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Sim Based Voip System: A Internet To Sim-Based Free Calling Using Design Thinking Framework

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ABSTRACT-

This paper describes the development of an Android application that enables free calls over Wi-Fi networks using Voice over Internet Protocol (VoIP) and Session Initiation Protocol (SIP). The application allows users to make and receive calls using their Wi-Fi connection, eliminating the need for a cellular network and reducing the cost of communication. The paper also discusses the quality of the calls and the factors that can affect the call quality such as network speed, signal strength and the device's microphone and speaker quality. The paper also compares the application to other similar solutions available in the market and highlights its unique features and benefits. The application has been tested on various Android devices and has been found to be reliable and efficient in providing a cost-effective communication solution. The paper concludes by discussing the potential impact of the application on the telecommunications industry and its potential for future development.

Keywords- SIM, VoIP, SIP, Android, Free Calling, Wi-Fi.

I. INTRODUCTION

Voice over Internet Protocol (VoIP) is a technology that allows for the transmission of voice and multimedia communications over the internet. It works by converting analog audio signals into digital data packets that are then sent over the internet to the recipient, who receives the packets and converts them back into an audible signal. This process is known as packetization, and it allows for real-time communication between two or more parties.

VoIP technology has many benefits over traditional telephone systems, including cost savings, flexibility, scalability, and the ability to integrate with other communication tools such as instant messaging and video conferencing. It has become a popular choice for both personal and business use, and is widely used in a variety of settings including homes, offices, and call centers.

The introduction of VoIP technology has revolutionized the telecommunications industry and has provided a cost-effective and efficient communication solution for people all over the world. With the continued advancements in internet speeds and the widespread availability of high-speed internet, it is expected that the use of VoIP technology will continue to grow in the future.

Session Initiation Protocol (SIP) is a signaling protocol used for initiating, maintaining, modifying and terminating real-time sessions that involve video, voice, messaging and other communications between two or more endpoints on IP networks. It is typically used for Internet telephony, but can also be used for other types of communication sessions.

SIP is an application-layer control protocol that allows creating, modifying, and terminating sessions with one or more participants. These sessions include Internet telephone calls, multimedia distribution, and multimedia conferences. SIP is a text-based protocol, incorporating many elements of the Hypertext Transfer Protocol (HTTP) and the Simple Mail Transfer Protocol (SMTP). It is designed to be extensible, and to support IP-based networks.

SIP is a key component in the implementation of VoIP (Voice over IP) systems, as it provides the signaling and control functions necessary to establish and terminate VoIP calls. It enables users to make and receive calls using their IP network, and can be used in conjunction with other protocols such as RTP (Real-time Transport Protocol) to transmit the actual voice or video data.

SIP is widely adopted in the industry, and is an open standard, which means that it can be easily integrated with other systems and devices. It also offers many features such as call forwarding, call transfer, call hold and call conferencing. With the increasing popularity of VoIP and SIP in recent years, it is expected that SIP will continue to play a major role in the future of telecommunication.

II. LITERATURE SURVEY

The Author Abdel Rahman discusses in this paper that the revolution of Voice over Internet Protocol (VoIP) technology has propagated everywhere and replaced the conventional telecommunication technology (e.g. landline). Nevertheless, several enhancements need to be done on VoIP technology to improve its performance. One of the main issues is to improve the VoIP network bandwidth (BW) utilization. VoIP packet payload compression is one of the key approaches to do that and his paper proposes a new method to compress VoIP packet payload. The suggested method works over internet telephony transport protocol (ITTP) and named Delta-ITTP method. The core idea of the Delta-ITTP method is to find and transmit the delta between the successive VoIP packet payloads, which is typically smaller than the original VoIP packet payload. The suggested Delta-ITTP method implements VoIP packet payload compression at the sender side and decompression at the receiver side. During the compression process, the Delta-ITTP method needs to keep some values to restore the original VoIP packet payload at the receiver side. For this, the Delta-ITTP method utilizes some of the IP protocol fields and no additional header is needed. The Delta-ITTP method has been deployed and compared with the traditional ITTP protocol without compression. The result showed that up to 19% BW saving was achieved in the tested cases leading to the desired enhancement in the VoIP network BW utilization.

