

DEEP LEARNING TECHNIQUES FOR FEATURE BASED RANKING OF AUDIO SIGNALS IN SEARCH ENGINES

Submitted in partial fulfillment of the requirements for the award of Bachelor of
Engineering degree in Electronics and Communication Engineering

by

SCINDHIA. M (37130363)



DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

SCHOOL OF ELECTRICAL AND ELECTRONICS

SATHYABAMA

INSTITUTE OF SCIENCE AND TECHNOLOGY

(DEEMED TO BE UNIVERSITY)

Accredited with Grade “A” by NAAC

JEPPIAAR NAGAR, RAJIV GANDHI SALAI, CHENNAI - 600 119

APRIL – 2021



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www.sathyabama.ac.in



DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

BONAFIDE CERTIFICATE

This is to certify that this Project Report is the bonafide work of and **SCINDHIA.M(37130363)** who carried out the project entitled “DEEP LEARNING TECHNIQUES FOR FEATURE BASED RANKING OF AUDIO SIGNAL IN SEARCH ENGINES” under my supervision from October 2020 to April 2021.

Internal Guide

Dr. N. M. Nandhitha; M.E., Ph.D., Dean

Head of the department

Dr.T.Ravi. M.E., Ph.D.,

Submitted for Viva Voce Examination held on _____

Internal Examiner

External Examiner

DECLARATION

I **SCINDHIA.M** (Reg.No: 37130363) hereby declare that the Project Report entitled **“DEEP LEARNING TECHNIQUES FOR FEATUREBASED RANKING OF AUDIO SIGNALS IN SEARCH ENGINES”** done by us under the supervision of **Dr. N.M.NANDHITHA. M.E., Ph.D., Dean** is submitted in partial fulfillment of the requirements for the award of Bachelor of Engineering degree in **ELECTRONICS AND COMMUNICATION ENGINEERING**.

DATE :

1)

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2)

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ABSTRACT

The need for audio authentication has become mandatory in the modern epoch. Audio recognition is also known as speech recognition, and voice recognition is a computer software programme or hardware system that can decode the human voice. Voice recognition is a method of controlling a computer, issuing instructions, or writing without the use of a keyboard, mouse, or buttons. Distance metrics such as Manhattan distance, Euclidean distance, and Maximum distance were taken into account in the previous scheme. The obtained distances are sorted and ranked in ascending order. The best match has the shortest distance and is ranked first. In the proposed system the audio information is pre-analyzed and the unique feature points are stored in the database. The deep learning technique is used in order to rank the audio signal from the speakers based on the quality of voice signal and error rat

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CHAPTER 1

INTRODUCTION

Discourse acknowledgment is an interdisciplinary branch of computational etymology that focuses on developing strategies and innovations that enable (programmed discourse acknowledgment) and PC discourse acknowledgment are other names for it. Phonetics, software engineering, and electrical engineering data and analysis are combined in this project.

Some discourse recognition systems necessitate "preparing" (also known as "enlistment"), which entails an individual speaker understanding text or secluded jargon and incorporating it into the system. "Speaker autonomous" frameworks are those that don't use any kind of preparation. "Speaker subordinate" frameworks are those that make use of preparation.

Voice UIs include voice dialling (for example, "call home"), call guiding (for example, "I may want to make a gathering decision"), domotic machine control, search catchphrases (for example, "find a webcast where unique words were verbally expressed"), simple knowledge passage (for example, entering a Visa number), and the readiness of structured data (for example, entering a Visa number). "Voice affirmation" or "speaker distinguishing proof" alludes to perceiving the speaker as opposed to the substance of their appearance. As far as creative mind, talk appreciation has a long history, with a couple of outstanding advances. Additionally by broad business appropriation of a scope of profound learning strategies in the planning and conveyance of talk affirmation structures.

Hidden Markov Models are mathematical models that produce a series of images or numbers. Since a discourse sign may be viewed as a piecewise fixed sign or a brief timeframe fixed sign, they are used in discourse

recognition. For certain stochastic reasons, discourse can be viewed as a Markov model.

Another factor HMMs are notable is that they are not difficult to make and use, just as being basic and computationally practical. Several frameworks employ ostensibly discriminative preparing techniques that avoid a straightforward factual approach to HMM boundary evaluation in favour of enhancing some order-related proportion of the preparation data. Most extreme shared data (MMI), least characterization error (MCE), and least telephone error are the models (MPE).

The Viterbi estimation will no doubt be utilized to detangle the talk (what happens when the structure is given another articulation and should enlist the most probable source sentence). A potential improvement to disentangling is to keep a group of good up-and-comers rather than only the best, and to rate these great competitors using a superior scoring skill (re scoring) so that we can select the best one based on this refined ranking.

DYNAMIC TIME WRAPING (DTW)

Dynamic time travel was once widely used for discourse recognition, but it has since been largely replaced by more effective HMM-based methodology. Dynamic time travel was a method of estimating the similarity of two groups that may vary in terms of schedule or speed. Similes in strolling examples, for example, will be known regardless of whether the individual was strolling slowly in one video and quickly in another, or whether there were speed increases and decelerations in one perception. DTW has been used to investigate film, sound, and designs, but it can be used to investigate any knowledge that can be turned into a clear representation.

Programmable discourse acknowledgment is a notable application that adapts to different talking rates. That is, the groupings are "distorted" to coordinate with one another in a non-direct manner. When it comes to covered up Markov models, this grouping structure technique is commonly used .

NEURAL NETWORKS

Neural organizations have a few characteristics that make them appealing acknowledgment models for discourse acknowledgment. Early neural organizations, despite their limited ability to display transient conditions, were sometimes useful for constant recognition activities, despite their feasibility in ordering brief timeframe units. Be that as it may, in this field, LSTM and related dreary neural associations (RNNs) and Time Delay Neural Networks (TDNNs) have as of late showed improved execution.

DNN models construct compositional models in which extra layers engage bits of lower-layer features, bringing about an enormous learning limit and the capacity to show complex instances of talk content. Microsoft Research has published a new Springer book, you'll find detailed audits of this turn of events, as well as the best in class as of October 2014. In recent outline posts, you can also read about the linked basis of programmed discourse acknowledgement and the impact of various AI ideal models, including profound learning.

One of the most important rules of profound learning is to avoid hand-drawn highlight designs and instead use rough highlights. This rule was demonstrated to be superior to Mel-Cepstral highlights, which use a couple of fixed change phases from spectrograms, in the design of a profound auto encoder using "raw" spectrogram or direct channel bank highlights. Waveforms, the true "rough" highlights of debate, have recently gotten a lot more attention, and they've been shown to deliver fantastic larger-scale dialogue acknowledgment outcomes.

END TO END AUTOMATIC SPEECH RECOGNITION

Together, start to finish models achieve proficiency with all of the components of the discourse recognizer. This is important because it affects the planning period and measurement. For example, all HMM-based frameworks require an n-gram language model, and one of the most common n-gram language

models can take several gigabytes of memory, making submission on a cell phone impractical. As a result, rather than the device being deployed locally, existing business ASR applications.

APPLICATIONS

Accordingly, CTC models can undoubtedly sort out some way to design talk acoustics for English characters, however they commit a great deal of fundamental spelling errors and need to depend on another dialect model to tidy up the papers. At that point, utilizing cosmically huge datasets, based on the work and exhibited some business footing in Chinese Mandarin and English. [Lip Net, the first start-to-finish sentence-level lip comprehension model, was released in 2016, and it used spatiotemporal convolutions and RNN-CTC engineering to outperform human-level execution in a restricted syntax dataset. Google Deep Mind launched CNN-RNN-CTC engineering with a huge reach in 2018. TAS stands for "Tune in, Attend, and Spell," and it is a model that "tunes in" to the acoustic sign, "considers" various sections of the sign, and "spells" out the record character by character. The ability to recognize voices varies depending on the make and model of the car. Real language discourse comprehension, rather than a preset series of instructions, is used in some of the more recent vehicle models. In the clinical field, talk affirmation might be performed at the front-end or back-finish of the clinical announcing stage. The despot is responsible for changing and affirming the record, which the provider guides into a talk affirmation motor. The voice is steered into a discourse acknowledgment unit by the supplier, and the perceived draught record is steered to the editorial manager alongside the first voice text, where the draught is changed and the report resolved. Conceded dialogue recognition is currently widely used in company.

The way that most EHRs haven't been explicitly altered to exploit voice-acknowledgment capacities is a more significant issue. The clinician's communication with the EHR is profoundly dependent on the reassurance and mouse, and course through the UI utilizing menus and tab/button clicks is a

significant piece of that. Voice-based course gives just slight ergonomic advantages. Then again, an enormous number of as of late overhauled radiology or pathology correspondence systems use voice "macros," in which the utilization of explicit articulations For instance, "ordinary report" will fill in countless default esteems and furthermore make a standard, which would shift dependent on the sort of assessment – for instance, a chest X-beam versus a gastrointestinal inconsistency game plan for radiology structure.

The results are promising, and the paper also provides material to the innovative work field for considering clinical documentation and vocabulary, as well as exhibition benchmarks and some management programming . The Mirage aeroplane programme in France, and various programmes in the United Kingdom handling a variety of aeroplane stages have all been noteworthy. Discourse recognizers have been successfully used in competitor aeroplanes in these programmes. Unlike what would be anticipated, no negative effects of the speakers' sloppy English were discovered. It was clear that unrestricted speech had fucked up the recognizer, which was to be expected. A limited jargon, or more every one of the, an appropriate grammar, could consequently be relied upon to improve acknowledgment precision generously.

The framework isn't utilized for a particular wellbeing or weapon-related errands, like weapon conveyance or bringing down the underside, yet it is utilized for a wide scope of other cockpit capacities. Visual and aural criticism are utilized to affirm voice orders. The framework is viewed as a critical arrangement for decreasing pilot responsibility, as it permits the pilot to disseminate centers to his plane with only two straightforward voice orders or to every one of his accomplices with only five.

In principle, Air regulator errands are additionally depicted as the critical yield of the regulator by exceptionally structured discourse, so reducing the trouble of the discourse acknowledgment undertaking should be possible. In certain cases, however, this is not the case. The FAA record 7110.65 clarifies the

expressions that air traffic controllers can use. Although the number of expressions supported by one of the reproduction merchants discourse acknowledgment frameworks is less than 150.

Regardless of the irrefutable level of coordination with word taking care of in every day individualized registering, ASR has not seen the standard extensions utilized in record improvement. Discourse recognition has become more affordable in mobile phones as portable processing speeds have improved.

PROBLEM DEFINITION

From the various applications, it is found that an efficient speech ranking system has to be established. The proposed work aims at developing an efficient speech ranking system using Convolutional neural networks. The performance of the proposed system is also to be measured in terms of metrics.

OBJECTIVE

The objectives are detailed as follows:

- To acquire the speech signals and preprocess the signals
- To identify a suitable forward transform and obtain the co-efficient
- To aggregate the co-efficient using appropriate statistical parameters
- To identify a suitable deep learning technique for classifying the speakers and hence perform the ranking
- To analyze the performance of the proposed classified

ORGANIZATION OF THE REPORT

This report is organized as follows: Section 2 provides the literature survey. Research database is given in Chapter 3. Proposed methodology is discussed in Chapter 4 and Chapter 5 concludes the research work.

CHAPTER 2

LITERATURE SURVEY

An exhaustive literature survey and the basic system design concepts are explained in this chapter. According to Neha Jain and Somya Rastogi (2020), in this country, people come with their natural ability to understand new things. Suppose people have to communicate by writing letters to each other, it's a real pain. And that is how people communicate with machines, because it would be so much better if computers could hear what people are thinking and if they could speak to them just to do stuff, and that people needed a very strong speech recognition system. Speech Recognition is a technique that can be used for the generation of text using a computer to recognize terms and phrases. The Speech Recognition System uses sound modeling and language modeling methods popularly called. Acoustic modeling constitutes the mathematical connection between the audio signal and phoneme linguistic segments, on the other hand language modeling describes the distribution of likelihood of words segments in the given word series. •Accuracy (percentage of speech and sentence conversion errors in text) •Speed (the extent to which the machine can keep up the pace with a human speaker) A device for speaking recognition is something you've dreamt of and been working on over decade. There are a range of software items that allow users to dictate their programmes and translate terms and phrases to text in a word processing document. Human beings will also use their speech orders to access control commands such as file opening, menu access, etc. Speech Recognition Technology will also assisted many individuals with disabilities who at times cannot write and now provide systems for speech recognition. But now people will just say those words to get what they need. And a kind of speech recognition script that implements speech recognition technologies, the mechanism that makes this possible.

The first Acoustic Mechanical Speech Machine was developed in 1784 by a student in Vienna (De Liang Wang, Jitong Chen, 2020) . Thus the first dictation system was invented in 1879 by Thomas Edison. Following the chain, Bell Laboratories in 1952 created the next method for the understanding of voice, which was capable of acquiring speaking numbers of 90%, but could only acknowledge the numbers spoken by the inventor. In 1970 a scholar was able to recognise more than 1000 speeches and could distinguish various pronunciations and sentences using the Harpy System. The recognition of speaking went on in the 80s with the advent of the Hidden Markov Model, which used a more mathematical approach in the analysis of sound waves. In 1986, the Hidden Markov model, which could be predictable in voice, was inaugurated in the 80's by IBM Tangora. In 2006, the NSA began to use speech recognition technology in the recorded speech to segment keywords. Then there was a surge in speech recognition technologies as Facebook, Google, Google Amazon, Microsoft and Apple, the world's top IT firms, began to sell this feature in different devices through services such as Google Home Amazon Echo and Apple. The objective of these leading technology firms is to have more exact answers to voice assistants. Yang Sun, Wen wu Wang, found the main senses achieve the perception of the outside world in the case of human beings. The visual and audio interpretation are the most critical of these. Nevertheless, speech has traditionally held a high priority in human conversation since it existed well before writing was introduced. As the main mode of communication between people, speech has the ability to be an effective mode of computer interaction. Speaking should be used from a distance such that the hand and eye can be busy in desired mode. In the world of robotics applications, signal processing technology is currently seeing rapid advances. The new field of speech signal processing, which led to phenomenal results, has greatly benefited from these advances. Speech processing has a great deal in the fields of defence, medicine, robotics and many other research and applications. Spoken word recognition, speech coding, speech synthesis and voice improvement are major fields covered in the field of word signals processing. A wearable electronic stick integrated with ASR systems, GPS and other connectivity devices will guide people with disabilities with the advancement of the wearable

electronic industry. Using speech acknowledgement provides links to blogs, dictation apps and others. Order, e-learning control guidance. Voice user interfaces (VUIs) use spoken technologies to enable users to access content, carry out the desired transactions and facilitate communication. VUIs rely on user input auto-language recognition. Since ASR applications use discourse as info, they keep away from or diminish the utilization of more regular manual information techniques (e.g., console, mouse), making them a feasible alternative for individuals with outrageous physical or neuromotor incapacities. VUIs permit human-PC exchanges that focus on a predefined assortment of discourse inputs that are required in a particular undertaking area and cover just a restricted range of human discussion. Analysts lean toward ASR due to its tweaked/customized approach in fields like safeguard, medication, robotization, and numerous others. A wide scope of ASR frameworks have been created because of the wide scope of possible applications. Because of their adaption to the pertinent application, discourse acknowledgment and discourse cognizance advancements have advanced into standard science.

II. Technique

Yaron Laufer and Sharon Gannot's VAD detection refers to the issue of the identification in an audio signal of speaker and non-speaker segments. It is the front end of many speech processing applications, including efficient speech recognition and low-bandwidth compression systems. Heavy and non-stationary noise present VAD systems with significant problems and research has been working on improving solid systems over recent years. Either the mixtures of voice and noise matching the application and marked with voice operation (monitored learning) or the noise data at most identical to the noise found in the application are used in a traditional modern VAD device (semi-supervised learning). In the above, methods implicitly presume that noise training data are usable since a noise model has to be initialised. The above semi-monitored approaches are based on parametric noise expectations (e.g. gaussianity), which can be severely infringed by non-stationary noise conditions. These advanced training details can be complex and laborious to collect.

It is also advantageous to build a VAD device, which is both unmonitored and resilient since it can work in different noise conditions across a broad spectrum of signal to noise ratios without training data. Previous VAD programmes, including G.729B and AMR, adopted a rule-based strategy, so no training data were needed.

They have been increasingly replaced by more stable and superior classification-based methods that include labeled training data. There was recently an interest in designing unattended VAD systems with supervised systems' efficiency benefits. The usual solution was to make current supervised and semi-monitored approaches more adaptive. 1 SW proposes a different method, based on a common paradigm in the source separation literature, the non-negative matrix factorization (NMF). Contrary to the above VAD methods, we model the sound blend directly (speech and noise). This has the drawback that the technique works in any noise situation provided one has a fair general model for expression. We will explain in depth how a universal language model is achieved in the next section, but this model is normally trained by a variety of speakers in a clean speech database. Once mastered, it can be used to detect speech in any noise context (from any unavailable speaker). Therefore, it is unattended from a customer point of view as the device is deployed. Our methodology also has the advantage of being entirely understandable: if you were to use this model for distinction, the characteristics used for classification precisely correspond to the relative levels of speech and noise.

Natural linguistic processing (NLP) is an Artificial Intelligence and Linguistics Tract dedicated to machine understanding of human language claims or sentences. In order to simplify the work of the user and fulfil the desire to connect with the machine naturally, the natural language processing came about. Since not all users are familiar with a single computer language, NLP offers users who don't have enough time to understand or master new languages. A collection of rules or a set of symbols may be described as a language. The symbol is merged and used for information transmission or transmission. The Rules tyrannise symbols. Natural linguistic processing (NLP) is an Artificial Intelligence and Linguistics Tract dedicated to machine understanding of human language claims or sentences. In order to simplify the work of the user and fulfil the desire to connect with the machine naturally, the natural language processing came about. Since not all users are familiar with a single computer language, NLP offers users who don't have enough time to understand or master new languages. A collection of rules or a set of symbols may be described as a language. The symbol is merged and used for information transmission or transmission. The Rules tyrannise symbols. Natural language processing can

essentially be divided into two parts, namely natural linguistic understanding and the generation of natural languages, which is the objective. Linguistics is a language science that involves sound phonology, word composition of morphology, the sentence construction of Grammar, the Semantics syntax and the interpretation of pragmatism. This can be split into two categories: higher speech recognition and lower natural language recognition. Automatic synopsis, resolution of co-references, discourse analysis, machine translation, morphological segmentation, name of individuals identification, recognition of optical character, part of speech synopsis, and so on are some of NLP's research activities. Machine translation, recognition of named entities, detection of optical characters, and other real-world applications are all possible with any of these features. An automatic description of a collection of text is produced in an understood way, including summaries or descriptive information on a known class of text.

It means a phrase or greater number of text, co-reference resolution, which specifies which word corresponds to the same object. Discourse analysis refers to the process of defining a linked text discourse system. Translations from one human language to another, automated translation of the text. Morphologic division which refers to individual morphemes separating word and identifying the grade of morphemes. A recognition entity named (NER), the text stream is defined and the objects in a text correspond to proper names. It provides an illustration of impressed text, Optical Character Recognition (OCR), which helps to define a matching or associated text. Part of the language tag defines a sentence which specifies the part of the language in a word. Although NLP tasks are clearly interwoven quite closely, they are often used for convenience. The task used for solventing bigger tasks, including automated summarisation, co-reference analysis, etc. Natural language processing's aim is to satisfy one or more specialisations' algorithms or methods. The human language technology metric, which is based on an algorithmic scheme, allows for the combining of language comprehension and language generation. It's also used to detect multilingual events. Rospocher et al. devised a one-of-a-kind standard method for crosslingual event extraction for English, Dutch, and Italian texts by employing distinct pipelines for each language. A common collection of leading multilingual linguistic communication process (NLP) tools is included in the scheme.

