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A PROPOSED AGGREGATION SCHEME TO IMPROVE THE PERFORMANCE OF VOIP OVER WLAN

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ملخص البحث:

تطبيقات نقل الصوت عبر الانترنت عبر الشبكات اللاسلكيه المحليه أصبحت من التطبيقات الهامه للأنترنت. على الرغم من ذلك يوجد مشكلتان مهمين، التعامل الغير فعال مع أى تأخير يحدث لحزم الصوت و الأ داء الغير مرضى لتطبيقات نقل الصوت عبر الأنترنت فى حالة وجود تداخل مع أحد التطبيقات الأخرى. السبب فى المشكلتان هى كبر حجم معلومات حزمة الصوت والتى تزيد بالحمل الذائد لطريقة الدخول على الشبكه. في هذا البحث، تم أقتراح خطه للدمج لتحسين أداء الشبكات اللاسلكيه المحليه للمعيار IEEE 802.11 عند أستخدام تطبيقات نقل الصوت عبر الانترنت، وذلك عن طريق دمج اى عينة صوت حدث لها تأخير مع العينه الجديده فى نفس حزمة البيانات، الخطه الجديده تعمل كأمتداد لمشفرات الصوت ولذلك فهى لاتحتاج إلى تغيير المعيار IEEE 802.11. وعن طريق محاكاة الشبكات وجد أن الخطه المقترحه حسنت قدرة الشبكه على التعامل مع مشكلة تأخير حزم الصوت بالأضافه إلى تحسين سعة نقل الصوت بحوالى ٢٧ ٪ مع قبول نوعية الخدمه.

ABSTRACT

Voice over Internet Protocol (VoIP) over a wireless local area network (WLAN) becomes an important Internet application. However, two major technical problems exist, inefficient treatment of any delayed voice packet and unacceptable VoIP performance in the presence of coexisting traffic from other applications. The reason for the two problems is the long voice packet header which is increased by channel access overhead. In this paper, an aggregation scheme is proposed to enhance the VoIP performance over the IEEE 802.11 WLAN, by aggregating any delayed voice samples in a single packet, the proposed scheme works as an extension to the voice codec, so it doesn't need to change the IEEE 802.11 standard. By simulation, the proposed scheme enhanced the capability of the network to deal with the delayed voice packets problem and increased the network capacity to around 27% with acceptable QoS.

KEYWORDS: VOIP, INFRASTRUCTURE IEEE 802.11, AGGREGATION, NETWORK CAPACITY, THROUGHPUT, PACKET LOSS RATE, ACCESS DELAY, JITTER, TCP.

1. INTRODUCTION

Voice over Internet Protocol (VoIP) is a way of making phone calls through the internet. Over time it gained popularity as a way to enable direct and free communication over the internet to people from all over the world. Any VoIP translates the analog speech from the user and converts them to digital form, this process done by the voice codecs. The codecs convert the audio signal into compressed digital form for transmission, and then back into an uncompressed audio signal for replay. Every codec transmits large number of packets (50 packets per second). Each packet contains a small amount of bytes that can be reconstructed into voice. Each packet in a codec covers a short period of time from 10 to 30 ms corresponding to the kind of codec [1]. Table 1, lists the attributes of GSM 6.10 voice codec [2] which will be used in this paper.

Codec	Bit rate (Kbps)	Framing interval (ms)	Payload(Bytes)	Packets/sec
GSM 6.10	13.2	20	33	50

Table 1: attributes of GSM 6.10 codec.

