and J.P. Thomas, eds, *Handbook of Perception and Human Performance*, Vol. I, *Sensory Processes and Perception*, chapter 15, Audition III: Loudness, pitch, localization, aural distortion, pathology. John Wiley & Sons, 1986.

[233] L.L. Scharf. *Statistical Signal Processing. Detection, Estimation, and Time Series Analysis*. Addison-Wesley, Reading, MA, 1990.

[234] F. Schiel. Speech and speech-related resources at BAS. In *Proceedings of International Conference on Language Resources and Evaluation*, Granada, Spain, May 1998.

[235] R. Schlüter and W. Macherey. Comparison of discriminative training criteria. In *Proceedings of ICASSP*, volume 1, pages 493–496, Seattle, WA, May 1998.

[236] K.-O. Schmidt. Neubildung von unterdrückten Sprachfrequenzen durch ein nichtlinear verzerrendes Glied. *Telegraphen- Fernsprech-Tech.*, **22**(1):13–22, 1933 (in German).

[237] K.-O. Schmidt and O. Brosze. *Fernsprech-Übertragung*. Fachverlag Schiele & Schön, Berlin, 1967 (in German).

[238] J. Schnitzler. *Breitbandige Sprachcodierung: Zeitbereichs- und Frequenzbereichskonzepte*. PhD thesis, RWTH Aachen, 1999 (in German).

[239] J.F. Schouten. The perception of pitch. *Philips Tech. Rev.*, **5**(10):286, 1940.

[240] J.F. Schouten, R.J. Ritsma, and B. Lopes Cardozo. Pitch of the residue. *J. Acoust. Soc. Am.*, **34**(8(Part 2)):1418–1424, 1962.

[241] M. Schug, A. Groschel, M. Beer, and F. Henn. Enhancing audio coding efficiency of MPEG layer-2 with spectral band replication (SBR) for digital radio (EUREKA 147/DAB) in a backwards compatible way. In *Proceedings of AES 114th Convention*, Paper 5850, Amsterdam, The Netherlands. Audio Engineering Society, 2003.

[242] H.W. Schüßler. *Digitale Signalverarbeitung*, Band I. Springer-Verlag, 2nd edition, Berlin, 1988 (in German).

[243] A. Seebeck. Beobachtungenüber einige Bedingungen der Entstehung von Tönen [Observations on some conditions for the creation of tones]. *Ann. Phys. Chem.*, **53**:417–436, 1841.

[244] A. Sek and B.C.J. Moore. Frequency discrimination as a function of frequency, measured in several ways. *J. Acoust. Soc. Am.*, **97**:2479–2486, 1995.

[245] R.N. Shepard. The analysis of proximities: multidimensional scaling with an unknown distance function. *Psychometrica*, **27**:Part I 125–140, Part II 219–246, 1962.

[246] R.N. Shepard. Multidimensional scaling, tree-fitting and clustering. *Science*, **210**:390–398, 1980.

[247] C.A. Shera, J.J. Guinan Jr., and A.J. Oxenham. Revised estimates of human cochlear tuning from otoacoustic and behavioral measurements. *Proc. Natl. Acad. Sci.*, **99**(5):3318–3323, 2002.

[248] L.J. Sivian, H.K. Dunn, and S.D. White. Absolute amplitudes and spectra of certain musical instruments and orchestras. *J. Acoust. Soc. Am.*, **2**(3):330–371, 1931.

[249] D. Slepian. On bandwidth. *Proc. IEEE*, **64**:292–300, 1976.

[250] D. Slepian and H.O. Pollack. Prolate spheroidal wave functions, Fourier analysis and uncertainty, I. *Bell Syst. Tech. J.*, **40**:43–64, 1961.

[251] A.M. Small and R.G. Daniloff. Pitch of noise bands. *J. Acoust. Soc. Am.*, **41**:506–512, 1967.

[252] R.H. Small. Vented-box loudspeaker systems part I: small-signal analysis. *J. Audio Eng. Soc.*, **21**(5):363–372, 1973.

[253] G.F. Smoorenburg. Combination tones and their origin. *J. Acoust. Soc. Am.*, **52**(2(Part 2)):615–632, 1972.

[254] H.W. Sorenson and D.L. Alspach. Recursive Bayesian estimation using Gaussian sums. *Automatica*, **7**:465–479, 1971.

[255] Sound Quality Assessment Material (recordings for subjective tests). European Broadcasting Union, 1988, No. 422 204-2.

[256] S.S. Stevens. Procedure for calculating loudness: mark VI. *J. Acoust. Soc. Am.*, **33**(11):1577–1585, 1961.

[257] S.S. Stevens. Perceived level of noise by mark VII and Decibels (E). *J. Acoust. Soc. Am.*, **51**(2(Part 2)):575–601, 1972.

[258] I. Stylianou. *Harmonic Plus Noise Models for Speech, Combined with Statistical Methods, for Speech and Speaker Modification*. PhD thesis, Ecole Nationale Supérieure des Télécommunications, Paris, 1996.

