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SIP Implementation for Ad-hoc Network

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***Abstract*— Telecommunication services are transforming from their traditional circuit-switched networks into packet-based networks that facilitate new services that combine data, voice, and video information. Voice over IP (VoIP) uses the Internet Protocol (IP) to transmit voice as packets over an IP network. The growth of this technology is due to the integration of voice and data traffic over wireless infrastructure that leads to Mobile VoIP (mVoIP). This paper deals with the implementing voice over Internet protocol (VoIP) over mobile devices using android platform. First, the paper discusses the factors involved in making a cheaper call. After a discussion of serving telecommunication service to achieve mobility**

***Keywords*— VoIP, mVoIP, voWi-Fi, SIP.**

I. INTRODUCTION

The telecommunication service is a commodity that is not affordable by every citizen. Calls from both mobile and fixed line service providers are billed per second or minute at rates that an average person can't afford. Telecommunications systems are very expensive to implement and therefore companies tend to cost their services very high in order to gain enough profits to make a good return on their investments. They have many cost affecting factors to consider (e.g. power, space and staff), since this systems are cumbersome and can take up a lot of room space and uses a lot of power. Access systems are also made up of expensive technologies which sometimes hinder them from deploying their services in low-populated remote rural areas, which in-turn makes the services inaccessible to the many small communities leaving in these areas. Nearly 70% of the Indian population leaving in rural areas, there is still a need for affordable and easily accessible telecommunications services for many.

With all this expensive methods of communication data packages that Telecommunication companies offer to their customer, at monthly flat rates where the monthly usage is not covered. Customers are keen to use the data services for downloading content or simple just serving the net.

This project therefore seeks to deploy a solution where users can enjoy telecommunication services whilst bypassing the traditional method based service provider. With the rapid deployment of mobile access networks combined with IP-based access to content, cheaper solutions to telecommunication should be feasible.

II. MOBILE VOIP

IEEE defines the standard 802.11 for implementing wireless local area networks (WLAN) in the 2.4, 3.6 and 5GHz frequency bands. The IEEE standard 802.11 has received a series of amendments over the years which are specifications for over the air modulation techniques that use the same basic protocol. The emergence of voice over 802.11 (Vo802.11) was made possible by simple moving VoIP over 802.11 as an access mechanism. Once the VoIP stream reaches the wired part of such a network via a wireless access point, it is transported on an IP network. Mobile VoIP only caters for the wireless access side of the VoIP network, whilst the servers discussed in the VoIP section are also active on the wired side of the network [5].

On the other side of communication, there is a vast growth in broadband and 3G technology where data is charged using flat/fixed fees which takes fewer of the tariff parameters in consideration. Service providers do not have different tariff tables for different destinations connected to the IP- based data network. Therefore utilizing a single network infrastructure to carry both voice and data reduces costs and introduces improved data management and communication efficiency within an organization. The convergence of voice and data gives rise to the term VoIP. By adopting VoIP, both customers and service providers will have benefit.

III. PROPOSED WORK

This section introduces proposed system "**SIP Implementation in Ad-Hoc network**". The main objectives of this system are:

1. It will enable the users to communicate with each other in low cost by integrating voice and data networks.
2. Implementing mVoIP either on Wi-Fi or mobile data that gives mobility to user.

A. *Design*

Design specifies the logical structure of a project and the plan, will be followed in its execution:

Suppose **A** (caller) wants to communicate with **B** (callee) through secured voice communication. **A** and **B** must have android based mobile handsets with mVoIP application installed on it which is responsible for voice communication between two mobiles [1][3].

1. The mobile handsets need to be registered at SIP server.
2. When **A** calls **B** the application installed on mobile handsets will convert it into digitized data packets.

3. The data packets will travel through Wi-Fi/mobile data channels.
4. The SIP server will route the call to the registered recipient **B**.
5. The application installed on **B** handset will perform the digital to voice conversion process.

