



**ACOUSTIC ANALYSIS OF SPEAKER USING ANSYS**

##### A MINOR PROJECT - III REPORT

###### ***Submitted by***

|  |  |
| --- | --- |
| **SETHU PRIAN VM** | **927621BEC193** |
| **SRIHARI M** | **927621BEC211** |
| **BALAJI N** | **927621BEC022** |
| **THIRUPUKAL G** | **927621BEC231** |

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**BONAFIDE CERTIFICATE**

Certifiedthatthis **18ECP105L - Minor Project III** report **ACOUSTIC ANALYSIS OF SPEAKER USING ANSYS** is the bonafide workof **SETHUPRIAN VM (927621BEC193), SRIHARI M (927621BEC211), BALAJI N(927621BEC022),THIRUPUKAL G (927621BEC231)** who carried out the project work under my supervision in the academic year **2023-2024 - ODD**.

**SIGNATURE SIGNATURE**

**Dr.A.KAVITHA B.E., M.E., Ph.D.,** **Dr.A.MURUGAN M.E., Ph.D.,**

**HEAD OF THE DEPARTMENT, SUPERVISOR,**

Professor, Professor,

Department of Electronics and Department of Electronics and

Communication Engineering, Communication Engineering,

M.Kumarasamy College of Engineering, M.Kumarasamy College of Engineering, Thalavapalayam, Thalavapalayam,

Karur-639113. Karur-639113.

This report has been submitted for the **18ECP105L – Minor Project-III** final review held at M. Kumarasamy College of Engineering, Karur on **13-10-2023.**

**PROJECT COORDINATOR**

**INSTITUTION VISION AND MISSION**

**Vision**

To emerge as a leader among the top institutions in the field of technical education.

**Mission**

**M1:** Produce smart technocrats with empirical knowledge who can surmount the global challenges.

**M2:** Create a diverse, fully -engaged, learner -centric campus environment to provide quality education to the students.

**M3:** Maintain mutually beneficial partnerships with our alumni, industry and professional associations

**DEPARTMENT VISION, MISSION, PEO, PO AND PSO**

**Vision**

To empower the Electronics and Communication Engineering students with emerging technologies, professionalism, innovative research and social responsibility.

**Mission**

**M1:** Attain the academic excellence through innovative teaching learning process, research areas & laboratories and Consultancy projects.

**M2:** Inculcate the students in problem solving and lifelong learning ability.

**M3:** Provide entrepreneurial skills and leadership qualities.

**M4:** Render the technical knowledge and skills of faculty members.

**Program Educational Objectives**

**PEO1:** **Core Competence:** Graduates will have a successful career in academia or industry associated with Electronics and Communication Engineering

**PEO2:** **Professionalism:** Graduates will provide feasible solutions for the challenging problems through comprehensive research and innovation in the allied areas of Electronics and Communication Engineering.

**PEO3:** **Lifelong Learning:** Graduates will contribute to the social needs through lifelong learning, practicing professional ethics and leadership quality

**Program Outcomes**

**PO 1: Engineering knowledge:** Apply the knowledge of mathematics, science, engineering fundamentals, and an engineering specialization to the solution of complex engineering problems.

**PO 2: Problem analysis:** Identify, formulate, review research literature, and analyze complex engineering problems reaching substantiated conclusions using first principles of mathematics, natural sciences, and engineering sciences.

**PO 3: Design/development of solutions:** Design solutions for complex engineering problems and design system components or processes that meet the specified needs with appropriate consideration for the public health and safety, and the cultural, societal, and environmental considerations.

**PO 4: Conduct investigations of complex problems:** Use research-based knowledge and research methods including design of experiments, analysis and interpretation of data, and synthesis of the information to provide valid conclusions.

**PO 5: Modern tool usage:** Create, select, and apply appropriate techniques, resources, and modern engineering and IT tools including prediction and modeling to complex engineering activities with an understanding of the limitations.

**PO 6: The engineer and society:** Apply reasoning informed by the contextual knowledge to assess societal, health, safety, legal and cultural issues and the consequent responsibilities relevant to the professional engineering practice.

**PO 7: Environment and sustainability:** Understand the impact of the professional engineering solutions in societal and environmental contexts, and demonstrate the knowledge of, and need for sustainable development.

