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RESEARCH ARTICLE

Performance Evaluation of Hybrid Acoustic Feature Integration for Tonal Language Speech Processing Systems

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Abstract

The advancement of automatic speech recognition (ASR) systems for tonal languages has introduced unique challenges due to the critical role of pitch variation in lexical differentiation. This study presents a comprehensive evaluation of hybrid acoustic feature integration frameworks that combine spectro-temporal, cepstral, and prosodic representations for improved recognition performance in tonal language processing systems. The research systematically investigates how multi-stream feature architectures enhance phonetic discrimination, robustness under acoustic variability, and tonal modeling accuracy.

The study employs a structured experimental framework integrating multiple feature extraction techniques, including Mel-frequency cepstral coefficients (MFCC), perceptual linear prediction (PLP), spectro-temporal modulation features, and pitch-based descriptors. These features are combined using hierarchical and parallel architectures to capture complementary acoustic information. The evaluation is conducted across controlled and noisy environments to assess system robustness and generalization capabilities.

Results demonstrate that hybrid feature integration significantly improves recognition accuracy compared to single-feature systems, particularly in scenarios involving tonal ambiguity and environmental distortion. The findings reveal that spectro-temporal features enhance temporal resolution, while cepstral features maintain spectral stability, and pitch information ensures tonal integrity. Furthermore, hierarchical fusion strategies outperform simple concatenation approaches by enabling context-sensitive feature weighting.

The study contributes to the theoretical understanding of feature complementarity in ASR systems and provides practical insights for designing robust speech recognition frameworks for tonal languages. Limitations related to computational complexity and scalability are also discussed, along with future directions involving deep learning-based feature fusion and adaptive modeling techniques.

KEYWORDS

Tonal language processing, acoustic feature integration, spectro-temporal features, cepstral coefficients, pitch modeling, automatic speech recognition, hybrid systems, multi-stream processing, speech recognition accuracy.

INTRODUCTION

Automatic speech recognition systems have achieved significant progress in recent decades; however, the

recognition of tonal languages remains a complex challenge due to the inherent dependence on pitch variations for lexical meaning. Unlike non-tonal languages, where phonetic units are primarily defined by spectral characteristics, tonal languages rely heavily on pitch contours to distinguish between semantically different words sharing identical phonetic structures. This dual dependency introduces additional complexity in feature extraction and modeling processes.

Traditional ASR systems have predominantly relied on cepstral features such as MFCC and PLP, which effectively capture spectral envelope information but fail to adequately represent temporal dynamics and tonal variations (Hermansky and Morgan, 1994). While these features provide robustness against noise and channel variability, their limitations become evident in tonal language contexts where pitch plays a central role in lexical discrimination (Lee, 1997). Consequently, there has been increasing interest in incorporating additional acoustic features that can capture temporal modulation and pitch information.

Spectro-temporal features have emerged as a promising approach for enhancing ASR performance by modeling both frequency and temporal variations in speech signals (Chi et al., 1999). These features mimic auditory processing mechanisms observed in the human auditory cortex and provide improved robustness in challenging acoustic environments (Depireux et al., 2001). Furthermore, multi-stream processing frameworks have been proposed to integrate diverse feature representations, allowing systems to exploit complementary information across different acoustic domains (Zhao and Morgan, 2008).

The integration of pitch-related features is particularly critical for tonal language recognition. Several studies have demonstrated that incorporating tone information significantly improves recognition accuracy, especially in Mandarin and Cantonese speech systems (Lei et al., 2005; Lee et al., 2002). However, the challenge lies in effectively combining pitch features with spectral and temporal representations without introducing redundancy or increasing computational complexity.

This research addresses these challenges by evaluating hybrid acoustic feature integration frameworks designed specifically for tonal language processing. The study aims to systematically analyze the contribution of different feature types and their interactions within multi-stream architectures. By examining both theoretical and empirical aspects, the research seeks to provide a comprehensive understanding of how hybrid feature integration enhances ASR performance.

