An Optimization Technique for Improved VoIP Performance over Wireless LAN

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***Abstract* - The performance of wireless LANs is greatly affected by pathloss, RF interference and other sources of signal attenuation, in addition to network congestion. The primary factors involved in effective real-time communication, namely delay and loss, must be within certain controlled limits in such a scenario. In this paper, we analyze the various factors driving IEEE 802.11b access points through extensive simulations and thereafter develop an optimization technique to configure the parameters of the access points. We simulate our test bed scenario and apply the developed algorithm. We then implement the configured parameters in our SIP enabled test bed to provide optimum Voice over IP (VoIP) performance. Thereafter, we further analyze and optimize the parameters of the communicating nodes being realized through Xlite soft phones that enhance VoIP performance to a much better extent both for voice and video calls. Test bed results confirm an improvement from 22% packet loss to 3% considering both AP and node configurations together. Finally, we propose optimization algorithm for proper selection of threshold parameters in active queue management system. Results verify significant performance improvement in VoIP under congested scenarios. Simulation and measured results have also been included.**

***Index terms* – Wireless LAN, VoIP, Access Point, IEEE 802.11b, Performance Optimization, Active Queue Management**

I. INTRODUCTION

In recent years, VoIP over wireless LANs has witnessed a rapid growth due to significant savings in network maintenance and operational costs and rapid rollout to new services. While being deployed widely in enterprise and home networks, the overall performance of VoIP strongly depends on the hardware properties of the concerned Access Points (APs). In order to maintain liveliness or a certain degree of interactivity, real-time traffic must reach the destination within a preset time interval with some tolerance [1]. Since VoIP based communication is targeted to handle real-time traffic exclusively, several protocols such as RTP, CRTP, RTCP, etc. are required for the proper transport of voice packets which are highly delay sensitive. Optimization of protocols is required to maintain Quality of Service (QoS) particularly under congested network condition.

Since we are dealing with VoIP, the communication must be real-time which means minimum delay and packet loss. So far, various optimization techniques have been proposed in terms of maintaining QoS with some acceptable limit. In [2], best-effort data control and admission control are proposed to guarantee QoS for real-time transmissions in the IEEE 802.11e wireless LANs. In [3], an error protection method is proposed for adaptive QoS. In [4], a traffic shaping scheme has been proposed for an AP from sending packets over unstable channel. Most schemes successfully try to avoid throughput degradation due to packet loss and retransmission on unstable links [4], [5]. However, there is further scope for improvement in optimizing the concerned parameters of the APs such as selection of Buffer Size, RTS threshold, Transmitter power, etc. Owing to frequent changes in network conditions coupled with diversity and environmental factors in addition to the unlicensed ISM band in IEEE 802.11b wireless LANs [6], it is difficult to develop a general algorithm for suitable optimization of AP parameters. In this paper, we have tried to resolve this issue by developing a suitable algorithm by carefully analyzing the direct and indirect factors guiding the APs and thereby suggesting an optimization technique that will help in configuring the AP parameters for optimal VoIP performance. Considering a snapshot of wireless network, close to the test bed as far as practicable, a simulated measurement analysis is done with NetSim. Results of this simulation are then applied to the developed algorithm to ascertain the optimized AP parameters. Next, we extend the work in optimizing the individual node parameters involved in the test bed after rigorous analysis for both voice and video calls, which have more stringent QoS requirements.

The algorithm developed by us is flexible in the sense that it can be used along with retransmission [7], link-level scheduling [4], channel error correction [3] and other optimization schemes. Implementation of active queue management [8] using optimization technique has also been incorporated to support our claim.

II. MOTIVATION

Maintaining Quality of Service (QoS) in VoIP has witnessed quick growth following rapid deployment of

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VoIP in various networks. Relevant optimizations have been carried out at different network layers and also in the call signaling protocols. Cross layer optimizations [9] have also been proposed for further improvement of VoIP performance particularly under congested scenarios. Currently, adaptive QoS optimization techniques [10], [11] have been proposed to deal with various unpredictable disturbances in WLANs.

Implementing VoIP in wireless networks has its own challenges [12]. Large deployments of 802.11 WLANs introduce several problems in handling real-time traffic for which proper solutions are highly needed [13]. Apart from optimizations in the links and traffic, the APs also need to be properly optimized in respect of queue length [14], retry limit [7], RTS threshold [15] etc. So far various optimization techniques have been proposed to enhance the performance of APs for real-time traffic.

Selection of APs has principally been focused when a node is in the vicinity of multiple APs. Selection based on the packet transmission delay associated with each AP has been considered in [16]. Minimizing the effect of low throughput node on other nodes is also a convenient technique to dynamically select AP in congested WLAN [17]. APs sending beacons for transmitting the R-values to all nodes in the wireless network enable the selection of best AP [18]. Selection based on a decision metric involving real-time throughput has been considered in [19] both for static as well as dynamic selection algorithm. AP association protocols based on throughput and frame loss rate [20], [21] have been considered as selection criteria.

For significant improvement in VoIP performance, apart from the selection of APs, stress has also to be given in controlling various parameters associated with the APs including the antennae. Significant optimization has been done in the domain of AP antennae for enhanced performance of VoIP networks [22], [23]. A scheme for increasing the number of simultaneous voice calls giving proper priority to APs has been highlighted in [24] with degradation of QoS well within the practically acceptable limit. The concepts of live migration and virtual AP [25] have also been suggested for the efficient deployment of APs in wireless LANs. Modified Deficit Round Robin (DRR) algorithm [26] replacing the standard PCF polling algorithm is the premium consideration behind the improvement in VoIP performance. However, suitable analysis of the packet access delay, based on M/G/I model provides an improved PCF algorithm [27] for better VoIP performance. Efficient handover to different APs [14] solves the problem of bottleneck in a highly congested network. A limit to maximum number of admission calls could be derived using Transmission Opportunity (TXOP) parameter of the MAC protocol [28].

From the literature survey, it is now observed that most of the work has focussed on the improvement in overall performance of the APs. However, very few works have been undertaken on the configuration of the AP parameters. Further, no work has been reported in respect of node (user agents) parameter control for the improvement in VoIP performance. Accordingly the basic motivation behind the work has been focussed towards the significant improvement of the VoIP performance by configuring both AP as well as individual node parameters and applying subsequently the existing optimization technique in the form of active queue management. Thus our work is based on three principal steps namely:

1. Analysis and optimization of AP
2. Extension to node parameter optimization and
3. Finally, the active queue management for getting best VoIP performance.

The reason for choosing IEEE 802.11b LANs is that it has been commercially deployed around the globe and our objective is to implement VoIP successfully in already established scenarios.

III. Analysis and Optimization of Access Point

IEEE 802.11b networks are governed by several factors [6], [29]. Our objective is to optimize the parameters of the APs. We focus on the factors that affect the performance of the APs and classify the salient features into two categories according to the control of AP on them. The controlled factors are buffer size, retransmission limit, RTS threshold, transmission power, antenna type, location factors and network load. Uncontrolled factors include path loss, fading, shadowing, interference, choice of codecs etc.

A. Analysis

In the analysis phase, we observe the performance of the network as a whole and the AP in particular under varying network conditions. We use NetSim [30], a simulation tool for this purpose and simulate a scenario where two nodes communicate each other in a congested environment. The parameters set for simulation are shown in Table I.