**PO 8: Ethics:** Apply ethical principles and commit to professional ethics and responsibilities and norms of the engineering practice.

**PO 9: Individual and team work:** Function effectively as an individual, and as a member or leader in diverse teams, and in multidisciplinary settings.

**PO 10: Communication:** Communicate effectively on complex engineering activities with the engineering community and with society at large, such as, being able to comprehend and write effective reports and design documentation, make effective presentations, and give and receive clear instructions.

**PO 11: Project management and finance:** Demonstrate knowledge and understanding of the engineering and management principles and apply these to one’s own work, as a member and leader in a team, to manage projects and in multidisciplinary environments.

**PO 12: Life-long learning:** Recognize the need for, and have the preparation and ability to engage in independent and life-long learning in the broadest context of technological change.

**Program Specific Outcomes**

**PSO1:** Applying knowledge in various areas, like Electronics, Communications, Signal processing, VLSI, Embedded systems etc., in the design and implementation of Engineering application.

**PSO2:** Able to solve complex problems in Electronics and Communication Engineering with analytical and managerial skills either independently or in team using latest hardware and software tools to fulfil the industrial expectations.

|  |  |
| --- | --- |
| **Abstract** | **Matching with POs, PSOs** |
| **<<Abstract keywords>>** | **<<PO1, PO2, PO3, PO4, PO5, PO6, PO7, PO8, PO9, PO10, PO11, PO12, PSO1, PSO2>>** |

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**ABSTRACT**

The acoustic analysis of a speaker using ANSYS is a crucial process in understanding and improving the speaker's performance. ANSYS provides a versatile platform for simulating and assessing various aspects of speaker design and functionality.In this analysis, ANSYS utilizes computational techniques to model the speaker's components, including the diaphragm, voice coil, magnet structure, and enclosure. These components interact in complex ways to produce sound, and ANSYS allows engineers to simulate these interactions to gain insights into the speaker's behavior.One key aspect of the analysis is the evaluation of the speaker's frequency response. ANSYS helps in predicting how the speaker reproduces sound across different frequencies, ensuring that it meets the desired audio specifications. Engineers can adjust parameters such as cone material, size, and shape to optimize the frequency response and achieve the desired sound characteristics.Furthermore, ANSYS aids in studying the directivity pattern of the speaker. By analyzing how sound radiates from the speaker at various angles, engineers can fine-tune the design to ensure even sound distribution and coverage in the intended application, whether it's for a home audio system or a concert venue.ANSYS also assists in assessing the efficiency of the speaker by analyzing the conversion of electrical energy into acoustic energy. This information is vital for designing energy-efficient and high-performance speakers.In addition, ANSYS enables engineers to investigate the impact of structural elements and damping materials on reducing unwanted vibrations and distortion, leading to improved sound quality.In summary, the acoustic analysis of a speaker using ANSYS is a comprehensive process that involves modeling, simulation, and optimization of various speaker components and characteristics. This analysis empowers engineers to design speakers that deliver exceptional audio quality, reliability, and efficiency for a wide range of application.

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**LIST OF ABBREVIATIONS**

|  |  |  |
| --- | --- | --- |
| **ACRONYM** |  | **ABBREVIATION** |
| FEA | - | Finite Element Analysis |
| CFD | - | Computational Fluid Dynamics |

CHAPTER 1  
INTRODUCTION

The world of acoustics is deeply intertwined with our everyday experiences, profoundly influencing the way we communicate, entertain, and interact with technology. Within this field, the design and optimization of audio systems, specifically speakers, hold a pivotal role. Acoustic analysis has grown in importance as it ensures the production of high-quality sound and meets the ever-increasing demands for enhanced audio performance across various domains, including consumer electronics, automotive technology, and industrial applications.A crucial tool that has revolutionized the study of acoustics and speaker performance is ANSYS. ANSYS, a leading simulation software suite, equips engineers and researchers with the means to comprehensively analyze intricate physical phenomena. In the context of speaker design, ANSYS has emerged as an invaluable tool, facilitating an in-depth examination of the science of acoustics and enabling manufacturers to engineer more efficient and high-fidelity speaker systems.The quality of sound produced by a speaker system hinges on multiple factors, such as the physical design, choice of materials, and the intricacies of sound wave propagation. Acoustic analysis stands as an indispensable practice, enabling engineers to meticulously assess and optimize these factors. It is this rigorous analysis that ensures that speakers meet their intended performance criteria. Whether the goal is to craft a home audio speaker, a professional sound reinforcement system, or a specialized automotive audio setup, acoustic analysis serves as the cornerstone of achieving exceptional sound quality, clarity, and efficiency.

