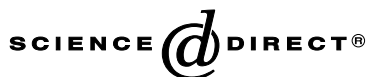




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A mixed-level switching dynamic system for continuous speech recognition [☆]

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Abstract

9 A two-level mixture linear dynamic system model, with frame-level switching parameters in the obser-
10 vation equation and with segment-level switching parameters in the target-directed state equation, is de-
11 veloped and evaluated. The main contributions of this work are: (1) the new framework for dealing with
12 mixed-level switching in the dynamic system and (2) the novel use of piecewise linear functions, enabled by
13 the introduction of frame-level switching, to approximate the nonlinear function between the hidden vocal-
14 tract-resonance space and the observable acoustic space. The approximation is accomplished by the frame-
15 dependent switching parameters in the observation equation. In this paper, in a self-contained manner, we
16 highlight the key algorithm differences from the earlier model having only single segment-level switching
17 that is synchronous between the state and observation equations. A series of speech recognition experi-
18 ments are carried out to evaluate this new model using a subset of Switchboard conversational speech data.
19 The experimental results show that the approximation accuracy is improved with an increased number of
20 switching-parameter values. The speech recognizer built from the new mixed-level switching dynamic
21 system model using an N-best re-scoring evaluation paradigm show moderate word error rate reduction
22 compared with using either single-level switching or no switching parameters.

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24

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25 1. Introduction

26 In recent years, a new approach to the challenging problem of conversational speech recognition
27 has emerged, holding a promise to overcome some fundamental limitations of the conventional
28 Hidden Markov Model (HMM) approach (e.g., Bridle et al., 1998; Deng, 1999; Deng and Ma, 2000;
29 Ma and Deng, 2003; Picone et al., 1999). This new approach is a radical departure from the current
30 HMM-based statistical modeling approaches. Rather than using a large number of unstructured
31 Gaussian mixture components to account for the tremendous variation in the observable acoustic
32 data of highly coarticulated spontaneous speech, the new speech model that we have developed
33 provides a rich structure for the partially observed (hidden) dynamics in the domain of vocal-tract-
34 resonances (VTRs) (Deng and Ma, 1999, 2000). In the design of the speech recognizer reported in
35 Deng and Ma (1999, 2000), we use a statistical nonlinear dynamic system to describe the physical
36 process of spontaneous speech production where knowledge of the VTR dynamic behavior in
37 speech production is naturally incorporated into the model training and decoding.

38 In the previous work documented in Deng (1999), Deng and Ma (1999, 2000) and Ma and
39 Deng (2003), a long-span coarticulatory model, one for each phone segment, was formulated in
40 mathematical terms as a constrained state-space nonlinear system. The state equation represents a
41 stochastic linear system, where the state variable represents the VTR hidden dynamics. The target-
42 directed or asymptotic behavior of the dynamics is established by forcing the system to enter the
43 asymptotic regime after large time steps. The measurement equation in the state-space model is a
44 static nonlinear mapping from the hidden dynamic (VTR) space to the observable acoustic space
45 (e.g., MFCC).

46 In our earlier work (Deng and Ma, 1999, 2000), due to the use of the nonlinear mapping
47 function, $h(Z)$, two approximations had to be made in the model parameter learning process. The
48 first approximation was $E[h(Z)] \approx h(E[Z])$. This amounts to making Taylor series expansion on
49 $h(E[Z(k)])$ and then truncating all high-order terms above the linear term. This overcomes the
50 difficulty in the calculation of the expectation of the nonlinear function, at the expense of an
51 unknown degree of reduced accuracy. The second approximation arises from the use of the ex-
52 tended Kalman filter (EKF) (Kitagawa, 1987; Mendel, 1995; Tanizaki, 1996) (due to the presence
53 of the nonlinear function), which was known to be non-optimal. In order to minimize the loss of
54 computational accuracy in parameter learning and likelihood calculation due to these approxi-
55 mations (but at the expense of a possible loss of modeling accuracy), the work reported in Ma and
56 Deng (2003) developed a mixture linear dynamic system model, where several (mixture) linear
57 regression mapping functions in the measurement equation were used to approximate the single
58 nonlinear mapping function. The switching between one mixture component to another occurred
59 at the phone-segment level [in a manner analogous to the segmental models described in Gish and
60 Ng (1993), Ostendorf et al. (1996)]; that is, the same mixture component was sustained across the
61 entire segment and the new mixture component may be switched to only at the new segment
62 boundary. Also, the same segmental constraint was applied to the switching of the parameters
63 (targets and time constants) in the state equation. Since the parameter switchings in the state
64 equation and in the measurement equation are synchronous, we call that model (Ma and Deng,
65 2003) as the single-level switching dynamic system.

66 In this paper, we introduce the mixed-level switching dynamic system where the parameter
67 switchings in the state equation and in the measurement equation are not synchronous. The

68 switching in the state equation remains at the segmental level, but that in the measurement
69 equation becomes instead at the frame level. Making the switching of mixture (i.e., multiple)
70 components in linear mapping functions of the measurement equation at the frame level has
71 advantages over the previous segment-level switching described in Ma and Deng (2003) as fol-
72 lows. For the frame-level switching, the multiple linear mappings become a piecewise linear ap-
73 proximation, at each frame, to the nonlinear function that defines the true mapping between the
74 VTR and observation spaces. At different frames, a different “piece” of the linear approximation
75 function may be optimally selected. This greatly increases the flexibility of the model in ap-
76 proximating the original complex nonlinear function. In contrast, for the segment-level switching
77 as developed in Ma and Deng (2003), only one linear approximation is made to the nonlinear
78 function for a given segment (consisting of many frames), although there is an inventory (mixture)
79 of possible linear functions available for each segment.

80 The organization of this paper is as follows. In Section 2, a description of the mixed-level
81 switching dynamic system model is provided, including its complete parameterization. In Section
82 3, an Expectation-Maximization-based algorithm is presented for parameter estimation of this
83 model. Some technical details will be referred to the earlier model of Ma and Deng (2003) with
84 single-level switching, which shares some similar steps of the algorithm derivation, and key al-
85 gorithm changes will be summarized. In Section 4, we report speech recognition experiments,
86 aimed to evaluate the new mixed-level switching model on the Switchboard database under the N-
87 best list re-scoring paradigm.