At the same time, the wireless local area network (WLAN) gains a very large number of users in the resent years, due to its convenience, mobility, and high-speed access. The most widely used standard in WLAN products is the IEEE 802.11 and its generations [3]. The basic building block of an IEEE 802.11 network is the basic service set (BSS). IEEE 802.11 standard defines two types of BSSs: independent and infrastructure. Stations in an independent BSS communicate directly with each other. In the other hand, stations in an infrastructure BSS communicate with each other via an access point (AP), therefore all traffic to and from a station must flow through the AP. This paper focuses on infrastructure BSS. The IEEE 802.11 standard defines two layers. The first layer is the Physical layer (PHY), which specifies the modulation scheme used and signaling characteristics for the transmission through radio frequencies. The second layer is the medium access control (MAC) layer, which determine how the medium is used. The MAC layer has two modes of operation [4]. The first mode is Point Coordination Function (PCF) which was designed to support voice and video transmission but it is not widely used and not supported in most 802.11 products. The second mode is Distributed Coordination Function (DCF) which based on Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) mechanism. The CSMA/CA is a very efficient mechanism, to reduce happening of collisions between any two or more nodes want to transmit data or voice packets in WLAN. To work this mechanism there are always waiting times for all nodes before and after it can transmit a single packet as shown in figure 1.

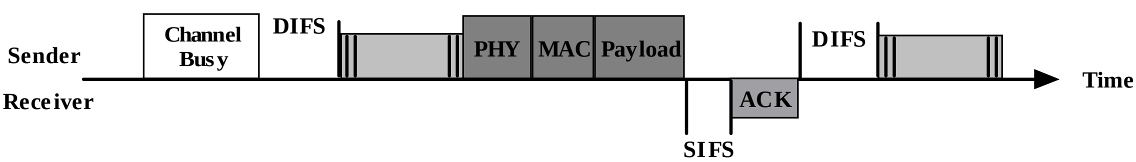

Figure 1: Basic access procedure for IEEE 802.11.

Before transmission a station will randomly choose a backoff time with a number of time slots. The station will decrease the backoff time counter while the channel is idle after a DCF interframe space (DIFS) interval, and pauses the timer if it senses that the channel is busy. When the backoff value reaches zero, the station will transmit its packet. Then the station will wait for the receiver to send back an ACK frame after a short interframe space (SIFS) interval. If it does not receive the ACK, the station assumes that the packet has been lost due to transmission errors or a collision. Thereafter it generates another backoff time and retransmits this packet following the same procedure as before.

When VoIP application is used over a WLAN there will be some challenges because the IEEE 802.11 standard was originally designed for data transportation, not for delay sensitive applications such as VoIP. The number of sessions that can coexist in an IEEE 802.11 VoIP network is smaller than expected when the rates of the network to the rates of the codecs are compared. Also, there is a famous problem which called the delayed packet problem. The delayed packet problem occurs as follow, when the codec generates a voice sample and sends it for transmission, the payload still in the transmission queue until the station get the right to send. If the transmission is delayed for any reason such as, a collision or a crowded traffic in the network due to any interference from other application or even a packet loss, which will need a retransmission, and the time for a new voice sample comes. The codec will generate another sample and send it for transmission. Therefore in the transmission queue there will be two packets need to be transmitted. Each one of them carries a header longer than the payload itself. At the IP layer the voice packet consists of 40 byte header and a payload ranging from 10 to 30 byte depending on the kind of codec. Also each one of them will wait in case of transmission for CW and DIFS times and if the first packet was sent successfully, the station will wait for SIFS before receiving the ACK. And then the station will wait for other stations to transmit their packets before the station transmit the next packet following the same procedure again. That way of dealing with the delayed packet is inefficient at all and will cause an extra delay for any left packets in the transmission queue without any clear reason from the view point of voice. The investigations of this paper revolve around finding a solution to the delayed packets problem, which will also reduce the effect of crowded network due to any interference traffic from other applications.

2. PREVIOUS WORKS

In this section, related work is briefly summarized on frame aggregation, in addition to other techniques that efficiently carry the voice traffic over IEEE 802.11 wireless LAN. Attention is been focused on solutions that do not require modifications on the IEEE 802.11 standard. In [5] the researchers proposed a Multiplex–Multicast scheme, in which the AP multiplexes packets from several VoIP streams into one multicast packet for transmission. The IEEE 802.11 multicast eliminates some overhead caused by the transmission of 802.11 ACKs, because the 802.11 multicast does not perform retransmission. However, for the same reason, the possibility of packet loss is very high and an additional delay is expected in composing such a composite packet. In [6, 7] two aggregation schemes were proposed, the main idea of both of them is to combine multiple data into a larger packet which exceeds the configuration packet limits, and send it by fragmentation, then reassemble it in the destination, but in both of them the IEEE 802.11 standard was modified to add identification header for identifying each single data from another. Also when there is multi voice data connection in the same station, the voice packets will suffer from extra delay caused by data packets.