As a result, the cross-lingual structure enables the understanding of activities, participants, places, and time, as well as their relationships. Individual pipeline output is intended to be used as input for a device that generates event-centric data graphs. All modules act like UNIX machine pipes: they all take standard input, process it, and output standard output, which is then used as the input for subsequent module pipelines. Furthermore, modular architecture allows for a variety of configurations as well as dynamic delivery. The majority of the add linguistic communication method is carried out by computer scientists, but other experts such as linguists, scientists, and philosophers have also expressed interest. The fact that human language technology contributes to the data of human language is one of the most ironic aspects of it. The field of linguistic communication is said with a variety of theories and techniques that trot out the subject of linguistic communication of computer communication. Ambiguity is a major flaw in linguistic communication that is often encountered at the grammar level, which includes subtasks such as lexical and morphology, which are concerned with the analysis of words and word formation. Any one of those levels will reveal ambiguities that can be resolved using the entire sentence's data. The anomaly can be resolved in a variety of ways, including minimization, conservation, interactive disambiguation, and weight uncertainty. Conserving uncertainty is one of the methods proposed by researchers to eliminate ambiguity.

As examined in and, four significant examination groups in discourse acknowledgment have discovered that DNN-based frameworks give altogether higher precision in nonstop telephone and word acknowledgment than the previous reformist GMM-based frameworks and the prior shallow neural organization frameworks in the course of the most recent couple of years. It is found that still there is a significant scope for research in the area of automated speech recognition and speaker recognition. MATLAB is used for the development of speech ranking software. Deep Learning toolbox is used for the development of LSTM based classifier.

CHAPTER 3

RESEARCH DATABASE

In order to perform the research work, an extensive research database has been created. It consists of 330 speech signals in three different languages namely Tamil, Telugu and English. In each language for every speaker, a total of ten samples are collected. While recording speech signals, speakers are advised to change their timbre and pitch.

Speech samples are shown in Figure 3.1. From subjective analysis, it is evident that these signals are non-stationary in nature. They also have bursts at certain intervals of time. Also each signal has voiced unvoiced and silent regions. Duration between these three regions is different for different samples. Though the speech signals shown in Figure 3.2 belong to a same speaker, these signals have different characteristics. However there is/are inherent feature/features that/which are common to these three signals (not evident to human eye).

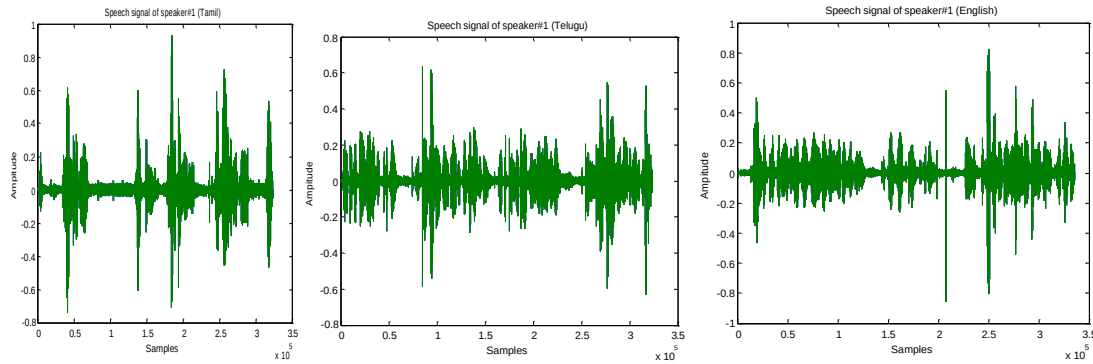


Figure 3.1. speech samples of Speaker #1 (English, Tamil and Telugu)

These characteristics can be represented using appropriate statistical parameters. Instead of considering the entire speech signal for feature extraction, 1024 samples of the signal are considered. On the preprocessed signal, short time energy is calculated. As the preprocessed signal consists of voiced, unvoiced and silent region, the determined short time energy can be regarded as the indicator of speech signal characteristics.

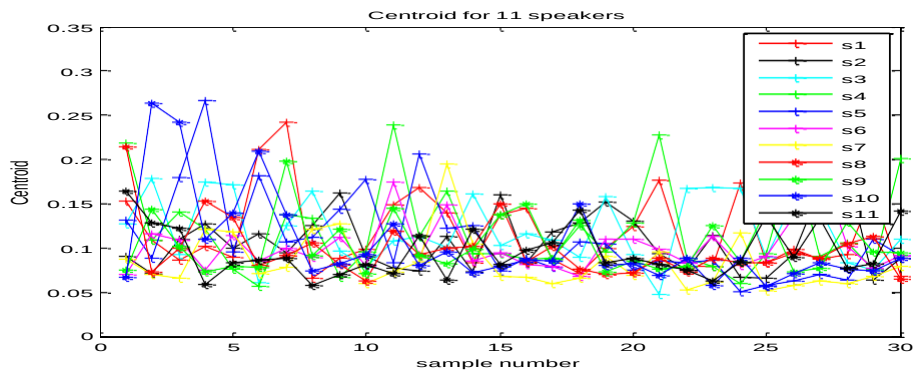
Flux and centroid (indicators of fundamental frequency) are obtained on Fourier Transform co-efficients. Table 3.1 shows these parameters for eleven speakers (three different languages).

Table 3.1 Short time Energy, centroid and Flux for speech signals

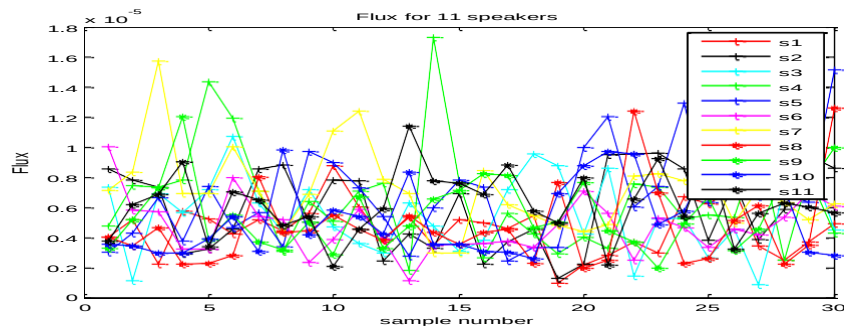
Speaker	Language	Centroid	Short Time Energy ($\times 10^6$)	Flux
1	English	0.153334	4.02	0.000476
	Tamil	0.149357	6.01	0.000308
	Telugu	0.176847	2.81	0.000779
2	English	0.090358	8.54	0.000257
	Tamil	0.077693	7.77	0.000256
	Telugu	0.093139	9.49	0.000395
3	English	0.127732	7.37	0.000196
	Tamil	0.107561	3.63	0.000151
	Telugu	0.048059	8.63	.0000980
4	English	0.218448	4.78	0.000898
	Tamil	0.239241	7.09	0.000327
	Telugu	0.227249	3.29	0.000856
5	English	0.131802	3.00	0.000672
	Tamil	0.083781	7.31	0.000335
	Telugu	0.085384	12.0	0.000239
6	English	0.070248	10.0	0.000224
	Tamil	0.17439	5.89	0.000971
	Telugu	0.098999	5.60	0.000408
7	English	0.087865	7.14	0.000197
	Tamil	0.073417	1.24	.0000198
	Telugu	0.095396	4.89	0.000363
8	English	0.214483	4.02	0.000942
	Tamil	0.118954	4.55	0.00028
	Telugu	0.088237	2.49	0.000374
9	English	0.075238	3.28	0.000257
	Tamil	0.144989	6.69	0.000435
	Telugu	0.076583	4.44	.0000454
10	English	0.067179	3.58	0.000247

11	Tamil	0.126979	5.38	0.000154
	Telugu	0.069022	9.72	0.0000995
	English	0.16457	3.74	0.00067
	Tamil	0.071261	4.56	0.000382
	Telugu	0.080976	2.18	0.000252

From the Table 3.1, it is found that the parameters flux, centroid and short time energy do not have linear relationship with the speaker i.e range cannot be fixed for individual speakers. It is because values overlap among the speakers. The above fact is also visible in the following graphs shown in Figure 3.2 a), b) and c). It is because, Fourier transform cannot capture the spectral co-efficients of a random signal.

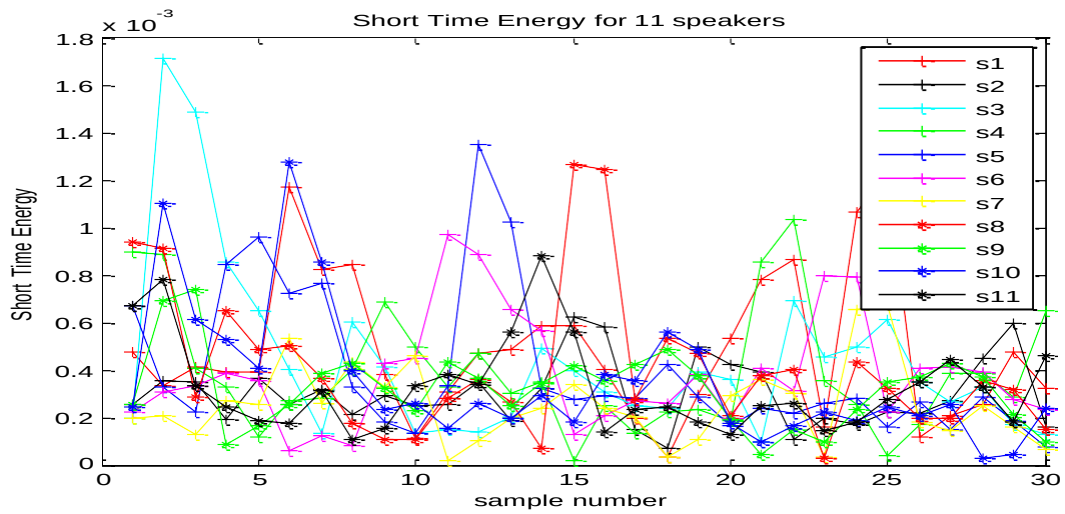


(a)



(b)

Figure 3.2 (Continued)



(c)

Figure 3.2 Relationship between the speakers and a) Centroid b) Flux c) Short time energy

From the above, it is evident that time domain analysis cannot be used for speech ranking system and hence , spectral domain techniques are to be used.

CHAPTER 4

RNN AND LSTM BASED IDENTIFICATION OF SPEECH SIGNALS

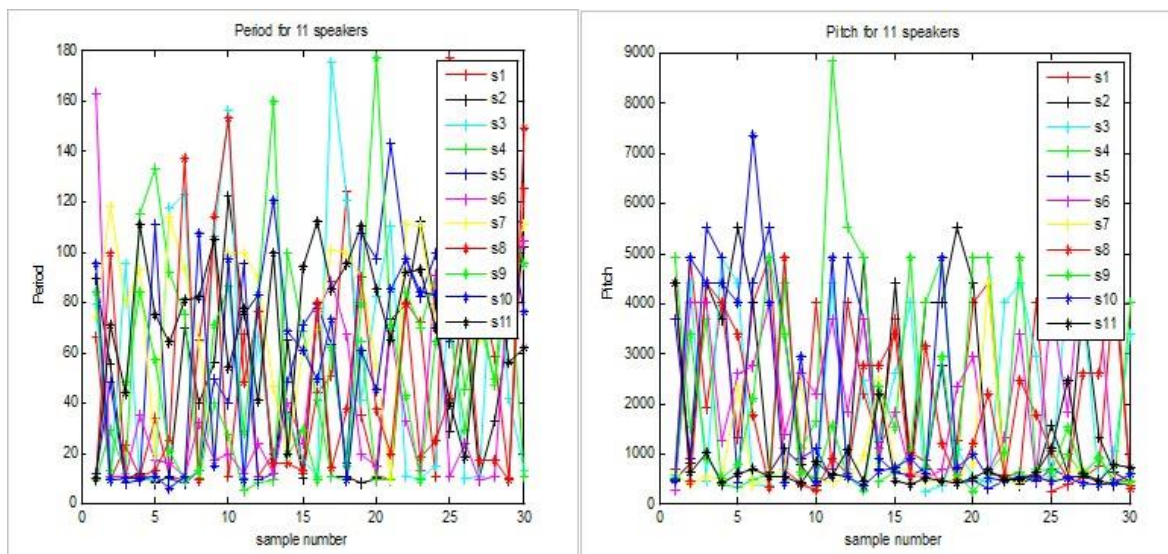
The two key modules in any speaker recognition system are feature extraction and classification. In the proposed work, autocorrelation features, Approximation statistical features and detailed statistical features are obtained from the speech signals. Discrete Meyer wavelet (dmey) is the wavelet used for the extraction of features. The extracted features are given to a classifier to identify the speaker based on the features. In order to improve the performance of the classifier the inter relationship among the features is to be considered. RNN and LSTM are the two classifiers considering the relationship between the input features.

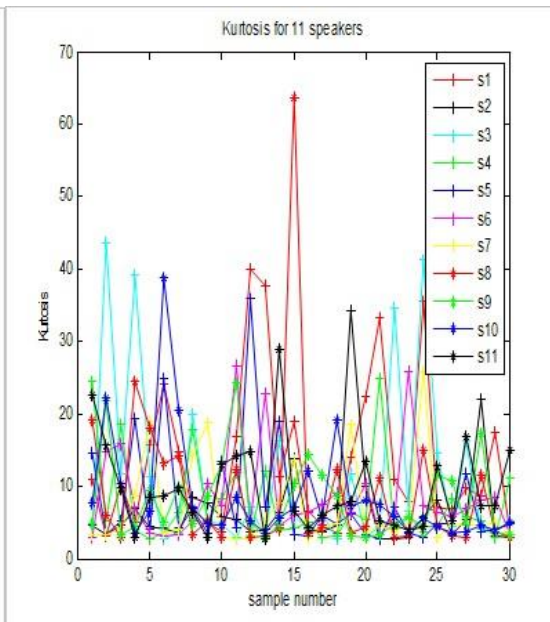
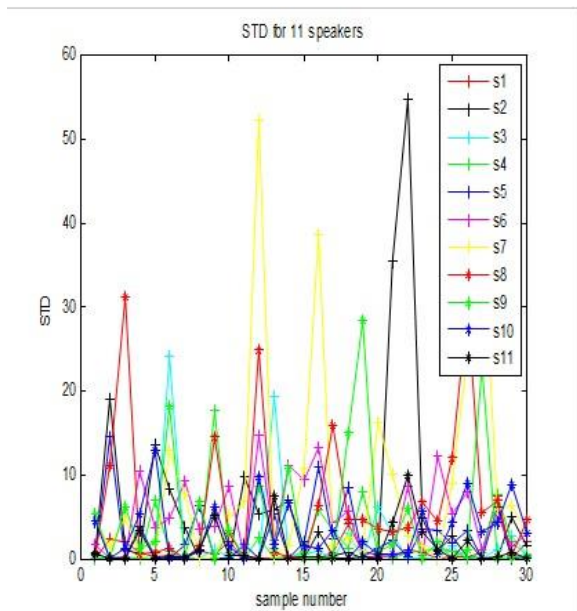
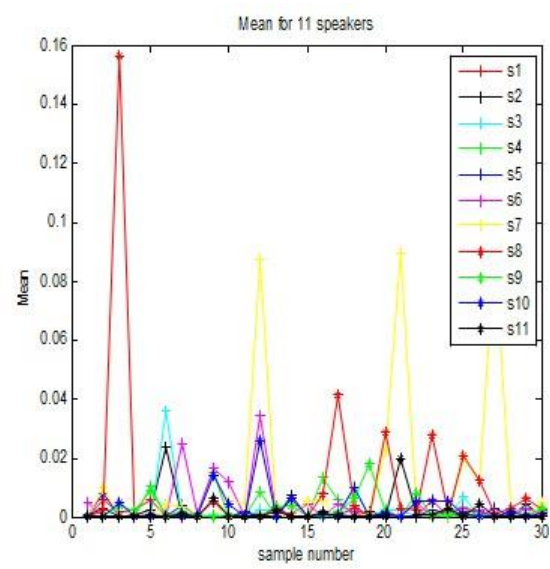
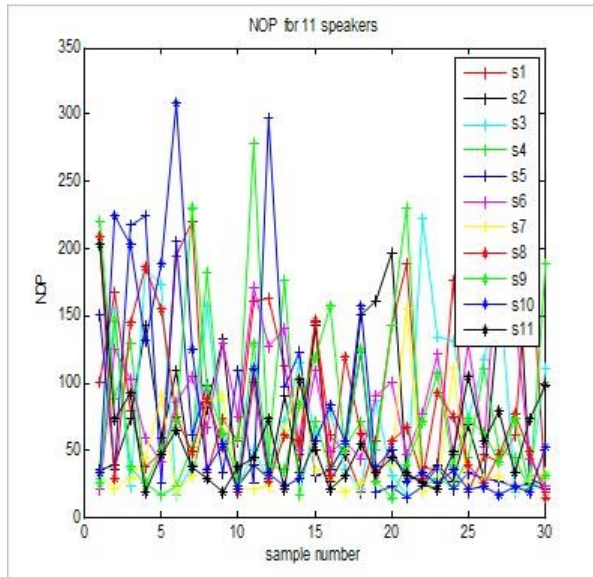
Recurrent Neural Network (RNN)

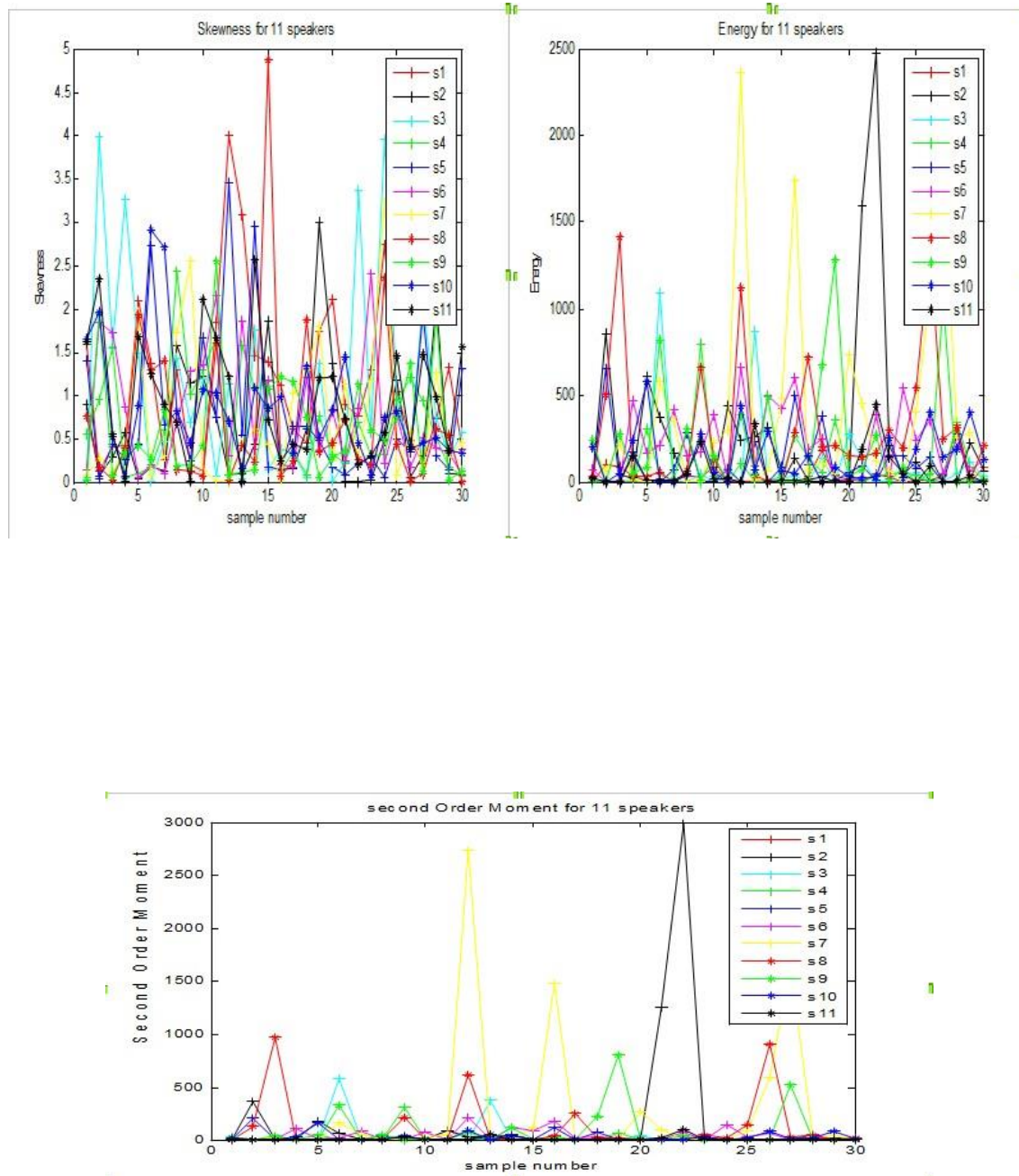
RNN is one of the subclass of Deep Learning Neural Network. The RNN architecture used in our work comprises of one hidden layer with ten neurons. Efficient learning for any complicated classification work is performed only by the hidden layer. For RNN based Speaker recognition system, nine speakers signal is considered. Features namely Pitch, Period, Number of Peaks, Mean, Skewness and Kurtosis were extracted from autocorrelation coefficients and given to RNN for speaker classification. In RNN, one feature is given as the input and output is predicted at an instant. Likewise, six features are given and the output is the corresponding speaker identified. Out of 270 signals, for training 135 signals and for testing 135 signals are used.