88 2. Model formulation

89 The mixed-level, switching dynamic system model developed in this study is a linear combi-
90 nation of standard linear dynamic models (a total of M). The m th linear dynamic model with the
91 first-level, segmental switching has the following form:

$$Z(k) = \Phi_m Z(k-1) + (I - \Phi_m) T_m + W_m(k-1), \quad (1)$$

$$O(k) = \dot{H}_m^{(l)}(k) \dot{Z}(k) + V_m(k), \quad (2)$$

94 where $\dot{H}_m^{(l)} = [a_m^{(l)}, H_m^{(l)}]$ and $\dot{Z}(k) = [1, Z(k)]'$. The state equation, Eq. (1), for each phone segment
95 is associated with M sets of parameters (Φ_m, T_m, Q_m) , but they are not switched from one set to
96 another until at the end of the current segment (hence segment-level switching). That is, the
97 parameter switching takes place at the boundary between two adjacent phones. In the measure-
98 ment equation, Eq. (2), however, the parameters $\dot{H}_m^{(l)}(k)$ are allowed to switch at each time frame
99 k . This gives rise to the second-level switching at the frame level. This frame-level switching is
100 constrained to take values from fixed finite sets: $\{\dot{H}_m^{(l)}, l' = 1, 2, \dots, L\}$ for each frame. Note that
101 the subscript m in Eq. (2) indicates that different segment-level mixture components correspond to
102 different sets of $\dot{H}^{(l)}$ values. Also note that when $L = 1$, the above model is reduced to the single-
103 level switching model described in Ma and Deng (2003).

104 To summarize, the complete set of model parameters consists of

$$\Theta = \{\pi_m, \Phi_m, T_m, Q_m, R_m, \dot{H}_m^l, \gamma_{m,l}, m = 1, 2, \dots, M, l = 1, 2, \dots, L\},$$

106 where π_m represents the mixture weight probability $P(m|\Theta)$ and $\gamma_{m,l}$ represents the \dot{H}_m^l -value weight
107 probability $P(l|m, \Theta)$. As discussed before, two levels of parameter switching have been designed.
108 First, the mixture component indexed by m switches at the segment (phone) level. Second, the \dot{H}_m^l
109 value indexed by l switches at the frame level. The first level of switching corresponds to the target
110 property of the VTR dynamics, which therefore is reasonably placed at the segment level. The
111 second level of parameter switching is designed to provide the flexibility for using multiple linear
112 regression functions to approximate the nonlinear relationship between the VTR and the mea-
113 surement variables. It is desirable that the parameter switching happens at the frame level because
114 the relationship approximated by the linear mapping can change at different frames. In this paper,
115 we also call the former switching *mixture switching* and the later one *H-value switching*. As will be
116 seen in Section 5, use of the additional *H-value switching* gives improved speech recognition
117 performance over the model of Ma and Deng (2003) which has the mixture switching only.

118 In order to implement the mixture switching, it is necessary to impose the *mixture-path con-*
119 *straint*, where for each sequence of the acoustic observation associated with a phone, the obser-
120 vation is restricted to be produced from a fixed mixture component, m , of the model. This means
121 that the target of the VTR in a phone is not permitted to switch from one mixture component to
122 another at the frame level.¹ The constraint of such a type is motivated by the physical nature of
123 the speech model – the target that is correlated with the phonetic identity is defined at the segment
124 (phone) level, not at the frame level. This constraint is imposed both the model training and on the
125 model scoring in implementing the speech recognizer. For the *H-value switching*, no such con-
126 straints are in place.

127 3. Learning model parameters

128 Due to the unobserved nature of the state in the model presented in the above section, Ex-
129 pectation-Maximization (EM) algorithm has been developed for model parameter estimation
130 (Dempster et al., 1977; Deng, 1993; Digalakis et al., 1993; Ostendorf et al., 1996). The approach
131 we are taking here has been inspired by that in Streit and Luginbuhl (1998), with substantial
132 modifications to suit our specific mixed-level, switching dynamic model of speech.

133 Before formally describing the EM algorithm, we first define a discrete random variable X ,
134 which provides the observation-to-mixture assignment for a sequence of observations. For ex-
135 ample, for a given sequence of observations of a phone, when $X = m$, ($1 \leq m \leq M$), it means that
136 the m th mixture component of the model is responsible for generating that observation sequence.
137 We need one additional discrete variable to represent the *H-value switching* on the measurement
138 equation. This is denoted by $Y = \{y_1, y_2, \dots, y_K\}$ (K is the length of the observation), where
139 y_k ($1 \leq k \leq K$) is a discrete random variable indicating which one of the \dot{H}_m^l 's ($1 \leq l \leq L$) is switched
140 onto at time frame k . For example, when $y_k = i$, it means the i th value, \dot{H}_m^i , is chosen at time k . (We
141 assume that y_1, y_2, \dots, y_K are independent random variables.) Finally, we define a discrete-variable
142 pair, $S = \{X, Y\}$, which represents the combination of X and Y . Note that in the single-level
143 switching model of Ma and Deng (2003), only one discrete random variable X was introduced.

¹ This same mixture-path constraint has been imposed earlier on the mixture-trended HMM; see Deng and Aksmanovic (1997).

144 In order to impose the mixture-path constraint discussed earlier, the joint variable pair is de-
145 fined at the segment level. Suppose we have N training tokens for a phone. We in this case define
146 the joint variable as

$$\{O, Z, S\}^N = \{(O^1, Z^1, S^1), (O^2, Z^2, S^2), \dots, (O^n, Z^n, S^n), \dots, (O^N, Z^N, S^N)\},$$

148 where $O^n = \{O^n(1), O^n(2), \dots, O^n(K_n)\}$ is the n th observation, Z^n is the corresponding hidden state
149 sequence, and $S^n = \{X^n, Y^n\}$ denotes the corresponding observation-to-mixture assignment by X^n
150 as well as the switching behavior of the measurement equation represented by

$$Y^n = \{y_1^n, y_2^n, \dots, y_{K_n}^n\}.$$

152 Here, X^n and Y^n are combined to jointly determine how the observation sequences are generated.

153 The following assumptions are made in the development of the learning algorithm:

- 154 • The N tokens are independent of each other. That is, S^n ($1 \leq n \leq N$) are independent of each
155 other.
- 156 • The random variables X^n , ($1 \leq n \leq N$) have identical (discrete) distributions.
- 157 • y_1^n, y_2^n, \dots , and $y_{K_n}^n$ are independent of each other. That is, the parameter switching does not de-
158 pend on the history, nor on the future.
- 159 • y_k^n ($1 \leq k \leq K_n$) have identical (discrete) distributions.

