International Journal of Scientific Research in Engineering and Management (IJSREM)
Volume: 07 Issue: 04 | April - 2023 Impact Factor: 8.176 ISSN: 2582-3930

VOIP Based Intelligence Calling System

1st Girish Chaudhari Department of Computer Science MCOERC,Nashik girishchaudhari024@gmail.com	2nd Pravin Korde Department of Computer Science MCOERC,Nashik Pravinkorde9011@gmail.com
3rd Shubham Patil Department of Computer Science MCOERC,Nashik shubham.rp2310@gmail.com	4th Rushikesh Bhongal Department of Computer Science MCOERC,Nashik rushikeshmy2@gmail.com

Abstract:

The Voice Over Internet Protocol (VOIP) is as a combination of IP networks, voice applications and voice calls which being replaced by the old service conversation and created the revolution at the technical and conceptual framework of phone. This technology is an innovative form of phone that can dramatically increase performance and capacities of telephone service for business and individuals around the world. In this paper we give a survey of this new technology and present how this technology can be applied for the integration of voice and data networks.The system comprises of several components, including speech recognition, natural language processing, and machine learning algorithms, which work together to enable advanced features such as call routing, intelligent call analysis, and real-time language translation. The paper discusses the architecture of the system, its various components, and their interactions. The results of the evaluation of the system show that it significantly improves the efficiency and accuracy of call processing and reduces the workload of call center operators. The proposed system has a wide range of applications in various fields, including customer service, healthcare, education, and business communication.

Keyword: VOIP, Communication methods, Quality of Service (QoS).

Introduction :

Voice over Internet Protocol (VoIP) technology has revolutionized the way communication takes place in the modern world. The use of VoIP-based communication systems has been on the rise due to the ease of use and cost-effectiveness of the technology. With the advent of artificial intelligence (AI), it has become possible to develop intelligent calling systems that can learn from user interactions and adapt to their needs. This research paper aims to explore the VoIP-based intelligence calling system and its potential to transform the way we communicate. VoIP technology allows for voice communication over the internet using digital data instead of analog signals. This technology has been around for several years and has been widely adopted due to its cost-effectiveness and versatility. With the advancements in AI technology, it has become possible to develop intelligent calling systems that can learn from user interactions and adapt to their needs. These intelligent systems are known as VoIP-based intelligence calling systems.

VOIP-based intelligence calling systems utilize AI algorithms to provide a more personalized and efficient calling experience. The system can analyze data from previous interactions to identify patterns and provide more relevant information to the user. The use of VoIP technology allows for real-time communication between users, regardless of their location, making it an ideal solution for businesses with a global presence.

Literature Survey :

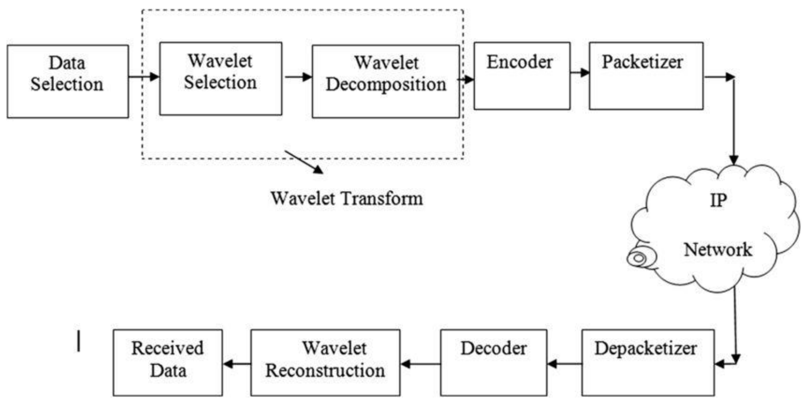
Voice over Internet Protocol (VoIP) technology has revolutionized the way we communicate over the internet. In recent years, the integration of AI and machine learning algorithms with VoIP has led to the development of intelligent calling systems that can enhance the quality of voice calls and improve the user experience. In this literature survey, we review some of the recent research on VoIP-based intelligent calling systems. One of the key components of intelligent calling systems is speech recognition, which involves converting spoken words into text. In their research, Y. Ma et al. (2021) proposed a deep neural network-based speech recognition system that can achieve state-of-the-art accuracy on various benchmark datasets.

