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Chapter # 2: Using HTK

HMM TOOL KIT HTK

Outline

- ✓ Introduction
- ✓ Data Preparation
- ✓ Creating Monophone HMMs
- ✓ Creating Tied-State Triphones
- ✓ Recognizer Evaluation
- ✓ Running the Recognizer Live
- Adapting the HMMs
- Semi-Tied and HLDA transforms

Introduction

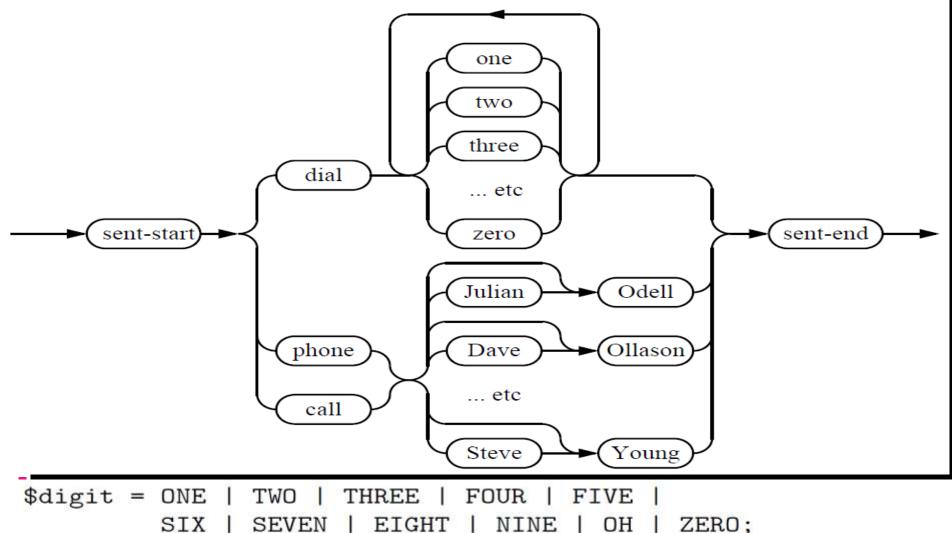
- The construction of a recognizer for simple voice dialing applications will be described
- The recognizer will be designed to recognize continuously spoken digit strings and a limited set of names

Data Preparation

- For training speech will be recorded from scratch and to do this scripts are needed to prompt for each sentence
- The prompt scripts will be used in conjunction with a pronunciation dictionary to provide the initial phone level transcriptions needed to start the HMM training process
- A task grammar will be used as a random generator for test data
- Before the data can be recorded
 - A phone set must be defined
 - A dictionary must be constructed to cover both training and testing
 - A task grammar must be defined

Step 1 - The Task Grammar

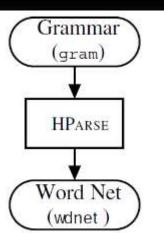
- HTK provides a grammar definition language for specifying simple task grammars
- It consists of a set of variable definitions followed by a regular expression describing the words to recognize
- For the voice dialing application, a suitable grammar might be as



Step 1 - The Task Grammar

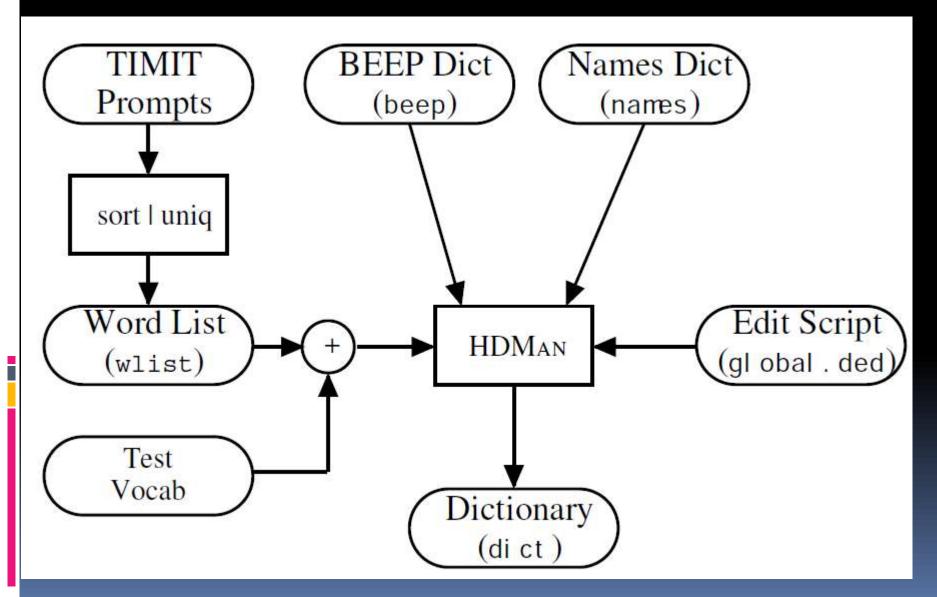
- The HTK recognizer requires a word network to be defined using a low level notation called HTK Standard Lattice Format (SLF) in which each word instance and each word-to-word transition is listed explicitly
- This word network can be created automatically from the grammar above using the HParse tool