the fusion of acoustic analysis with ANSYS software stands as a transformative approach that harnesses advanced simulation technology to elevate the design and performance of speaker systems. This synergy of engineering expertise and state-of-the-art software serves as a catalyst for the development of speakers capable of delivering sound quality that surpasses conventional standards. The reach of this impact extends across numerous applications, spanning from personal entertainment systems to professional audio equipment, underscoring the profound influence of acoustics in our daily lives. It boasts a rich repertoire of tools for finite element analysis (FEA), computational fluid dynamics (CFD), and multiphysics simulations. For the specific domain of acoustics, ANSYS offers the capability to simulate sound waves, their interaction with speaker components, and their behavior in different environments. This software empowers engineers to undertake intricate simulations, visualize results, and methodically refine designs. The culmination of these capabilities leads to superior speaker performance, providing engineers with a toolset to navigate the complexities of sound dynamics with precision. the fusion of acoustic analysis with ANSYS software stands as a transformative approach that harnesses advanced simulation technology to elevate the design and performance of speaker systems. This synergy of engineering expertise and state-of-the-art software serves as a catalyst for the development of speakers capable of delivering sound quality that surpasses conventional standards. The reach of this impact extends across numerous applications, spanning from personal entertainment systems to professional audio equipment, underscoring the profound influence of acoustics in our daily lives.

**1.1 OBJECTIVES**

The project had as its objectives the following aspects

* To delimitate the advantages, disadvantages and applications for the project.
* To present possible solutions, if any to solve or mitigate the impact of the disadvantages.

**1.2 SPEECH RECOGNITION SYSTEM**

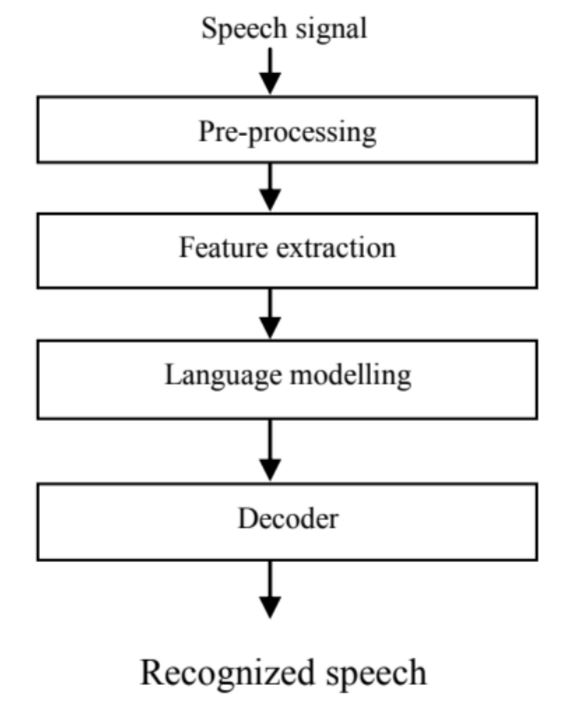
The speech sound is captured using microphone to convert it in to electrical signal. The purpose of sound card inside the computer is to change analog signal into digital signal. Sound card has capabilities to store and play this speech signal. There are following building blocks for general speech recognition system.

* Signal preprocessing
* Feature extraction
* Language model
* Decoder
* Speech Recognition

**1.2.1 SIGNAL PREPROCESSING**

Speech signal captured by microphone, telephone etc. are analog in nature so it is required to be digitized as per Nyquist theorem. This theorem states that a signal is to be sampled more than twice the rate of highest frequency present in

it.Generally sampling frequencies for speech signal are 8 KHz and 20 KHz. For telephonic speech signal it is recommended to have 8 KHz sampling rate while 16 KHz is generally used for normal microphones