For LSTM based speaker recognition system eight speakers signal is considered. Out of 30 speech signals from each speaker, 15 are considered for training and 15 are considered for testing. The proposed LSTM approach is applied with four different classes of inputs for analyzing suitable features for speaker recognition system.

Class I : Statistical boundaries in particular Pitch, Period, NOP, Mean, Standard Deviation, Kurtosis, Skewness, Energy, ZCR over autocorrelation coefficients increased by a factor of 100. Class II : Statistical boundaries to be specific Pitch, Period, NOP, Mean, Standard Deviation, Kurtosis, Skewness, Energy, ZCR, Second Order Moment and Third Order Moment over autocorrelation coefficients. Class III : Statistical boundaries to be specific Pitch, Period, NOP, Mean, Standard Deviation, Kurtosis, Skewness, Energy, ZCR over autocorrelation coefficients duplicated by a factor of 1000. Class IV : Ten Statistical boundaries specifically Pitch, Period, NOP, Mean, Standard Deviation, Kurtosis, Skewness, Energy, ZCR, Second Order Moment over the Detailed coefficients of Discrete meyer wavelet coefficients.







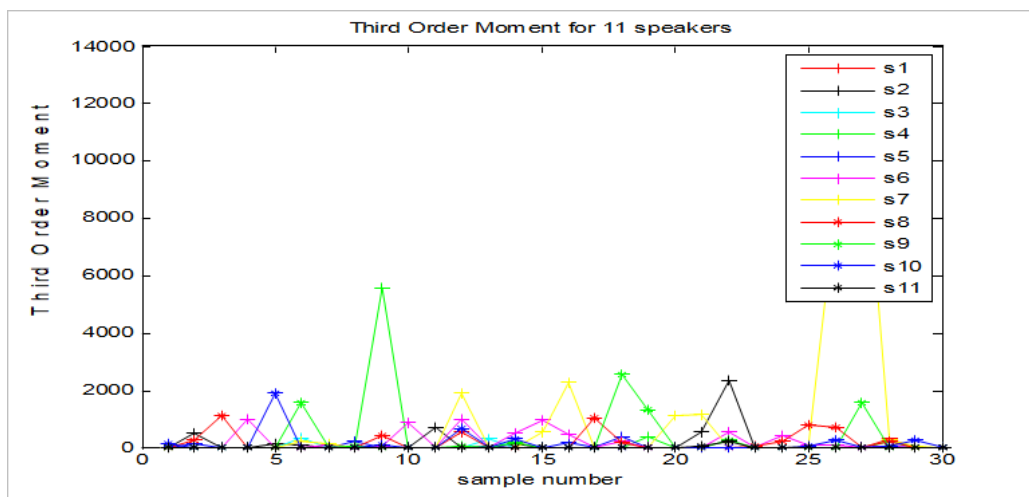
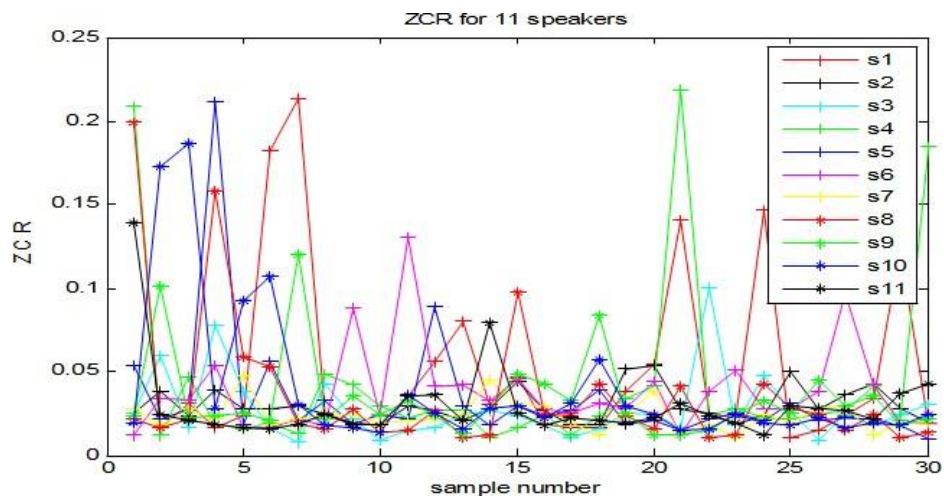
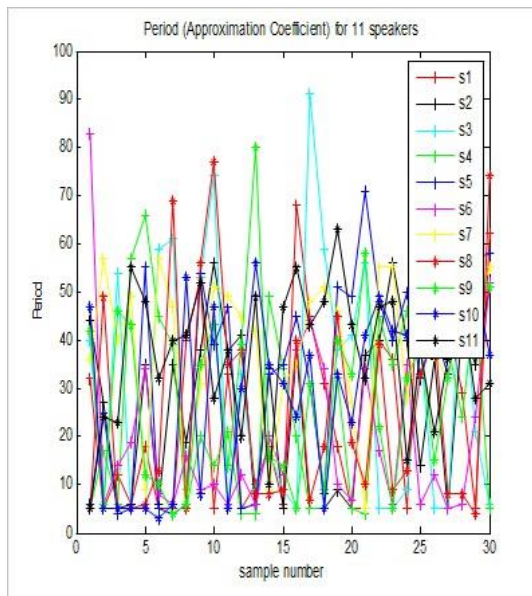
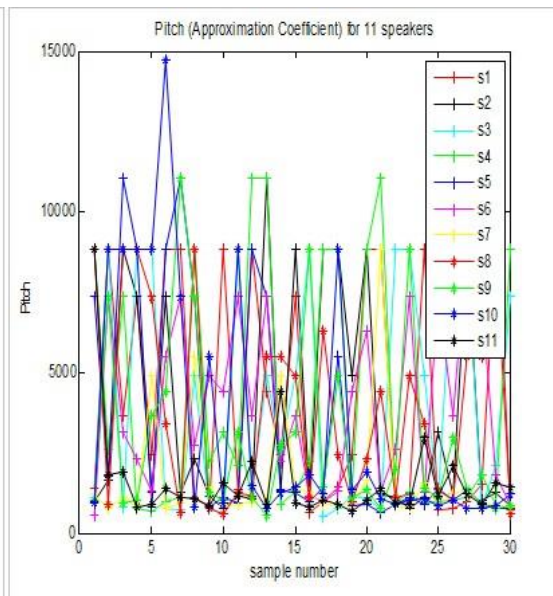


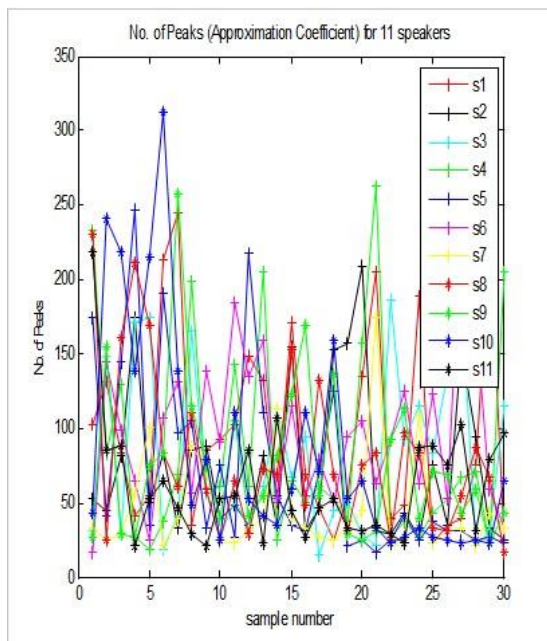
FIG.4.1 Class II parameters for representing speech signals (Autocorrelation)



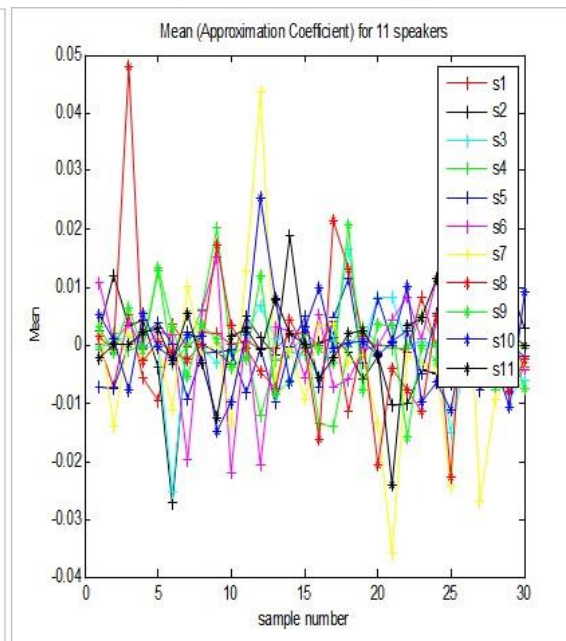
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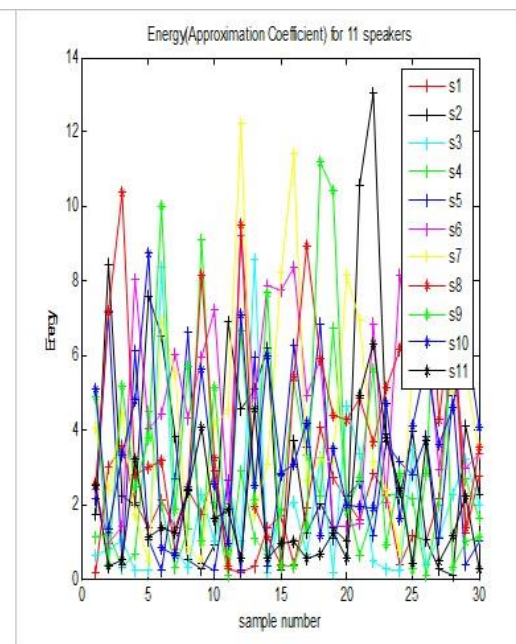
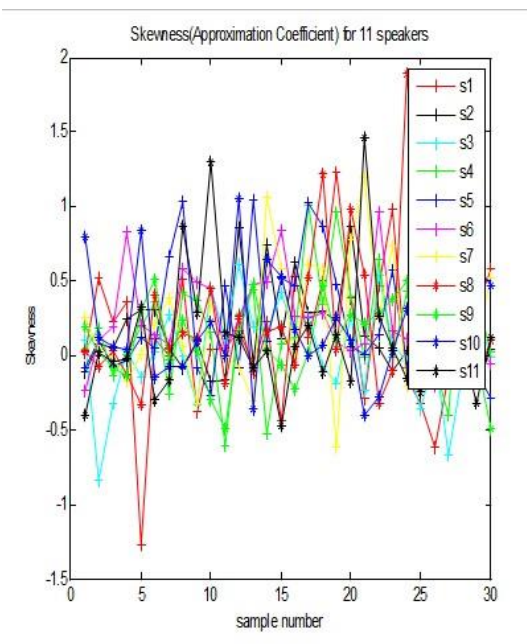
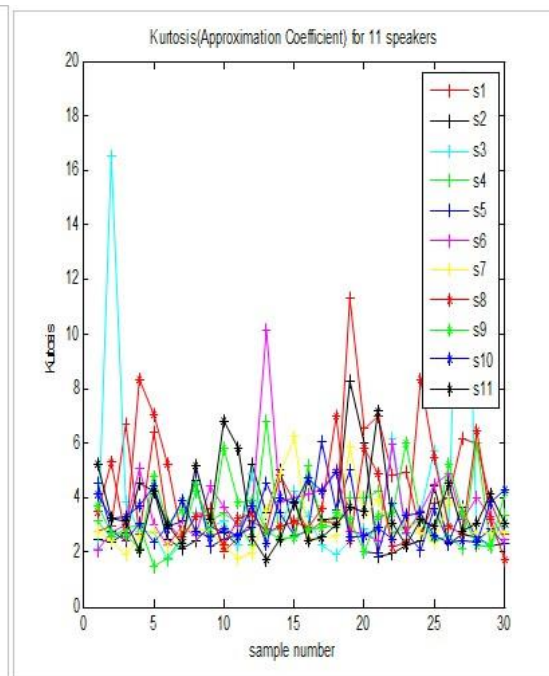
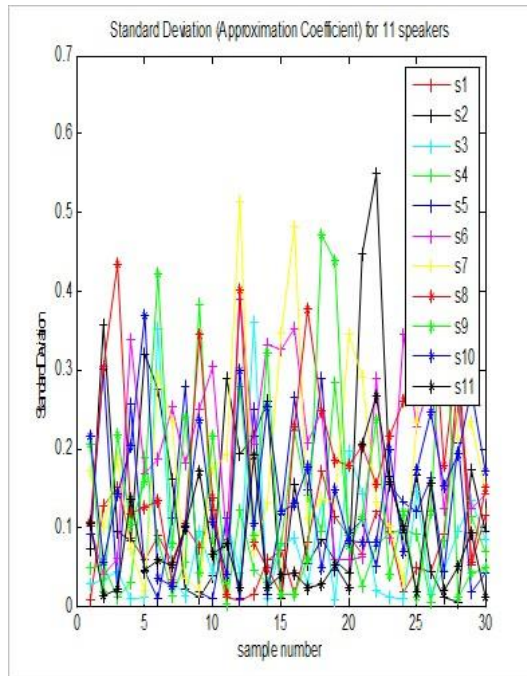
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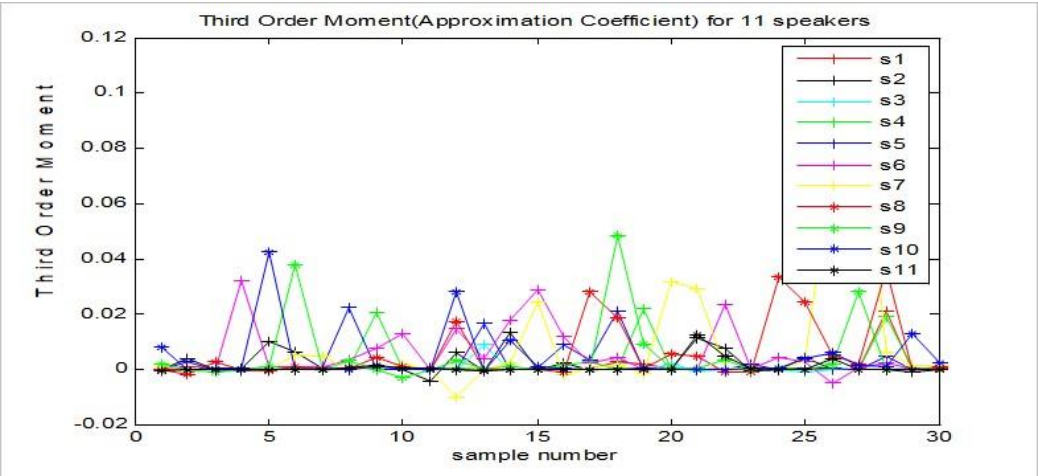
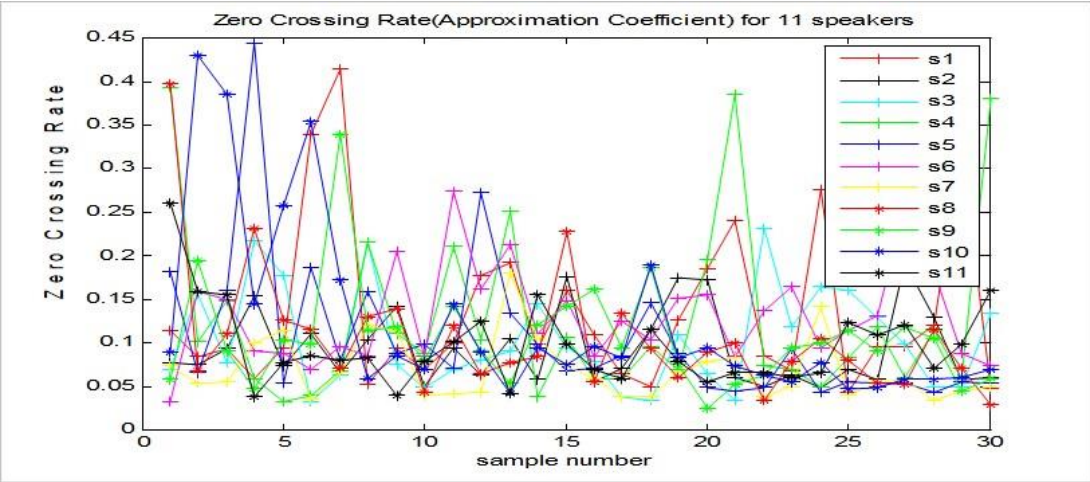


