

Performance issues of Voice over Wireless LAN (VoWLAN) and comparing it with Wired LAN

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Abstract—The emerging next generation networking environment presents an IP based core interconnecting many wireless radio access networks providing ubiquitous access to end users through a vast variety of wireless devices. VoIP over wireless LAN is freeing the telephone from cord is an obvious next step. The convergence of voice and data networks enables new applications and cost reduction. There are number of factors that inhibit wide spread adoption of VoWLAN. Data was reviewed on voice quality effects of excessive latency and jitter, degraded voice quality, poor convergence, interrupted voice service, roaming latency, security problem, retransmission and dropped packets, low capacity and reduced number of calls, voice and data coverage and power coverage environments. The subjective was to find out if VoIP works well enough in wireless LAN to be useful. For this comparison of the behavior of voice over IP in a 100MB/s Ethernet is done and quality of service parameters is measured in both environments. These days wireless environment is getting more and more important so the objective in concern of voice data was to discover if it is possible to do telephone calls in a wireless LAN with voice over IP with good enough speech quality

II. INTRODUCTION

H.323 is the ITU-T standard for real time multimedia communications and conferencing over packet based networks. These standards define how components that are built in compliance with H.323 setup calls, exchange (compressed) audio and video, participate in multiunit conferences and operate with non-h.323 endpoints. Evaluation phase had two focuses: QoS concerning voice data on one side and concerning signaling data on other side.

Concerning signaling data:-

With radio connection getting worse, there is a high probability of TCP calculating the call-setup-delay to extremely high levels; this is because of TCP not being adequate for wireless environments.

Call-setup signaling is the spine-spot of each call. It is possible that no connection can be set up, although an understandable conversation can be possible.

Once the call is established the ongoing-call-signal is extremely resistant against the pitfalls and threats that come in a wireless environment

Concerning voice data:-

If conditions are favorable, the voice data flow is not obstructed and a conversation is possible without problems.

With the radio connection getting worse, the speech quality first also gets a little worse but not annoying. But there is a certain limit beyond which conservation is not possible anymore.

H.323 OVERVIEW:

A signal call doing over a H.323 system can be divided into three stages:

1. Call Setup
 - dial signal
 - dialing
 - waiting for other party to hook the phone
 - the counterpart hooks off the phone
2. Ongoing Call
 - voice data is carried across the connection
3. Call Release
 - One part decides to finish the conversation
 - he/she hangs up
 - the other party gets a release

With in these different stages different things happen, on the signaling side as well as the voice data side.

Figure-1 below shows protocols used by H.323 compliant application. For me only the protocols in gray boxes are relevant. The voice data is transmitted over the protocol stack RTP/UDP/IP, the coding is G.711 and G.723. All other protocols are used for signaling.

Radio Connection Classes-There is a classification of quality of radio connection using classes given by Wave LAN driver software. These classifications are defined by the SNR of the radio connection and are displayed by the software in real-time. Class A is defined as a connection, which SNR is 20db or above, while Class B has SNR between 10 to 20 db and SNR of Class C is below 10db. The MAC packet loss and the distribution of how much packet is transmitted with a specific transmission rate are typical for each class... The smaller SNR of a connection, more the packet loss on the MAC layer. MAC layer and more packets are transmitted using a lower transmission rate.

III. SIGNALLING DATA

Step 1: *FILTERING*

To analyze the signaling behavior filtering of the relevant H.323 protocols packets used for signaling out the rest of IP-stream is done. This was by filtering of known ports. Next looked into these profiteered IP-packets and locate the H.323 PDU. This was done by searching for specific pattern within each IP-packet; the localized H.323 content was extracted from IP-packet and treated as H.323 packet.

Step 2: *DECODING*

Then extracted the relevant H.323-packets from IP-packets they contained, and putted that into a H.323 decoder to get its content. the decoded messages were needed to make necessary calculations and set-up and release-delay.H.323 decoder is provided by OpenH.323 project

Step 3: *SETUP AND RELEASE*

By knowing the packet messages it was able to reconstruct a typical call setup/release

IV. VOICE DATA

VoWLAN introduces a number of challenges that must be addressed for successful deployment.

