

Tamil e-Speech

C. Lakshmi Narayanan* & M. Mohamed Waseem[&]

III year B.E. Computer Science and Engineering

Sona College of technology

Salem – 5, Tamil Nadu, India.

*chandrulnarayanan@yahoo.co.in; [&]mohamedwaseemmk@yahoo.com

Abstract

Researches should be carrying out to uplift Tamil in Internet. The language of God should be improved for adapting all type of technical advances. This paper gives the technical support to Tamil in Internet. This compresses the activity of software in Internet for comfortable reading of Tamil. Today, there is a wide spread talk about improvement of the multimedia in computer for the ease of end user, as no longer people want to sit and read data from the monitor, since there is a painstaking effort to be taken, which involves strain to the eyes. In this aspect, text to speech gives more comfort to the user using Internet. The software Tamil Text to Speech is developed to give good Tamil pronunciations for the Tamil users. Use of Tamil text to speech software includes: improving Tamil learning ability; e-learning to the children in e-Tamil; improving the availability of Tamil in computer. This paper describes the activity of the software and its function over Internet. The aim of developing this software is to activate it in the Clint system when it is downloaded (free download) through the website (Tamil website). This free download and the mother tongue content would make the Tamil users uses the website more frequently, good content and good Tamil speech will surely embark the golden emblem to Tamil in Internet.

1. Introduction

This paper is based on Tamil Text To Speech that can be used as voice web. It is a well-known fact that the Information & Communication Technology has been playing a crucial rule in growing the economics of developing countries. Hence a large number of people are being spoken in their first language rather than English it is being developed to improve the Tamil through web. This paper concentrates on speech synthesis methods and algorithm in Tamil TTS software (being developed by us). Algorithm of Tamil TTS is discussed in two methods. First method basis is stored character sound play. The second method involves mathematical calculation to generate sound files in run time. This method uses one base voice, this base voice is used for standard pitch and tone of the speech. Implementation of this method involves various research results and sound manipulation. Audio parameters and similarity of Tamil character pronunciation (research result) is being carried out. Sound

2. Key aspect of Tamil language in speech (concept ATL)

In Tamil each character's pronunciation is independent to the next character. Thus the pronunciation parameter of each character in speech is almost constant.

In speech: $Audioparameter (Tamil\ character) =$
where is a constant

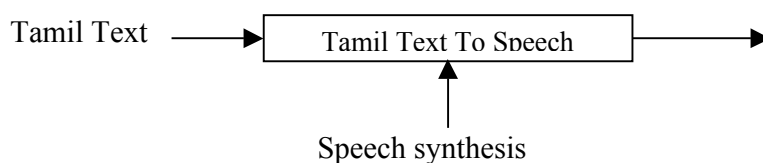
3. System Design:



The website is designed in the way that, the Tamil TTS is attached with the website itself and the user can download the software to convert the WebPage content to Tamil speech.

Note: The design and algorithm of Tamil Text To Speech is mentioning the Tamil TTS software that we are developing.

4. Design of Tamil Text To Speech:



Text is given to the Tamil TTS software. Speech output is synthesized by various speech synthesis techniques.

5. Techniques For Speech Synthesis

- * Waveform Encoding
- * Formant Frequency Synthesis
- * Digital Vocal Tract Modeling

Initially this Tamil TTS software is developed in Waveform Encoding now it is being developed in other speech synthesis techniques.

6. Implementation of Speech Synthesis in Tamil Text To Speech:

6.1 Waveform encoding (existing technique):

This algorithm is based on the concept ATL(explained above). In waveform encoding each character is recorded and played according to input given as text.

6.1.1 Algorithm of Tamil text to speech – by c l narayanan & m m waseem

In this algorithm the time complexity is very much reduced by direct access of sound files without much comparison

Normal comparison:

Best case: 1

Worst case: 247

In our algorithm, comparison is

Best case: 1

Worst case: 3

File storing technique: x_ascii

6.1.2 Algorithm:

Function Tamil_talk_WE(char_in as textfile)

begin

 While (end of char_in)

 begin

 Get each character from char_in

 Find ASCII of char_in(i)

 If I+1 exists then

 begin

 Find ASCII of char_in(I+1)

 end

 If I+2 exists then

 begin

 Find ASCII of char_in(I+2)

 end

 //for one char one sound

 as=char_in(i)

 //for two char one sound

 as = char_in(i)+char_in(I+1)

 //for three char one sound

 as = char_in(i)+char_in(I+1)+char_in(I+2)