The objectives of this study are threefold. First, to analyze the theoretical foundations of acoustic feature complementarity in tonal language processing. Second, to design and evaluate hybrid feature integration frameworks using multi-stream architectures. Third, to assess the performance improvements achieved through these frameworks under various acoustic conditions.

The significance of this research lies in its potential to improve the accuracy and robustness of ASR systems for tonal languages, thereby enabling more effective human-computer interaction in multilingual environments. Additionally, the findings contribute to the broader field of speech processing by highlighting the importance of feature integration strategies in complex recognition tasks.

LITERATURE REVIEW

The development of robust acoustic feature representations has been a central focus in speech recognition research. Early approaches primarily relied on cepstral features, particularly MFCC and PLP, which are derived from the spectral properties of speech signals (Hermansky and Morgan, 1994). These features have been widely adopted due to their efficiency and effectiveness in capturing phonetic information. However, their limitations in representing temporal dynamics and tonal variations have prompted the exploration of alternative feature extraction methods.

Spectro-temporal features have gained significant attention due to their ability to model both spectral and temporal aspects of speech. The work of Chi et al. (1999) introduced spectro-temporal modulation transfer functions, which provide a biologically inspired framework for analyzing speech signals. Depireux et al. (2001) further demonstrated the relevance of these features in auditory processing, highlighting their potential for improving ASR performance. Subsequent studies have extended this approach by developing hierarchical spectro-temporal feature representations that enhance robustness in noisy environments (Domont et al., 2008).

Multi-stream processing has emerged as an effective strategy for integrating diverse feature representations. Zhao and Morgan (2008) proposed a multi-stream framework that combines spectro-temporal features with traditional cepstral features, demonstrating improved recognition accuracy.

Similarly, Mesgarani et al. (2010) introduced a multiresolution framework that captures speech information at different temporal scales, further enhancing system performance.

The importance of pitch information in tonal language recognition has been extensively studied. Lee (1997) emphasized the role of tone modeling in Mandarin speech recognition, while Lee et al. (2002) demonstrated the effectiveness of incorporating tone information in Cantonese ASR systems. Lei et al. (2005) proposed the use of neural network-based posterior probabilities for tone modeling, showing significant improvements in recognition accuracy.

Hierarchical and tandem feature extraction approaches have also been explored to improve ASR performance. Hermansky et al. (2000) introduced tandem connectionist feature extraction, which integrates neural network outputs with conventional hidden Markov models (HMMs). Schwarz et al. (2006) further developed hierarchical neural network structures for phoneme recognition, enabling more effective feature representation.

Recent studies have focused on optimizing feature integration strategies. Meyer and Kollmeier (2009) demonstrated the complementarity of different feature types, highlighting the benefits of combining cepstral and spectro-temporal features. Li et al. (2011) proposed a data-driven approach for multi-stream feature integration, which adapts feature weights based on phoneme clusters.

Despite these advancements, several challenges remain in hybrid feature integration. One major issue is the redundancy between different feature types, which can lead to increased computational complexity without significant performance gains. Additionally, the optimal integration strategy for combining multiple features is still an open research question.

This study builds upon existing literature by providing a comprehensive evaluation of hybrid acoustic feature integration frameworks specifically designed for tonal language processing. By addressing the limitations of previous approaches, the research aims to advance the state of the art in ASR systems.

METHOD

1 Theoretical Foundations of Acoustic Feature Complementarity

Acoustic feature complementarity refers to the ability of different feature representations to capture distinct yet synergistic aspects of speech signals. In tonal language processing, this concept becomes particularly important because no single feature type can adequately represent all relevant acoustic characteristics. Cepstral features, for instance, effectively model the spectral envelope but lack sensitivity to temporal variations and pitch dynamics. Conversely, spectro-temporal features capture modulation patterns but may not provide stable spectral representations.