Firstly, we vary the buffer size of the AP and observe the variation in delay and loss as shown in Fig. 1. It is seen that increasing the buffer size results in decrease in loss as the AP can hold more number of packets in a congested network before transmitting them. However, the delay increases with buffer size in a congested network as the AP receives out of order packets and waits for earlier packets to arrive before the packets are sent to the receiver in the correct order. While this mechanism reduces jitter, the end-to-end delay increases which is highly undesirable for real-time traffic.

TABLE I.
PARAMETERS SET IN NETSIM

Parameter	Value
Buffer Size	Variable
Retry Limit	7
RTS Threshold	2347 bytes
Transmission Type	DSSS
Channel	1(Frequency:-2412 MHz)
Transmitter Power	100 mW
Channel Characteristics	Fading and Shadowing
Path-loss Exponent	3.5
Fading Figure	1 (Rayleigh Fading)
Standard Deviation (Shadowing)	12 (Ultra-high frequency)

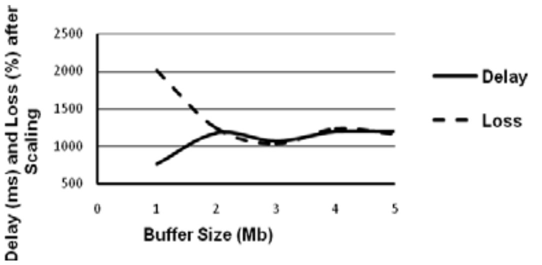


Figure 1. Variation in delay and loss with buffer size

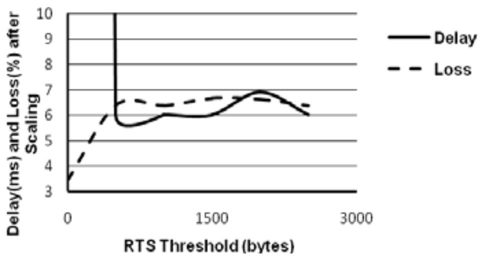


Figure 2. Variation in delay with increasing RTS threshold

Next, we study the effect of Request to Send (RTS) threshold parameter by varying it from 0 to 2347 bytes. Any packet whose length is greater than the RTS threshold is transmitted following a RTS-CTS (Clear to Send) exchange. NetSim allows RTS-CTS mechanism up to a threshold limit of 1500 bytes. The parameters of Table 1 are used to study the effect of buffer size and RTS threshold. Fig. 2 shows that delay decreases with increase in RTS threshold. A RTS threshold of 2347 bytes results in 6.04 ms of delay whereas a RTS threshold of 0 byte results in 523.83 ms of delay as a case study. Delay increases in the latter scenario as each packet waits for a RTS-CTS exchange before getting transmitted. Loss, unlike delay is directly proportional to the RTS threshold up to certain limit depending upon the nature of congestion as shown in Fig. 2. As RTS-CTS mechanism aims to minimize collisions among hidden stations, increasing RTS threshold increases collisions resulting in loss up to the above limit.

Fig. 3 shows the effect of retransmissions on the AP parameters. Increasing the retransmission limit also increases the delay and loss up to the above limit. Fig. 4 depicts the dependency of throughput of the network on retransmission limit for different path-loss exponent. The throughput starts decreasing with increasing retransmissions [31] for path-loss exponent beyond 3.5. So, keeping retransmission limit to a value such that loss and delay are at an optimum level without major throughput degradation is a matter of serious concern.

Finally, we analyze the performance under varying path loss condition. We consider Rayleigh fading with a standard deviation of 12dB for shadowing as the AP operates at Ultra High Frequency of 2412 MHz [32]. We vary the transmission power of the AP. Fig. 5 shows that for increasing path-loss, the minimum transmitter power required to keep the delay and loss to tolerable limits increases as nodes move away from the AP. So the AP must operate at maximum power. Our aim is to increase equivalent isotropically radiated power (EIRP) [33] which is given by (1),

$$EIRP = P_t \times G_t \qquad (1)$$

where, P_t denotes transmitted power and G_t stands for transmitted antenna gain.

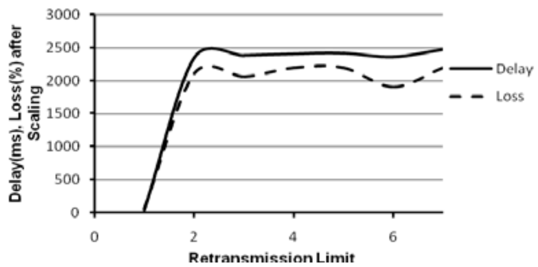


Figure 3. Increase in Delay and Loss with increasing retransmissions

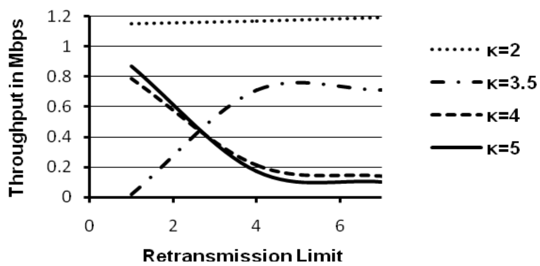


Figure 4. Variation of throughput with increasing retransmissions for various path-loss exponents

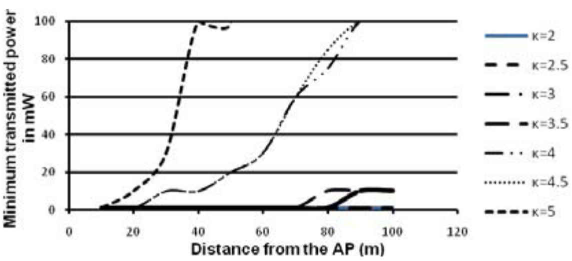


Figure 5. Minimal transmitter power requirement with increasing distance from the AP for path-loss exponents

As IEEE 802.11b APs operate in the unlicensed ISM band, any nearby RF network operating at similar frequencies may cause RF interference with APs. Such interference increases with increase in transmitter power. So, an optimum power must be determined. This governs the relative position of APs. As gain directly influences EIRP, a directional antenna with higher gain is advantageous than an omnidirectional antenna. However, directional antenna is not suitable with respect to real-time traffic as it involves more APs and greater handoff latencies. Therefore, omnidirectional antenna is preferred to directional antenna in VoIP communication.

B. Optimization Technique

B.1. Background Assumptions

Assuming that necessary error correction methods are in proper place, much importance is given to delay than packet loss particularly in the light of handling real-time traffic. The QoS metrics for ascertaining the performance of the network namely delay, jitter, packet loss, Mean Opinion Score (MOS) and R-Factor are categorized into *good*, *tolerable* and *poor* limits. Beyond the good limit is considered as the *threshold*. Further, path loss exponents are measured beforehand to have an idea regarding the variation of minimum transmitter power with distance from the AP for various path loss exponents as shown in Fig. 5. Moreover, omnidirectional antennae are assumed to be used. This algorithm is developed with consideration of the maximum range from AP where signal level deteriorates beyond 'good' value and delay and loss reach to maximum. It is also assumed that the average number of users in the network is known in advance. In other words, the algorithm is developed considering the worst case scenario and therefore sudden increase in users will not severely affect the performance.

