International Journal of Research Publication and Reviews

Journal homepage: www.ijrpr.com ISSN 2582-7421

Design of Real-Time Voice over Internet Protocol (VOIP) System under Bandwidth Network

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A B S T R A C T

Most people know the VOIP through the consumer of SKYPE that has found public recognition in recent years. However, SKYPE is just one example of the implementation of VOIP which have important technological history and a close relationship with the telecommunications industry. The ideas of Voice over IP were 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. Since 1970, telecom companies have begun to offer Actuator IP software for their telephony equipment. The human voice is an analogue wave signal and historically calls had been created on the network of analogue circuits which provided End to end links for every call and had been known as switching circuits. Most of the 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). Advances in VOIP technology were lead to the availability of telephone software of computers by many software providers. Gateway servers and voice processing cards are the interface between the PSTN network and the Internet that enable users to make calls through pc and the IP phone.

Keywords: Bandwidth, Attenuation,Data,Data Transfer Rate,Communication

1. Introduction

VoIP or Voice over Internet Protocol offers numerous benefits to businesses that incorporate it into their communication systems. VoIP systems cost less than traditional telephone lines and provide services for users to forward and round calls even in peak hours. Before consumers were hesitant to use VoIPs since back then, calls were of low quality. This makes conversing harder for users. Now, however, because of the immense competition of VoIP providers in the market, improvements in the quality of services are felt by consumers. Businesses are now more enticed to use VoIP systems. A large number of organizations today use business VoIP systems. Businesses like call centres, telemarketing companies, and software development solutions profit from VoIP. Your company can take advantage of the benefits effective VoIP systems can give you. It increases productivity and improves how to target customers view the kind of business you are running. Consumers need to understand the importance of VoIP systems. To do this, users need to be familiar with how VoIP differ from traditional phone service. In using the telephone, the service providers connect calls through a series of copper wires. However, phone companies provide multiple sets of these wirings to connect calls to different areas if in case you need to contact several people at a time (Kumar&Roy, 2021; Packer&Reuschel,2018).

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The aim of this paper is to Design and evaluate VOIP and how they can be used to facilitate communications access for users with disabilities (for example using real-time text, text-to-speech, and speech-to-text) and to know what the basic requirements are to provide such services give both practical and general knowledge concerning Voice over IP. The emphasis will be on the relevant protocols, what they are, how they can be used, and how they can be extended. Understand the key issues regarding quality-of-service and security.VOIP works by converting analogue voice signals into digitized data packets. The packets are sent out across the internet the same way as any other IP packets, using the internet's TCP/IP protocol. VOIP is often used abstractly to refer to the actual transmission of voice (rather than the protocol implementing it).

Voice over IP (also called VoIP, IP Telephony, and Internet telephony) refers to technology that enables the routing of voice conversations over the Internet or a computer network. To place calls via VOIP, a user will need a software-based sip phone program OR a hardware-based VOIP phone. Phone calls can be made to anywhere / anyone: Both to VOIP numbers as well as people with normal phone numbers.

For the past 50 years, companies have been using conventional PBX systems which require separate networks for voice and data communications. But with the new VOIP telephony revolution, businesses are quickly migrating to VOIP PBX systems, which offer the huge advantage of converging data and voice networks (McInnes et al., 2020; Nuño et al., 2020).

VOIP, which stands for Voice over Internet Protocol, is the transmission of voice traffic over IP-based networks. Initially designed for data networking, the Internet Protocol (IP) was adapted to voice networking following its successful positioning as the global standard for data networking.The human voice is an analogue wave signal and historically calls had been created on the network of analogue circuits which provided End to end links for every call and had been known as switching circuits. Most of the 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, analogue systems were replaced by digital networks and telephone exchanges were done by high-speed leased lines. This exchange was used from digital technology computers and digital signal protocols such as ISDN but communication was still established via circuit switching, and copper wires. Since 1990 the companies that have manufactured phone equipment and communications, have started to increase the use of digital data transmission ideas between exchanges through packets related to IP (Altrad&Abdulsalam,2019; Lee et al., 2019).

