<table>
<thead>
<tr>
<th>Title</th>
<th>Page</th>
</tr>
</thead>
<tbody>
<tr>
<td>Maximum entropy direct models for speech recognition</td>
<td>1</td>
</tr>
<tr>
<td>Design of fast LVCSR systems</td>
<td>7</td>
</tr>
<tr>
<td>Support vector machines for segmental minimum Bayes Risk decoding of continuous speech</td>
<td>13</td>
</tr>
<tr>
<td>Speaker recognition using prosodic and lexical features</td>
<td>19</td>
</tr>
<tr>
<td>Recognizing emotions from student speech in tutoring dialogues</td>
<td>25</td>
</tr>
<tr>
<td>Voice signatures</td>
<td>31</td>
</tr>
<tr>
<td>Automatic model complexity control using marginalized discriminative growth functions</td>
<td>37</td>
</tr>
<tr>
<td>Baum-Welch training for segment-based speech recognition</td>
<td>43</td>
</tr>
<tr>
<td>Two-stage continuous speech recognition using feature-based models : a preliminary study</td>
<td>49</td>
</tr>
<tr>
<td>Word-selective training for speech recognition</td>
<td>55</td>
</tr>
<tr>
<td>'Early recognition' of words in continuous speech</td>
<td>61</td>
</tr>
<tr>
<td>In search of optimal data selection for training of automatic speech recognition systems</td>
<td>67</td>
</tr>
<tr>
<td>Accurate hidden Markov models for non-audible murmur (NAM) recognition based on iterative supervised adaptation</td>
<td>73</td>
</tr>
<tr>
<td>Variational Bayesian approach for automatic generation of HMM topologies</td>
<td>77</td>
</tr>
<tr>
<td>Slovenian large vocabulary speech recognition with data-driven models of inflectional morphology</td>
<td>83</td>
</tr>
<tr>
<td>Comparing NN paradigms in hybrid NN/HMM speech recognition using tied posteriors</td>
<td>89</td>
</tr>
<tr>
<td>Phoneme-grapheme based speech recognition system</td>
<td>94</td>
</tr>
<tr>
<td>Transcribing Mandarin broadcast news</td>
<td>99</td>
</tr>
<tr>
<td>Recent advances in broadcast news transcription</td>
<td>105</td>
</tr>
<tr>
<td>Partial change accent models for accented Mandarin speech recognition</td>
<td>111</td>
</tr>
<tr>
<td>Pronunciation variation analysis based on acoustic and phonemic distance measures with application examples on Mandarin Chinese</td>
<td>117</td>
</tr>
<tr>
<td>Automatic pronunciation modelling for multiple non-native accents</td>
<td>123</td>
</tr>
<tr>
<td>Improvements in English ASR for the Malach project using syllable-centric models</td>
<td>129</td>
</tr>
<tr>
<td>Low memory decision tree method for text-to-phoneme mapping</td>
<td>135</td>
</tr>
<tr>
<td>Automatic indexing of key sentences for lecture archives</td>
<td>141</td>
</tr>
<tr>
<td>Improving the performance of a keyword spotting system by using support vector machines</td>
<td>145</td>
</tr>
<tr>
<td>Belief confirmation in spoken dialog systems using confidence measures</td>
<td>150</td>
</tr>
<tr>
<td>Semantic-oriented error correction for spoken query processing</td>
<td>156</td>
</tr>
<tr>
<td>Applying example-based error correction selectively</td>
<td>162</td>
</tr>
<tr>
<td>WebTalk : mining websites for automatically building dialog systems</td>
<td>168</td>
</tr>
<tr>
<td>RoBoDiMA : a dialog-object-based natural language speech dialog system</td>
<td>174</td>
</tr>
<tr>
<td>The relationship between dialogue acts and hot spots in meetings</td>
<td>180</td>
</tr>
<tr>
<td>Children's speech recognition with application to interactive books and tutors</td>
<td>186</td>
</tr>
<tr>
<td>Modality tracking in the multimodal Bell labs communicator</td>
<td>192</td>
</tr>
<tr>
<td>A hybrid barge-in procedure for more reliable turn-taking in human-machine dialog systems</td>
<td>198</td>
</tr>
</tbody>
</table>
Call-type classification and unsupervised training for the call center domain  

Issues in the evaluation of spoken dialogue systems using objective and subjective measures  

Acoustic correlates of user response to error in human-computer dialogues  

Balancing data-driven and rule-based approaches in the context of a multimodal conversational system  

Mandarin emotion recognition in speech  

Recognition of para-linguistic information and its application to spoken dialogue system  

Improved tone recognition for fluent Mandarin speech based on new inter-syllabic features and robust pitch extraction  

Pitch-based emphasis detection for characterization of meeting recordings  

Air- and bone-conductive integrated microphones for robust speech detection and enhancement  