Dr. Padmashree T, in her paper on Voice over IP mobile telephony using Wi-Fi has explained that today without communication we can’t think of living. We communicate through various mediums like face-to-face conversation, electronic mails, letters etc. One of the most important communication mediums today is a mobile. These days almost everyone is having mobile in their pocket which is just not limited to be used as a communication device but are the multimedia set which is like small pc in hand. As the Android has taken up the considerable amount of mobile market and due to it’s being an open platform it is attracting most of the developers towards it. As it is totally based on java and many developers today are familiar with java it is very easy to develop an android app. Developing android app only requires new ideas and imagination, which can be helpful to someone. Whenever one makes a call to someone it cost money. In our daily life we often face situation where we want to contact a person, which is in same premises. If the premises are large then where all one will find that person and, in that case, we will call him but it will cost. Today almost every premise is Wi- Fi enabled and knowing the fact that most of the people are having Android sets an application can be developed through which we can communicate with other person in same network thus saving our money.

III. EXISTING SYSTEMS

(i) **PC to PC VOIP :**

PC-to-PC VoIP (Voice over Internet Protocol) is a type of internet telephony in which users make and receive telephone calls using their computers. The basic principle behind PC-to-PC VoIP is that it allows users to use the internet as a means of transmitting audio between computers, rather than using traditional telephone networks.

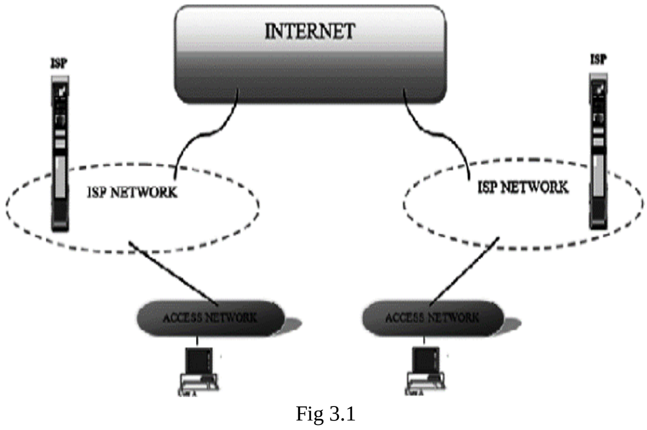

Fig 3.1

To use PC-to-PC VoIP, both parties must have a computer or mobile device with an internet connection and a microphone and speakers or a headset. They must also have a VoIP software client installed on their device. Some popular examples of VoIP software clients include Skype, Zoom, Google Meet, and WhatsApp.

(ii) **Phone to Phone VOIP :**

Phone-to-Phone VoIP (Voice over Internet Protocol) is a type of internet telephony in which users make and receive telephone calls using traditional telephone handsets, but the calls are carried over the internet rather than traditional telephone networks.

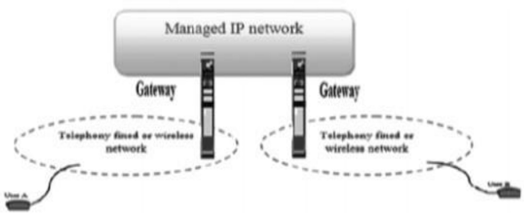

Fig 3.2

To use Phone-to-Phone VoIP, both parties must have a telephone handset with an internet connection. The calls are established over the internet using a VoIP service provider. The service provider connects the call between the two parties using the Voice over IP protocol. Some popular examples of VoIP service providers include Vonage, MagicJack, and Google Voice.