In [8] the researcher uses adaptive *CWmin* control and deadline-based MAC queue management. The MAC queue drops packets that miss the preset deadline, transferring the bandwidth to in-time packets. The concern with this scheme is that it is difficult to know if a real-time packet is indeed useless at the receiver, so the dropping decision is not useful to enforce inside the network.

In [9] the researcher proposed a Multiplexing scheme for internet telephony depending on header compression in [10, 11], the idea of this scheme is to combine many packets together in the multiplexer and send them as one large packet. Next, the demultiplexer restores the original packets and sends each one to its destination, but this scheme cannot be applied in the normal network architecture, the researcher consider two WLANs in two locations connecting together.

In [12, 13] the researchers proposed a dual queue to ensure priority of the voice packets over data packets, the dual queue enhances the QoS without changing the IEEE 802.11 standard, but it doesn't enhance the total network capacity.

3. The proposed scheme

The main idea of the proposed scheme is to dynamically combine speech frames which exist in the delayed voice packets in one single packet. The explanation for wasting bandwidth in case of delayed voice packets is that, a header exists for every voice packet and each packet needs to be acknowledged separately before transmission for the next packet. Therefore, the more speech frames put into a packet, the fewer headers and ACKs require [21]. To reduce the overall overhead introduced by the header and the wasting time to wait for ACK in each time a situation of a delayed voice packet occurred, multiple voice samples can be packed into a single frame to be transmitted. Any codec originally contains a static merging technique which can control the length of the voice samples during the voice call. In the proposed scheme, the length of the voice samples dynamically changed due to the network situations. If a situation of the delayed voice packets happens, the proposed scheme will merge the old voice samples (not the all packet which contains the voice sample and the header) with the new generated one in a longer voice sample, and replace it with the old packet in the real-time queue. The dual queue strategy is used as in [11, 12]. The idea of the dual queue is to implement two queues called real-time (RT) and non real-time (NRT) queues. It classified each packet to be transmitted into RT or NRT types. The dual queue is used to give priority to the voice packets over data packets (if exist) in the same station, and also to control changing the single payload packet by the aggregated payload packet. The proposed scheme can adapt the size of the aggregated payload in the merging process. The length of the aggregated payload is dynamically specified to be the length of the delayed payloads in the real-time queue. The length of the aggregated payload is specified first, and if the size of the merged payload equals the required length, the packet will stay in the real-time queue without any replacement in the future. The mechanism of the proposed scheme works as follows:

1.When the voice codec generate a voice sample, it will be copied and put into the codec buffer, the original payload will be sent to the real-time queue for transmission.

2.When the packet sent successfully, an ACK will be received, therefore any left samples in the codec buffer and in the real-time queue will be deleted.

3.In the other hand, when the codec generates a voice sample and there was another one in the codec buffer, those samples will be merged and packed into a single packet,

4.The old packet in the real-time queue will be replaced by the new one after the merging process.

5.If the size of the merged packet equal to the specified length, the complete packet will be sent and stay in the real-time queue without any replacement later.

The proposed scheme works as an extension to the voice codec, therefore it doesn't need any change in the IEEE 802.11 protocol. Also it can work with multi VoIP connections (conference). If there will be a conference between multi voice agents, all the stations will connect to a conference bridge which works by a mechanism called unicast receive and multicast send. All the connected station will send there packets to the conference bridge, which will receive all the packets from the stations and aggregate them in a single packet for transmission to all the stations as a downstream packet. The proposed scheme can also work with other proposed ideas which don't need changes in IEEE 802.11 protocol such as [5, 8]. The algorithm of the proposed scheme illustrated in figure 2.