HParse gram wdnet



- The first step in building a dictionary is to create a sorted list of the required words
- For robust acoustic models, the training data will consist of English sentences unrelated to the phone recognition task
- The first few items from the TIMIT DB might be as follows

SOOO1 ONE VALIDATED ACTS OF SCHOOL DISTRICTS
SOOO2 TWO OTHER CASES ALSO WERE UNDER ADVISEMENT
SOOO3 BOTH FIGURES WOULD GO HIGHER IN LATER YEARS
SOOO4 THIS IS NOT A PROGRAM OF SOCIALIZED MEDICINE
etc



 Create a new dictionary called dict by searching the source dictionaries beep and names to find pronunciations for each word in wlist

```
HDMan -m -w wlist -n monophones1 -l dlog dict beep names
```

- names is a manually constructed file containing pronunciations for the proper names used in the task grammar
- The general format of the dictionary is
 WORD [outsym] p1 p2 p3

ah sp ax sp ey sp CALL k ao l sp DIAL d ay ax 1 sp EIGHT ey t sp PHONE f ow n sp SENT-END sil SENT-START sil SEVEN s eh v n sp TO t ax sp TO t uw sp z ia r ow sp **ZERO**

Step 3 - Recording the Data

- The training and test data will be recorded using the HTK tool HSLab (a combined waveform recording and labelling tool)
- HSLab is invoked by typing

HSLab noname

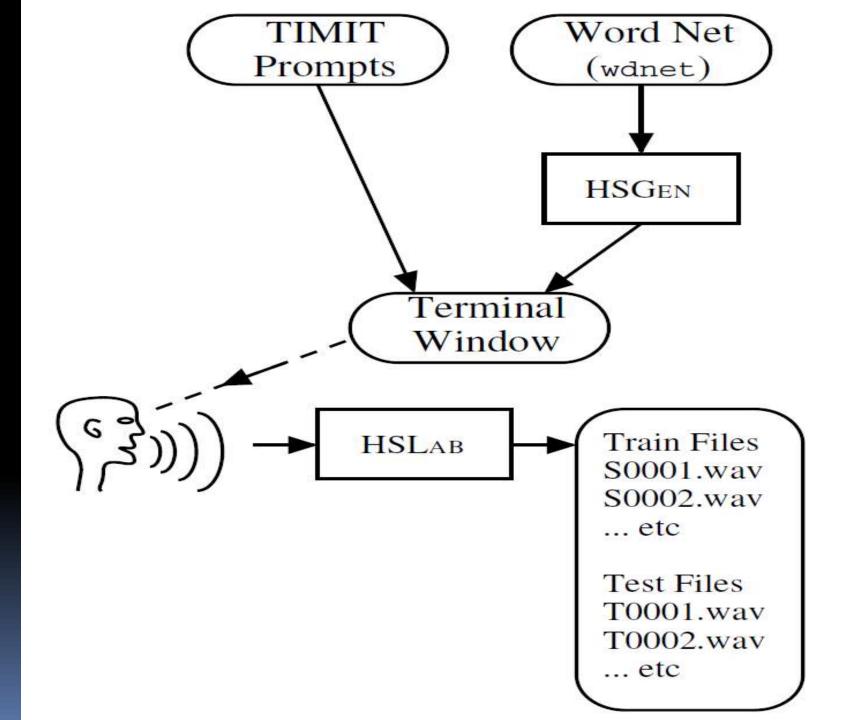
- Each time the record button is pressed, it writes the subsequent recording alternately to a file called noname_o. and to a file called noname_1.
- Thus, it is simple to write a shell script which for each successive line of a prompt file, outputs the prompt, waits for either noname_o. or noname_1. to appear, and then renames the file to the name prepending the prompt

Step 3 - Recording the Data

- The prompts for test sentences need to be generated before recording them
- The tool HSGen can be used to do this by randomly traversing a word network and outputting each word encountered

HSGen -l -n 200 wdnet dict > testprompts

- PHONE YOUNG
- 2. DIAL OH SIX SEVEN SEVEN OH ZERO
- 3. DIAL SEVEN NINE OH OH EIGHT SEVEN NINE NINE
- 4. DIAL SIX NINE SIX TWO NINE FOUR ZERO NINE EIGHT
- CALL JULIAN ODELL
- ... etc



