QoS for VoIP in GPRS, Wireless and Wired networks

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Submitted for the courses 2G1325 ‘Practical VoIP: SIP and related protocols’ and 2G1330 ‘Wireless and Mobile Network Architectures’
May 2005

Abstract

With the addition of new services and the active involvement of telephony providers, Voice over Internet Protocol (VoIP) has become the focus of attention. This paper focuses on the Quality of service for VoIP over wired, wireless and GPRS infrastructure. The emphasis is on the QoS parameters like delay or response time, packet loss, jitter and MOS (Mean Opinion Score) specifically over SIP (Session Initiation Protocol) and RTP (Realtime Transport Protocol). The impact of these parameters in different network scenarios is also studied.

Keywords

VoIP, SIP, MiniSIP, MIKEY, RTP, GPRS, 802.11, Access Points, 802.3 protocol.

Acknowledgement

This Research was performed at TS Lab in the Department of Microelectronics and Information Technology (IMIT) at Royal Institute of Technology (KTH), Stockholm, Sweden. The authors would like to thank Prof. G.Q. Maguire Jr. for his guidance and valuable comments that have improved this report. We would also like to thank Johan Montelius, Johan Bilien and Khurram Jehangir Khan for their help with the desired equipment and tests.

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1. Introduction

VoIP is one of the amazing revolutions in the Telecommunication field which allows to make phone calls over a data network like Internet instead of regular (analog) phone line. This packet voice system digitize and compress the voice signal into voice data packets, sends the packets over the internet then decode and reconstruct the signal at the other end and so one can talk to anyone over a regular phone number by using VOIP technology. Now the Telecommunication infrastructure has evolved with the introduction of IP telephony. With the addition of new services and the active involvement of telephony providers, Voice over Internet Protocol (VoIP) has become the focus of attention. Service providers can see huge returns for their investments in the technology and are therefore eager to improvement of Protocols like H.323, G.711, G.723 etc. [12] which are widely used in this Technology.

1.1. Problem Statement

Many issues have emerged with the introduction of voice data in the digital IP world. The quality of service for VoIP is of concern on public internet and a number of factors like delays, packet loss, jitter, R-values, MOS etc. (discussed later in the document) effect the overall quality of the call. In this report we have done some analysis for performance over VoIP protocol precisely (SIP and RTP) over different Network Scenarios like wired, wireless and GPRS by using different codecs.

2. VoIP

Voice over IP (VoIP) is the technology of merging voice and data that has the ability to transmit voice over data networks. It carries voice over IP-based, packet–switched Network (Local Area Network, Wide Area Network) [1]. The terms IP telephony or Internet telephony are alternatively used in place of VoIP. The technology has gained wide acceptance over the internet due to the low costs and scalability. It typically uses Realtime Transport Protocol (RTP).

3. SIP

Session Initiation Protocol (SIP) [2] is basically a physical layer control protocol which can set up, modify and end the session (e.g. IP telephony). SIP VoIP performs operation within the Internet domain and operates by dialing a Proxy Server to connect with a different user on an IP network. SIP based VoIP phone uses an IP address to place a call to other user instead of Fixed telephone number.

Session Description Protocol (SDP) [3] is used in conjunction with the SIP protocol for describing multimedia session for the intention of session announcement, session invitation describing, and other forms of multimedia session initiation. According to RFC 3261 [2], there are five types of services that SIP offers:

User Location: To find the location of the end system for communication.
User Availability: To find if the called party is willing to communicate.
User Capabilities: To negotiate and determine the media capabilities, e.g. a voice codec that is supported by both calling party and the called party.
Call (Session) Setup: Ringing and establishing call parameters at both called and calling party.
Session Management: The transfer and termination of the calls.

3.1 Minisip

Minisip is a SIP based client phone currently running on linux implements supplementary security features like Mutual Authentication, encryption and integrity of on going call. All these security functionality are work-in Progress under IETF standard (SRTP and MIKEY) [10].

4. RTP

The RTP protocol has significant properties of transport protocol it uses both unicast and multicast transport protocol [9]. RTP gives end-to-end network transport function appropriate for transferring real time data like audio, video over network services, the goal of RTP is to make easy delivery , monitoring, rebuilding, mixing and synchronization of data stream, it doesn't assure the quality of service for real-time services.

4.1 RTCP

The goal of RTP Control Protocol (RTCP) is to provide feedback on transmission quality, convey on-going RTP session's participants information,

5. QoS Parameters

VoIP Call quality over different Network Scenarios has been tested in this report and here is the definition of QoS terminologies which we measured in this research report.

5.1 Packet Loss

There are several reason for which Packet loss can be occur like Link failure, high levels of congestion that leads to buffer overflow in routers, connection loss over WLAN due to weak signal , in GPRS it might happen because of weak signal or handover.

5.2 Delay

User start noticing the delay when the transit delay for packets exceeds 100 milliseconds and if this delay go beyond 200 milliseconds then the user can notice the conversational difficulties due to the Breakdown in the usual conversational Protocol. Delay can also make echo problems.

5.3 Jitter

Transmit time of packet can vary because of network congestion, improper queuing, or configuration errors this variation in delay is called jitter or packet delay variation, this results in poor voice quality.

5.4 Mean Opinion Score (MOS)

The mean opinion score gives a numerical estimate on the human speech quality at the destination end of the circuit. An MOS score ranges from 1 for an unacceptable call to 5 for an excellent call. A typical range for Voice over IP would be from 3.5 to 4.2.

6. Resources

The test setup required a number of resources in order to achieve desired results.

6.1 Software

The software used in the research includes:

Debain Sarge and Red Hat 9 Linux OS
Microsoft Windows XP Professional
ClearSight Analyzer V4.1.1.18
Ethereal V0.10.11
MiniSIP V0.7 with LibMIKEY support
X-Lite SIP based softphone
MiKTeX V2.4.1461
Toshiba ConfigFree V4.00.06

6.2 Hardware

Sony Ericsson K700i phone with dual-end serial connectors.
10 MB multiport Hub.