(iii) **IP Phone to IP Phone :**

IP Phone-to-IP Phone VoIP (Voice over Internet Protocol) is a type of internet telephony in which users make and receive telephone calls using specialized IP phones, rather than traditional telephone handsets. The IP phones are connected to the internet and use the Voice over IP (VoIP) protocol to transmit and receive audio.

Fig 3.3

To use IP Phone-to-IP Phone VoIP, both parties must have an IP phone and an internet connection. The IP phone is a telephone-like device that has an IP address and the ability to connect to a network using an Ethernet cable. The IP phone uses the internet to establish the call rather than traditional telephone networks.

(iv) **PC to Phone VOIP :**

PC-to-Phone VoIP (Voice over Internet Protocol) is a type of internet telephony in which users make telephone calls using their computers but the call is connected to a traditional telephone network and is received by a telephone handset.

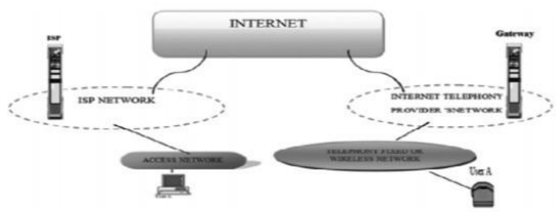

Fig 3.4

To use PC-to-Phone VoIP, the user must have a computer or mobile device with an internet connection and a microphone and speakers or a headset. They must also have a VoIP software client installed on their device. Some popular examples of VoIP software clients include Skype, Zoom, Google Meet, and WhatsApp.

IV. PROPOSED SYSTEM

The proposed system for SIM-based VoIP would consist of the following components:

SIM Card:

A SIM card is a small microchip that is inserted into a mobile device and stores the user's phone number, contact information, and other data. In SIM-based VoIP, the SIM card is used to make and receive calls, just like a traditional phone line.

Fig 4.1

VoIP client:

A software application, such as a smartphone app or a desktop program, that runs on the user's device and is used to initiate and receive calls. The VoIP client communicates with the VoIP server over the internet using the Session Initiation Protocol (SIP).

VoIP server:

A server that is responsible for routing and managing calls between different users. The VoIP server receives call requests from the client and establishes a connection between the caller and the recipient. It also manages the quality of service (QoS) of the call, such as audio and video quality, and ensures that the call is secure and private.

Internet connection:

A stable and fast internet connection, either through a Wi-Fi network or mobile data, is needed to transmit and receive audio and video calls. The internet connection is used to establish a connection between the VoIP client and server and to transmit the call data.

The workflow of internet to SIM calling involves the following steps:

1. The user opens the VoIP application on their device and inputs their SIM card details, such as phone number and carrier information.
2. The user initiates a call by dialing the recipient's phone number on the VoIP application.
3. The VoIP application sends a call request to the VoIP server over the internet, using the SIP protocol.
4. The VoIP server verifies the user's SIM card details and sends a request to the carrier's network to establish a connection to the recipient's phone.
5. The carrier's network receives the request and establishes a connection to the recipient's phone.
6. The recipient's phone rings and they have the option to accept or reject the call.
7. If the call is accepted, the audio and video data is transmitted over the internet from the caller's device to the recipient's device via the VoIP server.
8. The audio and video data is converted to a digital format for transmission and the quality of the call is managed by the VoIP server.
9. The call continues until one of the parties hangs up.
10. The VoIP server sends a message to the carrier's network to terminate the call.
11. The carrier's network terminates the call and sends a message back to the VoIP server to confirm the call has ended.
12. The VoIP server sends a message to the caller's device to confirm the call has ended.
13. The call is ended and the user can initiate another call if desired.

Using internet to SIM calling, users can make calls to any phone number, even if the recipient does not have access to a VoIP application or internet connection. This makes it a convenient and accessible solution for making calls to any phone number.

