DATA-DRIVEN CODEBOOK ADAPTATI ON I N PHONETI CALLY TI ED SCHMMS

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ABSTRACT

This paper reports the results of our experiments aimed at the automatic optimization of the number of parameters in the semi-continuous phonetically tied HMMbased speech ion systemthat is part of JANUS-2.

In ferent algorithm devised to determine the el parameters. In recognition exameous human-to-human timization of

smaller the average distance between a data point and its closest reference is. It is also computationally cheap. However, there is the need to define a threshold, which is highly ble for an unsupervised algorithm the recognition experiments performed with regiven below

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is criterion, we compute the probability p(training) and abort the process of adding new refven probability threshold has been vortional to the recognizer's gonly samples of theory of

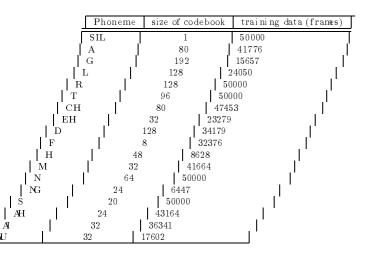


Table 2. Godebook size for some phonemes

ognition results

The recognition results shown in table 3 have been achieved context dependent generalized triphones and a dependent generalized triphones and a dependent you. Only first-best reach triphones were applied, and applied generalized. Dependent of the property weed of the property of the pr

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