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Modeling of Fundamental Frequency Contour of Thai Expressive Speech using Fujisaki's Model and Structural Model

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Abstract: Problem statement: In spontaneous speech communication, prosody is an important factor that must be taken into account, since the prosody effects on not only the naturalness but also the intelligibility of speech. Focusing on synthesis of Thai expressive speech, a number of systems has been developed for years. However, the expressive speech with various speaking styles has not been accomplished. To achieve the generation of expressive speech, we need to model the fundamental frequency (F0) contours accurately to preserve the speech prosody to preserve the quality of speech prosody. **Approach:** This study presents a comparison of two successful F0 models. One approach is based on the Fujisaki's model which has been applied for many tonal and toneless languages. Another one is based on the structural model which has been conducted primarily for Mandarin Chinese. It is based on the assumption that the behavioral characteristics of vocal-fold elongation in vibration could be approximated by those of a simple forced vibrating system. Therefore this approach has been applied to model Thai expressive speech with best-fit function. Our speech database consists of male and female speech and each one contains 4 different speech styles including angry style, sad style, enjoyable style and reading style. Five sentences are used for each speech style and each sentence includes 100 samples. The speech sample in each group is analyzed for an F0 contour, subsequently a number of Fujisaki's and structural modeling parameters are extracted for each contour. Thereafter, the parameters are used to synthesis the F0 contour and then the synthesized contour is compared with that of natural speech by calculating RMS error. **Results:** From the experimental analysis, it has been observed that RMS error of each speech style is different from the others for both models. It also reveals that the RMS error of the Fujisaki's model is higher than that of the structural model for all speech styles. In other words, the structural model gives the better fit for modeling of the F0 contour of the expressive speech than that of the Fujisaki's model. **Conclusion:** From the finding, it is a definite evidence that the structural model is more appropriate than that of the Fujisaki's model for modeling four different speech styles including angry style, sad style and enjoyable style and reading style.

Key words: Noised speech, speech analysis, fundamental frequency contour, speech enhancement, speech synthesis, speaker-independent, local-accent, model parameters, Fujisaki's model

INTRODUCTION

The fundamental frequency of voice speech is the most important feature among all of the features known to carry prosodic information which is an inherently supra-segmental feature of human speech. The F_0 contours of an utterance convey the stress, intonation and rhythmic structures, which determine the naturalness and intelligibility of synthetic speech. As a result, the appropriate modeling of F_0 contour plays a significant role in the speech processing area, e.g., speech recognition, speech synthesis, speech analysis and speech coding. A number of modeling techniques in the former studies have been performed in various levels of speech units, e.g., utterance level (Saito and Sakamoto, 2002; Li *et al*., 2004; Tao *et al*., 2006), word and syllable levels (Fujisaki and Sudo, 1971; Tran *et al*., 2006). In Thai speech, Fujisaki's model has been successfully applied for modeling of utterances, tones and words (Hiroya and Sumio, 2002; Seresangtakul and Takara, 2002; 2003). In the Thai speech synthesis, the statistical modeling of F_0 contour has been conducted by Chomphan and Kobayashi in the implementation of both speaker-dependent and speaker-

independent systems in 2007-2009 (Chomphan and Kobayashi, 2007; 2008; 2009). Lately, the Fujisaki's model has been applied within a speaker-independent system as extended modules. Moreover, it has also been exploited in the modeling of Thai expressive speech; i.e., sad, happy, angry styles (Chomphan and Kobayashi, 2008; 2009). Another study has been conducted by using a structural model which is based on the assumption that the behavioral characteristics of vocal-fold elongation in vibration could be approximated by those of a simple forced vibrating system (Ni and Hirose, 2006; Chomphan and Kobayashi, 2009). The RMS error calculation has been done for evaluation the modeling performance for both mentioned speech models and also for all speech styles including angry style, sad style, enjoyable style and reading style. This study mainly aims at comparing the Fujisaki's model and the structural model.

MATERIALS AND METHODS

Fujisaki's model: The F0 contour is treated as a linear superposition of a global phrase and local-accent components on a logarithmic scale, as depicted in Fig. 1 (Chomphan and Kobayashi, 2008; 2009).