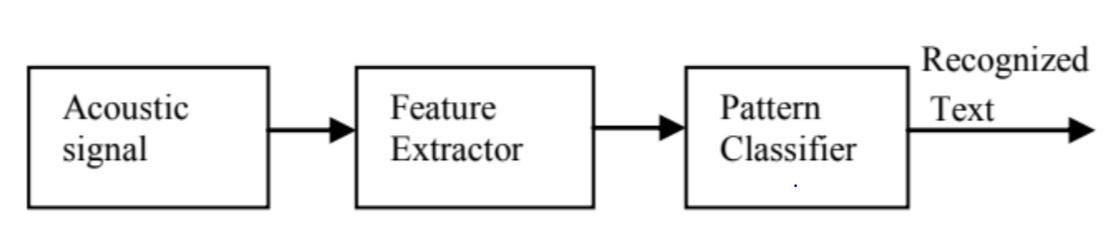


**Fig 1.1**

**1.2.2 FEATURE EXTRACTION**

Feature extraction is used to find a set of properties that are stable and acoustically correlated to each other. So it is a type of parameterization of speech signal.Such parameters can form the observation vectors. The goal of feature extractor is to identify relevant information for purpose of accurate classification.

Fig 1.2 shows the role of feature extraction in speech recognition.



**Fig 1.2**

**1.2.3 LANGUAGE MODELING**

Language modeling is used to find the correct word sequence by predicting nth words using (n-1) preceding words.Language modeling is of various types.

* Uniform model: where occurrence of each word is equally probable.
* Stochastic model: probability of present word depends on probability of word preceding it.
* Finite state languages: This language modeling use finite state network to define allowed word sequence.
* Context free grammar (CFG): It is used to encode allowed sequence of words in speech recognition system.CFG follows a mechanism for defining languages (sets of acceptable sentences).

CFG is used to model a language in speech recognition by expanding its special non-terminal symbols after application of production rules.

**1.2.4 DECODER**

This stage is involved to find most likely word sequence for the given observation sequence. Generally dynamic programming algorithms are used to solve this problem. The purpose of these algorithms is to search single path through the network to have best match for the given sequence, Viterbi algorithm is mostly used for this purpose. In case of large vocabulary, a beam search method is useful for Viterbi iteration .

**1.2.5 SPEECH RECOGNITION**

Speech recognition is completed in two phase: Training and testing phase. Training phase is just similar to identification of objects. It may be repeated many times for better recognition which improves the performance while testing. Testing phase includes the comparison between reference pattern scored while training, and spoken words at the time of testing. The extent of closeness in these two phases counts for improving the performance of the system. But variability encountered during recognition effects a lot for constant recognition rate.

**CHAPTER 2**

**LITERATURE SURVEY**

**2.1 DURING THE YEAR OF 1970-1980:**

Just as isolated word recognition was a key focus of research in the 1970s, the problems of connected word recognition was a focus of research in the 1980s. Here the goal was to create a robust system capable of recognizing a fluently spoken string of words(eg., digits) based on matching a concatenated pattern of individual words. Moshey J. Lasry has developed a featurebased speech recognition system in the beginning of 1980. Wherein his studies speech spectrograms of letters and digits[97].A wide variety of the algorithm based on matching a concatenated pattern of individual words were formulated and implemented, including the two level dynamic programming approach of Sakoe at Nippon Electric Corporation (NEC),the one pass method of Bridle and Brown at Joint Speech Research Unit(JSRU) in UK, the level building approach of Myers and Rabiner at Bell Labs , and the frame synchronous level building approach of Lee and Rabiner at Bell Labs. Each of these optimal matching procedures had its own implementation advantages, which were exploited for a wide range of tasks. Speech research in the 1980s was characterized by a shift in technology from template based approaches to statistical modeling methods especially the hidden Markov model approach. Although the methodology of hidden Markov modeling (HMM) was well known and understood in a few laboratories(Primarily IBM, Institute for Defense Analyses (IDA), and Dargon systems), it was not until widespread publication of the methods and theory of HMMs, in the mid 1980, that the technique became widely applied in virtually, every speech recognition research laboratory in the world. Today, most practical speech recognition systems are based on the statistical frame work developed in the 1980s and their results, with significant additional improvements have been made in the 1990s.

