Extraction of Speech Parameters from Speech Database using Festival

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ABSTRACT
Speech synthesis is the process of production of artificial speech. The system used for generation of speech from text is called as text-to-speech (TTS) system. In TTS system, text and voice models for a particular language or multiple languages are given as input to the system, which generates speech as output corresponding to the provided voice models. Speech synthesis systems can be extremely useful to people who are visually challenged, visually impaired and illiterate to get into the mainstream society. More recent applications include spoken dialogue systems and communicative robots. HMM (Hidden Markov Model) based Speech synthesis is the emerging technology for TTS. HMM based speech synthesis system consists of training phase and synthesis phase. In the training part, phone and excitation parameters are extracted from speech database and modeled by context dependent HMMs. In synthesis part, the system will extract the suitable phone and excitation parameters from the previously trained models and generates the speech. The main objective of this project is to build an HMM based speech synthesis system. In the training process, the system uses HTK (Hidden Markov Model Tool Kit) and SPTK (Signal Processing Tool Kit) developed at Cambridge University and Tokyo Institute of Technology respectively. Synthesis part is be done by „Festival". Festival is a speech synthesis tool for the generation of speech and it is language independent which is developed at the University of Edinburgh. The main advantage of this approach is its flexibility in changing speaker identities, emotions and speaking styles.

Keywords
HMM (Hidden Markov Model) , SPTK, HTK.

1. INTRODUCTION
HTS system consists of training part and synthesis part. In the training part, speech parameters are extracted from the speech database [1]. As training part is the heart of the HTS system, speech database has an important role in the implementation or design of the system [2]. A few software installations were required in order to create a working environment for implementation of this system [3]. HTS is the back-end part of the system. It does not contain text normalization module. So, in order to generate the speech, the HMM models are given to the front-end part of the system [4].

This Research paper explains the process undertaken to develop a speech synthesis system using Hidden Markov Models (HMM) [5]. It gives the overall information about the process starting from the selection of sentences for reading and the recording process. Further, gives the software tools required for the development of HMM Based Speech Synthesis System and also discusses the step by step procedure for the implementation of the system.

2. SPEECH DATABASE
In the development of HMM based speech synthesis system, the speech database is given to the training part of the system for the extraction of speech parameters. The speech database includes recorded wave files and corresponding utterances of the recorded wave files [6-8]. The sentences to be recorded are selected in a way such that the text should cover maximum all phone symbols belonging to that particular language. In this thesis, for the implementation of HTS system, an Indic-Marathi prompts are taken for recording which is prepared by the people of Speech Vision Lab.To record speech files for the chosen text, the speaker should be selected in such a way that he or she can speak with pleasant voice [9]. The speaker should not suffer from bad health condition like cold, cough etc. The selected speaker should have some awareness about the selected prompts to avoid mistakes in the recording process [10-25]. To record an efficient and noise-less speech database, high quality recording equipment is required. A noise less environment like a specially designed recording studio is required to avoid background noise while recording the speech files. Recording is done using a builtin laptop microphone. The speech files are recorded in noise-less clean environment. A high quality microphone and recorder are used for recording. The distance from the microphone to mouth, speaking volume and speaking style is kept constant till the completion of recording. A MATLAB program is used while recording for the generation of single wav files. At the starting and ending of sentence recording, a key is pressed such that the respective recorded speech file will be stored with the given name as a single wave file.Care should be taken to avoid mistakes in utterances due to wrong pronunciation or repeated pronunciation of a word while recording [11]. Any mistakes while recording can be recovered either by re-recording those utterances or by correcting the corresponding sentence to suit the sentence [12]. The recorded speech files are stored in wave format (.wav) [13-25].

3. TOOLS
Some software installations are required for the development of HMM based speech synthesis system. This HTS system was build on laptop with Ubuntu 14.04 as operating system on a 32-bit machine and can be build on any version with the supporting library files. These tools can be found at their respective sites [26-38].

3.1 List of required tools
The tools used for the implementation of HTS system are listed below:

- speech_tools-2.1-release.tar.gz
- festival-2.1.tar.gz
- festvox- 2.1-release.tar.gz
- HTK-3.4.1.tar.gz

- Hdecode-3.4.1.tar.gz
- HTS-2.2_for HTK-3.4.1.tar.gz
- SPTK-3.4.1.tar.gz
- hts_engine_API-1.05.tar.gz
- ActiveTcl8.4.19.6.295590-linux-ix86.tar.gz
- HTS-demo_CMU-ARCTIC-SLT.tar.bz2

4. IMPLEMENTATION OF HTS SYSTEM

Create a directory or folder on the desktop and move to that directory.

