# 1993 BENCHMARK TESTS FOR THE ARPA SPOKEN LANGUAGE PROGRAM

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# ABSTRACT

This paper reports results obtained in benchmark tests conducted within the ARPA Spoken Language program in November and December of 1993. In addition to ARPA contractors, participants included a number of "volunteers", including foreign participants from Canada, France, Germany, and the United Kingdom. The body of the paper is limited to an outline of the structure of the tests and presents highlights and discussion of selected results. Detailed tabulations of reported "official" results, and additional explanatory text appears in the Appendix.

# **1. INTRODUCTION**

Benchmark tests were implemented within the ARPA Human Language Technology research program during the period November 1993 - January 1994. As in tests conducted last year, the large-vocabulary continuous speech recognition technology tests made use of Wall Street Journal-based Continuous Speech Recognition (WSJ-CSR) corpus material which was collected at SRI International (SRI) under contract to the Linguistic Data Consortium (LDC). Spoken language understanding technology tests made use of ARPA Air Travel Information System (ATIS) material collected at several sites, processed at NIST, annotated at SRI, and provided to participating members of the LDC.

#### 2. WSJ-CSR TESTS

### 2.1. New Conditions

All sites participating in the WSJ-CSR tests were required to submit results for (at least) one of two "Hub" tests. The Hub tests were intended to measure basic speaker-independent performance on either a 64K-word (Hub 1) or 5K-word (Hub 2) read-speech test set, and included required use of either a "standard" 20K trigram (Hub 1) or 5K bigram (Hub 2) grammar, and also required use of standard training sets. These requirements were intended to facilitate meaningful cross-site comparisons.

The "Spoke" tests were intended to support a number of different challenges.

Spokes 1, 3 and 4 supported problems in various types of adaptation: incremental supervised language model adaptation (Spoke 1), rapid enrollment speaker adaptation for "recognition outliers" (i.e., non-native speakers) (Spoke 3), incremental speaker adaptation (Spoke 4). [There were no participants in what had been planned as Spoke 2.]

Spokes 5 through 8 supported problems in noise and channel compensation: unsupervised channel compensation (Spoke 5), "known microphone" adaptation for two different microphones (Spoke 6), unsupervised channel compensation for 2 different environments (Spoke 7), and use of a noise compensation algorithm with a known alternate microphone for data collected in environments when there is competing "calibrated" noise (radio talk shows or music) (Spoke 8). Spoke 9 included spontaneous "dictation-style" speech.

Additional details are found in Kubala, et al. [1], on behalf of members of the ARPA Continuous speech recognition Corpus Coordinating Committee (CCCC).

### 2.2. WSJ-CSR Summary Highlights

The design of the "Hub and Spoke" test paradigm, was such that opportunities abounded for informative contrasts (e.g., the use of bigram vs. trigram grammars, the enablement/disablement of supervised vs. unsupervised adaptation strategies, etc).

There were nine participating sites in the Hub 1 tests and five sites participating in the Hub 2 tests, and some sites reported results for more than one system or research team.

The lowest word error rate in the Hub 1 baseline condition was achieved by the French CNRS-LIMSI group [2,3]. Application of statistical significance tests indicated that the performance differences between this system and a system developed by Cambridge University Engineering Department using the "HMM Toolkit" approach [4-6], were not significant. The Cambridge University HMM Toolkit approach also yielded excellent results for the smallervocabulary Hub 2 tests. The lowest word error rate for an ARPA contractor on the Hub 1 test data, for the C1 condition permitting valid cross-site comparisons, was reported by the group at CMU [7-9]. The CMU results were not significantly different from the corresponding results for the Cambridge University HMM Toolkit system. The lowest word error rate for an ARPA contractor for the (less constrained) P0 condition was reported by the group at BBN.

It is difficult to summarize results of the spoke tests, except to note that there were results reported for 8 different "spoke conditions", with from 1 to 3 participants and systems typically involved in each spoke. Details are presented in the Appendix.

### 2.3. WSJ-CSR Discussion

In NIST's analyses of the results, displays of the range of reported word error rates for each speaker across all systems are sometimes informative. These displays tend to draw attention to particularly problematic speakers or systems. Figure 1 shows data for the 10 speakers and 11 systems participating in the required Hub 1 C1 test. The speakers have been ordered from low error rate at the top of the figure to high error rate at the bottom. The length of the plotted line indicates the range in word error rate reported over all systems, and the one-standard-deviation points about the mean are indicated with a "+" symbol.

Note that three speakers (40h, 40j, and 40f) have unusually high error rates relative to the other seven in this test set.

In previous tests involving the Resource Management Corpus, it was noted that high error rates seemed to be correlated, at least indirectly, with unusually fast or slow rate of speech. To see if this was the case for the present test data, NIST obtained estimates of the average speaking rate (words/minute) for each of the test speakers. These estimates were based solely on the total number of words uttered and the total duration of the waveform files, and more sophisticated measures would be desirable. Figure 2 shows a plot of the word error rate vs. speaking rate for the 10 speakers and 11 systems in the Hub 1 C1 test.

This figure, like Figure 1, indicates that speakers 40h, 40j and 40f not only have unusually high error rates relative to the other speakers in this test set, but it also indicates that for these speakers, the speaking rate is markedly higher than for the other seven. Whereas the speaking rate for the seven speakers ranges from approximately 115 to 145 words/minute, for the three speakers with high error rate, the speaking rate ranges from 165-175 words/minute.

There are at least two factors that may contribute to higher error rates at these fast speaking rates: within-word and across-word coarticulatory effects (e.g., phone deletions) associated with fast (possibly better described as "careless" or "casual") speech, and possible under-representation of these effects in the training material.

Chase, et al. [9], at CMU, noted that for the 4 speakers in Spoke 7 (40g, 40h, 40i, and 40j), two (40g and 40i) could be subjectively characterized as "careful speaker[s]", but that 40h was characterized as a "pretty fast speaker, [with] very low gain", and 40j as a "very, very fast speaker". These "fast speakers" appear in a number of the test sets.

NIST's analyses of the distributions of rate of speech for two sets of training material for the Hub 1 test (each consisting of approximately 30,000 utterances: "short-term" and "longterm" speakers) indicate that the distributions are rather broad, with the short-term speakers' distribution peaking at 130 words/minute, with a standard deviation of 30 words/minute, and the long-term speakers' distribution peaking at 145 words/minute, with an associated standard deviation of 30 words/minute. Note that speaking rates for the 3 "fast-talking" speakers fall just outside the "plus one standard deviation region" range relative to the peak of the distribution for the "short-term speaker" training set, and just inside the corresponding region relative to the "long-term"

Because a number of the measured performance differences between systems were small, and the results of the pairedcomparison significance tests validated the relevant null hypotheses, it has been observed that, in general, the use of larger test sets, especially for the Hub tests, would have been more informative, especially with regard to the results of significance tests requiring larger speaker populations (i.e., the Sign and Wilcoxon Signed-Rank tests). With larger populations of test speakers, it would be less likely to have such disproportionately large representation of "fast speakers" in the test sets.

Two spokes made use of microphones other than the "standard" Sennheiser close-talking microphone. (See, for example, the discussion in the Appendix of this paper for Spokes 5 and 6.) Too other spokes dealt with the issue of performance degradations that were presumably due to degradations in the signal-to-noise ratio. (See, for example, the discussion for Spokes 7 and 8.)

For the test data of Spokes 5-7, subsequent to the completion of the tests, NIST performed signal-to-noise ratio (SNR) analyses, using three different bandwidth (signal preprocessing) conditions: broadband, A-weighted, and 300 Hz-3000 kHz passband "telephone bandwidth". The filtered SNR's are generally higher than the broadband values. Figure 3 shows the results of these SNR analyses.

Figure 3 (a) indicates the SNRs measured for the data of Spoke 5, which includes 10 "unknown" microphones in addition to the simultaneously collected reference Sennheiser close-talking microphone data for each data subset, collected in the normal data collection environment. SRI's "normal offices for recording" speech data have A-weighted sound level values in the 46-48 dB range. There were 2 "tieclip" or lapel microphones, 5 stand-mounted microphones, a surfaceeffect microphone, a speakerphone, and a cordless telephone in this set of 10 test microphones.

Note that the SNR values for the Sennheiser microphone are typically about 45 dB for the both the broadband and Aweighted conditions, indicating that there is little lowfrequency energy in the spectrum of the noise in the Sennheiser microphone data. Sennheiser microphone data typically yield values of 50 dB for the telephone-bandwidth condition. For the alternate microphones, the broadband SNR's range from about 23 dB (for the Audio-Technica stand-mounted microphone) to 45 dB (for the GE cordless telephone). With filtration the SNR's are higher, as expected. Note that nearly all of the microphones provide at least a 30 dB telephone-bandwidth SNR, and that the AT Pro 7a lapel-mounted microphone provides approximately 40 dB.

Figure 3 (b) indicates the measured SNR's for the data of Spoke 6, which includes 2 "known" alternate microphones in addition to the reference Sennheiser close-talking microphone, collected in the normal data collection environment. For the Sennheiser close-talking microphone, the broadband SNR's are, as for Spoke 5, 45-46 dB. There is a substantial difference between the broadband and Aweighted SNRs for the Audio-Technica stand-mounted microphone, corresponding to low frequency noise picked up by this microphone, and for the telephone-bandwidth condition the SNR is approximately 35 dB. With the telephone handset, SNRs are 38 to 40 dB, depending on bandwidth.

The test set data for Spoke 7, shown in Figure 3 (c), involved use of two different microphones (an Audio-Technica standmounted microphone and a telephone handset in addition to the usual "reference" Sennheiser close-talking microphone), in two different noise environments, with background Aweighted noise levels of 58-68 dB.

In the quieter of the two "noisy" environments, a computer laboratory with a reported A-weighted sound level in the 58-59 dB range, the broadband SNR was approximately 34-36 dB for the Sennheiser microphone, and 35 dB for the telephone handset data, but only 17 dB for the Audio-Technica microphone. Spectral analyses of the Audio-Technica background noise data demonstrate the presence of significant low frequency energy as well as the presence of harmonic components with an approximately 70 Hz fundamental. These components may have originated in some rotating machinery (e.g., a cooling fan or disc drive).

In the noisier environment, a room containing machinery with conveyor belts for sorting packages, with a reported A- weighted sound level in the 62-68 dB range, the broadband SNR ratio for the Sennheiser data degraded to 27-29 dB (a decrease of approximately 7 dB), and 27 dB for the telephone handset data, and the Audio-Technica to 16 dB (a decrease of only 1 dB). With A-weighting, in the quieter environment, the SNR for the Sennheiser improved very slightly (less than 1 dB, relative to the broad band values), and for the Audio-Technica it was 25 dB, 8 dB higher than the broad band value.

In the noisier environment, the A-weighted S/N ratio for the Sennheiser data was approximately 29 dB, and the Audio-Technica 20 dB.

For the telephone handset data, both the telephonebandwidth-filtered and the A-weighted SNRs were higher than, but typically within one or two dBs, of the unweighted values, as might be expected.