Step 4 - Creating the Transcription Files

- Every file of training data must have an associated phone level transcription
- An orthographic transcription in HTK label format can be created using a text editor or a scripting language

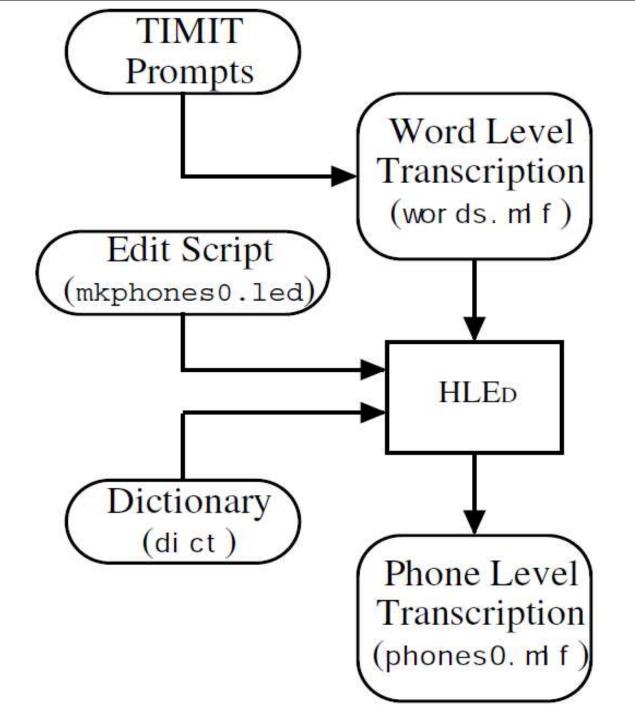
```
"*/S0002.lab"
                                "*/S0003.lab"
#!MLF!#
"*/S0001.lab"
                                BOTH
                TWO
                                FIGURES
ONE
                OTHER
                                (etc.)
VALIDATED
               CASES
ACTS
                ALSO
OF
                WERE
SCHOOL
                UNDER
DISTRICTS
                ADVISEMENT
```

Step 4 - Creating the Transcription Files

 Once the word level MLF has been created, phone level MLFs can be generated using the label editor HLEd

HLEd -1 '*' -d dict -i phones0.mlf mkphones0.led words.mlf

mkphones0	Output
EX	#!MLF!#
IS sil sil	"*/S0001.lab"
DE sp	sil
\$100 1	W
	ah
	n
	V
	ae
	1
	ih
	d
	etc



Step 5 - Coding the Data

- The final stage of data preparation is to parameterize the raw speech waveforms into sequences of feature vectors
- Coding can be performed using the tool HCopy configured to automatically convert its input into MFCC vectors

HCopy -T 1 -C config -S codetr.scp

Coding parameters TARGETKIND = MFCC_0 Step 5 TARGETRATE = 100000.0SAVECOMPRESSED = TThe firsavewithcrc = T parame WINDOWSIZE = 250000.0 sequen USEHAMMING = T Coding PREEMCOEF = 0.97 configu NUMCHANS = 26 MFCC V CEPLIFTER = 22 NUMCEPS = 12**HCopy**

ENORMALISE = F

1 .

ration is to veforms into

e tool HCopy : its input into

codetr.scp

```
# Coding parameters
         TARGETKIND = MFCC_O
Step 5
         TARGETRATE = 100000.0
         SAVECOMPRESSED = T
■ The firsavewithcrc = T
                                 ration is to
                                veforms into
  parame WINDOWSIZE = 250000.0
  sequen USEHAMMING = T
                                e tool HCopy
Coding PREEMCOEF = 0.97
  configu NUMCHANS = 26
                                 its input into
  MFCCVCEPLIFTER = 22
         NUMCEPS = 12
HCopy
                                codetr.scp
         ENORMALISE = F
```

/root/sjy/waves/S0001.wav /root/sjy/train/S0001.mfc /root/sjy/waves/S0002.wav /root/sjy/train/S0002.mfc /root/sjy/waves/S0003.wav /root/sjy/train/S0003.mfc /root/sjy/waves/S0004.wav /root/sjy/train/S0004.mfc (etc.)

