

# STUDY OF VOIP SERVICES AND ITS APPLICATIONS

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**Abstract**— Now a day's 'IP Telephony' is becoming an increasingly popular subject. Internet Telephony (Voice over IP) essentially means a voice message transmitted using the Internet Protocol. IP Telephony generally refers to the use of the SIP signalling protocol used to setup, control and manage voice. The Session Initiation Protocol (SIP) is a signaling protocol for initiating, managing and terminating voice sessions across packet networks. IP Telephony based on SIP through mobiles phone enables the user to make voice call at very low call rates. One can use SIP phones to call your friends and family at any geographical location. But the restriction is both the party communicating must be SIP users. Because of the use of Internet facility helps in reducing the call rates. IP Telephony based on SIP through mobiles phone provides mobile environment for VoIP calls. VoIP would fail if remained wired. So through this system one can make VoIP call from their mobile phones which are having Internet access facility. IP Telephony based on SIP through mobiles phone had achieved the feature of portability which is absent the traditional telephone call system.

**Index Terms**— VOIP, RTP, SIP, PROXY SERVER

## 1 INTRODUCTION

VOIP stands for Voice over IP. VOIP is a technology which is used for transmission of voice packets on packet switched IP networks. VOIP is also known as IP telephony or Internet telephony. The simple idea behind the concept of VOIP is to send the digitized samples of voice over data network and then replay these packets at receiver end to hear the original voice. We can send audio, video or timed text over VOIP. In general, it can be used for multimedia communication over IP network. Though it incurs cost for network connectivity, many applications like Skype [5], Yahoo Messenger [6], Google Chat [7], etc. allow "free" calls to their users. Here, free refers to a marginal cost of making the user call. Long distance phone calls such as Jumblo[8], are also cheaper than traditional phone calls. VOIP also reduces the infrastructure cost as a only a single network is required to carry the data as well as voice packets. Packets are only delivered when there is a media content to be delivered. The user can receive and make calls from anywhere. It is also possible to combine VoIP services with other systems such as email, conferencing, and so on. Some modes of operations for VoIP are [4]:

- 1) PC-to-PC,
- 2) Phone-to-Phone,
- 3) PC-to-Phone,
- 4) Phone-to-PC, and
- 5) Network to Network.

Some important technologies and protocols related to VoIP are:

- 1) H.323 [9],
- 2) Session Initiation Protocol (SIP) [3],
- 3) Media Gateway Control Protocol (MGCP) [2], and
- 4) Real-time Transport Protocol (RTP) [1].



Figure 1  
Typical VoIP configuration

Table 1 - VoIP Call Ratings

R value (lower limit)	MOS (lower limit)	User Perception
90	4.34	Very Satisfied
80	4.03	Satisfied
70	3.6	Some users dissatisfied
60	3.1	Many users dissatisfied
50	2.58	Nearly all users dissatisfied

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## 2 VOICE OVER IP

### 2.1 Threats / Risks

Many well known threats to internet applications also apply to the VOIP. A VOIP application may face different attacks like spam over internet telephony (SPIT), spoofing and identity theft.

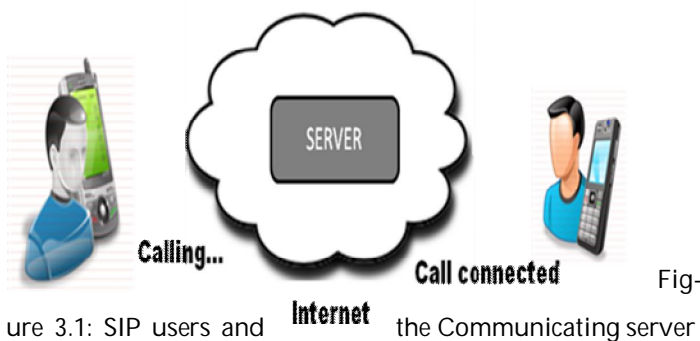
### 2.2 Spoofing

In this type of attack an attacker masquerades as another VoIP caller. For example, an attacker can try to insert a fake caller ID into an ordinary VoIP call so that the receiver believes the call to be coming from a known and trusted source (a bank, for example). A receiver may place trust on a user whose identity has been faked by the attacker. In such conversation, the receiver may be tricked to disclose his personal information like account numbers, social security numbers etc. This is same as the traditional phishing attacks where a user follows links in an unsolicited email and is tricked to provide personal information. The bits and pieces of the information provided by the may be used to complete the identity records of victims for identity theft.

### 2.3 Confidentiality concerns

VOIP voice/data packets may sometimes be sent unencrypted over the internet. This raises questions regarding the confidentiality of VOIP service. So, it is possible for an attacker to collect the data traveling over VOIP system. He may also try to reconstruct the conversation.