- mkdir directory_name
- cd directory_name

Copy all the required downloaded tools into the directory_name folder [4][5].

4.1 Installation of speech tools, festival and festvox

To build HMM models, the utterance files which consist of textual features and the duration of each unit in the text to be synthesized are required. To generate these utterance structures, speech tools, festival and festvox are needed. Run the commands given, for the installation of speech tools, festival and festvox respectively [4][5].

- Installation of speech tools
- tar –xvf speech_tools-2.1-release.tar.gz
- cd speech tools
- ./configure
- Make

The first command is for extraction tar files. The most popular archiving tool used in UNIX and Linux is the “tar” command. The second command gives the path to the speech files folder. Next command is for running the configure script. “Configure” is an executable script designed to help in developing a program to be run on computers and matches the libraries on the user’s computer, with those required by the program, before compiling it from its source code. By running this command, it will check some details about the machine on which the software is going to be installed and creates the Makefile to be used in the next step. The last command make is a general purpose workflow program, usually used for compilation [4][5].

- If error occurs during the execution of make command, follow the given steps:
- In speech_tools/include/EST_Titerrator, at line number 212:7 and 292:7, insert this-> and save.
- In speech_tools/include/EST_TNamedEnum.h, at line no 133:64, insert this-> and save.
- In speech_tools/base_class/EST_Tsimplematrix.cc, at line no 132:4, 130:11, and 101:4 insert this-> before set_values, just_resize.
- Add #include <string.h> in the header file section as this program consists of memcpy function.
- Inspeech_tools/base_class/EST_Tsimplevector.cc, at line no 74:7, insert this-> before just_resize. Add #inclue<string.h> in the header files section because this program consists of memset function.

- Run make

Installation of Festival

- cd..
- tar –xvf festival-2.1.tar.gz
- cd festival
- ./configure
- make

Installation of Festvox

- cd..
- tar -xvf festvox- 2.1-release.tar.gz
- cd festvox
- ./configure
- Make

4.2 Installation of HMM Toolkit (HTK) and Patch for HTS

HTK along with patch files provided for HTS are used to train contextindependent and context-dependent hidden Markov models. The procedure for installing these tools is given below:

- cd..
- tar –xvf HTK-3.4.1.tar.gz
- tar –xvf Hdecode-3.4.1.tar.gz
- tar –xvf HTS-2.2_for HTK-3.4.1.tar.gz

(Copy all the extracted files into a folder and name it as hts_patch)

- cd htk

Run the command given below to include a patch file for HTS.

- patch -p1 -d . < ../hts_patch/HTS-2.2_for_HTK-

4.3 PATCH

The executable such as Hcopy, HList, HInit, will be compiled in /usr/local/HTS-2.2beta/bin

- ./configure
- make
- sudo make install
- sudo make hlmtools install-hlmtools
- sudo make hdecode install-hdecode

4.4 Installation of HTS Engine

This synthesize speech waveform from trained HMMs

- cd..
- tar –xvf hts_engine_API-1.05.tar.gz
- cd hts_engine_API-1.05
- ./configure
- make
- sudo make install

4.5 Installation of SPTK (Signal Processing Toolkit)

The SPTK functions such as mgecp, x2x, lsp2lpc, etc will be compiled in /usr/local/SPTK/bin

- cd..
- tar –xvf SPTK-3.4.1.tar.gz
- ./configure
- make
- sudo make install

4.6 Installation of Active Tcl

Execute the following commands to install Active Tcl in /usr/local/Active Tcl-8.4/bin/

- cd ..
- tar –xvf ActiveTcl8.4.19.6.295590-linux-ix86.tar.gz
- cd ActiveTcl8.4.19.6.295590-linux-ix86
- sudo ./install.sh

In the window that appears, type /usr/local/ActiveTcl-8.4

5. GENERATION OF RAW FILES AND UTTERANCES

i) Generation of raw files from the wav files using ch_wave

$ESTDIR/bin/ch_wave -otype raw -F 16000 wav/$i.wav -o directory_name/HTS-demo_CMU-ARCTIC-SLT/data/raw/db_v_$i.raw ii) Generation of utterance files

Perform the steps given to set the environment variables for speech tools, festival and festvox directories respectively. These variables point to the paths where speech tools, festival and festvox tools are located.