Creating Monophone HMMs

- A set of identical monophone HMMs in which every mean and variance is identical is created
- These are then retrained, short-pause models are added and the silence model is extended slightly
- The monophones are then retrained
- The recogniser tool HVite is used to perform a forced alignment, a new phone level MLF is created in which the choice of pronunciations depends on the acoustic evidence
- This new MLF can be used to perform a final reestimation of the monophone HMMs

- The first step in HMM training is to define a prototype model
- For phone-based systems, a good topology to use is 3-state left-right with no skips

```
~o <VecSize> 39 <MFCC_0_D_A>
~h "proto"
<BeginHMM>
                                    <State> 4
 <NumStates> 5
                                       <Mean> 39
 <State> 2
                                         0.0 0.0 0.0 ...
    <Mean> 39
                                       <Variance> 39
      0.0 0.0 0.0 ...
                                         1.0 1.0 1.0 ...
    <Variance> 39
                                    <TransP> 5
      1.0 1.0 1.0 ...
                                    0.0 1.0 0.0 0.0 0.0
 <State> 3
                                    0.0 0.6 0.4 0.0 0.0
    <Mean> 39
                                     0.0 0.0 0.6 0.4 0.0
      0.0 0.0 0.0 ...
                                    0.0 0.0 0.0 0.7 0.3
    <Variance> 39
                                     0.0 0.0 0.0 0.0 0.0
      1.0 1.0 1.0 ...
                                   <EndHMM>
```

 The HTK tool HCompV will scan a set of data files, compute the global mean and variance and set all of the Gaussians in a given HMM to have the same mean and variance

HCompV -C config -f 0.01 -m -S train.scp -M hmm0 proto

- Given this new prototype model stored in the directory hmmo
 - A Master Macro File (MMF) called hmmdefs containing a copy for each of the required monophone HMMs is constructed
 - Relabeling it for each required monophone (including "sil")

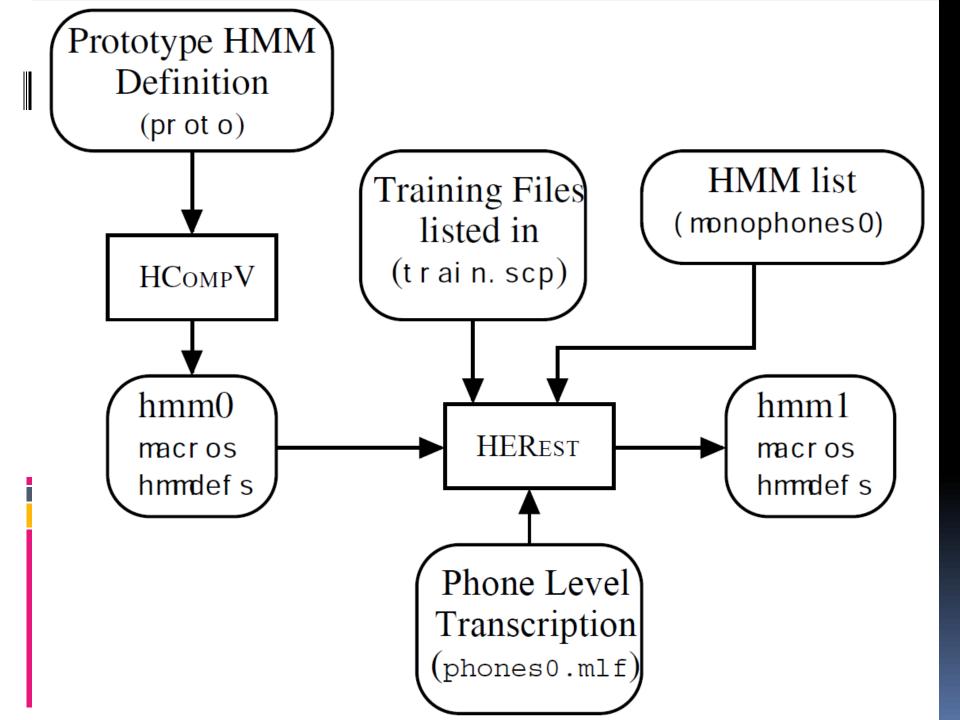
macros

hmmdefs

 The flat start monophones stored in the directory hmmo are re-estimated using the embedded re-estimation tool HERest invoked as follows