 //sound file

 call soundplay function

 end

 end

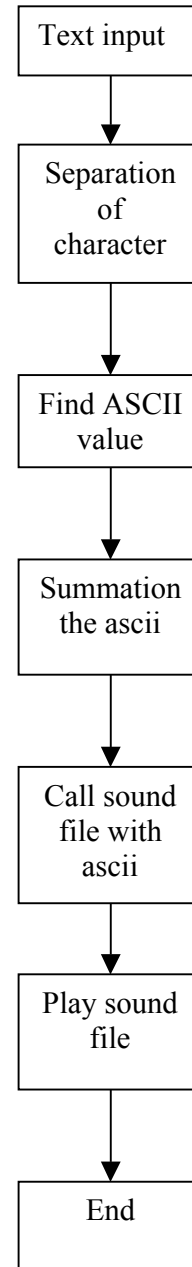
Function soundplay(ascii as integer)

begin

 open soundfile x_ascii

 play soundfile x_ascii

end



7. Other Techniques:

7.1 Formant Frequency Synthesis:

In the second basic technique for speech generation, formant frequency synthesis attempts to replicate the human vocal tract. In this method, bandpass filters are summed together to act as the various audio filters in the oral cavity. Obviously, this method allows the flexibility to utter many different sounds in succession in reduced data storage.

7.2 Digital Vocal Tract Modeling:

The third technique that was mentioned models the human vocal tract digitally. The methods that support this technique are very mathematical in nature, since they map the actions of the human vocal tract to equations. The most prevalent method has been linear predictive coded (LPC) speech.

In linear predictive coded speech, the basic concept is that current information on a speech sample helps to estimate future information on the speech sample.

7.3 Implementing other Speech Synthesis Techniques in Tamil TTS:

We are in research to find out the similarity in Tamil character pronunciation and audio parameters of each character. From these audio parameters we can generate the sound for each character from one base sound. This is done by manipulation of frequency. Individual characters are extracted from the input text. Identification of character involves some procedure, Tamil character have one character one sound, two characters one sound and three characters one sound, thus the individual single sound characters are identified by the following algorithm (from wave form encoding)

```
//for one char one sound
//e.g for sound ka
as=char_in(i)
//for two char one sound
//e.g. for sound kaa
as = char_in(i)+char_in(I+1)
//for three char one sound
//e.g. for sound kho
as = char_in(i)+char_in(I+1)+char_in(I+2)
```

7.4 To generate a sound from the base audio file in frequency manipulation, three modules are involved

- * Transform time domain to frequency domain
- * Modifying sound in frequency domain
- * Synthesis and Transform sound again to time domain

Transform time domain to frequency domain and vice versa is carried out by the tools in Matlab, this is also carried out by the languages C and C++. In Linux QT, C++ can be used.

7.4.1 Modifying sound in Frequency domain:

The base sound is sampled in to different samples S_b of frequency F_s and samples are separated by time $1/F_s$. To separate the frequencies windows are used.

$$\text{Samples} = t * F_s$$

Where t is the small time period formed from stepwise movement of window

There are different types of window Raised cosine window, Blackman window, Kaiser window etc.,

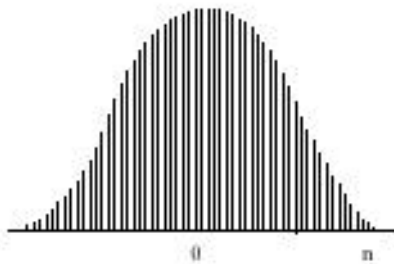
We can use hamming window for good frequency response,

The window sequence is of the form,

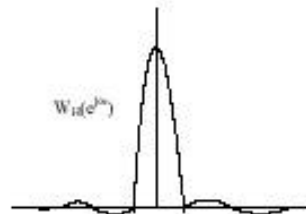
$$w_\alpha(n) = \alpha + (1-\alpha)\cos(2\pi n/N-1) \quad \text{for } -(N-1)/2 \leq n \leq (N-1)/2$$

The equation for Hamming window can be obtained by substituting $\alpha = 0.54$ in the above equation.

$$W_H(e^{j\omega}) = 0.54(\sin(\omega N/2)/\sin(\omega/2)) + 0.23(\sin(\omega N/2 - \pi N/n-1)/\sin(\omega/2 - \pi/N-1)) + 0.23(\sin(\omega N/2 + \pi N/n-1)/\sin(\omega/2 + \pi/N-1))$$



Hamming window sequence



Frequency response of hamming window

Each separations (separated by the time sample t) is converted in to frames by Discrete Fourier Transform (DFT)

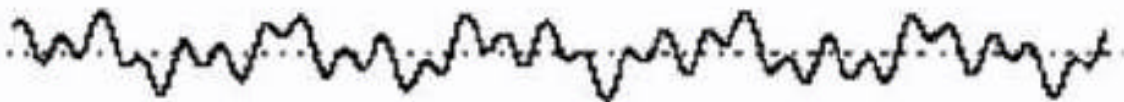
$$X(k) = \sum_{n=0}^{N-1} x(n) e^{-j2\pi kn/N} \quad 0 \leq k \leq N-1$$