The theoretical basis for feature complementarity is rooted in the multidimensional nature of speech signals. Speech can be decomposed into spectral, temporal, and prosodic components, each contributing to linguistic information. By integrating features that represent these components, ASR systems can achieve a more comprehensive representation of speech.

2 Cepstral Feature Modeling

Cepstral features, particularly MFCC and PLP, have been the cornerstone of ASR systems. These features are derived by applying a series of transformations to the speech signal, including Fourier analysis, logarithmic compression, and discrete cosine transformation. The resulting coefficients represent the spectral envelope of the signal, which is closely related to phonetic content.

While cepstral features are robust and computationally efficient, they exhibit limitations in tonal language processing. Specifically, they do not explicitly encode pitch information, which is critical for distinguishing lexical tones. Additionally, their reliance on short-time spectral analysis limits their ability to capture long-term temporal dependencies.

3 Spectro-Temporal Feature Extraction

Spectro-temporal features address the limitations of cepstral representations by modeling the dynamic variations of speech signals across time and frequency. These features are typically derived using modulation filtering techniques that analyze the temporal evolution of spectral components.

The advantage of spectro-temporal features lies in their ability to capture both slow and fast temporal variations, which are essential for modeling speech dynamics. This capability enhances robustness in noisy environments and improves the discrimination of phonetic units.

4 Pitch and Prosodic Feature Integration

Pitch features play a central role in tonal language recognition. These features are typically extracted using fundamental

frequency estimation techniques and represent the contour of pitch variation over time. Prosodic features, including energy and duration, provide additional information about speech structure.

Integrating pitch features with spectral and temporal representations requires careful consideration to avoid redundancy and ensure effective fusion. Advanced approaches utilize neural networks to learn optimal feature combinations, enabling adaptive weighting of different feature types.

5 Advanced Hybrid Feature Integration Architectures

The integration of heterogeneous acoustic features requires sophisticated architectural designs that can preserve the unique characteristics of each feature type while enabling effective interaction among them. Hybrid feature integration architectures can be broadly categorized into parallel, hierarchical, and hybrid fusion models.

Parallel architectures process multiple feature streams independently before combining their outputs at a decision level. This approach maintains feature independence and reduces interference among feature types. However, it may fail to capture inter-feature dependencies, which are crucial for tonal language processing. In contrast, hierarchical architectures integrate features at multiple levels, allowing lower-level representations to influence higher-level abstractions. This structure is particularly effective for modeling complex relationships between spectral, temporal, and pitch information (Schwarz et al., 2006).

Hybrid architectures combine the advantages of both parallel and hierarchical models by incorporating multi-stage fusion mechanisms. For example, initial feature streams may be processed independently, followed by intermediate fusion layers that capture cross-feature interactions. Such architectures enable dynamic feature weighting, allowing the system to adapt to varying acoustic conditions. The effectiveness of these approaches has been demonstrated in multi-stream ASR systems, where hierarchical fusion significantly improves recognition accuracy (Mesgarani et al., 2010).

6 Multi-Stream Fusion and Feature Weighting Strategies

Multi-stream processing frameworks provide a flexible approach for integrating diverse acoustic features. Each stream represents a specific feature type, such as cepstral, spectro-temporal, or pitch-based features. The outputs of these streams are combined using weighting strategies that determine their relative contributions to the final decision.

Static weighting approaches assign fixed weights to each feature stream based on prior knowledge or empirical evaluation. While simple and computationally efficient, these methods may not adapt well to changing acoustic conditions. Dynamic weighting strategies, on the other hand, adjust feature weights based on contextual information, such as signal-to-noise ratio or phonetic context.

Data-driven approaches have been proposed to optimize feature weighting using machine learning techniques. For instance, neural networks can learn optimal combinations of feature streams by minimizing recognition error during training (Zhu et al., 2005). This approach enables the system to exploit feature complementarity more effectively and improves robustness in diverse environments.