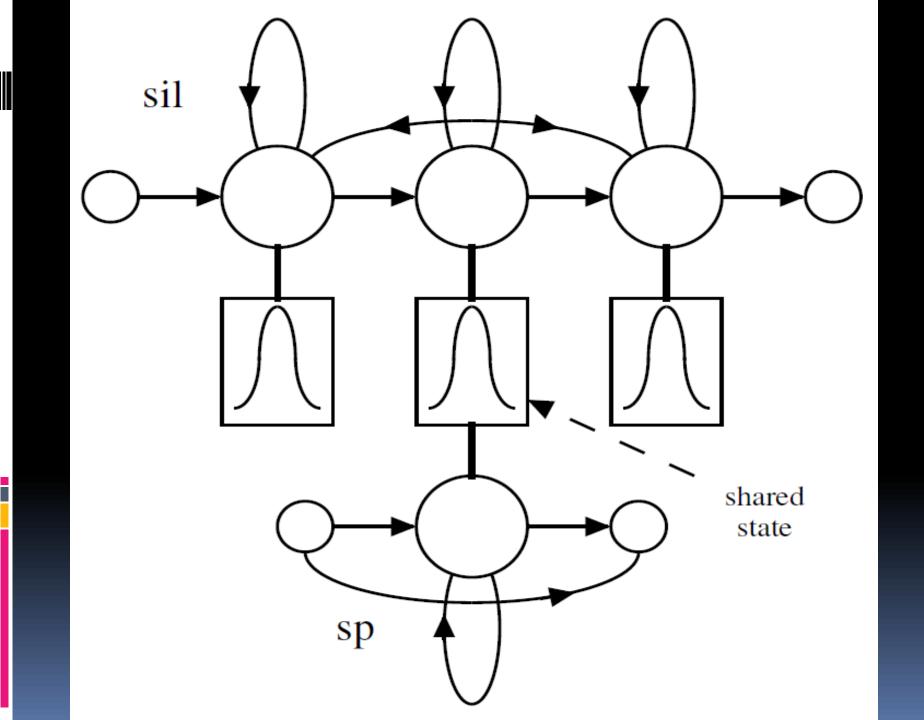
```
HERest -C config -I phones0.mlf -t 250.0 150.0 1000.0 \
-S train.scp -H hmm0/macros -H hmm0/hmmdefs -M hmm1 monophones0
```

- This loads all the models in hmmo which are listed in the model list monophoneso
- These are then re-estimated using the data listed in train.scp
- The new model set is stored in the directory hmm1
- -t option sets the pruning thresholds to be used during training
- Pruning limits the range of state alignments that the forward-backward algorithm includes in its summation



- Each time HERest is run it performs a single reestimation
- Each new HMM set is stored in a new directory
- Execution of HERest should be repeated twice more, changing the name of the input and output directories (set with the options -H and -M) each time, until the directory hmm3 contains the final set of initialized monophone HMMs

- Extra transitions from states 2 to 4 and from states 4 to 2 in the silence model are added
- The idea is to make the model more robust by allowing individual states to absorb the various impulsive noises in the training data while avoiding transit to the following word
- A 1 state short pause sp model should be created with a direct transition from entry to exit node
- This sp has its emitting state tied to the center state of the silence model



- These silence models can be created in two stages
 - Use a text editor on the file hmm3/hmmdefs to copy the center state of the sil model to make a new sp model and store the resulting MMF hmmdefs, which includes the new sp model, in the new directory hmm4
 - Run the HMM editor HHEd to add the extra transitions required and tie the sp state to the center sil state
- HHed applies a set of commands in a script to modify a set of HMMs

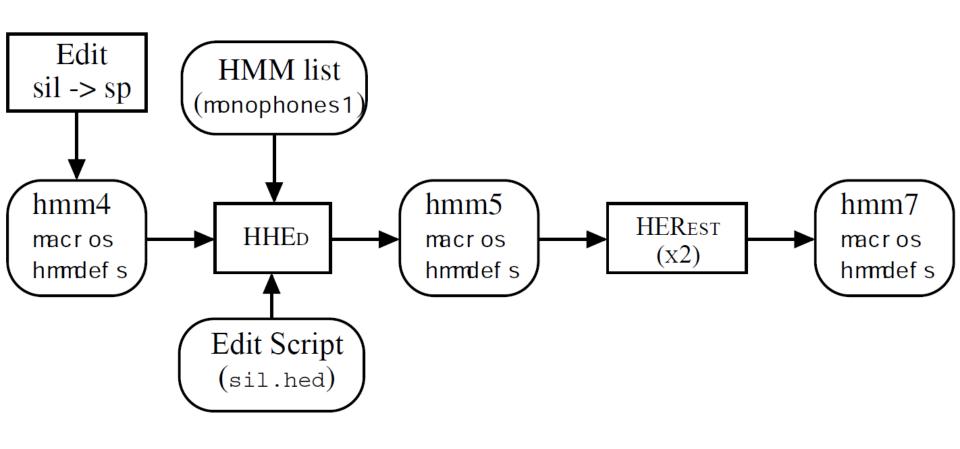
HHEd -H hmm4/macros -H hmm4/hmmdefs -M hmm5 sil.hed monophones1

```
AT 2 4 0.2 {sil.transP}
AT 4 2 0.2 {sil.transP}
AT 1 3 0.3 {sp.transP}
TI silst {sil.state[3],sp.state[2]}
```

 HHed applies a set of commands in a script to modify a set of HMMs

HHEd -H hmm4/macros -H hmm4/hmmdefs -M hmm5 sil.hed monophones1

- Two passes of HERest are applied using the phone transcriptions with sp models between words
- This leaves the set of monophone HMMs created so far in the directory hmm7