B.2. Proposed Algorithm

B.2.1. Selection of Optimal Buffer Size (n)

a) Set the buffer size at the minimum level of 1 Mb.
b) Calculate the delay and loss for the concerned buffer size.
c) Increase the buffer size by 1.
d) Repeat step 'b'
e) Repeat steps 'c', 'd' until the buffer size reaches the maximum value of 5.
f) Plot the two metrics against the buffer size in Microsoft Excel.
g) View trendline of the respective curves and get the relevant equations.
h) Plot the equations in MATLAB as shown in Fig. 6 to find the point of intersection *P* corresponding to the buffer size n'.
i) View the $delay_p$ and $loss_p$ corresponding to the point *P*.

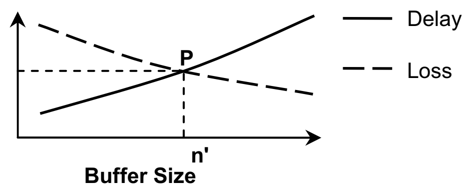


Figure 6. Point of intersection for delay and loss curves obtained in MATLAB

The intersection of the two curves may lead to two cases.

CASE – I: Both $delay_p$ and $loss_p$ are within the *threshold* limits.

Shift the intersection point *P* along the delay graph to the point *Q* corresponding to the buffer size n'' so that delay becomes lesser while loss increases and reaches its *threshold* value. Fig. 7 shows this shifting to utilize optimal buffer size with acceptable loss and delay.

CASE – II: Delay is unacceptable but loss is within *threshold*.

Shifting is done as in Case – I but the shifting is mandatory in Case – II as shown in Fig. 8 to maintain the QoS of the network while the shifting is optional in Case – I.

In both the cases, the optimal buffer size is selected as $n = n''$.

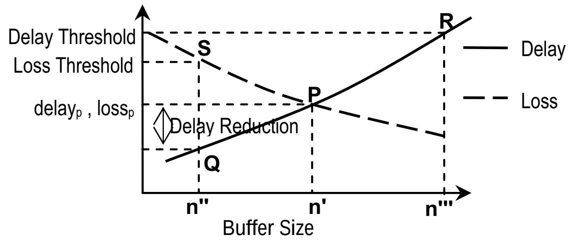


Figure 7. Delay Reduction for optimal buffer size n'' when $delay_p$ is within the threshold value

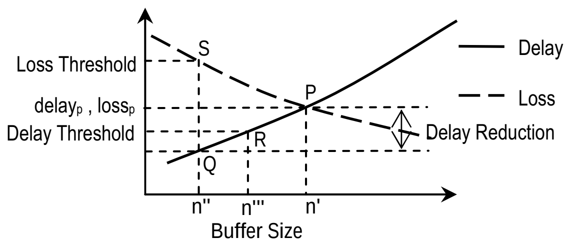


Figure 8. Delay Reduction for optimal buffer size n'' when $delay_p$ is above the threshold value

B.2.2. Reduction of packet loss.

In the previous sub-section, the optimal buffer size has been selected reducing the delay beyond the threshold value while the packet loss has been increased up to the safe limit. Now a new mechanism of RTS-CTS is being incorporated to reduce the loss beyond threshold with the optimal buffer size.

a) Plot the delay and loss for selected optimal buffer size n'' after appropriate scaling by decreasing the RTS threshold from 2347 to 0 byte.
b) View trendline of respective curves and get the relevant equations.
c) Plot the equations in MATLAB as shown in Fig. 9 to find the intersection point P'.
d) Move along the delay curve from P' towards Q' start until delay reaches the *threshold* limit. Let the corresponding RTS threshold be *thresh*.
e) Accept *thresh* as optimal RTS threshold if it is less than or equal to Maximum Transmission Unit (MTU) in Bytes.

Up to this sub-section, the optimal values of buffer size and RTS Threshold have been selected as n'' and *thresh* respectively.

B.2.3. Selection of Retransmission Limit (r)

a) It is applicable when both delay and loss are within the threshold limits while the path-loss exponent is less than 4.

b) Increase *r* with an initial value of 0 in steps of 1 until either the delay or loss reaches the threshold (whichever is earlier).

c) With one of the metrics reaching threshold limit, the number of retransmissions (*r*) of voice packet becomes fixed at *r′*. Thus the limit is set as *r′*.

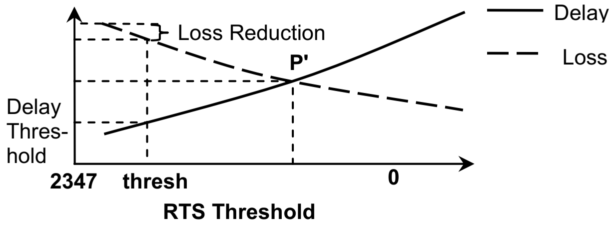


Figure 9. Loss reduction for optimal RTS threshold

B.2.4. Selection of Transmission Power (p) and Location (loc) of the new AP

Let A = number of APs available, d = distance of the receiver from the AP, p = transmitter power, κ = path-loss exponent, l = maximum distance up to which the signal is recognizable.

For d = 1 to l {
a) Select t_x from the graph of Fig. 5 for a given κ.
b) Measure the corresponding loss and delay from NetSim.
c) If both are within the threshold limit{
i. If there is no adjacent RF source {Select p = maximum power of the relevant AP.}
ii. Else {Select $p = t_x$.}
d) Else if *threshold* is crossed or network is out of range{
i. If($A > 0$){
a. Place a new AP at location *loc* such that *loc* = (d-10) m.
b. $A=A-1$.
c. Select a non-overlapping channel.
d. Repeat the For loop.}
ii. Else {
a. Increase t_x by external means to t_x'.
b. Select $p=t_x'$. } } }

So the developed optimization algorithm finally provides the buffer size (=*n′′*), the RTS threshold value (=*thresh*), retransmission limit (=*r′*), transmitter power (=*p*) and location (=*loc*) of a new AP. The various steps to be followed in order to find the optimum value of the above said parameters are outlined in the flowchart, Fig. 11 in the following page.

C. Implementation

The proposed algorithm is now implemented both in NetSim simulator and the test bed for improving the quality of voice calls in wireless LANs. Delay, packet loss and MOS thresholds are considered to be 180 ms, 5% and 3 respectively as call quality degrades beyond these values.

C.1. Implementation in NetSim

At first, we implement the proposed algorithm in NetSim. We create a scenario where two nodes are communicating through voice amidst the presence of data traffic from other nodes present there. The scenario is implemented for path-loss exponents κ =2 and κ =4.

For κ =2 and buffer size of 1 MB, the delay and loss are 91.18 ms and loss 2.502 % respectively. For buffer size of 5 MB, the delay and loss become 201.32 ms and loss 1.232 %. As per the proposed algorithm, the buffer size is considered to be 1 MB and retry limit is taken as 2. The RTS threshold limit is configured to 2347 bytes. Further, the minimum transmitter power is selected to be 20 mW and the maximum allowed location is taken to be 100m. The voice call is made in NetSim with the parameters configured as above. The final delay is 119.66 ms and the loss is 3.368 %, both of which are within threshold limits.

For κ =4, the delays are 150 ms and 493.6 ms for buffer size of 1MB and 5MB respectively. The corresponding loss rates are 7.912 % and 6.322 %. Applying the proposed algorithm, the buffer size is kept at 1 MB. The retry limit of 1 is accepted and RTS threshold limit is taken as 0 byte. The minimum transmitter power of 100 mW is selected with the maximum allowed distance to be 50 m. Simulation of voice call in NetSim after configuration of the parameters results in 132.477 ms of delay and 3.277 % of loss. Both of them are within tolerable limits. Thus the proposed algorithm helps to maintain voice calls within an acceptable delay and loss in diverse scenarios.

