**Title**  
Parametric Speech Synthesis Based on Gaussian Process Regression Using Global Variance and Hyperparameter Optimization

**Authors**  
Tomoki Koriyama, Takashi Nose, Takao Kobayashi

**Citation**  

**Pub. date**  
2014, 5

**Copyright**  
(c) 2014 IEEE. Personal use of this material is permitted. Permission from IEEE must be obtained for all other uses, in any current or future media, including reprinting/republishing this material for advertising or promotional purposes, creating new collective works, for resale or redistribution to servers or lists, or reuse of any copyrighted component of this work in other works.

**URL**  
http://www.ieee.org/index.html

**DOI**  
http://dx.doi.org/10.1109/ICASSP.2014.6854319

**Note**  
This file is author (final) version.
PARAMETRIC SPEECH SYNTHESIS BASED ON GAUSSIAN PROCESS REGRESSION USING GLOBAL VARIANCE AND HYPERPARAMETER OPTIMIZATION

Tomoki Koriyama1, Takashi Nose2, Takao Kobayashi1

1Interdisciplinary Graduate School of Science and Engineering, Tokyo Institute of Technology
2Graduate School of Engineering, Tohoku University

koriyama@ip.titech.ac.jp, tnose@m.tohoku.ac.jp, takao.kobayashi@ip.titech.ac.jp

ABSTRACT

This paper examines two issues of a statistical speech synthesis approach based Gaussian process (GP) regression. Although GP-based speech synthesis can give higher performance in generating spectral parameters than the HMM-based one, a number of issues still remain. In this paper, we incorporate global variance (GV) feature to overcome over-smoothing problem into the parameter generation. Furthermore, in order to utilize an appropriate kernel function in accordance with actual data, we propose an EM-based kernel hyperparameter optimization technique. Objective and subjective evaluation results show that using GV and hyperparameter estimation enhanced the performance in spectral feature generation.

Index Terms— statistical parametric speech synthesis, Gaussian process, global variance, kernel hyperparameter

1. INTRODUCTION

In the last decade, statistical parametric speech synthesis framework based on hidden Markov model (HMM) [1] has shown various advantages such as flexibility to transforming voice characteristics, robustness to language dependence, and realization of small foot-print systems [2]. However, there still remain problems in vocoding, which result from insufficient accuracy of acoustic modeling and over-smoothing effect. Recently, for the purpose of more accurate acoustic modeling, other approaches to the statistical parametric acoustic modeling have been proposed using deep neural networks [3] and deep belief network-Gaussian process hybrid model [4].

In this context, we have proposed an alternative speech synthesis approach based on frame-level Gaussian process (GP) regression [5, 6]. In this approach, frame-level contextual features are used as input features of GP regression [7], and acoustic features are used as output features. To make computational cost feasible, we adopted partially independent conditional (PIC) approximation [8] and showed that the GP-based approach achieved comparable or better performance compared with HMM-based approach in generation of spectral feature trajectories [6]. This could be attributed to the advantages of GPs, such as the flexibility to model complexity and the robustness against over-fitting.

Although the GP-based speech synthesis is promising, there exist a number of issues for the realization of practical systems. One of them is generation of acoustic feature trajectories from predictive distribution. We used the predictive mean sequence as the generated trajectories in the previous study [6]. However, it generally causes over-smoothing problem. In this study, therefore, we incorporate an alternative way considering global variance (GV) [9] which has been widely used to alleviate over-smoothing problem in the HMM-based speech synthesis. Another issue is selection of hyperparameter of kernel function used in GP. Fortunately, it is possible to use automatic hyperparameter optimization based on empirical Bayesian approach for GP. In this paper, we apply it to the speech synthesis framework using PIC approximation.

2. GP-BASED SPEECH SYNTHESIS

In the speech synthesis framework based on GP regression [5,6], we consider GP in which an input variable $x_n$ is a contextual feature set obtained by linguistic and phone boundary information, and an output variable $y_n$ is an acoustic feature normalized into zero mean. The correlation between two frames, whose input variables are $x$ and $x'$, are defined by a kernel function $k(x, x'; \theta)$, where $\theta$ is a hyperparameter vector of the kernel function. This enables frame-level modeling of acoustic features without using dynamic features that are used and essential in the HMM-based speech synthesis. Based on GP regression framework [7], predictive distribution of test (synthetic) data $y_T$ given training data $y$ are inferred, which is given by a Gaussian distribution

$$p(y_T | y, \theta) = N(y_T; \mu_{y_T | y}, \Sigma_{y_T | y})$$

and used for parameter trajectory generation. For example, we used mean sequence $\mu_{y_T | y}$ in the previous study [5].