In summary, for the quieter of the two environments used in collecting the data of Spoke 7, none of the data subsets in Spoke 7 had an average filtered SNR worse than about 25 dB, and in the noisier environment, the worst average filtered SNR for any data subset was approximately 20 dB. These SNR values would not ordinarily be regarded as indicative of severe noise-degradation.

Spoke 8 involved data collected in the presence of competing noise -- music and talk radio broadcasts. For the case of competing music, the broadband SNR for the reference Sennheiser microphone ranged from 44 DB for the so-called "20 dB" condition, to 36 dB for the "10 dB" condition, and 29 dB for the "0 dB" condition. For the Audio-Technica microphone, corresponding measured values were 25, 17, and 11 dB. NIST's measurements of SNR for the data containing competing speech were inconclusive because of the difficulty of distinguishing between the spoken test material and the competing talk radio.

## 3. ATIS TESTS

#### 3.1. New Conditions

Recent ATIS tests were similar in many respects to previous ATIS tests -- the primary difference consisting of expansion of the size of the relational air-travel-information database to 46 cities, and use of a body of newly collected and annotated data using this relational database [10]. As in prior years, tests included spontaneous speech recognition (SPREC) tests, natural language understanding (NL) tests and spoken language understanding (SLS) tests. For the first time, data collected at NIST was included in the test and training data. The NIST data was collected using systems provided to NIST by BBN and SRI.

In previous years, results for NL and SLS tests were presented and discussed in terms of a "weighted error" percentage, which was computed as twice the percentage of incorrect answers plus the percentage of "No Answer" responses. The decision to weight "wrong answers" twice as heavily as "no answer" responses was reconsidered within the past year by the ARPA Program Manager, and this year only unweighted NL and SLS errors are reported (i.e., incorrect answers count the same as "No Answer" responses). For most system developers, this change of policy has appeared to result in changed strategies for system responses, so that in this year's reported results, little use was made of the "No Answer" response.

### **3.2.** Summary Highlights

For the recent ATIS tests, results were reported for systems at seven sites. Lowest error rates were reported by the group at CMU [11]. The magnitude of the differences between systems is frequently small, and the significance of these small differences is not known.

As in previous years, error rates for "volunteers" are generally higher than for ARPA contractors, possibly reflecting a lesser level-of-effort.

Additional details about the test paradigm, and comments on some aspects by individual participants, are found in another paper in this Proceedings, by Dahl, et al., on behalf of members of the ARPA Multi-site ATIS Data COllection Working (MADCOW) Group [10]. Details about the technical approaches used by the participants, and their own analyses and comments, are to be found in references [11,23-28].

#### **3.3.** ATIS Discussion

This year, 46% of the utterances were classified as Class A and 34% in Class D, so that 80% of the test utterances were "answerable" (i.e., Class A or D). Last year's test set had about the same percentage of Class A queries (43%), but somewhat fewer classified as Class D (i.e., 25%), so that last year only 67% were answerable. One possible reason for this change (other than the test-set-to-test-set fluctuations) may be that the Principles of Interpretation document is continually being extended to cover phenomena that would have otherwise resulted in categorization of some queries as "unanswerable", and therefore Class X.

For text input (NL test), for last year's test material, the lowest unweighted NL error rate was 6.5% for the Class A+D subset, 6.5% for Class A, and 6.4% for Class D, in contrast with this year's corresponding figures of 9.3%, 6.0% and 13.8%. Note that this year's test set apparently had "more difficult" Class D queries, and that there was a larger fraction of the queries that were classified as Class D than last year (34% vs. 25%).

For speech input (SLS test), and for last year's unweighted test material, the unweighted SLS error rate was 11.0% for the Class A+D subset, 10.2% for Class A, and 12.5% for Class D, in contrast with this year's corresponding figures of 13.2%, 8.9% and 17.5%.

Note that while the lowest error rate for Class A queries is smaller this year (i.e., 8.9% vs. 10.2%), this year's best Class D error rate was substantially higher than last year's. It may be the case that this is related to the extended coverage provided by the current Principles of Interpretation document, so that queries that in previous years would have been classified as unanswerable, are now judged to be answerable, although context-dependent.

#### 4. ACKNOWLEDGEMENTS

The "Hub and Spokes" Test paradigm could not have been developed, specified, or implemented without the tireless and effective efforts of Francis Kubala, as Chair of the ARPA recognition Corpus Collection continuous speech Coordinating Committee (CCCC). The tests would also not have been possible without the dedicated efforts of Denise Danielson and her colleagues at SRI in collecting an exceptionally large and varied amount of CSR data for CSR system training and test purposes. In the ATIS community, Debbie Dahl served as Chair of the MADCOW group, and it is to her credit that new data was collected at several sites with the 46 city relational database and that participating sites reached agreement on the details of the current tests. Kate Hunicke-Smith and her colleagues at SRI International were again responsible for annotation of ATIS data and for assisting NIST in the adjudication process following preliminary scoring. It is a pleasure to acknowledge Kate's thoughtful and cheerful interactions with our group at NIST.

As in previous years, the cooperation of many participants in the ARPA data and test infrastructure -- typically several individuals at each site -- is gratefully acknowledged.

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#### NOTICE

Throughout this paper, a number of references are provided in order to refer readers to relevant papers and oral presentations by researchers at the individual sites participating in the tests. In some of these papers, results are cited that differ by small amounts from those tabulated in this paper. In some cases the authors cite unofficial or preliminary, "pre-adjudication" results. In other cases, the authors cite other unofficial test results conducted after the "official" test period closed.

The views expressed in this paper are those of the author(s). The results presented are for local, system-developerimplemented tests. NIST's role in the tests is one of selecting and distributing the test materials, implementing scoring software, and uniformly tabulating the results of the tests. The views of the author(s) and these results are not to be construed or represented as endorsements of any systems or official findings on the part of NIST, ARPA or the U.S. Government.

### APPENDIX: "BENCHMARK TEST RESULTS"

#### A.1. WSJ-CSR November 1993 Test Material

The 1993 WSJ-CSR tests make use of newly-collected training material, a new compressed waveform file format, new test paradigms, and new test sets.

The new training material for the WSJ-CSR task includes a substantial amount of data (31 CD-ROMs containing training and developmental test data) collected at SRI International under contract to the Linguistic Data Consortium (LDC).

In a collaborative effort involving NIST, Tony Robinson at Cambridge University's Engineering Department, and the LDC, the newly collected waveform data was processed with an "embedded" version (i.e., the file's SPHERE-format header is uncompressed, but the bulk of the file is compressed) of a lossless waveform compression algorithm ("shorten") using the NIST SPHERE file header convention, to reduce the storage requirements for this data by a factor of approximately 50% [12]. The CSR test material was released in November.

# A.2. WSJ-CSR Test Scoring and Adjudication

The CSR tests were conducted in November and December. Test and scoring protocols were similar to last year. However, new to the CSR benchmark tests this year was the addition of an official adjudication period. Following a preliminary scoring of recognition results, sites participating in the tests were permitted to submit requests for adjudication to NIST. Adjudication requests in the CSR domain contained requests for transcription modifications due to transcription errors, alternative transcriptions, etc.

A total of 22 bug reports were received from 6 sites. The bug reports contained requests for changes to 199 (151 unique) utterance transcriptions in all WSJ-CSR test sets. The NIST adjudicators carefully evaluated each request and ultimately revised transcriptions of 83 utterances (55% of the ones in question.)

Of the transcriptions that were revised, most were the result of judgements by the adjudicators that the transcriptions contained words which could have multiple orthographic representations (e.g., compound words, variant orthographic representations, etc.) or which were lexically ambiguous. In many of these cases, both the original transcription and an alternative transcription were permitted. This was implemented by mapping alternate word forms to a single form in both the transcriptions and the recognized strings. The remaining revisions were the correction of simple transcription errors.

#### A.3. WSJ-CSR Test Participants

United States participants in the WSJ-CSR tests included: BBN Systems and Technologies (BBN) [13], Boston University (BU) [14], Carnegie Mellon University (CMU) [7-9], Dragon Systems [15], the International Computer Science Institute (ICSI) at Berkeley [16], Massachusetts Institute of Technology's Lincoln Laboratory (MIT/LL) [17], and SRI International (SRI) [29,30].

Foreign participants included two British groups at Cambridge University's Engineering Department, one pursuing connectionist approaches (CU-CON) [18], and another, developers of the HMM Toolkit (CU-HTK) [4-6], a French group at CNRS-LIMSI (LIMSI) [2,3], and a German group at the Philips GmbH Research Laboratories in Aachen [20].

BU collaborated with BBN, making use of the N-best outputs of a BBN system, using an N-best rescoring formalism, a stochastic segment modelling approach, and the use of both BU and BBN knowledge sources.

# A.4. WSJ-CSR Benchmark Test Results

A.4.1. Hub 1: 64K Baseline. The intention of the two "Hub" tests was "to improve basic [speaker independent] performance on clean [read speech] data". For Hub 1, test data consisted of 200 utterances -- 20 from each of 10 speakers, using the primary (Sennheiser series HMD 410) microphone as used in prior tests.

All sites were required to provide results for a static (i.e., non-adaptive) Speaker-Independent (SI) baseline system that would permit cross-site comparisons, which would use the standard 20K word trigram "open vocabulary" grammar and use standardized training sets.

The results of that baseline system are tabulated in the column labelled "Contrast C1" in Table 1.

Results for (optional) use of the same system training, but with the 20K bigram grammar, are shown in the column labelled "Contrast C2". These 'contrastive' results were intended for comparison with results for optional 'primary' systems. The primary systems could use "any grammar or acoustic training", and these results are shown in the column labelled "P0".

In most cases, data from each site shows on a single line. The three BU "C1" systems each represent different N-best rescoring formalisms using the BU stochastic segment model recognition system in combination with the BBN Byblos system, using different knowledge sources to re-rank the Nbest hypotheses. The two different CMU systems are different in many ways, so that comparisons are non-trivial. For the baseline "C1" systems, word error rates ranged from 19.0% to 11.7%, with the lowest error rate reported for the LIMSI system.

In this table, and others of this sort in this paper, the results of contrastive comparisons are shown in the boxes labelled "COMPARISONS AND SIGNIFICANCE TESTS". The results of use of the NIST statistical significance tests that have been used in previous tests are also shown.

To illustrate interpretation of some of the tabulated results, note that BBN and MIT/LL achieved reductions in error rate of 13.9% and 9.8%, respectively, for their P0 systems when compared to the C1 baseline systems. In most cases, these reductions were shown to be significant. Refer to [13] and [17] for discussion of factors contributing to these reduction error rate.

When contrasting use of trigram and bigram grammars, a number of sites achieved reductions in error rate of from approximately 12% to 23% for the case of use of the trigram grammar.

Table 2 shows a matrix tabulation of the results of cross-site and, in some cases, within-site, paired comparison statistical significance tests for the baseline H1-C1 systems.

A.4.2. Hub 2: 5K Baseline. Because run times for full 20K systems were in some cases regarded as prohibitive, a second baseline Hub test, requiring only a 5K lexicon, was permitted. For Hub 2, the required static SI baseline C1 system made use of a standard 5K bigram closed vocabulary grammar and either of two smaller training sets, consisting of approximately 7200 sentence utterances.