Typically, the phrase command produces a baseline component, while the accent command produces the accent component of an F0 contour. Mathematically, the F0 contour of an utterance generated from an extension of the Fujisaki's model for tonal languages has the following expressions (Seresangtakul and Takara, 2003):

$$\ln F0(t) = \ln F_b + \sum_{i=1}^{I} A_{pi}[G_{pi}(t - T_{0i})] + \sum_{j=1}^{J}\sum_{k=1}^{K(j)} A_{t,jk}[G_{t,jk}(t - T_{1jk}) - G_{t,jk}(t - T_{2jk})] \quad (1)$$

$$G_{pi}(t) = \begin{cases} (\alpha_i^2 t)\exp(-\alpha_i t) & \text{for } t \geq 0 \\ 0 & \text{for } t < 0 \end{cases} \quad (2)$$

$$G_{t,jk}(t) = \begin{cases} [1 - (1+\beta_{jk} t)\exp(-\beta_{jk} t)] & \text{for } t \geq 0 \\ 0 & \text{for } t < 0 \end{cases} \quad (3)$$

Where:

$G_{pi}(t)$ = Represents the impulse-response function of the phrase-control mechanism

$G_{t,jk}(t)$ = Represents the step-response function of the tone-control mechanism

The symbols used in three equations denote that F_b is the smallest F0 value in the F0 contour of interest and A_{pi} and $A_{t,jk}$ are the amplitudes of the i-th phrases and of the j-th tone command. Here, T_{0i} is the timing of the i-th phrase command and T_{1jk} and T_{2jk} are the onset and offset of the k-th component of the j-th tone command. α_i and β_{jk} are time constant parameters, while I, J, K(j) correspond to the number of phrases, tones and components of the j-th tone contained in the utterance.

To find the optimal representative parameters, optimization is carried out by minimizing the mean squared error in the Ln F0(t) domain through the hill-climbing search in the space of model parameters (Seresangtakul and Takara, 2003). To use this model, the parameters are extracted from the speech database, utterance by utterance. The derived parameters are subsequently computed.

Derived parameters: From the conventional parameters, we proposed seven derived parameters which reflect the geometrical appearance of the F0 contour of an utterance as follows:

- Baseline frequency
- Numbers of phrase commands
- Numbers of tone commands
- Phrase command duration
- Tone command duration
- Amplitude of phrase command
- Amplitude of tone command

All of them have been extracted for four speech expressions of angry style, sad style, enjoyable style and reading style. Thereafter, the extracted parameters are used to resynthesize the F0 contour in the evaluation process.

Structural model: The voice F_0 contour is modeled in a logarithmic scale, as depicted in Fig. 2. The mathematical model has been applied (Ni and Hirose, 2006; Chomphan and Kobayashi, 2009) by using a structural control consisting of placing a series of normalized F_0 targets along the time axis, which are also specified by transition time and amplitudes. The transitions between targets are approximated by connecting truncated second-order transition functions.

From the background knowledge that the physical factors to regulate the frequency of vocal-fold vibrations are the mass, length and tension of vibrating structures, all of which are dynamically controlled primarily by the intrinsic and extrinsic muscles of the larynx and secondly by the sub-glottal pressure (Ni and Hirose, 2006).

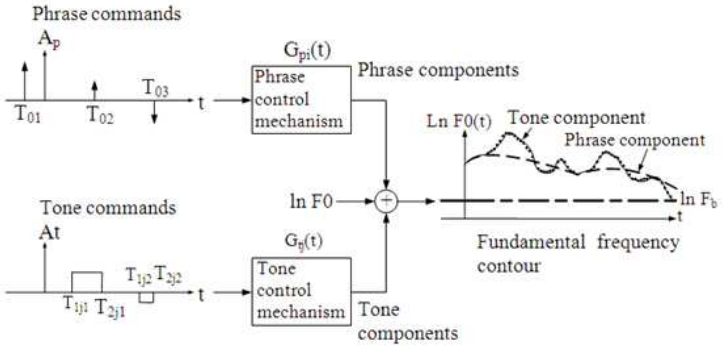

Fig. 1: An extension of Fujisaki's model for the generation of F0 contour

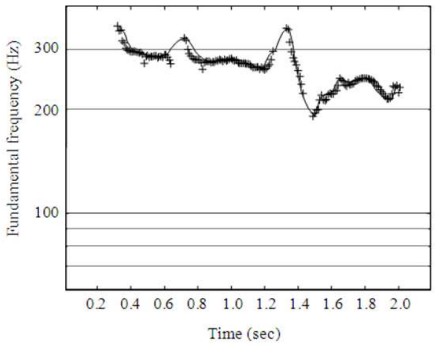

Fig. 2: F_0 contour with a trend line in a logarithmic scale

Fujisaki explained that logarithmic fundamental frequency varies linearly with vocal-fold elongation x (Fujisaki, 1983), which can be represented in the following mathematical term:

$$\ln f_0 = \frac{b}{2}x + \ln(\sqrt{a}c_0) \quad (4)$$

where, a, b and c_0 are constant coefficients (Fujisaki, 1983).

Assumption: The behavioral characteristics of vocal-fold elongation in vibration can be approximated by those of a simple forced vibrating system (Ni and Hirose, 2006).

Formulating the assumption, the behavioral characteristics of a simple forced vibrating system can be characterized by the amplifying coefficients of its vibrating amplitudes:

$$A\left(\frac{\omega_d^2}{\omega_f^2}, \zeta\right) = \frac{1}{\sqrt{\left(1-\left(1-2\zeta^2\right)\frac{\omega_d^2}{\omega_f^2}\right)^2 + 4\zeta^2\left(1-2\zeta^2\right)\frac{\omega_d^2}{\omega_f^2}}} \quad (5)$$

where, ω_d and ω_f denote the natural angular frequencies of the driven system and the driving force, respectively, ζ is called damping ratio indicating how tightly the driving force and the driven system are coupled together. Subsequently, replacing ω_d^2/ω_f^2 (square frequency ratio) by λ and substituting $A(\lambda, \zeta)$ expressed in Eq. 5 for x of Eq. 4, as a result, the logarithmic fundamental frequency can be expressed as:

$$\ln f_0 = \frac{b \times C}{2} A(\lambda, \zeta) + \ln(\sqrt{a}c_0) \quad (6)$$

where, C is a constant coefficient.

Typically, a speaker has an individual vocal range. Let f_{0t} and f_{0b} denote the top and bottom frequencies of the vocal range of a speaker and λ_t and λ_b denote two λ values that are one-to-one mapped to f_{0t} and f_{0b}. The relationship between f_0 within the vocal-range frequency interval and its corresponding λ is shown as follows:

$$\frac{\ln f_0 - \ln f_{0_b}}{\ln f_{0_t} - \ln f_{0_b}} = \frac{A(\lambda, \zeta) - A(\lambda_b, \zeta)}{A(\lambda_t, \zeta) - A(\lambda_b, \zeta)} \quad (7)$$

Since f_{0t} and f_{0b} are the top and bottom frequencies of the vocal range, λ_t and λ_b shall be determined regardless of ζ.

Practically, f_0 and λ can be determined through $f_0 = T_{f0}(\lambda, \zeta)$ and $\lambda = T_\lambda(f_0, \zeta)$, where they can be derived from Eq. 7 as followings:

$$T_{f_0}(\lambda, \zeta) = \exp\left(\frac{A(\lambda, \zeta) - A(\lambda_b, \zeta)}{A(\lambda_t, \zeta) - A(\lambda_b, \zeta)} \times \ln\frac{f_{0_t}}{f_{0_b}} + \ln f_{0_b}\right) \quad (8)$$

Table 1: Target values for sparsely specifying an F_0 contour

i	0	1	2	3	4	5	6	7
t_i (s)	0.32	0.46	0.57	0.72	0.89	1.00	1.18	1.25
f_{0i} (Hz)	351.00	296.00	287.00	323.00	275.00	280.00	266.00	296.00
λ_i	1.28	1.40	1.42	1.34	1.44	1.42	1.46	1.40