160 With these assumptions, the PDF of the joint variable $\{O, Z, S\}^N$, with the fixed model pa-
161 rameter set Θ , can be decomposed into

$$\begin{aligned} & p(\{O, Z, S\}^N | \Theta) \\ &= \prod_{n=1}^N \left\{ p(Z_0^n | \Theta) \left[\prod_{k=1}^{K_n} p(Z_k^n | Z_{k-1}^n, X^n, \Theta) p(O_k^n | Z_k^n, X^n, y_k^n, \Theta) P(y_k^n | X^n, \Theta) \right] P(X^n | \Theta) \right\}. \end{aligned} \quad (3)$$

163 This result is generalization of a corresponding result for the single-level switching model pre-
164 sented and derived in Ma and Deng (2003).

165 Further, to compute the auxiliary Q -function in the E-step of the EM algorithm, we need
166 several other PDFs.

167 First, we have

$$p(\{O, S\}^N | \Theta) = \prod_{n=1}^N \left[\prod_{k=1}^{K_n} p(O^n(k) | O_{1,k-1}^n, X^n, y_k^n, \Theta) P(y_k^n | X^n, \Theta) \right] P(X^n | \Theta). \quad (4)$$

169 Second, we decompose $p(\{O, X\}^N | \Theta)$ into

$$\begin{aligned} & p(\{O, X\}^N | \Theta) = \sum_{\{Y\}^N} p(\{O, S\}^N | \Theta) \\ &= \sum_{Y^1} \sum_{Y^2} \dots \sum_{Y^N} \prod_{n=1}^N \left[\prod_{k=1}^{K_n} p(O^n(k) | O_{1,k-1}^n, X^n, y_k^n, \Theta) P(y_k^n | X^n, \Theta) \right] P(X^n | \Theta) \\ &= \prod_{n=1}^N \left[\prod_{k=1}^{K_n} \sum_{y_k^n=1}^L p(O^n(k) | O_{1,k-1}^n, X^n, y_k^n, \Theta) P(y_k^n | X^n, \Theta) \right] P(X^n | \Theta), \end{aligned} \quad (5)$$

171 where independence between tokens and between switchings at different times has been used.

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172 Third, $p(\{O\}^N|\Theta)$ is computed as

$$\begin{aligned} p(\{O\}^N|\Theta) &= \sum_{\{X\}^N} p(\{O, X\}^N|\Theta) \\ &= \prod_{n=1}^N \sum_{X^n=1}^M \left[\prod_{k=1}^{K_n} \sum_{y_k^n=1}^L p(O^n(k)|O_{1,k-1}^n, X^n, y_k^n, \Theta) P(y_k^n|X^n, \Theta) \right] P(X^n|\Theta). \end{aligned} \quad (6)$$

174 Then, the conditional PDF $p(\{X\}^N|\{O\}^N, \Theta)$ can be derived to be

$$p(\{X\}^N|\{O\}^N, \Theta) = \frac{p(\{O, X\}^N|\Theta)}{p(\{O\}^N|\Theta)} = \prod_{n=1}^N \omega_m^n, \quad (7)$$

176 where

$$\omega_m^n = \frac{\left[\prod_{k=1}^{K_n} \sum_{l=1}^L p(O^n(k)|O_{1,k-1}^n, m, l, \Theta) P(l|m, \Theta) \right] P(m|\Theta)}{\sum_{m=1}^M \left[\prod_{k=1}^{K_n} \sum_{l=1}^L p(O^n(k)|O_{1,k-1}^n, m, l, \Theta) P(l|m, \Theta) \right] P(m|\Theta)}. \quad (8)$$

178 In the above, because X^n s have identical distributions, they are replaced by a common variable m
179 for notational simplicity. For the same reason, y_k^n is replaced by the common variable l .

180 Finally, we compute the conditional PDF of

$$p(\{Y\}^N|\{X\}^N, \{O\}^N, \Theta) = \frac{p(\{O, S\}^N|\Theta)}{p(\{O, X\}^N|\Theta)} = \prod_{n=1}^N \prod_{k=1}^{K_n} \zeta_{k,m,l}^n, \quad (9)$$

182 where

$$\zeta_{k,m,l}^n = \frac{p(O^n(k)|O_{1,k-1}^n, m, l, \Theta) P(l|m, \Theta)}{\sum_{l=1}^L p(O^n(k)|O_{1,k-1}^n, m, l, \Theta) P(l|m, \Theta)}. \quad (10)$$

184 Using the independence assumption among tokens, we obtain

$$p(Y^n|X^n, O^n, \Theta) = \prod_{k=1}^{K_n} \zeta_{k,m,l}^n. \quad (11)$$

186 The independence assumption among switchings at different times further gives

$$p(y_k^n|X^n, O^n, \Theta) = \zeta_{k,m,l}^n. \quad (12)$$

188 *3.1. EM algorithm: E-step*

189 Given all the joint PDF and conditional PDF computations discussed above, we describe the
190 EM algorithm. Since both $\{Z\}^N$ and $\{S\}^N$ are missing data, we compute integration over both of
191 them to obtain the Q -function:

$$Q(\Theta|\bar{\Theta}) = \sum_{\{S\}^N} \int \log p(\{O, Z, S\}^N|\Theta) \cdot p(\{Z\}^N|\{O, S\}^N, \bar{\Theta}) d\{Z\}^N p(\{S\}^N|\{O\}^N, \bar{\Theta}), \quad (13)$$

193 where $\bar{\Theta}$ denotes the parameter set at the immediately previous step of the EM algorithm.

194 Generalizing some derivation steps in Ma and Deng (2003), we can show that the auxiliary
195 function consists of three parts:

$$Q(\Theta|\bar{\Theta}) = Q_Z + Q_Y + Q_X. \quad (14)$$

197 Also, following the same steps as in Ma and Deng (2003), these three terms can be simplified to
198 (as before, we use m and l to denote X^n and y_k^n , respectively):

$$Q_Z = \sum_{n=1}^N \sum_{m=1}^M \int \left[\sum_{k=1}^{K_n} \log p(Z_k^n | Z_{k-1}^n, m, \Theta) \right] p(Z^n | O^n, m, \bar{\Theta}) dZ^n \cdot \bar{\omega}_m^n \\ + \sum_{n=1}^N \sum_{m=1}^M \int \left\{ \sum_{k=1}^{K_n} \sum_{l=1}^L \log p(O_k^n | Z_k^n, m, l, \Theta) \cdot \bar{\xi}_{k,m,l}^n \cdot p(Z^n | O^n, m, l, \bar{\Theta}) \right\} dZ^n \cdot \bar{\omega}_m^n, \quad (15)$$

$$Q_Y = \sum_{n=1}^N \sum_{m=1}^M \left[\sum_{k=1}^{K_n} \sum_{l=1}^L \log P(l|m, \Theta) \cdot \bar{\xi}_{k,m,l}^n \right] \bar{\omega}_m^n, \quad (16)$$

$$Q_X = \sum_{n=1}^N \sum_{m=1}^M \log P(m|\Theta) \bar{\omega}_m^n. \quad (17)$$

202 In the above the symbols $\bar{\omega}_m^n$ and $\bar{\xi}_{k,m,l}^n$ denote the corresponding variables of ω_m^n and $\xi_{k,m,l}^n$ (Eqs. (8)
203 and (10), respectively) for the preceding EM iteration.

