AN IMPROVED VOIP CONTROL MECHANISM FOR VOIP OVER WIRELESS NETWORK

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ABSTRACT.: *Voice over IP (VoIP) is becoming as a common application in wireless local area network. It uses to transfer the voice over Internet Protocol networks. Furthermore, it used over 802.11 standards, it is becoming greatly prevalent, in spite of these features, it stills suffer from many retractions like delay in reaching to destination and packet loss. So that, the main reasons for regression of voice quality occurred as a result voice loss and data delay. However, the traffic problem is considered one of the main technical challenges of VoIP. since the current wireless rate adaptation algorithm for having been used to enhance the delivered packets per time slot rather than the quality of the delivered voice. this research, study the main features of VoIP problems and the constraints of previous algorithms which deal with the adaptation of the data rate, and then improve the quality of transmitted voice over wireless area networks by improving the current algorithms of adaptation of the data rate. The proposed algorithm was used to avert the main physical channel problems which are: interference and fading .this paper calculate the results of the proposed algorithm By using MATLAB and the proposed algorithm exceeding other algorithms and many performance parameters*

Keywords: VOIP, wireless network, data rate algorithms.

INTRODUCTION

VoIP is an innovation for transferring voice, fax and videos over the data packets exchanged between systems. by utilizing VoIP, voice data is changed over into digital form and sent over the Internet, and changed over once again into analogue form in the wake of achieving the telephone recipient at the opposite end. VoIP is usually used in foundations, home systems and business. It is the main contender to the regular PSTN regarding effectiveness, cost and quality. It began with influencing call from PC to PC at that point to advanced to influencing calls from PC to telephone to and telephone to PC by means of passages now telephone to telephone. It has been acknowledged broadly by the business group as utilizing this innovation the entire communication framework can be given over existing IP systems to a small amount of cost of the customary telephone. Consequently, VoIP gives the convergence among voice and information systems. However, in every one of these focal points, there are a few issues as the call is routed over the IP system which is a good attempt. In ordinary call communication the physical channel is devoted but in VoIP the telephony is routed by using the IP system . so that consequently shares the downside of a packet exchanged technology. i.e. there is no ensured guaranteed of QoS. IP systems were not designed for voice communication but for information and they depend on the Best attempt rule which implies that data is not ensured to be delivered to the receiver. In this way the protocols of WLAN are in charge of the dependable transmission of information. For real-time requests information, the unwavering quality instruments are just too expensive regarding the delay. Also, an IP system is shared asset used by various applications that go after access to the channel. This opposition can prompt bottlenecks, delays or even lost packets which are resolved to ongoing applications. In order to obtain the end goal of VoIP and to make VOIP work satisfactorily, a few necessities must be satisfied or as there are requests on the Quality of Service from the system. The voice stream must not endure a delay in transmission higher than (150) ms, including preparing delays included toward the end frameworks in addition to the system's idleness as this would reduce the intelligence of the discussion. Moreover, contingent upon the codec used to change the analogue form into a digital form of packets a level of the lost packets must be reserved under a specific least, If this isn't planned it might be difficult to reproduce the voice at the receiver (listener) in a complete way. There are many technologies that give the required Quality of service by VoIP for particular networks but may don't bolster them.

Difficulties in VoIP over Wireless Local Area Network (WLAN)

There are two main difficulties for VoIP over Wireless Local Area Network. the first one is the means by which to expand the capacity with respect to voice clients. Initially, intended for data traffic, the Wireless Local Area Networks encounter bandwidth wastefulness when providing voice traffic because of the great overhead of The second difficulty is quality of service preservation for voice clients. the traffic of Voice is very critical to packet delay and jitter. In Wireless Local Area Networks(LAN), traffic of VoIP might intervene by other traffic of data, resulted by delay difference. In this way, it is important to upgrade Quality of Service to improve abilities of Wireless Local Area Networks.

Main features of VoIP Traffic

Not at all like the commonplace target of non-continuous information movement, i.e., augmenting its throughput, continuous VoIP traf_c requires ensured Quality of Service. Going for provisioning high QoS Voice Wireless Local Area Network
benefit, this paper first contemplates the qualities of VoIP traffic, and at that point talk about the way to assess its Quality of Service.