- export ESTDIR=/home/user_name/Desktop/directory_
- name/ speech_tools
- export FESTDIR=/home/user_name/Desktop/directory_
- name/ festival
- export FESTVOXDIR=/home/user_name/Desktop/
- directory_name/ festvox

To start a new project, a new folder is to be created. The folder to be created has to follow a particular standard which is given below. Create a directory in „directory_name" folder. The name of the folder should be in the format as shown below:

- mkdir institutename_dictonarylanguage_speaker name_type
- (ex. bamu_sangram_indic)
- Now move to the created directory to make setup. For this process, run the commands given below:
- Cd institutename_dictonarylanguage_speakername_typ e
- $FESTVOXDIR/src/clustergen/setup_cg

bamu sangram indic

After execution of the commands, required folders will be created and required shell script files will be copied in to the newly created directory. Next, we need to generate prompt file corresponding to the recorded speech wave files. It consists of file id followed by transcription. Transcription is quoted with double quotes and entire line is enclosed in brackets. As an example, the structure of prompt file is shown.

(mar0001 " kaarand~a aapalayaakad:ei tii padadhata naahii.")

The prompt file is copied into the etc folder in the current directory for the input wave files and it is renamed as txt.done.data. Next, copy the recorded speech files which are in the .wav format into the recording folder which is already created in the current directory. These wav files which are in the recording folder may not be compatible for labeling. So, these wav files must be modified into the required format. For this modification, a script file exists in bin folder in current directory. To execute this file script, run the command which is given below:

./bin/get_wavs recording/*.wav

Wave files are extracted one by one from the recording folder and given to the ch_wave executable file which is part of the speech tools in exported environment variable. This executable file takes the wav as input and converts the wav file to required format; here the scripts require a wav file with 16000 HZ sampling frequency and mono sound. The output wave is stored in wav folder which is in the current directory. Next, prompt files are to be generated for each and every speech file present in wav folder using txt.done.data prompt file which is in etc folder. This can be done by using script called do_build

./bin/do_build build_prompts

This generates prompt-wav files, prompt-lab files and prompt-utts files in existing folders. Now, the directory is ready to make labeling. The labeling is done by using EHMM labeler. An EHMM labeler is designed by festvox developing group. An EHMM labeler is available with festvox 2.1 release. The labeling procedure can be done by executing the command given below [4] [5].

./bin/do_build label

if error occurs during execution of this command, change the directory path at 148th line in do_ehmm script. ($FESTVOXDIR/src/ehmm/bin/do_ehmm) Here, the phones present in speech files are labeled and these labels are stored in lab folder. These generated labels are used to create utterances for speech files. Run the below command for the generation of utterances[4] [5].

./bin/do_build build_utts

The generated utterance files are in the format that can be used by the HTSdemo_CMU-ARCTIC-SLT and these utterances are stored in folder named utts which is in the festival directory.

5.1 Installation of HTS-demo_CMU-ARCTIC-SLT

The HTS-demo is unpacked using the tar command as given below:

- cd ..
- tar -xvf HTS-demo_CMU-ARCTIC-SLT.tar.bz2

Delete all files on the paths data/raw and data/utts. Next, copy the raw files and utterances which are generated in the above steps into the data/raw and data/utts folders. While copying, it should be noted that the raw files and utterances should be in the format given below respectively.

Raw files: db_v_mar0001.raw

Utterances: db_v_mar0001.utt

Where 'db' refers to the data set and 'v' refers to the speaker name.

All the necessary files are in the right places in their respective folders. The first stage in running the HTS-demo scripts is to configure it. Configuring the HTS-demo gives a guideline on how to set up these paths and alter certain parameters. It also indicates the packages required to run the HTS-demo. The command used to configure is given below:

- cd HTS-demo_CMU-ARCTIC-SLT

Setup HTS-demo_CMU-ARCTIC-SLT by running configure script given below:

./configure--with-tcl-search-path=usr/local/ActiveTcl-8.4/bin--with-fest-search- path=/home/

idntyti/Desktop/HTS/festival/examples--withsptk-search-path=/usr/local/SPTK/bin--with-hts-searchpath=/usr/local/HTS-2.2beta/bin--with-hts-engine-searchpath=/home/idntyti/Desktop /HTS/hts_engine_API-1.05/bin

There are certain files which need to be modified in order to run the demo successfully. A slight modification is to be done in the file Makefile found in the

/data/ which is in the main HTS-demo_CMU-ARCTIC-SLT directory.

makefile #setting

SPEAKER=v

DATASET=db

This modification depends on how the raw file (db_v_mar0001.raw) and utterance (db_v_mar0001.utt) are named. The speech parameters have to be changed according to recorded database in the Makefile, which is in the data folder in the

HTS-demo_CMU-ARCTIC-SLT/data/Makefile as given below: # speech analysis conditions

SAMPFREQ = 16000 # Sampling frequency (48 kHz)

FRAMELEN = 400 # Frame length in point (1200 = 48000 * 0.025)

FRAMESHIFT = 80 # Frame shift in point (240 = 48000 * 0.005)

Perform make command as given below with which HTS-demo runs: - make

With this make command; the script in the Makefile is executed and all the speech parameters required are extracted from the raw files.