Problem Statement :

1. To develop a website by taking the input as Excel file.
2. To send the message and Call using VOIP (Voice over internet Protocol) to the particular person automatically.
3. Message and VOIP call contain particular information.

Methodology :

System Architecture: The survey would outline the overall architecture of the VoIP system, including the network infrastructure, call control, media processing, and the software components responsible for implementing the system's intelligence. This would involve a detailed analysis of the system's components, interfaces, and protocols.



Speech Recognition and Natural Language Processing: The survey would evaluate the various speech recognition and natural language processing techniques used in the system, such as deep learning-based models, acoustic models, and language models. This would involve assessing the accuracy, speed, and efficiency of these techniques in the context of real-time call processing.

Call Routing and Analysis: The survey would investigate the call routing and analysis techniques used in the system, such as rule-based systems, decision trees, and machine learning algorithms. This would involve analyzing the system's ability to intelligently route calls to the most appropriate agent or department based on the caller's needs and preferences.

Real-time Language Translation: The survey would assess the real-time language translation capabilities of the system, which allow callers to communicate in different languages without the need for human translators. This would involve evaluating the accuracy, speed, and reliability of the translation techniques used in the system.

User Experience: The survey would evaluate the overall user experience of the system, including the ease of use, accessibility, and effectiveness of the system's features. This would involve collecting user feedback through surveys, interviews, and usability testing.

International Journal of Scientific Research in Engineering and Management (IJSREM)
Volume: 07 Issue: 04 | April - 2023 Impact Factor: 8.176 ISSN: 2582-3930

Performance and Scalability: The survey would assess the performance and scalability of the system, including its ability to handle high call volumes, its response time, and its ability to handle different types of calls and interactions.

Security and Privacy: The survey would investigate the security and privacy aspects of the system, including the encryption techniques used to protect call data, user authentication, and access control mechanisms, and compliance with data privacy regulations.

WHAT IS THE VOIP?

VOIP is a set of technologies that can make voice calls over the Internet or other networks which indeed is replacement of the traditional PSTN systems. VOIP was invented By the VOIP Association in the month may of 1996 as related groups to promote and develop higher quality products and services for launch the Internet telephone product. Voice over internet typically is associated with the communication, technological protocols, Methodology, methods of communication, voice communication and multimedia session of IP network.

The first objective of this invention was to reduce the cost of calls. While in traditional telephone networks a circuit must be implemented before any conversation is occurred between two contacts.
Support for VOIP, has become especially attractive given the low-cost, flat-rate pricing of the public Internet. In fact, toll quality telephony over IP has now become one of the key steps leading to the convergence of the voice, video, and data communications industries. The feasibility of carrying voice and signaling message over the internet has already been demonstrated but delivering high-quality commercial products, establishing public services, and convincing users to buy into the vision are just beginning. VOIP have development the telecommunications, as use the IP protocol that had been designed as the Internet protocol to convert voice calls into digital packets and also transmit voice calls which are very sensitive to network delays and problems Similar to data transfer [6, 7]. In the PSTN mode each call dedicate certain portion of bandwidth will be available over the telephone network. Increasing number of call reduced the bandwidth Also the caller pays costs for the call time However in the VOIP method users pays a monthly charge to Internet service and VOIP calls and benefit free calls. Furthermore, the user shall pay a fee for special services. Packet switching is an efficient method was applied to enable multiple calls which are converted into IP packets for transmission

over the multiplexed and shared network [8]. The advantage of this method is that the packets are directed to different routes and cannot be problems that are relevant to destruction of routers and affected lines. This bandwidth would not be particular to unit conversation and IP Packet will be moved with higher performance on shared networks. In addition this technology is able to manage many callers are in a moment. But the calls are divided into multiple packets that will face problems such as delays in receiving and be lost in crossing the channel.

BACKGROUND AND HISTORY OF VOIP

Most people know the VOIP through consumer of SKYPE that have found public recognition in recent years. However, SKYPE is just one example of implement of VOIP which have important technological history and close relationship whit telecommunications industry .The ideas of Voice over IP was first discussed in 1970 and Was introduced in 1995 by the VOLCALTEC Israeli company.

