Proposed System for Free Calls over Wi-Fi Network Using Voip and SIP: Android Application

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Abstract- *Today's era of technological advancement has made humans inclined to a great extent towards communication and dependent on free and procurable Internet services. Free and wide availability of Internet services has made communication feasible and more sustainable to mankind. In this paper, we have proposed the design of a Session Initiation Protocol (SIP) based client that uses Voice over Internet protocol (VoIP) at the core level, for a dedicated organization. This allows the stakeholders of the organization to enable and use the facility of audio calls over a public and private network. Peer-to-peer calling facility over Wi-Fi will help in reduction of bills over time. The outcome of this paper will be a Devoted, Dedicated and Dimensional (D3) client application, on Android platform.*

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

I. INTRODUCTION

Mobile phones and portable handsets have become an integral feature of the daily schedule of a human being. Communication over Wi-Fi network and using several other applications, software and open source licenses have contributed to more of this availability. Most of the software designed nowadays is on the most popular mobile operating system, Android. Android supports Java applications, which is more feasible and above all, Java supports the client-server architecture which makes it feasible to use these applications with effectiveness and efficiency. Today thousands of applications are available that support communication via app-to-app. In this paper, we have proposed the design of VoIP Calling app-to-app for a dedicated organization.

SIP (Session initiation protocol) is a protocol operated in VoIP (voice over Internet Protocol) conveyance authorizing users to make voice and video calls, predominately free or very cheap.SIP is correlative with VoIP considering it provides signalling function to it. Apart from VoIP, it is applicable in other multimedia technologies as well, online games, videos and many more services.

The principal sense for which people are profoundly taking bent towards VoIP and SIP is its cost. In era where communication serves as a foundation of planning, which in turn must interact so as to implement the plan. SIP is a way to cut down communication cost, add more features to communication and interaction.

Our purpose is to provide communication between students, staff and administration so that to render the organization more efficient systems. SIP handles communication sessions, on which parties communicate. It provides signalling for building, updating and terminating sessions with one or more communicating participants.

SIP allows peers around the globe to communicate using their mobile phones, PDAs and other devices, over the internet. It is an integral part of Internet telephony and grants us to have advantage of VoIP and a rich communication experience. Even wireless networks, popularly known by the name, Wi-Fi are accessible by each and every computing device and also help the users and the Smartphones, PDAs to communicate over a network. An advantage of such application includes the users to call each other with no monetary deductions and call costs.

VoIP, also known as communication protocol, helps provide communication facility with very less jitter and disturbance in it.

II. RESEARCH REVIEW ANALYSIS

The Author Bhushan R. Jichkar discusses in this paper that nowadays, Wi-Fi, also known as 802.11b, has become the chosen technology for communication over wireless media in both professional and personal environments. Designed initially for private access Wi-Fi has continued to emerge as an universally acceptable technology. User friendly interface and easy access has made this technology much in demand. Today the most cost effective use of Wi-Fi is calling over network. Hence to overcome such issue we are developing a system which allows free calling over Wi-Fi network using VoIP service. Our system allows peer to peer calling and an additional feature of group calling. For supporting group calling feature we are using SIP (Session Initiation Protocol). This technology will help us to find a way for free calling and thus ultimately helps to private organization for reducing bills over communication. In this paper, the author discusses that the system will be implemented as well as the performance will be tested by doing actual implementation of this system.

III. PRODUCT REVIEW ANALYSIS

Even before the proposal of this design, several Android applications and open-source softwares have already been released, and approved by the users and the people all over the globe.

Some of the leading softwares and their various features have been briefly discussed in the table designed below. The table elucidates the feature that provides a brief insight into the open source softwares.

Table 1: Available softwares for SIP Calling

Software	Asterisk	FreeSWITCH
Features	Voicemail, conference calling, interactive voice response (phone menus), and automatic call distribution.	Interactive voice response (IVR) services, video conferencing with chat and screen sharing
OS	Linux, NetBSD, OpenBSD, FreeBSD macOS, and Solaris	Linux, Windows, MacOS and FreeBSD.
Type	Free and Open software	Free and Open software
Supports	SIP (session initiation protocol), Media Gateway Control Protocol (MGCP) and H.323.	Has full support for encryption, ZRTP, DTLS, SIPS.

