Study of Enhanced VoIP Performance under Congested Wireless Network Scenarios

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Abstract— **Stringent QoS maintenance in a wireless environment for VoIP communication is a major challenge. VoIP inherently generates constant bit rate traffic and is highly sensitive to network delay. However, unpredictable network congestion makes a VoIP session so degraded that its QoS goes below a tolerable limit. Accordingly, a suitable solution is needed to adapt varying network conditions satisfying minimum QoS while maintaining transparency to the end user. We have extensively studied the effect of variable voice packet payload size with changing number of voice sample frames in the payload of the RTP packets. For this, we have used OPNET Modeler 14.5.A to observe the performance in terms of MOS, End-to-end delay through extensive simulations. Results are provided with valid discussions. Based on the observation, we propose an Adaptive VoIP (AdVoIP) algorithm that may be used in any real life VoIP network for further enhancement of the performance. The adaptive algorithm may use RTCP Receiver Reports to assess the network conditions in real wireless scenarios.**

Keywords-Voice over IP; Congested Wireless network; voice codec; payload size; delay; MOS; OPNET

I. Introduction

Real-time applications have their own set of benefits and requirements. Its broad spectrum provides us with facilities like IP telephony, teleconference, etc. However, on the other hand it gives us the burden of finding feasible solutions to meet its stringent time requirements. It is a well known fact that the Internet backbone, which is the transport medium for any information across the globe, is time-variant. The characteristics of the Internet backbone are not known in advance, since they depend on the behavior of the other connections throughout the network [1]. The connectivity may be hampered for several reasons rendering networking applications ineffectual. Often the networks suffer congestion i.e. traffics exceeding the capacity of the network are routed through it. The effect of this event is that the data packets suffer from high delay and loss while passing through the network. Such delay and loss are unacceptable in case of real-time applications. Of the various real-time applications we have concentrated on Voice over IP (VoIP) since it has gained importance over the past few years owing to its low cost and ease of interfacing between data and voice traffic [2].

VoIP applications generally send data at a constant rate and do not incorporate any control mechanisms. The main reason behind the absence of control mechanism is that the VoIP applications mainly resort to User Datagram Protocol (UDP) for their functioning because of the lower network overhead achieved for sending UDP packets [2]. But UDP does not have any congestion control mechanism. So, if the network gets overloaded, the VoIP sources do not change their data rate leading to further degradation in the network performance. Hence, we need some mechanism to decrease the rate of feeding data to an already congested network. Using adaptive rate VoIP is one of the solutions, where adaptive multi-rate (AMR) speech Coder-decoder (codec) is used. Some works have suggested using a set of speech codecs having different bit rates [1]. The bit rate is reduced in a congested situation. However, such reduction of bit rate degrades the voice quality. In a Wireless Local Area Networks (WLANs), communication suffers from increased bit error rate and higher delay and loss.

In this paper, we have tried to explore packetization as an alternative means of rate adaptation of VoIP sources when they are being operated in WLANs. We have varied the size of the voice packets by changing the number of voice sample frames in the payload of the RTP packet as per the situation. The bit rate is not varied and hence the voice quality is not degraded. Further, we have proposed an algorithm to vary the packet payload size in a systematic manner if the number of VoIP users is increased or the network gets congested by some other reasons. Our focus is to achieve better QoS under varying network condition.

II. Background Study

VoIP makes use of the already well established Internet backbone in order to transport voice packets. The Internet Protocol (IP) is the main pioneer for carrying out the network layer operations of VoIP. In the transport layer, both Transmission Control Protocol (TCP) and UDP can be used. UDP is preferred over TCP in order to meet the time sensitiveness of the voice traffic [2][3]. Moreover, TCP consumes more bandwidth than UDP for maintaining higher quality of voice traffic whereas UDP accommodates more sessions in the same network by reducing bandwidth consumption [3]. Real-time Transport protocol (RTP) and Real-time Transmission Control Protocol (RTCP) are used for time stamping and sequencing the packets [4]. This method is adopted because the voice packets may take different routes from the sender to the receiver and arrive out of order. The timestamps and sequence numbers help in rearranging the packets. RTCP plays a very important role in controlling

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network congestion. Since, VoIP mostly operates on UDP; it does not have any inherent congestion control mechanism. It also lacks from the reliability offered by TCP. RTCP working in tandem with RTP seeks to solve the above problems.