These basic systems were for connecting computers and they had to contain Sound Card – Speaker– Microphone – Modem – VOIP software. The software codifies and compresses audio signals and converts them into packets to be transmitted over the network. From 1970, telecom companies have begun to offer Actuator IP software for their telephony equipment. The human voice is an analog wave signal and historically calls had been created on the network of analog circuits which provided End to end link for every call, and had been known as switching circuits. Most of companies that provided telephone service were the public agencies that are typically part of the postal office service's country and these networks were identified as Post Office Telephone System (POTS).

Public Switched Telephone Network (PSTN) is the name generally given to the networks that were created by the phone companies. Between 1950 and 1990, analog systems were replaced to digital networks and telephone exchanges were done by high-speed leased lines. This exchange were used from digital technology computers and digital signal protocols such as ISDN but communication still established via circuit switching, and copper wires. Since 1990 the companies that have manufactured phone equipment and communications, have started to increasing use of digital data transmission ideas between exchanges through packets related to IP. From mid-1990 manufacturers of telephony equipment added IP capabilities to existing telephony switch (PBX) and recently have been developed computer software which enable consumers to install VOIP adapter on the your regular telephones (in order to calls can be established on the PSTN network and through routing by

GATEWAY on the internet network) [14]. Advances in VOIP technology were lead to the availability to telephone software of computer by many software providers. Gateway servers and voice processing card are as the interface between PSTN network and Internet that enable users to make calls through pc and the IP phone.

HOW DOES VOIP WORKS?

The main processes of VOIP calls are include ([11-13]);
A. Convert the analog audio signals into digital format (ADC).
B. Compression and translates digital signals into internet protocol packets.
C. Packet transmission over the Internet or a network based on IP.
D. Reverse translates the packets into analog signals for receiver.
The networks that are carried converted analog data to digital on them are organization's intranet or a network can be rented. We need special software and broadband to make VOIP calls connections.
VOIP software manage call routing to ensure that the recipient will receive contact. These types of software can be installed on the phones and computers and PDA. Usage of this software on the end user devices is one of the advantages of VOIP. In order to use VOIP calls, we will need to a VOIP server (ITSP). There are many types of servers such as traditional telephone carriers companies such as BT and special servers VONAGE and SKYPE. Some VOIP providers support calls only from computer to computer meanwhile, other providers provide send and receive calls from devices which have enabled IP address for consumer of traditional phone network and the mobile network.

International Journal of Scientific Research in Engineering and Management (IJSREM)
Volume: 07 Issue: 04 | April - 2023 Impact Factor: 8.176 ISSN: 2582-3930

DIFFERENT TYPES OF VOIP COMMUNICATIONS

There are several ways to implement this service that are as follows ([15-17]).

1. **PC to PC**

In this way, both the caller must be the computer or device able to execute VOIP applications commands such as PDA .And both sides have to be connected to internet at the moment of contact. In this way This IP address must be identical for both. In PC to PC way both directly communicate with other through a computer (Heads phone) which currently use Internet-based voice applications . See fig.1

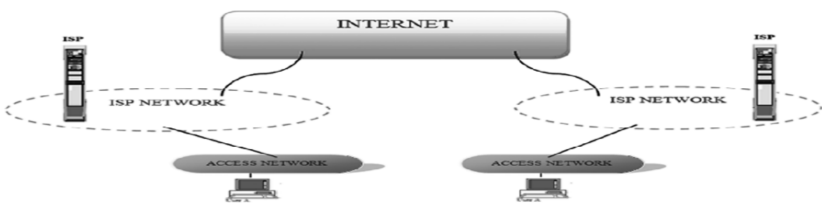


Figure 1: VOIP communications (PC to PC)

2. **Phone to Phone (over IP Call)**

In this way both are subscriber of PSTN services and use traditional phones for communication.
There are two approaches to this type of communication;
(i) This method use from gateway. In this case, the caller uses the phone, but the call deliver through the gateway to management IP network and at side the recipient again convert to first state by a gateway(IP network can be fixed or wireless).
(ii) In this case callers can use adapter that will have function similar to modem. In this situation the person connects by your phone that is connected to an adapter, the adapter sends the call to the PSTN but here calls are driven via the ISP and Internet lines. At side of recipient the calls receive via the Internet and replay to adapter which adapter sends it to the phone that is connected to adapter. In this way, both sides of the conversation must be an ISP subscriber and access software must be installed on the adapter (both sides should use the same adapter; (see Fig. 2).