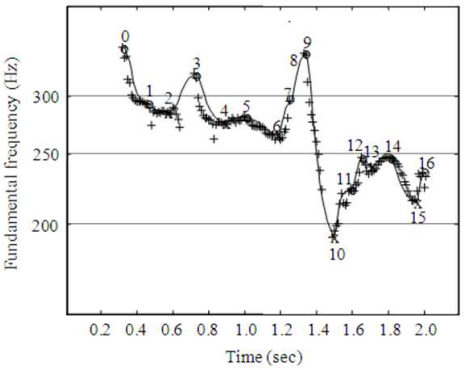

Fig. 3: Example of pitch target allocation on an F_0 contour

and $T_\lambda(f_0, \zeta)$ can be obtained by searching λ from 1 step-by-step in small increments (e.g., 0.0001), given $\lambda_b > \lambda_t$, until λ satisfies the following conditions:

$$\ln\frac{f_0}{f_{0_b}} > \frac{A(\lambda+0.0001,\zeta)-A(\lambda_b,\zeta)}{A(\lambda_t,\zeta)-A(\lambda_b,\zeta)}\times\ln\frac{f_{0_t}}{f_{0_b}} \quad (9)$$

and

$$\ln\frac{f_0}{f_{0_b}} < \frac{A(\lambda-0.0001,\zeta)-A(\lambda_b,\zeta)}{A(\lambda_t,\zeta)-A(\lambda_b,\zeta)}\times\ln\frac{f_{0_t}}{f_{0_b}} \quad (10)$$

Allocation of pitch targets: By applying the structural model, the pitch targets on an F_0 contour is allocated in advance. Thereafter the parameter of the structural model will be approximated. For example, Fig. 3 shows a sparser specification of the F_0 contours shown in Fig. 1, the 'o' signs indicate the tonal peaks, the 'x' signs indicate the tonal valleys and the square sign indicates a neutral target to reset an into national phrase.

Let t_i and f_{0i} denote the timing and F_0 value of the i^{th} target, respectively. Table 1 lists the first eight target points (t_i, f_{0i}) and corresponding $\lambda_i = T_\lambda(f_{0i}, \zeta_0)$, given $\lambda_t = 1$; $\lambda_b = 2$; $f_{0b} = 120$Hz; $f_{0t} = 420$Hz and $\zeta_0 = 0.156$. The i^{th} local F_0 movement, i = 0, ..., 15, is defined as a scope extending from the i^{th} target (t_i, f_{0i}) to the next. If $f_{0i} \le f_{0i+1}$, the local F_0 movement is therefore rising; otherwise, falling. The measured F_0 contours are subsequently approximated by connecting the rising and falling transitions through these target points.

For an F_0 falling movement, say i = 0, first compute $\lambda(t)$ for $t_0 \le t \le t_1$ by using following equation:

$$\lambda(t) = \lambda_p + \Delta\lambda\left(1-\left(1+\frac{4.8}{\Delta t}t\right)e^{-\frac{4.8}{\Delta t}t}\right), \quad \text{for } t \ge 0 \quad (11)$$

with the following parameters (based on Table 1): $\lambda_p (= \lambda_0) = 1.28$; $\Delta\lambda = \Delta\lambda_0/0.95$ and $\Delta t = \Delta t_0/0.95$, where $\Delta\lambda_0 = (\Delta\lambda_1 - \lambda_0) = 0.12$; $\Delta t_0 = (t_1 - t_0) = 0.14$. Second, synthesize contour by using $f_0(t) = T_{f0}(\lambda(t), \zeta_0)$ of Eq. 5. The thick line between the 0^{th} and 1^{st} targets shown in Fig. 3 indicates the re-synthesized F_0 contour.

For an F_0 rising movement, say i = 2, first compute $\lambda(t)$ for $t_2 \le t \le t_3$ by using Eq. 8 with the following parameters: $\lambda_p = 2-\lambda_2 = 0.58$; $\Delta\lambda = \Delta\lambda_2/0.95$, where $\Delta\lambda = (2-\lambda_3)-(2-\lambda_2) = 0.08$and $\Delta t = \Delta t_2/0.95$, where $\Delta t_2 = t_3 - t_2 = 0.15$. Then, synthesize contour by using $f_0(t) = T_{f0}(2-\lambda(t), \zeta_0)$ of Eq. 5. It has been noted that $A(\lambda, \zeta) = A(2-\lambda, \zeta)$ is applied to the computation. In Fig. 3, the thick line between the 2^{nd} and 3^{rd} targets indicates the re-synthesized F_0 contour.