C.2. Implementation in test bed

We further implement our proposed optimization algorithm in real test bed. Initially a model of the test bed is created in NetSim to obtain optimal parameter values applying the algorithm. Then the parameters are applied in test bed to improve VoIP performance.

C.2.1. Test bed Description

Our experimental test bed, as shown in Fig. 10 consists of fixed and mobile nodes for VoIP communication, a wireless AP, a switch and a SIP server. The Brekeke SIP server [34] used here is SIP Proxy Server [35] and Registrar. It authenticates and registers user agents such as VoIP device and soft phone, and routes SIP sessions such as VoIP calls between user agents. The server has record-route facility and performs proxy authentication to provide security. Further, we use ManageEngine VQ Manager [36], a powerful, web-based, 24x7 real-time QoS monitoring tool for VoIP networks, to analyze the various QoS metrics associated with an ongoing call.

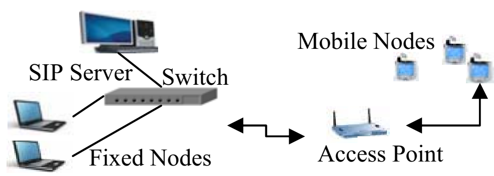


Figure 10. Test bed

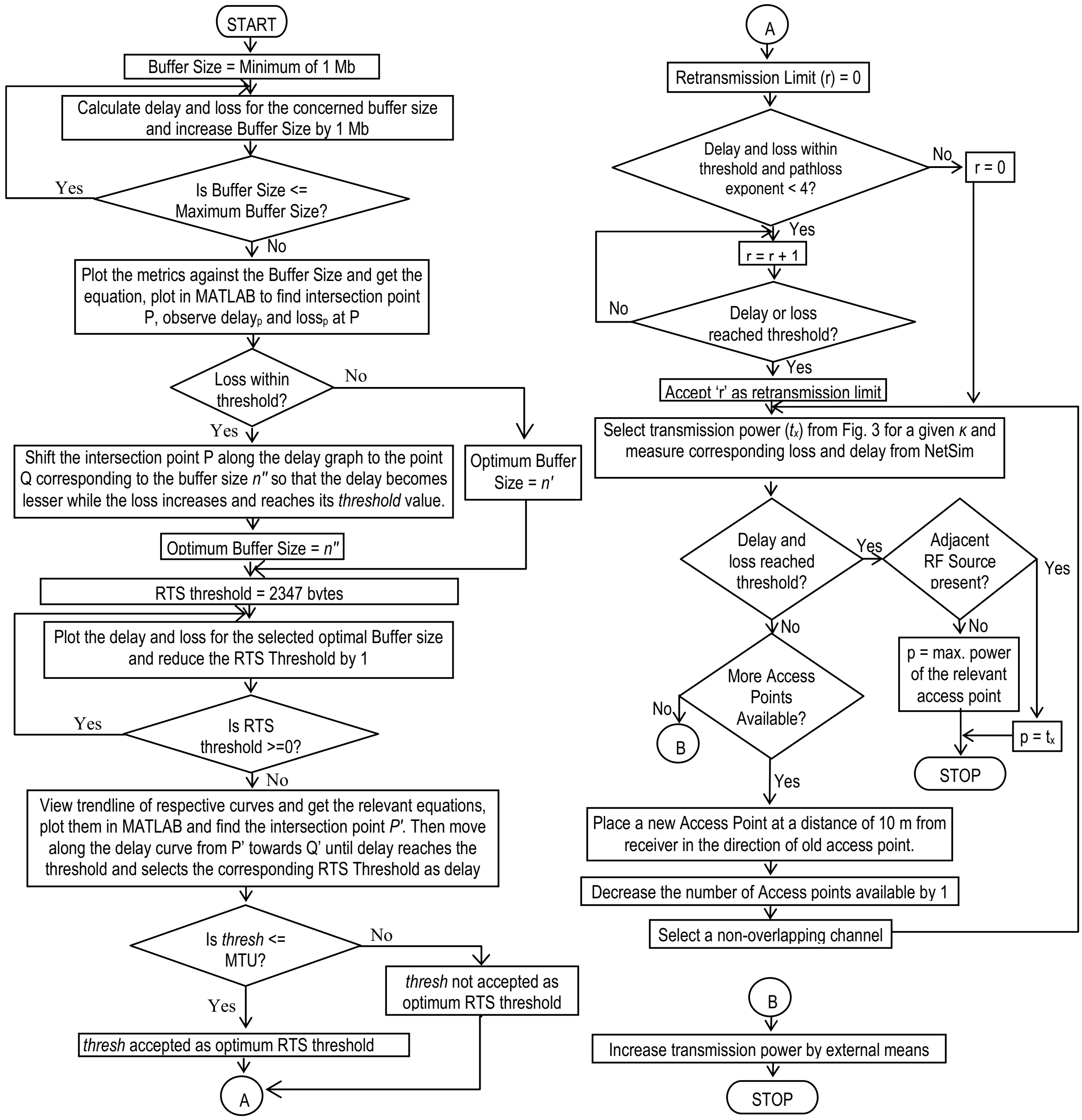


Figure 11. The various steps of the developed algorithm

C.2.2. NetSim Simulation of the test bed scenario

We simulate our test bed scenario in NetSim and apply the developed optimization algorithm. The relevant intersection points correspond to 4.52 Mb is shown in Fig. 12. The corresponding delay is 158.3 ms and the loss is 3.43%. Now as the delay is near the threshold (180 ms) in our case but the loss is within limits, we apply the technique to select the buffer size of 2.63 Mb where the delay is 155 ms and the loss is 5% as seen in Fig. 13. Following the algorithm, the retransmission limit is kept to 0 and the RTS threshold is kept at 2347 bytes. As there is no other source of RF interference in our test bed, we take the maximum transmitter power to increase the coverage.

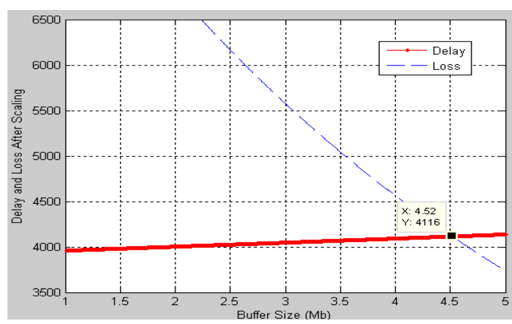


Figure 12. Intersection points of delay and loss in MATLAB

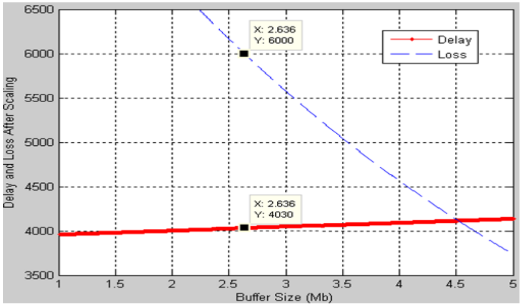


Figure 13. Optimal buffer size and the corresponding scaled delay and loss plotted in MATLAB

C.2.3. Calculation of Path Loss Exponent

The path loss exponent plays a very important role in the proper implementation of our algorithm. So, careful measurements are taken to find out the magnitude of the path loss in our test bed. For this purpose, we have used a mobile receiving device, a laptop in our case, equipped with a RF power measuring software, NetStumbler. We have measured the received signal strength at different distances from an AP. The data that we have collected, as shown in Fig. 14, is used to find out the value of the path loss exponent using the log normal shadowing model [37]. The model is described by (2) as follows.

$$P_{ls}(x)(dB) = P_l(x_0) + 10\kappa \log_{10}(x / x_0) + x_\sigma \qquad (2)$$

Where, $P_{ls}\ (x)$ is the path loss at a distance of x metre from the AP, $P_L\ (x_0)$ is the path loss at a reference distance x_0 metres, κ is the path loss exponent and X_σ is a zero mean Gaussian random variable.