RTCP is based on the periodic transmission of control packets to all participants in the session, using the same distribution mechanism as the data packets [4]. The primary function of RTCP is to provide feedback on the quality of data distribution. This feedback information obtained from the receiver reports (RR) allows the source to estimate the loss rates experienced by the receivers and to adjust its bandwidth accordingly. The RRs contain information about fraction lost, cumulative packet loss, inter-arrival jitter, etc. Moreover, RTCP also allows each participant in a VoIP session to have a unique Canonical Name (CNAME) as an identifier. Thus, it also helps each of the participants to keep a track of the number of VoIP users. However, if the number of participants in the session is very large, then the number of RTCP packets exchanged between the participants should be controlled and must not be allowed to exceed a certain value because it may further lead to worse congested network.

Speech, as we know, comes out of the source in analog form. In order to transmit it through the Internet backbone, it is mandatory to digitize the speech sample obtained [2][3]. This is done using several modulation techniques like Pulse Code Modulation (PCM), Adaptive Pulse Code Modulation (ADPCM), Algebraic Code Excited Linear Prediction (ACELP), Low-delay code excited linear prediction (LDCELP), Multi-Pulse - Maximum Likelihood Quantizer (MP-MLQ), Conjugate-Structured ACLP (CS-ACELP), etc. Compressing and packetizing the voice sample is required to make it feasible for successful transmission [3]. Compression is done so that the bandwidth consumed for sending the voice sample is reduced. Codecs play a very important role for making a successful VoIP session. They are responsible for the modulation, compression and packetization of the obtained voice sample. Voice codecs came into existence when the G.711 [5][6] was devised. Different codecs use different modulation techniques and bit rates.

The time sensitiveness of VoIP traffic makes it compulsory for a voice sample to reach the receiver within 150 ms of its creation in order to be acceptable for most of the user applications [7]. But the processing of the analog voice signals into a digitized form and packetizing it in order to transmit it through the network requires time. The time required for the voice packet to travel through the network is also taken into account. Moreover, the voice packet received at the receiver must also be processed to make it comprehensible to the receiver. Thus end-to-end delay increases which is comprised of packetization and network delays.

The voice signal received by the user is tested for quality so as to make sure that it meets the Quality of Service (QoS) agreements. The voice can be tested for quality in two ways, namely, subjective and objective. Humans perform the subjective voice testing by listening to the voice sample, whereas, objective tests are performed by computers [2]. A common subjective benchmark for quantifying the performance of the speech codec is the Mean Opinion Score (MOS) [2]. For performing MOS test, a voice sample is given to a group of listeners. They listen to the sample and give a rating on a scale where "excellent" quality is given a score of 5, "good" a 4, "fair" a 3, "poor" a 2, and ""bad" a 1. The ratings given by every member of the group is then averaged to get the MOS. The E-Model is most commonly used for objective measurements. The basic result of the E-Model is the calculation of the R-Factor. The R-factor is defined in terms of several parameters associated with a voice channel across a mixed Switched Circuit Network and a Packet Switched Network. The parameters included in the computation of the R-factor are fairly extensive covering such factors as echo, background noise, signal loss, codec impairments, and others [8]. R-factor can be expressed by (1).

$$R = 94.2 - I_d - I_{ef} \tag{1}$$

Where, I_d is the impairment associated with the mouth-to-ear delay of the path, I_{ef} is an equipment impairment factor associated with the losses within the gateway codecs. MOS is related to R-Factor by (2) [8].

For R<0: MOS=1

For R>100: MOS=4.5

For 0<R<100: $MOS = 1 + 0.035R + 7\times10^{-6} R(R-60)(100-R)$ (2)

Till date, some explorations have already been made in studying VoIP performance under altered packet payload sizes. VoIP performance has been studied over ethernet links in [9]. Some works have also been done to improve VoIP performance by changing the access point parameters [10]. The inspiration behind using packetization as a mean for adaptive VoIP can be found in the packet format of a VoIP packet. With a closer look into the packet format, we can see that at least 40 bytes of overhead is incurred in order to transmit a VoIP packet. To be more elaborate, 20 bytes of IP header, 8 bytes of UDP header and 12 bytes of RTP header comprises of the overhead. For a G.726 codec with a bit rate of 32kbps, the codec sample size is 20 bytes and the VoIP payload size is 80 bytes [9][11]. So it is clear that a large percentage of the transmitted information comprises of the header overhead. Using, [2][12][13] the encapsulation of VoIP packets can be illustrated as in Fig. 1.

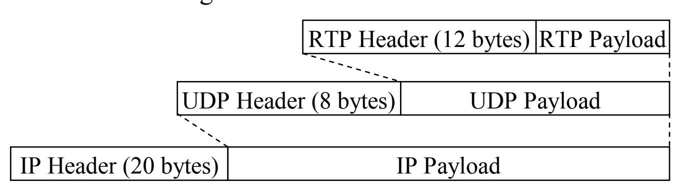

Figure 1. The encapsulation procedure of voice packets

The idea can be further illustrated by the following equations. Let the number of voice sample frames in the RTP packet be n. We know that the sample frame size for G.726 codec with a bit rate of 32kbps is 20 bytes.