6.2.1 Client A:
Compaq Presario 2100 with Intel Celeron 1.8GHz Processor, 256 MB RAM
Dual boot OS: Windows XP (Client A_win) and Red Hat 9 (Client A_rh)

Orinoco Silver WLAN card
NS PCI Fast Ethernet card
Stereo headphones

6.2.2 Client B:
Toshiba Satellite A45-S120 with Intel Celeron 2.6GHz Processor, 512 MB RAM.
Dual boot OS: Windows XP (Client B_win) and Debian Sarge (Client B_deb)
Orinoco Silver WLAN card
Intel Pro/100 Ethernet card
Stereo headphones

6.3 Monitoring

The main focus of the tests on either of the networks was based on SIP and RTP data. It is interesting to note that 'Client A' and 'Client B' also served as Protocol Analyzers for some calculations since the 'Analyzer' was sometimes unavailable or not necessary. The analysis tools used for the tests were ClearSight Analyzer and Ethereal.

6.3.1 Protocol Analyzer
The Analyzer was equipped with the following specifications.
OS: Windows XP
Protocol analyzer for SIP and RTP (ClearSight and Ethereal)
Orinoco Silver WLAN card
Fast Ethernet card

7. Testing

Since this study is based on actual measurements taken in real-time, a number of test scenarios were designed to calculate the performance parameters of VoIP calls over GPRS, wireless (802.11) and wired (802.3) networks. One of the first tests were performed with different softphones like X-Lite [7], Hotfoon [11], MiniSIP [10] etc. using the network infrastructure from WIDER3 at KTH [8]. Later, a setup over the internet was used in order to achieve real-time calculations.

The SIP based software client used for making calls to and from the laptop machines in the GPRS test scenario was X-Lite which is a product of Xten networks [7]. The software is quite simple to use and allows features like multiple simultaneous calls, call waiting, support for NAT etc. X-Lite can work with five different speech codecs namely G711u, G711a, GSM, iLBC and SPX. Out of these, G711u and GSM were selected for testing. G711u is the standard format for digital voice delivery in PSTN. It uses 8 KHz sampling rate and 64Kbps audio encoding while GSM uses an 8 KHz sampling rate with 13Kbps encoding [6]. This section provides information about the scenario and the various tests performed on the above mentioned networks.

8. GPRS Network

8.1 GPRS Test Scenario

The local Monaco base station at KTH was down due to unavoidable problems, therefore, GPRS subscription was taken from Comviq/Tele2 on private GSM connections for testing VoIP calls over GPRS. In the scenario with GPRS, Sony Ericsson's K700i modem driver was installed on 'Client A'. A dual-end serial cable was used to connect the phone with the laptop computer (Client A). A dial-up modem connection was created on the client machine which dials the external GPRS connection via the K700i interface. Due to expensive connection charges and the unavailability of another GPRS enabled GSM phone, it was decided to keep the other client over the Ethernet. This also provided the use of a hub in the middle that would allow sniffing the Ethernet packets on the fly in a convenient manner.

The SIP phone used for testing was X-Lite since the modem drivers were only available for Windows platform. Therefore, Minisip was not used in the tests with GPRS. Figure 1 demonstrates the network design for GPRS test scenario.

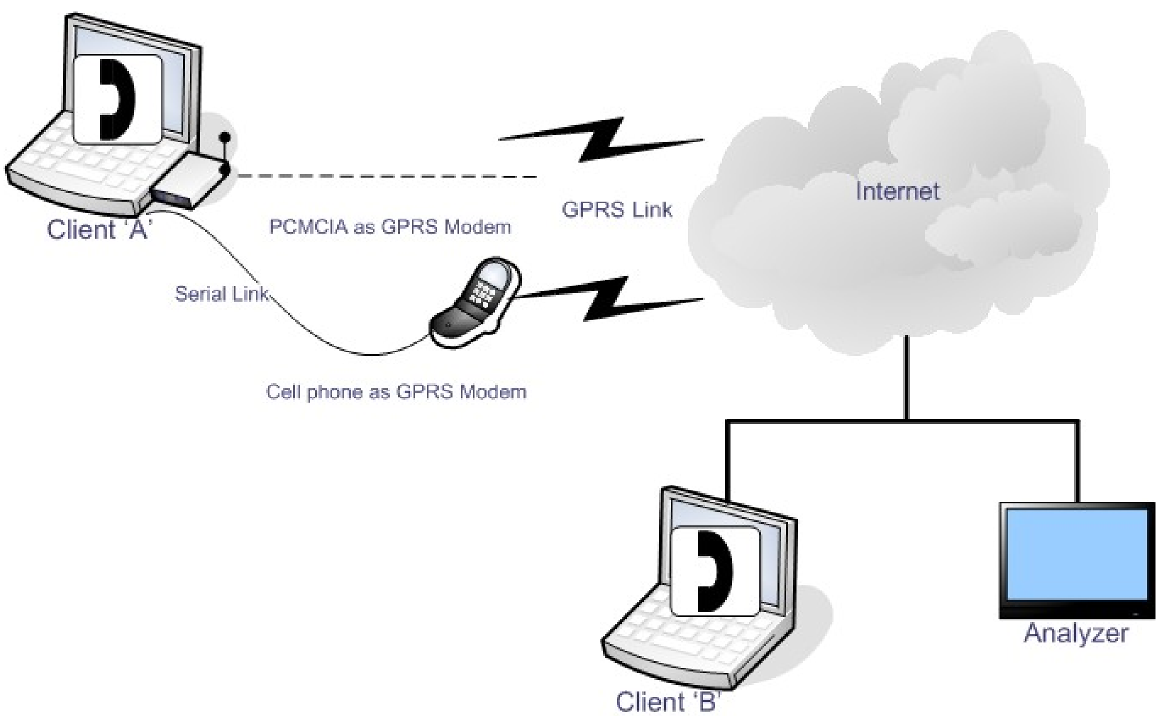


Figure 1: Network Design for GPRS Test Scenario

8.1.1 Analysis for G711u

Figure 2 demonstrates the results from the VoIP call made by 'Client B' to 'Client A' using X-Lite. The codec used in this test was G711u. Client B was connected to the

Internet via Ethernet while Client A uses the serial GPRS link to connect to the Internet. Client B was also the Analyzer in this scenario.

