Analyzing VoIP for 3G/4G Wireless Networks

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***Abstract*—An analysis has been done on the existing 3G wireless cellular network for VoIP based applications using tools to analyze protocols. QoS parameters like Delay, Jitter and Packet Loss have been given attention to. The results show that VoIP is still not ready to be a killer application as it plagued by excessive delays in some scenarios.**

***Index Terms*—VoIP, 3G,4G,WireSharks**

I. INTRODUCTION

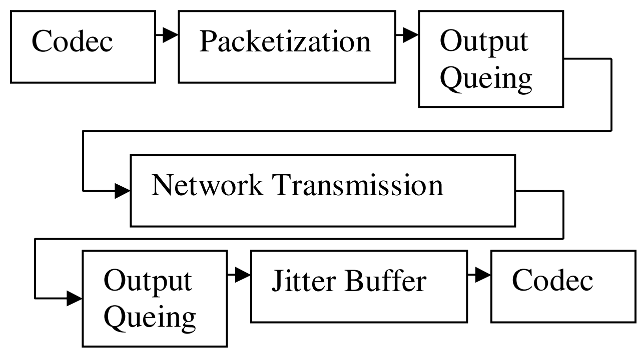
Trend towards VoIP

Change is the only constant they say and this statement is very true for wireless networks. New emerging trends always overshadow the old ones with new features and improvements. Deployment of 3G network services has been underway for a while and talks of 4G are already in the air. This new network architecture is supposed to provide features like high quality voice, high definition video and also features like global mobility support. With a data rate support of 50 Mbps and more the technology is attractive but there are several issues that need to be addressed before its commercial deployment.

Voice has always been the revenue generating feature of mobile phones but has traditionally been a circuit-switched technology. But the Next Generation Networks are going to be all-IP. Integrating all the services under the same IP based technology is one of the objectives of LTE (Long Term Evolution) referred to as 4G or B3G (Beyond Third Generation).This means transmitting all voice calls over IP. This shift has so far been considered positive because circuit switch based network does not offer the scalability provided by the IP based network. Also more features like video and text messaging can be integrated with voice in IP based architecture.

But in this process, special care has to be taken regarding voice traffic which has more strict requirements on delay and packet loss. ITU places a requirement of <150msec of one way delay for conversational voice, with a delay variation of <1% and a packet loss ratio (PLR) of <3%.

In calculating the delay budget for VoIP contribution from different segments and processes must be taken into account. The following figure will give an idea about the same.



The Figure above gives us an idea about the transmission flow from the sender to receiver with the network transmission consisting of uplink, backbone and then downlink transmission.

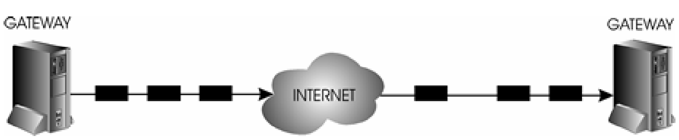
A brief overview of VoIP

VoIP converts voice signals into compressed data packets that can be sent over IP. The IP networks are best-effort networks and there is no guarantee of constant bit flow. In VoIP calls the digital speech stream must be transferred over network and packets should be arriving at constant speed at destination. There are delays and delay variations leading to jitter. There are packet losses too on the way to destination terminals. Overall the impairments affecting VoIP are summarized below:

Delay/Latency: This is the time taken for the speech to reach the receiver side from the sender. Each component in the transmission path as shown in the figure above- encoding, packetization, output queuing, packet transmission, input queuing, jitter buffer, decoding - adds delay. The delay increases with the number of router hops as well as the codec processing contributes to delay. Also greater the size of the jitter buffer ,more is the delay. However this parameter can be adjusted in accordance with network environment. The recommendation of ITU-T G.114 to achieve high quality of service is 300ms of round-trip delay.

As the jitter buffer size is an adjustable parameter, the delay caused by the network causes a deeper impact. Both one way and round trip delay can be measured by various means.

Jitter: It is the variation or regularity of packet inter-arrival time. It exists only in packet-based networks. This is because the delay inflicted by the network on each packet will be different. Unreasonable jitter makes speech unrecognizable. For high-quality voice, the average inter-arrival time at the receiver should be nearly equal to the inter packet gaps at the transmitter with a low standard of deviation. Jitter buffers are used to counteract this impairment of packet networks since the compression algorithms on the receiver terminal require the packets to have an equal spacing between them. The sequence number on the RTP packets are also used to re-sequence out of order packets



Packet spacing change leading to jitter

Packet Loss: Packet loss typically occurs either in bursts or periodically due to a consistently congested overloaded links, excessive collisions on a LAN and other errors. 5-10 % loss of all voice packets transmitted can deteriorate the voice quality.