## 3 THE PROPOSED SYSTEM



The input to the system is valid SIP user number, which the user gets while registration such as 101, 102. According to dialed number the call establishment is done with help of the server. The server here acts like a proxy which helps to connect call. The server here is responsible to initiate, manage and terminate the call. The callee can accept or reject the call. The subsystem i.e. the server will also generate different tones like busy tone, ringing tone etc. It also helps to add new user or delete users. The functionalities of subsystem are explained below

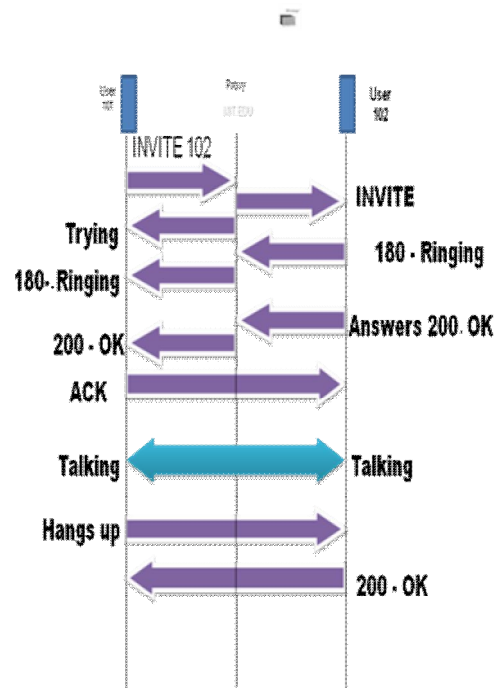


Figure 3.2 : Basic flow diagram showing communication between two SIP users

### 3.1 SIP Server

The SIP server is used to resolve usernames to IP addresses, so that requests sent from one user agent to another can be directed properly. A user agent registers with the SIP server, providing it with their username and current IP address, thereby establishing their current location on the network. This also verifies that they are online, so that other user agents can see whether they're available and invite them into a session. Because the user agent probably wouldn't know the IP address of another user agent, a request is made to the SIP server to invite another user into a session. The SIP server then identifies whether the person is currently online, and if so, compares the username to their IP address to determine their location. If the user isn't part of that domain, and thereby uses a different SIP server, it will also pass on requests to other servers.

In performing these various tasks of serving client requests, the SIP server will act in any of several different roles.

- Registrar server
- Proxy server
- Redirect server

### 3.2 Registrar Server

Registrar servers are used to register the location of a user agent who has logged onto the network. It obtains the IP address of the user and associates it with their

username on the system. This creates a directory of all those who are currently logged onto the network, and where they are located. When someone wishes to establish a session with one of these users, the Registrar server's information is referred to, thereby identifying the IP addresses of those involved in the session.

### 3.3 Proxy Server

Proxy servers are computers that are used to forward requests on behalf of other computers. If a SIP server receives a request from a client, it can forward the request onto another SIP server on the network. While functioning as a proxy server, the SIP server can provide such functions as network access control, security, authentication, and authorization.

## 4 ADVANTAGES OF THE SYSTEM

1. Reduce long distance calling cost: The main benefit that attracts multiple customers is the cost of phone calls over IP and to be more precise almost a complete absence of fees for international and long distance calls that are charged so high by phone companies.
2. Portability: Wherever you have an access to mobile Internet connection you can sign up with your SIP account and make and receive phone calls to your VoIP number.
3. Functionality: Since VoIP is digital one can offer more services together with voice such as call redirection, call waiting and call on hold, audio conference etc.
4. They simplify and enhance telecommunications for business users.

## 5 DISADVANTAGES OF THE SYSTEM

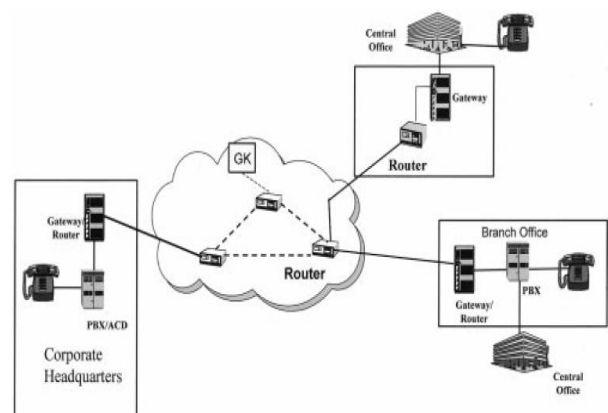
1. This SIP API is expensive to use.
2. SIP has disadvantages depend lot choices of client phone, hardware, network, and Internet telephony service provider.
3. Minimum Requirements: Dial-up service may be sufficient for your Internet needs, but it won't be powerful enough to carry voice.
4. High-speed service, usually broadband is usually considered a minimal necessity for successful SIP service.
5. No facility to make Emergency Calls.

## 6 APPLICATIONS OF THE SYSTEM

Mobile phones: Because SIP client software is lightweight; it can be embedded in mobile phones so that these services can cross all platforms. Using SIP as the signalling protocol means that sessions can be established between different SIP user that then negotiate the appropriate media capability.

LAN VoIP Calls: The softphone developed help the LAN user to communicate with each other for free. The maximum number of the SIP user supported is 20.

Business purpose and private network users: Help the SIP user in a virtual private network to establish call within its network. It would also provide facility of conferencing for the employee who is not physically present at office.



Business use of VOIP

## ACKNOWLEDGMENT

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