We get a value of 3.5 by following the above procedure.

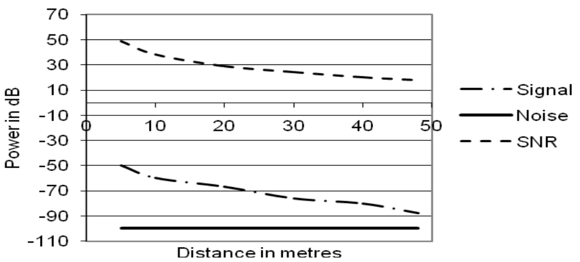


Figure 14. Received Signal Power with respect to Distance from AP

C.2.4. Implementation In The Real Test bed

All the parameters derived from the developed optimization algorithm and calculated value of κ are now added to our test bed through Network delay Simulator and Network Emulator for Windows Toolkit (NEWT). The outcome from the test bed comes through VQ Manager in terms of delay and loss as shown in Fig. 15. It is clearly seen from Table II that the results obtained after applying the optimization technique are optimal. Since the readings are taken at a distance of 30m from the AP and the path loss is calculated to be 3.5 in our test bed, we place a new AP at a distance of 20m applying the algorithm. This results in further optimization as the MOS increases to 4 from 3.4.

TABLE II.
IMPLEMENTATION RESULTS

Scenario	Avg. Delay	Avg. Loss%	MOS
Maximum Queue Size	112 ms	0	4.4
Minimum Queue Size	11 ms	22	3.1
Obtained Queue Size	96 ms	8	3.4

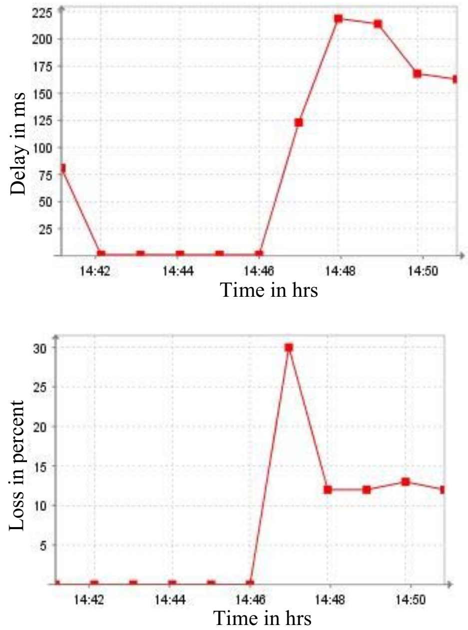


Figure 15. Optimization of Delay and Loss in Test Bed

IV. ANALYSIS AND OPTIMIZATION OF THE NODE PARAMETERS

The parameters of individual nodes (soft phone) such as buffer size, RTS threshold and retry limit must also be configured after proper analysis. As the number of calls increase, optimization of these parameters becomes a matter of great concern. However, this optimization must be done in conjunction with the configured AP parameters. The analysis includes the configuration of most general parameters.

A. Analysis

Each node has its buffer which has to be properly configured for effective voice communication. It is seen from simulations in NetSim that the buffer size of the individual nodes depends on the buffer size and the retransmission limit of the APs. Fig. 16 shows that increasing the buffer size of the APs increases the overall delay as well. Moreover, for each AP buffer size, there is a sudden increase in delay and loss with increase in retransmission limit [38]. So the AP buffer size must be decreased which results in further loss. Further, there has to be a minimum node buffer size for each retransmission limit to avoid considerable loss in packets as seen from fig. 17. Hence, it is mandatory to carefully initialize the buffer size of the individual nodes.

The RTS threshold parameter in each node has the same function as stated in Section IIIA. Extensive simulations in NetSim point to the fact that the RTS threshold parameter of each individual node along with the RTS threshold parameter and the buffer size of the AP affects the overall delay and packet loss. Increasing the RTS threshold of each node decreases the delay significantly and increases the loss as can be seen in fig. 18. Delay is further reduced with increase in AP RTS threshold parameter and decrease in AP buffer size. So, selecting an appropriate RTS threshold value for the individual nodes has to be done with utmost care.

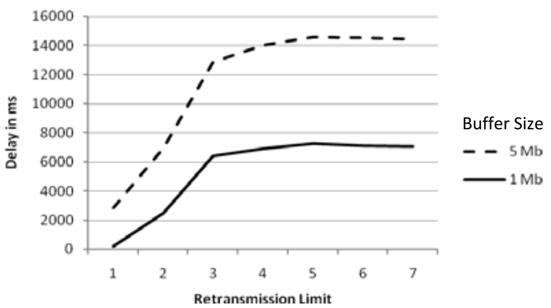


Figure 16. Variation of delay with increase in retransmission limit for different buffer sizes

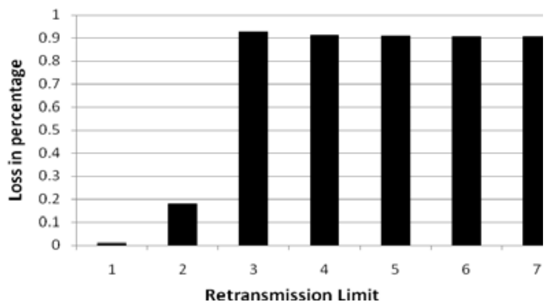


Figure 17. Variation of loss with increase in retransmission limit for buffer size of 1 Mb

B. Optimization Technique

B.1. Selection of Retransmission Limit (r_{node})

For the sake of simplicity, the retransmission limit r_{node} is kept the same as the retransmission limit (r) of the APs as obtained in Section IIIB.

B.2. Selection of Buffer Size (n_{node})

Two scenarios are possible as given under.

SCENARIO – I: r_{node} =0

If the node has memory constraints, n_{node} is selected as the maximum size available or else n_{node} is selected as the AP buffer size n as obtained in Section IIIB.

SCENARIO – II: r_{node} is not equal to 0

B.2.1. Selection of the minimum buffer size (n(r)) for each retransmission limit (r)

a) Set the retransmission limit r as 1.
b) Set the buffer size at the minimum level of 1 Mb.
c) Calculate the loss for the concerned buffer size.
d) Increase the buffer size by 1.
e) Repeat step ‘c’
f) Repeat steps‘d’, ‘e’ until the buffer size reaches the maximum value of 5.
g) Plot the metric against the buffer size in Microsoft Excel.
h) Select the buffer size $n(r)$ after which there is a sudden increase in loss.
i) Increase the retransmission limit by 1.
j) Repeat steps from ‘b’ to ‘h’.
k) Repeat steps ‘i’, ‘j’ until the retransmission limit reaches the maximum value of 7.