With the above impairments in mind, if VoIP is to succeed as the next revenue generating application for mobile phone market, it has to overcome these challenges to make the package attractive to the consumer.

The most important standardization bodies working on VoIP are ITU-T, IETF, IMTC and ATM Forum. In addition, there are smaller organizations working on VoIP such as MIT Internet Telephony Consortium, Technical Advisory Committee etc. The H.323, an ITU standard and SIP, an IETF standard are the two signaling protocols for VoIP.

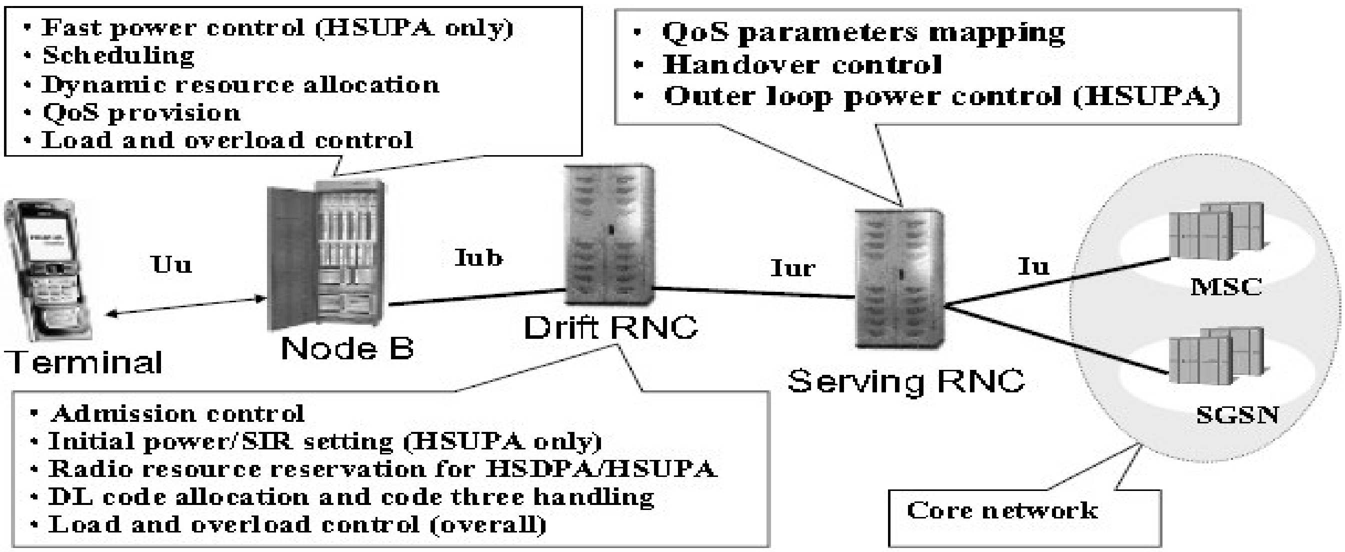
In the next session I will be giving a brief background on the efforts to deploy VoIP over 3G networks and its success so far .Then I will go on to present the efforts underway for VoIP over 4G networks which is interesting and challenging as well.

II. BACKGROUND

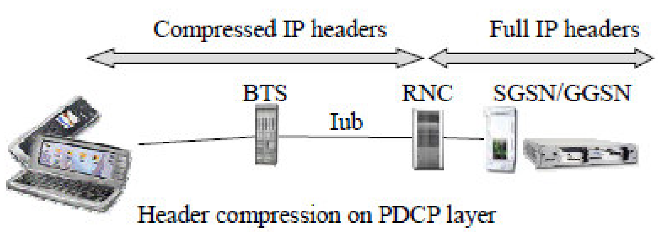
The road from VoIP over 3G to 4G

The 3rd Generation Partnership Project (3GPP), collaboration between groups of telecommunications associations in its Release 7 gave the specs for HSPA with a focus on decreasing latency, improvements to QoS and real-time applications such as VoIP. The introduction of 3G networks, including WCDMA Release 99, made it possible to run Voice-over-IP (VoIP) over cellular networks with reasonable quality, but with lower spectral efficiency than circuit switched voice. With the following releases namely Release 6 and 7 several improvements were made. The architecture for HSUPA and its influence on VoIP performance improvement is given briefly below:

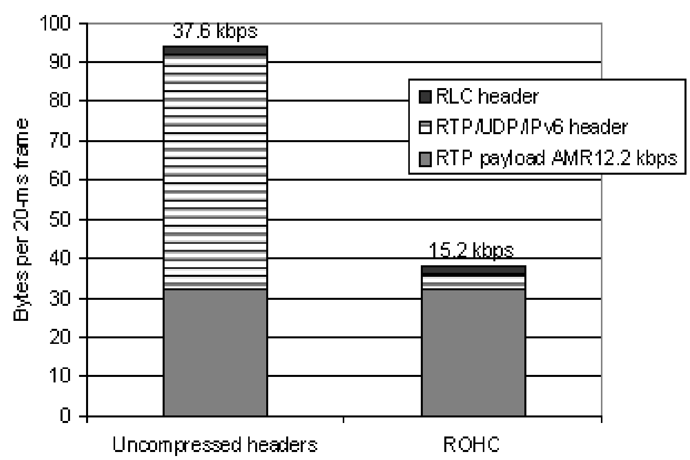
The RRM (Radio Resource Management) architecture has undergone changes with every 3GPP Release. For the HSDPA/HSUPA cellular network architecture the scheduling moved to Node B (3G term for BTS) from RNC (Radio Network Controller) in Release 6 .In the earlier release ie Release 99 the Node B(BTS) was mainly responsible for power control.Also in case there were two RNCs involved for a connection the scheduling was distributed between them .In Release 6 the SRNC (Serving RNC, the RNC connected to the core network) decides the QoS parameters as well as suitable handling of handovers. QoS management is taken care of by the Node B scheduler as well. As an example VoIP service can have a higher scheduling priority than any data service. Also the scheduler buffer can be controlled by a discard timer which indicates the maximum time packets should be kept in the buffer and then discard when the timer expires. The figure above gives us an idea of the features of Release 6 architecture. There is a new MAC layer MAC-hs added to the Node B to handle the scheduling. The terminal also has an additional MAC layer added to it namely MAC-es/s.



The HSDPA/HSUPA protocol architecture can be divided into the user plane part, handling user data, and the control plane part. The RRC layer in the control plane part handles all the signaling related to configuring the channels, mobility management. The Packet Data Convergence Protocol (PDCP) has as its main functionality header compression as mentioned before. It is done in the user equipment (UE) and in the radio network controller (RNC) therefore, it saves not only air interface capacity but also Iub transmission capacity.



This is because size of a full IPv6 header together with a Real Time Protocol/User Datagram Protocol (RTP/UDP) header is 60 bytes, while the size of a typical voice packet is 30 bytes. IP header compression allows using the transmission medium more efficiently for VoIP traffic over HSPA. There is a reduction in data rate from 40kbps to 16kbps using the ROHC (Robust Header Compression) approach.



Radio link control (RLC) handles the segmentation and retransmission for both the user and control data. It has three modes of operation, namely:

I. Transparent mode: In this mode no overhead is added to the RLC layer.

II. Unacknowledged mode: In this mode no RLC layer retransmission will take place. This is used with applications that can tolerate some packet loss, as is the case with VoIP, and cannot allow delay variation due to RLC level retransmission.

iii. Acknowledged mode operation, when data delivery is ensured with RLC layer retransmissions with applications that require all packets to be delivered.

The methods that have been used in HSDPA to improve the downlink packet data performance as well as capacity and bit rates are link adaptation, fast scheduling and physical layer retransmissions.

Moving towards 4G

As the third generation mobile systems are becoming commercialized, research focus has shifted towards 4G systems. The main transmission technology in 4G proposals such as evolutions of the 3GPP long term evolution (LTE) and IEEE 802.16 is orthogonal frequency division multiplexing (OFDM), which uses multiple carrier frequencies dedicated to a single data source.

Although not as well defines as 3G network

Supporting QoS in 4G networks will be challenging due to varying bit rates, channel characteristics, bandwidth allocation, fault-tolerance levels, and handoff support among heterogeneous wireless networks. Handoff delay is another important QoS-related issue in 4G wireless networks. The delay is more apparent in inter network handovers compared to intra network because of authentication procedures that require message exchange, multiple-database accesses, and negotiation-renegotiation due to a significant difference between needed and available QoS.

Handoff may lead to the user experiencing significant drop in QoS that will affect the performance of both upper-layer protocols and applications. Deploying a priority-based algorithm and using location-aware adaptive applications is said to reduce both handoff delay and QoS variability.

A SUMMARY OF STUDIES CONDUCTED ON VoIP SERVICE OVER HSDPA and LTE

Defining terms:
HSDSCH is the HSDPA transport channel and E-DCH is the HSUPA transport channel.

Summarizing simulation results with features in Release 7 HSDPA/HSUPA [2]:

Some of the features of this release were:

Uplink gating: The idea in the gating is to stop the transmission of the uplink control channel when there is no data to be sent on transport channel and no feedback signaling on physical control channel. The gating reduces the uplink interference levels, and therefore, increases the uplink capacity

Mobile equalizer: Reduction in intra-cell interference and improved capacity is another feature of Release 7 due to the introduction of a 2-antenna equalizer, which is a Release 7 enhancement.