VoIP QoS Index For GPRS with G711u Codec		
Date: 05/21/2005	Location: Forum 6th Floor	
Codec: G711u	Start time : 17:01:33	Call Duration: 131.252(sec)
Parameters.	Client 'B' IP Address: 130.237.214.83 MAC: 00:08:0D:DB:A6:CA	Client 'A' IP Address: 213.101.155.226 SonyEricsson K700i GPRS
Packets Sent	6564	1623
Packet Loss	0	4744
Packet Loss Rate	0.00%	292.30%
Out of Sequence Packets	0.00%	26.43%
Mean Jitter (ms)	14.190	33.226
Mean MOS	4.22*	3.81*
Mean R-value	85.88*	74.91*

Figure 2: Results for VoIP over GPRS using G711u codec

It is obvious that the results achieved from the 131.252 seconds call involves high packet losses at Client A. The out of sequence packets and mean jitter is also higher than the acceptable ranges while Mean MOS and Mean R-value is also lower than desired value. The analysis shows that the bandwidth over the GPRS connection is either insufficient or there is a lot of delay in transmission (e.g. interleaving [4] etc.). Detailed graphs of the different criteria are shown below.

Packet Loss Report

	Packets 130.237.214.83 <-> 213.101.155.226		Packet Loss 130.237.214.83 <-> 213.101.155.226		Packet Loss Rate 130.237.214.83 <-> 213.101.155.226	
Flow 1	6564	1623	0	4744	0.00%	292.30%

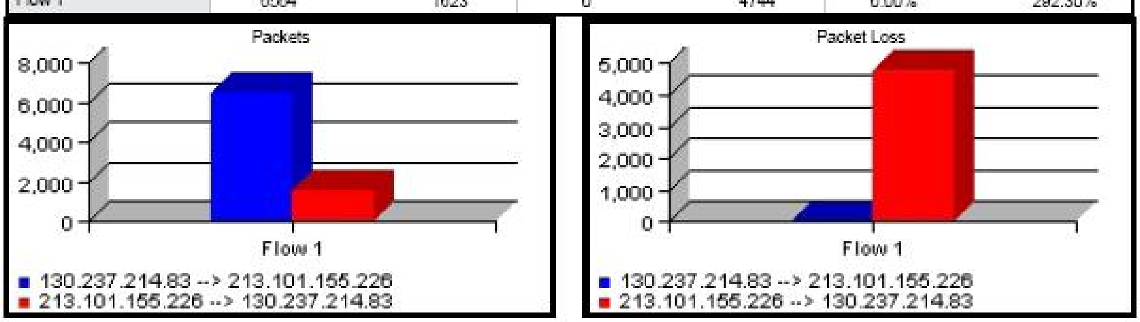


Figure 3: Packet Loss in GPRS on G711u codec

Jitter Analysis

	130.237.214.83 --> 213.101.155.226	213.101.155.226 --> 130.237.214.83
Flow 1(ms)	14.279	67.998

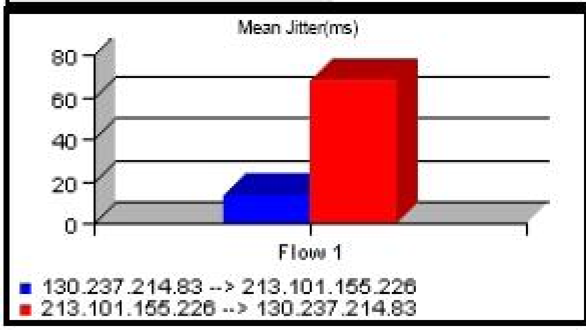
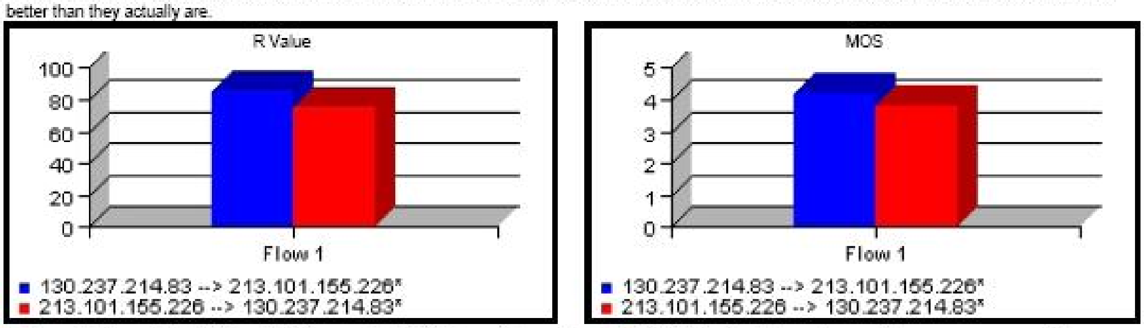


Figure 4: Jitter Analysis in GPRS on G711u codec

R Value and MOS

	130.237.214.83 --> 213.101.155.226	213.101.155.226 --> 130.237.214.83
Flow 1 R Value	85.88*	75.03*
Flow 1 MOS	4.22*	3.82*

* - The calculations for the MOS and R-Values marked with this symbol are missing the RTCP value for latency. As a result, scores may appear higher or better than they actually are.



* - The calculations for the MOS and R-Values marked with this symbol are missing the RTCP value for latency. As a result, scores may appear higher or better than they actually are.

Figure 5: R-value and MOS in GPRS on G711u codec

8.1.2 Analysis for GSM

The second test uses the same setup but uses a different codec called GSM. The results from the test are presented in Figure 6.

VoIP QoS Index For GPRS with GSM Codec		
Date: 05/21/2005	Location: Forum 6th Floor	
Codec: GSM	Start time : 17:10:04	Call Duration: 51.612(sec)
Parameters.	Client 'B' IP Address: 130.237.214.83 MAC: 00:08:0D:DB:A6:CA	Client 'A' IP Address: 213.101.155.226 SonyEricsson K700i GPRS
Packets Sent	2581	1537
Packet Loss	0	925
Packet Loss Rate	0.00%	60.18%
Out of Sequence Packets	0.00%	20.56%
Mean Jitter (ms)	14.225	15.103
Mean MOS	4.31*	4.30*
Mean R-value	88.97*	88.70*

Figure 6: Results for VoIP over GPRS using GSM codec

The measurements show that the performance parameters are better as compared with G711u. However, the results are still not very good. There was a lot of lag in the conversation and sometimes it was not understandable. The packet loss is still quite high and the MOS and R-values are also lower than required. The graphs for these parameters are displayed below.