A- *Voice codec delay:* in VoIP, an analogue signal is divided to samples and enter to encoder by using many codecs like ITU-T G.711/729 [6, 7], or (iLBC) [20], into a digital bit stream. periodically, The encoded voice is putting in packets and then is sent to the MAC as encoded frames. Without quiet concealment, which distinguishes noiseless terms of human discourse and does not create any voice packets amid that interim, voice traffic is essentially c (CBR) activity, i.e., voice packets of a settled size are created and transmitted occasionally. After packet reaching to the receiver through networks, each packet of voice exhausted a various delay, so that, the interval of inter-packet for the delivered voice traffic differs over time, which is called (*jitter*). In order to recompense the jitters, a receiver of VoIP utilizes a de-jitter store or (buffer), which adds a number of packets and then after that begins yielding them with a steady between parcel interim. At that point, the VoIP parcels are de-packetized, changed over to a simple voice flag, and played back to an audience. In outcome, a voice flag encounters end to end delay, which signifies the inertness that a voice flag takes from a mouth of the speaker to an audience's ear, comprising of packetization, preparing, de-jitter , wireless delay /wireline delays, and so on... It ought to be noticed that each voice packets, in the end, encounters a settled postponement, the postpone which the very first voice bundle encounters, due to the de-jitter support. A voice packet encountering an over the top delay, i.e., a postpone longer than the fixed delay, can't be played back to the audience, and is comparable to a parcel misfortune. G.711 codec, which is the easiest yet generally utilized voice codec, creates 64 kb/s voice movement with a consistent size of packets and interim [18].

B- *B- VOIP Performance metrics:* Recommendation of ITU-T G.107 determines E-design model, which gives a helpful quality metric. On account of voice codec G.711, R-score is communicated by the accompanying disentangled as shown in equation (1) [8]:

R value = 93:2 -Gdelay - Gloss; (1)

Given that Gdelay and Gloss represent the main factors which occur as a result of an end to end delay and packet losses of traffic of VoIP , respectively[9], and when the value of R is > or = to 80 is quality of the transferred voice.

Literature review

As the Voice over IP in wireless local area network service is becoming common, there have been many researchers in this fields, this research, mainly focus on the rate adjustment case of the VOIP so as to improve the Quality of Service. The main studies that study this subject as the following:

Lacagge *et al.* [1] presented a modified adaptive algorithm which adaptively controls the success value of data. Depending on Automatic Rate Fallback algorithm, the proposed algorithm moderates the over the head of probing value by increasable expanding the delivering threshold as a probing value drops.

Kimm et al.*[2]* presents collision aware adaptive rates mechanism that can experimentally recognize the physical channel faults, so that the data rate can be minimized only after a physical channel fault [3]. In this mechanism, a wireless local area network gives request to send transferring after a frame transferring error. Depended on the transferring the calculated results of the request to send packets and the packets frames will follow the request to send packets transferring, the Wireless Local Area wireless Network can tell whether the previous frame loss could because of physical collisions or channel errors.

collision aware adaptive rates is enhanced to fastly react to the physical channel dynamics in Vermma study , by adapting the resetting base of the calculating counter of automatic rate feedback[4,5]. In automatic rate feedback and collision aware adaptive rates, the success counter, which indicates the number of sequential successful transferring, gives the value of zero when frames transferring fails. In the gathering of these two algorithms, the counter has reset after every two sequential failures, which means that it reset after the packet frames error is occurred because of physical channel errors.

SIMULATION AND ASSUMPTIONS

To check the performance of a proposed algorithm, the system has implemented the solution by using MATLAB, the system assumes some assumption which are:

- The frames of the transferred data have been transferred without any prior ACK packet. Because it depends on Spanning Tree Algorithm STA. this packet is polled, then it either sends a half of the frame number or it sends an empty frame if there is no data needs to send.
- The sender always achieves the initial back- down before starting with the process of frame sending.
- *Bit Error Rate* probability for all bit error that is supposed to happen because of the physical channel noise which includes (fading, multipath, and interference).