Power Scheme: Discontinuous HSDPA reception for lower mobile power consumption

Packet bundling: Turbo code employed in HSDPA performs more effectively when up to VoIP packets are bundled together. Circuit switched voice which used convolution coding transmits 1 voice packet per 20-ms radio frame. The use of packet bundling further improves capacity.

Advanced Node-B HSDPA scheduler: HSDPA VoIP simulations assume proportional fair packet scheduling Code-multiplexing of users (M_{users}). The scheduler selects those M_{users} with highest priority from the scheduling candidate set for transmission in the next 2-ms transmission time interval (TTI).The criteria for the scheduling candidates is as follows:

1. Users that have a minimum M_{pkts} of VoIP packets buffered in the Node B. The value for M_{pkts} depends on the maximum allowed VoIP.

2. Users whose head-of-line packet delay is equal to or larger than (M_{pkts} -1) x 20 ms.

3. Users with pending retransmissions in their hybrid automatic repeat request (HARQ) manager.

These criteria help avoid scheduling for users with low amounts of buffered data in the Node B, which might cause a loss of system capacity. A VoIP packet with ROHC is roughly around 38 bytes or 304 bits while the HSDPA transport block size with, three high-speed downlink shared channel codes can be beyond 1500 bits. Therefore, a single transport block can carry multiple VoIP packets.

HSDPA/HSUPA Simulation Result Analysis:

- Gating improved the capacity to 30-40%.
- Use of the advanced receiver and the improved scheduling algorithm improved the capacity by 30 % in the experimental setup.

With the enhancements of Release 5, 6 and 7, the spectral efficiency of VoIP was improved compared to circuit switched Release 99 solution.

Summarizing simulation results with Release 8 (4G) features for VoIP performance [16]:

Application is what drives the mobile industry with each application having its own special requirement. Some are tolerant to delay and some to packet loss and some which are tolerant to neither, VoIP being one of them. One of the main contributors to delay in an IP based network is retransmissions. In this discussion I will summarize some aspects of 4G networks that have been simulated and studied. The delay requirements for VoIP, as stated by ITU-T is, <150msec of one way delay for conversational voice. However the contribution to delay in a 4G or 3G cellular network traffic is by wireless link and fixed link as well. The fixed network delay cannot be controlled. The buffer at the receiver to combat jitter contributes to the increase in total delay before playback. The traditional method of using FEC (where retransmissions are not required) for voice traffic is not optimal to capacity and is more complicated as it required different ARQ/FEC for varying traffic classes.

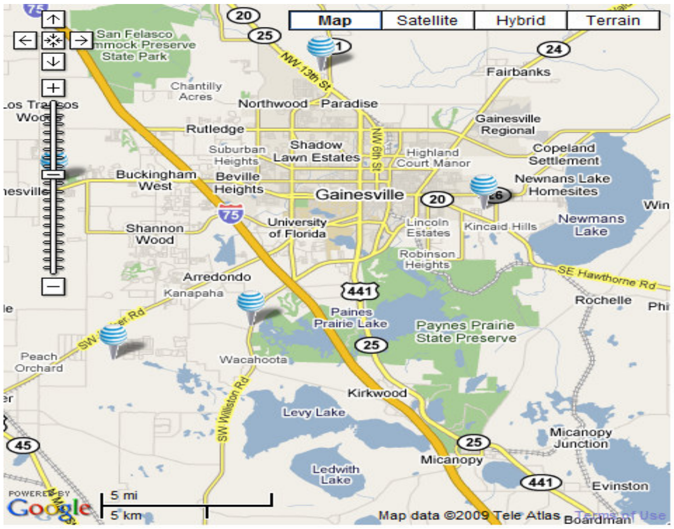
The system design in 3G cause high transmission delays, however 4G has a target of 8 ms HARQ RTT, as well as 2 ms HARQ RTT. The performance of VoIP applications can be measured in a number of ways, for example with regard to delay, jitter and packet loss. There are also perceptual models Mean Opinion Score (MOS), Perceptual Evaluation of Speech Quality (PESQ), and the E-Model which are based on the user experience. The simulation that will be summarized below has taken the following considerations:

1. In the downlink OFDMA coded transmissions divided in time and frequency was used: 1500x25 frames per second, over a 5 MHz channel in the 1900 MHz band. Each time-frequency slot, consists of 108 symbols. The symbols are modulated with 1 to 8 bits/symbol corresponding to uncoded BPSK to 256- QAM modulation. The modulation is adaptive so the data rate varies between limits.
2. Frame retransmissions are used till a limit is reached.
3. Queuing is not considered as transmissions occur below link capacity.