B.2.2. Selection of the optimal buffer size(n_{node})

a) Set the retransmission limit as r_{node}.
b) Set the buffer size as n (r_{node}).
c) Select the access point buffer size as obtained in Section IIIB.
d) Repeat Section IIIB.2.1 for each individual node.
e) Shift the intersection point P along the delay graph to the point Q corresponding to the buffer size n (r_{node}) so that the delay becomes lesser while the loss increases. This shifting is shown in fig. 19. Let the delay at Q be $delay_{node}$.
f) Shift the intersection point P along the delay graph to the point R corresponding to the buffer size n''

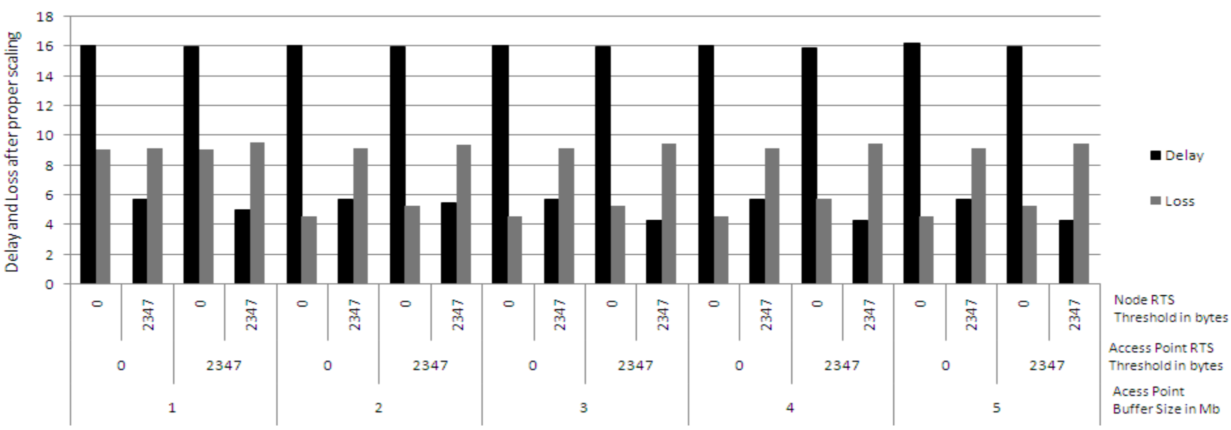


Figure 18. Variation in delay and loss for various node and AP RTS Threshold parameters with increase in AP buffer size

as shown in fig. 19 so that the delay reaches threshold. Let the delay at *R* be $delay_{thresh}$.

g) Shift the intersection point *P* along the loss graph to the point *S* corresponding to the buffer size n''' as shown in fig. 21 so that the loss reaches threshold. Let the loss at *S* be $loss_{thresh}$.

The above procedure may lead to two possible cases.

CASE – I: $n' <= n\ (r_{node})$

If $delay_{node} < delay_{thresh}$ the optimal buffer size is selected as $n_{node} = n\ (r_{node})$ as shown in fig. 19 else $n_{node} = n''$ as in fig. 20.

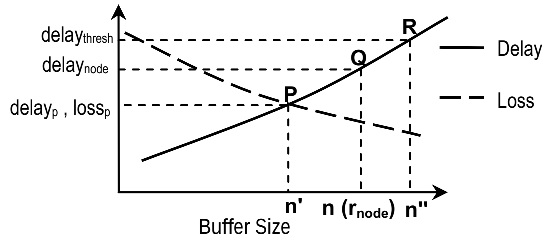


Figure 19. Selection of optimal buffer size when $n' <= n\ (r_{node})$ and $delay_{node} < delay_{thresh}$

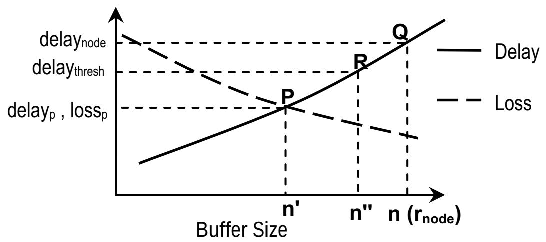


Figure 20. Selection of optimal buffer size when $n' <= n\ (r_{node})$ and $delay_{node} > delay_{thresh}$

CASE – II: $n' > n\ (r_{node})$

If $delay_{node} < delay_{thresh}$ the optimal buffer size is selected as $n_{node} = \min\ (n\ (r_{node}),\ n''')$ as seen in fig. 21 else $n_{node} = n''$ as in fig. 22

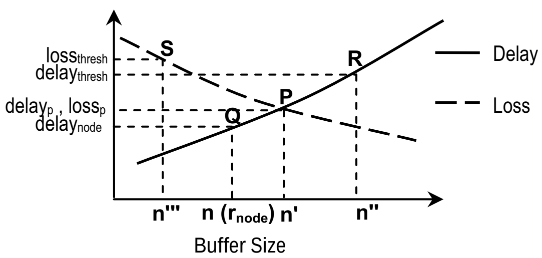


Figure 21. Selection of optimal buffer size when $n' > n\ (r_{node})$ and $delay_{node} < delay_{thresh}$

B.3. Selection of RTS threshold ($thresh_{node}$)

a) It is applicable in scenarios where either loss is greater than threshold or both delay and loss are less than threshold. In all other cases, the default RTS of 0 is selected.

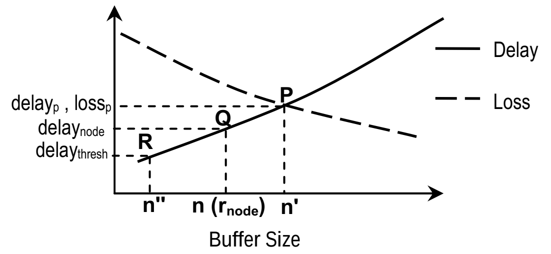


Figure 22. Selection of optimal buffer size when $n' > n\ (r_{node})$ and $delay_{node} > delay_{thresh}$

b) Plot the delay and loss for the selected optimal buffer size n_{node} after appropriate scaling by increasing the RTS threshold from 0 to 2347 bytes.

c) View trendline of respective curves and get the relevant equations.

d) Plot the equations in MATLAB to find the intersection point *P'*.

e) Move along the delay curve from *P'* towards *Q'* start until delay reaches the *threshold* limit. Let the corresponding RTS threshold be *thresh* as shown in fig. 23.

f) Accept *thresh* as $thresh_{node}$ if it is less than or equal to Maximum Transmission Unit (MTU) in Bytes.

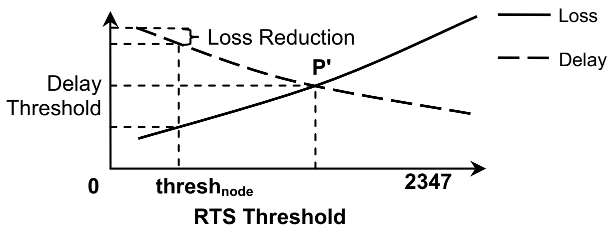


Figure 23. Selection of optimal RTS threshold

So the developed optimization algorithm finally provides the buffer size ($=n_{node}$), the RTS threshold value ($=thresh_{node}$) and the retransmission limit ($=r_{node}$) for the communicating nodes.