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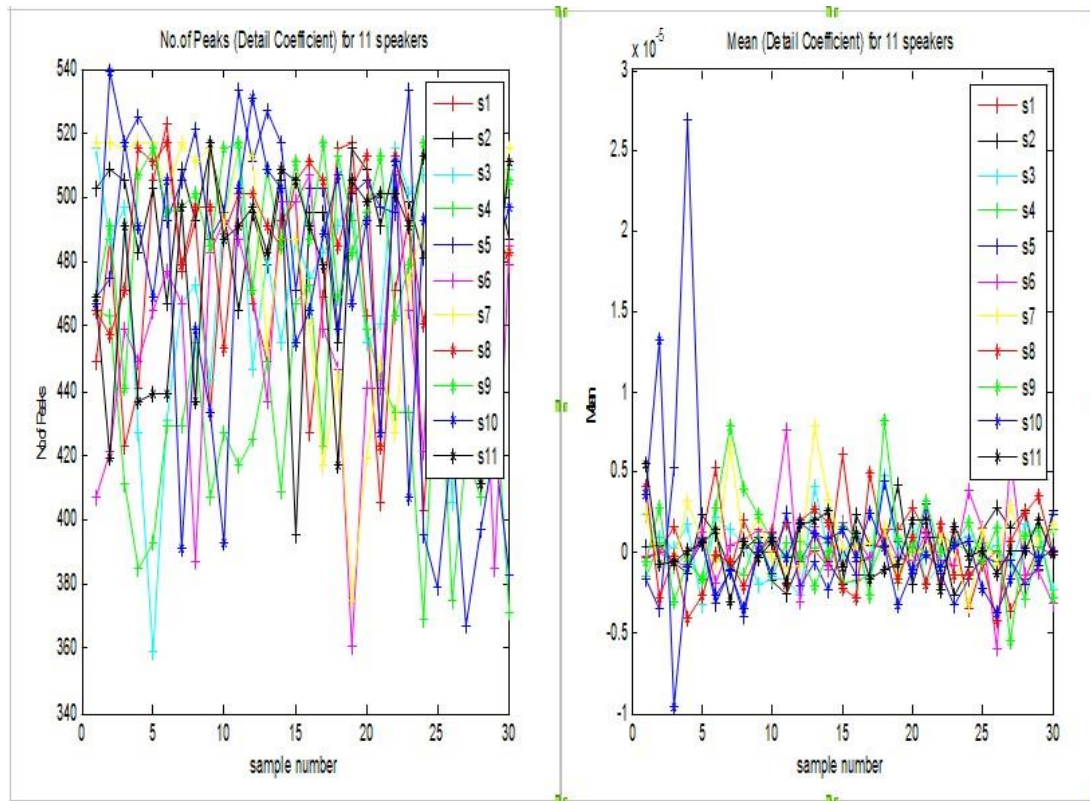
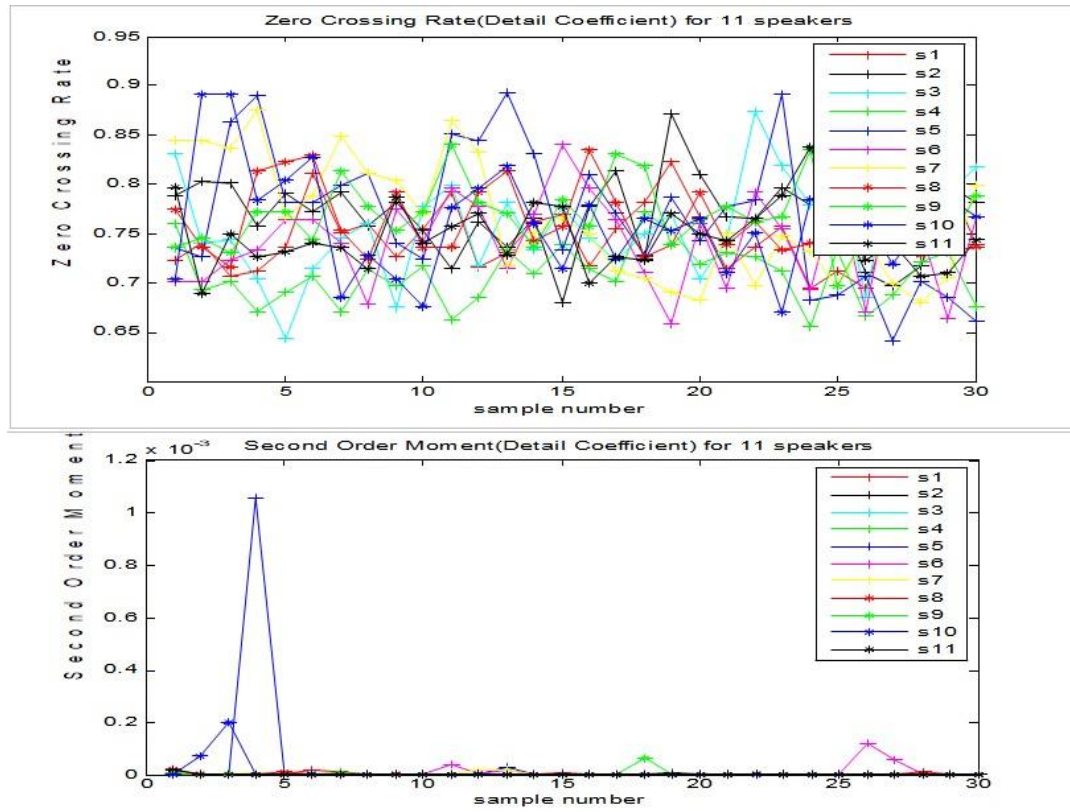
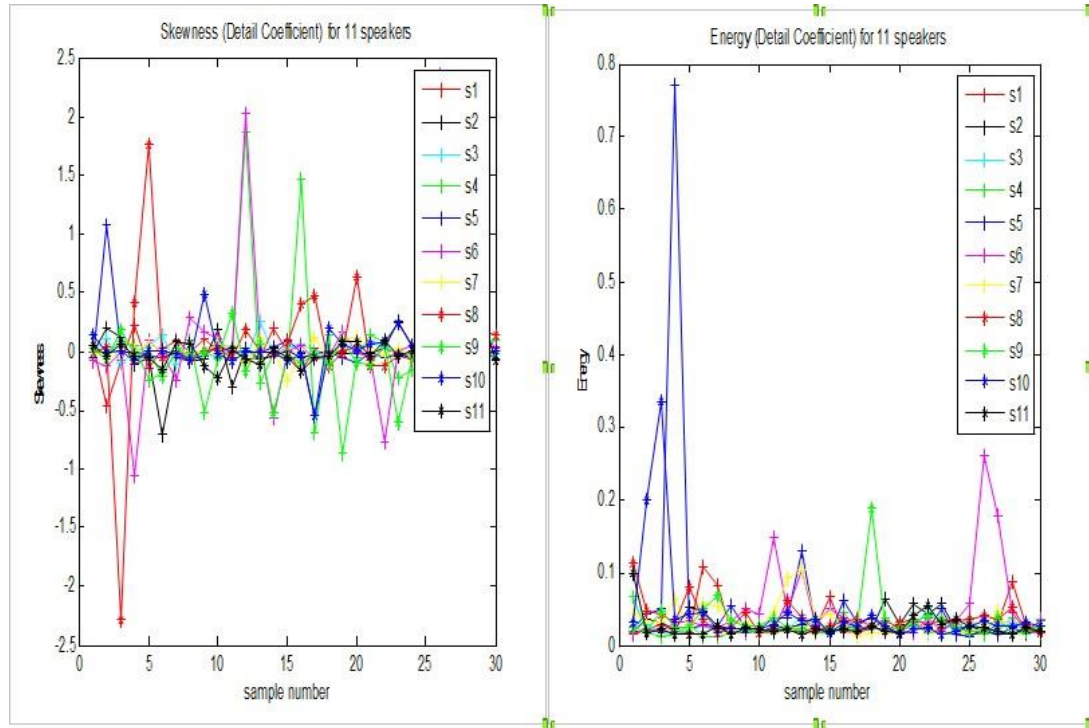
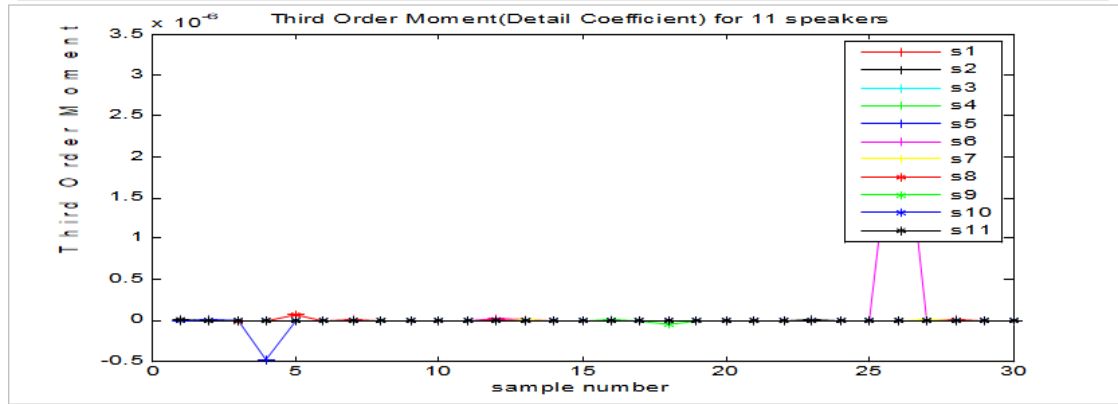


FIG 4.2 Class IV parameters for representing speech signals (Discrete Meyer Wavelet)





In Recurrent Neural Network (RNN) based speaker Recognition System, six statistical features of nine speakers are used . Using Confusion matrix , Sensitivity is calculated for all the nine speakers. The computed sensitivity values for all the speakers with the extracted features is shown in Table 1. From Table 1, it is observed that the statistical feature Skewness identifies all the speakers, whereas other features fails to identify all the speakers.

Table 1: Sensitivity using RNN Classifier

Speaker	pitch	period	NOP	Mean	Skewness	kurtosis
1	23	0	0	15.3	15.3	39
2	29	14.2	0	0	93	2
3	44	3	19	13	75	2
4	60	2	2	20	47	0
5	33.3	2	0	2	2	0
6	0	14	20	20	13.3	0
7	0	20	2	20	27	13.3
8	0	33.3	20	0	20	13.33
9	0	2	0	0	2	40

In LSTM network is applied with four different classes of inputs. For all the classes of inputs, confusion matrix is formed through which sensitivity is also calculated. The computed sensitivity values for all the speakers for all the set of input classes is shown in Table 2. From Table 2, it is observed that the statistical features from detailed coefficients of discrete meyer wavelet gives better results compared to other classes of inputs from other coefficients.

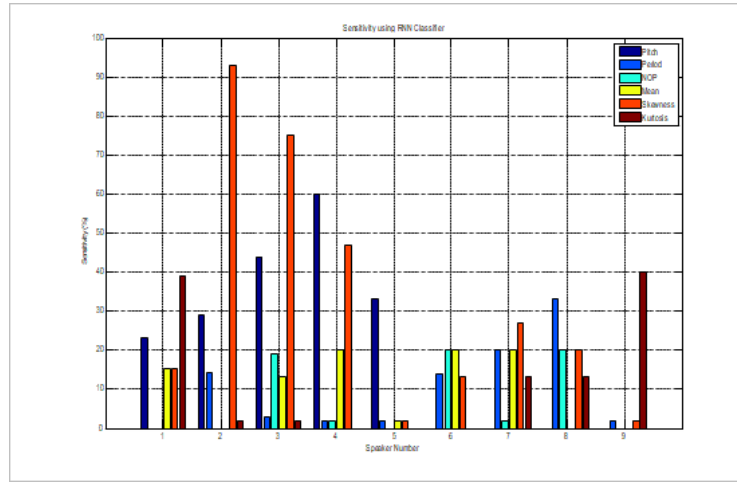


Fig.4.3Sensitivity Comparison for various speakers using RNN Classifier

Table 2: Sensitivity using LSTM Classifier

Speaker	Class I	Class II	Class III	Class IV
1	33.3	6.67	46.67	40
2	13.3	0	0	13.33
3	6.67	27	13.33	6.67
4	6.67	13.3	6.67	13.33
5	13.3	13.3	6.67	20
7	40	20	6.67	40
8	0	13.3	20	40
9	13.3	0	20	13.33

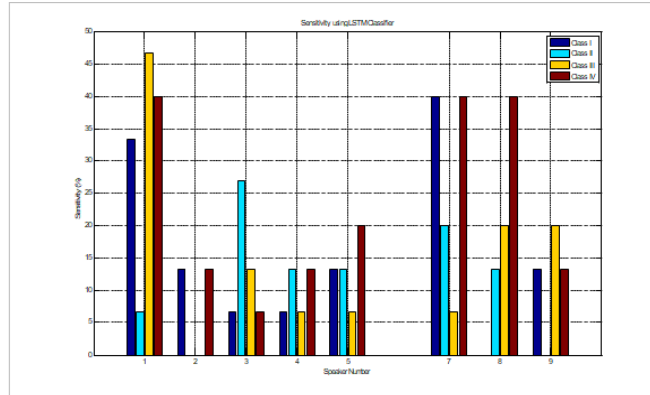


Fig. 2. Sensitivity Comparison for various speakers using LSTM Classifier

From the Tables, it is clear that LSTM classifier provides the best output for Class IV exemplars. That is the Discrete Meyer co-efficient provide the highest inter class variance and less intra class variation.

Proposed Work

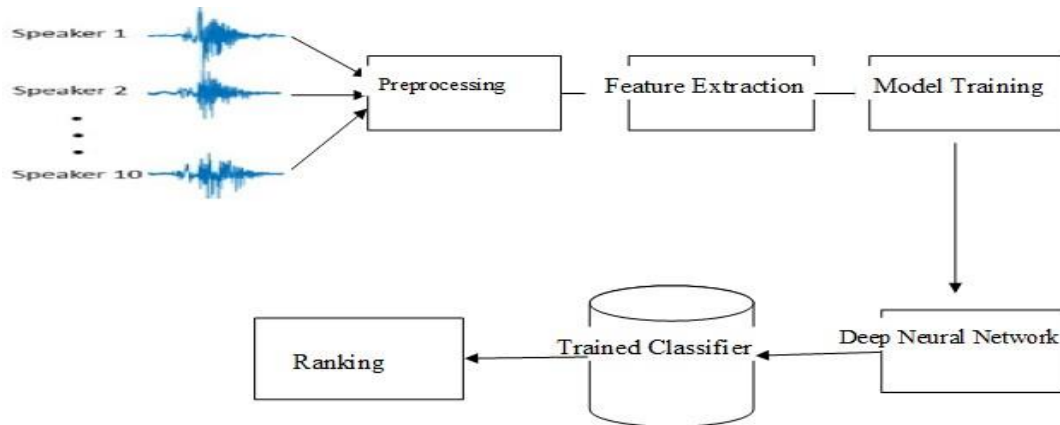


FIG 4.5 BLOCK DIAGRAM

The discussion signal is given as an information. By then the sound data is pre-dissected and the outstanding part habitats are dealt with in the enlightening file. Discrete Wavelet Transform (DWT) is utilized for the part extraction to discard the undesirable rehash of the signs. At long last the critical learning procedure is utilized to rank the sound sign from the speakers dependent on voice sign and mistake rate.

