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Design and Implementation of SIP based VoIP Application for Mobile Devices using J2ME

T. M. T. Khan*, S. M. Patil**, D. R. Raut***

* *AIKTC/Computer Engineering, New Panvel, India*

** *BVCOE/Information Technology, Khargar, India*

*** *WIEECT/Computer Engineering, Mumbai, India*

Abstract: Voice over Internet Protocol (VoIP) is a way to carry out a telephonic conversation over a data network. VoIP applications, with SIP and RTP, promise converged telecommunications and data services that are cheaper, more versatile and provide good voice quality as compared to traditional offerings. Although VoIP is widely used, VoIP on mobile devices is still in its infancy. Currently, there are number of VoIP solutions for mobile phones, however VoIP solutions developed using Java 2 Platform Micro Edition (J2ME) are not available. Java based solutions are widely compatible with many devices. In this paper strong focus has been given to cross-device compatibility through the use of the widely supported J2ME framework. The design and implementation details of VoIP client using J2ME are illustrated.

Keywords: VoIP, RTP, SIP, J2ME.

I. INTRODUCTION

To date, the Public Switched Telephone Network (PSTN) has been used to conduct telephone calls over a wired network. With the development of computing technology, Voice over Internet Protocol (VoIP) has been established as an alternative to traditional telephony networks. VoIP allows telephone conversation to take place over a data packet switched networks like the Internet. VoIP products promise converged telecommunications and data services that are cheaper, more versatile and provide improved voice quality as compared to traditional offerings. Although VoIP is widely used, VoIP on mobile devices is still an area of research.

II. TECHNOLOGICAL OVERWIEW

VoIP is the digitalizing of voice using an analog to digital converter (ADC), sending this data through a data network and the reassembling of this data to form the original analog format using a digital to analog converter (DAC). VoIP is made of two parts, signaling and data transport. The VoIP signaling function can be performed using protocols such as Session Initiation Protocol (SIP), H.323 and Media Gateway Control Protocol (MGCP). Data transport can be performed by The Real time Transport Protocol (RTP). This protocol is used to deliver voice data during conversation.

A. Session Initiation Protocol (SIP)

The Session Initiation Protocol (SIP) [1] is an application-layer control and signaling protocol. SIP is used to create, modify, and terminate multimedia sessions or conferences such as Internet Telephone calls. The SIP message format is similar to the Hyper Text Transfer Protocol (HTTP) message format. Two main components in SIP are user agent (UA) and servers. UAs are regarded as a client that can send the request and response together. This includes a user agent client and a user agent server. Servers are used to receive requests from clients for servicing and sending responses back to the clients. The servers are typed as redirect server, proxy server and registrar.

B. Real-time Transport Protocol (RTP)

Once signaling functions are implemented, voice data needs to be transmitted between clients. Real-time Transport Protocol (RTP) [2] is viewed as the most powerful protocol to deliver multimedia packets in a session. RTP is defined by Internet Engineering Task Force (IETF). It consists of two parts:

RTP: is used to carry voice data.

RTCP: is used to monitor the quality of services and information about the participants who are in a sessions.

C. Voice Codecs

In VoIP, there are many different audio codecs. The bandwidth required during a VoIP conversation naturally depends on the codec. The G.711 codec is used on most telephony systems all over the world. The G.729 codec provides the best voice quality. However, due to the native support of G.711 by mobile devices, it is more suitable to use G.711.

III. MOBILE APPLICATION DEVELOPMENT ENVIRONMENT

There are number of integrated development environments, languages, frameworks and libraries that can be used to develop the solution proposed in this paper. J2ME is chosen as the development tool in this paper, since most of the mobile phones has support for Java. Although

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RTP is not supported on J2ME, implementation of RTP on J2ME framework is one of key features of this paper.

A. PROPOSED ARCHITECTURAL FRAMEWORK

Depending upon the required functionality of SIP client under constraint specific mobile platforms, the system design is divided into two major proposed frameworks.

a. Proposed SIP Framework

SIP performs signaling functions in a session. In this paper, SIP functions are implemented based on the specifications of SIP Protocol, as shown below.

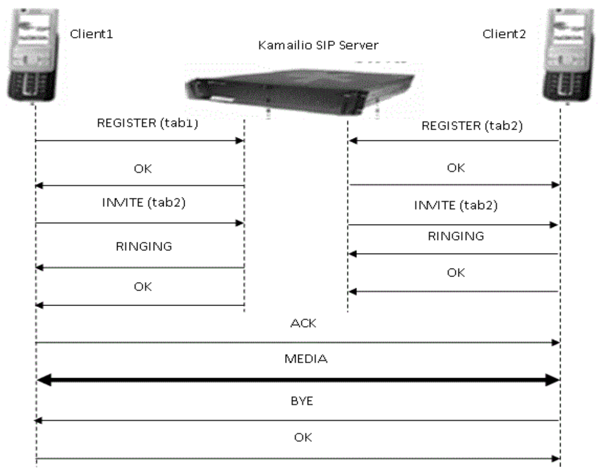

Figure. 1 SIP sequence diagram.