Merits of VOIP

This system works on the VoIP protocol. The technology establishes calls and sends/receives data over the existing internet network using IP protocol.SIP is the application layer protocol, a signalling protocol that is used for establishing multimedia sessions over IP networks in the internet telephony process. The emphasis will be on the relevant protocols, what they are, how they can be used, and how they can be extended (Alsoubaie, 2019; Naeem et al., 2020).

1. Users will be able to communicate effectively, speedily and most importantly, securely thereby enhancing the privacy and confidentiality of mobile communication.
2. Implementing voice encryption on third-generation GSM data or GPRS servers would in turn result in a better-encrypted voice speed and clarity.
3. The configuration and usage of the proxy server (SIP / Asterisk) through defining call routing and handset registration mechanisms.
4. It will enable the users to communicate with each other in an encrypted fashion.

2. Design Methodology and Materials

The general application set up of the VOIP is shown in Fig. 1.

Fig. 1 Application scenario of VOIP

Input Design:

Before the VoIP can start working, there is some setup requiring which are a follow;

With the new technology and call clarity advances of VoIP business phone systems in the last several years, it is a good time for businesses to make the switch from a traditional provider to a VoIP solution. In most cases, you can get similar or even more advanced features as a traditional phone system for much less cost. To make it easier for your business to switch, follow these seven steps:

1. Figure out how many users you will need

The first step to getting VoIP up and running in your office is to figure out how many employees or users you plan on having. Getting a solid idea of user numbers early on in the process makes certain decisions easier as you go along. It will help you determine how many lines you need, whether or not your Internet connection can support your expected call volume, and which features matter most. With some providers, you can save money if some employees only need an extension and not their number. Check with your potential providers to see what kind of deal they can give you.

2. Make sure your office Internet connection is fast enough

Once you know how many users you are going to have, then you are ready to test your office Internet connection to see if it can handle VoIP for that kind of user volume. This is important for two reasons:

Checking out your connection before getting a VoIP system will save you the headaches and frustration that result from the crappy call quality and dropped calls that accompany an insufficient connection.

It can potentially save you a lot of time and hassle. If you switch to a VoIP system only to find out your connection is insufficient, you either have to cancel your service and go back to what you had, or you have to upgrade the Internet connection in your office, either of which is a hassle.

You wouldn't want to sign up for VoIP only to discover it won't work with your existing Internet. Fortunately, it isn't difficult to check if your Internet can handle VoIP.

3. Sit down and set a VoIP budget

VoIP offers a wide range of features, and it's easy to get carried away when looking at the different add-ons and bonuses you can use. That's why setting your budget early on in the process is important. It will help keep you grounded and make sure your eyes are not bigger than your budget when you start looking at what features and plans are a good fit for your business. In general, the cost starts at around $30 per user to around $60 to $70 per user for premium plans. Whenever possible, pay annually and you will get significant monthly savings.

4. **Determine which features are most important here are lots of different features available**, such as unique phone numbers for each employee, an auto call router, call rules, conference calling, mobile apps and more. You'll want to figure out which specific features will be most useful for your business. For example, perhaps you want a home-based VoIP system for your office, but your business keeps you on the road a lot. If so, you'll want a solid mobile app. Or maybe you're getting a system with more than 10 lines. In that case, in-depth call routing rules are a must.

Here's what you should do to help narrow things down: Write down specifically why you are getting a business phone system, what you want it to accomplish, and what its primary uses will be (calling out, conference calling, calling while on the go, etc.).

When you get down to choosing your VoIP phone provider, the two feature lists you just created will make the selection process much quicker and more efficient.

5**. Choose a VoIP phone service**

Now it is time to choose your provider. Before you make the decision, make sure you have your number of users, budget and feature list handy for reference. When making your decision, you want to compare providers and find the one that:

For an in-depth breakdown, consult this comparison guide to the leading VoIP providers.