**2.1.1 HIDDEN MARKOV MODEL(HMM):**

HMM is one of the key technologies developed in the 1980s, is the hidden Markov model(HMM) approach . It is a doubly stochastic process which as an underlying stochastic process that is not observable (hence the term hidden), but can be observed through another stochastic process that produces a sequence of observations. Although the HMM was well known and understood in a few laboratories (primarily IBM, Institute for Defense Analysis (IDA) and Dragon Systems), it was not until widespread publication of the methods and theory of HMMs in the mid-1980s that the technique became widely applied in virtually every speech recognition research laboratory in the world. In the early 1970s, Lenny Baum of Princeton University invented a mathematical approach to recognize speech called Hidden Markov Modeling (HMM). The HMM pattern-matching strategy was eventually adopted by each of the major companies pursuing the commercialization of speech recognition technology (SRT).The U.S. Department of Defense sponsored many practical research projects during the 70s that involved several contractors, including IBM, Dragon, AT&T, Philips and others. Progress was slow in those early years.

**2.1.2 NEURAL NET:**

Another new technology that was reintroduced in the late 1980s was the idea of applying neural networks to problems in speech recognition. Neural networks were first introduced in the 1950s, but they did not prove useful initially because they had many practical problems. In the 1980s however, a deeper understanding of the strengths and limitations of the technology was achieved, as well as, understanding of the technology to classical signal classification methods. Several new ways of implementing systems were also proposed.

**2.1.3 DARPA PROGRAM:**

Finally, the 1980s was a decade in which a major impetus was given to large vocabulary, continuous speech recognition systems by the Defense Advanced Research Projects Agency (DARPA) community, which sponsored a large research program aimed at achieving high word accuracy for a 1000 word continuous speech recognition, database management task. Major research contributions resulted from efforts at CMU(notably the well known SPHINX system), BBN with the BYBLOS system, Lincoln Labs, and AT&T Bell Labs. The SPHINX system successfully integrated the statistical method of HMM with the network search strength of the earlier Harpy system. Hence, it was able to train and embed context dependent phone models in a sophisticated lexical decoding network. The DARPA program has continued into the 1990s, with emphasis shifting to natural language front ends to the recognizer and the task shifting to retrieval of air travel information. At the same time, speech recognition technology has been increasingly used within telephone networks to automate as well as enhance operator services.

**2.2 DURING THE YEAR OF 2000-2009**

* + 1. **General:**

Around 2000, a variational Bayesian (VB) estimation and clustering techniques were developed. Unlike Maximum Likelihood, this VB approach is based on a posterior distribution of parameters. Giuseppe Richardi have developed the technique to solve the problem of adaptive learning, in automatic speech recognition and also proposed active learning algorithm for ASR. In 2005, some improvements have been worked out on Large Vocabulary Continuous Speech Recognition system on performance improvement. In 2007, the difference in acoustic features between spontaneous and read speech using a large scale speech data base i.e, CSJ have been analyzed. Sadaoki Furui investigated SR methods that can adapt to speech variation using a large number of models trained based on clustering techniques. In 2008, the authors have explored the application of Conditional Random Field(CRF) to combine local posterior estimates provided by multilayer perceptions corresponding to the frame level prediction of phone and phonological attributed classes. De-wachter et.al., attempted to over-come the time dependencies, problems in speech recognition by using straight forward template matching method. Xinwei Li et.al., proposed a new optimization method i.e., semi definite programming(SDP) to solve the large margin estimation(LME) problem of continuous density HMM(CDHMM) in speech recognition. Discriminate training of acoustic models for speech recognition was proposed under Maximum mutual information(MMI). Around 2007 Rajesh M.Hegde et.al, , proposed an alternative method for processing the Fourier transform phase for extraction speech features, which process the group delay feature(GDF) that can be directly computed for the speech signal.

**2.2.2 DARPA program:**

The Effective Affordable Reusable Speech-to-Text (EARS) program was conducted to develop speech-to-text (automatic transcription) technology with the aim of achieving substantially richer and much more accurate output than before. The tasks include detection of sentence boundaries, fillers and disfluencies. The program was focusing on natural, unconstrained human speech from broadcasts and foreign conversational speech in multiple languages. The goal was to make it possible for machines to do a much better job of detecting, extracting, summarizing and translating important information, thus enabling humans to understand what was said by reading transcriptions instead of listening to audio signals .