One issue of GP-based speech synthesis is computational cost. To reduce the computational cost, we have incorporated partially independent conditional (PIC) approximation [8] into GP-based acoustic modeling [6]. PIC approximation uses a latent variable $f_M$ which is dependent on pseudo-data input features $\{x_m | m = 1, \ldots, M\}$, where the pseudo-data size $M$ is much smaller than the number of frames in training data, $N$. The joint distribution of $f_M$ is given by following Gaussian distribution.

$$p(f_M | \theta) = N(f_M; 0, K_M)$$

where $K_M$ is a covariance (Gram) matrix whose elements are kernel values of pseudo-data input features, which is calculated by a kernel function $k(x, x'; \theta)$. In PIC approximation, all training data $y$ is separated into several blocks $\{y_{m,s} | s = 1, \ldots, S\}$, and it is assumed that $y_{m,1}, \ldots, y_{m,S}$ are conditionally independent given $f_M$. By using this assumption, the joint probability of all training data

A part of this work was supported by JSPS Grant-in-Aid for Scientific Research 24300071, 25540065, and 25-8776.
\[ y = [y_1 \ldots y_N]^T \] is defined by
\[
p(y|\theta) = \int p(y|f_M, \theta)p(f_M|\theta)df_M
\]
\[
= \int \prod_{s=1}^{S} p(y_{BS}|f_M, \theta)p(f_M|\theta)df_M
\]
(3)

where
\[
p(y_{BS}|f_M, \theta) = N(y_{BS}; \mu_{BS|f_M}, \Sigma_{BS|f_M})
\]
(4)
\[
\mu_{BS|f_M} = K_{BS}M_K f_M
\]
(5)
\[
\Sigma_{BS|f_M} = K_{BS} - Q_{BS} + \sigma^2_s I
\]
(6)
\[
Q_{BS} = K_{BS}M_K K_{MB_s}.\]
(7)

Here, \( K_{BS}M = K_{MB_s} \) represents the covariance matrix that 
expresses the relationship between the block size and the pseudo-data, and \( K_{BS} \) and \( K_{MB_s} \) correspond to self-covariance matrices of the block size and the pseudo-data, respectively. \( \sigma^2_s \) is a noise variance.

By marginalizing out the latent variable \( f_M \) in (3), we get the following Gaussian distribution.
\[
p(y|\theta) = N(y; 0, K_N^{\text{PIC}} + \sigma^2_s I)
\]
(8)

PIC approximation adopts the independence assumption for test (synthetic) data in the same way as that for the training data. The predictive distribution of test data is expressed by a Gaussian distribution. Let \( D_T = \{ (x_t, y_t) \}_{t=1}^{T} \) be a test data set, and \( y_{TS} \) be a partial vector of \( y_T \), which consists of test data assigned to the block size. Then, we obtain following predictive distribution for the synthetic acoustic feature trajectory.
\[
p(y_T|y, \theta) = N(y_T; \mu_{y_T|y}, \Sigma_{y_T|y})
\]
(11)
\[
\mu_{y_T|y} = \left[ \begin{array}{c} \mu_{y_{T_1}|y} \\ \vdots \\ \mu_{y_{T_S}|y} \end{array} \right]
\]
(12)
\[
\Sigma_{y_T|y} = \left[ \begin{array}{ccc} \Sigma_{y_{T_1}|y} & & \\ & \ddots & \\ & & \Sigma_{y_{T_S}|y} \end{array} \right].
\]
(13)

