



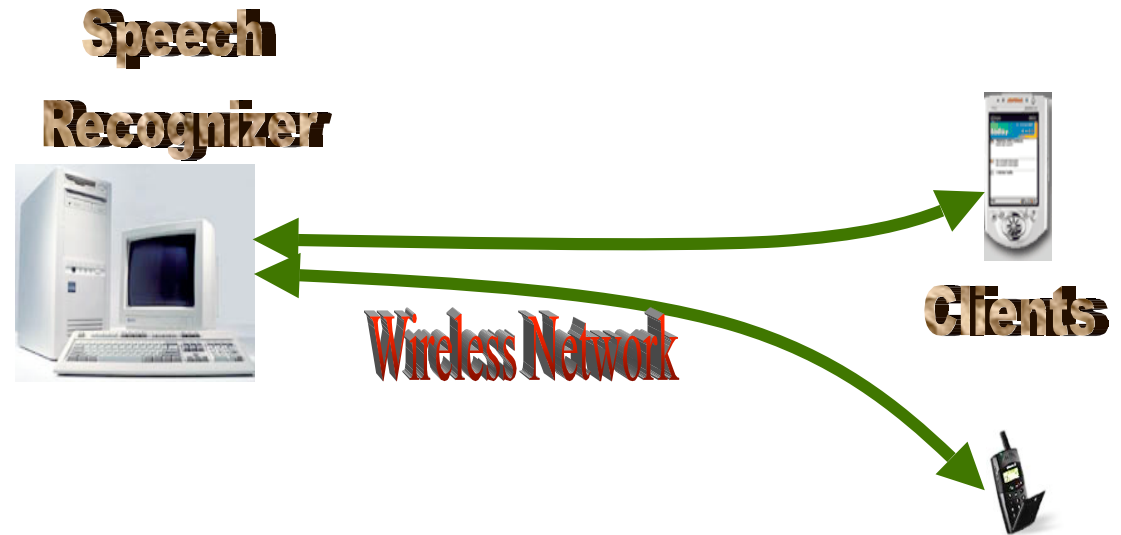
IMSC
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INTEGRATED MEDIA SYSTEMS CENTER
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Engineering Research Center at the
UNIVERSITY OF SOUTHERN CALIFORNIA

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Distributed Speech Recognition



USC STUDENTS, DEGREES

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BRIEF DESCRIPTION OF DEMONSTRATION

Distributed speech recognition (DSR), enables low complexity (mobile) wireless devices to support speech recognition applications. The acquired speech is transmitted across a wireless network to a remote desktop computer hosting the speech recognizer. To minimize the bandwidth requirement and maximize the battery life of the wireless device, the acquired speech is compressed before transmission. By optimizing the compression performance to the speech recognizer (rather than minimizing perceptual distortion) it is shown that good rate-recognition performance can be achieved. A phoneme recognizer is implemented on the desktop to achieve continuous speech recognition capability.

UNIQUE OR DISTINGUISHING CHARACTERISTICS RELATIVE TO STATE-OF-THE-ART

- Encoding algorithm optimized for speech recognition
- Layered encoding providing a multi-resolution bitstream
- Multi-stage scalable recognition at the server

APPLICATIONS <ul style="list-style-type: none"> • Remote recognition • Distributed estimation 	RECENT HIGHLIGHTS, LEVEL OF DEVELOPMENT, UPCOMING MILESTONES <ul style="list-style-type: none"> • Compression schemes made scalable to provide flexibility in degree of error correction and enabling data driven scalability in the speech recognizer • Mutual information between speech features and phonemes used as a metric for optimizing the encoders for speech recognition applications
UNDERLYING TECHNOLOGIES <ul style="list-style-type: none"> • DSR targets the general area of remote speech recognition, in applications where computation/memory requirements precludes light-weight mobile devices from providing state of the art speech recognition services to the user. • We propose encoding only the features used for recognition instead of the speech waveform to achieve better rate-recognition performance. Furthermore, to handle the problems associated with transmission of data over a wireless channel, the encoders are made scalable. Now the application is able to trade-off between source and channel rates enabling it to adapt to the channel condition and provide best service to the user. • To handle the computational burden at the server resulting in several clients accessing a server, scalable recognition architectures are adopted. Initial low complexity recognition stages pre-process the data to reduce the search space for later more complex recognizers. To alleviate the overall recognition latency, initial recognition stages make use of the base layers and the later recognition stages make use of the enhancement layers. 	
LIST OF PUBLICATIONS, REFERENCES, URLs <ul style="list-style-type: none"> • http://biron.usc.edu/~snaveen/Publications.html 	

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