TRAP-TANDEM: data-driven extraction of temporal features from speech  

Frequency-domain linear prediction for temporal features  

Blind normalization of speech from different channels  

Improved speaker adaptation using speaker dependent feature projections  

Discriminative adaptive training using the MPE criterion  

Gaussian mixture modeling with volume preserving nonlinear feature space transforms  

High resolution signal reconstruction  

Improved robust features for speech recognition by integrating time-frequency principal components (TFPC) and histogram equalization (HEQ)  

Feature pruning in likelihood evaluation of HMM-based speech recognition  

A feature-based filled pause detection system for Dutch  

On the feasibility of ASR in extreme noise using the PARAT earplug communication terminal  

A noise-robust ASR front-end using Wiener filter constructed from MMSE estimation of clean speech and noise  

Maximum entropy discrimination (MED) feature subset selection for speech recognition  

A cross-channel modeling approach for automatic segmentation of conversational telephone speech  

Analysis and effect of speaking style for dialogue speech recognition  

Analysis of acoustic correlates of British, Australian and American accents  

Unsupervised noise model estimation for model-based robust speech recognition  

Using noise reduction and spectral emphasis techniques to improve ASR performance in noisy conditions  

Analysis of different acoustic front-ends for automatic voice over IP recognition  

Experiments of in-car audio compensation for hands-free speech recognition  

On-line compensation for non-stationary noise  

Optimal filtering of noisy cepstral coefficients for robust ASR  

Warping and scaling of the minimum variance distortionless response  

Nonlinear spectral transformations for robust speech recognition
Mel-cepstrum modulation spectrum (MCMS) features for robust ASR
Using articulatory information for speaker adaptation
A robust speaker clustering algorithm
Hidden mode HMM using Bayesian network for modeling speaking rate fluctuation
A study of generic models for unsupervised on-line speaker indexing
Pronunciation modeling for names of foreign origin
Improving the robustness of prosody dependent language modeling based on prosody syntax dependence
Efficient handling of multilingual language models
Post-dialogue recognition confidence scoring for improving statistical language models using untranscribed dialogue data
Good-turing estimation from word lattices for unsupervised language model adaptation
An architecture for rapid retrieval of structured information using speech with application to spoken address recognition
Automatic junk e-mail filtering based on latent content
Topic segmentation using Markov models on section level
Tree matching for evaluation of speech interpretation systems
A look at NIST's benchmark ASR tests: past, present, and future
Are extractive text summarisation techniques portable to broadcast news?
Exploring the style-technique interaction in extractive summarization of broadcast news
Stochastic understanding models guided by connectionist dialogue acts detection
Language modeling using efficient best-first bottom-up parsing
Thematic text clustering for domain specific language model adaptation
Language modeling using a statistical dependency grammar parser
Segmenting spoken language utterances into clauses for semantic classification
Improved language model adaptation using existing and derived external resources
On the relation between additive smoothing and universal coding
Out-of-vocabulary word recognition with a hierarchical doubly Markov language model
State-space method for language modeling
Name entity recognition using language models
Effectiveness of the backoff hierarchical class N-gram language models to model unseen events in speech recognition
Confidence scoring for ANN-based spoken language understanding
Interactive grammar inference with finite state transducers
Is word error rate a good indicator for spoken language understanding accuracy
A data-driven spoken language understanding system
Combining classifiers for spoken language understanding
ASCII based transcription systems for languages with the Arabic script: the case of Persian
Automatic indexing of multimedia content by integration of audio, spoken language, and visual information
Audio packet loss over IP and speech recognition p. 607
Review of AMR speech codec- and distributed speech recognition-based speech-enabled services p. 613
Data collection and evaluation of AURORA-2 Japanese corpus p. 619
What will people say? Speech system design and language/cultural differences p. 624
Word alignment issues in ASR scoring p. 630
Experiments in word-reordering and morphological preprocessing for transducer-based statistical machine translation p. 634
Semantics synchronous understanding for robust spoken language applications p. 640
Forward-backward modeling in statistical natural concept generation for interlingua-based speech-to-speech translation p. 646
Approach toward speech-to-speech translation system by using a collection of sentences and utterances p. 652
User-oriented evaluation scheme for speech translation systems p. 658
Hand-held speech-to-speech translation system p. 664
Transonics : a speech to speech system for English-Persian interactions p. 670
VTLN-based cross-language voice conversion p. 676
A method for automatic extraction of F0 contour generation process model parameters for Mandarin p. 682
Using estimated formants tracks for formants smoothing in text to speech (TTS) synthesis p. 688
Expressive speech synthesis using American English ToBI : questions and contrastive emphasis p. 694
A unit selection approach to F0 modeling and its application to emphasis p. 706
Transformation of speaker characteristics for voice conversion p. 706
Fundamental frequency modeling for corpus-based speech synthesis based on a statistical learning technique p. 712

Table of Contents provided by Blackwell's Book Services and R.R. Bowker. Used with permission.