(A) *Delays*: The delay is affecting the speech quality, but it is always present. It gets only longer or shorter depending on the radio connection quality available. But it never exceeds the range of about 500 ms- 1000 ms. The delays are not always pure, i.e. the time between a packet is sent and received, but it is delay of speech. This is a sum of lots of delays and some of them are network delay, queuing delay, speech encoding/decoding delay, transmission delay. You can see that VoIP in wireless LAN has by far the longest delay

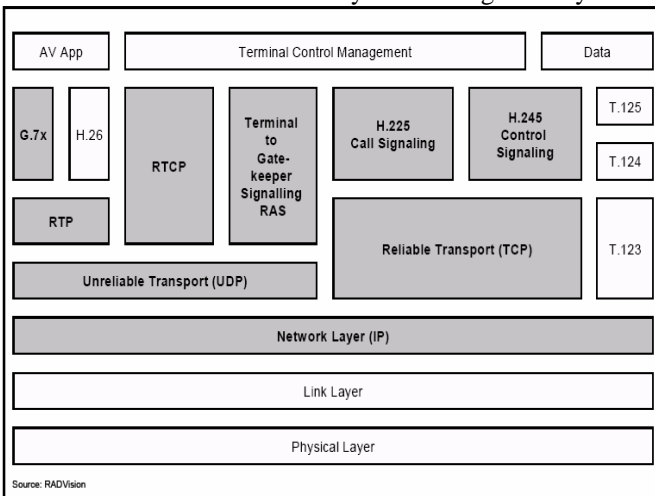


Figure-1 H.323 Format

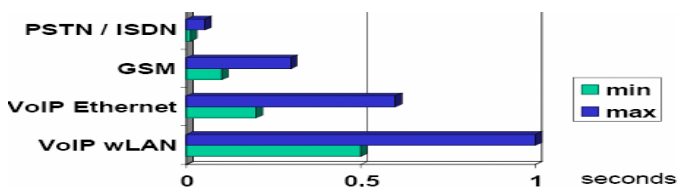


Figure 2 shows delay comparison

(B) *Excessive latency and jitter, degraded voice quality*: Loss and jitter occurs together and if they exceed certain

limits, they lead to interruptions in transmitted speech and if this interruption occurs too frequent, conversation become impossible. If the losses are small but frequent there will be a stuttering, which sounds like an echo. Figure 3 shows graphics of loss and jitter in one call.

(C) *Poor coverage, interrupted voice service*: To enable callers to move around freely, however their connection must be maintained without interruption; that is, in order to have a true mobility, coverage must be ubiquitous and continuous throughout. Cell planning technologies does not provide this. The limited number of channels results in limited overlap between cells, creating shadows or “black holes”. capacity for voice application is the number of packets per second that can be transmitted

(D) *Roaming Latency*: As mobile users move from cell to cell, inter-AP hands-off is required. As the cell size decreases to provide higher capacity, inter-AP hands-off occurs more frequently. A hands-off requires several steps, including AP discovery, re-association, security measures and higher-level protocol exchange. AP discovery typically takes 150-400 ms and introduce 40-100 ms of jitter in 802.11 b and 802.11 g, well in excess of acceptable voice latency. Re-association consists of several steps: station to AP handshake, inter-AP protocol (IAPP) exchanges and bridge notification

(E) *Security Problems*: Wired Equivalence Privacy (WEP) is the original security protocol for 802.11 networks but this is flawed. The interim WiFi protected access (WPA) includes temporal-key encryption (TKIP), message integrity access (MIC) and strong authentication.WPA provides adequate security mechanism and exchange of pair wise keys. This provides a latency of 500-1500 ms.

(F) *Retransmission and dropped packets*: Data applications do not tolerate dropped packets.TCP/IP retransmits erroneous or missing packets to avoid corruption.TCP/IP slow start result in degradation of throughput. Wireless mechanism has drop rate of 1% and above and above and result in degradation of throughput. Wireless medium is unreliable and therefore WLAN MAC includes a retransmission mechanism to avoid dropped packets. This is good solution but it produces jitter and latency. So to reduce retransmission, low drop rate is required.