**2.2.3 Spontaneous speech recognition:**

Although read speech and similar types of speech, e.g. news broadcasts reading a text, can be recognized with accuracy higher than 95% using state-of-the-art of speech recognition technology, and recognition accuracy drastically decreases for spontaneous speech. Broadening the application of speech recognition depends crucially on raising recognition performance for spontaneous speech. In order to increase recognition performance for spontaneous speech, several projects have been conducted. In Japan, a 5-year national project Spontaneous Speech: Corpus and Processing Technology was conducted . A world-largest spontaneous speech corpus, Corpus of Spontaneous Japanese (CSJ) consisting of approximately 7 millions of words, corresponding to 700 hours of speech, was built, and various new techniques were investigated. These new techniques include flexible acoustic modeling, sentence boundary detection, pronunciation modeling, acoustic as well as language model adaptation, and automatic speech summarization . The three analyses on the effects of spontaneous speech on continuous speech recognition performance are described in viz., (1) spontaneous speech effects significantly degrade recognition performance, (2) fluent spontaneous speech yields word accuracies equivalent to read speech, and (3) using spontaneous speech training data. These can significantly improve the performance for recognizing spontaneous speech. It is concluded that word accuracy can be improved by explicitly modeling spontaneous effects in the recognizer, and by using as much spontaneous speech training data as possible. Inclusion of read speech training data, even within the task domain, does not significantly improve performance.

**2.2.4 Robust Speech recogonition**

To further increase the robustness of speech recognition systems, especially for spontaneous speech, utterance verification and confidence measures, are being intensively investigated . In order to have intelligent or human-like interactions in dialogue applications, it is important to attach to each recognized event a number that indicates how confidently the ASR system can accept the recognized events. The confidence measure serves as a reference guide for a dialogue system to provide an appropriate response to its users. To detect semantically, significant parts and reject irrelevant portions in spontaneous utterances, a detection based approach has recently been investigated . The combined recognition and verification strategy work well especially for ill-formed utterances. In order to build acoustic models more sophisticated than conventional HMMs, the dynamic Bayesian network has recently been investigated . Around 2000, a QBPC, systems were developed to find the unknown and mismatch between training and testing conditions. A DCT fast subspace techniques has been proposed to approximate the KLT for autoregressive progress. A novel implementation of a mini-max decision rule for continuous density HMM-based Robust speech recognition is developed by combining the idea of mini-max decision rule with a normal viterbi search. Speech signal modeling techniques well suited to high performance and robust isolated word recognition have been contributed. The first robust Large vocabulary continuous speech recognition that uses syllable-level acoustic unit of LVCSR on telephone bandwidth speech is described in . In 2003, a novel regression based Bayesian predictive classification(LRBPC) was developed for speech Hidden markov model. Walfgang Rchichal has described the methods of improving the robustness and accurancy of the acoustic modeling using decision tree based state tying. Giuluva Garau et.al., investigated on Large vocabulary continuous speech recognition. Xiong Xiao have shown a novel technique that normalizes the modulation spectra of speech signal. Kernel based nonlinear predictive coding procedure, that yields speech features which are robust to nonstationary noise contaminated speech signal.Features maximally in sensitive to additive noise are obtained by growth transformation of regression functions that span a reproducing a kernel Hilbert space (RKHS). Soundararajan proposed a supervised approach using regression trees to learn non linear transformation of the uncertainty from the linear spectral domain to the cepstral domain. Experiments are conducted on Aurora-4 Database.

**CHAPTER 3**

**3.1 EXISTING SYSTEM:**

In the existing system, the mobile device is initially controlled through wired networks like switches, buttons etc. later it is controlled through wireless networks like remote etc. But in wireless network, there is no proper acknowledgement for the transmitted signal. As a result there is no reliability and robustness. We can use our computer only if keyboard and mouse works properly .In wireless network when controlling a mobile device there is no proper acknowledgement for the transmitted signal. As a result there is no reliability and robustness. There is no other alternative if any of the mouse or keyboard gets faulty.

**3.1.1 LIMITATIONS OF EXISTING SYSTEM:**

* No proper acknowledgement for the transmitted signal.
* No reliability and robustness.
* Consumes a lot of time.
* Requires direct role of the person to be involved in the work.

**CHAPTER 4**

**4.1 PROPOSED SYSTEM:**

In the proposed system about 95% of the existing system is being computerized. In the proposed system, the mobile device is controlled through speech signal. The speech signal is given through micro phone. Using this we can achieve the maximum amount of accuracy. Hence more acknowledgements are provided which results in reliability and robustness. Modern speech recognition systems can achieve high recognition rates, but their accuracy often decreases dramatically in noisy and crowded environments. This is usually dealt with by either requiring an almost noise-free environment or by placing the microphone very close to the speaker‘s mouth.