6**. Order phones and other hardware**

After settling on your service, it is time to order your hardware. Most VoIP services can be used without special VoIP phones. For example, you can often purchase an adapter that you can plug your analogue phones into if you want to continue using those. Or, in many cases, you can use your computer, tablet or mobile phone.

However, to fully use all the extra VoIP features, you may need to purchase a VoIP phone. The nice thing is that there is quite a wide variety of VoIP phones available, from basic and cheap to sophisticated and expensive. Check this article on the best VoIP phones if you have trouble deciding and need some help.

7**. Set up and configure your VoIP system**

In most cases, you just plug your phone into your Ethernet system, configure your settings on your phone, and you are good to go. In some cases, if you have a firewall in place, you might have to go back and forth with your provider a bit to get things working properly. If you are tech-savvy, you should be able to handle most VoIP setups.

To make things easier, each provider offers in-depth tutorials. If you are technically challenged, you may want to go with a provider such as Native, which will walk you through the setup process step-by-step over the phone (Ortega et al., 2019).

3. System Implementation, Result and Discussion

For a good program to be considered working perfectly and implemented, checking for errors carefully and debugging is required for better performance of the program. And to certain, the program was checked for errors before it could be uploaded on the server-side for the general public. So for this reason it will be better to carefully take out time to check for errors in the program and debug (remove the errors) for better operation of the proposed system.

Direct Cut-Over

This method involves the old system being completely dropped and the new system is completely implemented at the same time. The old system is no longer available. As a consequence, you must be sure that the new system is functional and operational. This conversion method is used when it is not feasible to continue operating two systems together. Any data to be used in the new system must be converted and imported from the old system. Users must be fully trained in the operation of the new software system before the conversion takes place.

How Application Works

Internet Phone Setting: Hit the settings/options button on your Android device. Select 'Settings' and then 'Call settings'. Scroll down to the bottom of this menu and you will see 'Internet call settings'. Select 'Accounts'. You will see the option to "add accounts" here. Again you will need your SIP account credentials, which you can find in the email we sent you when you signed up.

User Credentials

Select 'Add account' and enter in your credentials as follows.
Username > Username
Password > SIP Password
Server > Domain
Set as primary account (used for outbound calls) >Your choice
Open up 'Optional Settings'.
Authentication username > Authentication Username
Display name > Your choice
Outbound proxy address > Outbound proxy: sip.onsip.com
Hit the 'back' button to return to your accounts menu. The status of your account will be published here so you will know if you did anything wrong. You can activate the option "receive incoming calls" here.

Program Description

Now, a lot of people have been asking about the coding language of VoIP. Well, you see VoIP is voice over IP technology and it is divided into two major parts, the first part is voice streaming and the second part is called signalling. Now if you look towards the broader site then yes call signalling can include some coding and mostly it's SIP that is used when it comes to the signalling of the call. On the other hand when it comes to voice streaming then yes it also has some codifications for the transmission of the sound and all of this is carried out via Codec. SDK (Software Development Kit) can use to make the development easier too (Jiménez Herranz, 2018).