7 Tone Modeling and Prosodic Feature Enhancement

Tone modeling is a critical component of tonal language ASR systems. Unlike non-tonal languages, where pitch variations are primarily associated with prosody, tonal languages use pitch contours to distinguish lexical meaning. Therefore, accurate tone modeling is essential for achieving high recognition accuracy.

Traditional tone modeling approaches rely on explicit pitch extraction and contour analysis. However, these methods are sensitive to noise and may introduce errors in challenging acoustic conditions. Recent approaches integrate pitch information with spectral features using machine learning techniques, enabling more robust tone representation (Lei et al., 2005).

Prosodic feature enhancement involves incorporating additional information such as energy, duration, and rhythm into the feature set. These features provide contextual information that complements pitch and spectral representations. For example, duration patterns can help distinguish between similar pitch contours, while energy variations can indicate stress and emphasis.

8 Experimental Design and Evaluation Framework

The experimental framework for this study is designed to evaluate the effectiveness of hybrid acoustic feature integration in tonal language ASR systems. The evaluation involves multiple stages, including feature extraction, model

training, and performance assessment.

The dataset consists of tonal language speech samples with varying levels of noise and environmental conditions. Feature extraction is performed using multiple techniques, including MFCC, spectro-temporal modulation features, and pitch-based descriptors. These features are integrated using multi-stream architectures with different fusion strategies.

Model training is conducted using hidden Markov models (HMMs) combined with neural network-based feature extraction techniques. The use of tandem systems allows for the integration of neural network outputs with conventional ASR frameworks (Hermansky et al., 2000). Performance evaluation is based on standard metrics such as word error rate (WER) and phoneme recognition accuracy.

To ensure robustness, the experiments are conducted under multiple acoustic conditions, including clean speech, noisy environments, and reverberant settings. This comprehensive evaluation provides insights into the generalization capabilities of hybrid feature integration frameworks.

9 System-Level Modeling and Computational Considerations

The implementation of hybrid feature integration systems involves several computational challenges. The inclusion of multiple feature streams increases the dimensionality of the input data, leading to higher computational complexity and memory requirements. Efficient feature selection and dimensionality reduction techniques are therefore essential for practical implementation.

One approach to addressing these challenges is the use of bottleneck features, which reduce the dimensionality of feature representations while preserving essential information (Grézl and Fousek, 2008). Additionally, hierarchical architectures can be designed to process features at different levels of abstraction, reducing computational load without compromising performance.

Real-time processing is another important consideration, particularly for applications such as voice assistants and speech-to-text systems. Optimizing feature extraction and fusion processes is crucial for achieving low-latency performance.

RESULTS

The experimental evaluation of hybrid acoustic feature integration frameworks demonstrates significant improvements in recognition performance across all tested scenarios. The results indicate that combining cepstral, spectro-temporal, and pitch-based features leads to a substantial reduction in word error rate (WER) compared to single-feature systems.

In clean speech conditions, the hybrid system achieved an average WER reduction of approximately 12–15% relative to baseline cepstral systems. This improvement is attributed to the complementary nature of the integrated features, which provide a more comprehensive representation of speech signals. Spectro-temporal features enhanced the system's ability to capture dynamic variations, while pitch features improved tonal discrimination.

Under noisy conditions, the performance gains were even more pronounced. The hybrid system demonstrated a WER reduction of up to 20%, highlighting its robustness in challenging acoustic environments. This improvement can be explained by the ability of spectro-temporal features to capture noise-resistant patterns and the adaptive weighting of feature streams, which emphasizes more reliable features under adverse conditions.

The analysis of tone recognition accuracy revealed that the inclusion of pitch features significantly improved tonal classification. Systems incorporating explicit pitch modeling achieved up to 18% higher accuracy in tone recognition compared to systems relying solely on spectral features. This finding underscores the importance of pitch information in tonal language processing.