Improved VOIP control mechanism

This paper proposes an improved VOIP control mechanism in order to enhance the quality of the received voice. the proposed algorithm works as follows: as most multiple poll algorithm, general surveying data is sent just once toward the start of conversation free period (CFP) then each STA recover its comparing surveying data. Besides, each STA just transmits when it gets various straightforward survey messages (piggybacked with affirmation when important) that are determined in the multiple poll message, the multiple poll component keeps up the request of channel access of STA. through the number of basic survey messages, a STA ought to check before transmission.

For a traditional single- polling system supposing that: If there is M users, for every period time of 450 msec, So that the total transferring period will be equal to

450* M

So for 30 users, the transferring time will be = 450* 30= 13500 msec

But, for the multiple- polling the total transmission time is calculated as the following:

Size of multiple- polling frame multiple by the number of users

If the user is 30 so the total transferring time will be equal to:

192+ (690*8)/2= 2779 msec

We can conclude that it is clear the Total transferring time in multiple polling mechanisms is minimum than that in single - poll mechanism. For the poll size 30, the transferring period for single -poll is around 13500ms whereas it is 2779 sm in multiple- poll mechanism.

The efficiency of single and multiple mechanisms is checked. The system efficiency has been calculated by using Equation.2

ή= K/N (2)

Given that:

ή = efficiency

K = the number of data bits in the mainframes which sent successfully

N= the period for each poll in the frames

For single polls, the STA value participates in creating frames of data in the event that the STA effectively gets a CF Polls outline and has pending information edges to be effectively transmitted. The surveyed STA may endure the edge mistake because of the obstruction from outer STA. So that the (K) value can be calculated as shown in equation (3)

K= (1- n)*(frame no. /2)* G – error data)* (1-error of single polls)

The surveyed STA will give a reaction to the PC after a SIFS period for an effective CF-Poll. In the event that a surveyed STA does not react to the PC after a PIFS period for a flopped CF-Poll, the PC assumes control over the channel control and may send the following CFPoll outline. In the HCF, the RTS/CTS casings ought to be traded before the information transmission to keep the obstruction from different STAs. Since the HCF is being substituted for the PCF in the 802.11E, we consider the Single Polls of the HCF in our examination. So that the (N) value is calculated by using equation N =(N-1)+ (N of signal poll + N of the previous poll)+ (of signal poll+ N of current poll)*(1- β)* no. of frame)/2*(N data + N frames) *(1- error of the single poll)

RESULTS

The proposed improve VOIP control algorithm has compared with Lacagge *et al. algorithm which* presents a modified adaptive algorithm which adaptively controls the success value of data protocol as mentioned in previous studies, the main parameters which have been calculated are: number of dropped packets, efficiency and delay as following:

This number is little in multi poll and rises somewhat with the size of the survey. The variety of the survey measure in the two calculations ranges from 3 to 9 parcels. which can essentially affect the nature of the voice. Value dropped packets As appeared in Table 1 and Figure 1, the value of dropped parcels ascends with survey estimate for the two calculations. As the extent of the survey expands, interface disappointment likewise increments. Toward the beginning of the reenactment, this number is little, however, it ascends to half of the starting an incentive as size increments to 21. The ascent in the quantum of dropped bundles is identified with the enormous size of the survey and expanded connection disappointment.