Display of Graphical User Interface

Figs. 2. and 3are some of the interface screenshots

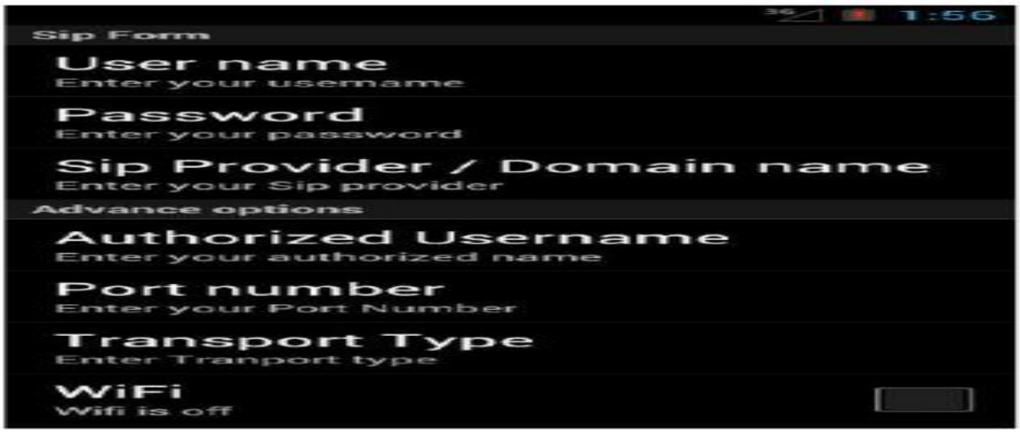

Fig.2. Screenshot of the IP dial pad

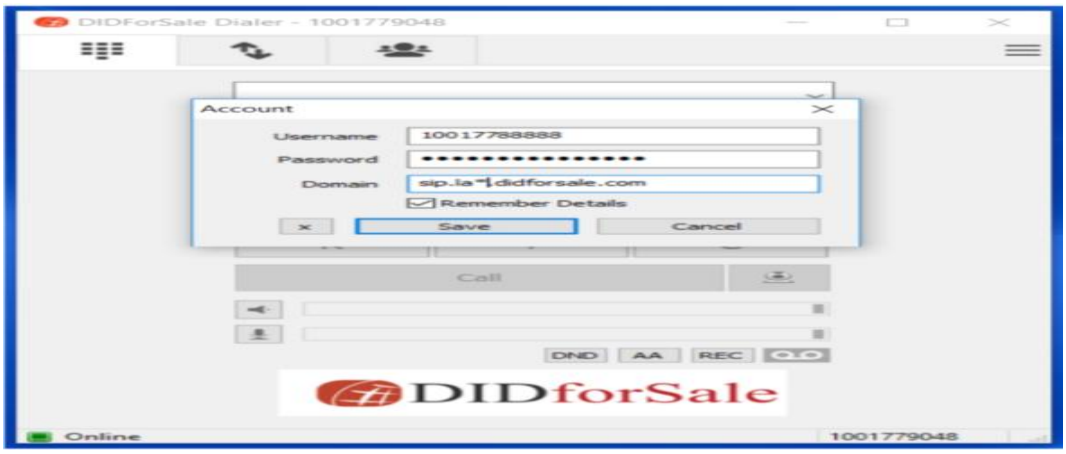

Fig 5. Screenshot of Sip Configuration Settings

4. Conclusion

VOIP is one of the emerging technologies used for communication where one can communicate with whoever they intend to in the world by using an appropriate VOIP service provider. Due to advantages, features and number of service providers, the cost of VOIP has become less expensive. So the main concern about this technology is QoS and security provided for the users. Among these two main concerns, this project mainly focuses on security. This document analyses the available encoding techniques of the voice signal and suggests an efficient technique for the network developers. Encoding techniques considered are G.711, G.723 & G.729. The usage of these techniques can be done in various networks however; one should know the Quality of service (QoS) that these techniques provide. With the QoS, one will know where these techniques fit, what level of security they provide, where they can be used and how efficient each technique is. In this dissertation, implementation is done in OPNET to compare the characteristics of each encoding technique. Implementation is done by designing two scenarios with a network that connects different networks over a wide area and the second scenario with a small area network. VoIP service is configured in both scenarios. Each scenario is then implemented with the encoding techniques considered. Finally, the characteristics of the encoding techniques are evaluated and analyzed using the graphical results obtained. There are several QoS parameters in place but Throughput, Load, MOS (mean opinion score), and Jitter and PDV (Packet delay variation) parameters are measured in this implementation as the characteristics can be evaluated for encoding technique comparisons graphically. These comparisons would allow us to carefully design the encoding techniques for security enhancements in future.

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