So, the size of the RTP packet = $20n$ + 12 bytes (3)

The size of the UDP datagram = ($20n$ + 12) + 8 bytes (4)

The size of the IP packet = ($20n$ + 20) + 20 bytes (5)

Therefore, the payload-to-overhead ratio = $^{20n}\!/_{40} = {}^{n}\!/_{2}$ (6)

Equation (6) indicates that as we increase the number of voice sample frames in the RTP packet the payload-to-overhead ratio increases. As a result we can send more data by using lower number of bit. This reduces network congestion.

III. The Simulator

OPNET (Optimized Network Engineering Tool) [14] provides comprehensive development environment supporting the modeling of communication networks and distributed systems. Both behavior and performance of the modeled systems can be analyzed by discrete event simulation. Tool for all phases of our study including model design, simulation, data collection and data analysis are incorporated in OPNET environment. Various constructs pertaining to communication and information processing are provided by OPNET. Thus it provides high leverage for modeling and distributed systems. Graphical specifications of a model are provided by OPNET most of the times. It provides a graphical editor to enter the network and model details. These editors provide an intuitive mapping from the modeled system to the OPNET model specification. OPNET provides 4 such type of editors namely the network editor, the node editor, the process editor and parameterized editor organized in a hierarchical way. It supports model level reuse i.e. models developed at one layer can be used by another model at a higher layer. All OPNET simulations automatically include support for analysis by a sophisticated interactive debugger. Technology developers leverage advanced simulation capabilities and rich protocol model suites to design and optimize proprietary wireless protocols.

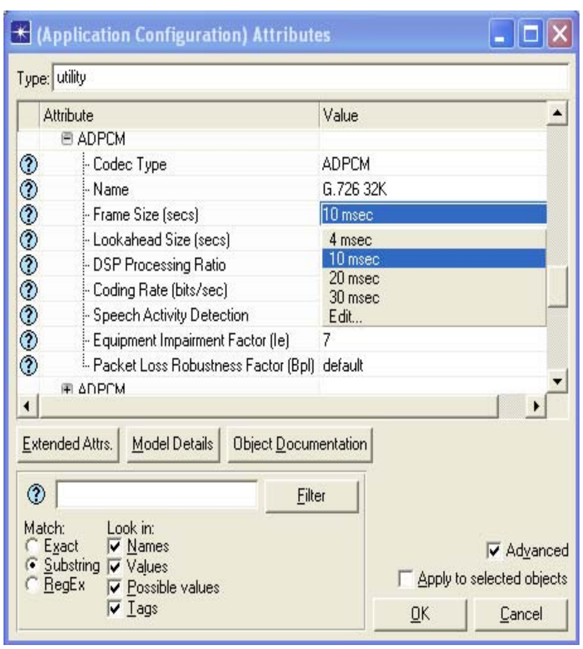

Figure 2. Screenshot of the SS node's attributes table in OPNET

In this paper, we take the advantages of OPNET Wireless modeler suites (OPNET 14.5.A). We have performed a series of simulation with the help of the WLAN model which provides very reliable results for wireless communication. Fig. 2 represents the screenshot of the SS node's attributes table providing the list of MCS that we can use for our simulation.

IV. Simulation Setup And Results

A. Simulation Setup

We have created a Wireless LAN Scenario with the help of OPNET Modeler 14.5.A. The setup consists of 8 nodes. A wireless access point connects them. The access point provides connectivity and routing functions.