204 By the definition of the model in Eqs. (1) and (2), $p(Z_k^n | Z_{k-1}^n, m, \Theta)$ is a Gaussian with mean:
205 $\Phi_m Z^n(k-1) + (I - \Phi_m) T_m$ and with covariance: Q_m . And $p(O_k^n | Z_k^n, m, l, \Theta)$ is also a Gaussian with
206 mean: $\dot{H}_m^l Z^n(k)$ and with covariance: R_m . Therefore, Q_Z can be re-written as

$$Q_Z = -\frac{1}{2} \sum_{n=1}^N \sum_{m=1}^M \left\{ K_n \log |Q_m| + \sum_{k=1}^{K_n} E_m \left[e1_{k,m}^n {}' (Q_m)^{-1} e1_{k,m}^n \right] \right\} \cdot \bar{\omega}_m^n \\ - \frac{1}{2} \sum_{n=1}^N \sum_{m=1}^M \left\{ K_n \log |R_m| + \sum_{k=1}^{K_n} \sum_{l=1}^L E_{ml} \left[e2_{k,m,l}^n {}' (R_m)^{-1} e2_{k,m,l}^n \cdot \bar{\xi}_{k,m,l}^n \right] \right\} \cdot \bar{\omega}_m^n + \text{const.}, \quad (18)$$

208 where $e1_{k,m}^n$ and $e2_{k,m,l}^n$ are defined as

$$e1_{k,m}^n = Z^n(k) - \Phi_m Z^n(k-1) - (I - \Phi_m) T_m,$$

$$e2_{k,m,l}^n = O^n(k) - \dot{H}_m^l Z^n(k).$$

211 In Eq. (18), $E_m[\cdot]$ denotes the conditional expectation $E[\cdot | O^n, m, \bar{\Theta}]$ and $E_{ml}[\cdot]$ denotes the con-
212 ditional expectation $E[\cdot | O^n, m, l, \bar{\Theta}]$. These conditional expectations, $E_m[\cdot]$ and $E_{ml}[\cdot]$, are com-
213 puted from the Kalman smoothing algorithm that will be discussed in detail later.

214 3.2. EM algorithm: M-step

215 With the Q -function computed above, we now go to the M-step of the EM-algorithm.

216 *Estimate for π_m .* The final form of the reestimate for π_m is

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$$\hat{\pi}_m = \frac{\sum_{n=1}^N \bar{\omega}_m^n}{\sum_{n=1}^N \sum_{m=1}^M \bar{\omega}_m^n} \quad \text{for } 1 \leq m \leq M. \quad (19)$$

218 This is identical to the estimate for the single-level switching model of Ma and Deng (2003), where
219 a derivation was provided.

220 *Estimate for $\gamma_{m,l}$.* Note in the Q -function, only Q_Y is related to the “switching” probability
221 $\gamma_{m,l} = P(l|m, \Theta)$. With the constraint, $\sum_{l=1}^L \gamma_{m,l} = 1$, the Lagrangian is

$$L_Y = Q_Y + \lambda(1 - \sum_l \gamma_{m,l}).$$

223 Taking the derivative of L_Y with respect to $\gamma_{m,l}$ and set it to zero, we obtain

$$\frac{\partial L_Y}{\partial \gamma_{m,l}} = \sum_{n=1}^N \left[\sum_{k=1}^{K_n} \frac{\bar{\xi}_{k,m,l}^n}{\gamma_{m,l}} \right] \bar{\omega}_m^n - \lambda.$$

225 Solving the above for $\gamma_{m,l}$ and normalizing with the Lagrangian finally give the final form of the
226 re-estimation formula

$$\hat{\gamma}_{m,l} = \frac{\sum_{n=1}^N [\sum_{k=1}^{K_n} \bar{\xi}_{k,m,l}^n] \bar{\omega}_m^n}{\sum_{n=1}^N K_n \bar{\omega}_m^n}. \quad (20)$$

228 Note that the single-level switching model of Ma and Deng (2003) did not have the parameter
229 $\gamma_{m,l}$ to estimate.

230 *Estimates for Φ_m , T_m and Q_m .* Note that in the Q -function only the first term of Q_Z is related to
231 these parameters in the state equation. We first introduce the following notations:

$$\begin{aligned} A0_m^n &= \sum_{k=1}^{K_n} E_m[Z^n(k-1)Z^n(k-1)'], & A1_m^n &= \sum_{k=1}^{K_n} E_m[Z^n(k)Z^n(k)'], \\ A2_m^n &= \sum_{k=1}^{K_n} E_m[Z^n(k)Z^n(k-1)'], & C_m &= (I - \hat{\Phi}_m)\hat{T}_m, \\ B0_m^n &= \sum_{k=1}^{K_n} E_m[Z^n(k-1)], & B1_m^n &= \sum_{k=1}^{K_n} E_m[Z^n(k)]. \end{aligned}$$

233 Then the final estimation formulas for these parameters are

$$\hat{T}_m = \frac{(I - \hat{\Phi}_m)^{-1} \sum_{n=1}^N \{B1_m^n - \hat{\Phi}_m B0_m^n\} \cdot \bar{\omega}_m^n}{\sum_{n=1}^N K_n \cdot \bar{\omega}_m^n}, \quad (21)$$

$$\hat{\Phi}_m = \left\{ \sum_{n=1}^N (A2_m^n - B1_m^n \hat{T}_m' - \hat{T}_m B0_m^{n'} + K_n \hat{T}_m \hat{T}_m') \cdot \bar{\omega}_m^n \right\} \cdot \left\{ \sum_{n=1}^N (A0_m^n - B0_m^n \hat{T}_m' - \hat{T}_m B0_m^{n'} + K_n \hat{T}_m \hat{T}_m') \cdot \bar{\omega}_m^n \right\}^{-1}, \quad (22)$$

$$\hat{Q}_m = \frac{\sum_{n=1}^N \sum_{k=1}^{K_n} E_m[e1_{k,m}^n e1_{k,m}^{n'}] \cdot \bar{\omega}_m^n}{\sum_{n=1}^N K_n \bar{\omega}_m^n}. \quad (23)$$

237 Again, these results are identical to the estimates for the single-level switching model of Ma and
238 Deng (2003), since these estimated parameters are in the state equation and do not subject to the
239 frame-level switching.