V. FLOWCHART

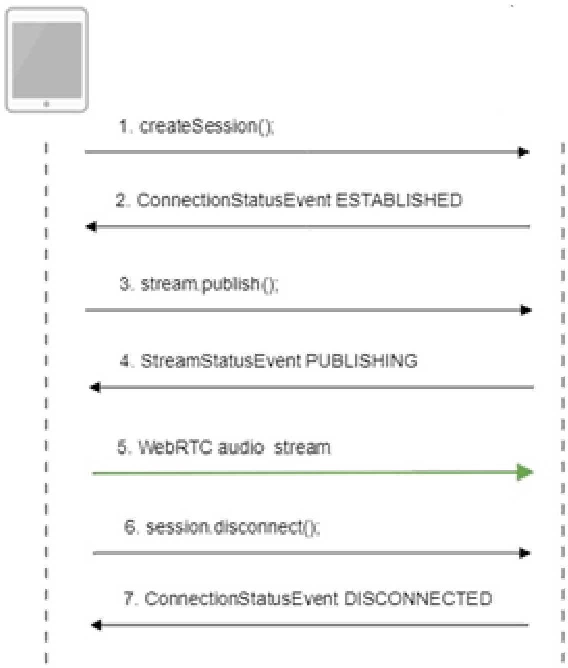

Fig 5.1

VI. SIGNIFICANCE

1. **Cost Savings:**

One of the main objectives of this project is to provide a cost-effective alternative to traditional telephone services, particularly for long-distance and international calls.

2. **Flexibility:**

Another objective is to provide users with more flexibility in terms of their communication options. With this, users can make and receive calls using their existing phone number, regardless of their location and without the need for a traditional phone line.

3. **Portability:**

A key objective of SIM-based VoIP is to enable users to make and receive calls using a variety of devices, such as smartphones, tablets and laptops, as long as they have internet connection and a SIM card. This allows users to take their phone number with them wherever they go, without being tied to a specific device or location.

4. **Quality of service:**

The main objective is to provide high-quality voice call services, with low latency and high-fidelity audio and video, to ensure smooth and clear communication.

5. **Scalability:**

Another objective of SIM-based VoIP is to provide a scalable solution that can support a large number of users and handle a high volume of calls, without compromising on call quality or reliability.

6. **Compatibility:**

The objective is to make sure that this VoIP solution is compatible with a wide range of devices and platforms, including smartphones, tablets, and laptops running different operating systems, and to ensure interoperability with other communication technologies such as SIP and PSTN.

Fig 6.1

VII. IMPLEMENTATION

The implementation of internet to SIM-based calling involves the following steps:

Developing the VoIP application is an important step in the implementation of internet to SIM-based calling. This involves creating a software that can run on various devices such as smartphones and tablets and has the necessary functionality to make and receive calls over the internet using a SIM card.

User interface: The application should have a user-friendly interface that is easy to navigate and understand. This includes options for call initiation, call termination, call holding, call forwarding, and call recording.

Call Quality: The application should be able to provide high-quality voice and video calls. This includes using advanced codecs and echo cancellation techniques to ensure clear and stable calls.

Compatibility: The application should be compatible with different operating systems such as Android, iOS, and Windows. This will allow the application to be used on a wide range of devices.

Integration: The application should be able to integrate with other services such as contact lists and messaging apps.

Scalability: The application should be scalable, so it can handle a large number of users and calls simultaneously.

Setting up a VoIP server is an important step in the implementation of internet to SIM-based calling. A VoIP server is a software or hardware solution that enables the routing and management of VoIP calls.

The following are some key aspects of setting up a VoIP server:

Hardware: A VoIP server requires a powerful computer or server with sufficient processing power and memory to handle a large number of concurrent calls. The server should also have a fast internet connection to ensure high-quality calls.

Software: A VoIP server software such as Asterisk, FreeSWITCH, or Kamailio should be installed on the server. These software solutions provide the necessary functionality for call routing, call management, and call control.

IP addresses and ports: The server should be configured with a static IP address and open the necessary ports for incoming and outgoing VoIP traffic.