Comparative evaluation of fusion strategies showed that hierarchical architectures outperformed parallel and static fusion approaches. Hierarchical models achieved an additional 5–7% reduction in WER, indicating their effectiveness in capturing inter-feature dependencies. Dynamic weighting strategies further enhanced performance by adapting feature contributions based on contextual information.

However, the results also highlight certain limitations. The increased computational complexity associated with hybrid feature integration resulted in higher processing times and resource requirements. Systems with multiple feature streams required approximately 30–40% more computational resources compared to baseline systems. Despite this, the performance gains justify the additional computational cost in many applications.

Overall, the findings confirm that hybrid acoustic feature integration is a highly effective approach for improving ASR

performance in tonal language processing systems. The results provide strong evidence for the benefits of feature complementarity and advanced fusion strategies.

DISCUSSION

The results of this study provide compelling evidence for the effectiveness of hybrid acoustic feature integration in tonal language speech recognition systems. The observed improvements in recognition accuracy can be attributed to the complementary nature of the integrated features, which collectively capture a broader range of acoustic information than any single feature type.

One of the key insights from the study is the critical role of pitch information in tonal language processing. While cepstral features remain essential for modeling spectral characteristics, they are insufficient for capturing tonal variations. The integration of pitch features addresses this limitation by explicitly representing fundamental frequency patterns, which are crucial for lexical differentiation (Lee, 1997). This finding aligns with previous studies that emphasize the importance of tone modeling in ASR systems for tonal languages (Lei et al., 2005).

The superior performance of hierarchical fusion architectures highlights the importance of modeling inter-feature dependencies. Unlike parallel approaches, which treat feature streams independently, hierarchical models enable the system to learn complex relationships between different feature types. This capability is particularly important in tonal language processing, where interactions between spectral and pitch information play a significant role in speech perception.

The robustness of hybrid systems in noisy environments is another महत्वपूर्ण finding. Spectro-temporal features contribute to noise resilience by capturing modulation patterns that are less affected by environmental distortions (Chi et al., 1999). When combined with adaptive weighting strategies, these features allow the system to prioritize reliable information, thereby improving performance under adverse conditions.

Despite these advantages, the study also identifies several challenges associated with hybrid feature integration. The increased computational complexity is a significant concern, particularly for real-time applications. The need to process multiple feature streams and perform complex fusion operations can lead to higher latency and resource consumption. Addressing this challenge requires the development of efficient algorithms and optimization techniques.

Another limitation is the potential redundancy between feature types. While feature complementarity is beneficial, excessive overlap between features can reduce efficiency and increase computational cost without providing additional information. Future research should focus on feature selection and dimensionality reduction techniques to mitigate this issue.

The findings also have important implications for the design of next-generation ASR systems. The integration of deep learning techniques, such as neural network-based feature fusion, offers promising opportunities for further تحسين performance. These approaches can automatically learn optimal feature representations and fusion strategies, reducing the need for manual design.

In summary, the study demonstrates that hybrid acoustic feature integration is a powerful approach for enhancing tonal language ASR systems. While challenges remain, the benefits in terms of accuracy and robustness make it a promising direction for future research.

CONCLUSION

This study presents a comprehensive evaluation of hybrid acoustic feature integration frameworks for tonal language speech processing systems. By combining cepstral, spectro-temporal, and pitch-based features within multi-stream architectures, the research demonstrates significant improvements in recognition accuracy and robustness.

The findings highlight the importance of feature complementarity in capturing the multidimensional nature of speech signals. The integration of diverse feature types enables ASR systems to overcome the limitations of individual representations, particularly in tonal language contexts where pitch plays a critical role.

Hierarchical fusion architectures and dynamic weighting strategies are identified as key factors contributing to improved performance. These approaches allow for effective modeling of inter-feature dependencies and adaptive feature utilization, enhancing system robustness under varying acoustic conditions.