[259] H. Suzuki and J. Tichy. Sound radiation from convex and concave domes in an infinite baffle. *J. Acoust. Soc. Am.*, **69**(1):41–49, 1981.

[260] H. Suzuki and J. Tichy. Sound radiation from an axis symmetric radiator in an infinite baffle. *J. Acoust. Soc. Jpn. (E)*, **3**(3):167–172, 1982.

[261] S.-E. Tan, W.-S. Gan, C.-W. Toh, and J. Yang. Application of virtual bass in audio cross-talk cancellation. *IEEE Electron. Lett.*, **36**(17):1500–1501, 2000.

[262] Y. Tanaka and N. Hatazoe. Reconstruction of wideband speech from telephone-band speech by multilayer neural networks. In *Spring Meeting of ASJ*, pages 255–256, 1995.

[263] Y. Tannaka and T. Koshikawa. Correlations between soundfield characteristics and subjective ratings on reproduced music quality. *J. Acoust. Soc. Am.*, **86**(2):603–620, 1989.

[264] Y. Tannaka, K. Muramori, M. Kohashi, and T. Koshikawa. Correlations between harmonic distortion, sound field characteristics and reproduced sound quality change in listening tests for loudspeakers. *J. Acoust. Soc. Jpn. (E)*, **11**(1):29–42, 1990.

[265] R. Taori, R. J. Sluijter, and A. J. Gerrits. Hi-BIN: an alternative approach to wideband speech coding. In *Proceedings of ICASSP*, volume 2, pages 1157–1160, Istanbul, Turkey, June 2000.

[266] E. Terhardt. Pitch, consonance, and harmony. *J. Acoust. Soc. Am.*, **55**:1061–1069, 1974.

[267] E. Terhardt. *Akustische Kommunikation: Grundlagen Mit Hörbeispielen*. Springer, Berlin, 1998 (in German).

[268] A.N. Thiele. Loudspeakers in vented boxes: Part I. *J. Audio Eng. Soc.*, **19**(5):382–392, 1971.

[269] E.A. Thompson. *The Soundscape of Modernity: Architectural Acoustics and the Culture of Listening in America, 1900-1933*. MIT Press, 1st edition, ISBN: 0 26 22 01 380, 2002.

[270] L.L. Thurstone. A law of comparative judgement. *Psychol. Rev.*, **34**:273–286, 1927.

[271] P. Tichavsky and A. Nehorai. Comparative study of four adaptive frequency trackers. *IEEE Trans. Signal Process.*, **45**(6):1473–1484, 1997.

[272] F.E. Toole. Listening tests–turning opinion into fact. *J. Audio Eng. Soc.*, **30**(6):431–445, 1982.

[273] F.E. Toole. Subjective measurements of loudspeaker sound quality and listener performance. *J. Audio Eng. Soc.*, **33**(1/2):2–32, 1985.

[274] F.E. Toole. Loudspeaker measurements and their relationship to listener preferences Part I. *J. Audio Eng. Soc.*, **34**(4):227–235, 1986.

[275] F.E. Toole. Loudspeaker measurements and their relationship to listener preferences Part II. *J. Audio Eng. Soc.*, **34**(5):323–348, 1986.

[276] W.S. Torgerson. *Theory and Methods of Scaling*. John Wiley & Sons, 1958.

[277] A. Uncini, F. Gobbi, and F. Piazza. Frequency recovery of narrow-band speech using adaptive spline neural networks. In *Proceedings of ICASSP*, volume 2, pages 997–1000, Phoenix, AZ, May 1999.

[278] P.P. Vaidyanathan. Homogeneous time-invariant systems. *IEEE Signal Process. Lett.*, **6**(4):76–77, 1999.

[279] J.-M. Valin and R. Lefebvre. Bandwidth extension of narrowband speech for low bit-rate wideband coding. In *IEEE Speech Coding Workshop*, pages 130–132, Delavan, WI, September 2000.

[280] V. Valtchev, J.J. Odell, P.C. Woodland, and S.J. Young. MMIE training of large vocabulary recognition systems. *Speech Commun.*, **22**(4):303–314, 1997.

[281] A.W.M. van den Enden and N.A.M. Verhoeckx. *Discrete-Time Signal Processing: An Introduction*. Prentice Hall, 1989.

[282] L.J. van der Pauw. The trapping of acoustical energy by a conical membrane and its implications for loudspeaker cones. *J. Acoust. Soc. Am.*, **68**(4):1163–1168, 1980.

[283] J. Vanderkooy. A model of loudspeaker impedance incorporating eddy currents in the pole structure. *J. Audio Eng. Soc.*, **37**:119–128, 1989.

[284] J. Vanderkooy, P.M. Boers, and R.M. Aarts. Direct-radiator loudspeaker systems with high Bl. *J. Audio Eng. Soc.*, **51**(7/8):625–634, 2003.