Packet Loss Report

	Packets		Packet Loss		Packet Loss Rate	
	130.237.214.83 <->	213.101.155.226	130.237.214.83 <->	213.101.155.226	130.237.214.83 <->	213.101.155.226
Flow 1	2581	1537	0	925	0.00%	60.18%

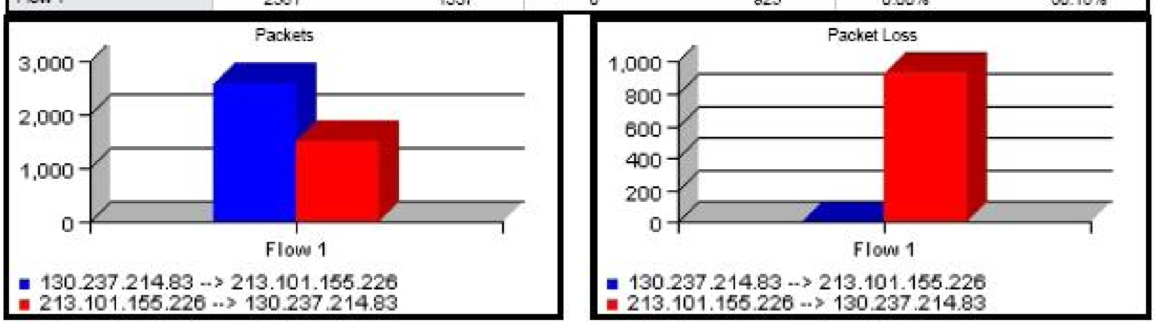


Figure 7: Packet Loss in GPRS on GSM codec

Jitter Analysis

	130.237.214.83 --> 213.101.155.226	213.101.155.226 --> 130.237.214.83
Flow 1(ms)	14.225	15.103

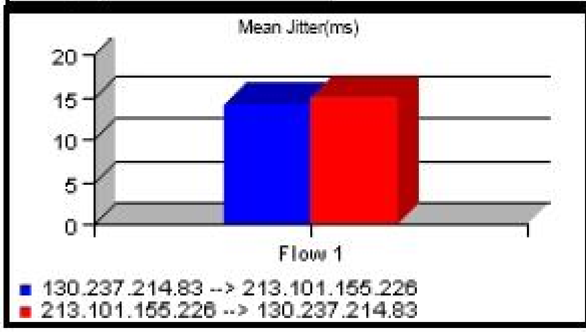
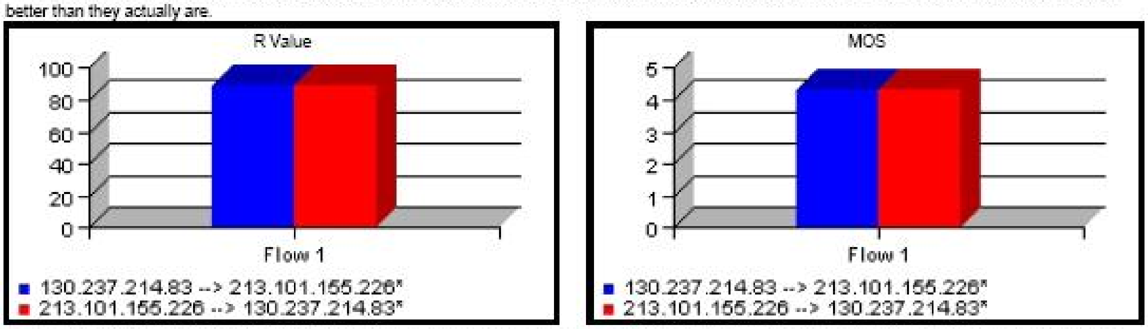


Figure 8: Jitter Analysis in GPRS on GSM codec

R Value and MOS

	130.237.214.83 --> 213.101.155.226	213.101.155.226 --> 130.237.214.83
Flow 1 R Value	88.97*	88.7*
Flow 1 MOS	4.31*	4.3*

* - The calculations for the MOS and R-Values marked with this symbol are missing the RTCP value for latency. As a result, scores may appear higher or better than they actually are.



* - The calculations for the MOS and R-Values marked with this symbol are missing the RTCP value for latency. As a result, scores may appear higher or better than they actually are.

Figure 9: R-value and MOS in GPRS on GSM codec

9. Wireless Network

StockholmOpen.net's [14] Public Access Points which is an Operator Neutral network in the Forum building of IT-University in Kista, Stockholm was used to perform the desired tests over wireless network. The available network equipment is based on 802.11b which was selected as a testbed for wireless.

9.1 Wireless (802.11) Test Scenario

The QoS measurements for VoIP on wireless networks are achieved through PCMCIA Wireless cards connected to 'Client A', 'Client B' and the 'Analyzer'. These Clients connected to the Internet through StockholmOpen in the Forum building of IT-University, Kista. One of the access points was configured privately which connected

'Client A' and 'Analyzer'. The other 'Client B' was used to connect to different access points in the building and also served as the roaming node. Figure 10 demonstrates the wireless scenario.