Here, let
\[
K_N^{\text{PIC}} = \left[ Q_{BS} \cdots \right]_{BS+1}K_{BS}Q_{BS} \cdots Q_{BS+1}.
\]
(14)
then the partial vector \( \mu_{y_{TS}|y} \) and matrix \( \Sigma_{y_{TS}|y} \) are given by
\[
\mu_{y_{TS}|y} = \left( K_{N}^{\text{PIC}} + \sigma^2_s I \right)^{-1} \begin{bmatrix} \mu_{y_{T_1}|y} \\ \vdots \\ \mu_{y_{T_S}|y} \end{bmatrix}
\]
(15)
\[
\Sigma_{y_{TS}|y} = Q_{TS} + \delta_{TS} \left(K_{TS} - Q_{TS} + \sigma^2_s I \right)
\]
(16)

\[
K_{N}^{\text{PIC}} = \left( K_{N}^{\text{PIC}} + \sigma^2_s I \right)^{-1}.
\]
Since the matrix \( (K_{N}^{\text{PIC}} + \sigma^2_s I) \) is the sum of the low rank matrix \( Q_N \) and the block diagonal matrix, we can speed up the inversion of \( (K_{N}^{\text{PIC}} + \sigma^2_s I) \) using the Woodbury, Sherman & Morrison formula [10]. When the maximum block size is \( B \), the computational cost for training results in \( O((B + M)^3) \), whereas GP without any approximation methods needs \( O(N^3) \) computational cost.

3. PARAMETER TRAJECTORY GENERATION USING PREDICTIVE DISTRIBUTION WITH GV

A simple way to generate a speech parameter trajectory from the predictive distribution is using the acoustic feature trajectory \( y_T \) that maximizes the likelihood of predictive distribution \( p(y_T|y, \theta) \). Since the predictive distribution is a Gaussian distribution, the trajectory that maximizes the likelihood becomes the mean sequence \( \mu_{y_T|y} \). However, the resultant trajectories tend to be overly smoothed. To alleviate this effect, we incorporate global variance (GV) constraint [9] into trajectory generation.

GV is an utterance-level feature which is obtained by
\[
v(y_T) = \frac{1}{T} \sum_{t=1}^{T} (y_t - m(y_T))^2
\]
(17)
\[
m(y_T) = \frac{1}{T} \sum_{t=1}^{T} y_t
\]
(18)

where \( T \) is the number of frames of an utterance. To incorporate GV into GP-based speech synthesis, we define the following likelihood
\[
L_GV = p(y_T|y, \theta)^a p(v(y_T)|\mu_s, \sigma^2_s)
\]
(19)

\[
\mu_s \text{ and } \sigma^2_s \text{ correspond to the mean and variance of GVs of training utterances, respectively.}
\]

The parameter \( \omega \) adjusts the weight of Gaussian process likelihood and GV likelihood. In this study, we use \( \omega = 1/T \), i.e., the dimension ratio of \( v \) to \( y_T \).

We generate a speech parameter trajectory by maximizing \( L_GV \). Practically, in order to search the optimum trajectory, we use a steepest descent algorithm with the derivative
\[
\frac{\partial L_GV}{\partial y_T} = -\omega \Sigma_{y_T|y}^{-1}(y_T - \mu_{y_T|y})
\]
(20)
\[
- \frac{2}{T} \frac{(v(y_T) - \mu_s)}{\sigma^2_s} (y_T - m(y_T)) \cdot 1.
\]

4. OPTIMIZATION OF KERNEL HYPERPARAMETERS

Here we derive how to estimate an appropriate hyperparameter set \( \theta \) automatically for PIC approximation. In the same way as usual GPs, we take the empirical Bayesian approach, where we search optimum hyperparameter set \( \theta \) that maximizes the marginal likelihood \( p(y|\theta) \). Since PIC approximation includes a latent variable \( f_M \), EM algorithm can be applied to the maximization. Q-function is defined by
\[
Q(\theta, \tilde{\theta}) = \int p(f_M|y, \theta) \log \left( p(y|f_M, \tilde{\theta})p(f_M|\theta) \right) df_M
\]
(21)

where \( \tilde{\theta} \) is a new hyperparameter set. Using the assumption of PIC approximation, the Q-function is decomposed into two Q-functions \( Q_s \) and \( Q_M \) as follows:
\[
Q_s(\theta, \tilde{\theta}) = \sum_{s=1}^{S} Q_s(\theta, \tilde{\theta}) + Q_M(\theta, \tilde{\theta})
\]
(22)
\[
Q_s(\theta, \tilde{\theta}) = \int p(f_M|y, \theta) \log p(y_{BS}|f_M, \tilde{\theta}) df_M
\]
(23)
\[
Q_M(\theta, \tilde{\theta}) = \int p(f_M|y, \theta) \log p(f_M|\theta) df_M.
\]
(24)
In the E-step, we calculate the posterior distribution of pseudodata variables by

\[
p(f_M | y, \theta) = \mathcal{N}(f_M; \mu_{f_M} | y, \Sigma_{f_M} | y).
\]