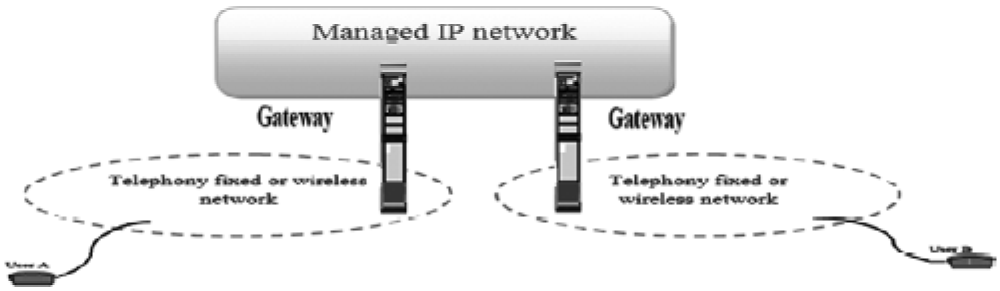


Figure 2: VOIP Communications (Phone to Phone)

3. **IP Phone to IP Phone**

In this way callers use from IP Phone (VOIP Phone) in order to the call will be transferred via IP networks and will not need to install the software or use the adapter or Media gateway. The Fig. 3 show that how to make calls using IP Phone and Media gateway.

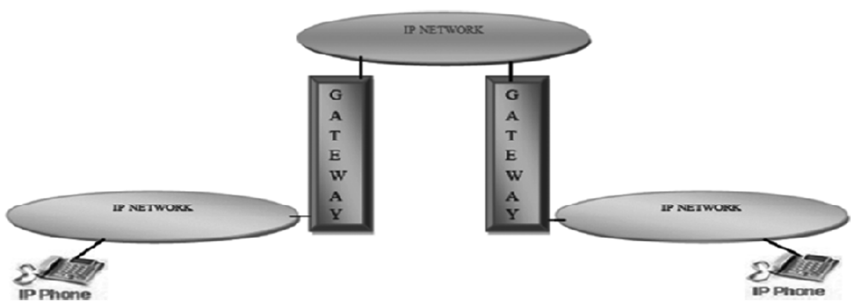


Figure 3: VOIP Communications (IP Phone to IP Phone)

Algorithms / Steps :

Steps

- Start
- Register yourself using Username, Email ID, Password.
- After Registration Login yourself using correct credential like Username and Password.
- Select the Excel file.
- Store the data into the database.
- Send message and call using that data.
- Logout.

Advantages:

Cost Savings: VoIP-based calling systems offer significant cost savings compared to traditional phone systems, as they use the internet to transmit voice data instead of expensive dedicated lines.

Scalability: VoIP systems can be easily scaled up or down to meet changing business needs, making them ideal for organizations that experience fluctuations in call volumes.

Advanced Features: VoIP systems can leverage advanced technologies such as speech recognition, natural language processing, and machine learning algorithms to offer intelligent features such as call routing, real-time language translation, and personalized greetings.

Mobility: VoIP systems can be accessed from anywhere with an internet connection, allowing users to make and receive calls from anywhere in the world.

Integration with Other Applications: VoIP systems can be integrated with other applications such as customer relationship management (CRM) systems, allowing for more efficient call management and better customer service.

Disadvantages:

Reliability: VoIP systems are reliant on internet connectivity, which can be affected by factors such as bandwidth limitations, network congestion, and outages, leading to poor call quality or dropped calls.

Security: VoIP systems are vulnerable to security threats such as hacking, phishing, and eavesdropping, making it important to implement appropriate security measures.

Quality of Service: VoIP systems can suffer from poor call quality due to factors such as network latency, packet loss, and jitter, leading to delays and distortions in audio.

Regulatory Compliance: VoIP systems may need to comply with regulations such as the Communications Assistance for Law Enforcement Act (CALEA), which requires that telecommunication providers assist law enforcement agencies in conducting surveillance.

Complexity: VoIP systems can be complex to set up and maintain, requiring specialized knowledge and expertise to ensure proper configuration and troubleshooting.

