Voice over Internet Protocol (VoIP) for Mobiles.


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Keywords: VoIP (Voice over Internet Protocol), SIP (Session Initiation Protocol), SDP (Session Description Protocol), GPRS (General Packet Radioing Service), EDGE (Enhanced Data-rates for GSM Evolution), GSM (Global System for Mobiles), CDMA (Code Division Multiple Access), PBX (Public Branch eXchange), RTP (Real-Time Transfer Protocol), CODEC (COmpression DECompression), QoS (Quality of Service), Asterisk, Mobile Networks, Speex, ITU G series, iLBC, 2G/3G (second/third generation).

ABSTRACT

VoIP - Routing of voice conversation over the Internet or any other IP based network. It’s a technology that allows telephone calls to be made over computer networks, instead of a regular (or analog) phone line.

VoIP application for mobiles were developed using Opensource libraries in C++. These libraries were converted to suite WinCE platforms which were originally meant for standard Windows platform. SIP was used as the application layer protocol for connection establishment between caller parties and SDP was used to exchange session description between those parties. RTP was used as the application layer protocol to exchange voice data after establishment of the connection. Asterisk was used to route calls between users.

System was experimented in GPRS-EDGE and 3G simulated networks with WinCE mobile emulators running the VoIP application. Mobile networks such as GPRS are commonly known for Packet Loss which eventually results in degrading the QoS. Therefore to preserve the QoS, high bit-rate CODEC’s such as G711 were avoided, instead low bit-rate CODEC’s such as iLBC, GSM and Speex were used in the application for voice encoding in 2G networks. 3G networks didn’t encounter such an issue. This was the first VoIP application that was developed in Sri Lanka for 2G networks for GPRS-EDGE connectivity and for 3G.

INTRODUCTION

In the current socio-economic environment of the world, communication plays a crucial role in the world of business. Hence to gain the competitive advantage, people communicate and conduct business while on the move. VoIP is essential in mobile since it provides the benefit of economical and reliable means of communication. In fact, majority of VoIP applications today are PC based and doesn’t support the requirement for mobile communication. Currently, very few mobile application versions exist which were for Wi-Fi and Wireless LANs since they provide wider bandwidth. These wireless networks only provide a dreadfully limited geographical coverage.

These limitations prompted the notion to develop the VoIP application for mobile devices which could be deployed in 2G/3G networks, since these mobile networks are vastly spread and widely used. To enhance the bandwidth of 2G networks, GPRS/EDGE technologies are combined to enable VoIP in mobile.

Currently there are many researches and developments in progress based on 2G and 3G networks for VoIP and some mobile operators are attempting to extend 2G/3G voice services for their
subscriber base across a variety of private and roamed networks. As voice data flows over a general purpose packet-switched network, we used SIP as the VoIP protocol and GPRS-EDGE as the connectivity.

Extremely Low Cost rates in making calls is the most prominent of VoIP since IP systems offer a more economical means for providing communication connections. The other major advantages are user mobility, higher reliability, efficiency and support for innovation.

We have proven the strength and innovativeness of this concept and the technical approach to what extent does the proposed project build upon or moves beyond the current state-of-the-art and to show the significance of the technical and economic benefits of the proposed work. So our scope of this research was to drill the technology to mobile phones and we believe that it will be highly applicable in future and will give lots of benefits to its users.

**METHODOLOGY**

Initially the application registers the user with the PBX and then invites the peer to connect to a conversation. Once we get the “OK” message from the peer, both parties exchange media descriptions. Then the application captures voice data, encode them from Analog to Digital, break into packets and send as digitized packets to the peer. Those packets are sent across through the IP network as real-time streaming media, the same technique used to transfer Real-time audio/video data across the Internet. At the receiving end, packets are assembled in the correct sequence and converts from Digital to Analog. If one peer needs to terminate the session, that particular peer will send a “BYE” message to the other peer involved in the conversation. Then all session data will be destroyed.

We used SIP in the application to register clients to the PBX and to establish call sessions with clients. In order to exchange session data in the application we used SDP. Asterisk is used as the PBX for the implementation. RTP is used to exchange voice data through the established communication session. CODEC’s such as ITU G-series 1, iLBC, GSM or Speex were used to capture and encode user voice and to break them into digitized packets. Modules were written to handle all aspects of the intended application in embedded C++ including the PBX operations.

In order to experiment the GPRS-EDGE applicability of the application, we used a bandwidth limiter to limit a normal network to accommodate GPRS-EDGE bandwidth. By using the simulated network, calls were made between parties to test the application. VoIP applications were executed on WinCE Emulators.

SIP is the IETF (Internet Engineering Task Force) protocol for VoIP and other text and multimedia sessions. It is an application-layer control (signaling) protocol for creating, modifying, and terminating sessions with one or more participants.

SDP is intended for describing multimedia sessions for the purposes of session announcement, session invitation, and other forms of multimedia session initiation.
RTP is a packet based communication protocol that adds timing and sequence information to each packet to allow the reassembly of packets to reproduce real time audio and video information.

To route calls between users, it requires a PBX (Public Branch Exchange). We are using the Asterisk open-source PBX for this purpose. Asterisk is a complete PBX in software.

Our main goal of the project is to deploy VoIP in mobile networks (2G/3G) as well. For this purpose our research study will focus on GSM, GPRS-EDGE, and CDMA mobile networks.