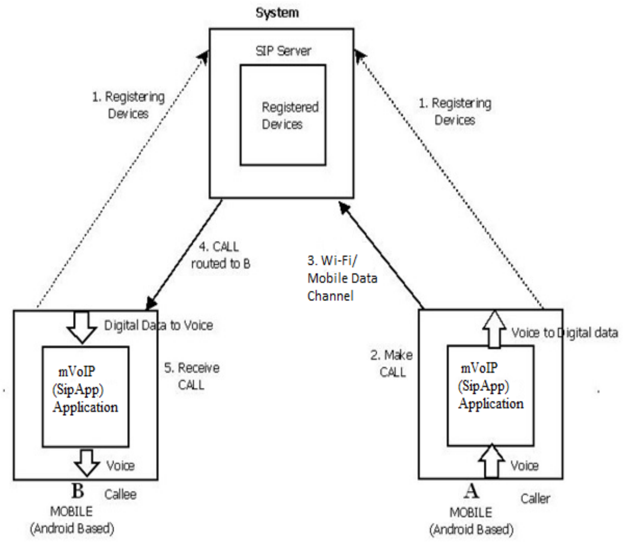

Figure 1: Block diagram of Proposed System.

B. ***Workflow of the Application***

1. Mobile voice communication can occur in 2 ways namely [1][6]:
   a. Wi-Fi (without the SIM card in Android handsets)
   b. Mobile data (with SIM card in Android handsets)
2. There would be one SIP server and minimum two Android handsets with mVoIP application installed on them. The handsets need to be registered at SIP server.
3. SIP client (mVoIP app) is configured on both the Android handsets.
4. The caller will launch the mVoIP Android application on his/her handset and dial the receiver's SIP URL. Once the call is made, the request will go the SIP server wherein the callee's URL will be checked and the call will be routed on its handset
5. Once the callee receives the call through the same mVoIP Android application, caller starts the conversation. All the voice data is digitized on the caller's handset sent to the SIP server for routing to the callee's handset. Logically, every handset will be assigned a SIP URL in the SIP server during configuration through which it will become identifiable.
6. The digitized voice data will travel through either Wi-Fi or Mobile data medium.

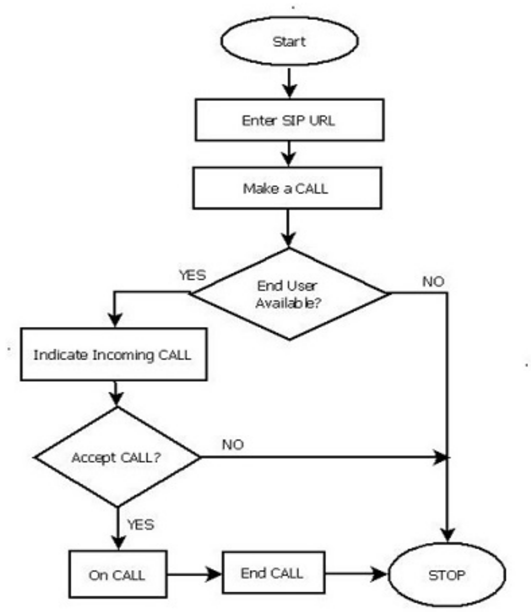

Figure 2: Work Flow of the System.

IV. mVoIP Application Implementation & Testing

Each mobile application has its own set of unique challenges; a customized approach of mVoIP application development follows a sequential set of steps. These steps are [4][7]:

A. Concept sketching- highlights basic functionality of an application.

B. Research & Strategies - identifies classes and interface to make functionality of system. To develop mVoIP system android development platform is used.

➢ **Introduction of SIP API**

Android provides an API that supports the Session Initiation Protocol (SIP) which enables a developer to add SIP-based Internet telephony features to an application. Android includes a full SIP protocol stack and integrated call management services that lets applications easily set up IP telephony, without having to manage sessions, transport-level communication, or audio record or playback directly.

According to the Android website (2016), the SIP API is available in the **android.net.SIP** package. The key class is **SIPManager**, which applications use to :

- Create a SIP Session.
- Initiate and receive SIP calls.
- Register and Unregister with a SIP service provider.
- Verify session connectivity.