240 *Estimates for \hat{H}_m^l and R_m .* Finally, we derive the re-estimation formulas for the parameters, $\hat{H}_{m,l}$
241 and R_m , contained in the measurement equation. In the Q -function, only the second term in Q_Z is
242 related to these parameters. Due to the frame-level switching involved in the measurement
243 equation, the results presented below are different from the corresponding estimation formulas
244 described in Ma and Deng (2003).

245 The derivative of Q_Z with respect to \hat{H}_m^l is

$$\frac{\partial Q_Z}{\partial \hat{H}_m^l} = -R_m^{-1} \left\{ \sum_{n=1}^N \left(\sum_{k=1}^{K_n} E_{ml}[(\hat{H}_m^l \dot{Z}^n(k) - O_k^n)(\dot{Z}^n(k))'] \bar{\xi}_{k,m,l}^n \right) \bar{\omega}_m^n \right\}. \quad (24)$$

247 Setting the derivative to zero, we obtain

$$\widehat{H}_m^l = \left\{ \sum_{n=1}^N \bar{\omega}_m^n \sum_{k=1}^{K_n} O^n(k) E_{ml}[\dot{Z}^n(k)]' \cdot \bar{\xi}_{k,m,l}^n \right\} \left\{ \sum_{n=1}^N \bar{\omega}_m^n \sum_{k=1}^{K_n} E_{ml}[\dot{Z}^n(k)(\dot{Z}^n(k))'] \cdot \bar{\xi}_{k,m,l}^n \right\}^{-1}, \quad (25)$$

249 where $E_{ml}[\dot{Z}^n(k)] = [1, E_{ml}[Z^n(k)]']'$ and

$$E_{ml}[\dot{Z}^n(k)(\dot{Z}^n(k))'] = \begin{bmatrix} 1 & E_{ml}[Z^n(k)]' \\ E_{ml}[Z^n(k)] & E_{ml}[Z^n(k)(Z^n(k))']' \end{bmatrix}.$$

251 The derivative of Q_Z with respect to R_m^{-1} is

$$\frac{\partial Q_Z}{\partial R_m^{-1}} = \frac{1}{2} \sum_{n=1}^N \left\{ \sum_{k=1}^{K_n} \sum_{l=1}^L (R_m - E_{ml}[e2_{k,m,l}^n (e2_{k,m,l}^n)']) \cdot \bar{\xi}_{k,m,l}^n \right\} \cdot \bar{\omega}_m^n. \quad (26)$$

253 Setting this to zero, we obtain the estimate for R_m :

$$\hat{R}_m = \frac{\sum_{n=1}^N \left(\sum_{k=1}^{K_n} \sum_{l=1}^L E_{ml}[e2_{k,m,l}^n (e2_{k,m,l}^n)'] \cdot \bar{\xi}_{k,m,l}^n \right) \cdot \bar{\omega}_m^n}{\sum_{n=1}^N K_n \bar{\omega}_m^n}, \quad (27)$$

255 where $E_{ml}[e2_{k,m,l}^n (e2_{k,m,l}^n)']$ is calculated according to

$$E_{ml}[e2_{k,m,l}^n (e2_{k,m,l}^n)'] = O^n(k) O^n(k)' - E_{ml}[\dot{Z}^n(k)] (\widehat{H}_m^l)' \\ - \widehat{H}_m^l E_{ml}[\dot{Z}^n(k)] + \widehat{H}_m^l E_{ml}[\dot{Z}^n(k) \dot{Z}^n(k)'] (\widehat{H}_m^l)'. \quad (28)$$

258 3.3. Sufficient statistics computed by Kalman smoother

259 As we showed earlier, in order to obtain the re-estimates for the model parameters, a set of
260 conditional expectations, which form the sufficient statistics for the estimation problem, need to

261 be calculated during the M-step of the EM algorithm. These sufficient statistics include $E_m[Z^n(k)]$,
262 $E_m[Z^n(k)Z^n(k)']$, $E_m[Z^n(k)Z^n(k-1)']$, $E_{ml}[Z^n(k)]$ and $E_{ml}[Z^n(k)Z^n(k)']$.

263 The conditional expectation $E_m[\cdot] = E[\cdot | O^n, m, \bar{\Theta}]$ is the Kalman smoother of the m th
264 mixture component for the n th observation. However, the conventional Kalman smoother
265 can not be directly applied here because the current model has parameter switching occurring
266 in the measurement equation. This situation is similar to that presented in Shumway and
267 Stoffer (1991), where the filtering and smoothing algorithms were derived. Based on the
268 solution given in Shumway and Stoffer (1991), the conditional expectation $E_{ml}[\cdot] =$
269 $E[\cdot | O^n, m, l, \bar{\Theta}]$ in our problem becomes the smoother of the m th mixture component for the n th
270 observation under an extra condition of $\dot{H}_m(k) = \dot{H}_m^l$ (recall that we use l to represent
271 $y_k^n = l$).

272 The basic theory of Kalman filtering and smoothing can be found in Tanizaki (1996), Kitagawa
273 (1987), Mendel (1995), etc. In the following we list the filtering and smoothing algorithms for the
274 new mixed-level switching model. These algorithms generalize the ones presented in Ma and Deng
275 (2003), which became the special cases when the number (L) of frame-level or H -value switching
276 possibilities is reduced to one.