Inverse Discrete Fourier Transform (IDFT):

$$x(n) = \frac{1}{N} \sum_{k=0}^{N-1} X(k) e^{j2\pi kn/N} \quad 0 \leq n \leq N-1$$

where $X(K)=F_1(k)$ and $x(n)=S_k(n)$

Base voice



Seperated in to frames

Frame 1

Frame 2

Frame 3



Frame 4



Changing the amp. and freq. according to the audio parameters of extracted chatacter

Changed frames

Frame 1

Frame 2

Frame 3



Frame 4



Combining all frames to get the sound of the extracted character

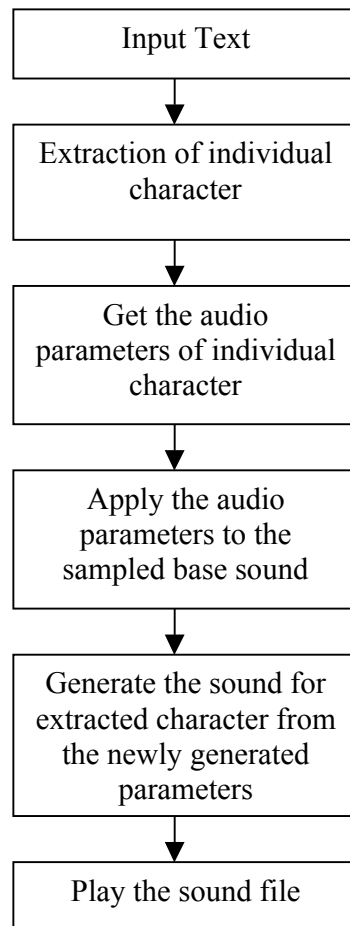


8. Extended Algorithm of Tamil TTS

```

Function main
begin
    Array text
    Get the input text in text
    i=0
    While (end of text)
        begin
            If i+1 exists then
                begin
                    Find ASCII of text(i+1)
                end
            If i+2 exists then
                begin
                    Find ASCII of text(i+2)
                    end
                    //for one char one sound
                    chr = text(i)
                    //for two char one sound
                    chr = text(i)+text(i+1)
                    //for three char one sound
                    chr = text(i)+text(i+1)+text(i+2)
                    call function play_sound(chr)
                end
            end
        end
    end
end

```



```

function play_sound(char chr)
begin
    //call function getAP to get audio parameters for the char. arg. chr
    AP=getAP(chr)
    //call function generate_sound to generate character sound
    generate_sound(AP)
end

```

```

function getAP(char chr)
begin
    get the stored audio parameter
    return audioparameters
end

function generate_sound(Audio parameters AP)
begin
    get the base audio file
    convert the time domain into frequency domain
    sample the base audio file
    convert the samples to frames by DFT
    change the audioparameters of frames with AP
    apply the changed audioparameters to each frames
    apply IDFT and combine the samples
    convert the frequency domain to time domain
    play the audio file
end

```

Using waveform encoding the size of the audio files is increased. In extended technique of Tamil TTS algorithm the size of the audio files is very much decreased. Optimizations in fluency and response time have to be done.

9. Future enhancement:

9.1 Making speech with expression

To get expression special character is added before the sentence/paragraph Process:

By inferring the special character the predefined audioparameters of each special character is mixed with the existing audioparameters of the Tamil characters and play the character sound. This would give some expressions.

10. Applications:

Some of the applications of this Tamil text to speech is:

10.1 PC & Multimedia:

Screen reading, language and literacy training.

10.2 Industrial:

Voice web, speaking alarm system and announcement systems.

10.3 Assisting:

Reading aids for the blinds, communication aids for vocally impaired people

11. Conclusion:

This is functioned in voice web. Around the world a lot of Tamil web users are there, but not all the Tamil web users are accessing Tamil websites, because they may not be comfort with reading Tamil, by making Tamil text to speech in web would develop the accessing of Tamil websites and developing Tamil in web users. This will also enhances e-learning. The Tamil TTS gives the foundation for the future generations to involve more researches in e-Tamil

Currently the software is developed in waveform encoding. By applying the methods as mentioned above will decrease the space very much but the optimization of the response time is needed. Researches is being carried out in above said methods and implementing expressions in speech.

Bibliography:

- [1] John G. Proakis and Dimitus G.Manolakis,"digital signal processing principle, Algorithms and Applications, Prentice Hall of India, New Delhi 3rd edition, 2002.
- [2] Speech and Language Processing – An introduction to Natural Language Processing, Computational Linguistics and Speech Recognition. Prentice Hall, NJ, USA. Jurafsky Daniel, Martin H. James, 2000.
- [3] "Speech synthesis models: a review", by Breen A, Electronics & Communication Engineering Journal, v. 4(#1), pp. 19-31, Feb. 1992.

Related links:

www.cda.co.za
www.festvox.org
www.tcts.fpms.ac.be
www.ncvs.org
www.kuralosai.com

TTS Window Design