(G) *Low capacity and reduced number of calls*: Capacity determines the number of concurrent calls that can be supported. Because of the VoIP small packet size and large overhead, the usual methods of measuring network capacity do not apply to voice. Several factors limit the PSS of a WLAN network, the most important of which are contention window,ACK packets, retransmissions and data adaption.The recently proposed wireless scheduled multimedia standard introduces contention free access based on 802.11 e’s hybrid coordination function replaces client competition for airtime with centralized client polling. By avoiding contention, contention free access methods utilize 75% of the maximum bandwidth. Contention free provides a higher PPS, resulting in double the number of calls.

(H) *Power consumption requirements*: Mobile phones must be

light and easily portable, which limits the size, poor and lifetime of the battery. To extend battery life mobile phones must take advantage of 802.11 power save mode when mobile is inactive. Power save mode increases the latency of setting up a call but does not affect the call. To extend the battery life in a active call, mobile user should be as close as possible to AP. Mobile users that are close to AP are able to transmit at highest data rates. Choices of transmission standard, OFDM or CCK also effect the power consumptions.

OFDM provides higher data rates, reducing transmission time and high data rates require extremely short distances from AP and consume more power during packet perception. CCK uses less power during reception of packets, is more power efficient and provides greater distances and supports low data rates as compared to OFDM.

V.COMPARISONS

Ethernet-Ethernet

No loss at all and the jitter is as little as a few nanoseconds. The speech quality is Very good – no interruptions, no stuttering. The delay is noticeable, but not annoying, if You don’t hear yourself speaking. The delay is noticeable, but not annoying, if You don’t hear yourself speaking *and* your own speech coming from the loudspeaker of Your call party. Figure 4 shows one loss and one jitter graph of a call.

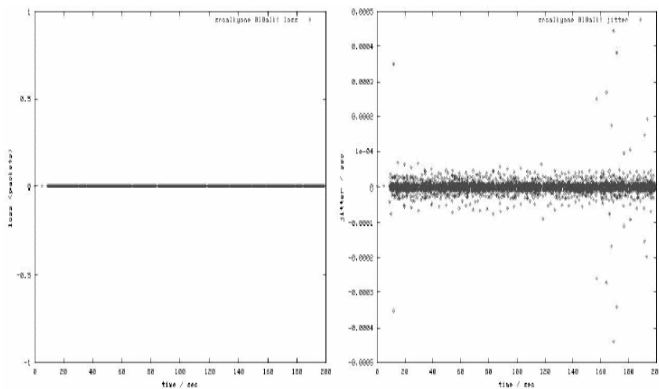


Figure 4 Typical loss and delay in scenario Ethernet ↔ Ethernet

Ethernet-Wireless

As long as the wireless host is either in an A or B class area, the speech quality is very Good, but if the wireless host enters the C class area, there will be partially long Interruptions. Even if the wireless host is located in the B class area, you can notice very short interruptions, but these don’t really disturb the conversation. Figure 5,6 shows the graph.