277 *Forward recursion (or filtering):*

$$\hat{Z}_{k|k-1,m}^n = \Phi_m \hat{Z}_{k-1|k-1,m}^n + (I - \Phi_m) T_m, \quad (29)$$

$$\Sigma_{k|k-1,m}^n = \Phi_m \Sigma_{k-1|k-1,m}^n \Phi_m + Q_m, \quad (30)$$

$$\tilde{O}_{k,m,l}^n = O^n(k) - \dot{H}_m^l \hat{Z}_{k|k-1,m}^n, \quad l = 1, 2, \dots, L, \quad (31)$$

$$\Sigma_{\tilde{O}_{k,m,l}^n}^n = H_m^l \Sigma_{k|k-1,m}^n H_m^{l'} + R_m, \quad (32)$$

$$K_{k,m,l} = \Sigma_{k|k-1,m}^n H_m^{l'} \left(\Sigma_{\tilde{O}_{k,m,l}^n}^n \right)^{-1}, \quad (33)$$

$$\hat{Z}_{k|k,m,l}^n = \hat{Z}_{k|k-1,m}^n + K_{k,m,l} \tilde{O}_{k,m,l}^n, \quad (34)$$

$$\Sigma_{k|k,m,l}^n = \Sigma_{k|k-1,m}^n - K_{k,m,l} \Sigma_{\tilde{O}_{k,m,l}^n}^n K_{k,m,l}', \quad (35)$$

$$\psi_{k,m,l} = \frac{\gamma_{m,l} \cdot \mathcal{N} \left(O^n(k); \dot{H}_m^l \hat{Z}_{k|k-1,m}^n, \Sigma_{\tilde{O}_{k,m,l}^n}^n \right)}{\sum_{l=1}^L \gamma_{m,l} \cdot \mathcal{N} \left(O^n(k); \dot{H}_m^l \hat{Z}_{k|k-1,m}^n, \Sigma_{\tilde{O}_{k,m,l}^n}^n \right)}, \quad (36)$$

$$\hat{Z}_{k|k,m}^n = \sum_{l=1}^L \psi_{k,m,l} \cdot \hat{Z}_{k|k,m,l}^n, \quad (37)$$

$$\Sigma_{k|k,m}^n = \sum_{l=1}^L \psi_{k,m,l} \cdot \Sigma_{k|k,m,l}^n, \quad (38)$$

288 where $\hat{Z}_{k|k-1,m}^n$ is the predictor and $\Sigma_{k|k-1,m}^n$ its error covariance. $\hat{Z}_{k|k,m}^n$ is the filter and $\Sigma_{k|k,m}^n$ its error
289 covariance. $\mathcal{N}(O^n(k); \hat{H}_m^l \hat{Z}_{k|k-1,m}^n, \Sigma_{\hat{O}_{k,m,l}}^n)$ is a Gaussian density with mean $\hat{H}_m^l \hat{Z}_{k|k-1,m}^n$ and covariance
290 $\Sigma_{\hat{O}_{k,m,l}}^n$. This is the density of the innovation sequence at time k .

291 *Backward recursion (or smoothing):*

$$A_{k,m}^n = \Sigma_{k|k,m}^n \Phi_m' (\Sigma_{k|k-1,m}^n)^{-1}, \quad (39)$$

$$\hat{Z}_{k|K_n,m}^n = \hat{Z}_{k|k,m}^n + A_{k,m}^n [\hat{Z}_{k+1|K_n}^n - \hat{Z}_{k+1|k,m}^n], \quad (40)$$

$$\Sigma_{k|K_n,m}^n = \Sigma_{k|k,m}^n + A_{k,m}^n [\Sigma_{k+1|K_n,m}^n - \Sigma_{k+1|k,m}^n] A_k'. \quad (41)$$

295 Note that the above smoothing is for the computation of $E[\cdot | O^n, m, \bar{\Theta}]$. For the computation of
296 $E[\cdot | O^n, m, l, \bar{\Theta}]$ at the given time point k , $\hat{Z}_{k|k,m}^n$ and $\Sigma_{k|k,m}^n$ in the above smoothing algorithm are
297 simply replaced by $\hat{Z}_{k|k,m,l}^n$ and $\Sigma_{k|k,m,l}^n$, respectively, and correspondingly, $\hat{Z}_{k|K_n,m}^n$ becomes $\hat{Z}_{k|K_n,m,l}^n$
298 and $\Sigma_{k|K_n,m}^n$ becomes $\Sigma_{k|K_n,m,l}^n$. At other points, smoothing remains unchanged.

299 Using the above Kalman smoothing results, the conditional expectations as sufficient statistics
300 are computed by

$$E_m[Z^n(k)] = \hat{Z}_{k|K_n,m}^n, \quad (42)$$

$$E_m[Z^n(k)Z^n(k)'] = \Sigma_{k|K_n,m}^n + \hat{Z}_{k|K_n,m}^n (\hat{Z}_{k|K_n,m}^n)', \quad (43)$$

$$E_m[Z^n(k)Z^n(k-1)'] = \Sigma_{k,k-1|K_n,m}^n + \hat{Z}_{k|K_n,m}^n (\hat{Z}_{k-1|K_n,m}^n)', \quad (44)$$

$$E_{ml}[Z^n(k)] = \hat{Z}_{k|K_n,m,l}^n, \quad (45)$$

$$E_{ml}[Z^n(k)Z^n(k)'] = \Sigma_{k|K_n,m,l}^n + \hat{Z}_{k|K_n,m,l}^n (\hat{Z}_{k|K_n,m,l}^n)', \quad (46)$$

306 where $\Sigma_{k,k-1|K_n,m}^n$ is recursively calculated by Shumway (1982)

$$\Sigma_{k,k-1|K_n,m}^n = \Sigma_{k|k,m}^n A_{k-1,m}^{n'} + A_{k,m}^n (\Sigma_{k+1,k|K_n,m}^n - \Phi_m \Sigma_{k|k,m}^n) A_{k-1,m}^{n'} \quad (47)$$

308 for $k = K_n, \dots, 2$, where the end point is

$$\Sigma_{K_n, K_n-1|K_n,m}^n = \sum_{l=1}^L \psi_{K_n,m,l} (I - K_{K_n,m,l} H_m^l) \Phi_m \Sigma_{K_n-1|K_n-1,m,l}^n. \quad (48)$$

310 3.4. Computation of ω and ξ

311 To compute ω_m^n and $\xi_{k,m,l}^n$ according to Eqs. (8) and (10), respectively, it suffices to compute
312 $p(O_k^n | O_{1,k-1}^n, m, l, \bar{\Theta})$. This is actually the PDF of the innovation sequence, and hence it is com-
313 puted straightforwardly by

$$p(O_k^n | O_{1,k-1}^n, m, l, \bar{\Theta}) = (2\pi)^{-d/2} \left| \Sigma_{\hat{O}_{k,m,l}}^n \right|^{-1/2} \exp \left\{ -\frac{1}{2} (\tilde{O}_{k,m,l}^n)' [\Sigma_{\hat{O}_{k,m,l}}^n]^{-1} \tilde{O}_{k,m,l}^n \right\}. \quad (49)$$

315 4. Likelihood computation

316 Efficient computation of the likelihood of a sequence of observation vectors using the model is
317 a key requirement ² for implementing the speech recognizer (in the recognizer testing phase). Such
318 computation is discussed in this section.