As for Hub 1, the Hub 2 test data consisted of 200 utterances -- 20 from each of 10 speakers, using the primary microphone.

Not surprisingly, error rates for the 5K systems were lower than for the 20K systems.

Table 3 shows that for the baseline C1 systems, error rates ranged from 17.7% to 8.7%, with the lowest error rate reported by the Cambridge University's HTK research group [4-6]. For the P0 systems, for which "any grammar or acoustic training" were permissible, lower error rates were to be expected, and were achieved, typically with reductions in error rate of from 25% to almost 50%. In this case, also, one of the HTK configurations achieved the lowest word error rate: 4.9%.

Table 4 shows a matrix tabulation of the results of cross-site and, in some cases, within-site, paired comparison statistical significance tests for the baseline H2-C1 systems.

A.4.3. Spoke 1: Language Model Adaptation. The stated goal for this language model adaptation spoke was "to evaluate an incremental supervised language model (LM) adaptation algorithm on a problem of sublanguage adaptation". The sole participant was Rosenfeld et al. at CMU [21]. Test data consisted of read speech data from four speakers, each reading 1 to 5 articles consisting of approximately 20-25 sentence utterances, with the Sennheiser microphone. NIST's scoring was done on four successive 5sentence utterance blocks throughout the articles (i.e., utterances 1-5, 6-10, 11-15, and 16+). Use of the statistical significance tests was not thought to be appropriate since these tests assume independence of errors across sentences, and this assumption is probably not valid when using an adaptive language model.

Table 5 presents the results for Spoke 1. The column labelled P0 shows results with incremental unsupervised adaptation enabled: word error rates vary from 16.5% on the first block of 5 sentences to 18.2% on the last block. In contrast, with language model adaptation disabled, the word error rates correspondingly vary from 20.5% to 21.1%. Comparisons between P0 and C1, involving enabling/disabling of supervised LM adaptation, indicate reductions in word error rate of between 9.8% to 19.4%, with lesser reductions for the P0:C2 comparisons involving unsupervised LM adaptation.

**A.4.4.** Spoke 3: SI Recognition Outliers. The stated goal for this spoke was "to evaluate a rapid enrollment speaker adaptation algorithm on difficult speakers (e.g., non-native speakers of American English)". The sole participant was BBN [13]. Test data consisted of read speech from ten speakers, each reading 40 sentence utterances, with the Sennheiser microphone. For each speaker, the 40 "rapid enrollment" utterances were available for use with the "rapid enrollment" speaker adaptation.

Table 6 presents the results for Spoke 3. The column labelled P0 shows results with rapid enrollment adaptation enabled: word error rate for the 400 utterance test set is 14.5%. In contrast, with adaptation disabled, the word error rate is 32.0%. Alternatively, the P0:C1 contrast indicates a reduction in error rate 54.7%, which was shown to be significant using all of the significance tests applied by NIST.

A.4.5. Spoke 4: Incremental Speaker Adaptation. The stated goal for this spoke was "to evaluate an incremental speaker adaptation algorithm". Two sites participated: Dragon [15] and MIT/LL [17]. In this spoke, there were only four test speakers, with 100 sentence utterances for each. NIST's scoring was done on four successive 25-sentence utterance blocks (i.e., utterances 1-25, 26-50, 51-75, and 76+).

Table 7 presents the results for Spoke 4.

For the Dragon results, word error rates for the P0 condition (with incremental unsupervised adaptation enabled) range from 15.5% to 14.3%. For MIT/LL, the corresponding variation is 10.9% to 11.1%. There is evidence of significant reductions in error of the order of 20% to 30% for the P0:C1 contrasts for the Dragon results (e.g., note the reduction of from 19.4% to 15.5% for the first block of 25 utterances).

For the corresponding MIT/LL results, the magnitudes of the reductions are not as large. For both sites, the incremental changes in error rates between the P0 and C2 cases, involving unsupervised/supervised adaptation, in most cases are not shown to be significant, and range from approximately 4% to 16%.

A.4.6. Spoke 5: "Microphone Independence". The stated goal of this spoke was to "evaluate an unsupervised channel compensation algorithm". The different "channels" in this case were different microphones -- each of the ten speakers in this test set used a different (unknown) microphone. Similar, but not identical, microphones had been incorporated in training and development material. For the 200 utterances in each portion of this test set, both the unknown microphone data (in "wv2" data files) and corresponding Sennheiser microphone data (in "wv1" files) were available.

Both CMU [22] and SRI [30] participated in this spoke.

Table 8 presents the results for Spoke 5.

With unsupervised channel compensation enabled, the CMU system achieved an error rate of 15.1%, in contrast to 20.9% with compensation disabled -- a 27.8% reduction in word error rate. SRI achieved a comparable reduction of 24.2%, and with slightly lower error rates. With compensation enabled, the CMU system achieved 9.7% word error for the corresponding Sennheiser data, while the SRI system achieved 6.6% word error. Enabling/disabling the channel compensation made essentially no difference for the case of the Sennheiser data subset, as might be suspected.

A.4.7. Spoke 6: Known Alternate Microphones. The stated goal of this spoke was to "evaluate a known microphone adaptation algorithm". There were two different microphones -- an Audio Technica stand-mounted microphone, and a telephone handset which was to be connected to the data collection apparatus "over external lines", in addition to the Sennheiser (wv1) data. Two-channel microphone adaptation data -- for each of the two microphones and the (reference) Sennheiser microphone was provided from "devtest data". There were ten speakers for the data for each of the two microphones, with 20 sentence utterances per speaker. In NIST's analysis of the results, data are separately tabulated for the Audio-Technica (at) data, and for the telephone handsets (th).

Three sites participated: BBN [13], Dragon [15], and SRI [30].

Table 9 presents the results for Spoke 6.

For the case of the microphone adaptation disabled (C1), for the Audio-Technica microphone's data, word error rates were 6.4% for the SRI system, 10.4% for the BBN system, and 18.5% for the Dragon results. For telephone handset data, the SRI system had 19.1\%, the BBN system had 29.3% and Dragon 65.4%. These results for the telephone handset data were probably somewhat worse than might have been expected because of inadvertent channel differences between development test and evaluation test sets.

Considering the adaptation enabled/disabled P0:C1 contrast, BBN and Dragon achieved 9.4% and 11.7% reductions in word error rate for the Audio-Technica microphone, and 57.4% (from 29.3% to 12.5% word error) and 11.7% for BBN and Dragon, respectively. On corresponding Sennheiser data, the BBN and SRI systems with adaptation disabled achieved word error rates ranging from 5.9% to 8.4%, while the Dragon results were 13.8% and 14.6%.

**A.4.8.** Spoke 7: "Noisy Environments". The stated goal of this spoke was to "evaluate a noise compensation algorithm with known alternate microphones" in two different data-collection environments with background A-weighted sound level of from 55 to 68 dB. Two different microphones were used, the same microphones as were used for Spoke 6, (the Audio-Technica and a telephone handset). Utterances for the microphone/channel adaptation (Sennheiser to known alternate microphone) were available from development test data, and there were files with background noise (but no speech) for each microphone-noise-environment-speaker condition. The two noise environments ("e1" and "e2") consisted of computer laboratory (e1), and a room with package sortation machinery in operation ("e2").

The sole participant in this spoke was SRI [30].

Table 10 presents the results for Spoke 7.

As might be expected, the word error rate was smallest for the lower of the two noise conditions with the alternate highquality (but not close-talking) Audio-Technica microphone (8.5%) (for which the A-weighted S/N ratio was approximately 26 dB), and markedly higher for both alternate microphones in the higher noise environment (17.4% and 28.8%). For corresponding data from the close-talking Sennheiser microphone, in the two different noise environments, error rates of from 6.3% to 9.1% were obtained.

A.4.9. Spoke 8: "Calibrated Noise Sources". The stated goal of this spoke was to "evaluate a noise compensation algorithm with a known alternate microphone on data corrupted with calibrated noise sources". Data was collected using the Audio-Technica microphone, which was also used in Spokes S6 and S7, in the presence of competing noise (from a "boom box" radio-tape player situated nearby). The competing noise was either a variety of musical selections ("mu") or talk radio ("tr"). The competing noise was "calibrated" in the sense that the level of the competing noise was intended to be set so as to be 20 or 10 dB less than the speech peak level, or equal to (or potentially greater than) the speech peak level, the "0 dB condition". Note however that NIST's measurements of SNR do not agree well with these desiderata, as discussed in Section 2.3 of this paper except in some qualitative sense.

CMU [22] was the sole participant in this spoke.

Table 11 presents the results for Spoke 8.

Data were submitted for the 3 competing noise conditions, both microphones (Sennheiser and Audio-Technica), and with noise compensation enabled and disabled -- a total of 24 conditions, permitting many cross-comparisons.

With compensation disabled, there were reductions in error rate with use of the close-talking, noise cancelling Sennheiser microphone when comparing results for the two different microphones (C3:C1). With compensation enabled, and again comparing the two different microphones (C3:P0), the differences in error rate are reduced, but are still significant in most cases.

There is evidence of significant reductions in error rate when considering compensation enabled/disabled (P0:C1) for both music and talk radio at the 10 dB and 0 dB conditions.

Further, enabling compensation appears to be beneficial for much of the data obtained with the close talking Sennheiser microphone (see, for example the C3:C2 comparisons).

**A.4.10.** Spoke 9: Spontaneous WSJ Dictation. The stated goal of this spoke was to "improve basic performance on spontaneous dictation-style speech". There were 10 speakers (all journalists, but with varying experience in dictation), each dictating 20 spontaneous Wall Street Journal-like sentence utterances, and using the Sennheiser microphone.

BBN [13] was the sole participant in this spoke.

Table 12 presents the results for Spoke 9.

Using the same system as used for the C1 condition in Hub 1 (which achieved a word error rate of 14.2% on the Hub 1 test data), a word error rate of 24.7% was achieved on the S9 data, indicating that the spontaneous dictation S9 test set is substantially more challenging. BBN's S9 system achieved an error rate of 19.1% on the S9 data, a significant reduction in word error rate of 22.8% over the H1-C1 system.

#### A.5. ATIS November 1993 Test Material

The final, adjudicated set of test material consisted of 965 test utterances and was collected at 5 sites -- BBN, CMU, MIT, NIST and SRI. As in previous years, it was selected by NIST staff from set-aside material previously collected within the MADCOW community [10]. The test set was selected so as to balance the number of utterances per data collection site (~200 utterances per site.) Because of differences in the scenarios and data collection systems used at the different collection sites, it was not possible to balance the

test set for number of subjects or the difficulty of scenarios per collection site. No "pre-filtering" of the test data was performed except to attempt to exclude subject-scenarios with mostly repetitive queries. The ATIS test material was released in November, 1993.

#### A.6. ATIS Scoring and Adjudication

The ATIS scoring and adjudication process took place in December and early January. ATIS test and scoring protocols were similar to those of previous benchmark tests. After the scored ATIS results were released in December 1993, approximately 140 adjudication requests ("bug reports") were sent to NIST. NIST worked in conjunction with SRI to resolve the requests, about 10 of which were duplicates.