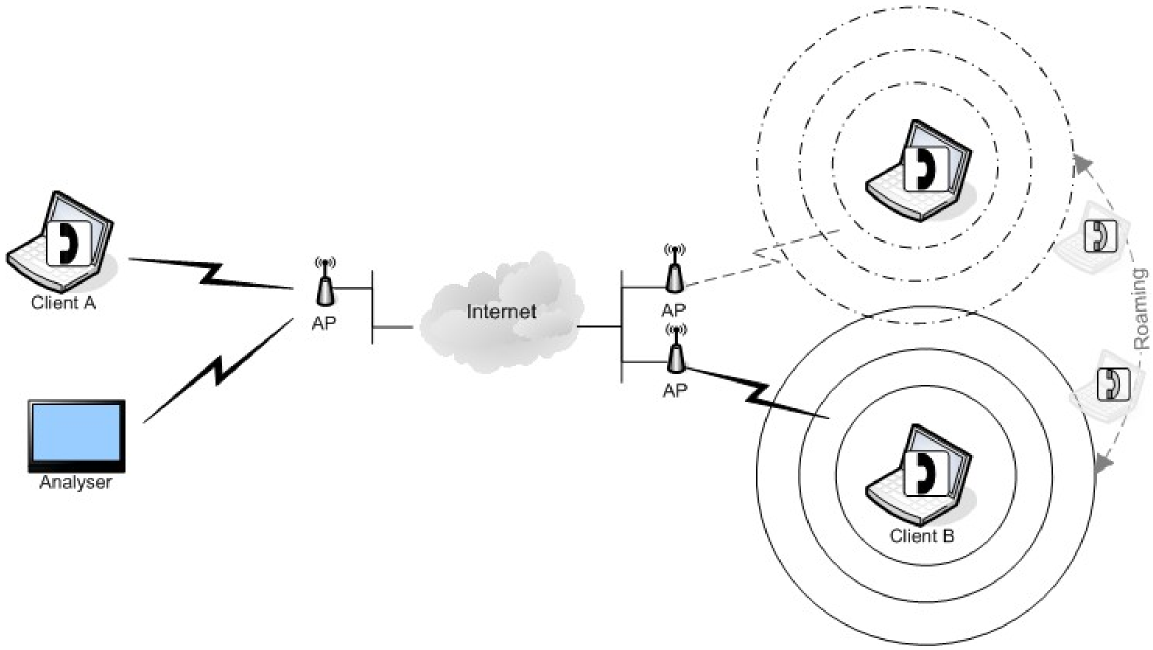


Figure 10: Network Design for Wireless Test Scenario

9.1.1 QoS Index over Single Access Point

Measurements were taken both for the secure and insecure SIP clients when both Client A and Client B connected to a single Access point in the network. Client A served as the SIP caller as well as the Analyzer for the current session. A brief description of each setup follows:

Insecure SIP client: X-Lite

X-Lite is a simple and easy to use SIP client which is used for this test to measure the performance of VoIP calls over wireless network. SIP accounts were created at iptel.org [5] namely sip:whaque@iptel.org and sip:rizkhan@iptel.org. The Clients connected to the SIP proxy at iptel.org. The calls were made from both ends alternatively to confirm the results.

Client A: sip:whaque@iptel.org

Client B: sip:rizkhan@iptel.org

SIP Proxy: iptel.org:5060

The results are demonstrated in the Figure 11.

VoIP QoS Index For WLAN Over Single AP		
Date: 05/18/2005	Location: Forum 8th Floor (Working Area)	
Codec: G711u	Start time : 21:26:05	Call Duration: 308.925(sec)
Parameters.	Client 'A' IP Address: 130.237.251.182 MAC: 00:02:2D:1D:67:06	Client 'B' IP Address: 130.237.251.141 MAC: 00:02:2D:00:60:6D
Packets Sent	28137	30064
Packet Loss	649	267
Packet Loss Rate	2.31%	0.89%
Out of Sequence Packets	16.99%	3.57%
Mean Jitter (ms)	11.785	2.094
Mean MOS	4.22*	4.22*
Mean R-value	85.96*	85.88*
Min Response time(sec)	0.0000	0.000275
Max Response time(sec)	0.0000	0.037428
Average Response time (sec)	0.0000	0.007567

Figure 11: Results for VoIP over Wireless using G711u codec

The chart shows that packet loss and jitter is more on Client A but the MOS and R-values are within acceptable range and therefore the sound quality was good although there was some lag in the conversation. Below follows the graphs for some of the VoIP performance parameters.

Packet Loss Report

	Packets		Packet Loss		Packet Loss Rate	
	130.237.251.182	<-> 130.237.251.141	130.237.251.182	<-> 130.237.251.141	130.237.251.182	<-> 130.237.251.141
Flow 1	28137	30064	649	267	2.31%	0.89%

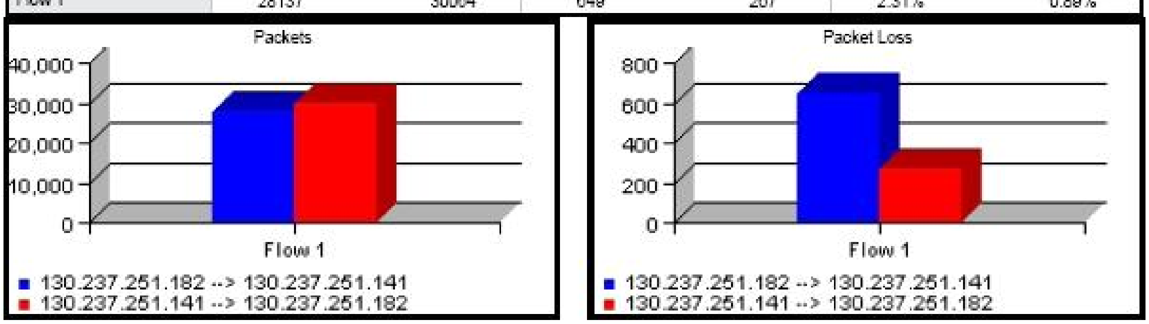


Figure 12: Packet Loss in Wireless on G711u codec

Jitter Analysis

	130.237.251.182 --> 130.237.251.141	130.237.251.141 --> 130.237.251.182
Flow 1(ms)	11.785	2.094

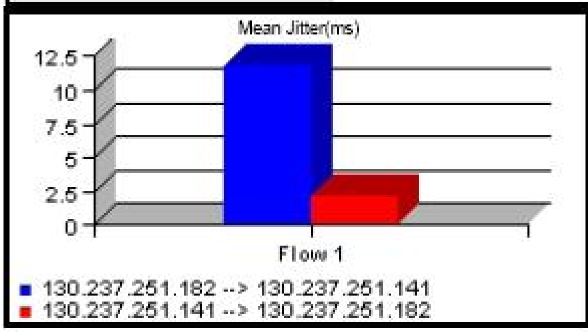
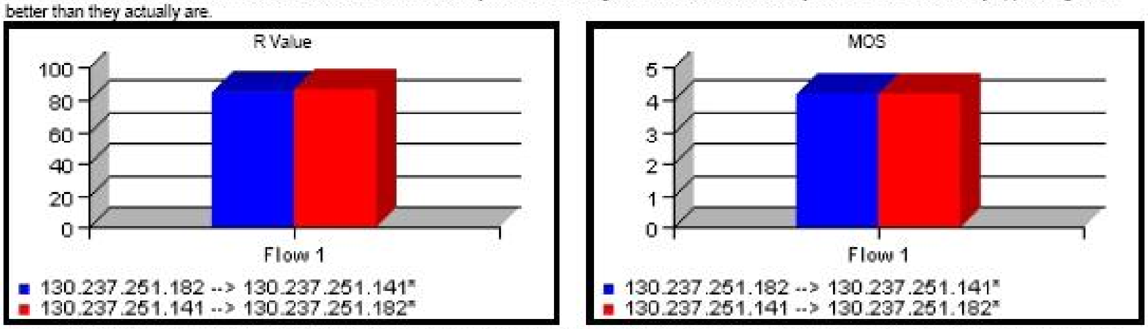


Figure 13: Jitter Analysis in Wireless on G711u codec

R Value and MOS

	130.237.251.182 --> 130.237.251.141	130.237.251.141 --> 130.237.251.182
Flow 1 R Value	85.76*	85.94*
Flow 1 MOS	4.22*	4.22*

* - The calculations for the MOS and R-Values marked with this symbol are missing the RTCP value for latency. As a result, scores may appear higher or better than they actually are.