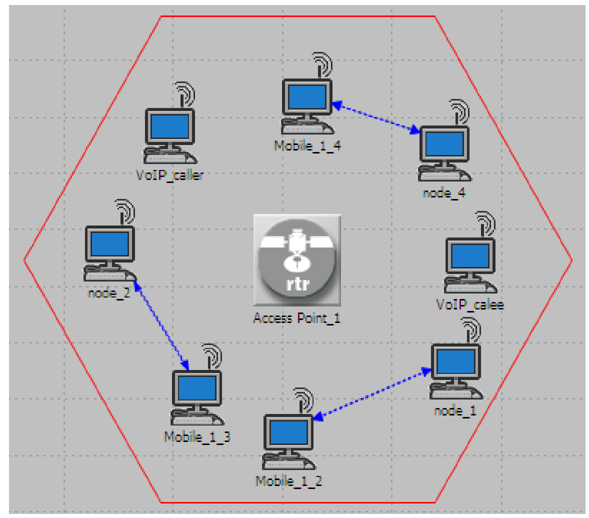

Figure 3. The Simulation Setup

It can be seen from Fig. 3 that one of the nodes acts as the VoIP caller and another node acts as the VoIP callee. The other nodes exchange information between themselves, which are not related to the VoIP session and form the network cross traffic. Earlier studies have shown that the packet size distribution in the Internet is centered about three values. To be more precise, 60% of the packets are 40 bytes, 25% are 550 bytes and 15% are 1500 bytes [15]. In the simulations, we have used packet sizes for the cross traffic in accordance with these findings. The cross traffic also pass through the access point, creating a network bottleneck. ***Thus, we create a congested network to carry out our simulations.*** Moreover, the traffic handling capacity of the access point can also be varied. For our cause, we have used 1 Mbps and 2 Mbps as the capacity of the access point so as to make it one of the deciding factors in our simulations. The red hexagon indicates the coverage area of the access point and the blue lines represent the cross load on the wireless access point. The VoIP Source uses G.726 ADPCM coder with a voice bandwidth of 32 kbps and it produces traffic at a constant rate.

We have varied the cross traffic as a percentage of the access point's traffic handling capacity and studied its impact on the VoIP session. Next, we have altered the voice packet payload size from 40 to 400 bytes by varying the number of voice sample frames in the RTP packet and measured its impact on the VoIP session for each of the above mentioned cross traffic load. Finally, we have increased the number participants of the VoIP session. Thus we have investigated how the network behaves for different levels of network congestion, packetization and number of participants in a VoIP session. In order to get further results in similar conditions but for a network with higher capacity, we have changed the traffic handling capacity of the access point from 1 Mbps to 2 Mbps.

We have measured the mean (time average) end-to-end delay and the mean MOS at the receiving end after each simulation. The end-to-end delay and the MOS at the receiver give us a clear indication about the quality of the VoIP session. Fig. 4 and Fig. 5 shows how the end-to-end delay and MOS varies if the packet payload size is kept the same but the cross traffic is increased. On the other hand, Fig. 6 and Fig. 7 shows how the end-to-end delay and MOS varies if the packet payload size is altered but the cross traffic is maintained at a certain high percentage.

B. Effect of Packet Payload Size on VoIP performance

The end-to-end delay and the MOS value provide sufficient information to assess the perceived quality of VoIP. Initially, we use only two participants in a VoIP session and we vary the cross traffic and packetization to study the performance of VoIP. It can be seen from Fig. 4 that as the network load increases the end-to end delay increases for a default G.726 32 kbps packet. Consequently, the MOS also degrades as the network cross traffic percentage increases as shown in Fig. 5.

Now, if we increase the packet payload size by increasing the number of sample frames in the payload, the payload-to-overhead ratio increases and we need to send lesser number of voice packets as already shown in Section II. As a result, we can send more useful information by transmitting lesser number of bits. Thus a better utilization of the network bandwidth is achieved. However, this method has a disadvantage. As we incorporate more sample frames to a single VoIP packet, we have to wait longer for more sample frames to be generated and thus the packetization delay increases. For delay sensitive VoIP, this is not desirable. However, reducing the network overhead helps to reduce the network delay up to a certain limit. This is because the primary network delay is caused due to the queuing delay. If the network capacity is too low, the network spends too much time to queue the packets and the network delay increases. This gives a very high end-to-end delay for small packet payload sizes. The delay decreases with increase in packet payload size up to an optimum value because the network load reduces and consequently the queuing delay also reduces. However, after that optimum packet payload size, the packetization increases as shown in Fig. 6. Hence on further increase in the packet payload size, the end-to-end delay increases. The delay characteristics of a VoIP system with network capacity of 1Mbps and a cross traffic of 80% and 5 users operating in tandem is shown in Fig.6. It is evident that the network queuing delay decreases drastically up to a certain voice sample size. This results in a fall in end-to-end delay even though there is an increase in packetization delay with the introduction of each voice sample frame to the voice packet payload. However, if more packet samples are included, the end-to-end delay increases owing to the fact that the network queuing delay does not decrease any further but the packetization delay increases with the incorporation of more voice samples. This increases the end-to-end delay. Similarly, after the optimum packet payload size the MOS decreases indicating degradation in performance as shown in Fig. 7. Hence, it is mandatory that we choose a tradeoff between the above mentioned delays, so as to get the best possible VoIP performance.

However, networks characteristics are variable. So, choosing a predetermined packetization would not be able to cope up with the changing network. An optimum packetization value must be chosen based on the network conditions so as to get the best out of an adaptive VoIP.