RESULT

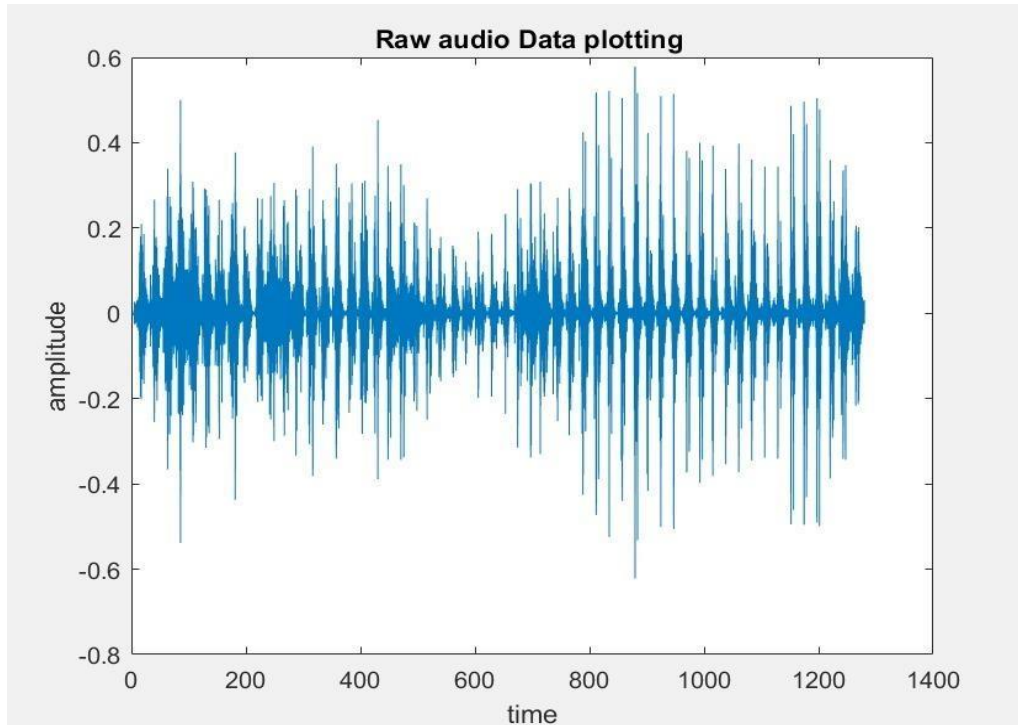


FIG 4.6 RAW AUDIO DATA PLOTTING

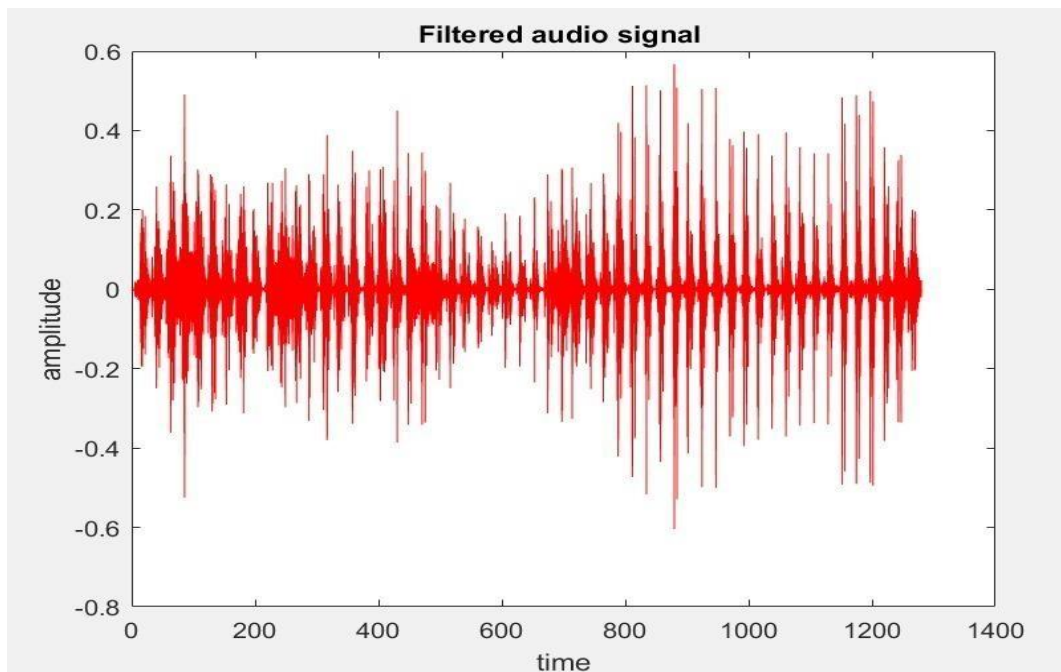


FIG 4.7 FILTERED AUDIO SIGNAL

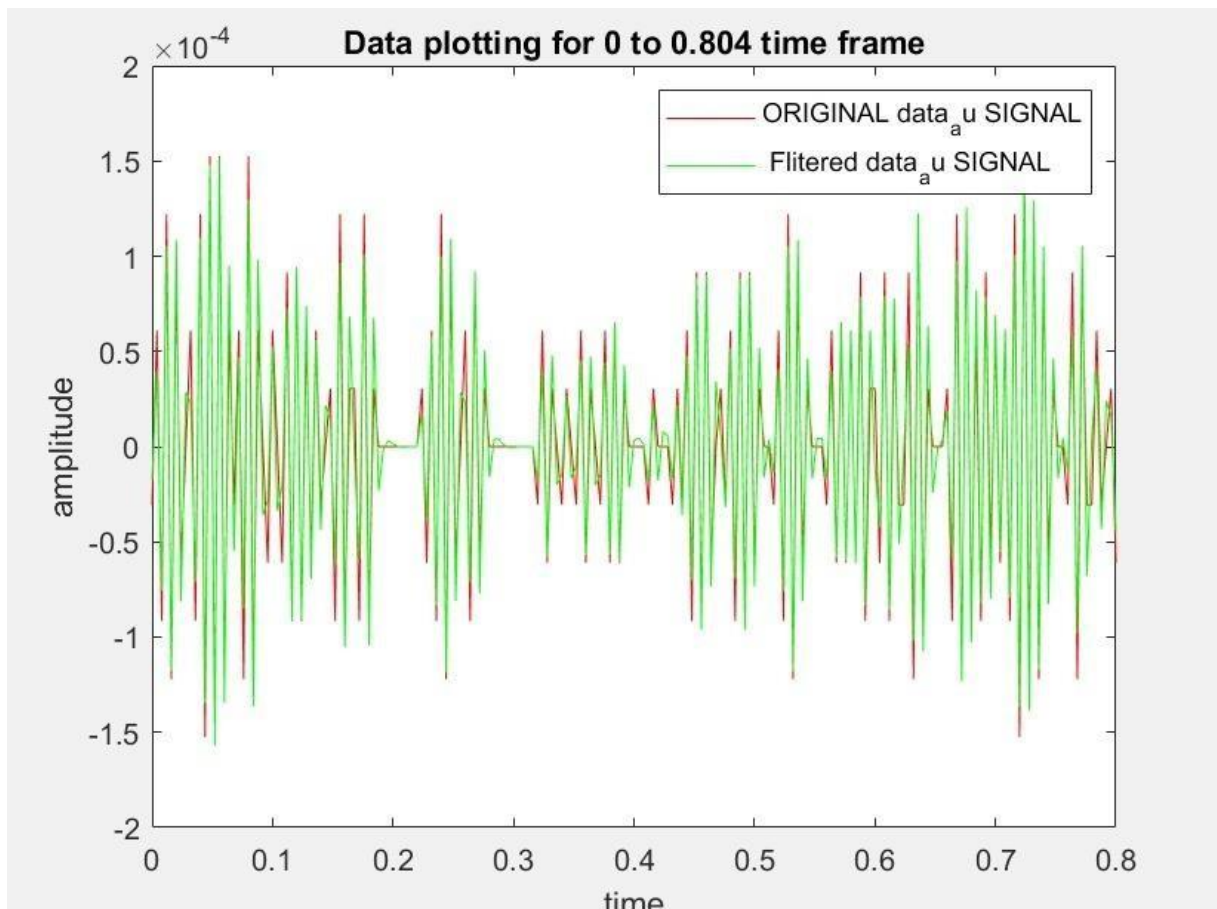


FIG 4.8 DATA PLOTTING FOR 0 TO 0.804 TIME FRAME

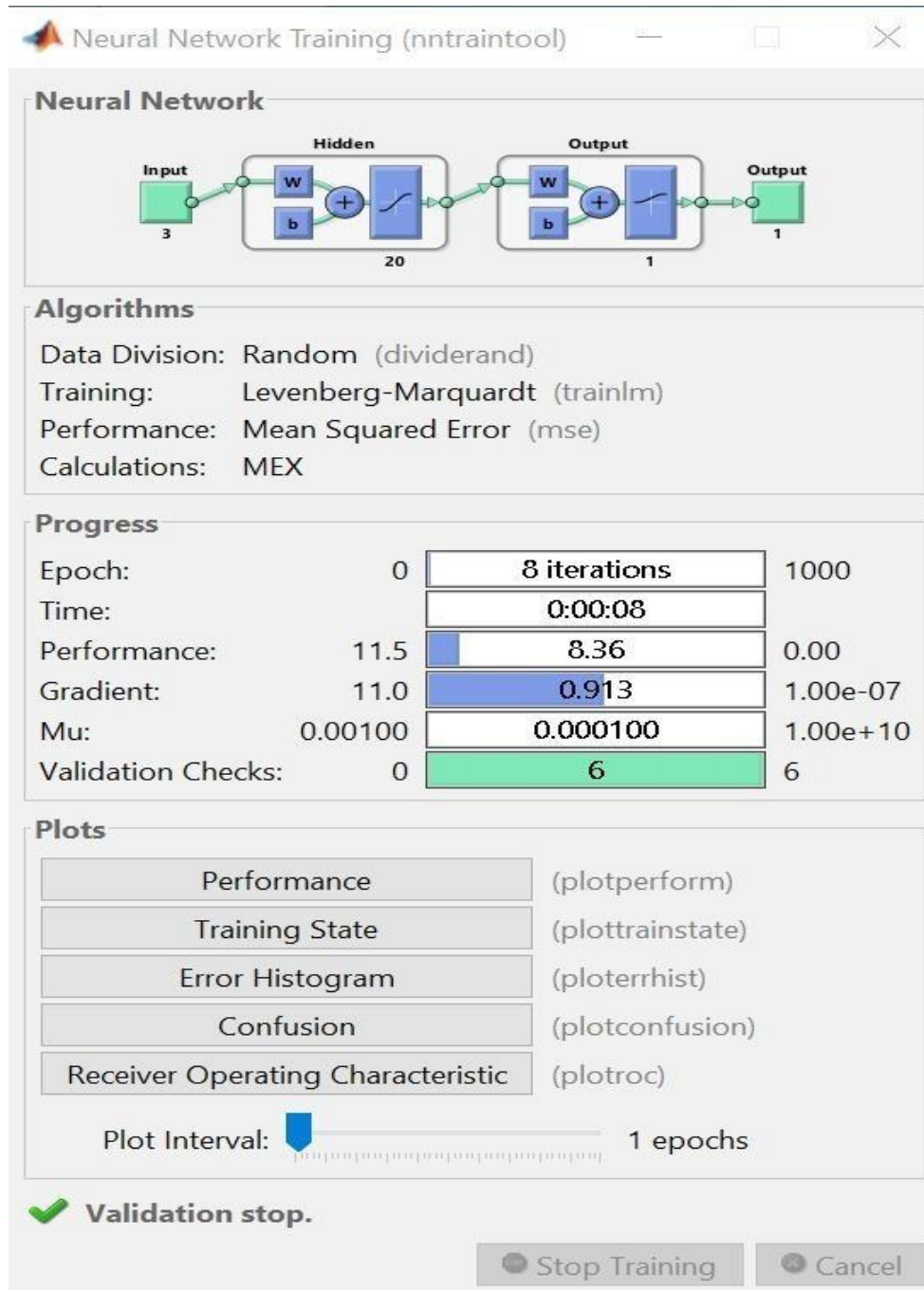


FIG 4.9 NEURAL NETWORK TOOL BOX

CHAPTER 5

CONCLUSION AND FUTURE WORK

Speaker recognition has a very important role in various applications like forensic department for identification of anti-social elements, biometric authentication systems etc. In this paper, various features involved in development of automated speaker recognition system is identified with an extensive real time research database. Also the two categories of Deep Learning neural network classifier namely RNN and LSTM is also used for speaker recognition. Based on the sensitivity calculations it is identified Skewness and the statistical features over the detailed coefficients of discrete meyer wavelet gives better results. These features are fed to RNN and LSTM for speaker recognition. LSTM has higher sensitivity and accuracy than RNN.

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report

by sneha v

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**DEEP LEARNING TECHNIQUES FOR FEATURE BASED RANKING OF AUDIO
SIGNALS IN SEARCH ENGINES**

³ Submitted in partial fulfillment of the requirements for the award of Bachelor of
Engineering degree in Electronics and Communication Engineering

by

SNEHA. V (37130389)

SCINDHIA. M (37130363)



³ **DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING**

SCHOOL OF ELECTRICAL AND ELECTRONICS

SATHYABAMA

INSTITUTE OF SCIENCE AND TECHNOLOGY

(DEEMED TO BE UNIVERSITY)

Accredited with Grade "A" by NAAC

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APRIL – 2021



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DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

BONAFIDE CERTIFICATE

5

This is to certify that this Project Report is the bonafide work of **SNEHA.V(37130389)** and **SCINDHIA.M(37130363)** who carried out the project entitled "DEEP LEARNING TECHNIQUES FOR FEATURE BASED RANKING OF AUDIO SIGNAL IN SEARCH ENGINES" under my supervision from October 2020 to April 2021.

Internal Guide

Dr. N. M. Nandhitha; M.E., Ph.D., Dean

Head of the department

Dr.T.Ravi. M.E., Ph.D.,

Submitted for Viva Voce Examination held on _____

Internal Examiner

External Examiner

DECLARATION

We SNEHA.V (Reg.No: 37130389) and SCINDHIA.M (Reg.No: 37130363) hereby declare that the Project Report entitled "DEEP LEARNING TECHNIQUES FOR FEATURE BASED RANKING OF AUDIO SIGNALS IN SEARCH ENGINES" done by us under the supervision of Dr. N.M.NANDHITHA. M.E., Ph.D., Dean is submitted in partial fulfillment of the requirements for the award of Bachelor of Engineering degree in ELECTRONICS AND COMMUNICATION ENGINEERING.

DATE :

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PLACE:

2)

SIGNATURE OF CANDIDATE:

ACKNOWLEDGEMENT

We are pleased to acknowledge our sincere thanks to Board of Management of SATHYABAMA for their kind encouragement in doing this project and for completing it successfully. We are grateful to them.

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ABSTRACT

The need for audio authentication has become mandatory in the modern epoch. Audio recognition is also known as speech recognition, and voice recognition is a computer software programme or hardware system that can decode the human voice. Voice recognition is a method of controlling a computer, issuing instructions, or writing without the use of a keyboard, mouse, or buttons. Distance metrics such as Manhattan distance, Euclidean distance, and Maximum distance were taken into account in the previous scheme. The obtained distances are sorted and ranked in ascending order. The best match has the shortest distance and is ranked first. In the proposed system the audio information is pre-analyzed and the unique feature points are stored in the database. The deep learning technique is used in order to rank the audio signal from the speakers based on the quality of voice signal and error rat

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CHAPTER 1

INTRODUCTION

Discourse acknowledgment is an interdisciplinary branch of computational etymology that focuses on developing strategies and innovations that enable (programmed discourse acknowledgment) and PC discourse acknowledgment are other names for it. Phonetics, software engineering, and electrical engineering data and analysis are combined in this project.

Some discourse recognition systems necessitate "preparing" (also known as "enlistment"), which entails an individual speaker understanding text or secluded jargon and incorporating it into the system. "Speaker autonomous" frameworks are those that don't use any kind of preparation. "Speaker subordinate" frameworks are those that make use of preparation.

Voice UIs include voice dialling (for example, "call home"), call guiding (for example, "I may want to make a gathering decision"), domotic machine control, search catchphrases (for example, "find a webcast where unique words were verbally expressed"), simple knowledge passage (for example, entering a Visa number), and the readiness of structured data (for example, entering a Visa number). "Voice affirmation" or "speaker distinguishing proof" alludes to perceiving the speaker as opposed to the substance of their appearance. As far as creative mind, talk appreciation has a long history, with a couple of outstanding advances. Additionally by broad business appropriation of a scope of profound learning strategies in the planning and conveyance of talk affirmation structures.

Hidden Markov Models are mathematical models that produce a series of images or numbers. Since a discourse sign may be viewed as a piecewise fixed sign or a brief timeframe fixed sign, they are used in discourse

recognition. For certain stochastic reasons, discourse can be viewed as a Markov model.

Another factor HMMs are notable is that they are not difficult to make and use, just as being basic and computationally practical. Several frameworks employ ostensibly discriminative preparing techniques that avoid a straightforward factual approach to HMM boundary evaluation in favour of enhancing some order-related proportion of the preparation data. Most extreme shared data (MMI), least characterization error (MCE), and least telephone error are the models (MPE).

The Viterbi estimation will no doubt be utilized to detangle the talk (what happens when the structure is given another articulation and should enlist the most probable source sentence). A potential improvement to disentangling is to keep a group of good up-and-comers rather than only the best, and to rate these great competitors using a superior scoring skill (re scoring) so that we can select the best one based on this refined ranking.

1.1.DYNAMIC TIME WRAPING (DTW)

Dynamic time travel was once widely used for discourse recognition, but it has since been largely replaced by more effective HMM-based methodology. Dynamic time travel was a method of estimating the similarity of two groups that may vary in terms of schedule or speed. Similes in strolling examples, for example, will be known regardless of whether the individual was strolling slowly in one video and quickly in another, or whether there were speed increases and decelerations in one perception. DTW has been used to investigate film, sound, and designs, but it can be used to investigate any knowledge that can be turned into a clear representation.

Programmable discourse acknowledgment is a notable application that adapts to different talking rates. That is, the groupings are "distorted" to coordinate with one another in a non-direct manner. When it comes to covered up Markov models, this grouping structure technique is commonly used .

1.2.NEURAL NETWORKS

Neural organizations have a few characteristics that make them appealing acknowledgment models for discourse acknowledgment. Early neural organizations, despite their limited ability to display transient conditions, were sometimes useful for constant recognition activities, despite their feasibility in ordering brief timeframe units. Be that as it may, in this field, LSTM and related dreary neural associations (RNNs) and Time Delay Neural Networks (TDNNs) have as of late showed improved execution.

DNN models construct compositional models in which extra layers engage bits of lower-layer features, bringing about an enormous learning limit and the capacity to show complex instances of talk content. Microsoft Research has published a new Springer book, you'll find detailed audits of this turn of events, as well as the best in class as of October 2014. In recent outline posts, you can also read about the linked basis of programmed discourse acknowledgment and the impact of various AI ideal models, including profound learning.

One of the most important rules of profound learning is to avoid hand-drawn highlight designs and instead use rough highlights. This rule was demonstrated to be superior to Mel-Cepstral highlights, which use a couple of fixed change phases from spectrograms, in the design of a profound auto encoder using "raw" spectrogram or direct channel bank highlights. Waveforms, the true "rough" highlights of debate, have recently gotten a lot more attention, and they've been shown to deliver fantastic larger-scale dialogue acknowledgment outcomes.

1.3.END TO END AUTOMATIC SPEECH RECOGNITION

Together, start to finish models achieve proficiency with all of the components of the discourse recognizer. This is important because it affects the planning period and measurement. For example, all HMM-based frameworks require an n-gram language model, and one of the most common n-gram language

models can take several gigabytes of memory, making submission on a cell phone impractical. As a result, rather than the device being deployed locally, existing business ASR applications.

1.4.APPLICATIONS

Accordingly, CTC models can undoubtedly sort out some way to design talk acoustics for English characters, however they commit a great deal of fundamental spelling errors and need to depend on another dialect model to tidy up the papers. At that point, utilizing cosmically huge datasets, based on the work and exhibited some business footing in Chinese Mandarin and English. [Lip Net, the first start-to-finish sentence-level lip comprehension model, was released in 2016, and it used spatiotemporal convolutions and RNN-CTC engineering to outperform human-level execution in a restricted syntax dataset. Google Deep Mind launched CNN-RNN-CTC engineering with a huge reach in 2018. TAS stands for "Tune in, Attend, and Spell," and it is a model that "tunes in" to the acoustic sign, "considers" various sections of the sign, and "spells" out the record character by character. The ability to recognize voices varies depending on the make and model of the car. Real language discourse comprehension, rather than a preset series of instructions, is used in some of the more recent vehicle models. In the clinical field, talk affirmation might be performed at the front-end or back-finish of the clinical announcing stage. The despot is responsible for changing and affirming the record, which the provider guides into a talk affirmation motor. The voice is steered into a discourse acknowledgment unit by the supplier, and the perceived draught record is steered to the editorial manager alongside the first voice text, where the draught is changed and the report resolved. Conceded dialogue recognition is currently widely used in company.

The way that most EHRs haven't been explicitly altered to exploit voice-acknowledgment capacities is a more significant issue. The clinician's communication with the EHR is profoundly dependent on the reassurance and mouse, and course through the UI utilizing menus and tab/button clicks is a

significant piece of that. Voice-based course gives just slight ergonomic advantages. Then again, an enormous number of as of late overhauled radiology or pathology correspondence systems use voice "macros," in which the utilization of explicit articulations For instance, "ordinary report" will fill in countless default esteems and furthermore make a standard, which would shift dependent on the sort of assessment – for instance, a chest X-beam versus a gastrointestinal inconsistency game plan for radiology structure.

The results are promising, and the paper also provides material to the innovative work field for considering clinical documentation and vocabulary, as well as exhibition benchmarks and some management programming . The Mirage aeroplane programme in France, and various programmes in the United Kingdom handling a variety of aeroplane stages have all been noteworthy. Discourse recognizers have been successfully used in competitor aeroplanes in these programmes. Unlike what would be anticipated, no negative effects of the speakers' sloppy English were discovered. It was clear that unrestricted speech had fucked up the recognizer, which was to be expected. A limited jargon, or more every one of the, an appropriate grammar, could consequently be relied upon to improve acknowledgment precision generously.

The framework isn't utilized for a particular wellbeing or weapon-related errands, like weapon conveyance or bringing down the underside, yet it is utilized for a wide scope of other cockpit capacities. Visual and aural criticism are utilized to affirm voice orders. The framework is viewed as a critical arrangement for decreasing pilot responsibility, as it permits the pilot to disseminate centers to his plane with only two straightforward voice orders or to every one of his accomplices with only five.

In principle, Air regulator errands are additionally depicted as the critical yield of the regulator by exceptionally structured discourse, so reducing the trouble of the discourse acknowledgment undertaking should be possible. In certain cases, however, this is not the case. The FAA record 7110.65 clarifies the

expressions that air traffic controllers can use. Although the number of expressions supported by one of the reproduction merchants discourse acknowledgment frameworks is less than 150.

Regardless of the irrefutable level of coordination with word taking care of in every day individualized registering, ASR has not seen the standard extensions utilized in record improvement. Discourse recognition has become more affordable in mobile phones as portable processing speeds have improved.

1.5.PROBLEM DEFINITION

From the various applications, it is found that an efficient speech ranking system has to be established. The proposed work aims at developing an efficient speech ranking system using Convolutional neural networks. The performance of the proposed system is also to be measured in terms of metrics.

1.6.OBJECTIVE

The objectives are detailed as follows:

- To acquire the speech signals and preprocess the signals
- To identify a suitable forward transform and obtain the co-efficient
- To aggregate the co-efficient using appropriate statistical parameters
- To identify a suitable deep learning technique for classifying the speakers and hence perform the ranking
- To analyze the performance of the proposed classified

1.7. ORGANIZATION OF THE REPORT

This report is organized as follows: Section 2 provides the literature survey. Research database is given in Chapter 3. Proposed methodology is discussed in Chapter 4 and Chapter 5 concludes the research work.

CHAPTER 2

LITERATURE SURVEY

An exhaustive literature survey and the basic system design concepts are explained in this chapter. According to Neha Jain and Somya Rastogi (2020), in this country, people come with their natural ability to understand new things. Suppose people have to communicate by writing letters to each other, it's a real pain. And that is how people communicate with machines, because it would be so much better if computers could hear what people are thinking and if they could speak to them just to do stuff, and that people needed a very strong speech recognition system. Speech Recognition is a technique that can be used for the generation of text using a computer to recognize terms and phrases. The Speech Recognition System uses sound modeling and language modeling methods popularly called. Acoustic modeling constitutes the mathematical connection between the audio signal and phoneme linguistic segments, on the other hand language modeling describes the distribution of likelihood of words segments in the given word series. •Accuracy (percentage of speech and sentence conversion errors in text) •Speed (the extent to which the machine can keep up the pace with a human speaker) A device for speaking recognition is something you've dreamt of and been working on over decade. There are a range of software items that allow users to dictate their programmes and translate terms and phrases to text in a word processing document. Human beings will also use their speech orders to access control commands such as file opening, menu access, etc. Speech Recognition Technology will also assisted many individuals with disabilities who at times cannot write and now provide systems for speech recognition. But now people will just say those words to get what they need. And a kind of speech recognition script that implements speech recognition technologies, the mechanism that makes this possible.

The first Acoustic Mechanical Speech Machine was developed in 1784 by a student in Vienna (De Liang Wang, Jitong Chen, 2020) . Thus the first dictation system was invented in 1879 by Thomas Edison. Following the chain, Bell Laboratories in 1952 created the next method for the understanding of voice, which was capable of acquiring speaking numbers of 90%, but could only acknowledge the numbers spoken by the inventor. In 1970 a scholar was able to recognise more than 1000 speeches and could distinguish various pronunciations and sentences using the Harpy System. The recognition of speaking went on in the 80s with the advent of the Hidden Markov Model, which used a more mathematical approach in the analysis of sound waves. In 1986, the Hidden Markov model, which could be predictable in voice, was inaugurated in the 80's by IBM Tangora. In 2006, the NSA began to use speech recognition technology in the recorded speech to segment keywords. Then there was a surge in speech recognition technologies as Facebook, Google, Google Amazon, Microsoft and Apple, the world's top IT firms, began to sell this feature in different devices through services such as Google Home Amazon Echo and Apple. The objective of these leading technology firms is to have more exact answers to voice assistants. Yang Sun, Wen wu Wang, found the main senses achieve the perception of the outside world in the case of human beings. The visual and audio interpretation are the most critical of these. Nevertheless, speech has traditionally held a high priority in human conversation since it existed well before writing was introduced. As the main mode of communication between people, speech has the ability to be an effective mode of computer interaction. Speaking should be used from a distance such that the hand and eye can be busy in desired mode. In the world of robotics applications, signal processing technology is currently seeing rapid advances. The new field of speech signal processing, which led to phenomenal results, has greatly benefited from these advances. Speech processing has a great deal in the fields of defence, medicine, robotics and many other research and applications. Spoken word recognition, speech coding, speech synthesis and voice improvement are major fields covered in the field of word signals processing. A wearable electronic stick integrated with ASR systems, GPS and other connectivity devices will guide people with disabilities with the advancement of the wearable

electronic industry. Using speech acknowledgement provides links to blogs, dictation apps and others. Order, e-learning control guidance. Voice user interfaces (VUIs) use spoken technologies to enable users to access content, carry out the desired transactions and facilitate communication. VUIs rely on user input auto-language recognition. Since ASR applications use discourse as info, they keep away from or diminish the utilization of more regular manual information techniques (e.g., console, mouse), making them a feasible alternative for individuals with outrageous physical or neuromotor incapacities. VUIs permit human-PC exchanges that focus on a predefined assortment of discourse inputs that are required in a particular undertaking area and cover just a restricted range of human discussion. Analysts lean toward ASR due to its tweaked/customized approach in fields like safeguard, medication, robotization, and numerous others. A wide scope of ASR frameworks have been created because of the wide scope of possible applications. Because of their adaption to the pertinent application, discourse acknowledgment and discourse cognizance advancements have advanced into standard science.