A mathematical model: Subsequently, let $F_0(t)$ represent an F_0 contour as a function of time t in a vocal range [f_{0b}, f_{0t}] (Ni and Hirose, 2006; Nicknam *et al*., 2009). Assume $\Lambda(t)$ to indicate a sequence of virtual tone graphs in λ-time space to specify the underlying lexical tone structures. Additionally, assume a latent scale $\zeta(t)$ to characterize the intonation components. Thus, the F_0 contour on the logarithmic scale of fundamental frequency is expressed as a scale transformation from $\Lambda(t)$ to $F_0(t)$, corresponding to the syllabic tones fitting themselves with sentence intonation in the vocal range:

$$\frac{\ln F_0(t)-\ln f_{0_b}}{\ln f_{0_t}-\ln f_{0_b}} = \frac{A(\Lambda(t),\zeta(t))-A(\lambda_b,\zeta(t))}{A(\lambda_t,\zeta(t))-A(\lambda_b,\zeta(t))} \quad (12)$$

, for $t \ge 0$

Where:

$$A(\lambda,\zeta) = \frac{1}{\sqrt{(1-(1-2\zeta^2)\lambda)^2+4\zeta^2(1-2\zeta^2)\lambda}}, \quad (13)$$

for $\lambda \ge 1$

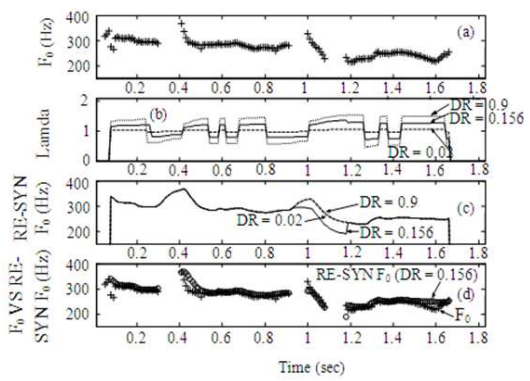

Fig. 4: An example of re-synthesis of F_0 contour by using the structural model (RE-SYN denotes "resynthesized", DR denotes "damping ratio")

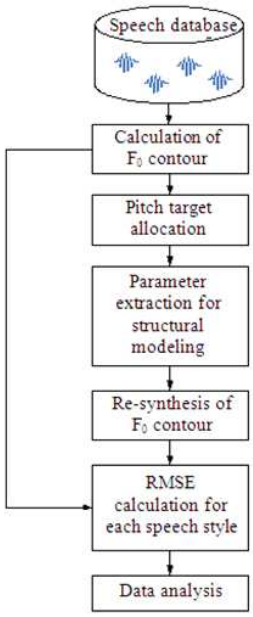

Fig. 5: Work flow for the experimental process

Equation 12-13 jointly indicate a structural formulation of the control process of coupling the syllabic tones and sentence intonation together to form a final sentence melody. Equation 12 states that F_0 contour $F_0(t)$ is a transformation of a sequence of virtual tone graphs $\Lambda(t)$ on a latent scale $\zeta(t)$. $\Lambda(t)$ is expressed as a concatenation of n parametric bell-shaped patterns lining up in series on the time axis with the following definition:

$$\Lambda(t) = \Lambda_{r_1}(t) + \sum_{i=1}^{n-1} \min(\Lambda_{f_i}(t), \Lambda_{r_{i+1}}(t)) + \Lambda_{f_n}(t) \quad (14)$$

where, $\Lambda_{ri}(t)$ and $\Lambda_{fi}(t)$ indicate the rising and falling transitions of the i^{th} bell-shaped pattern, respectively. Their definitions are as follows:

$$\Lambda_{r_i}(t) = \begin{cases} \lambda_{p_i} + \Delta\lambda_{r_i}(1 - D_{r_i}(t_{p_i} - t)), & \text{for } t_{p_{i-1}} \le t < t_{p_i} \\ 0, & \text{otherwise,} \end{cases} \quad (15)$$

$$\Lambda_{f_i}(t) = \begin{cases} \lambda_{p_i} + \Delta\lambda_{f_i}(1 - D_{f_i}(t - t_{p_i})), & \text{for } t_{p_i} \le t < t_{p_{i+1}} \\ 0, & \text{otherwise,} \end{cases} \quad (16)$$

Where:

$$D_{x_i}(t) = \left(1 + \frac{4.8t}{\Delta t_{x_i}}\right) e^{-\frac{4.8t}{\Delta t_{x_i}}}, \quad x \in \{r, f\} \quad (17)$$