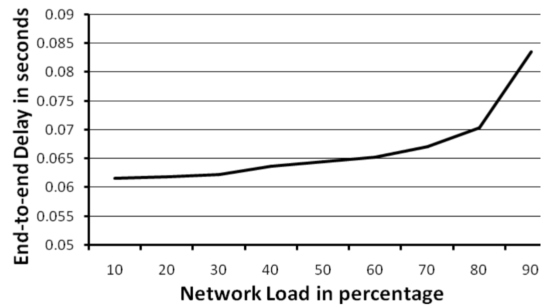

Figure 4. Increase in end-to-end delay w.r.t increase in network cross traffic

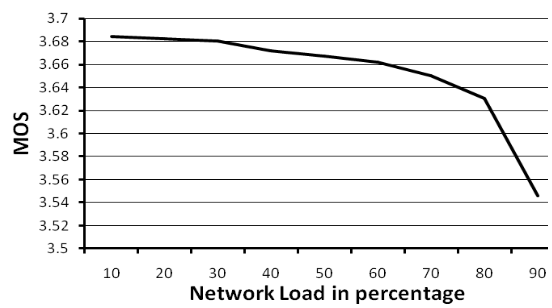

Figure 5. Decrease in MOS w.r.t increase in network cross traffic

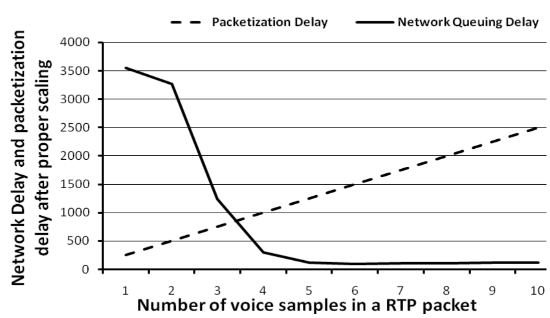

Figure 6. Effect of packetization on Network delay and packetization delay in a congested network

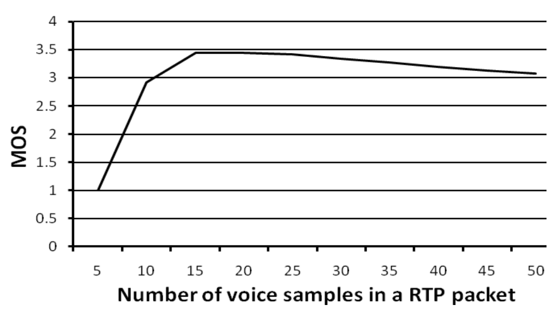

Figure 7. Effect of packetization on MOS in a congested network

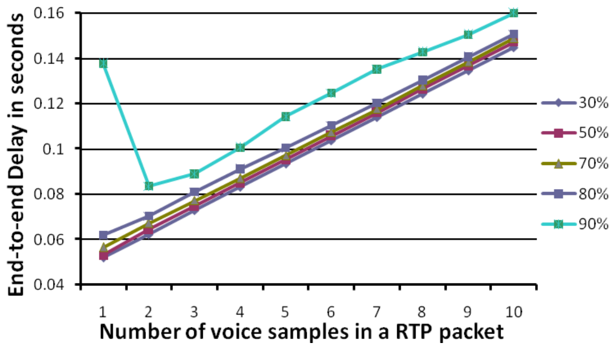

Figure 8. Effect of packetization on end-to-end delay for two VoIP users with Accesspoint capacity of 1Mbps

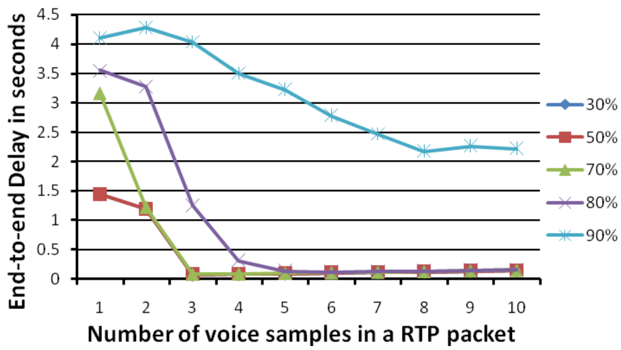

Figure 9. Effect of packetization on end-to-end delay for five VoIP users with Accesspoint capacity of 1Mbps

