

# DATA-DRIVEN CODEBOOK ADAPTATION IN PHONETICALLY TIED 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 recognition system that is part of JANUS-2. Different algorithms devised to determine the optimal number of parameters are evaluated in recognition experiments. In recognition experiments, the automatic optimization of the number of parameters is compared to the manual optimization of the number of parameters.

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 variable for an unsupervised algorithm. The recognition experiments performed with are given below.

### **k e l i h o o d - c r i t e r i o n**

In this criterion, we compute the probability  $p(\text{training})$  and abort the process of adding new references when the probability threshold has been reached. This is proportional to the recognizer's performance on only samples of the training set. The theory of

Phoneme	size of codebook	training data (frames)
SIL	1	50000
A	80	41776
G	192	15657
L	128	24050
R	128	50000
T	96	50000
CH	80	47453
EH	32	23279
D	128	34179
F	8	32376
H	48	8628
M	32	41664
N	64	50000
NG	24	6447
S	20	50000
AH	24	43164
A	32	36341
AU	32	17602

Table 2. Codebook size for some phonemes

### Recognition results

The recognition results shown in table 3 have been achieved with context dependent generalized triphones and a perplexity of 70. Only first-best re-  
 triphones were applied, and  
 took place. D-  
 used *Codebook HMM Speech*  
*Recognition Systems*, Proc. ICASSP 1994

Justin and R. Schwartz, *A Comparison of Several Approximate Algorithms for Finding N-best Hypotheses*, ICASSP 1991, vol 1, pp 701-704.

- [8] T. Schultz and I. Regina, *Acoustic and Language Modeling of Human and Nonhuman Noises for Human-to-Human Spontaneous Speech Recognition*, Internal report of Interactive Systems Labs., Universität Karlsruhe, 1994, submitted to ICASSP 95.