* - The calculations for the MOS and R-Values marked with this symbol are missing the RTCP value for latency. As a result, scores may appear higher or better than they actually are.

Figure 14: R-value and MOS in Wireless on G711u codec

Secure SIP Client: Minisip

Minisip is a secure SIP client designed at KTH. 'Client A' and 'Client B' were running Minisip and the SIP URI used for testing the calls were sip:whaque@ssvl.kth.se and sip:rizkhan@ssvl.kth.se. Pre-shared keys were configured at both ends to make secure calls. The Clients were connecting to the SIP proxy at KTH (sip.ssvl.kth.se). The Client B served as the analyzer which eavesdropped the communication between the two Clients.

Client A: sip:whaque@ssvl.kth.se

Client B: sip:rizkhan@ssvl.kth.se

SIP Proxy: sip.ssvl.kth.se

VoIP QoS Index For MiniSIP		
Date: 05/20/2005	Location: Forum 6th Floor	
Codec: G711u	Start time : 21:26:05	Call Duration: 308.925(sec)
Parameters.	Client ‘B’ IP Address: 130.237.214.85 MAC: 00:08:0D:DB:A6:CA	Client ‘A’ IP Address: 130.237.251.214 MAC: 00:0B:CD:55:97:36
Packets Sent	13870	13864
Packet Loss	0	0
Packet Loss Rate	0.00%	0.00%
Out of Sequence Packets	0.00%	0.00%
Mean Jitter (ms)	0.011	2.052
Mean MOS	4.22*	4.22*
Mean R-value	85.91*	85.89*
Min Response time(sec)	0.0000	0.000393
Max Response time(sec)	0.0000	0.019770
Average Response time (sec)	0.0000	0.009335

Figure 15: Results for VoIP over Wireless using Minisip

It was observed that there was negligible amount of delay as compared to the unsecured SIP conversation. Only the length of the SIP INVITE message was increased a few bytes to accomodate the pre-shared key and the rest of the RTP communication was done in the same way as X-lite. This is a benefit with Minisip that it provides security at such a low cost.

9.1.2 Roaming between multiple access points

Measurements were taken to see the handoff times to switch between wireless access points. For this reason, Client A was stationary and was connected to an access point. Client B was the roaming node which moved between access points. The 2658.125 seconds call was made by Client B which also acted as the Analyzer for that session. The results are shown in Figure 16.

VoIP QoS Index For WLAN Over Multiple AP's		
Date: 05/18/2005	Location: Forum 6th & 8th Floor	
Codec: G711u	Start time : 20:08:15	Call Duration: 2658.125(sec)
Parameters.	Client 'B' IP Address: 130.237.251.141 MAC: 00:02:2D:00:60:6D	Client 'A' IP Address: 130.237.251.182 MAC: 00:02:2D:1D:67:06
Packets Sent	145776	98371
Packet Loss	785	543
Packet Loss Rate	0.538%	0.552%
Out of Sequence Packets	2.05%	1.62%
Mean Jitter (ms)	17.738	4.352
Mean MOS	4.22*	4.22*
Mean R-value	85.87*	85.94*
Min Response time(sec)	0.0000	0.000263
Max Response time(sec)	0.0000	0.051143
Average Response time (sec)	0.0000	0.010760

Figure 16: Results for VoIP over Wireless in roaming

Figure 17 shows the list of all the access points traversed by 'Client B' during the test alongwith their location and the switching times. It is interesting to see high values of packet loss and jitter which is due to the continuous switching between access points. The quality of the conversation was variable at different locations and times.

Roaming Over Multiple AP's				
No.	MAC Address	Channel (Freq:2.4GHz)	Switching Time (min)	Location
1	00:11:20:68:9E:10	11	0	Working Area 8th Floor
2	00:11:20:63:07:40	10	04:20	Kitchen Area 8th Floor
3	00:02:2D:2D:92:DE	7	08:56	7th Floor Stairs
4	00:07:85:B4:13:3F	13	10:30	7th Floor near Elevator B
5	00:02:2D:09:AB:FD	1	25:22	6th Floor Kitchen Area
6	00:0C:85:6E:FC:30	9	31:38	6th Floor Reading Room
7	00:02:2D:02:88:80	2	34:25	6th Floor Student Expedition
8	00:07:50:D6:59:B4	2	36:08	7th Floor Elevator A
9	00:02:2D:02:88:80	2	38:03	8th Floor Elevator A
10	00:02:2D:02:89:5B	1	39:50	8th Floor Working Area
11	00:11:20:68:9E:10	11	42:56	8th Floor Working Area

Figure 17: Roaming with various Access Points

10. Ethernet

The Ethernet in the Forum Building of IT-University (KTH) at Floor 6 was used as the testbed for measurements on 802.3 network. A 10 Mbps Hub was also used on the network to monitor the flow of traffic between the Clients.

10.1 Ethernet test scenario

The scenario for performance measurement of VoIP over wired network involved 'Client A' and 'Analyzer' attached to the internet via the 10Mbps hub in the StockholmOpen network. Similarly, 'Client B' connected via the Ethernet on the network at KTH as shown in Figure 18.