Once a call is established, applications can mute calls, turn on speaker mode and more. Applications can also use the **SIPManager** to create generic SIP connections.

C. Wire framing - plans and design the application views.

The first step in the Wire-framing processes is to plan and design the application views. The application is made out of a series of views which are

- SIP Settings, and
- Make a Call

Note that "receive a call" function will only be invoked once for an incoming call and is not part of the root view.

D. User interface design - converts wire frames into attractive interface.

Figure 3 shows the screen that the user will see once the application is initiated. The screen provides the user with the options to

- Make a call
- Add/Change the SIP Settings
- End a call (for Incoming Call).

Figure 3: mVoIP when initiated

E. Development involves coding of application.

F. Testing phase tests an application for various criteria.

- **Physical Android Device Testing**

TEST OBJECTIVE: The test is conducted to establish one-to-one SIP communication between the mVoIP devices. This is necessary to confirm that the mVoIP application is functioning as it should.

TEST CASE 1: Making a Call
Figure 4 shows live RTP being exchanged between the mVoIP device and another SIP Client

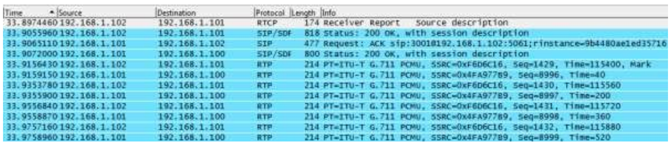

Figure 4: Making a call from mVoIP device

TEST CASE 2: Receiving a Call
Figure 5 shows RTP packets being exchanged between the mVoIP and another SIP client.

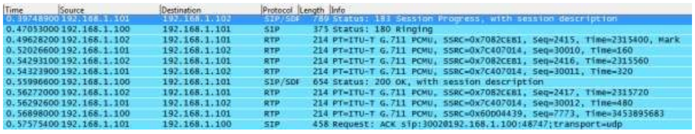

Figure 5: Making a Call to mVoIP device

TEST OUTCOME: Both test has proved that it is possible to make and receive calls from the mVoIP client device and also proves that the SIP protocol stack on the mobile device is fully operational.

- **Network Usage**

TEST OBJECTIVE: The test was conducted to know the network bandwidth usage for upward and downward network traffic.
TEST CASE: Figure 6 shows data traffic exchanged between the two devices.

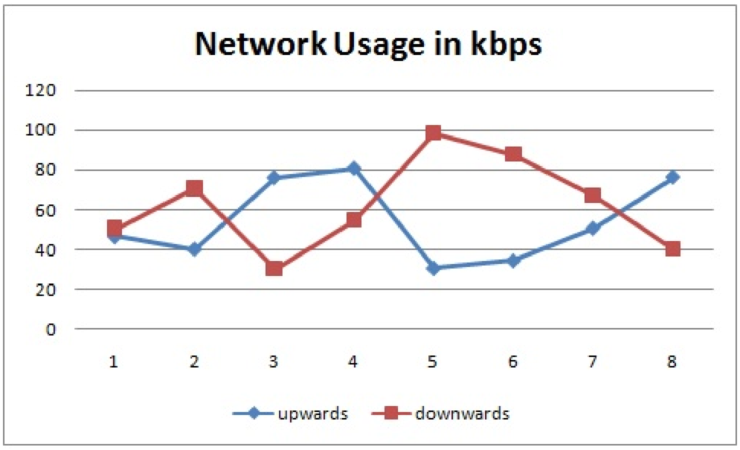

Figure 6: Network usages for mVoIP device.

V. Conclusion

The study has satisfied its primary objective of implementing a mobile VoIP which can be hosted by open source software running on inexpensive backend servers. In its current state, the mVoIP system might not be able to replace a traditional voice system, but it does lay the foundation of how inexpensive systems can be implemented.

The mVoIP system can provide telecommunications services. Looking in the future, the mVoIP project should be enhanced in order for it to compete with traditional voice services. This will require additional research and testing in the different aspects of the system for example, Quality of Service, High Availability and Coverage.

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