The model parameters in Eq. 12-17 are defined as follows:

$[f_{0_b}, f_{0_t}]$: Bottom and top frequencies of the vocal Range in hertz
$[\lambda_b, \lambda_t]$: Bottom and top values of normalized vocal ranges in λ
$\zeta(t)$: Latent scales
n: Number of the bell-shaped patterns
(t_{p_i}, λ_{p_i}): i^{th} peak coordinate in λ-time space; $t_{p_0} = 0$ and $t_{p_{n+1}} = \infty$
Δt_{r_i}: i^{th} rising transition time
$\Delta\lambda_{r_i}$: i^{th} rising transition amplitude
Δt_{f_i}: i^{th} falling transition time and
$\Delta\lambda_{f_i}$: i^{th} falling transition amplitude, $i = 1, ..., n$

Figure 4 shows an example of re-synthesis of F_0 contour by using the structural model. Figure 4a shows the F_0 contour extracted from the natural speech, while Fig. 4b shows corresponding value of λ for three different fixed damping ratios ζ (0.156, 0.02 and 0.9). Figure 4c shows the re-synthesized F_0 contour with the three damping ratios in Fig. 4b, while Fig. 4d compares the F_0 contour extracted from the natural speech and the re-synthesized F_0 contour with limited samples (Geravanchizadeh and Rezaii, 2009).

An experimental design: The flow chart in Fig. 5 shows the core process for our experiment. At first the speech corpus has been implemented. There is male and female speech in the corpus. Each of them has four speech styles including happy, sad, angry and reading

styles. Each style consists of 5 sentences with 100 samples of utterances. Therefore our speech corpus contains 4,000 utterances. At the beginning, the F_0 values of an utterance have been calculated and then the pitch targets have been allocated by using local minimum/maximum criteria. In between any two adjacent pitch targets used as fixed points, an exponential function has been approximated to minimize the difference between the approximated function and the F_0 contour. The corresponding parameters from all of the functions along the utterance will be used as its representatives. Subsequently, the resynthesis of F_0 contour from the parameters has been conducted. Thereafter, the RMS error between the natural F_0 contour and the resynthesized F_0 contour has been executed. Finally, we analyzed the summarized data from the previous stages.

RESULTS

The RMS error calculation process has been performed, thereafter the experimental data can be summarized in the following five bar charts (Fig. 6-10). The averaged RMS errors from five different sentences have been calculated. Each bar chart represents one sentence and contains those of four speech styles including happy, sad, angry and reading styles. Moreover each bar chart contains four lines; two are of the Fujisaki's model including male and female speech, while the other twos are of the structural model including male and female speech.

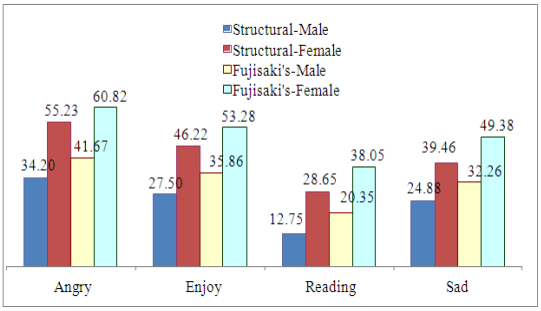

Fig. 7: Averaged RMS error for sentence "τHΦ⌉-παφ-να&φ-μA⌉" in IPA "where have you been?" in English

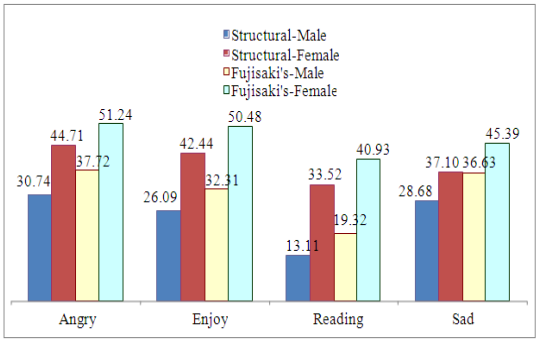

Fig. 8: Averaged RMS error for sentence "χ^{η}α&ν-χ^{η}α∃-κλA∃β-βα⊥⌉ν-λE≅⌉ω-να≅" in IPA "I will go back home." in English