The majority of the bug reports dealt with transcription issues, in some cases pointing to limitations in our community's procedures for transcribing ATIS-domain spontaneous speech. One utterance, in particular, which was classified as Class X (and thus did not affect the NL or SLS scores), but was included in the ATIS SPREC scoring, included low-level remarks by the experimenter, as a result of an inadvertent "open mike" condition. Originally, this block of speech was transcribed as "unintelligible", but in adjudication, it was fully transcribed, partially because a number of sites had objected to having been scored with significant numbers of insertion errors. After adjudication, most sites continued to do very poorly on this one utterance, but were now penalized for substitutions and deletions as well. It alone accounts for an increment of approximately 0.3% in the Class A+D+X word error for most sites, and a substantially larger fraction of the Class X error rate. In retrospect, it is clear that this problematic utterance (and the entire subject-scenario) ought not to have been included in the test set because of the "open mike" condition.

Besides the recurrent complaints of bad transcriptions, a problem involving fare IDs or flight IDs not appearing in the maximal reference answer files (the "rf2s") (which came to be known as "Joe's Fare Bug") was brought to our attention. This bug was attributed to about 21 of the test utterances before scoring. The bug was fixed by SRI and new .rf2s were generated prior to rescoring.

### A.7. ATIS Test Participants

United States participants in the ATIS tests included: AT&T Bell Laboratories (AT&T) [23], BBN Systems and Technologies (BBN) [24], Carnegie Mellon University (CMU) [11], Massachusetts Institute of Technology's Laboratory for Computer Science (MIT/LCS) [26], and SRI International (SRI) [27], and Unisys (UNISYS) [28]. There was one foreign participant: (CRIM) [25], from Canada.

AT&T collaborated with CMU, using an AT&T-developed

ATIS-domain speech recognition system and the CMU ATIS natural language system, and Unisys collaborated with BBN, using a set of N-best outputs for a BBN ATIS-domain speech recognition system as input for Unisys-developed natural language technology.

### A.8. ATIS Benchmark Test Results

**A.8.1.** SPontaneous speech RECognition (SPREC) Tests. Table 13 presents the results for the SPREC tests for all systems and subsets of the ATIS test data, using the Sennheiser close-talking microphone. For the case of the subset of all answerable queries, Class A+D, the word error rates ranged from 3.3% to 9.0%.

Table 14 presents a matrix tabulation of the ATIS SPREC results for the Class A+D subset. The overall word error rate across all tested systems for the data from the several collecting sites ("Overall Totals" row along the bottom of the Table) ranges from 3.6% for the CMU-collected data to 6.8% for the NIST-collected data, reflecting differences in subject populations and other factors.

Table 15 presents the results, in matrix form, of the application of 4 paired-comparison significance tests for the SPREC systems for the Class A+D subset. Among other things, note that the performance differences between the BBN and the CMU systems are not shown to be significant, and that the differences between the MIT, SRI and one of the Unisys systems are also not shown to be significant. Note also that significant differences are shown between the BBN results and those for the two Unisys systems, which make use of BBN-provided N-best results.

A.8.2. Natural Language (NL) Understanding Tests. Table 16 presents a tabulation of the results for the NL tests for all systems and all sets of "answerable" ATIS queries, Class A+D, Class A and Class D.

For the set of all answerable queries, Class A+D, the unweighted error rate ("UW. Err.") ranges from 43.1% to 9.3%. For Class A queries, the range is 28.6% to 6.0%, and for Class D, the range is 63.1% to 13.8%. In each case (and as in last year's results), the lowest error rates were reported by the CMU system.

As noted in Section A9 of this paper, the AT&T NL system was the results of a collaborative agreement with CMU, thus it is not surprising that the performance is nearly identical to that of the CMU system.

There are, in some cases, more than one set of results submitted by individual sites, corresponding to different systems. The differences between systems were specified in the "Systems Descriptions" provided to NIST at the time results were submitted. Space limitations prohibit discussion of these differences in this paper. After preliminary scoring had been completed, Moore at SRI advised NIST that a bug had been found in the code that produced results submitted to NIST for the SRI NL and SLS systems, with the effect of reporting results that were "essentially the output of [the SRI] system with the robust processing component turned off", because a "No\_Answer" response over-wrote the answer produced by the robust processing component (a "template matcher"). With the permission of the ARPA Coordinating Committee, SRI later resubmitted results for the debugged systems, and these SRI results are shown as "late, debugged" results.

Table 17 presents a matrix tabulation of the official NL results for the several subsets of test material. There is some indication of varying degrees of difficulty presented by the different subsets of data from the different sites, subject-scenarios, and subject populations: note that the unweighted error rates reported in the "Overall Totals" row ranges from 28.1% to 16.0%, but also note that both these values were obtained with BBN systems -- one at BBN, and the other at NIST. These differences probably are not significant since the numbers of speakers in the individual test sets is small.

**A.8.3.** Spoken Language System (SLS) Understanding Tests. Table 18 presents a tabulation of the results for the SLS tests for all systems and all sets of "answerable" ATIS queries, Class A+D, Class A and Class D.

For the set of all answerable queries, Class A+D, the unweighted error rate ("UW. Err.") ranges from 46.8% to 13.2%. For Class A queries, the range is 33.5% to 8.9%, and for Class D, the range is 65.2% to 17.5%. For the Class A+D and Class A results, the lowest error rates were obtained by the CMU system, but for the Class D results, the lowest error rates were obtained by the MIT/LCS system.

Table 19 presents a matrix tabulation of the official SLS results for the several subsets of Class A+D test material from different sites. Note that there is some evidence of "local adaptation" to locally collected data (e.g., error rates for the CMU system are substantially lower for the CMU-collected data).

Note also that some sites (typically the "volunteers") continued to use the "No\_Answer" option more frequently than others, which would be a beneficial strategy in a system in which "wrong answers" were penalized more heavily than "no answer". In some cases, use of this option was more prevalent for data from some originating sites than others, perhaps reflecting differences between subject populations or subject-scenario subsets.

#### RANGE ANALYSIS ACROSS SPEAKERS FOR THE TEST: November 1993 Hub 1, Contrast 1 by Speaker Word Error for Speakers

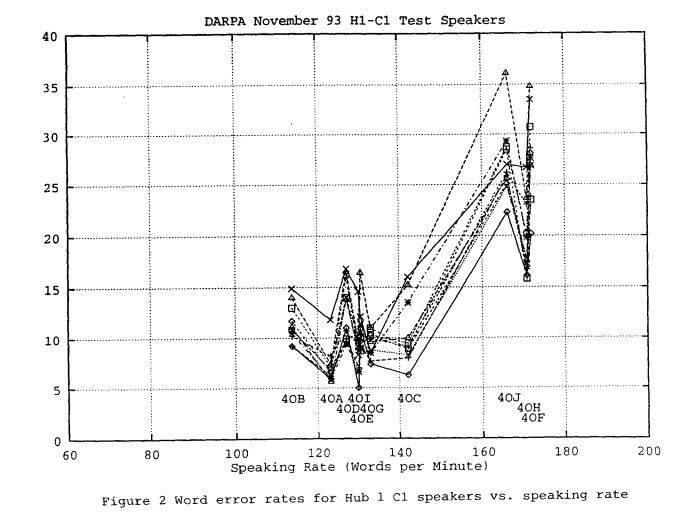
	ļ –									PE	RCENT	AGES									
SPKR	0 	5 	10 	15 	20 	25 	30 	35 	40 	45 	50 	55 	60 	65 	70 	75 	80 	85 	90 	95 	10
40A 40I 40G 40C 40E 40B 40D 40D 40H 40J 40F			+ -+   +- - + -	 -+ -+ -{+ +-	!	-+ -+	+· - +														

-> shows the mean

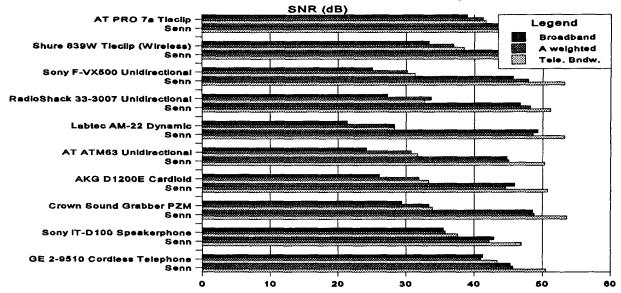
Word Error Rate

-> shows plus or minus one standard deviation

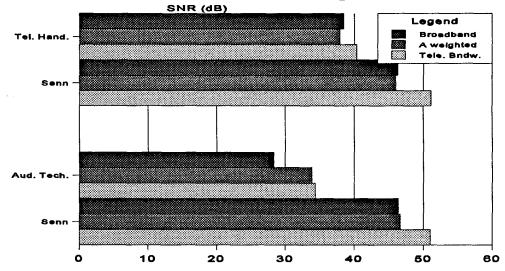
Figure 1 Range of Word error rates for the 10 speakers of the Hub 1 C1 test set for 11 systems



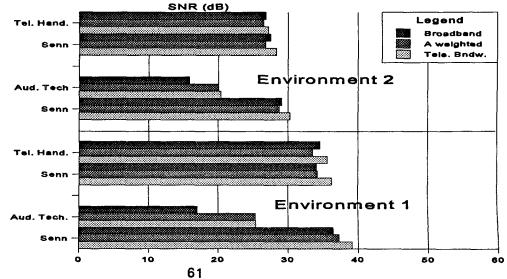
# Fig. 3A: SNR Measurements for Spoke 5











·						
		Hub and S 5 1: 64K F			ac	
GOAL: DATA:	10 speal	basic SI kers * 20 ennheiser	utts = $20$		ean data. IK-word read	d WSJ
	Prim	ary and Co	ntrast Con	nditions		
PO		ies and ut			ning, sessio an as side	on
C1	open-vo		r and cho:		20K trigra ther short	
C2	open-voo		r and cho		20K bigram ther short	
SIDE INFO:		boundarie 90 only.	s and utte	erance of	der are kn	own
	Prima	гу РО	Contra	st C1	Contras	t C2
	Word E		Word Er	r. (%)	Word Err	. (%)
bbn1 bu2 bu3 cmu1 cmu2	11	2.2	14 15 14 14	.7 .3 .5		
cu-htk1 dragon1	+.		12 19		14.	4
limsi1 mit-111	10	5.8	11		15.	2
philips2 sri1			14 14	.4	17. 16.	
**********	COMPAI	RISONS AND	SIGNIFIC		 rs	******
	Test Comp.	<pre>% Change W.E.</pre>	MAPSSWE		nce Tests: Wilcoxon	MCN
bbn1 mit-111	P0:C1 P0:C1	13.9% 9.8%	P0 P0	same PO	P0 P0	P0 P0
	Test Comp.	<pre>% Change W.E.</pre>			ce Tests: Wilcoxon	McN
limsi1 philips2	C1:C2 C1:C2 C1:C2	11.7% 22.7% 14.0%	C1 C1 C1	C1 C1 C1	C1 C1 C1	same C1 same
sri1	C1:C2	13.0%	C1	C1	C1	C1

