

# DATA-DRIVEN CODEBOOK ADAPTATION IN PHONETICALLY TIED SCHMMS

*Thomas Kemp*

*<kemp@ira.uka.de>*

Interactive Systems Laboratories,  
Department of Computer Science,  
University of Karlsruhe,  
76131 Karlsruhe, Germany

## 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, the automatic optimization of the number of parameters is compared to manual optimization. The results show that the automatic optimization of the number of parameters leads to a significant improvement in recognition performance.

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-  
 tri-grams were applied, and  
 were used. Codebook HMM Speech  
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