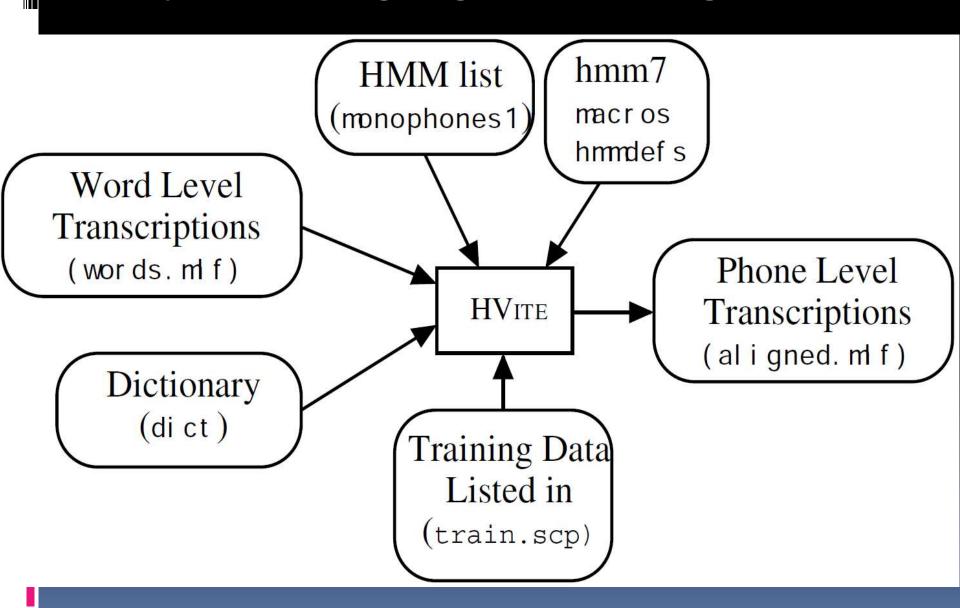
Step 8 - Realigning the Training Data

- The dictionary contains multiple pronunciations for some words, particularly function words
- The phone models created so far can be used to realign the training data and create new transcriptions
- This can be done with a single invocation of the HTK recognition tool Hvite

```
HVite -l '*' -o SWT -b silence -C config -a -H hmm7/macros \
   -H hmm7/hmmdefs -i aligned.mlf -m -t 250.0 -y lab \
   -I words.mlf -S train.scp dict monophones1
```

 Once the new phone alignments have been created, another 2 passes of HERest can be applied to reestimate the HMM set parameters again

Step 8 - Realigning the Training Data



Creating Tied-State Triphones

- Given a set of monophone HMMs, the final stage of model building is to create context-dependent triphone HMMs
 - The monophone transcriptions are converted to triphone transcriptions and a set of triphone models are created by copying the monophones and reestimating
 - Similar acoustic states of these triphones are tied to ensure that all state distributions can be robustly estimated

- Context-dependent triphones can be made by simply cloning monophones and then reestimating using triphone transcriptions
- To convert the monophone transcriptions in aligned.mlf to an equivalent set of triphone transcriptions in wintri.mlf we use

```
HLEd -n triphones1 -l '*' -i wintri.mlf mktri.led aligned.mlf
```

 At the same time, a list of triphones is written to the file triphones1

```
sil th ih s sp m ae n sp ...

sil th+ih th-ih+s ih-s sp m+ae m-ae+n ae-n sp ...

WB sp

WB sil

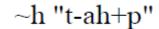
TC
```

 The cloning of models can be done efficiently using the HMM editor HHEd

```
HHEd -B -H hmm9/macros -H hmm9/hmmdefs -M hmm10 mktri.hed monophones1
```

 The edit script mktri.hed contains a clone command CL followed by TI commands to tie all of the transition matrices in each triphone set

```
CL triphones1
TI T_ah {(*-ah+*,ah+*,*-ah).transP}
TI T_ax {(*-ax+*,ax+*,*-ax).transP}
TI T_ey {(*-ey+*,ey+*,*-ey).transP}
TI T_b {(*-b+*,b+*,*-b).transP}
TI T_ay {(*-ay+*,ay+*,*-ay).transP}
```



<transP> 0.0 1.0 0.0 ..

0.0 0.4 0.6 ..

~h "t-ah+b"

<transP>

0.0 1.0 0.0 ..

0.0 0.4 0.6 ..

HHED Tie Command

~t "T ah"

<transP> 0.0 1.0 0.0 .. 0.0 0.4 0.6 ..

