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Syllable Specific Unit Selection Cost Function Using a Tone Modeling Technique for Automatic Phonetic Segmentation of Hindi Speech Using HMM

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***Abstract* —This paper presents a technique of improving tone correctness in speech synthesis of a tonal language based on an average-voice model trained with a corpus from nonprofessional speakers speech. Unit selection-based concatenative synthesis is one of the widely used speech synthesis approaches. This approach overcomes the limitations of other synthesis techniques such as articulatory synthesis and formant synthesis.The automatic phoneme segmentation framework evolved imitates the human phoneme segmentation process.Public announcements are generally made by creating a library of a set of prerecorded standard sentences for giving only routine information to the passengers of various transport systems like roadways, aircraft and ships.**

***Keywords* — Text to speech synthesis,target cost,concatentation cost,unit selection, Hidden Markov models.**

I. Introduction

Unit selection-based concatenative synthesis is one of the widely used speech synthesis approaches. In this approach, depending on text input, prerecorded speech segments are selected from a large database and concatenated to produce speech. Unit selection framework selects the most appropriate natural segments of speech using syllable-specific concatenation cost and target cost. The automatic phoneme segmentation framework evolved imitates the human phoneme segmentation process. Public announcements are generally made by creating a library of a set of pre-recorded standard sentences for giving only routine information to the passengers of various transport systems like roadways, aircraft and ships.The goal of this paper is modeling tones to improve the tone correctness of synthetic speech.Quantized F0 symbols in the proposed technique were utilized in two different ways based on phone and sub-phone boundary information.

.II. Background

Festival speech synthesis engine by default uses phones as the basic units. The choice of unit size depends on the characteristics of the language. Hybrid systems have been developed by combining unit selection and statistical parametric synthesis approaches. In order to exploit the merits of the two approaches, hybrid systems have been developed.Hybrid systems have been developed by combining unit selection and statistical parametric synthesis
approaches.In order to exploit the merits of the two approaches, hybrid systems have been developed[1]. A more flexible system that can synthesize speech using only a small amount of automatically labeled speech data from nonprofessional speakers is required to widen the applications of speech synthesis. Introducing generative F0 models is another approach to avoiding the problem with insufficient amounts of training data. T-Tilt model, which is an expansion of Tilt model, was proposed to model the F0 contours of tonal languages[2]. Hidden Markov model is an extension of discrete Markov model in which states are hidden, The hidden Markov models may be used to represent the sequence of sounds within a section of speech.
Phoneme, an elemental speech sound, can be modeled by an individual HMM. In this paper work is done on the automatic phonetic segmentation is to modify an HMM based recognizer to perform the task of phonetic segmentation., in order to reduce manual efforts and speed up the segmentation process[3].

. III. Related Work Done

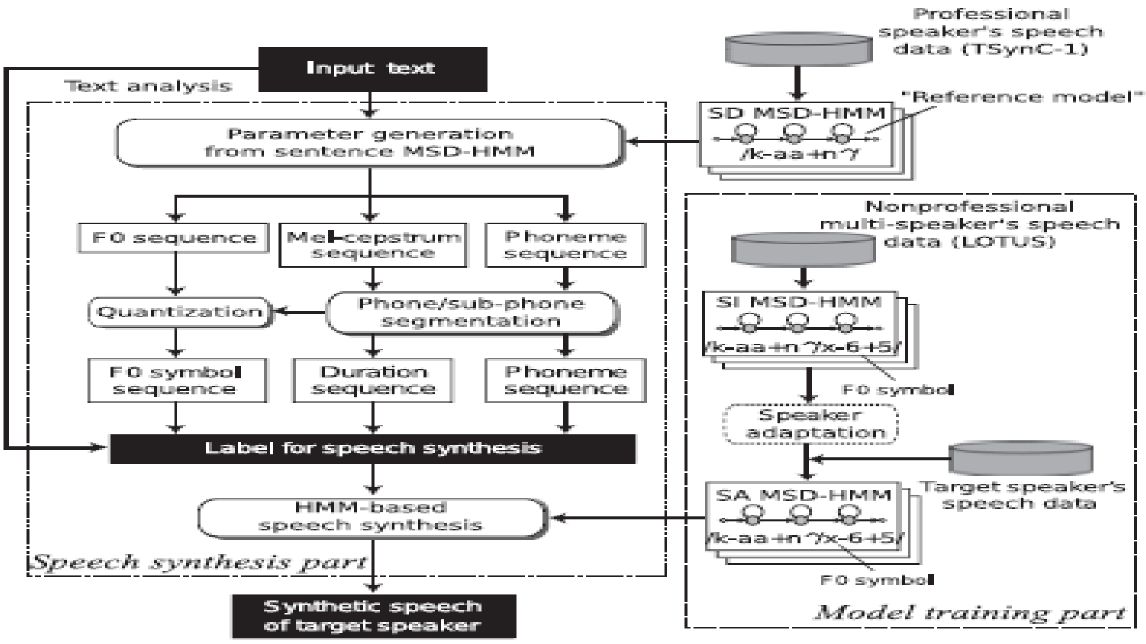
Concatenation cost gives an estimate of the quality of the join between candidate units $ui-1$ and ui for the desired target units. It predicts the degree of perceived discontinuity. Concatenation cost can be split into sub costs, where each of the subcosts, indicates one continuity metric.
In the Festival speech synthesis engine, concatenation cost is calculated based on optimal coupling technique, but the problem in that is the cost of concatenating two units at their best position of join is determined. To overcome this problem the proposed segmental concatenation cost is defined based on the type of segments present at the syllable joins[1]. Two approaches study to obtain the tonal context labels, i.e.,phone-based and sub- phone-based quantized F0 symbols,but the generated phone-based F0 symbol sequence is insufficient to represent correct information on tone contours, so to overcome this problem, propose subphone- based F0 symbols where quantize the F0 contour based on units smaller than phones[2].Hidden Markov model is an extension of discrete Markov model in which states are hidden,The hidden Markov models may be used to represent the sequence of sounds within a section of speech.Performing manual segmentation is considered to be a precise method of obtaining segmented database and information about its contents(labeling). But, the main limitation of this is, manual segmentation is time consuming, labor intensive and prone to error. So, in this paper work is done on the automatic phonetic segmentation is to modify an HMM based recognizer to perform the task of phonetic segmentation., In order to reduce manual efforts and speed up the segmentation process, it is desirable to design an efficient method for automatic phonetic labeling, especially when the size of the speech corpus is large[3].