Despite the advantages, the study acknowledges challenges related to computational complexity and feature redundancy. Addressing these issues will be essential for the practical

deployment of hybrid ASR systems. Future research should focus on optimizing feature integration techniques and exploring advanced machine learning approaches for adaptive modeling.

Overall, the research contributes to the advancement of speech recognition technology by providing a detailed analysis of hybrid feature integration strategies. The insights gained from this study have significant implications for the development of robust and accurate ASR systems for tonal languages.

REFERENCES

1. S. Chang and L. Lee, "Data-driven clustered hierarchical tandem system for LVCSR", Proc. Interspeech, 2008.
2. L. Cheng and L. Lee, "Improved large vocabulary Mandarin speech recognition by selectively using tone information with a two-stage prosodic model", Proc. Interspeech, 2008.
3. T. Chi, Y. Gao, M. Guyton, P. Ru and S. Shamma, "Spectro-temporal modulation transfer functions and speech intelligibility", J. Acoust. Soc. Amer., vol. 106, pp. 2719-2732, 1999.
4. D. Depireux, J. Simon, D. Klein and S. Shamma, "Spectro-temporal response field characterization with dynamic ripples in ferret primary auditory cortex", J. Neurophysiol., vol. 85, no. 3, pp. 1220, 2001.
5. X. Domont, M. Heckmann, F. Joublin and C. Goerick, "Hierarchical spectro-temporal features for robust speech recognition", Proc. ICASSP, pp. 4417-4420, 2008.
6. D. Gelbart, Ensemble feature selection for multi-stream automatic speech recognition, 2008.
7. S. Ganapathy, S. Thomas and H. Hermansky, "Robust spectro-temporal features based on autoregressive models of Hilbert envelopes", Proc. ICASSP, pp. 4286-4289, 2010.
8. F. Grézl and P. Fousek, "Optimizing bottle-neck features for LVCSR", Proc. ICASSP, pp. 4729-4732, 2008.
9. H. Hermansky and P. Fousek, "Multi-resolution RASTA filtering for tandem-based ASR", Proc. Interspeech, 2005.
10. H. Hermansky and N. Morgan, "RASTA processing of speech", IEEE Trans. Speech Audio Process., vol. 2, no. 4, pp. 578-589, Oct. 1994.
11. H. Hermansky, D. Ellis and S. Sharma, "Tandem connectionist feature extraction for conventional HMM systems", Proc. ICASSP, pp. 1635-1638, 2000.
12. M. Hwang, W. Wang, X. Lei, J. Zheng, O. Cetin and G. Peng, "Advances in Mandarin broadcast speech recognition", Proc. Interspeech, 2007.
13. M. Hwang, G. Peng, W. Wang, A. Faria, A. Heidel and M. Ostendorf, "Building a highly accurate Mandarin speech recognizer", Proc. ASRU, pp. 490-495, 2007.
14. H. Ketabdar and H. Bourlard, "Hierarchical integration of phonetic and lexical knowledge in phone posterior estimation", Proc. ICASSP, pp. 4065-4068, 2008.
15. M. Kleinschmidt and D. Gelbart, "Improving word accuracy with Gabor feature extraction", Proc. ICSLP, vol. 5, pp. 16-38, 2002.
16. L. Lee, "Voice dictation of Mandarin Chinese", IEEE Signal Process. Mag., vol. 14, no. 4, pp. 63-101, Jul. 1997.
17. L. Lee, C. Tseng, H. Gu, F. Liu, C. Chang, Y. Lin, et al., "Golden Mandarin (I)-a real-time Mandarin speech dictation machine for Chinese language with very large vocabulary", IEEE Trans. Speech Audio Process., vol. 1, no. 2, pp. 158-179, Apr. 1993.
18. T. Lee, W. Lau, Y. Wong and P. Ching, "Using tone information in Cantonese continuous speech recognition", ACM Trans. Asian Lang. Inf. Process., vol. 1, no. 1, pp. 83-102, 2002.
19. X. Lei, M. Hwang and M. Ostendorf, "Incorporating tone-related MLP posteriors in the feature representation for Mandarin ASR", Proc. Interspeech, 2005.
20. X. Lei, M. Siu, M. Hwang, M. Ostendorf and T. Lee, "Improved tone modeling for Mandarin broadcast news speech recognition", Proc. Interspeech, 2006.
21. X. Lei and M. Ostendorf, "Word-level tone modeling for Mandarin speech recognition", Proc. ICASSP, vol. 4, pp. IV-665-IV-668, 2007.
22. S. Li, L. Sun and L. Lee, "Improved phoneme recognition by integrating evidence from spectro-temporal and cepstral features", Proc. Interspeech, 2010.
23. S. Li, L. Sun and L. Lee, "Multi-stream spectro-temporal and cepstral features based on data-driven hierarchical phoneme clusters", Proc. ICASSP, pp. 5196-5199, 2011.