[285] P. Vary, K. Hellwig, R. Hofmann, R.J. Sluyter, C. Galand, and M. Rosso. Speech codec for the European mobile radio system. In *Proceedings of ICASSP*, volume 1, pages 227–230, New York, April 1988.

[286] P. Vary, U. Heute, and W. Hess. *Digitale Sprachsignalverarbeitung*. Teubner-Verlag, Stuttgart, 1998 (in German).

[287] S.V. Vaseghi. *Advanced Signal Processing and Digital Noise Reduction*. John Wiley & Sons, Teubner, 1996.

[288] W. Verhelst. Overlap-add methods for time-scaling of speech. *Speech Commun.*, **30**(4):207–221, 2000.

[289] W. Verhelst and M. Roelands. An overlap-add technique based on waveform similarity (WSOLA) for high quality time-scale modification of speech. In *Proceedings of ICASSP*, volume 2, pages 554–557, Minneapolis, MN, April 1993.

[290] G. von Békésy. *Experiments in Hearing*. Acousticl Society of America, 1989. Originally published in 1960; Reprinted by ASA in 1989.

[291] S. Voran. Listener ratings of speech passbands. In *IEEE Speech Coding Workshop*, pages 81–82, Pocono Manor, PA, September 1997.

[292] W.A. Wagenaar and P. Padmos. Quantitative interpretation of stress in Kruskal's multidimensional scaling technique. *Br. J. Math. Stat. Psychol.*, **24**:101–110, 1971.

[293] K. Walliser. Über ein Funktionsschema für die Bildung der Periodentonhöhe aus dem Schallreiz. *Cybernetik*, **6**:65–72, 1969.

[294] S. Wang. *Low Bit-Rate Vector Excitation Coding of Phonetically Classified Speech*. PhD thesis, University of California, Santa Barbara, CA, August 1991.

[295] R.L. Wegel and C.E. Lane. The auditory masking of one sound by another and its probable relation to the dynamics of the inner ear. *Phys. Rev.*, **23**:266–285, 1924.

[296] P.J. Westervelt and R.S. Larson. Laser-excited broadside array. *J. Acoust. Soc. Am.*, **54**(1):121–122, 1973.

[297] H. Yang, S. van Vuuren, and H. Hermansky. Relevancy of time-frequency features for phonetic classification measured by mutual information. In *Proceedings of ICASSP*, volume 1, pages 225–228, Phoenix, AZ, May 1999.

[298] Z.R. Yang and M. Zwolinski. Mutual information theory for adaptive mixture models. *IEEE Trans. Pattern Anal. Machine Intell.*, **23**(4):396–403, 2001.

[299] H. Yasukawa. Spectrum broadening of telephone band signals using multirate processing for speech quality enhancement. *IEICE Trans. Fundam. Electron. Commun. Comput. Sci.*, **E78-A**(8):996–998, 1995.

[300] M. Yoneyama and J-I. Fujimoto. The audio spotlight: an application of nonlinear interaction of sound waves to a new type of loudspeaker design. *J. Acoust. Soc. Am.*, **73**(5):1532–122, 1983.

[301] Y. Yoshida and M. Abe. An algorithm to reconstruct wideband speech from narrowband speech based on codebook mapping. In *Proceedings of ICSLP*, pages 1591–1594, Yokohama, Japan, 1994.

[302] W.A. Yost, A.N. Popper, and R.R. Fay. *Human psychophysics*. Springer-Verlag, 1993.

[303] F.W. Young. Scaling. *Ann. Rev. Psychol.*, **35**:55–81, 1984.

[304] J. Zera and D.M. Green. Detecting temporal onset and offset asynchrony in multicomponent complexes. *J. Acoust. Soc. Am.*, **12**:47–65, 1940.

[305] E. Zwicker. Ein Verfahren zur Berechnung der Lautstärke. *Acustica*, 10: 1960.

[306] E. Zwicker. 'Negative afterimage' in hearing. *J. Acoust. Soc. Am.*, **36**:2413–2415, 1964.

[307] E. Zwicker. Dependence of level and phase of the $(2f_1 - f_2)$ cancellation tone on frequency range, frequency difference, level of primaries, and subject. *J. Acoust. Soc. Am.*, **70**(5):1277–1288, 1981.

[308] E. Zwicker. *Psychoakustik*. Springer-Verlag, New York, 1982.

[309] E. Zwicker and H. Fastl. *Psychoacoustics. Facts and Models*. Springer, 2nd edition, Berlin, Heidelberg, New York, 1999.

[310] E. Zwicker, H. Fastl, and C. Dallmayr. Letter to the editors: BASIC-program for calculating the loudness of sounds from their 1/3-oct band spectra according to ISO 532 B. *Acoustica*, **55**(1):63–67, 1984.

[311] E. Zwicker and R. Feldtkeller. *Das Ohr als Nachrichtenempfänger*. S. Hirzel Verlag, Stuttgart, 1967.