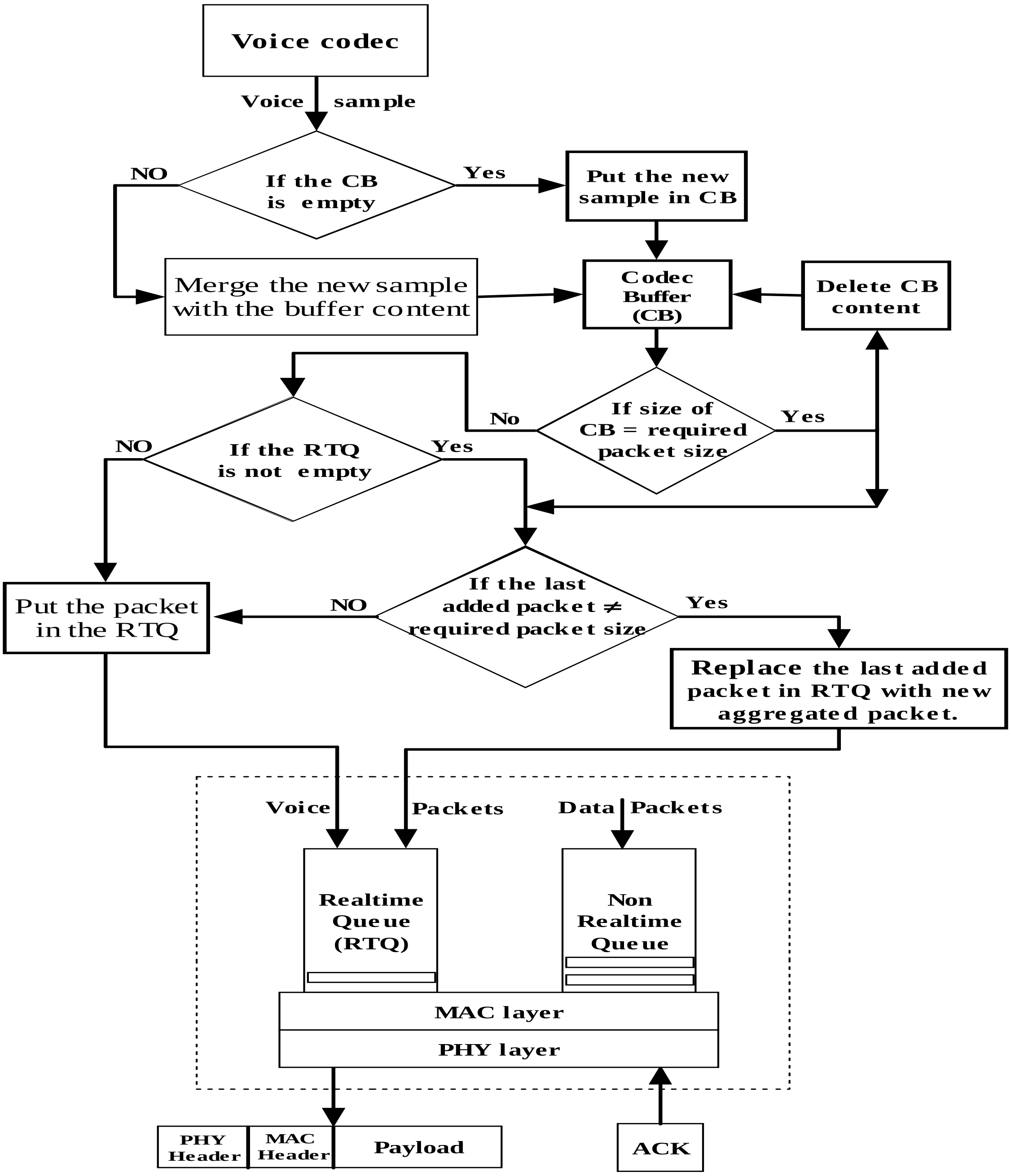

Figure 2: the proposed scheme algorithm.