In this setup the traffic was transmitted from a fixed sender to a mobile receiver. This gave a packet stream of 50 packets/s with 172 byte payload. The traffic was generated and captured with the simulation tool used in the experimental setup. Reliability and delay has a tradeoff where retransmissions for reliability lead to delay and limiting retransmissions reduce delay with a negative impact on reliability. It is observed that retransmission delay impacts the delay in more profound way. So short delay loops as promised by 4G will lead to reduced delay. A link retransmission delay of 2 ms resulted in a packet delay of 20 ms over the wireless link, and 8 ms link retransmission delay resulted in a packet delay of 50 ms. The highest link retransmission delay tested, 16 ms, resulted in an 80 ms upper delay bound which is a good margin for 150 ms end-to-end delay for voice over IP.

IV. EXPERIMENT

Most of the experiments and simulations are done with some limitations and considering optimal conditions. I have procured a 3.5G (marketed as 3G by AT &T) data card. The following map gives a coverage area and the cell tower locations of AT & T around Gainesville.



The specific tower locations were procured from AT & T site. I used VoIP raider and Xlite using the sip server (sip.gatorphone.com).The summary of the field tests conducted under different test conditions and the results are given below. Also the scenarios that could not be tested and the reasons thereof are also explained below.

Limitation: For this scenario due to the availability of one data card test could be performed with only one of the calling parties having 3G and the other being either 2G (Cellular Phone) or High Speed Cox Internet Connection delivering 50 Mbps of data rate.

Tool Used for the Analysis: WireShark Version 1.0.5, a free Network Protocol Analyzer distributed and released under the GNU General Public License.

Scenario # 1 : Both the callers are in a standing static position (With no motion)

Two laptops, Widows Vista Basic operating system with Xlite installed and configured to sip.gatorphone.com with phone numbers acquired from IPKall as well as another VoIP tool VoipRaider. One of the System had 3.5G card with typically 700 Kbps - 1.7 Mbps download / 500 Kbps - 1.2 Mbps uplink and the other with standard 40Mbps internet connection.

Between 3.5G and standard High Speed Internet (HSI) (Cox Connection):

Observations: It is observed that the drop by jitter buffer is more in the direction from the 3G to the HIS Connection user than the reverse direction. However the RTP packet loss is not very significant in both directions.

A print shot of the graph analysis done by Wireshark is shown below :

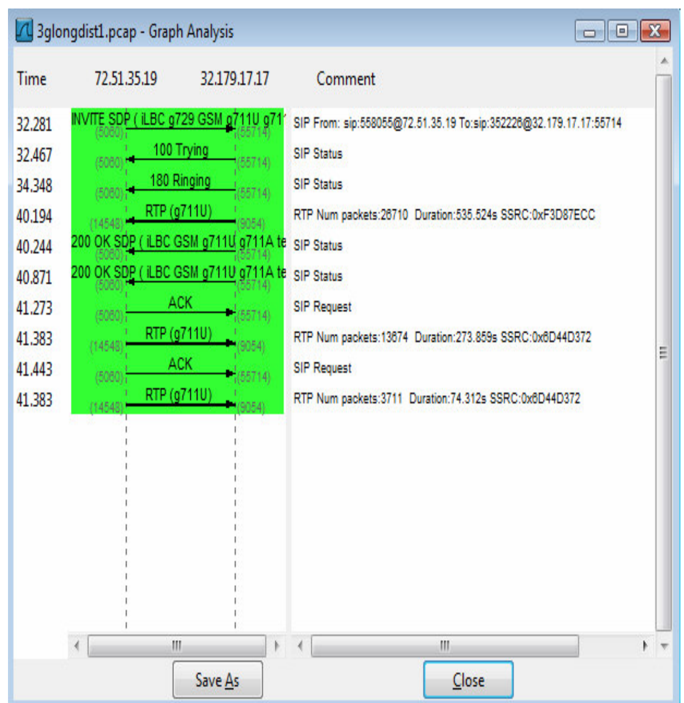


Fig 1

As we see in the Fig 1 above the following information is provided by the graph analysis:

Two columns representing the IP addresses of the two callers.

