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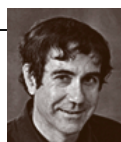
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Springer Handbook of Speech Processing
Benesty, J.; Sondhi, M.M.; Huang, Y. (Eds.)
2008, XXXVI, 1176 p. 456 illus., Hardcover
ISBN: 978-3-540-49128-6