II. Technique

Yaron Laufer and Sharon Gannot's VAD detection refers to the issue of the identification in an audio signal of speaker and non-speaker segments. It is the front end of many speech processing applications, including efficient speech recognition and low-bandwidth compression systems. Heavy and non-stationary noise present VAD systems with significant problems and research has been working on improving solid systems over recent years. Either the mixtures of voice and noise matching the application and marked with voice operation (monitored learning) or the noise data at most identical to the noise found in the application are used in a traditional modern VAD device (semi-supervised learning). In the above, methods implicitly presume that noise training data are usable since a noise model has to be initialised. The above semi-monitored approaches are based on parametric noise expectations (e.g. gaussianity), which can be severely infringed by non-stationary noise conditions. These advanced training details can be complex and laborious to collect.

It is also advantageous to build a VAD device, which is both unmonitored and resilient since it can work in different noise conditions across a broad spectrum of signal to noise ratios without training data. Previous VAD programmes, including G.729B and AMR, adopted a rule-based strategy, so no training data were needed.

They have been increasingly replaced by more stable and superior classification-based methods that include labeled training data. There was recently an interest in designing unattended VAD systems with supervised systems' efficiency benefits. The usual solution was to make current supervised and semi-monitored approaches more adaptive. 1 SW proposes a different method, based on a common paradigm in the source separation literature, the non-negative matrix factorization (NMF). Contrary to the above VAD methods, we model the sound blend directly (speech and noise). This has the drawback that the technique works in any noise situation provided one has a fair general model for expression. We will explain in depth how a universal language model is achieved in the next section, but this model is normally trained by a variety of speakers in a clean speech database. Once mastered, it can be used to detect speech in any noise context (from any unavailable speaker). Therefore, it is unattended from a customer point of view as the device is deployed. Our methodology also has the advantage of being entirely understandable: if you were to use this model for distinction, the characteristics used for classification precisely correspond to the relative levels of speech and noise.

Natural linguistic processing (NLP) is an Artificial Intelligence and Linguistics Tract dedicated to machine understanding of human language claims or sentences. In order to simplify the work of the user and fulfil the desire to connect with the machine naturally, the natural language processing came about. Since not all users are familiar with a single computer language, NLP offers users who don't have enough time to understand or master new languages. A collection of rules or a set of symbols may be described as a language. The symbol is merged and used for information transmission or transmission. The Rules tyrannise symbols. Natural linguistic processing (NLP) is an Artificial Intelligence and Linguistics Tract dedicated to machine understanding of human language claims or sentences. In order to simplify the work of the user and fulfil the desire to connect with the machine naturally, the natural language processing came about. Since not all users are familiar with a single computer language, NLP offers users who don't have enough time to understand or master new languages. A collection of rules or a set of symbols may be described as a language. The symbol is merged and used for information transmission or transmission. The Rules tyrannise symbols. Natural language processing can

essentially be divided into two parts, namely natural linguistic understanding and the generation of natural languages, which is the objective. Linguistics is a language science that involves sound phonology, word composition of morphology, the sentence construction of Grammar, the Semantics syntax and the interpretation of pragmatism. This can be split into two categories: higher speech recognition and lower natural language recognition. Automatic synopsis, resolution of co-references, discourse analysis, machine translation, morphological segmentation, name of individuals identification, recognition of optical character, part of speech synopsis, and so on are some of NLP's research activities. Machine translation, recognition of named entities, detection of optical characters, and other real-world applications are all possible with any of these features. An automatic description of a collection of text is produced in an understood way, including summaries or descriptive information on a known class of text.

It means a phrase or greater number of text, co-reference resolution, which specifies which word corresponds to the same object. Discourse analysis refers to the process of defining a linked text discourse system. Translations from one human language to another, automated translation of the text. Morphologic division which refers to individual morphemes separating word and identifying the grade of morphemes. A recognition entity named (NER), the text stream is defined and the objects in a text correspond to proper names. It provides an illustration of impressed text, Optical Character Recognition (OCR), which helps to define a matching or associated text. Part of the language tag defines a sentence which specifies the part of the language in a word. Although NLP tasks are clearly interwoven quite closely, they are often used for convenience. The task used for solventing bigger tasks, including automated summarisation, co-reference analysis, etc. Natural language processing's aim is to satisfy one or more specialisations' algorithms or methods. The human language technology metric, which is based on an algorithmic scheme, allows for the combining of language comprehension and language generation. It's also used to detect multilingual events. Rospocher et al. devised a one-of-a-kind standard method for crosslingual event extraction for English, Dutch, and Italian texts by employing distinct pipelines for each language. A common collection of leading multilingual linguistic communication process (NLP) tools is included in the scheme.

As a result, the cross-lingual structure enables the understanding of activities, participants, places, and time, as well as their relationships. Individual pipeline output is intended to be used as input for a device that generates event-centric data graphs. All modules act like UNIX machine pipes: they all take standard input, process it, and output standard output, which is then used as the input for subsequent module pipelines. Furthermore, modular architecture allows for a variety of configurations as well as dynamic delivery. The majority of the add linguistic communication method is carried out by computer scientists, but other experts such as linguists, scientists, and philosophers have also expressed interest. The fact that human language technology contributes to the data of human language is one of the most ironic aspects of it. The field of linguistic communication is said with a variety of theories and techniques that trot out the subject of linguistic communication of computer communication. Ambiguity is a major flaw in linguistic communication that is often encountered at the grammar level, which includes subtasks such as lexical and morphology, which are concerned with the analysis of words and word formation. Any one of those levels will reveal ambiguities that can be resolved using the entire sentence's data. The anomaly can be resolved in a variety of ways, including minimization, conservation, interactive disambiguation, and weight uncertainty. Conserving uncertainty is one of the methods proposed by researchers to eliminate ambiguity.

As examined in and, four significant examination groups in discourse acknowledgment have discovered that DNN-based frameworks give altogether higher precision in nonstop telephone and word acknowledgment than the previous reformist GMM-based frameworks and the prior shallow neural organization frameworks in the course of the most recent couple of years. It is found that still there is a significant scope for research in the area of automated speech recognition and speaker recognition. MATLAB is used for the development of speech ranking software. Deep Learning toolbox is used for the development of LSTM based classifier.

CHAPTER 3

RESEARCH DATABASE

In order to perform the research work, an extensive research database has been created. It consists of 330 speech signals in three different languages namely Tamil, Telugu and English. In each language for every speaker, a total of ten samples are collected. While recording speech signals, speakers are advised to change their timbre and pitch.

Speech samples are shown in Figure 3.1. From subjective analysis, it is evident that these signals are non-stationary in nature. They also have bursts at certain intervals of time. Also each signal has voiced unvoiced and silent regions. Duration between these three regions is different for different samples. Though the speech signals shown in Figure 3.2 belong to a same speaker, these signals have different characteristics. However there is/are inherent feature/features that/which are common to these three signals (not evident to human eye).

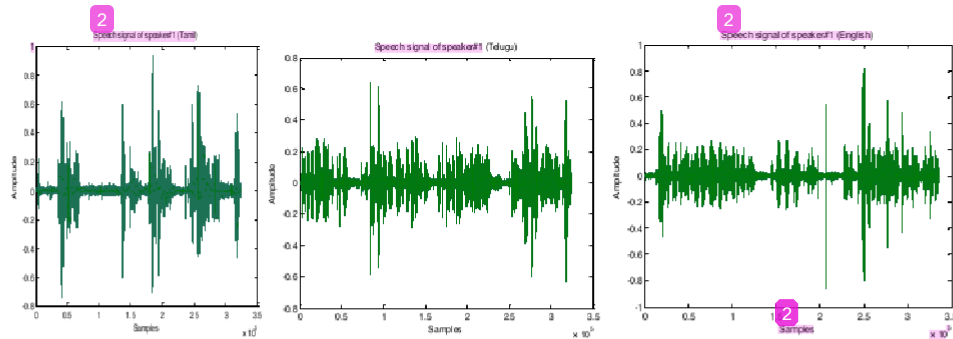


Figure 3.1. speech samples of Speaker #1 (English, Tamil and Telugu)

These characteristics can be represented using appropriate statistical parameters. Instead of considering the entire speech signal for feature extraction, 1024 samples of the signal are considered. On the preprocessed signal, short time energy is calculated. As the preprocessed signal consists of voiced, unvoiced and silent region, the determined short time energy can be regarded as the indicator of speech signal characteristics.

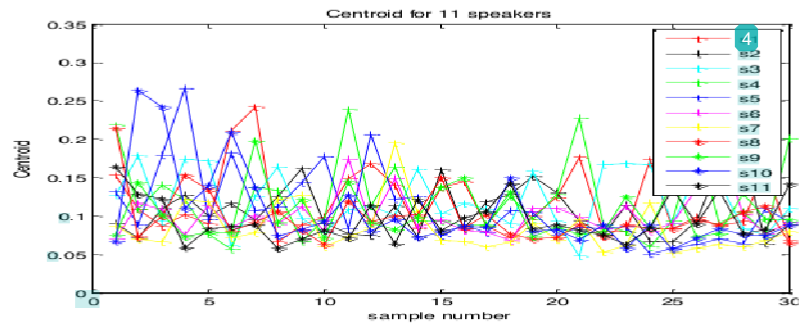
Flux and centroid (indicators of fundamental frequency) are obtained on Fourier Transform co-efficients. Table 3.1 shows these parameters for eleven speakers (three different languages).

Table 3.1 Short time Energy, centroid and Flux for speech signals

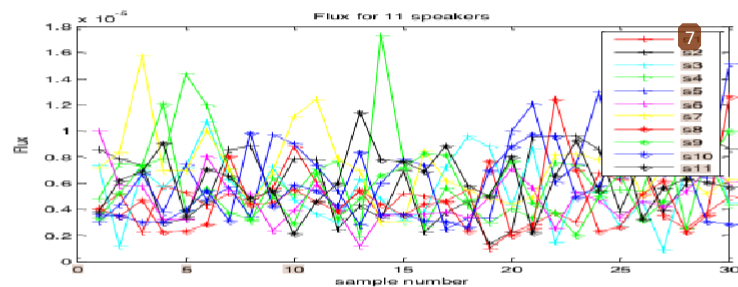
Speaker	Language	Centroid	Short Time Energy ($\times 10^6$)	Flux
1	English	0.153334	4.02	0.000476
	Tamil	0.149357	6.01	0.000308
	Telugu	0.176847	2.81	0.000779
2	English	0.090358	8.54	0.000257
	Tamil	0.077693	7.77	0.000256
	Telugu	0.093139	9.49	0.000395
3	English	0.127732	7.37	0.000196
	Tamil	0.107561	3.63	0.000151
	Telugu	0.048059	8.63	.0000980
4	English	0.218448	4.78	0.000898
	Tamil	0.239241	7.09	0.000327
	Telugu	0.227249	3.29	0.000856
5	English	0.131802	3.00	0.000672
	Tamil	0.083781	7.31	0.000335
	Telugu	0.085384	12.0	0.000239
6	English	0.070248	10.0	0.000224
	Tamil	0.17439	5.89	0.000971
	Telugu	0.098999	5.60	0.000408
7	English	0.087865	7.14	0.000197
	Tamil	0.073417	1.24	.0000198
	Telugu	0.095396	4.89	0.000363
8	English	0.214483	4.02	0.000942
	Tamil	0.118954	4.55	0.00028
	Telugu	0.088237	2.49	0.000374
9	English	0.075238	3.28	0.000257
	Tamil	0.144989	6.69	0.000435
	Telugu	0.076583	4.44	.0000454
10	English	0.067179	3.58	0.000247

11	Tamil	0.126979	5.38	0.000154
	Telugu	0.069022	9.72	0.0000995
	English	0.16457	3.74	0.00067
	Tamil	0.071261	4.56	0.000382
	Telugu	0.080976	2.18	0.000252

From the Table 3.1, it is found that the parameters flux, centroid and short time energy do not have linear relationship with the speaker i.e range cannot be fixed for individual speakers. It is because values overlap among the speakers. The above fact is also visible in the following graphs shown in Figure 3.2 a), b) and c). It is because, Fourier transform cannot capture the spectral co-efficients of a random signal.

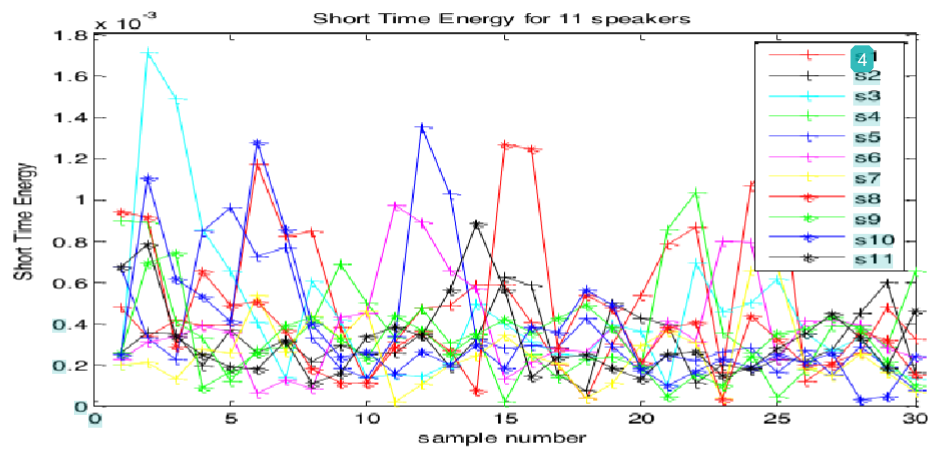


(a)



(b)

Figure 3.2 (Continued)



(c)

Figure 3.2 Relationship between the speakers and a) Centroid b) Flux c) Short time energy

From the above, it is evident that time domain analysis cannot be used for speech ranking system and hence , spectral domain techniques are to be used.

CHAPTER 4

RNN AND LSTM BASED IDENTIFICATION OF SPEECH SIGNALS

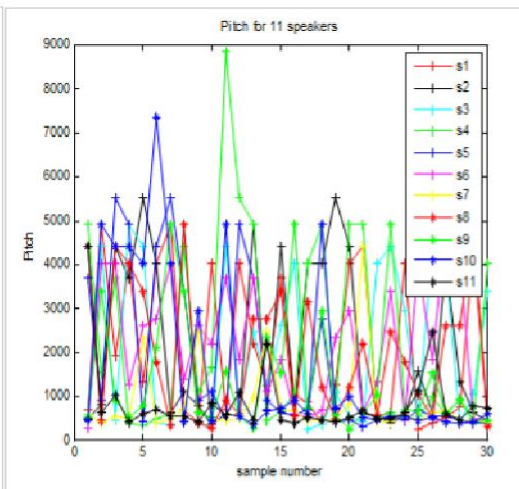
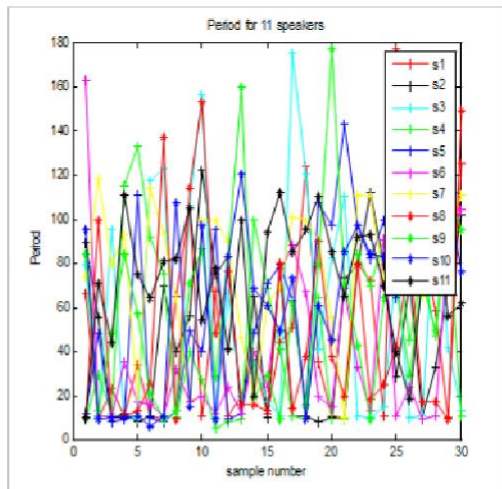
The two key modules in any speaker recognition system are feature extraction and classification. In the proposed work, autocorrelation features, Approximation statistical features and detailed statistical features are obtained from the speech signals. Discrete Meyer wavelet (dmey) is the wavelet used for the extraction of features. The extracted features are given to a classifier to identify the speaker based on the features. In order to improve the performance of the classifier the inter relationship among the features is to be considered. RNN and LSTM are the two classifiers considering the relationship between the input features.