Table 1: single poll and multi poll Number of dropped packets versus poll size

Size of poll	Single poll	Multi poll
2	14	12
5	17	13
8	19	15
10	20	15
14	22	17

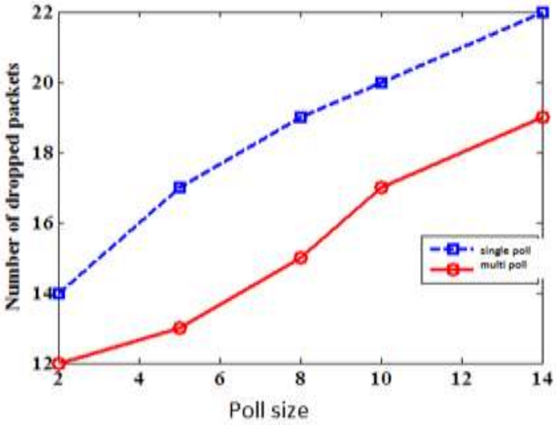

Figure 1: single poll and multi poll Number of dropped packets versus poll size

Efficiency

As shown in Table 2 and Figure 2, the effectiveness ascends with survey measure. In the two models, the overhead from the two calculations is viewed as high. Toward the beginning of the reproduction, this distinction in proficiency shows a reasonable addition, however, when the survey estimate touches 9, this esteem increments by 0.2 to 0.4. The proposed numerous survey calculation demonstrates a lower efficiency esteem contrasted with the single survey and changes from using different survey that reaches from 0.7 to 0.13.

Table 2: single poll and multi poll efficiency versus poll size

Size of poll	Single poll	Multi poll
2	0.77	0.84
5	0.87	0.88
8	0.88	0.91
10	0.89	0.92
14	0.91	0.96

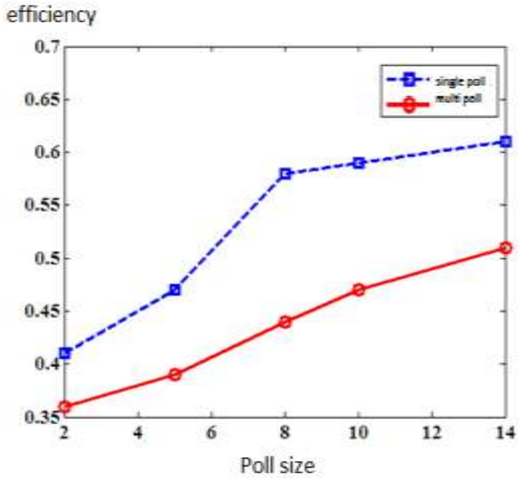

Figure 2: single poll and multi poll efficiency versus poll size

delay

the delay was linearly greater as the size of poll increased for both algorithms as presented in 3 and Figure 3. At the start of the simulation, the lowered delay has registered. After a certain duration of time, because of the increased size of the poll, the delay is also extended in the whole network. Figure 3 illustrates that the delay of the entire system was slightly longer in the (2 to 8) poll size, but this value went up significantly as the size of the poll were increased to 12. The multi-poll shows a smaller delay than single poll between 0.037 and 0.077 sec.

Table 3: single poll and multi-poll delay versus poll size

Size of poll	Single poll	Multi poll
2	163.81	110.33
5	187.16	120.25
8	198.23	125.24
10	207.02	135.15
14	222.15	171.22

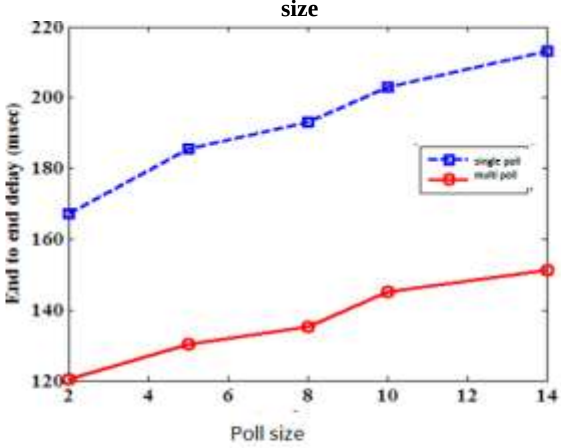

Figure 3: single poll and multi-poll delay versus poll size

CONCLUSION

Because of the constraints of single poll previous work, the 802.11e presents multiple polls which provides a great number of voice calls in comparison to single poll mode. But multiple polls also have some constraints. Because it needs more time to recover the great overheads which result from polling. The proposed algorithm can use to improve the reliability, efficiency and reduce the end to end delay and in order to cope with the unobserved hop problems.

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