4. Capacity analysis

In the capacity analysis, continuous bit rate (CBR) voice source are considered. Voice packets are generated at the voice codec rate (50 packets per second as using GSM 6.10 codec). All the formulas in the analysis here as used in [5, 13].

Let *Nmax* be the maximum number of sessions can be supported. The transmission times for downlink and uplink packets are *Tdn* and *Tup*, let *Tav* be the average transmission time of two packets. In 1 second, there are totally 1/*Tav* packets transmitted by the AP and all the stations.

$$1/Tav = \text{number of streams} * \text{number of packets sent by one stream in one second} \quad (1)$$

In the ordinary VoIP case, there are *N* downlink and *N* uplink unicast streams. On average, for every uplink packet there is a corresponding downlink packet

$$\text{Number of streams} = 2Nmax \quad (2)$$

$$1/Tav = 2*Nmax * Np$$

$$Nmax = 1 / (2 * Np * Tav) \quad (3)$$

Where, *Np* is the number of packets sent by one stream and

$$Tav = (Tup + Tdn) / 2 \quad (4)$$

For VOIP packets the total overhead time (*Tov*) for one transmission is

$$Tov = Sender\_ov + Receiver\_ov + Header\_ov \quad (5)$$

Where, *Sender_ov* is the overhead of the sender, *Receiver_ov* is the overhead of the receiver and *Header_ov* is the overhead due to the headers of UDP, RTP, IP, and IEEE 802.11b MAC header

$$Header\_ov = H_{UDP} + H_{RTP} + H_{IP} + H_{MAC} \quad (6)$$

At the MAC layer, the overhead at the sender is

$$Sender\_ov = DIFS + CWavg + PHY \quad (7)$$

Where *CWavg* is the average contention window and it is equal to

$$CWavg = slot\ time * (CWmin-1) / 2 \quad (8)$$

In the unicast transmission, the overhead caused by the receiver is:

$$Receiver\_ov = ACK + SIFS \quad (9)$$

In this case, transmission time for uplink and downlink are equal therefore,

$$Tup = Tdn = Tav = (payload) * 8 / data\ rate + Header\_ov + Sender\_ov + Receiver\_ov \quad (10)$$

The values of IEEE 802.11b parameters are listed in Table 2, assuming that GSM 6.10 is used; payload is 33-B and *Np* is 50. Solving (3, 10) *Nmax* = 11.2 sessions.

Parameter	DIFS	SIFS	Slot Time	CW min	CW max	Data Rate	Basic Rate	PHY header	MAC header	ACK
Value	50 µsec	10 µsec	20 µsec	32 slot time	1023 slot time	11 Mbps	2 Mbps	192 µsec	34 bytes	248 µsec

Table 2: Parameter values of IEEE 802.11b DCF.

To calculate the capacity in case of several samples per packet, let *Ns* be the required number of sample per packet, for original transmission *Ns* = 1, *Np* = 50. For the proposed scheme with 2 merging samples per packet, *Ns* = 2 sample per packet, therefore *Np* =25, and the payload size in this case equal 66-B, by solving [3, 10] with the new parameters, *Nmax* = 21.9 sessions. The max capacity for several numbers of samples in merging process is calculated as shown in table 3.

Scheme	IEEE 802.11	Proposed scheme with			
		2 samples	3 samples	4 samples	5 samples
Capacity (session)	11.2	21.9	32	41.6	50.8

Table 3: capacity results for original scheme vs. the proposed scheme with several merging sizes.

Framing overhead is one of the bottleneck problems; therefore enlarging the RTP payload size will increase the number of supportable calls. The system capacity in case of voice transmission must achieve some constrains which recommended by ITUT [5-3]. Therefore some important parameters must be measured to define the max capacity of the voice over WLAN with a good QoS [14, 15].

- Average one-way delay**,** is the most critical parameter for VoIP. If it is too long, the performance of the conversation flow would decrease, ITUT recommend a one-way delay of up to 150 milliseconds, but the 150 ms delay is the total end-to-end delay in one direction, therefore in the WLAN to achieve this requirement the average access delay of the voice packets must not exceeds 30 ms [5, 13].
- Packet loss rate**,** for VoIP, packet loss rates of up to 1% are generally acceptable [16].
- Mean jitter**,** the objective of this parameter is to minimize the mean jitter and keeping the packet loss rate due to jitter effect less than 1%.