Table 1 Hub 1 Results

Composite Report of All Significance Teste For the himed Test

Abbrev. Test Name

	Ż	SI	IM	Ŧ
	Matched Pair Sentence Segment (Word Error) Test	Signed Paired Comparison (Speaker Word Accuracy) Test	Wilcowon Signed Rank (Speaker Word Accuracy) Test	McNemar (Sentence Error) Test

		Lame Lame Same Same	same same same	a Amés Famés Pames Pames	aame same same same	l1_c1 eame eame	2222	d_c1 eme		same same same bane	
10-10-111			same same same sri1-h1_c1			cu-htki-hi_ci ame same	#ri1-b1_c1 #ri1-b1_c1 #ri1-b1_c1 #ri1-b1_c1	limeil-h1_c1 same limeil-h1_c1	eri1-b1_c1 eamo eri1-b1_c1 eri1-b1_c1		
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		<b>BISHB</b>		46F9	9579	***	2023	1619 	4519 		
#1-17-17-17-14	bbm1-b1_c1 bbm1-b1_c1 bbm1-b1_c1 bbm1-b1_c1	bul-bl_ct same bul-bl_ct same	bu3-b1_c1 bu3-b1_c1 bu3-b1_c1	bu3-b1_cd bu3-b1_cd bu3-b1_cd bu3-b1_cd	cmui-bi_ci same cmui-bi_ci cmui-hi_ci	cu-btki-bi_ci cu-btki-bi_ci cu-btki-bi_ci cu-btki-bi_ci	91195 911195 911195	limsi1-h1_c1 limsi1-h1_c1 limsi1-h1_c1 limsi1-h1_c1			
_	8559 	<u> </u>	9579 	<u> </u>	<u> </u>	보이부정		9579 			
138811-11-CI	limeit-b1_c1 limeit-b1_c1 limeit-b1_c1 limeit-b1_c1	limi1-b1_c1 limi1-b1_c1 limi1-b1_c1 limi1-b1_c1	limeil-b1_c1 anne limeil-b1_c1 limeil-b1_c1	limeit-b1_c1 limeit-b1_c1 limeit-b1_c1 limeit-b1_c1	limet1-b1_c1 same same limet1-b1_c1	91149 91149 91149 91149	limeil-b1_c1 limeil-b1_c1 limeil-b1_c1 limeil-b1_c1				
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-	<b>4</b> 51 <b>8</b>	<b>B</b> 2 3 4 9	E S H E	- IS IN							
12-11-End	2 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2 2	bu3-b1_c1 eame bu3-b1_c1 eame	900 8 900 8 900 8	1 1 1 1 1 1 1 1 1		4 5 6 8 1 1 1 1 1 1 1 1 1 1 1 1				1 1 1 4 4 4 4 4 4 4 4 4 4 4 4 4 4 4 4 4	
_	<b>W</b> S I W		8559								
bu2-h1_c1	same same same	bu2-bi_c1 same bu2-bi_c1 same		1 1 1 1 4 4 4 4 4 4 1 1 1	4 6 8 8 8 8 8 8 8 8 8 8 8 8 8 8 8 8 8 8	1 1 7 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1	5 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1				
_	<b>B</b> 15 <b>H</b> 19	<b>8</b> 579									
bul-bl_cl	bbm1-h1_c1 same bbm1-h1_c1 bbm1-h1_c1										
	AN IS IN										
12-14-1mm											
-	bbn1-h1_c1	bu1-h1_c1	bu2-h1_c1	bu3-h1_c1	cmu1-h1_c1	cu-htk1-h1_c1	dragon1-h1_c1	limsi1-h1_c1	mit-111-h1_c1	philips2-h1_c1	sri1-h1_c1

.

Table 2 Significance Test Results: Hub 1 (64K Baseline) C1 Systems

·						:
	Nov	93 Hub and S Hub 2: 5K Rea			ac	
GOAL: DATA:	10 6	ove basic SI p peakers * 20 p , Sennheiser p	utts = 200	e on cle utte 51	ean data. (-word read	WSJ
1	P	rimary and Co	ntrast Con	ditions		
PO	boun	) any grammar daries and ut rmation.				on
C1	clos	) Static SI to ed-vocab gram t-term or long ).	mar and ch	oice of	either	.2K
SIDE INFO:		ion boundarie H2-P0 only.	s and utte	rance of	rder are kn	own
	1	Primary	PO	1 4	Contrast C1	
System	I	Word Err.	. (%)	W W	ord Err. (%	)
bu1 bu2		6. 5.	4		11.6 10.3	
bu3 cu-con1		5.1	8	1	10.8 13.5	
cu-htk2	1	4.9	9		8.7	
cu-htk3					12.5	
icsi1					17.7	
limsi2 philips1	1	5.:			9.3 12.3	
philips2	1	6.4		1		i
		************			============	
	COM	PARISONS AND	SIGNIFICAN	CE TESTS	5	
	Teet	% Change			nce Tests:	/
	Comp				Wilcoxon	McN
bu1	P0:C	42.4%	₽0	PO	PO	PO
	P0:C	47.4%	PO	PO	PO	PO
bu3	P0:C	46.6%	PO	₽0	PO	PO
	P0:C		PO	PO	PO	PO
	PO:C	1 43.7%	PO PO	P0 P0	P0 P0	PO PO
burribar	PO:C.	1 25.5%	1	r0	PU	PU [

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Table 3 Hub 2 Results

Composite Report of All Significance Tests For the h2\_c1 Test

Test Name Matched Pair Sentence Segment (Word Error) Test Signed Paired Comparison (Speaker Word Accuracy) Test Wilcovn Signed Marks (Speaker Word Accuracy) Test MeNemar (Sentence Error) Test

Abbrev. MP SI WI MN

	bu1-h2_c1	-	bu2-h2_c1	-	bu3-h2_c1	-no	cu-con1-h2_c1	<u>з</u>	cu-htk2-h2_c1	៩	cu-htk3-h2_c1	Ŧ	lcs11-h2_c1	7	lims12-h2_01	hd	philipsi-h2_c1
bu1-h2_c1		UN IS IN	bu2-h2_c1 same bu2-h2_c1 same	M S I S M	same same same same	AP IS IN	bu1-h2_c1 same same	UN IN IN	cu-htk2-h2_c1 cu-htk2-h2_c1 cu-htk2-h2_c1 cu-htk2-h2_c1 cu-htk2-h2_c1	de la fe	same same same same same same		bu1-h2_c1 bu1-h2_c1 bu1-h2_c1 bu1-h2_c1 bu1-h2_c1	du I I M	lims12-h2_c1 lims12-h2_c1 lims12-h2_c1 lims12-h2_c1 lims12-h2_c1	dw Is IM	s ring s ring s ring s ring s ring
bu2-h2_c1				du I I Mu	bu2-h2_c1 same bu2-h2_c1 bu2-h2_c1 same	MP IS IM	bu2-h2_c1 same same	MP IS IM	cu-htk2-h2_c1 same same cu-htk2-h2_c1	성망부정	bu2-h2_c1 same same	du in In	bu2-h2_c1 bu2-h2_c1 bu2-h2_c1 bu2-h2_c1	du is is is	lims12-h2_c1 lims12-h2_c1 same lims12-h2_c1	dw IN NW	bu2-h2_c1 same same
bu3 -h2_c1						MP IN NW	bu3-h2_c1 same same	AW IS IM	cu-htk2-h2_c1 cu-htk2-h2_c1 cu-htk2-h2_c1 cu-htk2-h2_c1 cu-htk2-h2_c1		bu3-h2_c1 same same	du is in w	bu3-h2_e1 bu3-h2_e1 bu3-h2_e1 bu3-h2_e1 bu3-h2_e1	AN IN N	limsi2-h2_c1 limsi2-h2_c1 limsi2-h2_c1 same limsi2-h2_c1	AM 18 IM	bu3-h2_c1 same same same
cu-con1-h2_c1								AW IN MW	cu-htk2-h2_c1 cu-htk2-h2_c1 cu-htk2-h2_c1 cu-htk2-h2_c1 cu-htk2-h2_c1	du IS IN M	same same same same same same		cu-con1-h2_d1 cu-con1-h2_d1 cu-con1-h2_d1 cu-con1-h2_d1	du li su Nu	lims12-h2_c1 lims12-h2_c1 lims12-h2_c1 lims12-h2_c1 lims12-h2_c1	•	MP same SI philips1-h2_c1 WI philips1-h2_c1 MN same
cu-htk2-h2_c1										AN IS IN	cu-htk2-h2_c1 cu-htk2-h2_c1 cu-htk2-h2_c1 ou-htk2-h2_c1		ou-htk2-h2_01 cu-htk2-h2_01 ou-htk2-h2_01 cu-htk2-h2_01 cu-htk2-h2_01	dw IS IN	emes emes emes	MP IN IN	cu-htk2-h2_c1 cu-htk2-h2_c1 cu-htk2-h2_c1 cu-htk2-h2_c1 cu-htk2-h2_c1
cu-htk3-h2_c1													eu-htk3-h2_e1 eu-htk3-h2_e1 eu-htk3-h2_e1 eu-htk3-h2_e1 eu-htk3-h2_e1	dw I I M	lims12-h2_c1 lims12-h2_c1 lims12-h2_c1 lims12-h2_c1 lims12-h2_c1	AM IN IN	Same Same Same Same
ics11-h2_c1											·			A I I M M	llms12-h2_c1 llms12-h2_c1 llms12-h2_c1 llms12-h2_c1 llms12-h2_c1		MP philips1-h2_c1 SI philips1-h2_c1 WI philips1-h2_c1 WN philips1-h2_c1
l i ms i 2 - h 2_c1																MP SI MN	11ms12-h2_c1 same 11ms12-h2_c1 11ms12-h2_c1
philips1-h2_c1										4 4 8		t t 1 1					

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MCN 50 10 B spkrs \* 40 utts = 400 utts (rapid enrollment from S3 speakers, used for S3-P0) 10 A spkrs \* 40 utts = 400 utts (rapid enrollment from Hub speakers, used for S3-C2) 5K-word read, WSJ data, Semnhelser mic, collected from non-native speakers of American English (British, European, Asian dialects, etc.). Word Err. (%) | Word Err. (%) speaker identity is known for P0, C1, and C2, session boundaries and utterance order is known Contrast C2 evaluate a rapid enrollment speaker adaptation algorithm on difficult speakers. 10 B spkrs \* 40 utts = 400 utts (test) Significance Testa: MAPSSWE Sign Wilcoxon 10.7 (req) S3~P0 system with speaker adaptation disabled (req) rapid enrollment speaker adaptation 90 Nov 93 Hub and Spoke CSR Evaluation Spoke 3: SI Recognition Outliers COMPARISONS AND SIGNIFICANCE TESTS Primary and Contrast Conditions 04 Contrast C1 (req) S3-P0 system on H2 data 32.0 04 & Change W.E. 54.7% Word Err. (%) Primary PO 14.5 for C3. Test Comp. P0:C1 GOAL: DATA SIDE INFO: 40111111100 111111 System ប 50 ដ bbn2 bbn2