IV. TONE MODELLING TECHNIQUE

Major methodology works on the waveform generation module or back end of syllable TTS, which selects the units from the database based on a unit selection algorithm selects the units from the database [1]. Two approaches study to obtain the tonal context labels, i.e.,phone-based and sub-phone-based quantized F0 symbols , The generated phone-based F0 symbol sequence is insufficient to represent correct information on tone contours[2].To solve this problem, propose subphone- based F0 symbols where quantize the F0 contour based on units smaller than phones.Performing manual segmentation is considered to be a precise method of obtaining segmented database and information about its contents (labeling). But, the main limitation of this is, manual segmentation is time consuming, labor intensive and prone to error. Most of the previous methods do not elaborate the issue of error analysis, such as what categories of phonemes tend to be error-prone and how to deal with them[3].

V. OVERVIEW OF TTS SYATEM

In this paper Proposed Methodology using Model training & Speech synthesis. In Model training first, an average voice model is trained using multiple data from nonprofessional speakers' speech. The spectrum and F0 are modeled with multi-stream HMMs in which the output distributions for the spectral and F0 parts are modeled using a continuous probability distribution for the former and a multi-space probability distribution. & in speech synthesis Reference model of a professional speaker to generate the proposed labels.When an input text is given, the conventional context labels are automatically generated from the given text by text analysis and a synthetic F0 contour is generated from the reference model using conventional labels shown in fig. The F0 context labels are created using the F0 quantization.fig shows overview of proposed TTS system[2].



VI. ANALYSIS & DISCUSSION

The parser takes the text as the input, extracts the phonemes and arranges them according to the sequence of occurrence in the given sentence, and the list of the phonemes is given as the output. The overall segmentation results in terms of percentage of phonemes falling into different ranges of deviation. It provides the number of tokens occurring for the every phoneme present in the database. The three average-voice models were constructed using different tonal context labels, i.e., conventional, phone-based F0 symbol and sub-phonebased F0 symbol. The reference model was trained using 2500 sentences from the TSynC-1 speech data uttered by a professional speaker, where the conventional tonal context labels were used. The tonal context labeling propose can generate an F0 contour closer to that of natural speech than the other techniques. The test-set consisted of 10 sentences. All syllables present in the test-set were available in the training database. Unit selection cost functions, namely concatenation cost and target cost, are proposed for syllable based synthesis. Concatenation costs are defined based on the type of segments present at the syllable joins.

CONCLUSION

Separate sets of concatenation costs and target costs are proposed appropriate to syllables.Concatenation costs are proposed which is based on the type of segments present at syllable joins. A technique of modeling tones to improve the tone correctness of synthetic speech. Quantized F0 symbols in the proposed technique were utilized in two different ways based on phone and sub-phone boundary information.The phoneme boundary estimate by 36 dimensions HMM, . a comparison has been made against hand-segmented and hand-labeled phonemes.For each of the phoneme occurring in the speech database,the difference between estimated start-end points & actual start-end points was computed and average was calculated. This is called deviation and is used as a measure of the performance of the segmentation.

FUTURE WORK

The proposed cost metrics have reduced perceptual discontinuities at the syllable joins. In addition to linear regression models, different nonlinear models such as ANN(artificial neural network) and SVM(support vector machines)can be explored for further improvement.The segmentation performance,& the better segmentation rates can be obtained by increasing the number of training speech sentences for future work.

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