~h "t-ah+p"

~t "T_ah"

~h "t-ah+b"

~t "T_ah"

 The cloning of models can be done efficiently using the HMM editor HHEd

```
HHEd -B -H hmm9/macros -H hmm9/hmmdefs -M hmm10 mktri.hed monophones1
```

 The edit script mktri.hed contains a clone command CL followed by TI commands to tie all of the transition matrices in each triphone set

```
CL triphones1
TI T_ah {(*-ah+*,ah+*,*-ah).transP}
TI T_ax {(*-ax+*,ax+*,*-ax).transP}
TI T_ey {(*-ey+*,ey+*,*-ey).transP}
TI T_b {(*-b+*,b+*,*-b).transP}
TI T_ay {(*-ay+*,ay+*,*-ay).transP}
```

 The cloning of models can be done efficiently using the HMM editor HHEd

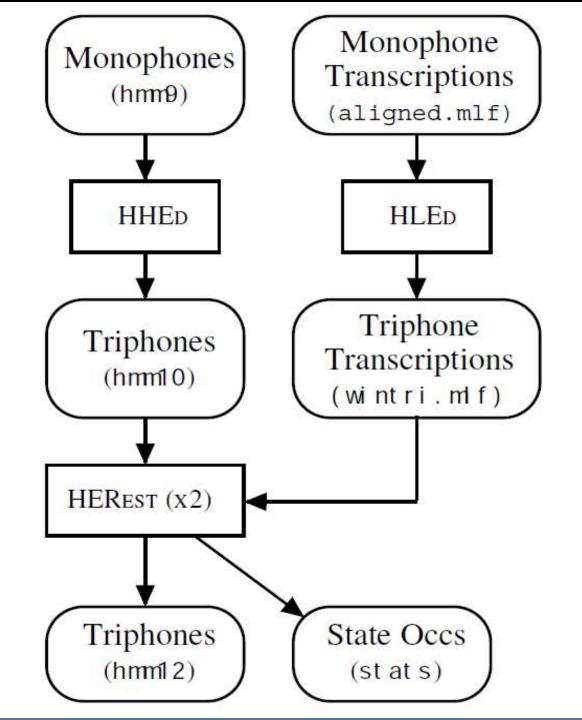
```
HHEd -B -H hmm9/macros -H hmm9/hmmdefs -M hmm10 mktri.hed monophones1
```

The edit script mktri.hed contains a clone contributed by Ti contributed tie all of the transition matrices in each triphone set

```
TI T_ah {(*-ah+*,ah+*,*-ah).transP}
TI T_ax {(*-ax+*,ax+*,*-ax).transP}
TI T_ey {(*-ey+*,ey+*,*-ey).transP}
TI T_b {(*-b+*,b+*,*-b).transP}
TI T_ay {(*-ay+*,ay+*,*-ay).transP}
```

- The transition parameters do not vary significantly with acoustic context but nevertheless need to be estimated accurately
- Once the context-dependent models have been cloned, the new triphone set can be re-estimated using HERest
- For the final pass of HERest, the -s option should be used to generate a file of state occupation statistics called stats.

HERest -B -C config -I wintri.mlf -t 250.0 150.0 1000.0 -s stats \
-S train.scp -H hmm11/macros -H hmm11/hmmdefs -M hmm12 triphones1

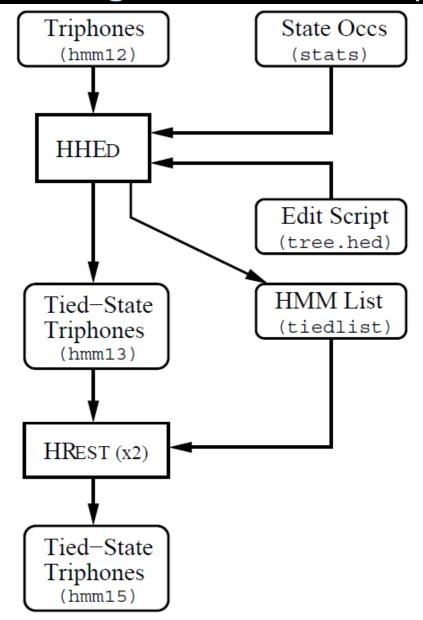


- The outcome of the previous stage is a set of triphone HMMs with all triphones in a phone set sharing the same transition matrix
- When estimating these models, many of the variances in the output distributions will have been floored since there will be insufficient data associated with many of the states
- The last step in the model building process is to tie states within triphone sets in order to share data and thus be able to make robust parameter estimates
- The performance of the recognizer depends crucially on how accurate the state output distributions capture the statistics of the speech data

- HHEd provides two mechanisms which allow states to be clustered and then each cluster tied
 - Data-driven: using a similarity measure between states
 - Decision trees: based on asking questions about the left and right contexts of each triphone
- Decision tree state tying is performed by running HHEd as follows