It is observed from Fig. 8 and Fig. 9 that we cannot choose a specific packet payload size beforehand. It is reasonable to choose an appropriate packet payload size based on the network conditions. From Fig. 9, we can see that for 30% cross traffic a packet payload size with 3 sample frames would be sufficient, after that the performance degrades. However, for a cross load of 80% a 5 sample frame packet payload size would be optimum. This statement can be further validated by Fig. 10 and Fig. 11 which shows the MOS for the similar network conditions. In both the figures, we can find that using the optimal packet payload size, we get the best MOS value and hence the best VoIP performance. It is to be noted that in case of the five user VoIP system, using 90% cross traffic gives results that are outside the tolerable range as can be seen in Fig. 9 and Fig. 11. Following the same procedure for a WLAN Access Point of traffic handling capacity of 2 Mbps, we get another set of data. Again, we see that the optimum packet payload size has changed even though the other parameters, i.e. number of VoIP users and the percentage of cross traffic has been the same. From Fig. 12 it can be concluded that by increasing the capacity of the network, the VoIP performance is enhanced even at high network congestion. Comparing Fig. 8 and Fig. 12 we can clearly see the improvement. Fig. 13 also indicates improvement in VoIP performance in comparison to Fig. 9. It is apparent from Fig. 12 that packetization does not provide any improvement over the VoIP performance. However, here also packetization helps in improvement, if we use smaller packet sizes. But, the sample size of the codec used puts a limit to the smallest possible packet payload size we can use. It is 20 bytes in case of G.726 codec operating at 32 kbps. So, we cannot use a packet of lower packet payload size.

In a 2 Mbps network, we can carry on a VoIP session between five participants even at 90% cross traffic, i.e. a heavily congested network. It is possible because, for a network with capacity 2 Mbps, even 90% cross traffic means that about 200 kbps bandwidth is still available. This bandwidth is sufficient for holding the VoIP streams produced by five participants. However, for a network with capacity of 1 Mbps, 90% cross traffic leaves only about 100 kbps which is not sufficient for holding the VoIP stream produced by five participants. So, the VoIP QoS goes below tolerable limits. Lastly, the use of altered packet payload size has also helped improve the quality of the VoIP traffic. It can be verified by looking at Fig. 14 and Fig. 15 where the MOS values are plotted using the same network parameters. Table-I illustrates the optimum number of voice samples for our simulated network.

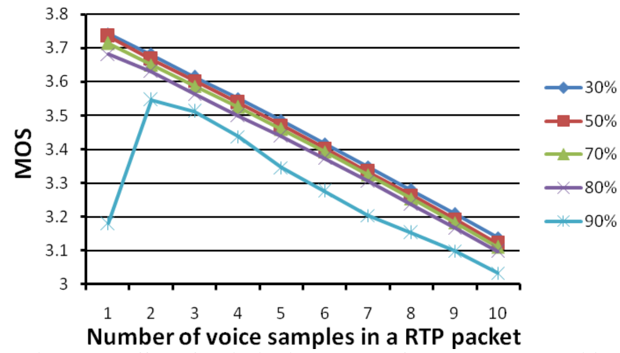

Figure 10. Effect of packetization on MOS for two VoIP users with Accesspoint capacity of 1Mbps

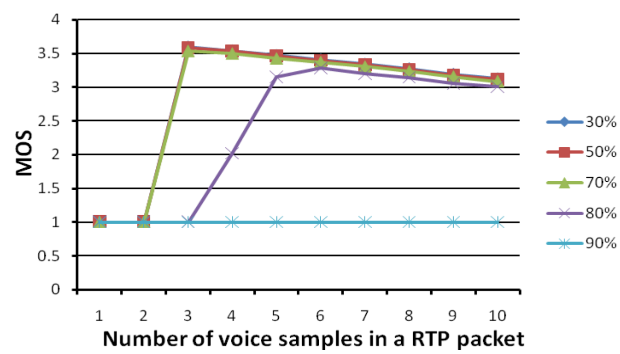

Figure 11. Effect of packetization on MOS for five VoIP users with Accesspoint capacity of 1Mbps

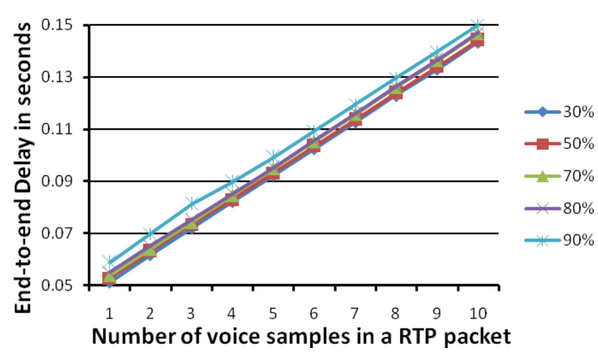

Figure 12. Effect of packetization on end-to-end delay for two VoIP users with Accesspoint capacity of 2Mbps