Word Err. (%) | Word Err. (%) | Word Err. (%) | Contrast C2 (opt) incremental unsupervised LM adaptation evaluate an incremental supervised LM adaptation algorithm on a problem of sublanguage adaptation. 4 A spirs \* 1-5 articles (~100 utts) = 400 utts. Read unfiltered WSJ data from 1990 ubblioations in TIPSTER corpus, Semhelser mic, minimum of 20 sentences per article. (req) incremental supervised LM adaptation, closed vocabulary, any LM trained from 1987-89 WSJO texts session boundaries and utterance order are 16.9 18.3 18.9 Change W.E. Change W.E. Change W.E. 19.4% 9.8% 13.6% 14.0% 2.18 5.38 9.98 9.98 98 98 98 98 98 17.78 4.88 9.18 10.58 (req) S1-P0 system with LM adaptation disabled Nov 93 Hub and Spoke CSR Evaluation Spoke 1: Language Model Adaptation COMPARISONS AND SIGNIFICANCE TESTS Primary and Contrast Conditions Contrast C1 20.5 19.2 18.4 21.12 P0:C1 P0:C1 P0:C1 P0:C1 P0:C1 Test P0:02 P0:02 P0:02 P0:02 Test Comp. Test Comp. 22222 22222 Primary PO 16.5 17.3 18.2 known cmu3 Utts 1-5 cmu3 Utts 6-10 cmu3 Utts 11-15 cmu3 Utts 16+ cmu3 Utts 1-5 cmu3 Utts 6-10 cmu3 Utts 11-15 cmu3 Utts 16+ Utts 1-5 Utts 6-10 Utts 11-15 Utts 16+ cmu3 Utts 1-5 cmu3 Utts 6-10 cmu3 Utts 11-15 cmu3 Utts 16+ GOAL: DATA: SIDE INFO: cmu3 cmu3 cmu3 cmu3 0d ដ 3 System

Table 6 Spoke 3: SI Recognition Outlier Results

Table 5 Spoke 1: Language Model Adaptation Results

Nov 33 Hub and Spoke CSR Evaluation Spoke 5: Microphone Independence	GOÀL: evaluate an unsupervised channel compensation algorithm DATA: 10 A spkrs * 20 utts = 200 utts (2 channels, same speech as H2	5K-word read WSJ data), 10 different mics not in training or development test. NOTE: No speech	from the test microphones can be used.	Primary and Contrast Conditions	) (req) unsupervised channel compensation enabled on wv2 data	i (req) S5-P0 system with compensation disabled on wv2 data	(reg) S5-P0 system on Sennheiser (wv1) data	(opt) S5-C1 system on	INFO: Microphone identities are not known		13.1 17.2 6.6 6.6   Comparisons and significance tests		Test & Change Significance Tests: Comp. W.E. MAPSSWE Sign Wilcoxon McN	P0:C1     27.8%     P0     same     P0     same       P0:C1     24.2%     P0     same     P0     same	Test & Change Significance Tests: Comp. W.E. MAPSSWE Sign Wilcoxon MCN	C2:P0 36.1% C2 C2 C2 C2 C2:P0 49.5% C2 C2 C2 C2 C2	Test & Change Significance Tests: Comp. W.E. MaPSSWE Sign Wilcoxon MCN	C2:C1 53.8% C2 same C2 C2   C2:C1 61.7% C2 C2 C2 C2	Test & Change Significance Tests: Comp. W.E. MAPSSWE sign Wilcoxon MCN	C31PO 35.9% C3	Test & Change Significance Tests: Comp. W.E. MAPSSWE sign Wilcoxon MCN	C31C1     53.7%     C3     same     C3     C3	Test & Change Significance Tests: Comp. W.E. MAPSSWE sign Wilcoxon MCN	C2:C3 0.3% same same same same C2:C3 0.0% same same same same
	<u> </u>				0d	5	62	Ü	SIDE		sri2 ======	1		cmu4 sr12		cmu4 sr12		cmu4 sr12		cmu4 sri2		cmu4 sr12		cmu4 sr12
<u> </u>	eaker utts (test)	tts (rapid in S3)	nhelser mic.		ised speaker	eaker	eđ	n boundaries	own <i>;</i> er the fact.		14.5	11.4	10.2 9.6 11.0	4 4 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1		e F	same same		same	4 <sup></sup> C1	icance Tests: Kence Tests:			same same same
: CSR Evaluation peaker Adaptation	n incremental speak. algorithm. * 100 utts = 400 ut	40 utts = 16 from A speak	ad WSJ data, Sennheiser mic.	st Conditions	emental unsupervised	0 system with speaker disabled	emental supervised	nditions: session boundaries	are known; correct own after the	Word Err	19.4 19.6 20.3 13.3				CTCT.	e Significance Te MAPSSWE	0 0 0 0 6 6 6 6			.me ratio 54_DU/54_C1 0.914 1.024	Significance Mabecur		c C 2 2 c C	same same C2 same
93 Hub and Spoke CSR Evaluation 4: Incremental Speaker Adaptation	n incremental algorithm. * 100 utts =	ts = 16 A speak				<i>i</i> th	(opt) incremental supervised adaptation	for all conditions: session boundaries	utterance orden  tional for C2: nscription is kr		15.5 19.4 14.5 15.3 19.6 14.5 15.4 20.3 13.3	11.6	11.6 10.4 12.0	::::::::::::::::::::::::::::::::::::::	ALL DESCRIPTION OF THE PARTY OF	* Change   Significance Te W.E.   MAPSSWE		5.88 same	7.0% same	14 - 1 25 - 1 25 - 1	significance	120 6.2% same	P0 13.9% C2	-4.1% same 11.8% same 1.4% same 1.4% same

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Table 7 Spoke 4: Incremental Speaker Adaptation Results

"Microphone Independence" Results

Table 8 Spoke 5:

5K-word read WSJ data, same 2 secondary mics as in 56, collected in two environments with a background the sold for a bout 55-68 Db. WOTE: the 800 feareo midrophone adaptation utterances will come from the devtest and are the only data from the target mics that is allowed. The only data available for adaptation to the environment will be from the S7 Spoke of the devtest data. Microphone identities are known. Use of the stereo environment-adaptation data will be allowed for the \$7-P0 condition only evaluate a noise compensation algorithm with known alectrate mic. 10 Å spkrs \* 10 utts \* 2 mics \* 2 envs = 400 utts (test, 2 channels) MGN g Word Err. (%) Contrast C2 (req) S7-P0 system with compensation disabled wv2 data Significance Tests: E Sign Wilcoxon 6.3 9.1 (req) S7-P0 system on Sennheiser (wv1) data 8 8 4 6 Nov 93 Hub and Spoke CSR Evaluation Spoke 7: Noisy Environments ....... COMPARISONS AND SIGNIFICANCE TESTS Primary and Contrast Conditions sign \*\*\*\*\*\*\* ......... MAPSSWE Word Err. (%) 8888 Contrast C1 8.5 17.4 29.1 28.8 % Change W.E. 25.08 47.88 71.28 71.48 \*\*\*\*\*\*\*\* Test Comp. sri2 at\_e1 sri2 at\_e2 sri2 th\_e1 sri2 th\_e1 sri2 th\_e2 sri2 at\_e1 sri2 at\_e2 sri2 th\_e1 sri2 th\_e2 DATA: GOAL: SIDE INFO System ដ ដ

SK-word read WSJ data, from an Audio-Technica directional stand-mounted mic and telephone handset over external lines, plus stereo mic adaptation data. NOTE: the 800 stereo microphone adaptation utterances will come from the devtest and are the only data from the target mics that is allowed. The data available for adaptation to the environment will be from the S' Spoke of the Microphone identities are known. Use of the stereo mic-adaptation data will be allowed for the S6-P0 condition only (req) supervised mic adaptation enabled on wv2 data Fame PO PO PO MaN McN (req) S6-P0 system with mic adaptation disabled on Word Err. (%) 8888 McN evaluate a known microphone adaptation algorithm. 10 A spkrs \* 20 utts \* 2 mics = 400 utts (test, 2 channels 10 D spkrs \* 40 utts \* 2 mics = 800 utts (mic-adaptation from devtest, 2channels) Contrast C2 Significance Tests: MAPSSWE Sign Wilcoxon Wilcoxon Significance Tests: E Sign Wilcoxon (req) S6-C1 system on Sennheiser (wv1) data Tests PO PO Same PO C gue C C Significance Nov 93 Hub and Spoke CSR Evaluation Spoke 6: Known Alternate Microphone COMPARISONS AND SIGNIFICANCE TESTS Primary and Contrast Conditions same PO same PO sign sign Word Err. (%) 00000 gg 0000 Contrast C1 10.4 18.5 19.5 19.5 19.5 19.5 MAPSSWE MAPSSWE 0000 m 0 04040 8888 Change W.E. Change W.E. Change 9.98 57.48 11.78 61.38 18.38 32.78 15.58 42.38 26.48 71.38 77.78 77.78 7.78 62.58 Word Err. (%) W.E. Primary PO devtest data. 9.4 12.5 25.3 25.3 æ data P0:C1 P0:C1 P0:C1 C2:P0 C2:P0 C2:P0 C2:P0 Test Comp. Test Comp. Test Comp. WV2 bbn3 at bbn3 th dragon3 at dragon3 th sr12 at sr12 th GOAL: DATA: at th at th at th SIDE INFO: bbn3 at bbn3 th dragon3 a dragon3 t bbn3 at bbn3 th dragon3 a dragon3 t bbn3 at bbn3 th dragon3 a dragon3 t sr12 at sr12 th \*\*\*\*\* РО បី ដ System bbn3 bbn3