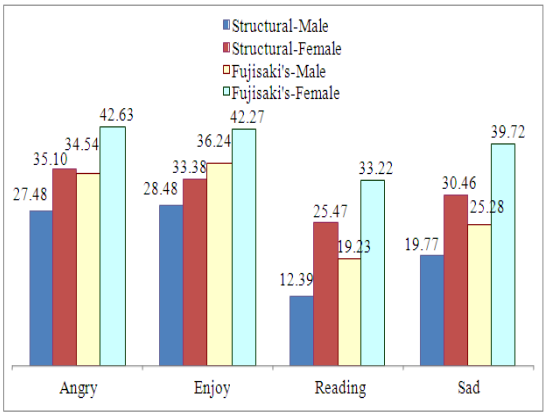

Fig. 6: Averaged RMS error for sentence "κ^{η}υν-ταμ-NAν-σε∃δ-ρ⟩\&⌉-φαN" in IPA "have you finished your work?" in English

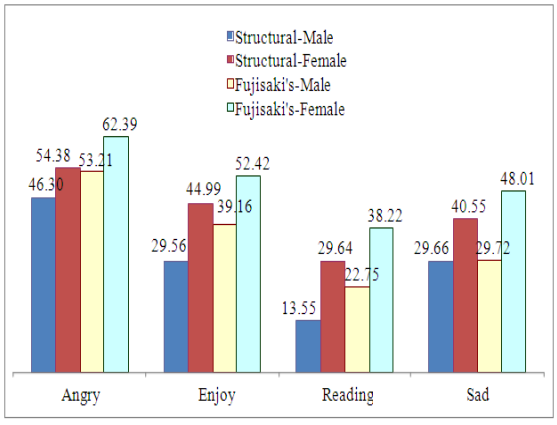

Fig. 9: Averaged RMS error for sentence "χ^{η}α&ν-ρα≅κ-κ^{η}υν-τHι⊥⌉-συ∃τ-ναφ-λο⊥κ" in IPA "I love you most in the world." in English

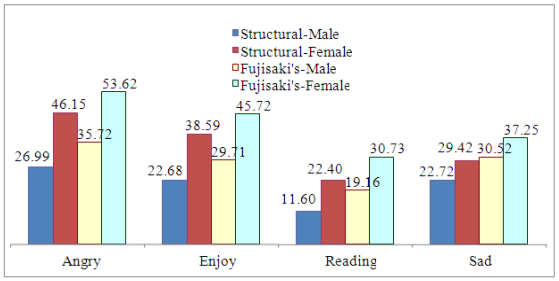

Fig. 10: Averaged RMS error for sentence "ραω-μα⊥φ-δA⊥φ-παφ-τHιο⊥ω-τHα≅-λε˥" in IPA "We do not go to the sea." in English

DISCUSSION

The experimental results in Fig. 6-10 show that the averaged RMS error of the angry speech is the highest level; meanwhile the averaged RMS error of the reading speech is the lowest level. The averaged RMS errors of the happy and sad speech are in the middle level. It can be obviously seen from all Figures that all 5 sentences have the corresponding results. When considering the differences between genders, we found that the averaged RMS error of female speech is above that of male speech. Last but not least, that the RMS error of the Fujisaki's model is mostly higher than that of the structural model for all speech styles. In other words, the structural model gives the better fit for modeling of the F0 contour of the expressive speech than that of the Fujisaki's model. The results of all sentences confirm this observation (Suhartono, 2011; Souleymane *et al*., 2009; Geravanchizadeh and Rezaii, 2009; Camminatiello and Lucademo, 2010; Nicknam *et al*., 2009).

CONCLUSION

This study proposes a comparison of two successful F0 models. First, the model is based on the Fujisaki's model. Second, the model is based on the structural model. The applied speech database consists of male and female speech and each one contains 4 different speech styles including angry style, sad style, enjoyable style and reading style. Five sentences are used for each speech style and each sentence includes 100 samples. From the experimental results, it has been seen that RMS error of each speech style is different from the others for both models. It also reveals that the RMS error of the Fujisaki's model is higher than that of the structural model for all speech styles. In other words, the structural model gives the better fit for modeling of the F0 contour of the expressive speech than that of the Fujisaki's model.

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