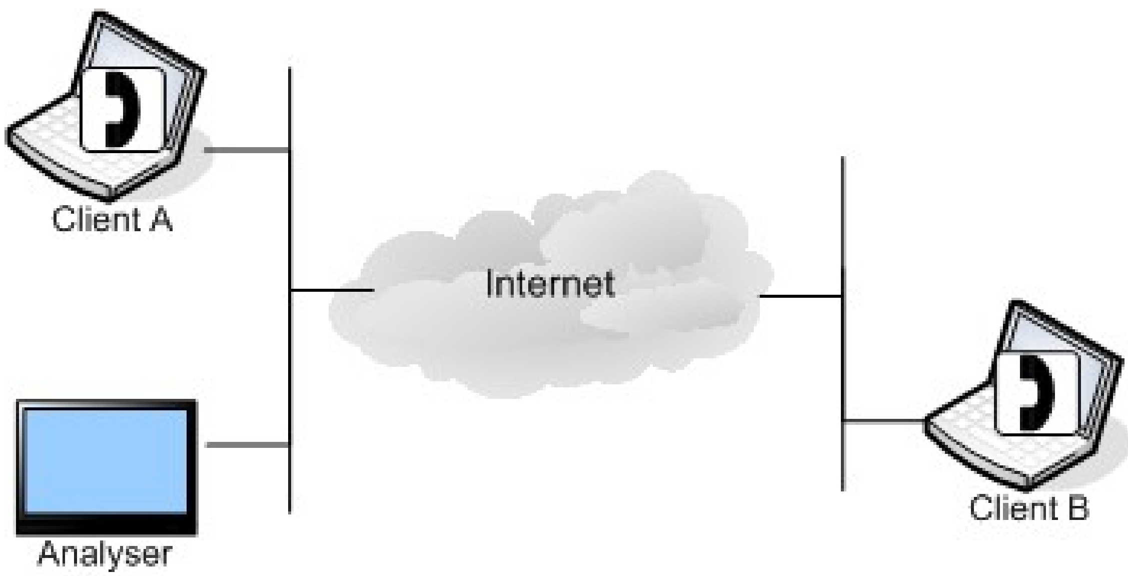


Figure 18: Network Design for Ethernet Test Scenario

'Client A' and 'Client B' were connected with their SIP accounts, sip:whaque@iptel.org and sip:rizkhan@iptel.org respectively, connected to the iptel SIP proxy (iptel.org:5060) to measure the performance of the VoIP calls.

Client A: sip:whaque@iptel.org
Client B: sip:rizkhan@iptel.org
SIP Proxy: iptel.org:5060
The measurements for the VoIP QoS index over Ethernet are shown in Figure 19.

VoIP QoS Index For Ethernet.		
Date: 05/18/2005	Location: Forum 6th Floor	
Codec: G711u	Start time : 22:53:15	Call Duration: 81.854(sec)
Parameters.	Client 'A' IP Address: 192.16.125.195 MAC: 00:08:0d:db:a6:ca	Client 'B' IP Address: 192.16.125.203 MAC: 00:0b:cd:55:97:36
Frames Sent	4080	3746
Packet Loss	0	0
Packet Loss Rate	0.00%	0.00%
Out of Sequence Packets	0.00%	0.00%
Mean Jitter (ms)	14.243	3.895
Mean MOS	4.22	4.22
Mean R-value	85.88	85.9
Min Response time(sec)	0.0000	0.005780
Max Response time(sec)	0.0000	0.036072
Average Response time (sec)	0.0000	0.021879

Figure 19: Results for VoIP over Ethernet on G711u codec

The call is considerably short however it is still interesting to see that there is no packet loss during communication since both clients are connected to the ethernet. The quality of the conversation was also better because the MOS and R-value is also within preferred range.

Packet Loss Report

	Packets		Packet Loss		Packet Loss Rate	
	192.16.125.195 <->	192.16.125.203	192.16.125.195 <->	192.16.125.203	192.16.125.195 <->	192.16.125.203
Flow 1	4080	3746	0	0	0.00%	0.00%

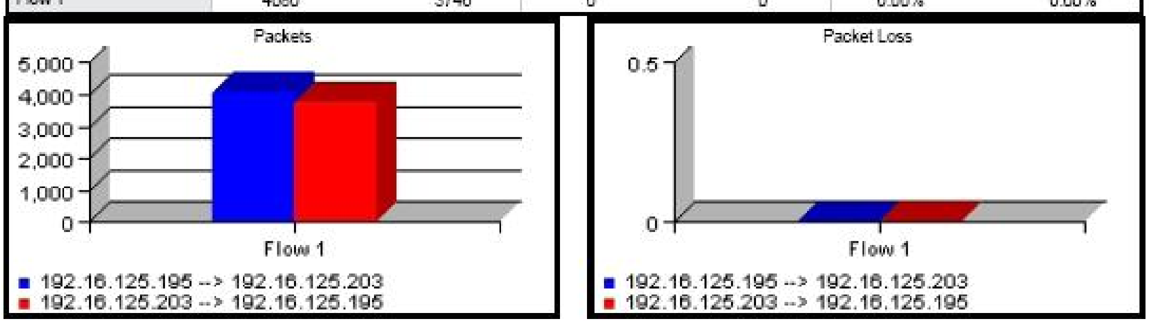


Figure 20: Packet Loss in Ethernet on G711u codec

Jitter Analysis

	192.16.125.195 --> 192.16.125.203	192.16.125.203 --> 192.16.125.195
Flow 1(ms)	14.243	3.895

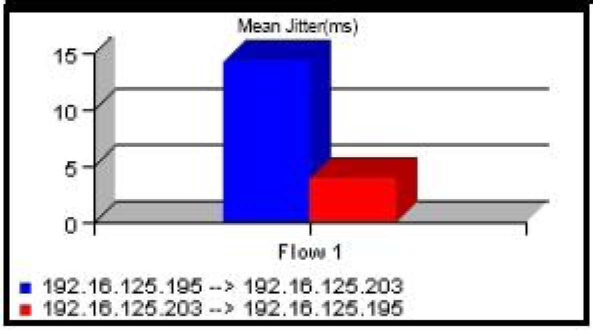
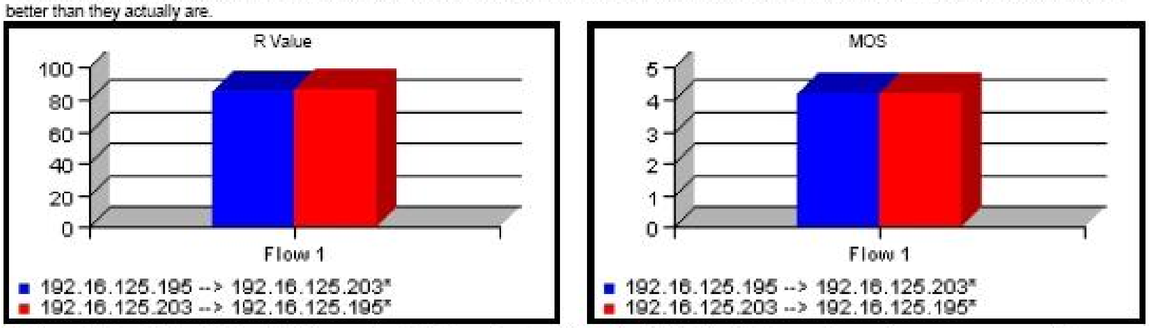