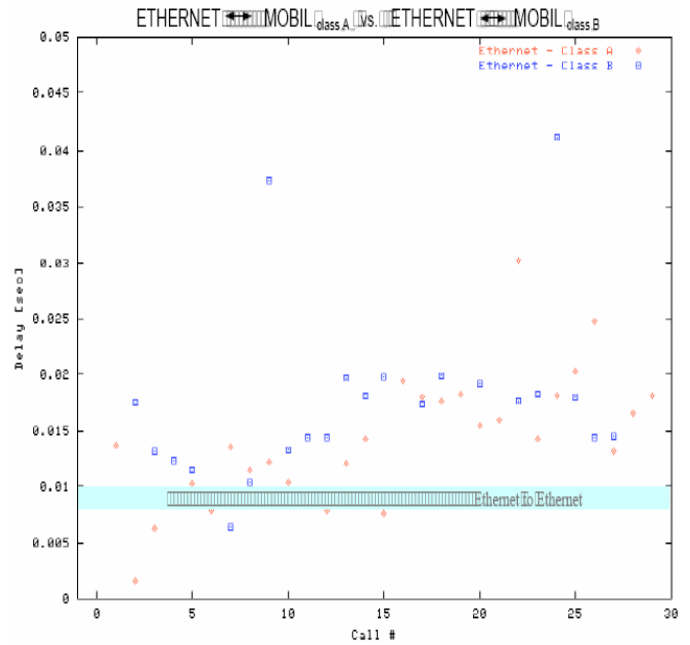


Figure-5

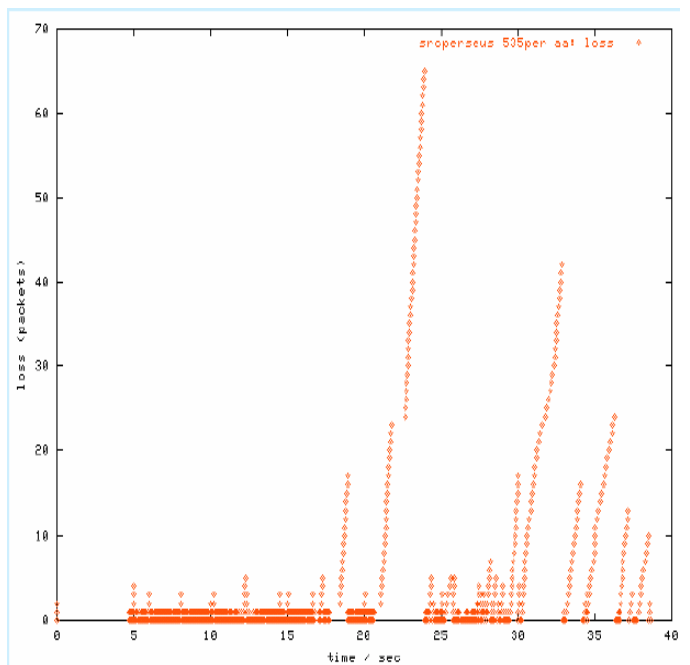


Figure 3 shows a Load and jitter in one call

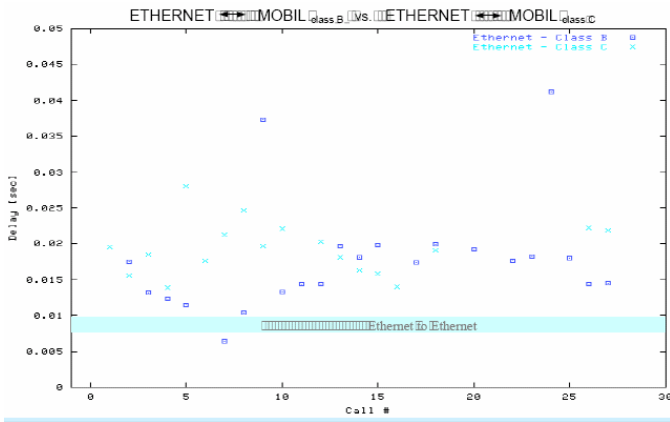


Figure - 6

Wireless-Wireless

If both wireless hosts are in the A class area, nothing special happens. The speech quality is good. Sometimes a little stuttering, but no serious disturbances. Figure 7 shows an example. Mostly only one packet at once lost, seldom more. When both hosts are in area B, good calls are hardly possible. The loss is with about 13% just around the limit. There are long phases of stuttering and sometimes long interruptions.

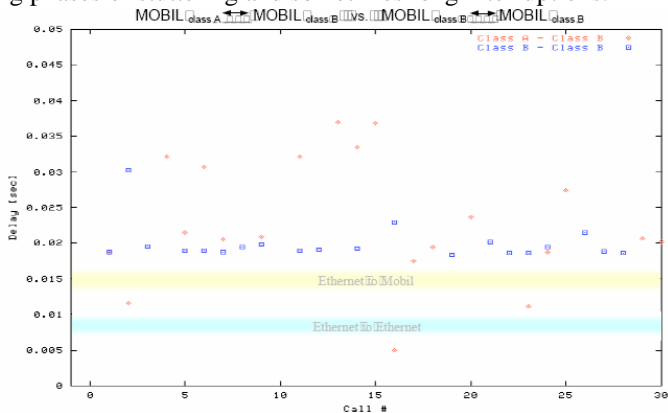


Figure-7

VI SUGGESTIONS FOR VoWLAN

Since signaling seems to be more vulnerable to bad radio conditions than the pure voice data, it should be subject to improvements first.