Skype, Whatsapp Messenger, Facebook Messenger have ruled and continue their journey of being the magnificent still, with atleast one Android application being released all over the globe. However, the feature that distinguishes our proposed Android application design is its working for a dedicated organization.

Table 2: Available softwares for SIP Calling

Software	FreePBX	Mysipswitch	SIP Express Router
Features	Voicemail, Unlimited IVR's, Unlimited Extensions	videoconferencing and instant messaging application	SIP telephony system, SIP load balancer, SIP security firewall
Operating System	Linux	Windows, Linux, Mac OS X and Android.	Linux, BSD, Solaris
Type	Open source GUI	Free and Open software	Free and Open software

IV. OBJECTIVES

1. A service that provides the facility of free calling and communication, with effective and friendly environment.

2. For a dedicated organization, the administration will be relieved of the vast problem of communication gap and provide easy and feasible interaction.

3. The users will be provided with a dedicated number, like the general GSM and CDMA providers.

4. The server will be dedicated for one single organization.

5. The Android application provides an active feature, which shows whether the user or that particular SIP Client is available within the Wi-Fi zone or not.

V. RESEARCH DESIGN

Research Design elucidates the logical structure of the project which is to be researched upon and the order of the plan of execution. The proposed technical infrastructure will work on devices with the installed Android application and the mobile phones with Wi-Fi working, of that particular organization.

The several technicalities that are required are:

- Both the parties should be registered at the SIP Server.
- Both the parties should have the Android Application installed on their handsets.
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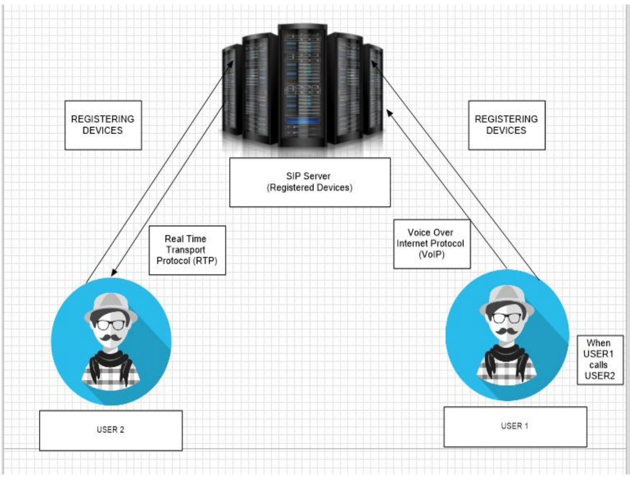

Fig 1: Proposed Design chart

The proposed system stores the Domain server address and a contact book is also present, for easy and flexible access for calling and communication, within the organization. The clients register themselves and proceed with the calling over the Wi-Fi network.

1. When the caller calls the receiver, Session Initiation Protocol (SIP) is initialised between the two connecting parties and a connecting session is established.
2. The receiver acknowledges the incoming request for a call, and answers to take the incoming call requests.
3. Real Time Transport Protocol (RTP) channel is established between the two parties to facilitate communicates in an effective, efficient and well-connected environment.
4. Finally, the two users can communicate over VoIP using Wi-Fi network.
5. When call terminates, the session initialised by the SIP Server is also destroyed.

VI. FLOWCHART OF THE APPLICATION

1. The caller handset installs and configures the Android application, to activate the free call over the Wi-Fi to the receiver. The caller searches for the contact over the address book to make a call.