319 We combine M different linear dynamic models (mixture models) using different weights to
320 represent a phone's VTR dynamics. After the weights and all model parameters are trained as
321 described in the preceding section, the likelihood of a phone for a sequence of observation vectors
322 is computed by

$$\begin{aligned}
 l(O|\Theta) &= \sum_S p(O, S|\Theta) = \sum_X \sum_Y p(O|Y, X, \Theta)p(Y|X, \Theta)p(X|\Theta) \\
 &= \sum_{m=1}^M \left\{ \prod_{k=1}^K \sum_{l=1}^L p(O_k^n | O_{1,k-1}^n, m, l, \Theta) \cdot \gamma_{m,l} \right\} \cdot \pi_m. \quad (50)
 \end{aligned}$$

324 Note that when $L = 1$ (and hence $\gamma_{m,l} = 1$), Eq. (50) is reduced to the corresponding likelihood
325 computation formula in Ma and Deng (2003).

326 5. Speech recognition experiments

327 In this section, we first introduce the experimental paradigm and design of the new recognizer
328 built on the dynamic speech model presented so far. We then report the evaluation results of the
329 new recognizer on the Switchboard data. The evaluation is conducted with reference to the
330 conventional triphone HMM recognizer under the identical experimental conditions.

331 5.1. Experimental paradigm and recognizer design

332 In all the experiments reported in this section, we use the N-best re-scoring paradigm to
333 evaluate the new recognizer on the Switchboard spontaneous speech data. The N-best lists
334 (transcription hypotheses) and their phone-level segmentation (or alignment) are obtained from a
335 conventional triphone-based HMM system, which also serves as the benchmark to gauge the
336 recognizer's performance improvement via use of the new speech model. The benchmark HMM
337 system has been described in detail in Bridle et al. (1998), Picone et al. (1999) and will not be
338 described in this paper.

339 We built the switching dynamic system models for a total of 44 distinct phone-like symbols,
340 including eight context-dependent phones. The phone-like symbol inventory are listed here:

341 Context-independent phones: aa ae ah eh ao ax er ih iy uh uw l el r w y en n s z sh zh th dh d t
342 sil sp

343 Context-dependent phones: f v b g p k m ng _f_ _v_ _b_ _g_ _p_ _k_ _m_ _ng_

² In fact, this is the only requirement if the N-best re-scoring scheme is used to evaluate the recognizer, as is reported in this paper.

344 The VTR targets of the above eight context-dependent phones are affected by the anticipatory
345 tongue position associated with the following phone. The targets of those phones with subscript
346 “_” are conditioned on the following phones being “front” vowels (iy, ih, eh, ae and y) and their
347 correspondences without subscript “_” conditioned on the following phones being the “non-
348 front” phones.

349 Physically, silence (sil) and short pause (sp) have no VTR targets. In our experiments, we as-
350 signed pseudo-targets to them so that they can be distinguished from other phones.

351 5.2. Baseline system

352 An HMM triphone system was trained on the “train-ws97-a” Switchboard training data (about
353 160 h). It served as the baseline system for the Workshop’97, and hence we call it the “ws97-
354 baseline” system. The performance of this HMM baseline system is listed as row 3 in Table 1,
355 where “Ref + 100” and “100 best” mean 100-best hypotheses with and without reference included,
356 respectively. We also add the “Oracle” and “By chance” performance into Table 1 to calibrate the
357 recognizer’s performance. The “Oracle” WER is calculated by always choosing the best one
358 hypothesis and the “By chance” WER is computed by randomly selecting one out of all hy-
359 potheses.

360 5.3. Training and test sets

361 In our experiments, we used several sets of training data with an increasing size to train the
362 parameters of the switching dynamic system models of speech. The smallest set of training data
363 consist of one male speaker’s data (speaker ID: 1028) extracted from the Switchboard training set
364 “train-ws97-a”. It contains several telephone conversations with a total of 30 min long. Due to the
365 use of training data from only one speaker, we avoid the speaker normalization problem for both
366 the VTR targets and for the MFCC observation. We name this set as “1/2 hour” training set to be
367 used in describing the experimental results in the remaining of this section.

368 An HMM system has also been trained on this “1/2 hour” training set, which we call the
369 “HMM-baseline” system. Its performance, measured by the percentage word error rate (WER) is
370 listed in row 5 of Table 1.

371 To investigate how the amount of training data and the number of mixture components affect
372 the recognizer performance, we also extracted multiple speakers’ data from the same “train-ws97-
373 a” training set to train the speech models. we first added another half an hour data to the original
374 “1/2 hour” training set, where the new data came from 30 different speakers. This increased
375 dataset is called “1 hour” training set. we then added one more hour of training data to the “1

Table 1
Performance (WER) of baseline HMM system

Systems	Ref + 100	100 Best
Oracle	0.0	32.5
By chance	59.6	60.2
ws97-baseline	56.2	56.9

hour” training set. This additional one hour of data came from 50 different speakers. This gives a new, expanded “2 hour” training set.

The test data in all of our experiments consist of all the male speakers from the WS’97 DevTest set. They comprise a total of 23 male speakers, 24 conversations, 1243 utterances, 9970 words, and 50 min of speech. All the 100-best hypotheses for each of the 1243 utterances were generated by the “ws97-baseline” HMM system (Bridle et al., 1998; Picone et al., 1999). All the experiments reported in this paper have used the VTR dynamic regimes derived sub-optimally from the phone alignments provided by this baseline HMM system.

5.4. Experiment I: models trained with one speaker’s data

In these experiments, the “1/2 hour” training set is used for model training. First, we use a single mixture component (1-mix) for the mixture-linear dynamic model (MLDM) with switching parameters and we increase the number of H -switching values from one to three. The percentage-WER results are tabulated in Table 2. It is observed that use of H -switching (two or three H values) reduces errors compared with no use of H -switching (one H value only).

We then use two mixture components (2-mix) while again gradually increasing the number of H -switching values. The WER results are listed in Table 3. We observe a similar pattern of error reduction to the previous experiment while uniformly raising the overall recognizer performance level somewhat.

Under identical conditions, compared with the “HMM-baseline” system trained on the same data, the new switching dynamic system model with two mixture components and two H switching values achieves 2.3% absolute WER reduction on the “100-best” case and more than 10% relative WER reduction on the “Ref + 100” case.