From Eqs. (2) and (4), we can derive the parameters

\[
\mu_{f_M} | y = \Sigma_{f_M} | y K_M^{-1} + \sum_{s=1}^S K_M B_s \Sigma_{y,s}^{-1} y b_s
\]

\[
\Sigma_{f_M} | y = K_M^{-1} + K_M^{-1} \sum_{s=1}^S K_M B_s \Sigma_{y,s}^{-1} y b_s K_M^{-1}.
\]

When the block size is \( B \) and pseudodata size is \( M \), we need \( O(S(B + M)^3) \) calculations for the each E-step, which is equal to the number of calculations of the model training with PIC approximation.

Based on the previous study [5, 6], we defined the following kernel

\[
\begin{align*}
\text{the first term represents the core of this kernel. The superscripts 1, 2, and 3 correspond to relevance, scale parameters and, noise term have equal relevance. On the other hand, we used}\; \\
\text{the second term of the right side of (30) is introduced to model short time correlation within the same utterance, where } \text{utt}(n) \text{ is the} \\
\text{utterance index of frame } n. \text{The input feature of kernel } k_r(\cdot, \cdot) \text{ is the time in the utterance. We used the following SE kernel}\;
\end{align*}
\]

\[
k_r(x_m, x_n) = \sum_{i \in \{ -1, 0, 1 \}} \sum_{j \in \{ -1, 0, 1 \}} u_{i,m} u_{j,n} k_p(p_{i,m}^{(i)}, p_{j,n}^{(i)}) k_c(c_{i,m}^{(i)}, c_{j,n}^{(j)})
\]

\[
+ \delta_{\text{utt}(m) \neq \text{utt}(n)} k_r(\tau_m, \tau_n) + \delta_{mn} \sigma^2_{\text{floor}}.
\]

Fig. 1. Log likelihood of predictive distribution for each dimension using optimized/non-optimized hyperparameters

where \( \theta_{pa,k} \) and \( \theta_{ph,k} \) correspond to relevance and scale parameters for input feature \( p_{n,k}^{(i)} \). For the phone context kernel \( k_c(c_{i,m}^{(i)}, c_{j,n}^{(j)}) \), we used a linear kernel in the same way as [5] defined by

\[
k_c(c_{i,m}^{(i)}, c_{j,n}^{(j)}) = \sum_{k_1=1}^{3} \theta_{pa,k} \theta_{ph,k} \exp \left( -\left( \frac{p_{i,m}^{(i)} - p_{j,n}^{(j)}}{2\theta_{pa,k}} \right)^2 \right)
\]

\[
\text{The last term in (30) is a flooring value to keep } \mathbf{K}_M \text{ positive definite since } \mathbf{K}_M^{-1} \text{ needs to be calculated.}
\]

In summary, the hyperparameter set used in this study became

\[
\theta = (\theta_{pa,1}, \ldots, \theta_{pa,3}, \theta_{ph,1}, \ldots, \theta_{ph,3}, \theta_{c,1}, \ldots, \theta_{c,3}, \theta_{r}, \theta_{pa,01}, \theta_{pa,02}, \sigma_e).
\]

\[
\theta_{pa,01} = \theta_{pa,02} = 0.209 \text{ which is equal to the standard deviation of the uniform distribution on } [0, 1]. \text{The other scale hyperparameters }
\]

\[
\theta_{ph,01} \text{ and } \theta_{ph,02} \text{ were set to 0.10s to consider correlations of frames of shorter range than } \theta_{ph,1} \text{ and the scale hyperparameter for short-time correlation kernel, } \theta_{pa,01}, \text{ was set to 0.20s.}
\]

5.2. Experimental conditions

We used speech data of a Japanese female voice actress. The speaker uttered 503 phonetically balanced sentences with a reading style.
Table 1. Mel-cepstral distances between original and synthetic speech [dB].