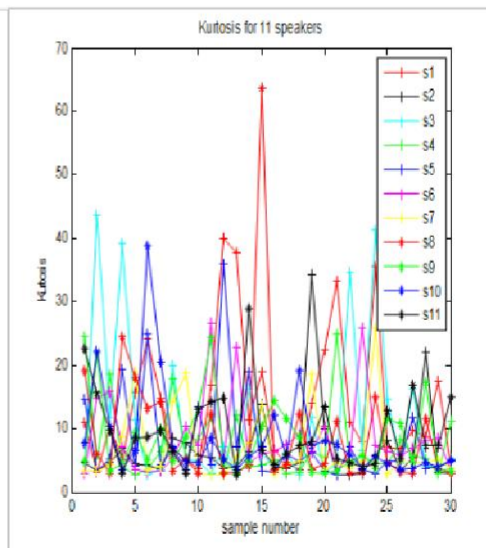
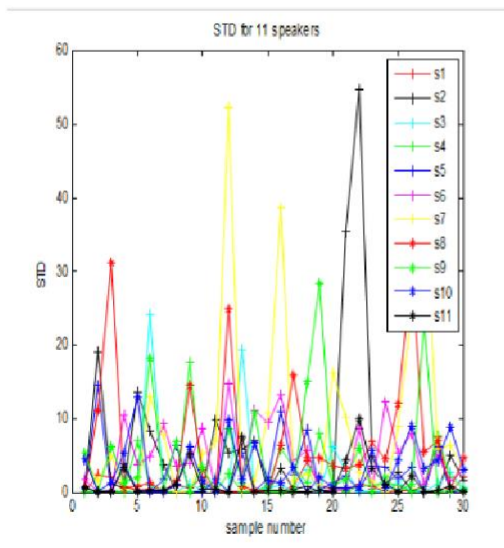
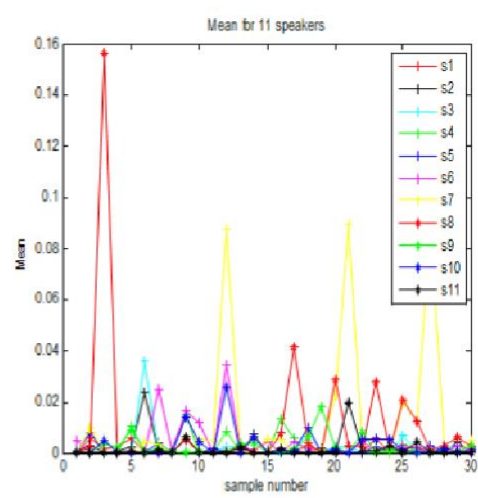
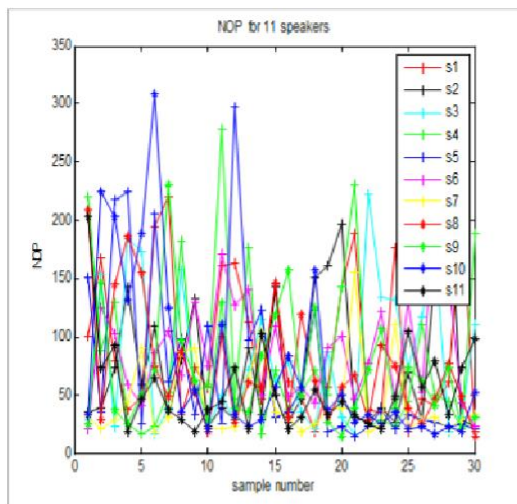
4.1. Recurrent Neural Network (RNN)

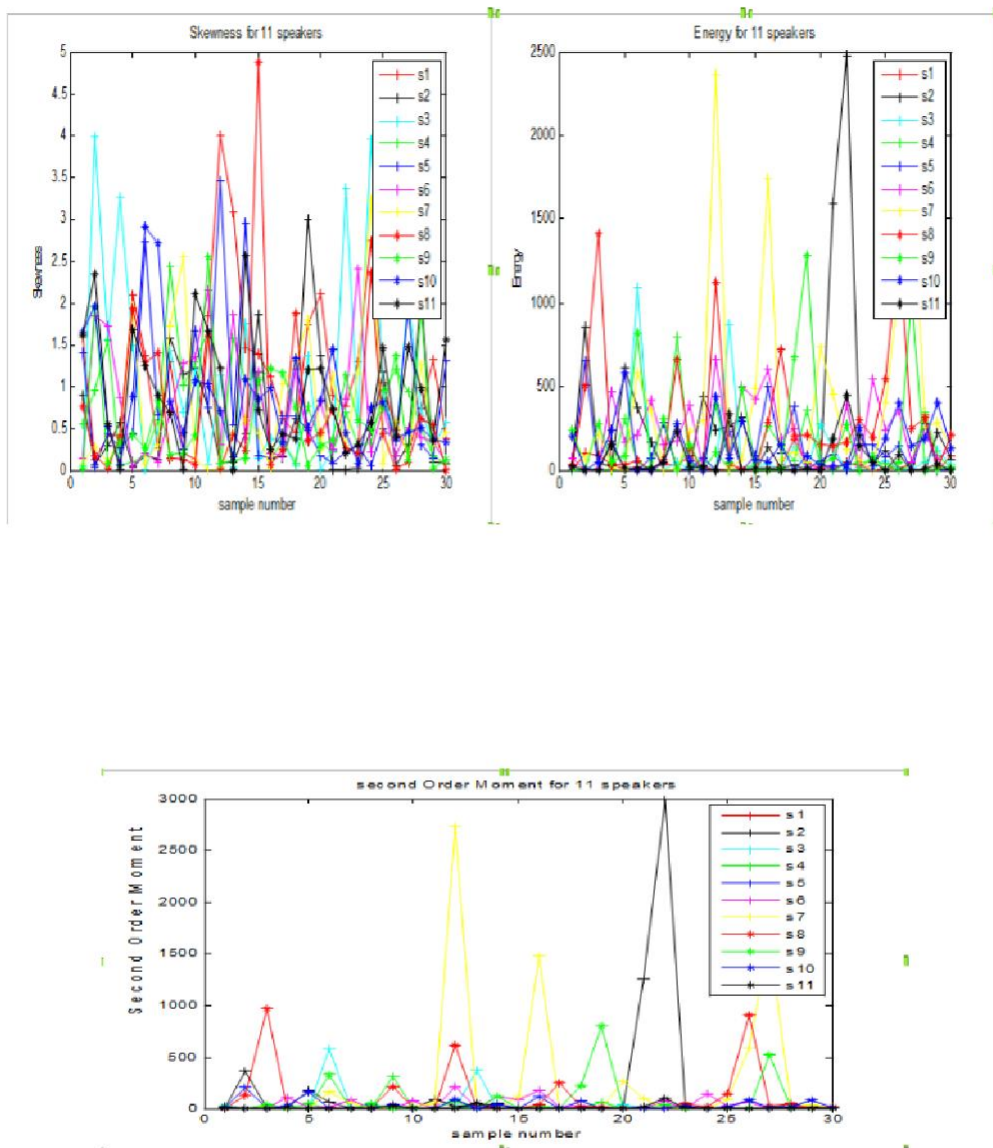
RNN is one of the subclass of Deep Learning Neural Network. The RNN architecture used in our work comprises of one hidden layer with ten neurons. Efficient learning for any complicated classification work is performed only by the hidden layer. For RNN based Speaker recognition system, nine speakers signal is considered. Features namely Pitch, Period, Number of Peaks, Mean, Skewness and Kurtosis were extracted from autocorrelation coefficients and given to RNN for speaker classification. In RNN, one feature is given as the input and output is predicted at an instant. Likewise, six features are given and the output is the corresponding speaker identified. Out of 270 signals, for training 135 signals and for testing 135 signals are used.

For LSTM based speaker recognition system eight speakers signal is considered. Out of 30 speech signals from each speaker, 15 are considered for training and 15 are considered for testing. The proposed LSTM approach is applied with four different classes of inputs for analyzing suitable features for speaker recognition system.

Class I : Statistical boundaries in particular ²Pitch, ²Period, NOP, Mean, Standard Deviation, Kurtosis, Skewness, Energy, ZCR over autocorrelation coefficients increased by a factor of 100. Class II : Statistical boundaries to be specific ²Pitch, ²Period, NOP, Mean, Standard Deviation, Kurtosis, Skewness, Energy, ZCR, ²Second Order Moment and ²Third Order Moment over autocorrelation coefficients. Class III : Statistical boundaries to be specific ²Pitch, ²Period, NOP, Mean, Standard Deviation, Kurtosis, Skewness, Energy, ZCR over autocorrelation coefficients duplicated by a factor of 1000. Class IV : Ten Statistical boundaries specifically ²Pitch, ²Period, NOP, Mean, Standard Deviation, Kurtosis, Skewness, Energy, ZCR, ²Second Order Moment over the Detailed coefficients of Discrete meyer wavelet coefficients.







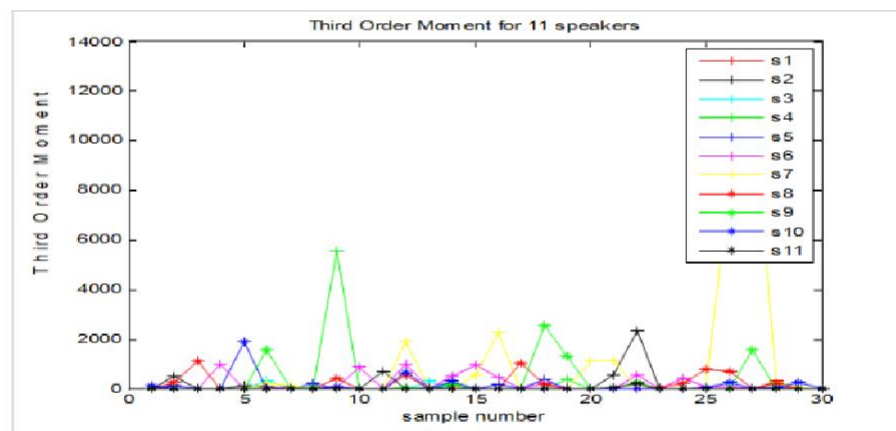
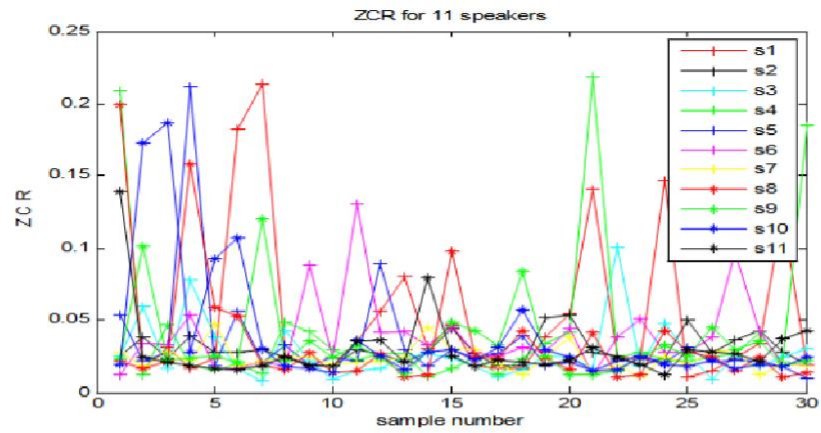
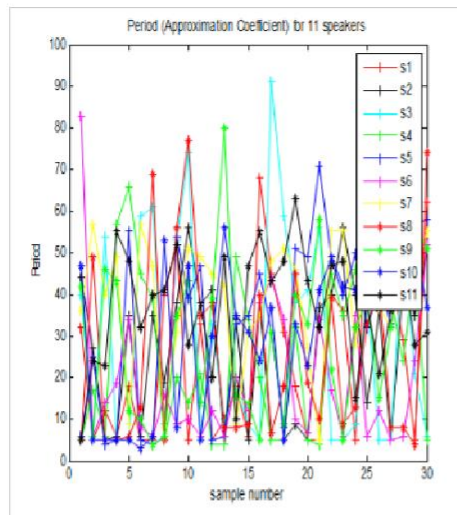
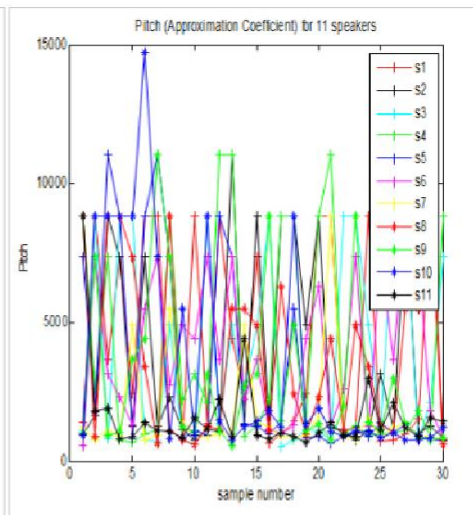


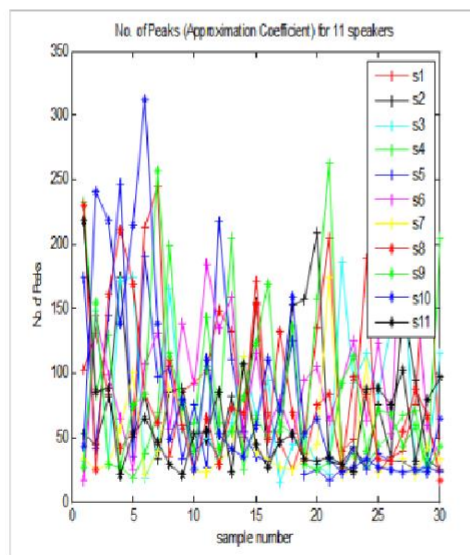
FIG.4.1 Class II parameters for representing speech signals (Autocorrelation)



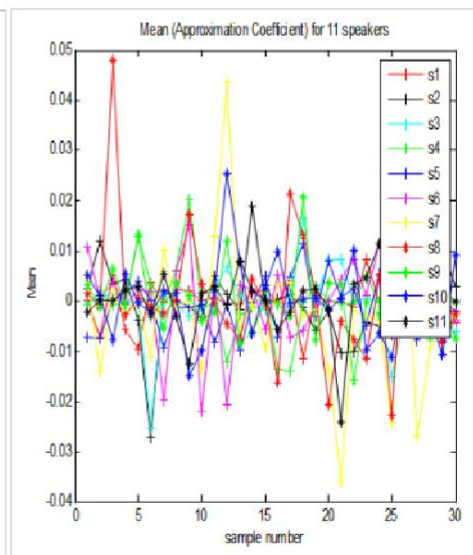
(a)



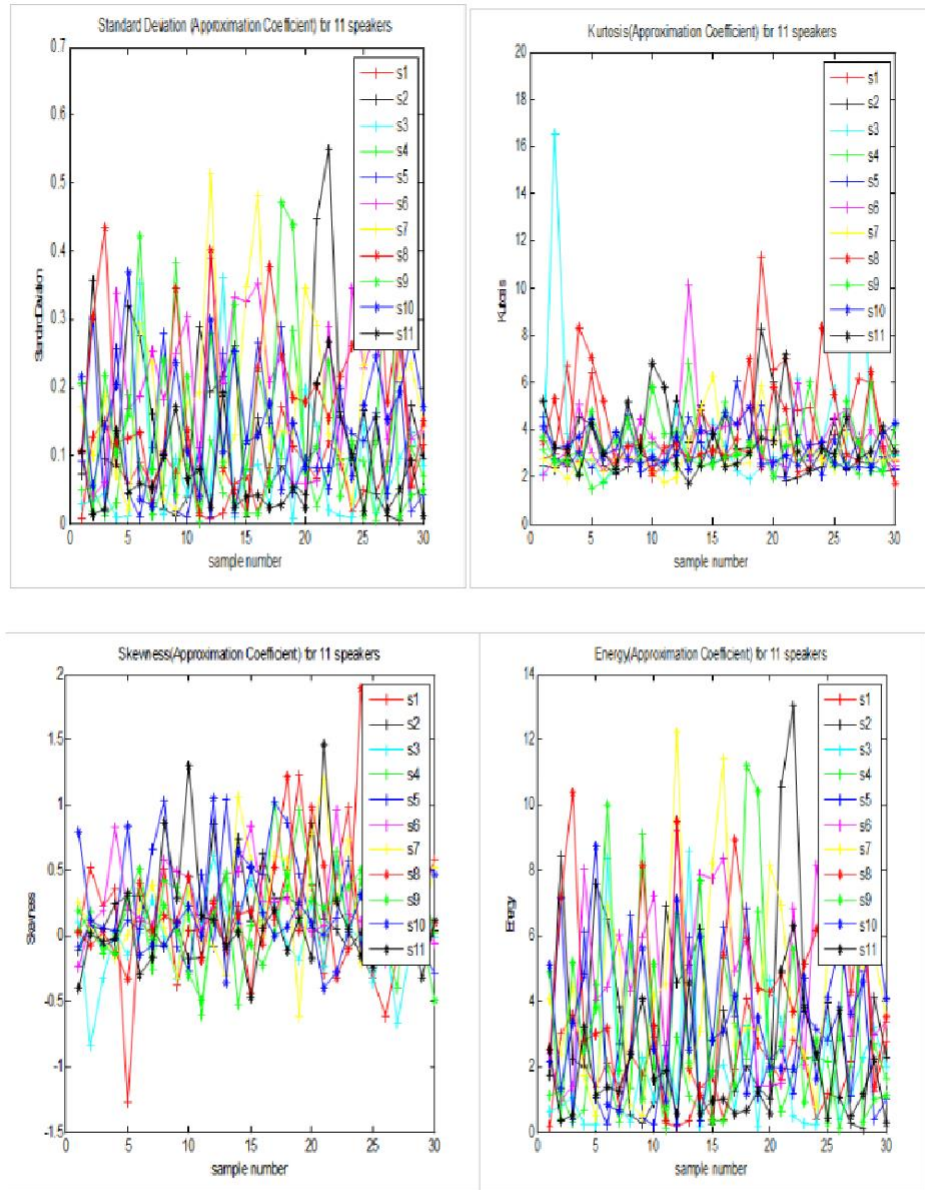
(b)

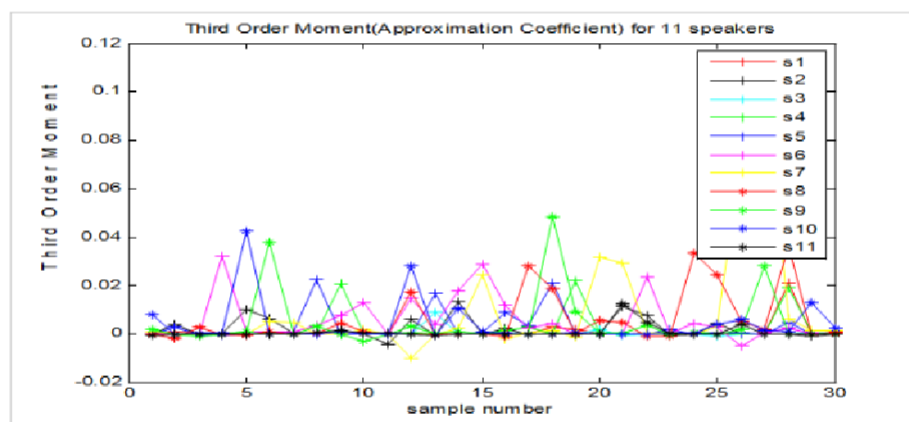
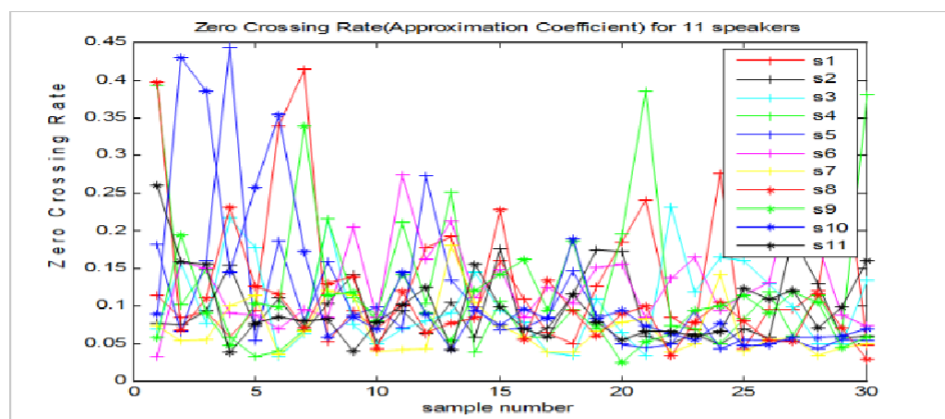


(c)



(d)