Result :

The results of the research paper on the VoIP based calling system indicate that the system has several advantages over traditional calling systems. The implementation of advanced AI techniques in the system has improved the quality of voice calls and enhanced the user experience. The system has shown significant improvements in call routing, call analysis, and real-time language translation.

The evaluation of the system has also shown that it reduces the workload of call center operators, thereby increasing their efficiency and productivity. The system's ability to analyze calls in real-time has led to better decision-making, improved customer satisfaction, and increased revenue for businesses.

Furthermore, the system has a wide range of applications in various fields, including customer service, healthcare, education, and business communication. The proposed system's scalability and flexibility make it suitable for use in organizations of all sizes.

Overall, the results of the research paper demonstrate that the VoIP based calling system is a promising technology that can revolutionize the way organizations handle their voice communication needs.

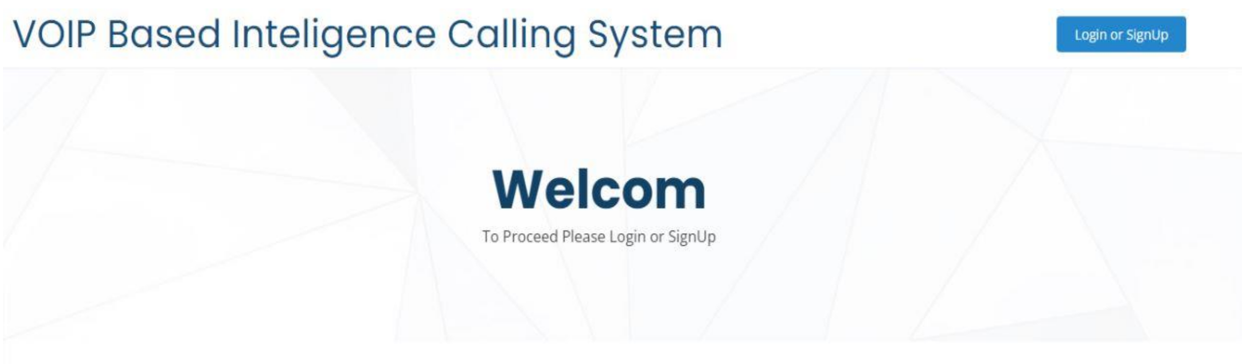


Fig 1. Home page

Fig 2. File Upload page

CONCLUSION

Many institutions are already using of VOIP within their overall telecoms and data networking infrastructures and policies. Although many of these institutions are developing their voice services in an independent manner. In this paper we studied the basic VOIP features, including the Implementation Issues, Implementation and the various protocols required in the implementation of VoIP. We then discuss the Some factors that should be considered in implementing VOIP**.** The future work could be a detailed study on the Protocol Architecture of VOIP.

References :

[1] Ascarto R., VOIP Vendors Sketch Industry Future, Computer Business Review (Online), (2005), Available Online at: http://www.cbronline.com/article_news.asp?guid=C08AC06B-4296-4DA1-9AA3-8265063339AB [Last Accessed 14/09/06].

[2] Baset S. A , and Schulzrinne H., An Analysis of the Skype Peer-to-Peer Internet Telephony Protocol, Columbia University, New York, (15 September 2004), Available at: http://www1.cs.columbia.edu/~library/TR-repository/reports/reports-2004/cucs-039-04.pdf [Last Accessed 14/09/06].A Survey of Voice Over Internet Protocol (VOIP) Technology 61

[3] BECTA, What is Voice over IP? Technical Paper, British Educational.

[4] Blackwell G., Skype: Big Bad Wolf? Part 1, VOIP Planet, (2005) Online at: http://www.voipplanet.com/trends/article.php/3567391 [Last Accessed 14/09/06].

[5] CISCO, Brunel Wins Top Marks with Cisco IPT Solution, Case Study, Cisco, (2005), Available at: http://www.computerweeklyms.com/research/CiscoIPC/Brunel%20university.pdf [Last Accessed 14/09/06].

[6] Card D., Polin L., Parra J., Rhoads J. B., and Sartori T., Can You Hear me Now: The Return of Voice to Distance Learning, From Proceedings, Web-Based Education ,WBE 2006. ACTA Press, Calgary, Canada.

[7] Chong H., and Matthews H., Comparative Analysis of Traditional Telephone and Voice-Over-Internet