<table>
<thead>
<tr>
<th>Generation</th>
<th>HMM</th>
<th>GP-INIT w/o GV</th>
<th>GP-OPTIM w/o GV</th>
<th>HMM w/ GV</th>
<th>GP-INIT w/ GV</th>
<th>GP-OPTIM w/ GV</th>
</tr>
</thead>
<tbody>
<tr>
<td>w/o GV</td>
<td>5.38</td>
<td>5.11</td>
<td>5.03</td>
<td>5.38</td>
<td>5.11</td>
<td>5.03</td>
</tr>
<tr>
<td>w/ GV</td>
<td>5.95</td>
<td>5.59</td>
<td>5.55</td>
<td>5.95</td>
<td>5.59</td>
<td>5.55</td>
</tr>
</tbody>
</table>

Fig. 2. Global variances of natural and generated mel-cepstrum sequences. These values show GV means over all test utterances. The optimized hyperparameters were used for generation.

These sentences were taken from ATR Japanese speech database set B [12]. In this study, we modeled and generated spectral features only. Speech samples were synthesized using generated spectral features, while F0s, aperiodicity features, and phone durations were taken from the original speech.

The number of training sentences was 450. The remaining 53 sentences, which were not included in the training data, were used as test data. The phone boundary information was annotated manually. Speech signals were sampled at a rate of 16kHz, and the frame shift was 5ms. The 0-39th mel-cepstral coefficients derived from the spectral envelop extracted by STRAIGHT [13] were used as the spectral features. The maximum number of frames $B$ of each block was set to 1000. The number of pseudo data sets $M$ was set to 200 and the samples of pseudo data sets were randomly chosen from the training data. The number of iterations of EM algorithm in hyperparameter optimization was 5.

For comparison, we also evaluated the HMM-based speech synthesis using minimum generation error (MGE) training [14]. The model topology was 5-state, left-to-right, no-skip hidden semi-Markov model (HSMM). The output distribution in each state was modeled with a single Gaussian pdf, with diagonal covariance matrices. The feature vector included delta and delta-delta dynamic features as well as the static one. Triphones were used for the context set for the HMM training. In the decision-tree-based context clustering for parameter tying, MDL was used as a stopping criterion [15].

5.3. Objective evaluation

To evaluate the effectiveness of the hyperparameter estimation objectively, log predictive likelihoods and spectral distortions are calculated. Figure 1 shows the average log predictive likelihoods of original utterances obtained by the predictive distribution $p(y_f|y, \theta)$. GP-INIT and GP-OPTIM represent the GP models using the initial kernel hyperparameters and optimized ones. It can be seen that the hyperparameter optimization consistently increased the predictive likelihood especially in lower dimensions. Table 1 shows the spectral distortions between original and synthetic speech. The table includes the generation methods without (w/o) and with (w/) GV. In both generation methods, the hyperparameter optimization increased the score in the case of “w/o GV”. In addition, considering GV increased the scores in the GP-based method, whereas the global variances of the generated trajectories got closer to natural speech, by comparing average global variance as shown in Fig. 2. It should also be noted that the distortions of GP-based method were smaller than HMM-based ones.

5.4. Subjective evaluation

The naturalness of the synthetic speech was evaluated by a mean opinion score (MOS) test. The number of participants was six. Each participant listened to the synthetic speech samples and rated the naturalness of synthetic speech on a five-point scale: 5: excellent, 4: good, 3: fair, 2: poor, and 1: bad. Fifteen sentences were randomly chosen from 53 sentences for each participant. Figure 3 shows the mean opinion scores (MOSs). The error bars indicate 95% confidence intervals. Note that every MOS score was higher than ordinary TTS ones because the original prosodic features were used as it were in the waveform generation. From the figure, it is seen that hyperparameter optimization increased the score in the case of “w/o GV”. In addition, considering GV increased the scores in the GP-based approach as well as in the HMM-based one, and GP-OPTIM with GV gave the highest score among the methods. There was a significant difference between HMM with GV and GP-OPTIM with GV at a 5% significance level.

6. CONCLUSIONS

In this paper, we proposed a generation method of acoustic feature trajectory considering GV in GP-based statistical speech synthesis approach. Moreover hyperparameter optimization for PIC approximation using EM algorithm are introduced. The proposed method which uses GV and hyperparameter optimization outperformed the conventional HMM-based approach by subjective evaluation. In our future work, we will add contexts such as accent and part of speech that are used in general TTS systems and sub-phone or state information. Then, the relevance of these contexts will be automatically determined by the hyperparameter optimization process. Furthermore, we will examine GP-based models of F0 and duration features using both prosodic contexts.
7. REFERENCES