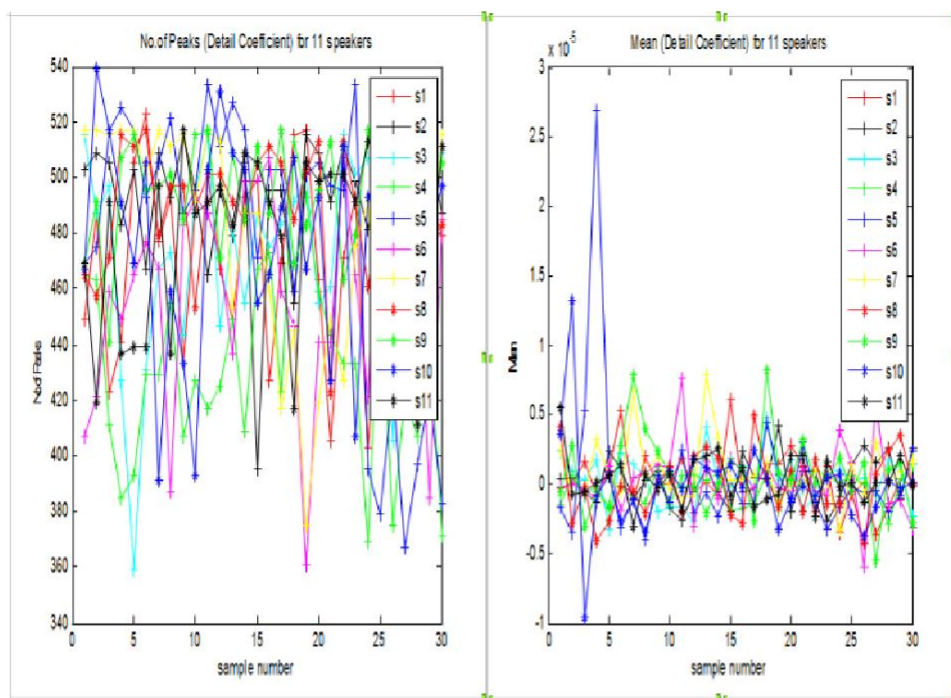
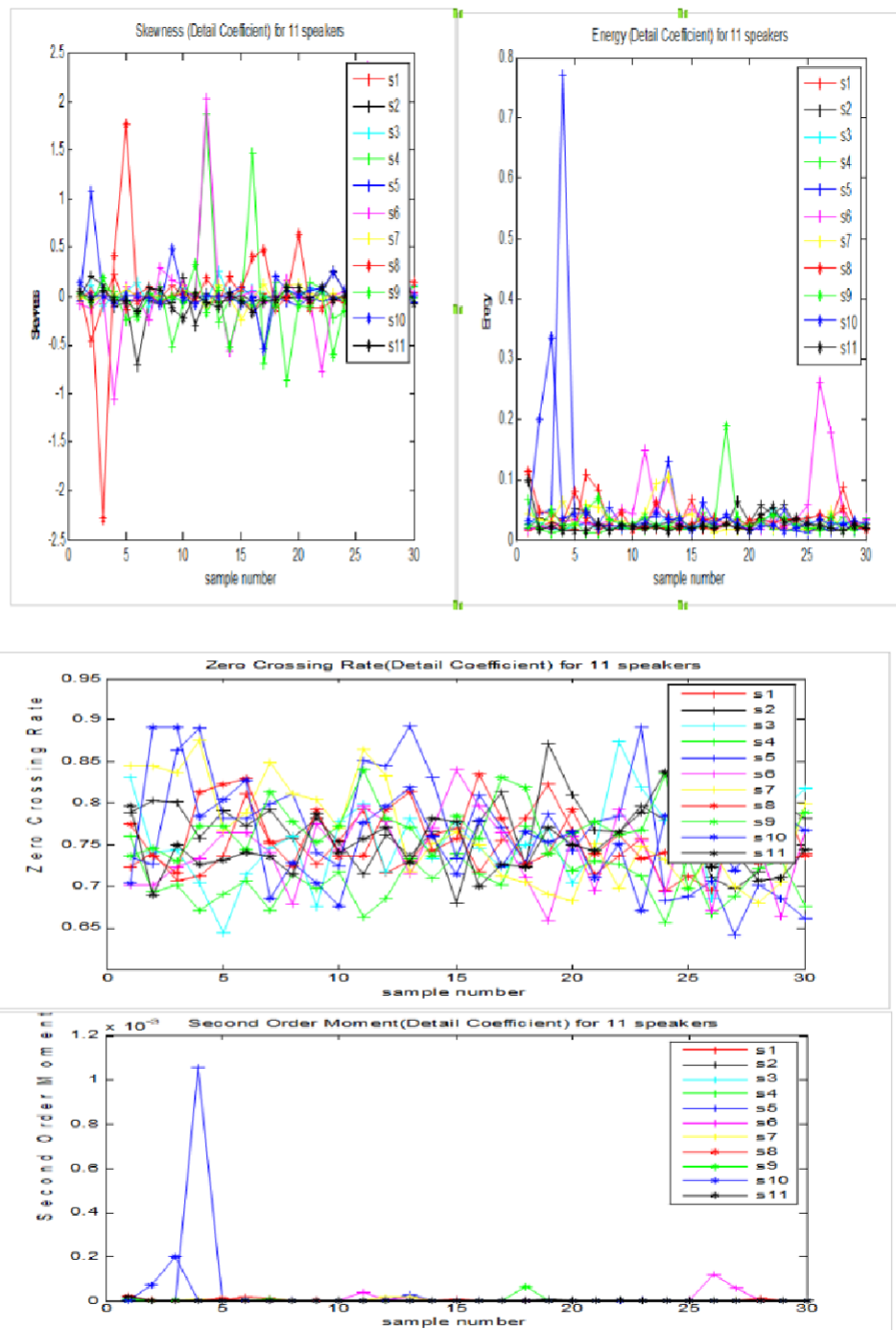
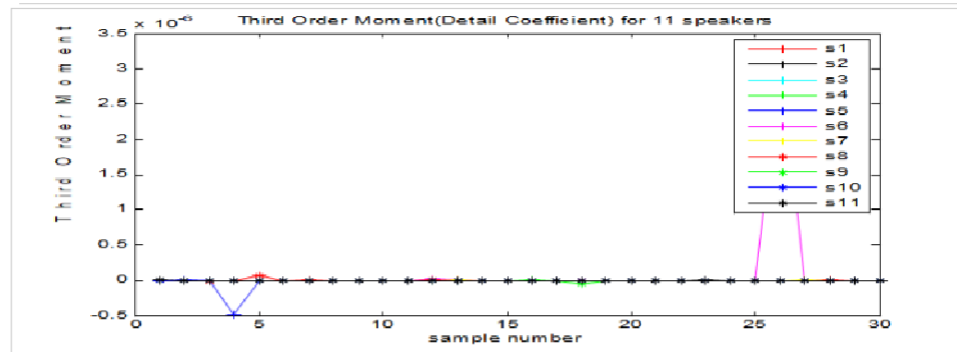


FIG 4.2 Class IV parameters for representing speech signals (Discrete Meyer Wavelet)





In Recurrent Neural Network (RNN) based speaker Recognition System, six statistical features of nine speakers are used . Using Confusion matrix , Sensitivity is calculated for all the nine speakers. The computed sensitivity values for all the speakers with the extracted features is shown in Table 1. From Table 1, it is observed that the statistical feature Skewness identifies all the speakers, whereas other features fails to identify all the speakers.

Table 1: Sensitivity using RNN Classifier

Speaker	pitch	period	NOP	Mean	Skewness	kurtosis
1	23	0	0	15.3	15.3	39
2	29	14.2	0	0	93	2
3	44	3	19	13	75	2
4	60	2	2	20	47	0
5	33.3	2	0	2	2	0
6	0	14	20	20	13.3	0
7	0	20	2	20	27	13.3
8	0	33.3	20	0	20	13.33
9	0	2	0	0	2	40

In LSTM network is applied with four different classes of inputs. For all the classes of inputs, confusion matrix is formed through which sensitivity is also calculated. The computed sensitivity values for all the speakers for all the set of input classes is shown in Table 2. From Table 2, it is observed that the statistical features from detailed coefficients of discrete meyer wavelet gives better results compared to other classes of inputs from other coefficients.

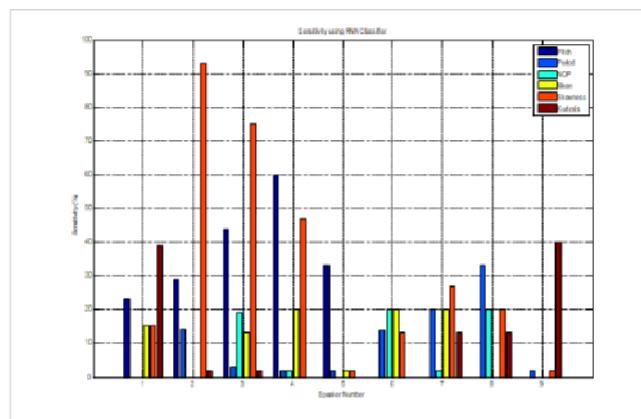


Fig.4.3Sensitivity Comparison for various speakers using RNN Classifier

Table 2: Sensitivity using LSTM Classifier

Speaker	Class I	Class II	Class III	Class IV
1	33.3	6.67	46.67	40
2	13.3	0	0	13.33
3	6.67	27	13.33	6.67
4	6.67	13.3	6.67	13.33
5	13.3	13.3	6.67	20
7	40	20	6.67	40
8	0	13.3	20	40
9	13.3	0	20	13.33

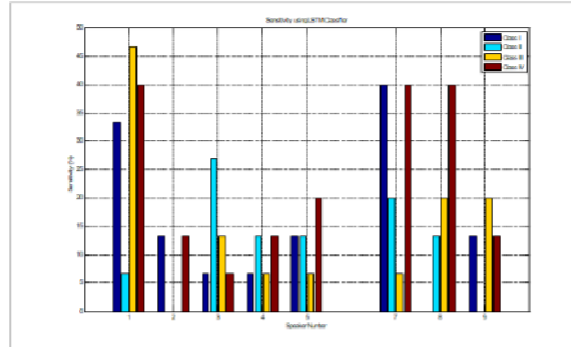


Fig. 2. Sensitivity Comparison for various speakers using LSTM Classifier

From the Tables, it is clear that LSTM classifier provides the best output for Class IV exemplars. That is the Discrete Meyer co-efficient provide the highest inter class variance and less intra class variation.

4.2 Proposed Work

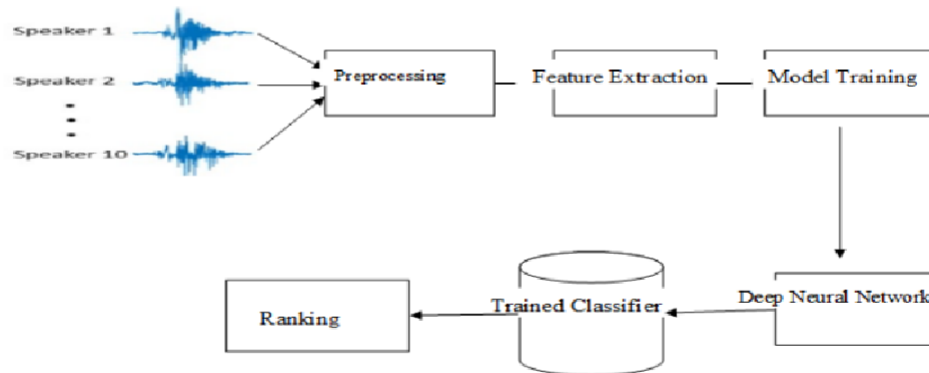


FIG 4.5 BLOCK DIAGRAM

The discussion signal is given as an information. By then the sound data is pre-dissected and the outstanding part habitats are dealt with in the enlightening file. Discrete Wavelet Transform (DWT) is utilized for the part extraction to discard the undesirable rehash of the signs. At long last the critical learning procedure is utilized to rank the sound sign from the speakers dependent on voice sign and mistake rate.

4.3 RESULT

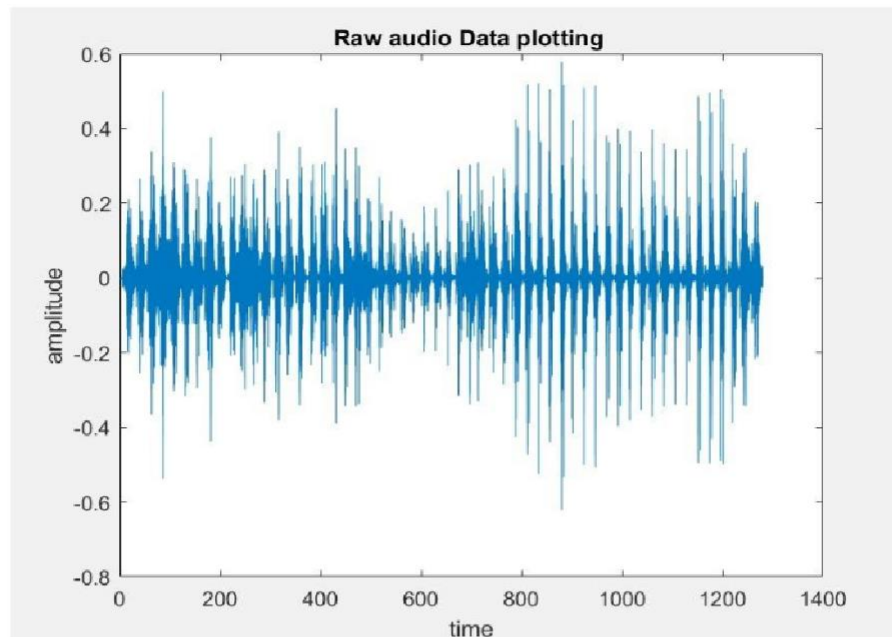


FIG 4.6 RAW AUDIO DATA PLOTTING

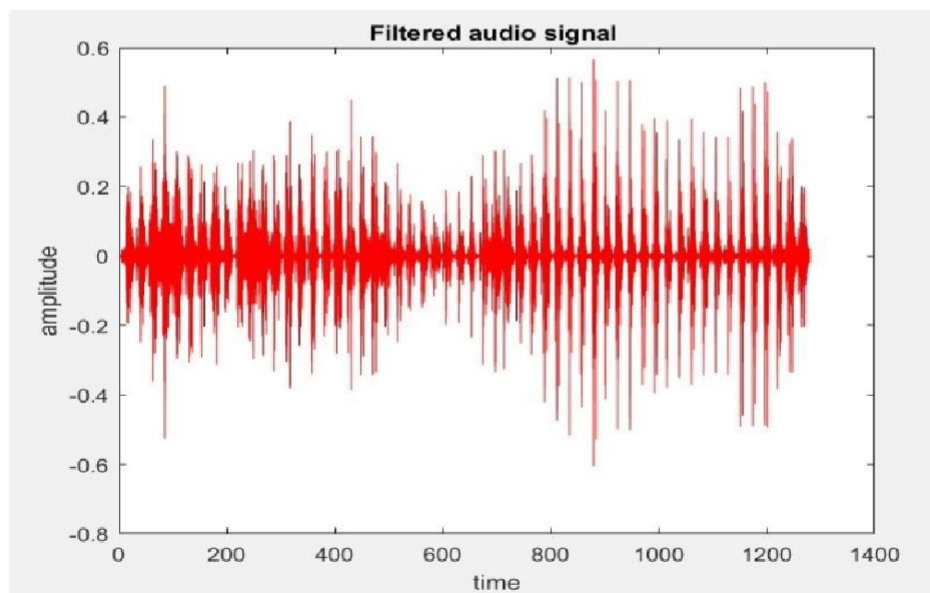


FIG 4.7 FILTERED AUDIO SIGNAL

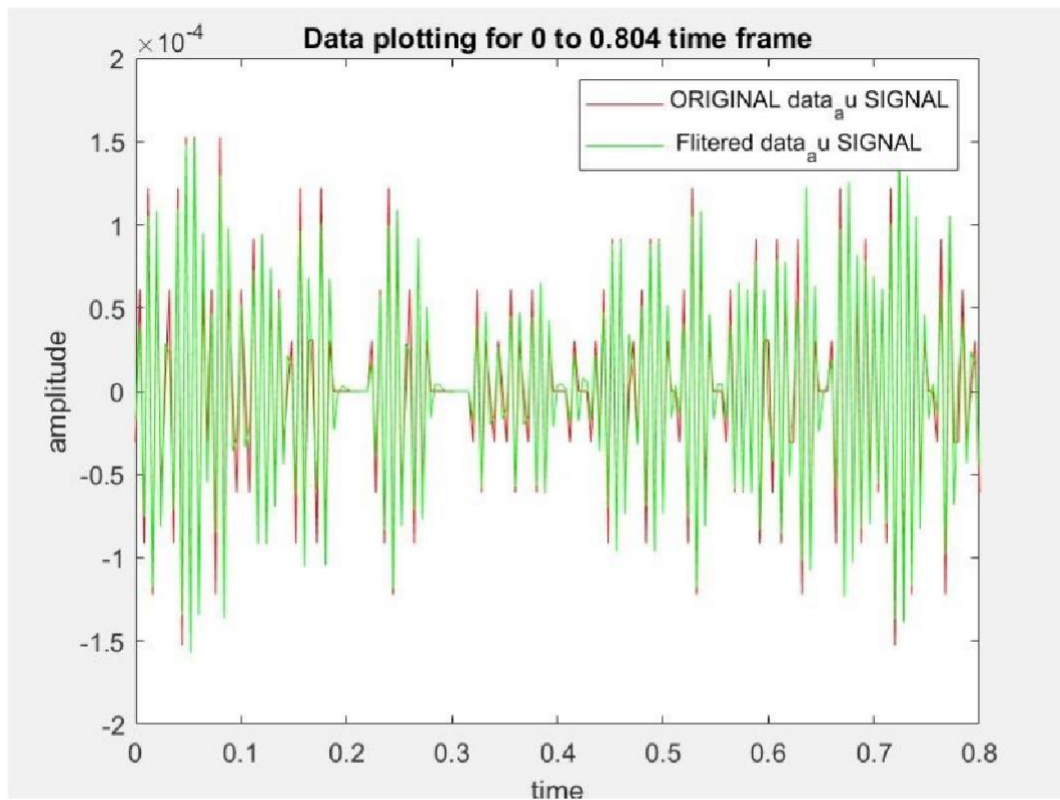


FIG 4.8 DATA PLOTTING FOR 0 TO 0.804 TIME FRAME

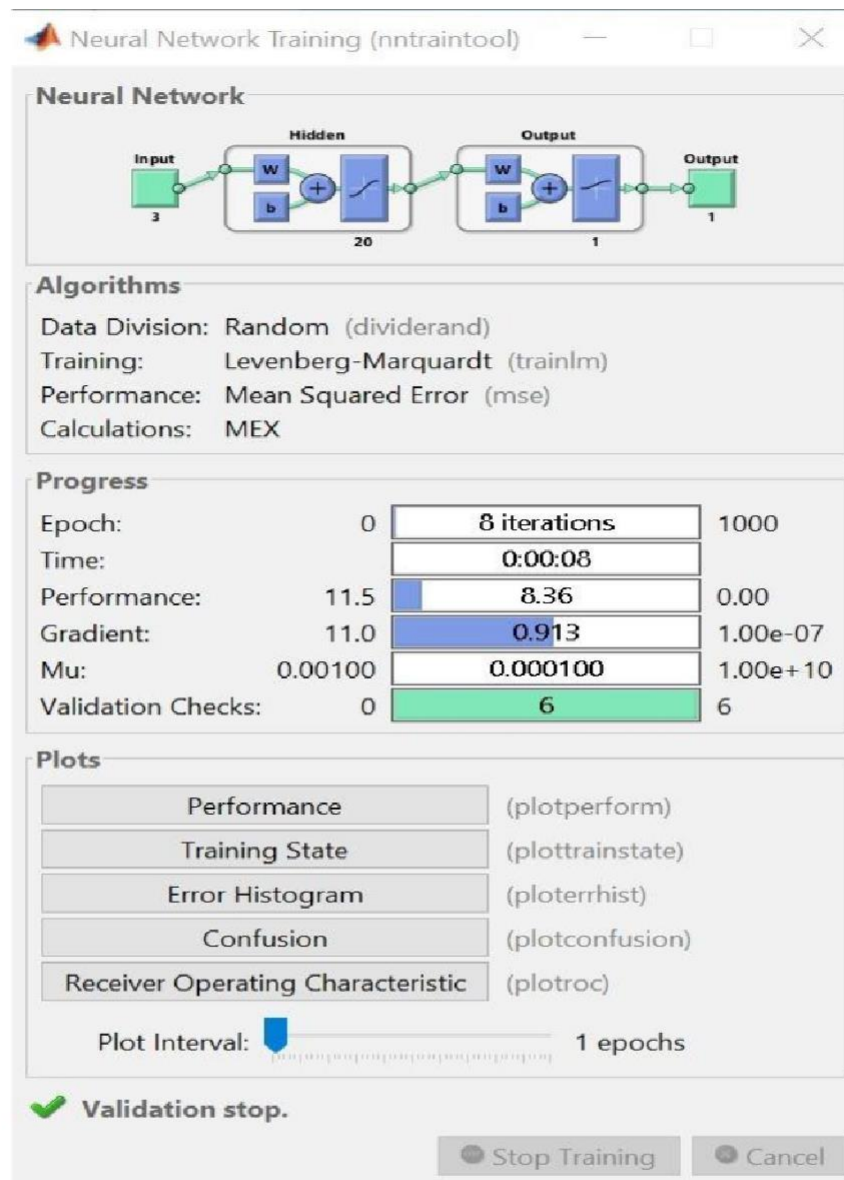


FIG 4.9 NEURAL NETWORK TOOL BOX

CHAPTER 5

CONCLUSION AND FUTURE WORK

Speaker recognition has a very important role in various applications like forensic department for identification of anti-social elements, biometric authentication systems etc. In this paper, various features involved in development of automated speaker recognition system is identified with an extensive real time research database. Also the two categories of Deep Learning neural network classifier namely RNN and LSTM is also used for speaker recognition. Based on the sensitivity calculations it is identified Skewness and the statistical features over the detailed coefficients of discrete meyer wavelet gives better results. These features are fed to RNN and LSTM for speaker recognition. LSTM has higher sensitivity and accuracy than RNN.

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