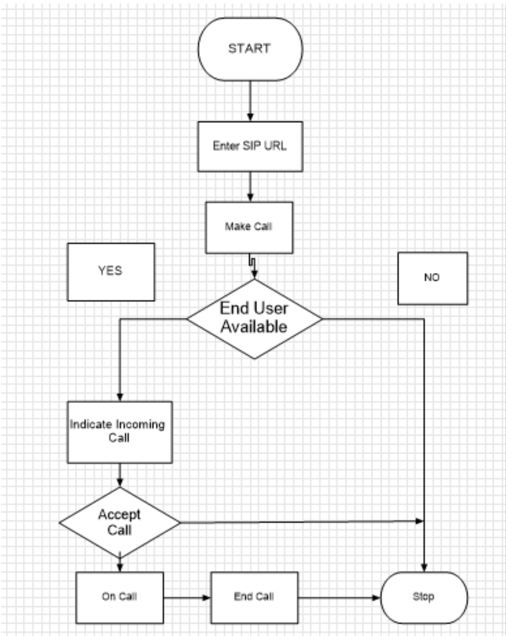

Fig 2: Flowchart

2. Before the call is connected, both the sender and the receiver parties have to record themselves with the assigned SIP server in order to register their calling SIP number, which is also signed up on the Android client so that the user's account gets activated.
3. Once the SIP account is activated, calling can be initiated over the Wi-Fi network, free of cost and also, facilitates much of ease and availability of the user.

This Android application requires a good and strong bandwidth Wi-Fi connection, and even registering within the SIP Server of that particular dedicated organization.

VII. SIGNIFICANCE

1. The proposed design benefits the users of that specific organization to access free calls over the public and private network.
2. The design specifically works for the WiFi network of that particular organization.
3. The caller browses the contact book, which is saved initially, within the database deployed.
4. As the proposed Android application design works for a dedicated organization, all the users can just register themselves on the SIP server and can save their numbers on the database directly.
5. There are least chances of any discrepancy from the numbers or contacts being misused, as the WiFi of that dedicated organization can only be used to access the SIP calling.

VIII. IMPLEMENTATION

(i) **Configuration of client on SIP Server:** The client is registered on the server with a username and password. The generic details are fed into the log of the SIP server and a logical number is assigned to the client, which is the telephone number of the user.

The user is shown inactive, right after his registration onto the SIP server. An active status is shown, when the user configures on the Android application, as well.

(ii) **Installing application on Android handsets:** The Android handset is connected to the workstation via a USB Cable, and the USB Debugging is enabled. The Android application is accessed on the Android IDE, Android Studio.

The application is installed and the home screen interface appears on the screen. The inflator on the handset is accessed and the configuration is further performed.

(iii) **Activation of SIP Client:** The application is accessed on the Android handsets. The user configures the SIP account by entering the login credentials – username, password, telephone number and the domain address, all of which are designated by the SIP Server.

The SIP account will be activated, and it will be configured on the Android handsets, where the application has been installed.

IX. TESTING AND RESULTS

The implementation requires the developer to test the developed product and conduct test cases for all the available conditions, from best to worst and to manual labour.

The Android application for free calling over the SIP Server was tested on various cases and time allotment and access points.

The following observations were noted down:

Table 3: Observation Table

Position Of The Contacts In The Address Book	Best Case Access Time	Manual Case Access Time	Worst Case Access Time
1	0	1	1
2	1	2	3
3	2	3	5
4	3	5	10
5	4	9	15

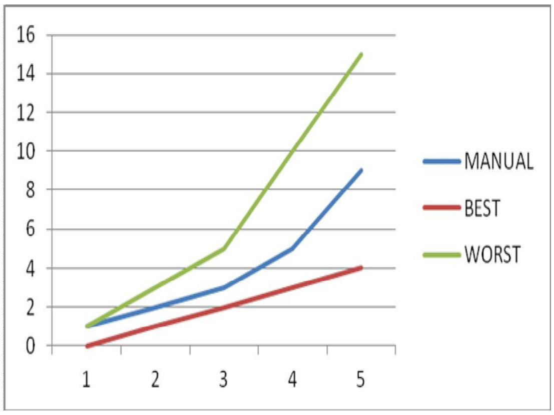

Fig 3: Observation Graph

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