HHEd -B -H hmm12/macros -H hmm12/hmmdefs -M hmm13 \
tree.hed triphones1 > log

```
TR 2
RO 100.0 stats
TR 0
                                             TB 350.0 "aa_s2" {(aa, *-aa, *-aa+*, aa+*).state[2]}
QS "L_Class-Stop" \{p-*,b-*,t-*,d-*,k-*,g-*\}
                                             TB 350.0 "ae_s2" {(ae, *-ae, *-ae+*, ae+*).state[2]}
QS "R_Class-Stop" {*+p,*+b,*+t,*+d,*+k,*+g}
                                             TB 350.0 "ah_s2" \{(ah, *-ah, *-ah+*, ah+*).state[2]\}
QS "L_Nasal" {m-*,n-*,ng-*}
                                             TB 350.0 "uh_s2" {(uh, *-uh, *-uh+*, uh+*).state[2]}
QS "R_Nasal" {*+m,*+n,*+ng}
QS "L_Glide" {y-*,w-*}
                                             TB 350.0 "y_s4" {(y, *-y, *-y+*, y+*).state[4]}
QS "R_Glide" {*+y,*+w}
                                             TB 350.0 "z_s4" \{(z, *-z, *-z+*, z+*).state[4]\}
                                             TB 350.0 "zh_s4" {(zh, *-zh, *-zh+*, zh+*).state[4]}
QS "L w" {w-*}
QS "R_w" {*+w}
                                             TR 1
QS "L_y" {y-*}
QS "R_y" {*+y}
                                             AU "fulllist"
QS "L_z" {z-*}
                                             CO "tiedlist"
QS "R z" {*+z}
                                             ST "trees"
```



Recogniser Evaluation

- The recognizer is now complete and its performance can be evaluated
- The recognition network and dictionary have already been constructed, and test data has been recorded
- All that is necessary is to run the recogniser and then evaluate the results using the HTK analysis tool HResults

Step 11 - Recognizing the Test Data

 Assuming that test.scp holds a list of the coded test files, then each test file will be recognised and its transcription output to an MLF called recout.mlf

```
HVite -H hmm15/macros -H hmm15/hmmdefs -S test.scp \
    -l '*' -i recout.mlf -w wdnet \
    -p 0.0 -s 5.0 dict tiedlist
```

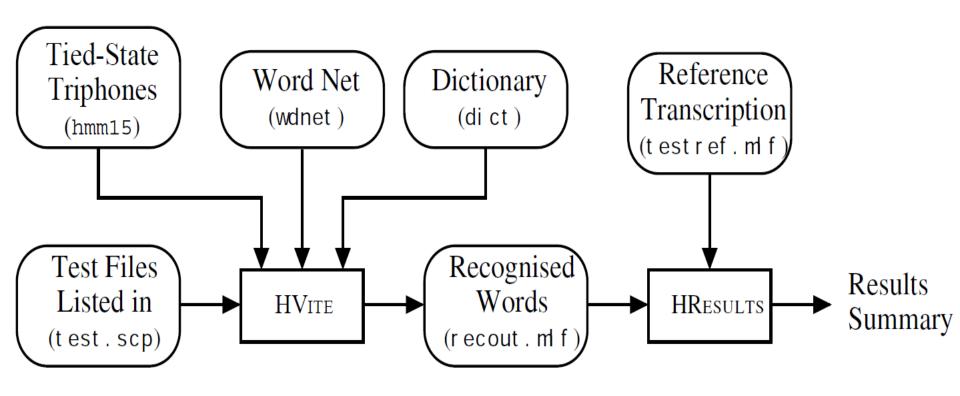
- The options -p and -s set the word insertion penalty and the grammar scale factor
- Assuming that the MLF testref.mlf contains word level transcriptions for each test file

Step 11 - Recognizing the Test Data

```
HResults -I testref.mlf tiedlist recout.mlf
Date: Sun Oct 22 16:14:45 1995
 Ref: testrefs.mlf
 Rec : recout.mlf
        ------ Overall Results ------
SENT: %Correct=98.50 [H=197, S=3, N=200]
WORD: %Corr=99.77, Acc=99.65 [H=853, D=1, S=1, I=1, N=855]
```

 Assuming that the MLF testref.mlf contains word level transcriptions for each test file

Step 11 - Recognizing the Test Data



Running the Recogniser Live

The recognizer can also be run with live input

```
# Waveform capture
SOURCERATE=625.0
SOURCEKIND=HAUDIO
SOURCEFORMAT=HTK
ENORMALISE=F
USESILDET=T
MEASURESIL=F
OUTSILWARN=T

HVite -H hmm15/macros -H hmm15/hmmdefs -C config2 \
-w wdnet -p 0.0 -s 5.0 dict tiedlist
```

ThankYou