C. Implementation

C.1. Voice Call

The proposed optimization technique is now applied in voice calls in the test bed as described in Section IIIC. The parameters of each node are obtained after initial configuration of the AP parameters as discussed in Section III and further simulation of the test bed scenario in NetSim. As softphones are used as the nodes, there are no memory constraints. Hence the buffer size of each node is selected to be 2.63 Mb according to the proposed algorithm. Further, the RTS threshold is configured to be 0 byte with the retry limit kept at 0 following the algorithm. The parameters obtained are now applied in a voice call in the test bed. It is seen from Table III that both delay and loss decrease further from 96 ms and 8 % respectively, as mentioned in Table II, to 82ms and 3% respectively when the optimization techniques are applied to both the APs and the node parameters. Moreover fig. 24 suggests that delay and loss remain nearly constant thereby minimizing the chances of high jitter.

C.2. *Video Call*

The optimization algorithm is further applied to video calls in a wireless network, which have more stringent QoS requirements in terms of delay and loss than voice calls. A snapshot of the original scenario is shown in fig. 25(a). Implementation of video calls in the test bed which is described in Section IIIC, leads to 22% of packet loss initially as observed in fig. 25(b) where the picture is unrecognizable. Implementation of the AP optimization technique, as described in Section IIIB, results in reduction of packet loss to 8 %. A snapshot of the video call at this moment is shown in fig. 25(c). Here the video is almost recognizable. Finally, the optimization technique of the individual nodes, as stated in Section IVB is applied. A snapshot of the video call in fig. 25 (d) shows further reduction in delay and loss to 82ms and 3 % respectively and the video call quality is almost the same as the initial one as MOS is further enhanced to 4.4. Thus the video call is optimized in a wireless medium after application of our proposed algorithm in APs and individual nodes.

TABLE III.
IMPLEMENTATION DETAILS FOR A VOICE CALL AFTER APPLICATION OF OPTIMIZATION TECHNIQUES

Scenario	**Avg. Delay**	**Avg. Loss%**	**MOS**	**Comments**
Maximum Queue Size	112 ms	0	4.4	Delay is high
Minimum Queue Size	11 ms	22	3.1	Loss is high
Obtained Queue Size of the APs	96 ms	8	3.4	MOS is acceptable but loss is > threshold
Obtained Queue Size of the individual nodes along with the obtained Queue Size of the APs	82ms	3	4.4	Both loss and delay <threshold and MOS is high

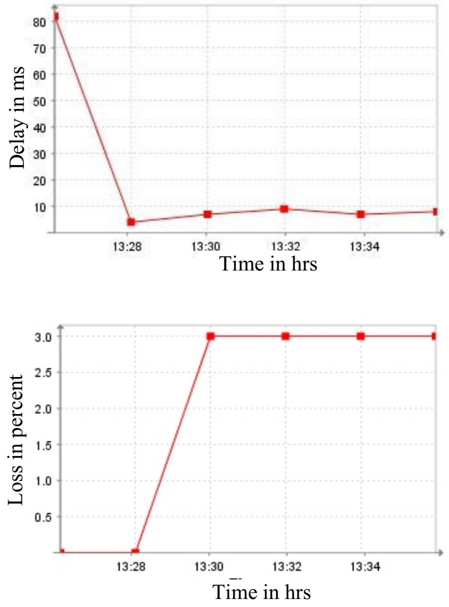
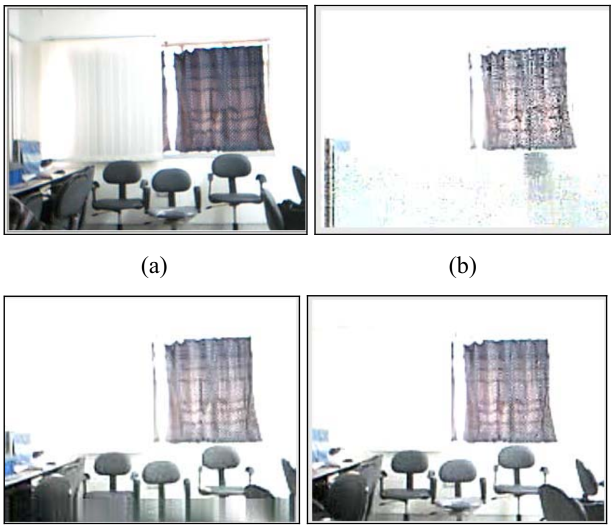


Figure 24. Optimization of delay and loss in test bed



(a) (b)

(c) (d)

Figure 25. Snapshots of video calls with loss of (a) 0% (b) 22 % (c) 8 % (d) 3 %)

V. ACTIVE QUEUE MANAGEMENT OPTIMIZATION

Queues are essential to store and forward packets following certain algorithms. A good buffer management scheme must have the following features.

a) It should maintain the network in a region of high throughput and low loss.
b) It should avoid bias against bursty traffic which is common in buffers with tail-drop mechanism.
c) It should avoid global synchronization. When loss occurs due to packet drop, all TCP flows reduce their window size to half and then increases slowly. This must be avoided.

Static buffers drop packets on being filled up. The packet drops are implemented with tail drop, random drop on full and drop front on full strategies. So these buffers cannot always satisfy the aforementioned features. Active queues drop packets before the queue is full to ensure the above features. The Random Early Detection algorithm [8] is one of the active queue management policies.

It has two parts namely,

1. Estimation of average queue size

   The average queue size is given by (3) as follows.

$$Q_{avg} = (1 - W) \times Q_{avg} + W \times Q_{sample}, 0 < W < 1 \quad (3)$$

Where, Q_{avg}= average queue size
Q_{sample}= instantaneous queue size
W= weight

2. Decision of packet drop

The decision to drop the packet must depend on the average and not the instantaneous queue size. Two thresholds are defined as Min_{th} and Max_{th}.

The decision to drop a packet as shown in fig. 26 is given as follows:

If $Min_{th}<Q_{avg}<Max_{th}$, packet is dropped with a probability 'p'.

If $Q_{avg}<Min_{th}$, no packet is dropped.

If $Q_{avg}>Max_{th}$, every packet is dropped.

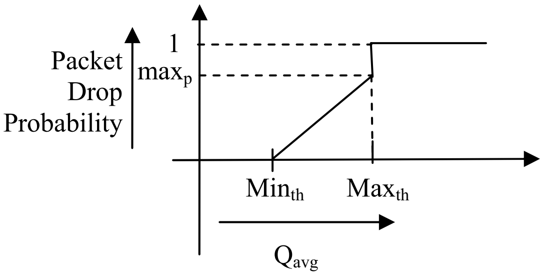


Figure 26. Variation of packet drop probability with increasing average queue size

The probability p is given by (4) as follows.

$$p = \max{}_p\{(Q_{avg} - Min_{.th})/(Max_{th} - Min_{.th})\} \quad (4)$$

Where, max_p= maximum packet drop probability
Q_{avg}= average queue size
Min_{th} = minimum threshold
Min_{th} = maximum threshold

So selection of appropriate threshold limits is crucial for significant performance improvement after implementation of active queue management system. Our objective is to select the optimal threshold limits namely Min_{th} and Max_{th} with the help of our proposed optimization algorithm for the APs as described in Section IIIB.

A. Optimization Technique

There are two possible cases that need to be dealt with.