5.5. Experiment II: models trained with multiple speakers’ data

In these experiments, the amount of training data is increased from the earlier “1/2 hour” training set. The results obtained from use of “1 hour” training dataset are listed in Table 4. For

Table 2

Performance (WER) of MLDM with a single mixture component and different H values (trained with “1/2 hour” training set)

Systems	Ref + 100	100 Best
MLDM:1-mix, 1- H	55.7	58.9
MLDM:1-mix, 2- H	55.0	57.7
MLDM:1-mix, 3- H	55.1	57.2

Table 3

Performance (WER) of MLDM with two mixture components and different H values (trained with “1/2 hour” training set)

Systems	Ref + 100	100 Best
MLDM:2-mix, 1- H	50.7	57.0
MLDM:2-mix, 2- H	50.4	56.6
MLDM:2-mix, 3- H	50.5	56.8

Table 4

Performance (WER) of MLDM with various H values and with various mixture components (trained with “1 hour” training set)

Systems	Ref + 100	100 Best
MLDM:2-mix, 1- H	51.0	57.1
MLDM:2-mix, 2- H	50.1	56.4
MLDM:2-mix, 4- H	50.0	56.4
MLDM:4-mix, 1- H	49.8	56.6
MLDM:4-mix, 2- H	49.6	56.5

Table 5

Performance (WER) of MLDM with various H values and with fixed four mixture components (trained with “2 hour” training set)

Systems	Ref + 100	100 Best
MLDM:4-mix, 1- H	49.5	56.0
MLDM:4-mix, 2- H	48.8	55.8

401 the two mixture component (2-mix) case, we observe a WER reduction from the use of one H
402 value to the use of more than one H values. Similar observations are made for the four mixture
403 component (4-mix) case, although the WER reduction is of less magnitude.

404 For the four mixture component (4-mix) case, we further experimented with using “2 hour”
405 training set. The WER results are shown in Table 5. A greater error reduction is observed moving
406 from one H value to two H values when compared with the earlier result of Table 4 with use of
407 fewer training data.

408 5.6. Some analysis of the model behavior

409 In this section, we provide some analysis on the behavior of the trained switching dynamic
410 model of speech and on the experimental results. The analysis is based on the fact that the noise
411 covariance matrix R (or noise variance if R is treated as diagonal as in our model implementation)
412 is estimated according to Eq. (27), where e_2 is the difference of the actual MFCC and the output
413 of the $h(\cdot)$ function. Thus, if the function $h(\cdot)$ accurately describes the relation between the VTR
414 space and the MFCC space, the estimated R will be small. Otherwise, the estimated R will be large.
415 Therefore, the accuracy of approximating the physically nonlinear relation between the VTR
416 space and the MFCC space using a piecewise linear function as implementing by switching- H
417 values can be assessed by examining the size of the estimated noise variance, R .

418 As typical examples, the diagonal values of the estimated R for phone models “aa”, “d”, and
419 “n” are shown in Tables 6–8, respectively. (Column three lists the average values of these diagonal
420 elements.) We observe that these estimated variance values are strictly decreasing with the in-
421 creasing number of H -switching values. This suggests that the approximation accuracy by using
422 the piecewise linear function is improved with the use of more linear functions. When the number
423 of H -switching values is increased from one to four, the average R value is decreased from 22.1 to
424 18.6 for “aa”, from 22.2 to 19.6 for “d”, and from 20.0 to 17.6 for “n”, respectively.

Table 6

Values of diagonal elements of R noise variance for the phone model “aa” as a function of the number of H switching values.

No. of H	Diagonal elements of R	Average
1 H	10.0 14.6 17.1 22.5 21.7 18.9 32.3 28.4 29.4 23.5 24.4 21.8	22.1
2 H	8.87 13.4 14.6 20.0 21.3 20.1 29.7 27.5 30.9 21.7 23.4 21.8	21.1
4 H	6.92 11.5 13.4 18.6 17.8 19.2 26.9 19.7 27.9 23.4 17.6 19.7	18.6

Table 7

Values of diagonal elements of R noise variance for the phone model “d” as a function of the number of H switching values

No. of H	Diagonal elements of R	Average
1 H	8.9 17.5 18.8 16.9 19.8 25.6 27.4 27.2 30.2 27.2 26.7 20.7	22.2
2 H	5.6 16.8 16.4 15.1 18.8 24.9 26.4 23.4 31.4 23.6 24.4 19.8	20.6
4 H	5.2 17.0 16.7 15.0 17.0 24.8 24.9 22.9 27.4 21.8 24.7 19.6	19.6

Table 8

Values of diagonal elements of R noise variance for the phone model “n” as a function of the number of H switching values

No. of H	Diagonal elements of R	Average
1 H	6.0 15.9 16.8 15.5 18.7 20.7 18.9 27.2 31.0 26.3 21.4 22.0	20.0
2 H	6.1 15.6 15.0 16.6 18.1 21.1 17.8 26.9 24.1 24.4 20.8 19.6	18.8
4 H	5.5 14.5 13.9 16.4 18.4 17.9 16.0 26.1 20.9 23.9 18.7 19.0	17.6

425 It is interesting to note that the mild improvement in the linear piecewise approximation ac-
426 curacy as reflected by the reduced R value is correlated with the mild WER reduction in the speech
427 recognition results presented earlier in this section.

428 6. Conclusions

429 A new version of the switching dynamic model of speech, the linear dynamic system model with
430 mixed-level (segment and frame) switching parameters, is presented in this paper. The segment-
431 level switching parameters in the target-directed state equation is the same as that in the earlier
432 single-level switching model (Ma and Deng, 2003), and the new frame-level switching parameters
433 in the observation or measurement equation is introduced in the current model. The use of the
434 frame-level switching parameters effectively provides piecewise linear functions to approximate
435 the physically nonlinear function between the partially observable VTR space and the observable
436 MFCC space in the observation equation.

437 A series of speech recognition experiments have been carried out to evaluate the new mixed-
438 level switching model. The experimental results show that the approximation accuracy is im-
439 proved with an increasing number of H -switching values (about a 10% reduction in the estimated
440 measurement noise variances). The speech recognizer built from the mixed-level switching

441 dynamic system model using the N-best rescoring evaluation paradigm also shows some varying
442 degrees of word error rate reduction compared with using the single-level switching model. The
443 new recognizer built with mixed-level switching parameters achieved a lower error rate than a
444 baseline HMM system evaluated under identical experimental conditions.

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