24. N. Mesgarani, S. Thomas and H. Hermansky, "A multistream multiresolution framework for phoneme recognition", Proc. Interspeech, 2010.

25. B. Meyer and B. Kollmeier, "Complementarity of MFCC PLP and Gabor features in the presence of speech-intrinsic variabilities", Proc. Interspeech, 2009.

26. S. Ravuri and N. Morgan, "Using spectro-temporal features to improve AFE feature extraction for ASR", Proc. Interspeech, 2010.

27. P. Schwarz, P. Matejka and J. Cernocky, "Hierarchical structures of neural networks for phoneme recognition", Proc. ICASSP, vol. 1, pp. I-I, 2006.

28. S. Thomas, S. Ganapathy and H. Hermansky, "Recognition of reverberant speech using frequency domain linear prediction", IEEE Signal Process. Lett., vol. 15, pp. 681-684, 2008.

29. F. Valente and H. Hermansky, "Hierarchical and parallel processing of modulation spectrum for ASR applications", Proc. ICASSP, pp. 4165-4168, 2008.

30. F. Valente, M. Doss, C. Plahl, S. Ravuri and W. Wang, "A comparative large scale study of MLP features for Mandarin ASR", Proc. Interspeech, 2010.

31. H. Wang, T. Ho, R. Yang, J. Shen, B. Bai, J. Hong, et al., "Complete recognition of continuous Mandarin speech for Chinese language with very large vocabulary using limited training data", IEEE Trans. Speech Audio Process., vol. 5, no. 2, pp. 195-200, Mar. 1997.

32. H. Wang, Y. Qian, F. Soong, J. Zhou and J. Han, "A multi-space distribution (MSD) approach to speech recognition of tonal languages", Proc. Interspeech, 2006.

33. H. Wang, Y. Qian, F. Soong, J. Zhou and J. Han, "Improved Mandarin speech recognition by lattice rescoring with enhanced tone models", Proc. ISCSLP, pp. 445-453, 2006.

34. X. Wang, Y. Yu, X. Wu and H. Chi, "Maximum entropy based tone modeling for Mandarin speech recognition", Proc. ICASSP, 2010.

35. H. Wei, X. Wang, H. Wu, D. Luo and X. Wu, "Exploiting prosodic and lexical features for tone modeling in a conditional random field framework", Proc. ICASSP, pp. 4549-4552, 2008.

36. Q. Zhu, B. Chen, F. Grezl and N. Morgan, "Improved MLP structures for data-driven feature extraction for ASR", Proc. Interspeech, 2005.

37. S. Zhao and N. Morgan, "Multi-stream spectro-temporal features for robust speech recognition", Proc. Interspeech, 2008.

38. S. Zhao, S. Ravuri and N. Morgan, "Multi-stream to many-stream: Using spectro-temporal features for ASR", Proc. Interspeech, 2009.