The question is simply how to be able to improve the used transport mechanism (TCP) or maybe use another way of adopting TCP to the conditions that come with a wireless connection. TCP is based on the fact that the throughput of a connection is rather determined by possible congestion than on the link-state quality (and therefore a higher error-probability). This is the case for WANs and wired LANs where TCP works great. This is because in WANs throughput will be limited by congestion and in an empty LAN by the maximum window size. But in both cases the error-rate will be near zero. But for a WLAN different conditions can be found. The probability for losing the first SYN packet is pretty high. This loss is not caused by congestion but simply by bad radio connection!

But TCP adapts to this by assuming congestion, which leads to a far too long timeout-value! Instead of waiting, it should retransmit almost immediately.

So the actual problem is: How to make TCP behave in a way that fits a WLAN conditions best?!

For this the basic solutions which can be implemented are:

1. Buffer parameters of closed TCP-connections to use for the next connection, within a certain time-period to a specific destination (buffered values for example: last Timeout, last congestion-window-size)
2. Use a completely different transport mechanism which is best-adapted to a WLAN environment (maybe leading to faster retransmissions, larger window-sizes, setting transfer rates actively)

3. Enable client mobility with zero latency roaming. Zero latency eliminates the latency and jitter associated with inter-AP hands-off.

4. Increase the maximum number of calls by allowing all stations to operate at the highest data rate. Ultra thin AP supports several channels; the number of calls is multiplied further. Provide blanket coverage with zero latency roaming, slack capacity is not required, allowing the full utilization of increased capacity. Blanket coverage and multiple AP's on the same channel greatly reduce the distance between the mobile stations and APs. So this allows a significant reduction in power consumption. Single channel blanket covers the static and dynamic load balancing. By deploying several blankets throughout, slower VoWLAN devices are placed on separate channel, freeing other channels for high-data rate applications.

VII. CONCLUSIONS

WLAN provides excellent opportunity to enable voice, since it combines the cost Effectiveness of VoIP solutions with cordless mobility. VoWLAN deployment have special needs in order to be effective. VoWLAN requires a strong uplink to reduce jitter and latency. It requires complete coverage and seamless mobility that allows strong security, without interrupting services with constant hands off. In addition increased capacity is needed to provide sufficient number of simultaneous voice calls. Users must always be close to an AP to cope with power constraints of VoWLAN devices, allowing them to transmit at highest data rates and avoid retransmissions, dropped packets or using lower transmission powers.

REFERENCES

- [1] "Putting QoS into VoIP" by Horacio Miranda de Oliveira, International Telecommunications, August 2000
- [2] "Understanding Latency in IP Telephony" by Alan Percy, Brook trout Technology white paper
- [3] "Measure VoIP Networks for Jitter and loss" by Martin Rowe, Test and measurement world
- [4] www.openh323.org Open H.323 Project web site

[5]www.techguide.com/comm/sec_html/voiceip.shtml

general information about Voice over IP

[6] TCP/IP Illustrated Vol 1: The Protocols

W. Richard Stevens

[7]Understanding Delay in packet voice networks white paper,cisco,2003

[8]Extricom white papers

[9]power consumption and energy efficiency comparisons of WLAN products,Atheros communication,2003

[10]Figure 3,4,5,6,7 are courtesy of Berthold Rathke.

BIOGRAPHY

Ashish Bhatia(ashishb@net.isquareit.ac.in) is a student of Advance Networking student at International Institute of Information Technology. He is B.E(Information Technology) from D.C.E. . His major field of research interest lies in the field of Wireless and voice Devices. Currently he is doing research on QoS in Wi-Max and many other research projects. He has great interest in VoIP and its security.