Table 10 Spoke 7: "Noisy Environments" Results

Table 9 Spoke 6: Known Alternate Microphones Results

	MCN	same	P0 Pues	same	same same	MCN	same	C2	same C2 C2	2	McN	same	30	same C2 C2	MaN	ទទ	Ü	same C3 C3	MCN	same C3	C S	000	WCN	ະ ບ	same	c3	Same	
	significance Tests: Sign Wilcoxon	same	04	Bane	04 6	cance Tests: Wilcoxon	same	5 5	same C2	C2	ance Tests: Wilcoxon	same	3 8	same C2 C2	ance resus: Wilcoxon	ខទ	ទ	same C3 C3	nce Tests: Wilcoxon	ទទ	ទេខ	388	nce Tests: Wilcoxon	ព	ទ	38	ទ	
rvaluation s Sources ANCE TESTS	significan	same	04	same	04 04	significan sign	same	same C2	same C2	8	significa sign	same	3 8	same C2 C2	 signicia	ទទ	ប	same C3 C3	Significance Sign W	.same C3	ទខ	រ ប ប	sign W	ប	same 22	c.3 Bame	ទ	
Spoke CSR F rated Nolfs D SIGNIFIC	MAPSSWE	same	04	same	04	MAPSSWE	same	55	Bame C2		MAPSSWE	same	38	888	 MAPSSWE	ខ១	ប	ខខខ	MAPSSWE	ទទ	ទទ	000	MAPSSWE	C	8	C3	C)	
OV 93 Hub and Spoke CSR Evaluation Spoke 8: Calibrated Noise sources COMPARISONS AND SIGNIFICANCE TESTS	& Change W.E.		39.58	4.98	30.9% 12.9%	* Change W.E.	•	31.046 69.148	14.18	66.0 <del>8</del>	& Change W.E.		76.7 <del>8</del>	18.48 63.78 70.48	W.E.	22.68 38.68	71.2%	21.98 52.88 66.08	e Change W.E.	23.5% 62.9%	78.3%	67.48 70.48	& Change W.E.	16.0%	11.18	9.18	10.28	
Nov Spo	Test Comp.	P0:C1	P0:C1	Di Ci	P0:C1	Test Comp.	C2: P0	C2: P0	C2: P0	C2:P0	Test Comp.	53 10 10 10		555	Comp.	C3: P0 C3: P0	C3 : P0	C3: P0 C3: P0 C3: P0	Test Comp.	C3.C1	88 19 19	555 555	Test Comp.	C3 : C2	330	322	C3:C2	
- - - - - - - - - - 					cmu5 tr_10 cmu5 tr_0				cmu5 tr_20 cmu5 tr_10			cmu5 mu_20		cmu5 tr_20 cmu5 tr_10 cmu5 tr 0		cmu5 mu_20 cmu5 mu_10	cmu5 mu_0	cmu5 tr_20 cmu5 tr_10 cmu5 tr_0		cmu5 mu_20 cmu5 mu 10		cmu5 tr_10 cmu5 tr_0					cmu5 tr 10	

	rith known brated	ils = 600	competing ackground at -Technica . NOTE: the rances will Y data from		data	disabled on	data	data	rast	Word Err. (%)	11.3	16.9	12.1	12.0
Evaluation se Sources	compensation algorithm with known data corrupted with calibrated	sources * 3 levels	d with In the b ie Audic from S6 from S6 on utte the onl	Conditions	enabled on wv2 d	compensation disa	Sennheiser (wv1) da	Sennheiser (wv1) da	trast	Word Err. (%)	13.5	1.81	13.3	13.4
Hub and Spoke CSR Ev 8: Calibrated Noise		* 10 utts * 2 2 channels)	5K-word read WSJ data collected recorded music or talk radio in 0, 10, and 20 Db SNR, using the directional stand-mounted mic fr directional stand-mounted and arc tic 400 stereo microphone adaptation come from the devtest and arc tit the target mic that is allowed.	and Contrast	compensation	system with	system on	system on	1 1 1	Word Err. (%)	14.8	1.25	16.3	36.9 86.3
Nov 93 Hul Spoke 8:	evaluate a noise alternate mic on noise sources	10 A spkrs utts (test,	5X-word rea recorded mu 0, 10, and directional 400 stereo come from t the target	Primary	(req) noise	(req) 58-P0 wv2 data	(req) 88-P0	(opt) S8-C1	Primary PO	Word Err. (%)	14.0	58.7	15.5	25.5
	GOAL :	DATA:			PO	C1	C2	ប		System	cmu5 mu 20	cmu5 mu_0		cmu5 tr_10   cmu5 tr_0

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Table 11 Spoke 8: "Calibrated Noise Sources" Results

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	Dec	93 ATIS	S SPREC	Test R	esults		
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unisys3-adx	• •		2.9	6.0		23.5	964
		Clai	se A+D	Subset			
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10		97.5	0.0	5 C	8 4 0 6	18.0	173 
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unisys3-d		96.4		0.6	• •	10	325
		Clas	ss X Su	bset			
		- 1		9	\$		
	W. Brr 15.5	Corr 87.3	9.7	3.0	2.8 2.8	U. EFF 62.8	# UCC. 191
bbn3-x	1		• •		-	~	0
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unisys2-x unisvs3-x	• •		• •		• •	÷ .	<u>ግ ወ</u>
2	•	;	•	•			

spontaneous way biccatton i SI performance on spontaneous le speech. i * 20 utts = 200 utts isy-like dictations (business news inheiser mic		D.			Contrast C2		13.7   24.7		e Tests: Wilcoxon MaN	P0 P0
s on spont 10 utts 10ns (busi	onditions	ic training	10	8	c1	3 (8)		F	ificanc Sign	04
. performance on speech. 20 utts = 200 ut 11te dictations siser mic	Contrast Conditions	or acoustic	on H1 data	on s9 data	Contrast	Word Err	13.	D SIGNIFICANCE	sign MAPSSWE	PO
style 5 style 5 ers * 2 ts WSJ-1 Sennhei	Primary and C	any grammar	S9-PO system	H1-C1 system	mary P0		19.1   13.1	COMPARISONS AND	& Change W.E.	22.8%
improve ba dictation- 10 C speak Spontaneou storles),	Ρr	(req) a	(req) S	(reg) H	Primary	Word	· · · · · · · · · · · · · · · · · · ·	COM	Test Comp.	P0:C2
GOAL: DATA:		04	C1	C2		system	bbn4			4114

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Table 13 ATIS SPREC Benchmark Test Results

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Table 12 Spoke 9: Spontaneous WSJ Dictation Results

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#### Dec93 ATIS SPREC Test Results '

		BBN (146 Utt.)	CMU	Class A+D Originating Site MIT   (132 Utt.)	NIST-SRI (77 Utt.)	SRI (166 Utt.)	Overall Totals 773	Foreign Coll. Site Totals
4	att2	6.3 1.8 1.3 9.4 49.3	3.4 1.4 1.0 5.8 25.2		8.6 2.6 2.0 13.2 49.4			5.4 2.1 1.5 9.0 42.2
1	bbn3		1.4 0.4 0.6					
	cmu2	2.5 0.6 0.4 3.5 23.3	1.3 0.5 0.7			2.0 0.4 0.7 3.1 16.9		
s Y	crim3	3.0 0.4 1.3 4.8 24.0	2.1 0.7 2.1 4.9 23.3			4.1 0.8 2.3 7.2 32.5	3.6 0.7 2.3 6.6 31.3	
T E M	mit_lcs2	2.2 0.8 0.4 3.4 23.3	1.7 1.5 0.5 3.7 16.6	3.2 1.4 1.0 5.6 31.8				
	sri3	2.8 1.0 0.7 4.6 27.4	1.8 1.0 0.6 3.5 14.1	2.6 1.4 0.7 4.6 30.3			2.5 1.4 0.7 4.6 23.3	2.6 1.4 0.8 4.8 25.2
	sri4	2.2 1.0 0.5 3.8 23.3	1.8 1.0 0.6 3.4 13.5	2.4 1.2 0.8 4.4 28.0				2.4 1.4 0.9 4.6 23.7
	unisys2	1.3 0.5 0.7 2.6 14.4	1.4 0.3 1.3 3.0 14.7			2.6 0.6 0.6 3.9 20.5		2.3 0.7 1.0 4.0 21.9
	unisys3	0.6 0.1 0.5	1.5 0.4 1.1 3.0 15.3			3.1 0.9 0.3 4.3 20.5	3.9 19.7	2.3 0.6 0.9 3.9 19.7
Ove	arall als	2.4 0.7 0.7 3.9 22.1	1.8 0.8 1.0 3.6 16.2			3.1 1.0 0.8 4.9 22.4	1	
	eign tem		1.9 0.8 1.0 3.7 16.8					*Sub *Del *Ins W.Err *Utt.Err

Matrix tabulation of results for the Dec93 ATIS SPREC Test Results, for the Class A+D Subset.

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Matrix columns present results for Test Data Subsets collected at several sites, and matrix rows present results for different systems.

Mumbers printed at the top of the matrix columns indicate the number of utterances in the Test Data (sub)set from the corresponding site.

"Overall Totals" (column) present results for the entire Class A+D Subset for the system corresponding to that matrix row. "Foreign Coll. Site Totals" present results for "foreign site" data (i.e., excluding locally collected data) for the Class A+D Subset.

"Overall Totals" (row) present results accumulated over all systems corresponding to the Test Data (sub)set corresponding to that matrix column. "Foreign System Totals" present results accumulated over "foreign systems" (i.e., excluding results for the system(s) developed at the site responsible for collection of that Test Data subset.)

Table 14 ATIS SPREC Results: Class (A+D) by Collection Site

Composite Report of All Significance Testa For the Dec93 ATIS SPREC Class A+D Test Results Test

Test Name

Abbrev. ...... MP SI WI MN Matched Pair Sentence Segment (Word Error) Test Matched Paired Comparison (Speaker Word Accuracy) Test Wilcoxon Signed Rank (Speaker Word Accuracy) Test Molemar (Sentence Error) Test

-	att2-a_d	а —	bbn3-a_d	5	cmu2-a_d	0	crim3-a_d	mit_lcs2-a_d	_	sri3-a_d	¢i	sri4-a_d	an	unisys2-a_d	ת _	unisys3-a_d
att2-a_d		du IS MA	bbn3-a_d bbn3-a_d bbn3-a_d bbn3-a_d bbn3-a_d	MP IS MN MN	cmu2-a_d cmu2-a_d cmu2-a_d cmu2-a_d	AM IN IM	crim3-a_d crim3-a_d crim3-a_d crim3-a_d crim3-a_d	MP mit_lcs2-a_d SI mit_lcs2-a_d WI mit_lcs2-a_d MN mit_lcs2-a_d		8713-8.0	AN IS IM	8r14-2.d 8r14-2.d 8r14-2.d 8r14-2.d	AM IN IM	unisys2-a_d unisys2-a_d unisys2-a_d unisys2-a_d	MP SI MN MN	unisys3-a_d unisys3-a_d unisys3-a_d unisys3-a_d unisys3-a_d
bn3-a_d				MP IS IN		MP SI MN	bbn3-a_d bbn3-a_d bbn3-a_d bbn3-a_d bbn3-a_d	MP bbn3-a_d SI 8ame WI bbn3-a_d MN bbn3-a_d	<b>4</b> 5 <b>5</b> 4	bbn3-a_d bbn3-a_d bbn3-a_d bbn3-a_d bbn3-a_d	AW Ng Ng	bbn3-a_d bbn3-a_d bbn3-a_d bbn3-a_d	MP SI MN	bbn3-a_d bbn3-a_d bbn3-a_d bbn3-a_d	MP IN IN	bbn3-a_d same bbn3-a_d bbn3-a_d same
cmu2-a_d						dw IS IM	cmu2-a_d cmu2-a_d cmu2-a_d cmu2-a_d cmu2-a_d	MP cmu2-a_d SI same WI same MN cmu2-a_d		cmu2-a_d same same cmu2-a_d	AN IS NO	Cmu2 - a_d same same	AN IN IN	Cmu2-a_d Bame Bame Bame	AM IN NM	000000 E E E E E E E E
crim3-a_d								MP mit_lcs2-a_d SI mit_lcs2-a_d WI mit_lcs2-a_d MN mit_lcs2-a_d		8r13-8_0 8r13-8_0 8r13-8_0 8r13-8_0 8r13-8_0	MR S I N	8r14-9_0 8r14-9_0 8r14-9_0 8r14-9_0	MR IN NW	unisys2-a_d unisys2-a_d unisys2-a_d unisys2-a_d	MP SI MN MN	unisys3-a_d unisys3-a_d unisys3-a_d unisys3-a_d unisys3-a_d
mit_lcs2-a_d					r r l l l l l l				dis in M	вате вате вате вате	d I S NA	9 8 9 9 9 8 9 8 9 8 8 8 8 8 8 8 8 8 8 8	MP IS IM	S S T C S S S T C S S S T C S S S T C S S S T C S T C S	AM IS IM	unisys3-a_d same same unisys3-a_d
sri3-a_d						·					AN IS NO	9 11 9 11 9 11 9 11 9 11 9 11 9 11 9 1	MP IN MM	8 ай 8 ай 8 ай 8 ай 8 ай	AM IS IM NM	unisys3-a_d same same unisys3-a_d
sri4-a_d						• • •							MP IS IM NM	зате Зате Зате Зате	MP IS IN MW	8 8 8 8 8 8 8 8 8 8 8 8 8 8 8 8 8 8 8
unisys2-a_d				; ; ; ; ;	 										MP SI, NM	same same same un <i>isys</i> 3-a_d
unisys3-a_d															·	