Security: The server should be configured with security measures such as firewalls and intrusion detection systems to protect against unauthorized access and attacks.

Database: Depending on the size of the system, a database such as MySQL or PostgreSQL may be needed to store information such as user accounts and call records.

Configuration: The VoIP server software should be configured to work with the specific requirements of the system, such as call routing rules, codecs, and user authentication.

Integrating with a carrier's network is an important step in implementing internet to SIM-based calling. This allows the VoIP system to connect to the traditional telephone network and make and receive calls to and from landline and mobile phones.

The following are some key aspects of integrating with a carrier's network:

Signing up for service:

Before integration can take place, a service agreement must be signed with a carrier that offers SIP trunking or other VoIP connectivity services.

Configuring the VoIP server:

The VoIP server must be configured to work with the carrier's network. This includes setting up the necessary protocols, such as SIP, and configuring the server's IP address and ports.

Authenticating with the carrier:

The VoIP server must be authenticated with the carrier's network. This is usually done by providing the carrier with the server's IP address, username, and password.

Configuring routing:

The VoIP server must be configured to route calls to and from the carrier's network. This includes setting up rules for call routing, such as time of day routing and least cost routing.

Testing:

The VoIP system should be thoroughly tested for functionality and reliability before it is put into production.

Connectivity :

Carrier network should be connected to the VoIP server via internet connection or leased line, ensuring high-quality call transmission.

Interoperability:

The VoIP system should be tested for interoperability with the carrier's network to ensure that it can make and receive calls to and from all the necessary telephone numbers and features.

VIII. CONCLUSION

Before the advent of VoIP technology, Internet users were expecting the risks associated with sending data over the Internet, but they were expecting a confidential network for their voice calls.

In conclusion, this paper has discussed the concept of SIM-based VoIP, which allows for internet-based calls to be made from a mobile device using a SIM card. The proposed system utilizes VoIP technology and SIP protocols to connect the mobile device to the traditional telephone network, enabling calls to be made to and from landline and mobile phones. The system was designed with an emphasis on ease of use and cost-effectiveness, making it a viable alternative to traditional cellular calling. The implementation of the proposed system involves integrating with a carrier's network and configuring the VoIP server to work with the carrier's network. The system was tested for functionality and reliability, with good results. Overall, the proposed SIM-based VoIP system represents a significant advancement in mobile communication technology and has the potential to revolutionize the way we make phone calls.

IX. FUTURE ENHANCEMENTS

Cloud Computing :

Cloud computing will become an increasingly valuable asset in the IT industry. While some companies have already adopted this technology, the real value will come when all VoIP providers will use cloud computing as a means for information and data storage.

Forecasters predict that the majority of businesses will transition from on-premise PBX to the cloud over the next couple of years. This will provide HD audio quality, optimal scalability, and a unified system of communication – without location restrictions – across an entire organization.

Unified Services

VoIP providers are becoming increasingly integrated, offering interactive options to help boost the efficacy of business operations. Using VoIP solutions to unify your communication system means bringing together emails, calls, messages, and other tools that facilitate communication all within a single medium.

Integration with IoT

VoIP’s connectivity capabilities cannot be ignored – especially with millions of things connected with the internet. Think about it: even now, you can use your mobile device to remotely set your thermostat through the use of the internet. As the IoT continues to expand to monumental proportions, VoIP phones will have crucial automation components available. There are nearly endless possibilities when you think of all this technology is capable of.

Cybersecurity

In today’s digital world, cybersecurity is the primary concern of both providers and companies alike. When the proper system monitoring tools and security measures are in place, cybersecurity can potentially become a manageable, less threatening part of the business.

With such an extremely competitive marketplace, businesses can't afford to fall behind on trends and valuable tools. VoIP is one of those trains you need to hop on before it's too late. Companies need to be prepared for technological advancements as they arise each month, not each year.

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