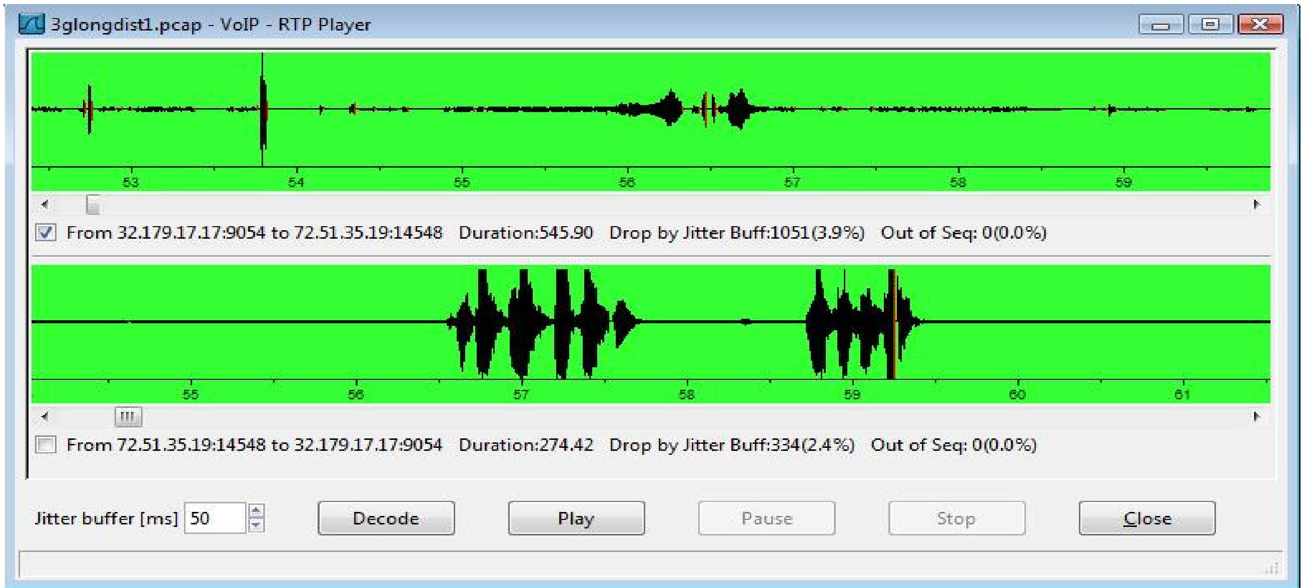


Fig 2

- An arrow showing the direction of each packet in the calls
- The label on top of the arrow shows message type. It shows the codecs available namely iLBC, g729, GSM, g711A, g711U and speex.
- Shows the UDP/TCP source and destination port per packet.
- Protocol dependent information is shown in the last column.

The first message is an INVITE message to start the call.After an exchange of few more messages the RTP stream is established between the two points through the Session Description Protocol (SDP) which carries the information about codecs, IP addresses and port numbers that is necessary for VoIP to work. RTP streams being unidirectional, a full duplex conversation is setup by setting up separate RTP streams in each direction using two separate SDP messages.

Fig 2 is a screenshot of the RTP Player. It shows the RTP streams available for the calls made.

The RTP packets that are dropped by jitter buffer as well as out of sequence packets are reported in the window. Between 3.5G and GSM 2G Phone:

Observations: It is observed here that RTP packet loss in both directions is more significant than the drop by jitter buffer.

The Graph display for this experiment is very similar to the one described above with the same sequence of events as described above for the test scenario. Fig 3 is a screenshot of the RTP screen analysis showing the out of sequence packets during the course of the 3G to 2G voice call.

An average data rate of 90Kbps was reported in WireShark for separate LAN scenario and data rate of 7 Kbps was reported between 3G and 2G call.

The table below gives a summary of results for Scenario #1

Scenario	Average Packet Loss	Maximum Jitter(ms)	Average Delay(ms)
VoIP using 3G to HIS VoIP separate LAN	0.00-0.06%	12	623(3G–HIS) 174 (Opp.)
VoIP using 3G to HIS VoIP same LAN	No packet loss	6.14	60(3G–HIS) 105 (Opp.)
VoIP using 3G and 2G	0.08-1.37%	23	56(3G–2G) 102(Opp.)