Table 15 Significance Test Results: ATIS SPREC Systems

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	Class A+D 773 Utts.	Class A 448 Utts.	Class D 352 Utts.
system	UW Err.	UW Err.	UW Err.
att1	10.2	7.4	14.2
bbn1	14.7	9.6	21.8
bbn2	22.4	16.1	31.1
cmu1	9.3	6.0	13.8
crim1	36.4	21.7	56.6
crim2	20.8	14.7	29.2
mit_lcs1	12.5	10.0	16.0
sri1	21.9	14.3	32.3
sri5 **	18.2	10.5	28.9
unisys1	43.1	28.6	63.1

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Table	16	ATIS	$\mathbf{NL}$	Test	Results

	********		BEN		CMU	Originating   NIT		NIS	Test Dat T-SRI		ST-BEN		SRI	Over Tota	18	Foreign Coll. Site	
	att1	124 85	146 22 0 15 0 15.1	153	163 10 0 6 0 6.1	132 121 11 92 8 8.3	0	67 87	77 10 0 13 0 13.0	   BC   90		149 90	166 17 0 10 0 10.2	694 7 90 1	9 0	Totals 694 79 90 10 10.2	0
	bbni	124   85	21 1 14 1 15.1	141   87		117 15   89 11   11.4	0	57 74	20 0 26 0 26.0	82	7 0	138 83	28 0 17 0 16.9	659 11	.3 1	535 92 85 15 14.7	0
	bbn2	104	41 1 28 1 28.8	127   78	36 0 22 0 22.1	112 20 85 15 15.2	0	54 70	23 0 30 0 29.9	69	20 0 22 0 22.5	134 81	32 0 19 0 19.3	600 17 78 2		496 131 79 21 20.9	0
	cmu1	128 88		153   94		120 12 91 9 9.1	0	67 87	10 0 13 0 13.0	80		153 92	13 0 8 0 7.8	701 7	2 0 9 0 9.3	548 62 90 10 10.2	0
S Y S	criml	76 52	61 9 42 6 47.9	11 <b>4</b>   70		88 36 67 27 33.3	8	40 52	32 5 42 6 48.1	79   89	10 0 11 0 11.2	95	57 14 34 8 42.8	492 22 64 2 3		492 227 64 29 36.4	7
T E M S	crim2	112		133   82		109 23   83 17   17.4	0	57 74	17 3 22 4 26.0	76   85		125 75	41 0 25 0 24.7	612 15 79 2 2		612 153 79 20 20.8	8
	mit_lcs1	111   76	35 0 24 0 24.0	150   92	13 0 8 0 8.0	120 12   91 9   9.1	0	62 81	15 0 19 0 19.5	83		150   90	16 0 10 0 9.6	676 9 87 1		556 85 87 13 13.3	0
	sri1	103 71	17 26 12 18 29.5	130   80		111 8 84 6 15.9	10	54 70	21 2 27 3 29.9	75   84	14 0 16 0 15.7	131   79	30 5 16 3 21.1	604 10 78 1 2		473 77 5 78 13 22.1	57
	sri5 **	113 77	30 3 21 2 22.6	142   87	17 4 10 2 12.9	117 14 89 11 11.4	1	54 70	21 2 27 3 29.9	75	14 0 16 0 15.7	131   79	30 5 18 3 21.1	82 1		501 96 83 16 17.5	2
	unisys1	52	31 39 21 27 47.9	i	20 26 46.0	91 26 69 20 31.1	11   		31 17 48.1	55	27 13 30 15 44.9	58	12 30 42.2	4	1 22 3.1	57 21 2 43.1	22
0v		1071		1331			37   3		193 25	748	128 14 14 2 16.0	1302	284 74	11		Legend:	222
Sγ	reign stem tals	843 72		1178 80	206 83 14 6 19.7	986 165 83 14 17.0	3	552 72			128 14 14 2 16.0	1040			#T   %T   % Ur	#F #NA %F %NA h-Weighted A	A

Matrix tabulation of results for the Dec 93 ATIS NL Test Results - Using Minimal/Maximal Scoring Criterion, for the Class (A+D) Subset.

Matrix columns present results for Test Data Subsets collected at several sites, and matrix rows present results for different systems.

Numbers printed at the top of the matrix columns indicate the number of evaluable utterances in the Test Data (sub)set from the corresponding site.

"Overall Totals" (column) present results for the entire Class (A+D) Subset for the system corresponding to that matrix row. "Foreign Coll. Site Totals" present results for "foreign site" data (i.e., excluding locally collected data) for the Class (A+D) Subset.

"Overall Totals" (row) present results accumulated over all systems corresponding to the Test Data (sub)set corresponding to that matrix column. "Foreign System Totals" present results accumulated over "foreign systems" (i.e., excluding results for the system(s) developed at the site responsible for collection of that Test Data subset.)

\*\* Late and for a debugged system.

Table 17 ATIS NL Results: Class (A+D) by Collection Site

	Class A+D 773 Utts	Class A 448 Utts.	Class D 352 Utts.
system	UW Err.	UW Err.	UW Err.
att1	24.6	22.1	28.0
bbn1	17.5	13.8	22.5
cmu1	13.2	8.9	19.1
crim1	43.3	28.6	63.7
crim2	28.2	23.7	34.5
mit_lcs1	14.2	11.8	17.5
sri1	24.8	16.5	36.3
sri2	25.4	18.5	34.8
sri5 **	20.7	14.1	29.8
sri6 **	21.2	13.8	31.4
unisys1	46.8	33.5	65.2

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Table	18	ATIS	SLS	Test	Results
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			BBN 146		CNU 163	1	Class ( ating Si NIT 132	te of	Set Test Dat ST-SRI 77		ST-BBN 89		SRI 166	Overa Total: 773	5	Foreign Coll. Site Totals
	att1	106 73		138   85	25 0 15 0 15.3	1	32 0 24 0 24.2		19 0 25 0 24.7		17 0 19 0 19.1		34 0 34.3	583 190 75 25 24		583 190 0 75 25 0 24.6
	bbn1	121 83	24 1 16 1 17.1	128 79	35 0 21 0 21.5	117 89	15 0 11 0 11.4	61 79	16 0 21 0 20.8		14 0 16 0 15.7	136	30 0 18 0 18.1	638 134 83 17 17	1 0	517 110 0 82 18 0 17.5
	cmu1		19 0 13 0 13.0	152 93	11 0 7 0 6.7	114 86	18 0 14 0 13.6		13 0 17 0 16.9		13 0 15 0 14.6	138	28 0 17 0 16.9	671 102 87 13 13	Ó	
	crim1		65 8 45 5 50.0		44 19 27 12 38.7	82 62			35 4 45 5 50.6	66 74	21 2 24 2 25.8		72 15 43 9 52.4	438 279 57 36 43	7	438 279 56 57 36 7 43.3
S Y	crim2		40 2 27 1 28.8	121 74	40 2 25 1 25.8	99 75	33 0 25 0 25.0		20 2 26 3 28.6	69 78		107 64	58 1 35 1 35.5	555 209 72 27 28	9 1 .2	555 209 9 72 27 1 28.2
-	mit_lcs1		36 0 25 0 24.7	148 91	15 0 9 0 9.2	116   88			14 0 18 0 18.2	81   91	8 0 9 0 9.0	145	21 0 13 0 12.7	663 110 86 14 14	0 0 .2	547 94 0 85 15 0 14.7
M S	sri1		19 33 13 23 35.6	140 86	20 3 12 2 14.1	99 75	11 22 8 17 25.0		16 15 21 19 40.3		20 1 22 1 23.6		28 <b>4</b> 17 2 19.3	581 114 75 15 24	10	447 86 74 74 14 12 26.4
	sri2	100 68		121 74			12 17 9 13 22.0		23 1 30 1 31.2	67 75		133   80	29 4 17 2 19.9	577 126 75 16 25	9	444 97 66 73 16 11 26.9
	sri5 **		38 3 26 2 28.1	140   86	20 3 12 2 14.1		19 1 14 1 15.2		21 2 27 3 29.9		20 1 22 1 23.6	134 81	28 4 17 2 19.3	613 146 79 19 20	2	479 118 10 79 19 2 21.1
	sri6 **	111 76	32 3 22 2 24.0	133 82	27 3 17 2 18.4		19 1 14 1 15.2	53 69	23 1 30 1 31.2		22 2	133   80	29 4 17 2 19.9	609 150 79 19 21	2	476 121 10 78 20 2 21.6
	unis <b>ys1</b>	49	36 38 25 26 50.7	88 54	31 44 19 27 46.0	64	33 15 25 11 36.4		38 19 57.1	55		51	18 31 48.8	411 188 53 24 46	23 8	411 188 174 53 24 23 46.8
0ve		1123	364 119 23 7 30.1	1409 : 79			250 64 17 4 21.6	578	229 40 27 5 31.8	758	200 21 20 2 22.6		410 83			Legend:
Sys			340 118 23 8 31.4			77	234 64 18 5 22.6	68	229 40 27 5 31.8		200 21 20 2 22.6		296 67 25 6 31.2		#T %T % Ur	#F #NA %F %NA h-Weighted Ern

Matrix tabulation of results for the Dec 93 ATIS SLS Test Results - Using Minimal/Maximal Scoring Criterion, for the Class (A+D) Subset.

Matrix columns present results for Test Data Subsets collected at several sites, and matrix rows present results for different systems.

Numbers printed at the top of the matrix columns indicate the number of evaluable utterances in the Test Data (sub)set from the corresponding site.

"Overall Totals" (column) present results for the entire Class (A+D) Subset for the system corresponding to that matrix row. "Foreign Coll. Site Totals" present results for "foreign site" data (i.e., excluding locally collected data) for the Class (A+D) Subset.

"Overall Totals" (row) present results accumulated over all systems corresponding to the Test Data (sub)set corresponding to that matrix column. "Foreign System Totals" present results accumulated over "foreign systems" (i.e., excluding results for the system(s) developed at the site responsible for collection of that Test Data subset.)

\*\* Late and for a debugged system.

Table 19 ATIS SLS Results: Class (A+D) by Collection Site