We used the Object Oriented Design methodology (OOD) as the architectural design model. Therefore the modeling environment we used was UML. It helped to specify, visualize, and document models of software systems, including their structure and design.

As the integrated development environment, we used Visual Studio 2005, initially with C# and then moved to embedded C++ along with the OpenSource SIP and RTP libraries.

RESULTS & DISCUSSION

Table 1. CODEC bandwidth consumption

<table>
<thead>
<tr>
<th>CODEC</th>
<th>BR</th>
<th>NEB</th>
</tr>
</thead>
<tbody>
<tr>
<td>G.711</td>
<td>64 kbps</td>
<td>87.2 kbps</td>
</tr>
<tr>
<td>G.729</td>
<td>8 kbps</td>
<td>31.2 kbps</td>
</tr>
<tr>
<td>G.723.1</td>
<td>6.4 kbps</td>
<td>21.9 kbps</td>
</tr>
<tr>
<td>G.723.1</td>
<td>5.3 kbps</td>
<td>20.8 kbps</td>
</tr>
<tr>
<td>G.726</td>
<td>32 kbps</td>
<td>55.2 kbps</td>
</tr>
<tr>
<td>G.726</td>
<td>24 kbps</td>
<td>47.2 kbps</td>
</tr>
<tr>
<td>G.728</td>
<td>16 kbps</td>
<td>31.5 kbps</td>
</tr>
<tr>
<td>iLBC</td>
<td>15 kbps</td>
<td>27.7 kbps</td>
</tr>
</tbody>
</table>

BR = Bit Rate, NEB = Nominal Ethernet Bandwidth (one direction).

VoIP Bandwidth consumption naturally depends on the codec used. Normal conversation includes a lot of silence. So even if one voice call sets up two 64 kbps RTP streams over UDP over IP over Ethernet (which adds overhead), the full bandwidth is not used at all times.

One of the key issues with all traditional VoIP protocols is the wasted bandwidth used for packet headers. Typically to send a G.723.1 5.6 kbps compressed audio path will require 18 kbps of bandwidth based on standard sampling rates. The difference between the 5.6 kbps and 18 kbps is packet headers. There are number of bandwidth optimization techniques used such as silence suppression and header compression. This can typically save 35% on bandwidth used.

At the network layer, mobile SIP performance will be influenced by packet loss and delay due to handovers between mobile IPv4 subnets. TCP multimedia flows over wireless links are negatively impacted by server-side congestion control and slow-start algorithms. Wireless links are notorious for packet loss at the data-link level. In particular, GPRS is known for high end-to-end latency peaks and low practical data rates.

QoS is fundamental to the operation of a VoIP network. The main QoS issues associated which affects VoIP performance is,

Latency - Time it takes for a voice transmission to go from its source to its destination. The ITU-T (International Telecommunication Union) Recommendation for upper bound is 150 ms for one-way traffic. For international calls, a delay of up to 400 ms is acceptable.

Jitter - Refers to non-uniform packet delays. It is often caused by low bandwidth situations in VoIP and can be exceptionally detrimental to the overall QoS.

Packet Loss - VoIP is exceptionally intolerant of packet loss. Packet loss can result from excess latency, where a group of packets arrives late and must be discarded in favor of newer ones. A packet loss of 5% caused VoIP traffic encoded with G.711 to drop below the QoS levels of the PSTN.
Benefits of VoIP.

- Lower Cost
- Higher Reliability
- User Mobility
- Greater Efficiency
- Supporting Innovation

VoIP Security issues.

- Interception of calls.
- Denial of Service Attacks (DoS) ¹.
- Theft of service.

One of the common techniques of providing security is encryption. Encryption can provide security against interception of calls and theft of service of above scenarios. For tackling DoS, Some Session Border Controllers have DoS countermeasures built in ¹.

CONCLUSION

Assuming that two parties (Caller and Callee) using the same codec for communication, codec’s with higher bit rates such as G711 (64 kbps) cannot accommodate in GPRS-EDGE or CDMA2000 Mobile networks since it consumes more bandwidth than they are allowed by those networks. This is due to the IP and RTP headers added by the ISO OSI model that increases the packet size. Further, mobile networks are more prone to packet loss, a little as 5% packet loss in higher bit rate codec’s leads to degrade the QoS and user dissatisfaction where it finds no difference with PSTN (Public Switched Telephone Network).

G711 - 174 kbps both ways,

GPRS-EDGE – 160 kbps,
CDMA2000 – 144 kbps,

Codec’s such as Speex, G.729, iLBC, GSM can be used to avoid the above mentioned QoS, Bandwidth and packet loss problems while preserving the audio quality. In 3G networks, high bit rate codec’s such as G711 are possible to implement since it provides a minimum bandwidth of 700 kbps.

In low bit rate codec’s, we encountered bandwidth wastage where the bit rate varies from 5.3 – 8 kbps and the total packet size ranging 21- 31 kbps approximately, again due to IP and RTP headers. As solutions, echo cancellation and voice suppression can be used.

VoIP uses RTP (streaming media) for exchanging voice data between caller parties. Providing encryption for streaming media is considered difficult due to the fact that it has to be performed in Real-Time. Providing encryption mechanisms for VoIP systems is difficult since it consumes more processing power & memory. In fact mobile phones and devices cannot afford to provide such an encryption, regardless of software or hardware level, due to the fact that they are only integrated with limited processing capabilities & memory ³ ⁴.

REFERENCES