Step by step execution procedure of Makefile:

- Extraction of mel generalized cepstral coefficients (mgc) from raw files. These coefficients are stored in HTS-demo_CMU-ARCTIC-SLT/data/mgc.
- Extraction of log-fundamental frequencies (lf0) from raw files. These parameters are stored in HTS-demo_CMU-ARCTIC-SLT/data/lf0.
- Generation of training data files from mgc and lf0 files. These files are stored in HTS-demo_CMU-ARCTIC-SLT/data/cmp.
- Extraction of monophone and full context labels from utterance files. Monophone
- labels are stored in HTS-demo_CMU-ARCTIC-SLT/data/labels/mono and full context labels are formed in HTS-demo_CMU-ARCTIC-SLT/data/labels/full.
- Generation of full context model list file. This file is stored in HTS-demo_CMU-ARCTIC-SLT/data/lists/full.list.
- Generation of full context model list file which includes unseen models and this file is in HTS-demo_CMU-ARCTIC-SLT/data/lists/full_all.list.
- Generation of monophone list file. This file is stored in HTS-demo_CMU- ARCTIC-SLT/data/lists/mono.list.

Generation of training data script and generation label script. These scripts are

- stored in HTS-demo_CMU-ARCTIC-SLT/data/scp/train.scp and HTS-demo_CMU-ARCTIC-SLT/data/scp/gen.scp respectively.

Once the features and the context-dependent label files are obtained, contextdependent models are built and speech is synthesized by the following command. Config.pm contains a list of parameters such as the stream weights, feature order, etc., along with the model structure, file locations, and HTS commands and options.

- perl scripts/Training.pl scripts/Config.pm

With the execution of this command, in order to synthesize, the necessary files are formed in their respective folders.

Step by step execution procedure of Training.pl script is given below:

- Creation of required directories namely configs, models, stats, edfiles, trees, gv, voices, gen and proto in HTS-demo_CMUARCTIC-SLT.
- Generation of set of configuration files used by different modules in the HTS-demo_CMU-ARCTIC-SLT/configs.
- Creation of prototype of an HMM. The prototype is created in HTS-demo_CMU-ARCTIC-SLT/proto. Monophone (context-independent) models for each phone in the database are to be trained.

The synthesized wave files can be played using the play command. The waveform can be viewed using wave surfer. The properties of the synthesized speech can be viewed by using ch_wave. The models generated in the voice folder are used to generate speech for a given text when these models are provided to the front-end part of the synthesis system.

6. GENERATION OF SPEECH WITH HMM MODELS FOR A GIVEN TEXT

HTS is the backend of the synthesis system. Using only the HTS-demo scripts, speech cannot be synthesized from the given text. We need to plug HTS to a front-end. Festival can be used as front-end. Download the tar files from the link given below:

http://homepages.inf.ed.ac.uk/jyamagis/release/MarathiHTSVoices-ver1.0.tar.gz

Copy the downloaded tar file into the directory_name folder. Follow the given commands to extract the downloaded tar file.

- cd..
- tar –xvf MarathiHTSVoices-ver1.0.tar.gz

In the folder named MarathiHTSVoices, replace the models in the hts_voice_cmu_us_arctic_slt-1.03.tar.gz and hts_voice_cmu_us_arctic_slt-1.03.tar.gz with the models generated in the voice folder. Next, run the below commands for the extraction of tar files present in the folder.

- cd MarathiHTSVoices
- sh do_build

After execution of this command, a new folder "build" is formed which contains the extracted files. A text document named "example.txt" exists in MarathiHTSVoices folder for the purpose of giving text as input. So, the text input is given through this document and have to save it. Replace the sampling frequency in runvoice.sh file by 16000 and frame rate by 80. After that, run the command given below:

- ./runvoice.sh

After execution of this command, wave files are generated in the MarathiHTSVoices folder. Finally, the synthesized waveforms can be played by using the play command.

7. CONCLUSION

The process undertaken to develop a speech synthesis system using hidden Markov models is explained in detail. The overall information about the process, starting from the selection of sentences is given clearly in this chapter. At last, the procedure for implementation of HTS system is described in step by step procedure.

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