**4.1.1 ADVANTAGES OF PROPOSED SYSTEM:**

* Ability to carry maximum number of operations on an operating system.
* Security of data.
* Ensures the maximum accuracy.
* Greater efficiency.
* User friendly and interactive.
* Minimum time required.
* Flexibility of the system.
* Highly advantageous for people with both physical and mental disabilities.
* Useful during natural calamities or typical situations.
* Can also be used as voice controlled operating system.
* Reduction in the cost of software as much as possible

**CHAPTER 5**

**SYSTEM STUDY**

**5.1 FEASIBILITY STUDY:**

A feasibility study is an evaluation and analysis of the potential ofthe proposed project which is based on extensive investigation and research to givefull comfort to the decisions makers. Feasibility studies aim to objectively andrationally uncover the strengths and weaknesses of an existing business or proposedventure, opportunities and threats as presented by the environment, the resourcesrequired to carry through, and ultimately the prospects for success.

* Technical Feasibility
* Operational Feasibility
* Economical Feasibility

**5.1.1 TECHNICAL FEASIBILITY:**

The assessment is based on an outline design of system requirements, to determine whether the company has the technical expertise to handle completion of the project. When writing a feasibility report, the following should be taken to consideration:

* A brief description of the business to assess more possible factor/s which could affect the study
* The part of the business being examined
* The human and economic factor
* The possible solutions to the problems At this level, the concern is whether the proposal is both technically and feasible.

**5.1.2 OPERATIONAL FEASIBILITY:**

Operational feasibility is a measure of how well a proposed system solves the problems, and takes advantage of the opportunities identified during scope definition and how it satisfies the requirements identified in the requirements analysis phase of system development.

* Is there sufficient support for the management from the users?
* Will the system be used and work properly if it is being developed and implemented?
* Will there be any resistance from the user that will undermine the possible application benefits?

Under this category of service we conduct a study to analysis and determine whether your business need can be fulfilled by using a proposed solution. The result of our operational feasibility Study will clearly outline that the solution proposed for your business is operationally workable and conveniently solves your problems under consideration after the proposal is implemented. This is sometimes referred to as Feasibility Evaluations‘. We would precisely describe how the system will interact with the systems and persons around. Our feasibility report would provide results of interest to all stake holders

**5.1.3 ECONOMIC FEASIBILITY:**

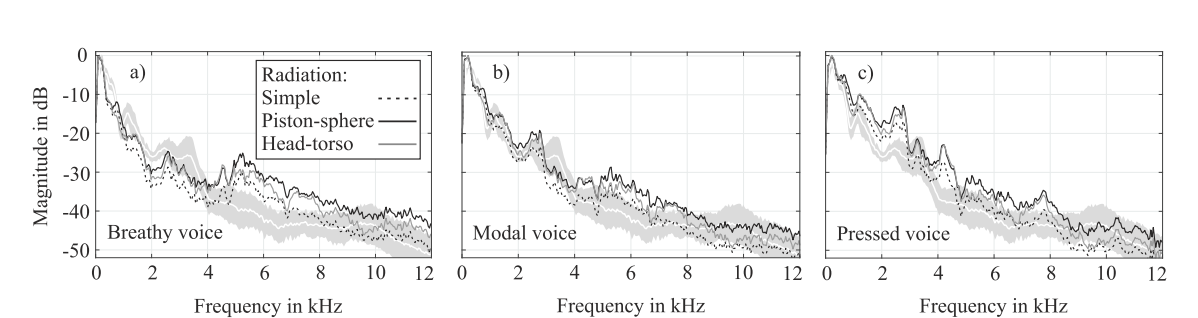
The purpose of an economic feasibility study (EFS) is to demonstrate the net benefit of a proposed project for accepting or disbursing electronic funds/benefits, taking into consideration the benefits and costs to the agency, other state agencies, and the general public as a whole. The EFS is composed of two required forms:

* Business Case
* Cost Benefit Analysis

**CHAPTER 6**

**RESULT AND DISCUSSION**

Figure 6.1 shows the LTAS for each of the 9 synthesis conditions (3 phonation types × 3 radiation characteristics) as the 586 black, gray and dashed lines. The white lines surrounded by 587 the gray areas indicate the average LTAS of the real speakers 588 and the interval of ±1 standard deviation. For each phonation 589 type, the differences between the 3 synthetic LTAS reflect 590 exactly the spectral differences between the 3 radiation characteristics, i.e., the high frequencies have the lowest amplitude for the simple radiation model, the highest amplitude for 593 the piston-in-sphere model, and an intermediate amplitude for 594 the head-torso model. For the synthetic modal voice, the three 595 LTAS curves are most similar to the LTAS of natural speech, 596 and deviate more for the synthesis with breathy and pressed 597 voice. This is plausible because the speakers were instructed 598 to speak with their ‘‘normal’’ voice, which should correspond 599 to a modal voice when we average across multiple speakers. 600 With regard to the spectra of the synthetic breathy voice it is 601 noteworthy that the magnitude above 4 kHz is relatively high 602 despite the steep spectral slope at lower frequencies. This 603 boost at high frequencies can be attributed to the aspiration 604 noise, which becomes the dominant excitation (compared to 605 the periodic voice source) in breathy phonation at higher 606 frequencies.



**Fig 6.1**

**CHAPTER 7**

**CONCLUSION AND FUTURE WORK**

Speech is basic mode of communication between human beings, so a feasible interface is required to connect human with machines. Although this field has gained a wide approval to automate the services and applications but there are several parameters which affect the accuracy and efficiency of speech recognition system. The most of speech variability involves speech rate, environmental conditions, channel and context of utterance. Robustness of speech system depends on some stable parameters/ features of speech signal. To enhance the power of speech recognition system, it is required to design speech recognizers in local languages. Multilingual is new evolving field in area of speech recognition. There is a lot of development and research in the field of foreign languages but to enhance its power and utility for native people, it’s essential to use this technology in native languages.ANSYS is a powerful tool for accurate acoustic analysis of speakers, aiding in design optimization and issue resolution. It saves time and costs by reducing the need for physical prototypes and validates performance against specifications. Advanced material modeling and non-linear behavior analysis for improved accuracy. Integration of multi-physics simulations to consider structural, thermal, and electromagnetic factors. Sensitivity analysis and optimization algorithms for automated design improvement. Real-world testing validation to confirm simulation accuracy. Environmental considerations, such as humidity and temperature effects.AI and machine learning for automation and predictive modeling. User feedback analysis to address real-world issues. Miniaturization and portability to meet market demands. Sustainability considerations for eco-friendly designs. In summary, ANSYS is a valuable tool for speaker analysis, and future work should focus on enhancing accuracy, multi-physics integration, automation, sustainability, and meeting user expectations. It excels in providing precise simulations that faithfully represent the acoustic behavior of the speaker. This accuracy lends itself to design optimization, enabling fine-tuning of critical parameters like diaphragm shape, materials, and enclosure geometry, resulting in an improved performance and enhanced sound quality. Furthermore, ANSYS plays a crucial role in troubleshooting by identifying and resolving acoustic issues such as unwanted resonances, distortion, and suboptimal sound projection. This translates into substantial cost and time savings, as it reduces the necessity for physical prototypes and facilitates rapid design iterations. Ultimately, ANSYS aids in performance validation, ensuring that the speaker meets its required specifications. Looking ahead, several areas warrant attention in the future development of acoustic analysis using ANSYS.

Advanced material modeling should be explored to enhance the accuracy of simulations, including the consideration of non-linear material behavior and materials with varying properties. The integration of multi-physics simulations, encompassing structural, thermal, and electromagnetic aspects, would provide a comprehensive view of the speaker's performance, addressing issues such as heat dissipation, mechanical integrity, and electromagnetic interference. Sensitivity analysis and the application of optimization algorithms would automate the process of design improvement, saving both time and resources. Real-world testing validation is essential to confirm the accuracy of ANSYS models and to ensure that the speaker performs as expected under various conditions. Environmental factors, such as humidity and temperature effects, should be incorporated into simulations to design speakers that perform reliably in diverse environments. The integration of AI and machine learning for automation and predictive modeling could further streamline the design process. Analyzing user feedback will provide insights into practical issues, enhancing user satisfaction.

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