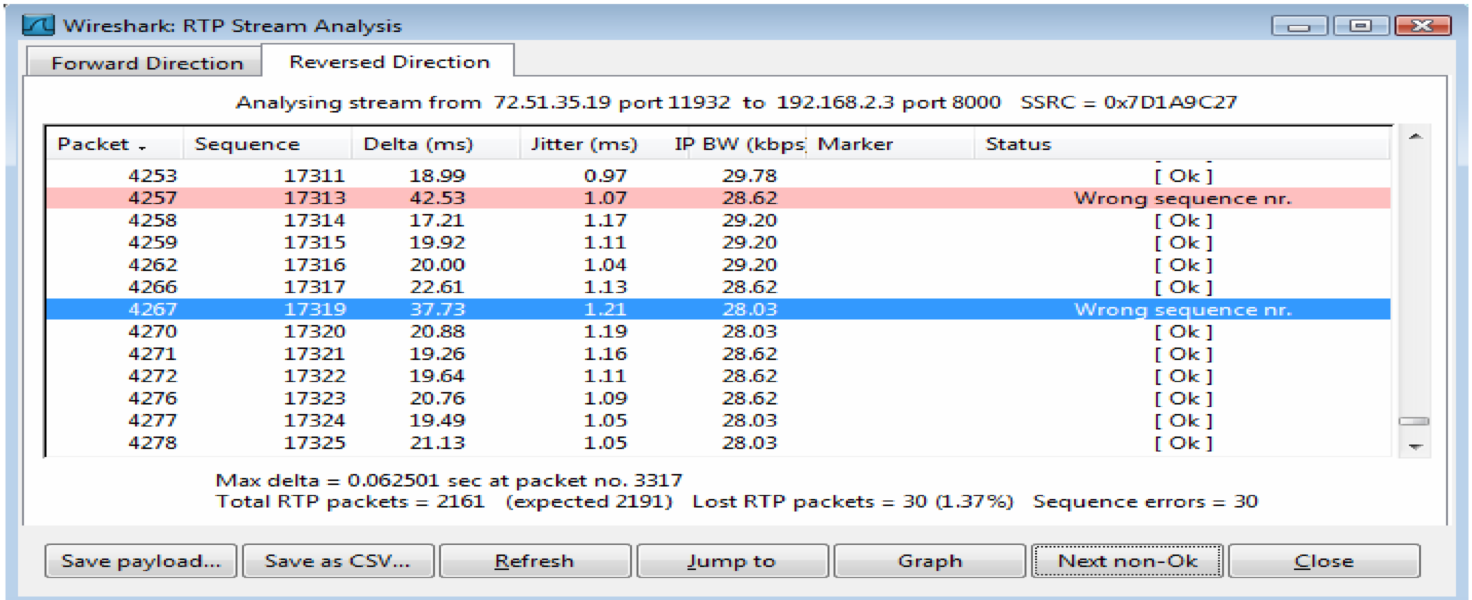


Fig 3

Scenario # 2: Caller with 3G card is in motion in the area spanning two cell towers while the second caller is static.

Between 3.5G and standard High Speed Internet (Cox Connection):

Observations: Packet loss, Delay as well as Jitter drastically increases. Also the codec that is used for voice changes from g711U in the previous scenario to iLBC (**internet Low Bit Rate Codec**). The bit rate supported by it is 15.2 kbps. It has a controlled response to packet loss and jitter. It was also observed that the packet delay from 3G to HIS was less than that from HIS to 3G. Using Xlite configured to sip.gatorphone.com caused more echo and packet loss compared to Voip Raider.

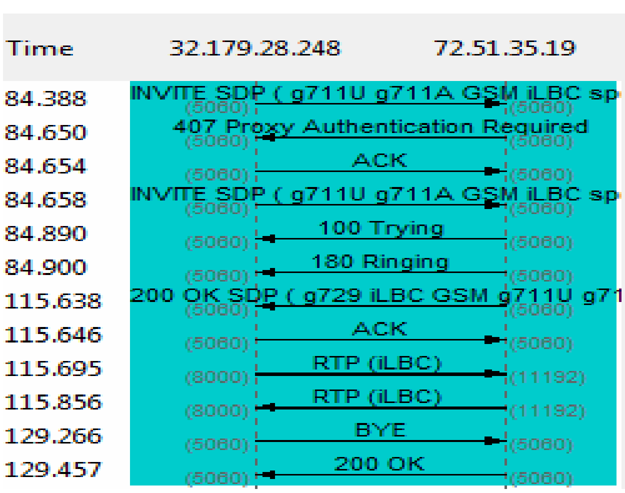


Fig. 4

In Fig 4 we see that the receiver asking for Proxy Authentication and iLBC codec being used for voice transfer.

Also it is observed that in extreme cases the delay has gone up to 12 seconds with 3.25 % packet loss as shown in Fig 5. In this case the call had to be reestablished for proper communication between the two VoIP clients. There were two instances of call drops in the course of calls due to heavy packet-loss and jitter. The average data rate reported was 37.2 Kbps.