Figure 21: Jitter Analysis in Ethernet on G711u codec

R Value and MOS

	192.16.125.195 --> 192.16.125.203	192.16.125.203 --> 192.16.125.195
Flow 1 R Value	85.88*	85.9*
Flow 1 MOS	4.22*	4.22*

* - The calculations for the MOS and R-Values marked with this symbol are missing the RTCP value for latency. As a result, scores may appear higher or better than they actually are.



* - The calculations for the MOS and R-Values marked with this symbol are missing the RTCP value for latency. As a result, scores may appear higher or better than they actually are.

Figure 22: R-value and MOS in Ethernet on G711u codec

11. Observations

- GPRS connections are not available in various countries and those where it is available, it is still quite expensive. In Sweden and most countries in Europe, a four to five minute VoIP call over GPRS requires one Mb of data transmitted over the link and costs around 20 SEK (3USD approx.) at each end. In case two GPRS users talk on VoIP, it will require the same contribution from both ends (20x2=40SEK) which makes it quite expensive and still the quality is not comparable to a traditional long distance call or the SIP calls using PCs. Therefore, the charges for usage of GPRS should be reduced once it gets widely deployed.

- Another interesting observation was made during the Roaming tests on the Wireless network at KTH. Client A was connected to a single access point and was stationary while Client B was roaming between different floors of the university building to test the amount of delay introduced due to handoff between different access points in the building. At one point during this time, Client B could not re-connect to any of the APs for around 5 minutes and hence the local Kerberos authentication session at KTH was lost. However, as soon as Client B could find another Access point to connect, it was surprising to see that the SIP session was still established and RTP stream was still active. The most interesting part is that this time Client B needed to be re-authenticated to the Kerberos system and the browser asked for a login to connect to the network at KTH, but this SIP call was not concerned to this in any way and was still using the previous authentication key.

11.1 Problems Encountered

- The team had decided to perform GPRS based tests using a PCMCIA GPRS card that was requested from TSLab at KTH for tests over the GPRS network. However, the team could not get the card although it was expected to arrive in a few days but it never came.
- The test GSM base station at KTH, Monaco was down during the period of testing. Therefore, a private subscription from a telecom services provider called Comviq was bought. It was expensive and connected at variable speeds at different times. Moreover, it was also complicated to signup for the GPRS service since the interface was only available in Swedish language.
- During the critical testing phases, SIP proxy services from Iptel went out of function without any notice. The team wasted a considerable amount of time in trying to make the system work. Later, it was discovered that the problem source was not in our network but it was iptel itself.

12. Conclusion

This paper has discussed the different issues regarding QoS for VoIP calls over different network scenarios i.e. GPRS, WLAN and wired. Analysis of different measurements with the change of network scenarios has been made. It is observed that the QoS for voice (response time, packet loss, jitter etc.) over wired network is more dependable and reliable than other network scenarios especially the packet loss and sequence error on both clients is almost negligible.

In WLAN scenario, some response time delays with sequence error and packet loss is noticed during the call especially when one Client changes its location/Access point but surprisingly still the Call is not terminated. Amazingly, the SIP connection remains established even if Client B loses its authentication from the Network during roaming over WLAN.

A tremendous difference is noticed over GPRS connection as compared to other Network scenarios especially with the G711u codec in which there were great delays involved, high packet loss, large number of out of sequence packets and jitter value. By changing the codec G711u with GSM, considerable difference in voice quality (response time, packet loss and jitter) is seen.

Moreover, long delays were encountered in establishing the internet connection through GPRS modem but once it was established it stayed connected until it was disconnected manually.

12.1 Future Work

Some future directions from this work are listed here:

- Due to time constraints, the team could not complete all the tasks especially detailed testing. The paper could be expanded with more accurate results using multiple analyzing tools.
- There are a number of security issues especially the strength of MIKEY in Minisip and the effects of channel encryption on the communication that could be measured as well.
- A future work could be to measure the delays added due to the PPP serial link between the GPRS based phone and the PC and other similar links that could add more delay.
- It would be interesting to look at ways to optimize interleaving in GSM networks in such a way that could make GPRS communication faster.
- A future work could be to measure the handoff times during roaming in Wireless networks.
- During the tests with roaming in WLAN, the SIP connection was persistent even after the network authentication was lost. The reasons for this behaviour could be studied into more details.
- Only a couple of codecs could be tested for QoS at this time. There are more efficient codecs that require less bandwidth and are faster e.g. G.729 etc. These could be tested in more detail.
- Packet Loss can occur due to a number of reasons like link failure, congestion, buffer overflow in routers, ethernet problems, and the occasional misrouted packet. A future work could be to identify the source of packet loss by examining packet metrics available from switches and routers along the voice path.
- IP Multimedia Subsystem (IMS) is under development and aims to connect large amount of services to any platform. The tests could be performed over these services to measure QoS parameters in these services.

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List of Abbreviations

AP	Access Point
GPRS	General Packet Radio Service
GSM	Global System for Mobile Communication
IEEE	Institute of Electrical and Electronics Engineers
IETF	Internet Engineering Task Force
IMIT	Department of Microelectronics and Information Technology
IMS	IP Multimedia Subsystem
IP	Internet Protocol
KTH	Kungliga Tekniska Högskolan
LAN	Local Area Network
NAT	Network Address Translator
OS	Operating System
PC	Personal Computer
PSTN	Packet Switched Telephone Network
RAM	Random Access Memory
RFC	Request for Comments
RTP	Realtime Transport Protocol
RTCP	RTP Control Protocol
SDP	Session Description Protocol
SIM	Subscriber Identity Module
SIP	Session Initiation Protocol
TCP	Transmission Control Protocol
VoIP	Voice over Internet Protocol
WLAN	Wireless Local Area Network

List of Figures