As signaling functions are provided, SDP is used to negotiate media session description between clients. The media session description includes encoding format, payload type and sampling rate of voice packets. Also, IP address and port number for receiving voice packets is included.

b. Proposed RTP Framework

After both parties reach the same media session description, the RTP protocol is used to deliver voice data. Figure 2 shows the details of RTP Framework that is used as a basis for the implementation. The implementation carries out the functions illustrated in Fig. 2.

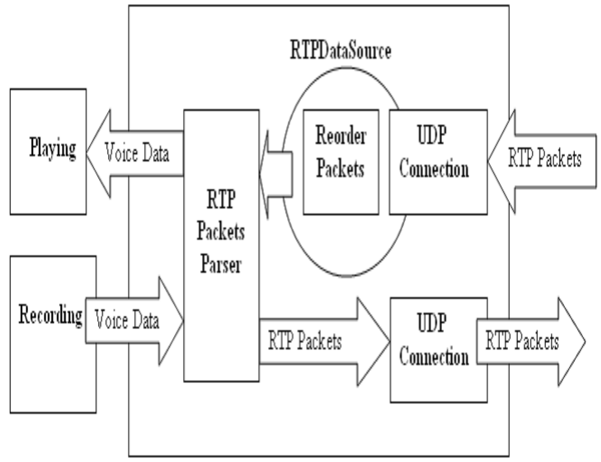

Fig. 2. RTP Framework.

B. IMPLEMENTATION OF PROPOSED FRAMEWORK

In this section, the implementation of the proposed architecture is going to be discussed. Requirements for setting up development environment is presented. SIP for J2ME is implemented to initiate VoIP calls on mobile emulator. Moreover, Session Description Protocol (SDP) is used for negotiating media session description. RTP is used to deliver voice data in this implementation.

Requirements of Proposed Architectural Framework

A number of additional softwares and tools are required for implementation and evaluation. They performs different functions during the implementation process. They are outlined as follows:

a) Kamailio is a free and open source SIP proxy server..
b) Sun Java Wireless Toolkit is a mobile phone emulator.
c) Ethereal is a network protocol analyzer that is used to capture network packets during transmission.
d) J2ME is used to develop the VoIP application.

SIP for J2ME

JSR 180 is an optional package that supports basic SIP API on J2ME [7]. It can run on devices with limited memory. There are six SIP request methods are explained as following in SIP specification: REGISTER, INVITE, ACK, CANCEL, BYE, OPTIONS. Two main requests are introduced in the following:

- REGISTRATION: registration is the first essential part for starting a session. A SIP registration message is generated and sent to Kamailio server. According to SIP specification, the transactions between clients can only be done if they are registered to the SIP registrar server. In this implementation, kamailio acts as registrar server.
- INVITATION: The invitation is the second step that a VoIP call needs to do after registration. According to

SIP API, SIP Invitation Message is generated as follows:

```
INVITE sip:tab2@192.168.1.3
To: tab2<sip:tab2@192.168.1.3>
From: tab1 <sip:tab1@192.168.1.2>
tag=1928301774
Call-ID: a84b4c76e66710
CSeq: 314159 INVITE
Contact: <sip:tab1@192.168.1.2>
Content-Type: application/sdp
Content-Length: 142
```

SDP for J2ME

When a client wants to communicate with another client, the session description details must be received by both parties. Then, the callee and caller will negotiate to choose a common media for voice packets. Session Description Protocol(SDP) [5] is used to describe the media details for clients. The functions of SDP are to define where the voice data should be sent such as the port number and IP address and define the codec to be used by other party and also identify the channel and sample rate of voice data. The following is one of the SDP message example according to SDP specification.

```
String SDP = " " +
"v=0\n"+
"o=tabl 0 0 IN IP4 192.168.1.2 \n" +
"s=-\n"
"c=IN IP4 192.168.1.2 \n" +
"t=0 0\n" +
"m=audio 10086 RTP/AVP 8\n" +
"a=rtpmap:0 PCMU/8000\r\n";
```

RTP on J2ME

As J2ME does not RTP, it is necessary to implement the RTP features. It is widely known that RTP packets house in UDP packets. Experiments shows that audio recording of 1 second generates around 8000 bytes of data. The RTP header size is 32 bytes but in implementation, only 12 bytes for header is used for shortening the total packet size and thereby improving voice quality. The recommended audio recording of 20 milli seconds generates about 160 bytes of data to which RTP header is added resulting into packet of size 172 bytes. Then this data is encapsulated into UDP packet and sent to the destination.

According to RTP specification, the following features are required:

- Constructing/Extracting RTP packets from UDP packets.
- Receiving/Sending UDP packets.
- Streaming and playing voice data.

IV. CONCLUSION

In this paper, the background and related work of VoIP on mobile devices were discussed. The proposed design and implementation were also detailed. This paper proves that a VoIP client can be developed using J2ME followed the standard and deployed on a mobile phone with the necessary features. The features of the implemented client are suitable for mobile devices. Although the implemented client is compatible with the VoIP standard, the client is not implemented for all the existing Mobile Operating systems.

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