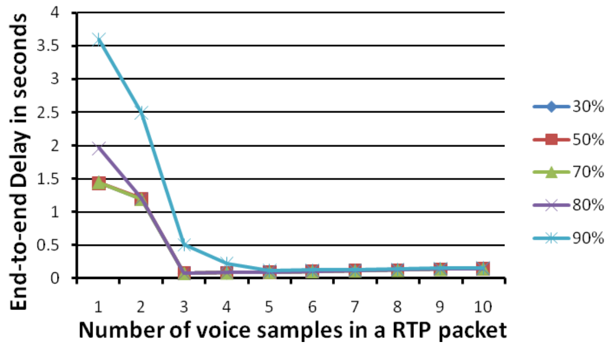

Figure 13. Effect of packetization on end-to-end delay for five VoIP users with Accesspoint capacity of 2Mbps

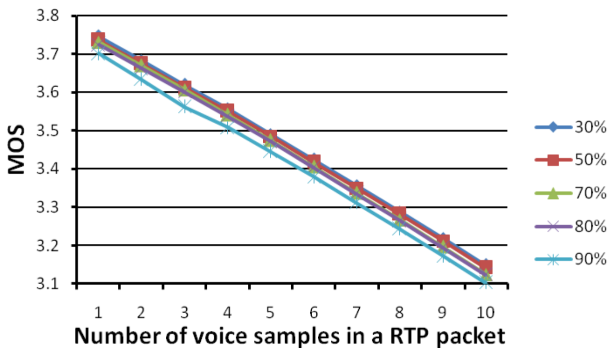

Figure 14. Effect of packetization on MOS for two VoIP users with Accesspoint capacity of 2Mbps

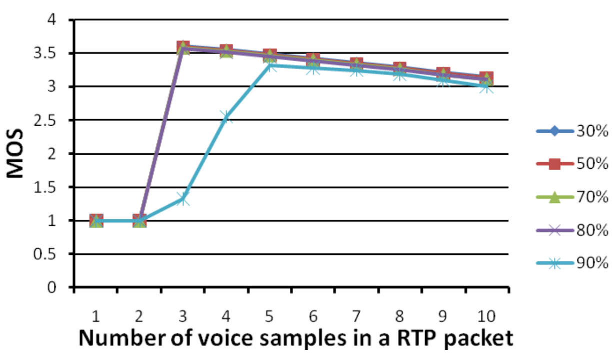

Figure 15. Effect of packetization on MOS for five VoIP users with Wireless Accesspoint capacity of 2Mbps and by varying cross traffic

TABLE I. THE OPTIMUM NUMBER OF VOICE SAMPLE FRAMES TO BE USED FOR DIFFERENT NETWORK CONDITIONS

% of cross traffic	**Capacity of the Network**							
	1 Mbps				***2 Mbps***			
	2 users	***3 users***	***4 users***	***5 users***	***2 users***	***3 users***	***4 users***	***5 users***
10	1	1	2	3	1	1	1	3
20	1	2	2	3	1	2	2	3
30	1	2	2	3	1	2	2	3
40	1	2	2	3	1	2	2	3
50	1	2	2	3	1	2	2	3
60	1	2	3	3	1	2	2	3
70	1	2	3	3	1	2	2	3
80	1	2	3	6	1	2	2	3
90	2	6	9	N/A	1	2	3	5

V. PROPOSED ADAPTIVE ALGORITHM FOR EFFICIENTLY USING THE PACKETIZATION SCHEME

From Table-I, we can observe that the optimum number of voice sample frames varies depending upon network cross traffic, capacity of the network and number of VoIP users in the session. So, it is clear that the use of altered packet payload size is beneficial. However, a tricky question arises as to when one should increase or decrease the packet payload size in order to get the best results. The network is always variable. Normally, the desired QoS negotiation is done during connection setup phase and the network guarantees this quality, barring catastrophic failures [16]. This method has a drawback. Suppose, a user connects when the network is congested, then a lower QoS is guaranteed. Afterwards, if the network becomes free, the user will still get the lower QoS, which is not desirable. On the other hand, if the network is free at the time of connection, the user is permitted a higher QoS. However, if the network becomes congested at a later stage, the network will have to turn down many users who want to join the network. Our packetization scheme, aims to maintain the same QoS even if the network condition changes by altering the packet payload size. However, this adaptive approach of changing the packet size, based on network delay may lead to a certain problem. Varying the packet payload size imposes a change in end-to-end delay which may again trigger the change in packet payload size and so on, thereby resulting in oscillation. The problem of any such possible oscillation has been taken care of in our proposed algorithm as discussed in *Step 8* below. The algorithm that we propose based on the simulated results can be summarized as follows. It has also been illustrated in Fig. 16.