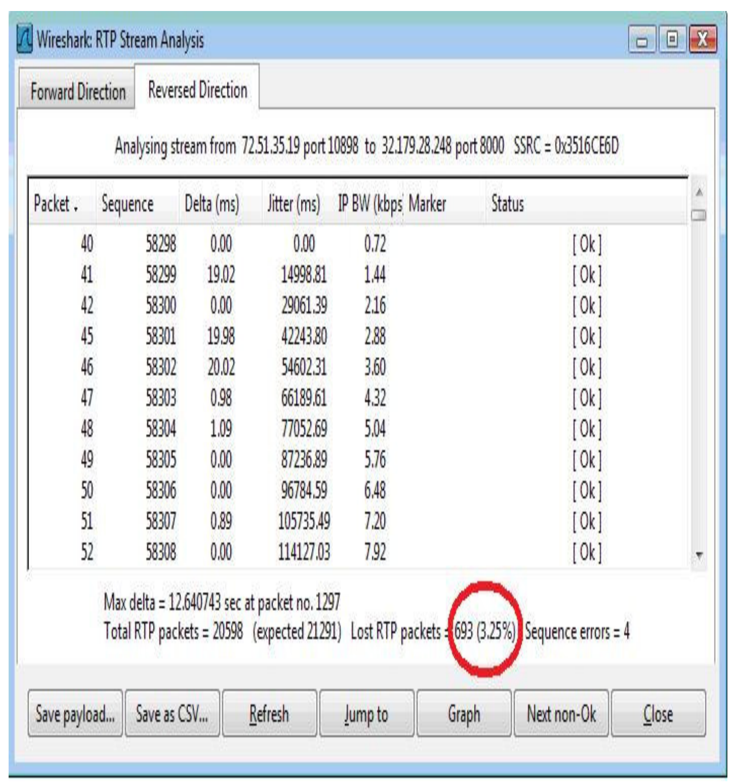


Fig.5

Also the drop by jitter varied from .7% to 15 % is both directions which contributed by the excessive packet loss due to the jitter buffer

Between 3.5G and GSM 2G Phone:

Observation: The results of this scenario are predictable and regulated without any erratic behavior. Even in this case the packet loss and delay in the direction from 3G system to 2G phone is more significant. Also the data rate reported in WireShark was 29 Kbps

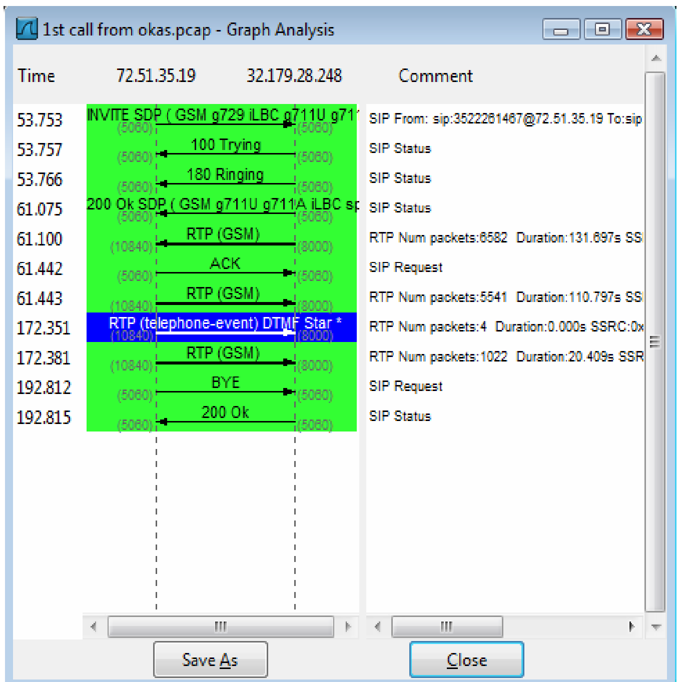


Fig 6

From the screenshot of graph analysis window above we see that of the various messages exchanged between the terminals the codec used is GSM which is the codec used by GSM phones. We also notice a DTMF (Dual Tone Multi Frequency) signal.

Dual-tone multi-frequency (DTMF) signaling is used for telecommunication signaling over analog telephone lines in the voice-frequency band between telephone handsets and other communications devices and the switching center.

In the table below are the results of Scenario #2:

Scenario	Average Packet Loss	Maximum Jitter (ms)	Average Delay(ms)
3G to HIS	0.00-.21%		99(3G-HIS) 564 (Opp.)
3G to 2G	0.00-0.10%	29	83(3G-HIS) 342(Opp.)

It is to be mentioned here that during the testing of scenario # 2, the laptop with a platform consisting of Intel Core II Duo and Vista Home Basic drained out of battery an hour earlier. In other words the backup lasted for 2 and ½ hours instead of 3 and ½.

V CONCLUSION AND FURTHER STUDY

From the results we see that delay in Part 1 of the first scenario namely between two separate LANs using 3G and HIS on the two sides and delay while the 3G card user is in motion far exceeds the ITU-T recommendations.

It is also to be noted that in situations where both the HSI and 3G were available, both the connections were active. So the operating system should be configured to relinquish the low speed connection when a higher speed connection is available.

Further study has to be done on 3G to 3G in both the scenarios as well as using 4G card as and when they are available.

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