CASE – I

1. It is applicable for Case 1 and Case 2 of Section III B.2.1 and is shown in fig. 27 and fig. 28 respectively.
2. Select Min_{th}=current buffer size n''.
3. Select Max_{th}=buffer size corresponding to delay threshold n'''.

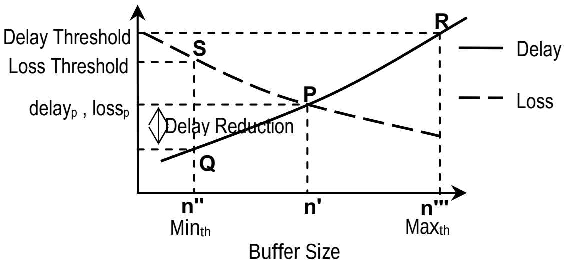


Figure 27. Selection of threshold parameters when both delay and loss are within threshold

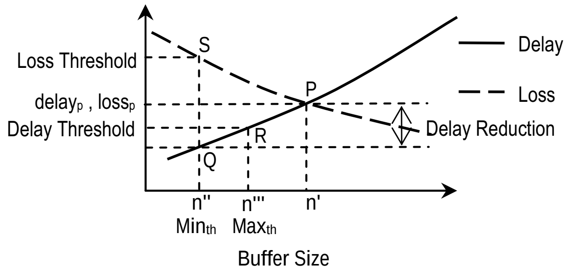


Figure 28. Selection of threshold parameters when delay is unacceptable

CASE – II

1. It is applicable when both delay and loss are greater than the threshold and is shown in fig. 29.
2. Select Max_{th} = current buffer size n''.
3. Plot delay and loss curves as referred to in Section III.
4. Shift the intersection point P along the loss graph to the point Q corresponding to the buffer size n''' so that the delay becomes lesser while the loss increases by 2 %. Let the loss at Q be $loss_Q$.
5. Select $Min_{th} = n'''$.

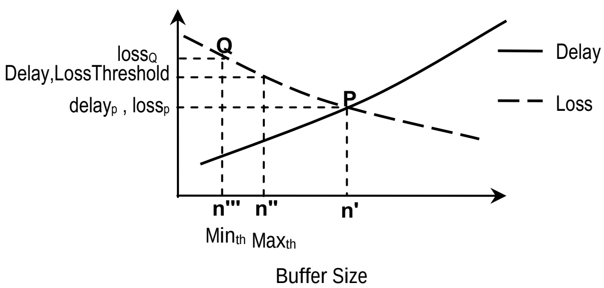


Figure 29. Selection of threshold parameters when both delay and loss are above threshold

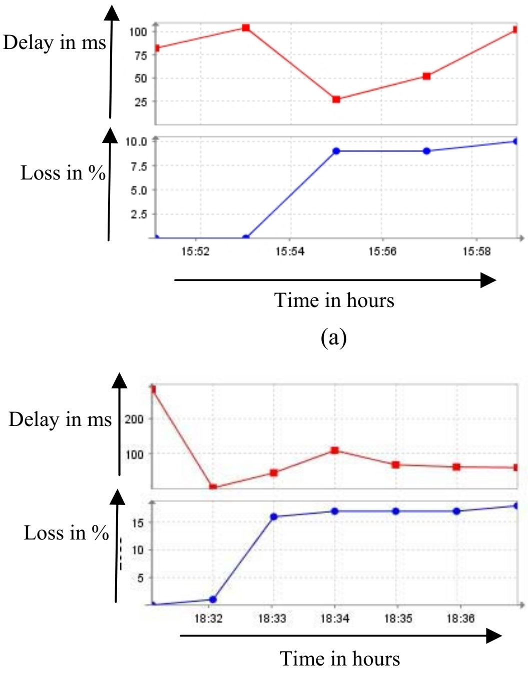
B. Implementation

The proposed optimization technique is applied in the test bed to select the minimum and maximum threshold values. As per the algorithm, the buffer size of 2.63 Mb is selected as the minimum threshold Min_{th}. The Max_{th} is taken corresponding to the delay threshold of 180 ms. As it is more than the maximum AP buffer size of 5 Mb, Max_{th} is selected as the maximum buffer size of 5 Mb.

Initially RED is implemented with its default threshold parameters and voice calls are made. Thereafter, RED is implemented in voice calls after configuration of the threshold parameters with the values obtained above. As observed in fig. 30(a) and fig. 31(a), in a low congested medium, delay and loss are comparable for RED implementations with default and configured threshold parameters respectively. With increase in background traffic, loss increases when RED is implemented with default parameters as shown in fig. 30(b). However, RED with configured parameters keeps both delay and loss within tolerable limits as shown in fig. 31(b). Thus the configuration of the AP parameters also helps to configure the threshold parameters in RED which, in turn, provide an optimal performance as shown in Table IV.

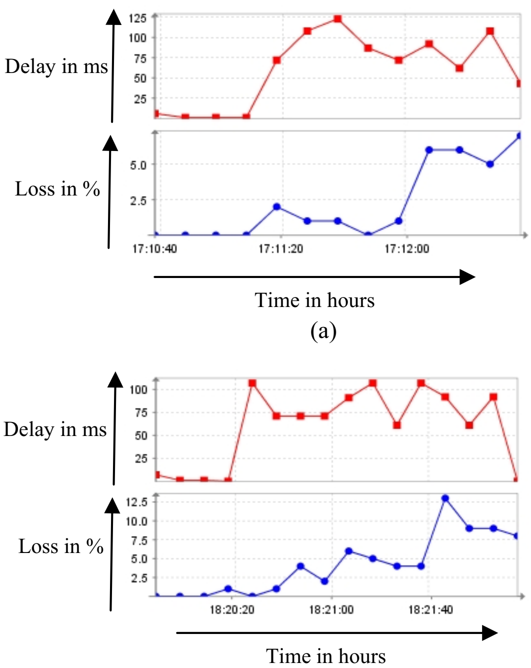
TABLE IV.
RED IMPLEMENTATION DETAILS

Parameters	Background traffic 1kbps		Background traffic 10kbps	
	Default	Confi-gured	Default	Configured
Delay in ms	66	61	70	63
Loss in %	6	6	14	4



(b)

Figure 30. Variation of delay and loss for default RED threshold parameters in (a) uncongested medium (b) congested medium



(b)

Figure 31. Variation of delay and loss for the optimized RED threshold parameters in (a) uncongested medium (b) congested medium

VI. CONCLUSION

In this paper, we address the VoIP performance optimization following a three step procedure. At first, AP parameter optimization is performed for the improved VoIP performance. After analyzing the various factors, related to AP, we observe that for optimum performance, a certain trade-off is required among all the parameters. While this decision depends on the current network condition, we have developed a general algorithm based on simulated measurements that is applied in all such scenarios to get a fair understanding of the optimal values required for each parameter. The optimal values of buffer size, different threshold limits, power, and location have been ascertained with respect to real-time traffic. Secondly, we have dealt with the problem of optimal configuration of the node parameters and devised optimization technique which further enhances VoIP performance. Finally, we establish optimization technique to select threshold parameters while implementing active queue management systems. Our algorithm has resulted in 8% packet loss considering optimization of AP parameters in comparison with 22 % without any optimization. The inclusion of node parameters in the algorithm further reduced the packet loss to 3% immediately after the call initiation process. The proposed optimization process is flexible enough to incorporate active queue management also by the selection of threshold parameters only.

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