Let the delay inferred from the latest RTCP RR be d_{last} and the delay inferred from the last but one RTCP RR be d_{prev}. d_{last} and d_{prev} should be updated on getting the RTCP RR every time. The steps of the algorithm are as follows.

1. When a user joins a VoIP session, first negotiate the QoS standards, and start with the optimum packet payload size for the negotiated codec.

2. From the RTCP RRs, estimate the network congestion by using the packet loss and delay jitter values.

3. If the network condition changes or a new user joins the VoIP session, proceed to step 4 else go to step 10.

4. If $d_{last} < d_{prev}$ and number of codec sample frames present in the RTP packet payload > 1then decrease the packet payload size by decreasing one codec sample frame and go to step 7.

5. If $d_{last} < d_{prev}$ and number of codec sample frames present in the RTP packet payload = 1, then go to step 10.

6. If $d_{last} > d_{prev}$ then add another codec sample frame to the RTP packet payload else go to step 10.

7. Wait for another RTCP RR.

8. If the observed end-to-end delay at the receiver for the current packet payload size is lower than the observed end-to-end delay at the receiver for the previous packet

payload size accept the current RTP packet payload size and set $d_{prev} = d_{last}$, d_{last} = current delay else revert to the previous packet payload size and another RTCP RR is examined to eliminate any possible oscillation.

9. If the new RTP packet payload size has been accepted go to step 4 else proceed to step 10.
10. Repeat from step 2 until the user is disconnected from the network.

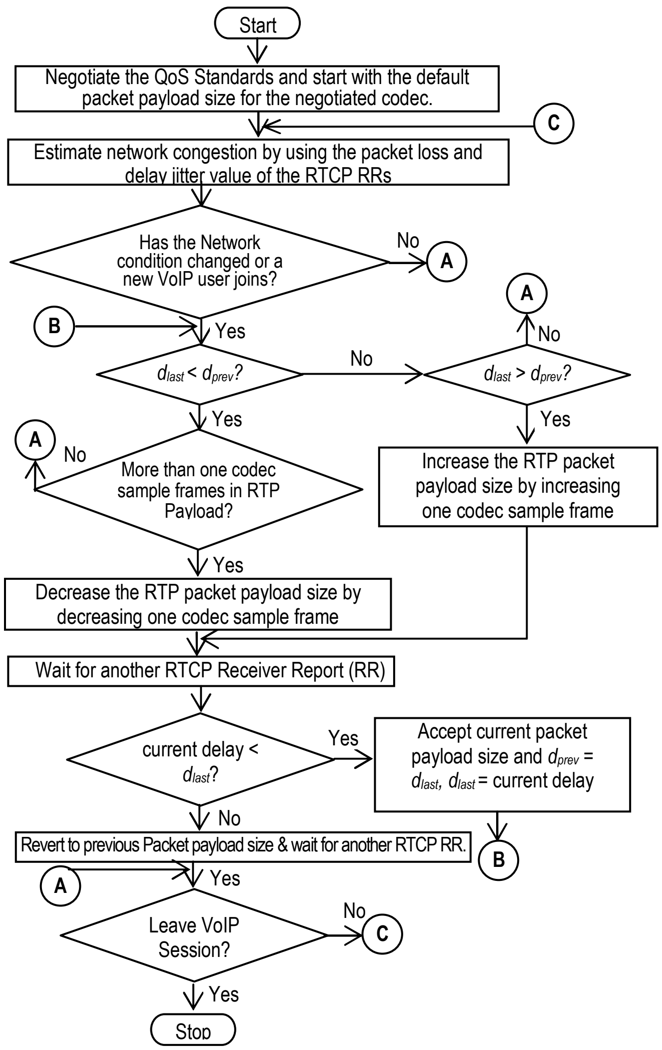

Figure 16. Flowchart describing the Proposed Algorithm

VI. CONCLUSION

In this paper, we studied the VoIP performance in a wireless congested network to maintain QoS with different packet payload sizes without changing the effective transmission rate. The procedure does not induce a drastic reduction in the MOS value of the call that leads us to the inference that there is no noticeable change in the call quality. Therefore, it is expected that the procedure will remain transparent to the user. However, one has to be careful in choosing the optimum value for efficient communication. Surpassing the optimum value and taking larger number of sample frames would lead to higher packetization delay. As a result, the end-to-end delay will increase and quality of the call will be lowered. From the observation of simulation results we propose an algorithm that would improve VoIP performance to a better extent in changing network conditions. We are on the process of implementing the proposed algorithm in a real life wireless VoIP test bed.

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