When the three constrains is achieved, it is expected that, the maximum capacity will be lower than the capacity in the analysis, because the approach of creating a large voice payload will increase the access delay and the mean jitter of the aggregated packet.

5. SIMULATION

The simulation was built using Microsoft visual basic.net 2005 depending on Pythagor wireless network simulator source code [17]. Pythagor simulator performs a very detailed simulation of the MAC protocol, based on the IEEE 802.11 specification. Also it is an open source code in order to implement any modified protocol. In simulation, five parameters are measured throughput, access delay, mean jitter, packet loss rate and the system capacity. The packet loss rate is not measured separately but it is measured impeded in each parameter. The system capacity is defined to be the number of VoIP sessions that can be supported, while maintaining the packet loss rate of every stream to be below 1% (including the jitter effect) and the average access delay of every stream below 30 ms, as considered in many researches such as [5, 13, 18, and 19]. In simulation topology, it is assumed that all voice streams are between stations in different BSSs or different locations out of the local WLAN, since users seldom call their neighbors in the same BSS. Within a BSS, there are two streams for each VoIP session, the uplink stream is for voice originating from the local station to the AP. The downlink stream is for voice originating from the other side of the VoIP session to the station, which flows from the remote station through the internet to the local AP to the station as shown in figure 3.

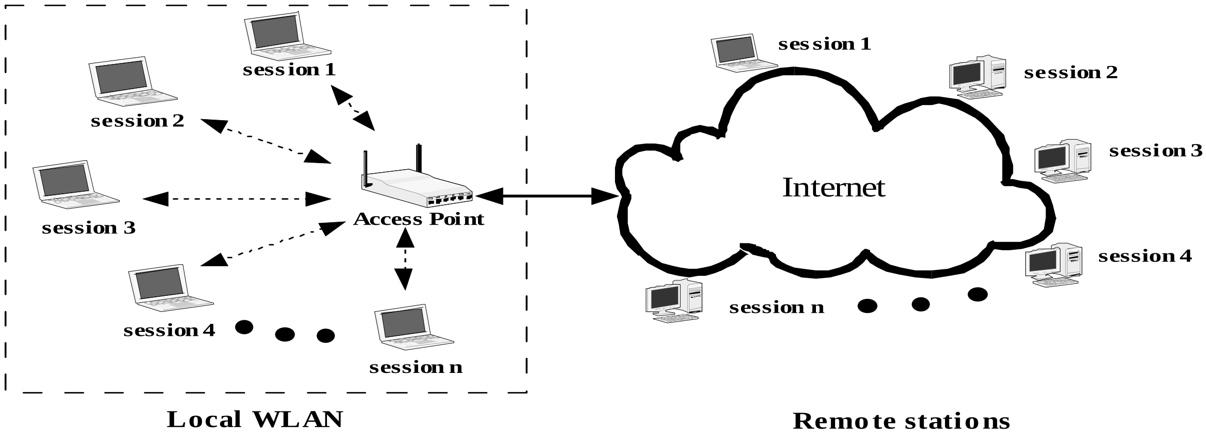

Figure 3: Simulation architecture for the infrastructure WLAN.

5.1 Capacity performance

In this subsection, system capacity is defined to be the number of VoIP sessions that can be supported while maintaining the packet loss rate of every stream to be below 1%. When the ordinary IEEE 802.11b is simulated it is noticed that. The network could handle 11 sessions with a packet loss rate does not exceed 1%. This result matches the capacity analysis very well. The number of sessions is also increased beyond the full capacity, but this leads to increasing the packet loss rate. For example when the 12th session is added, the packet loss rate jumps to around 5%. This result is due to the insufficient bandwidth to accommodate all the transmissions of the stations and the AP. However, when exceeds the system capacity, the extra traffic will be delayed, leading to a large packet loss rate.

In the proposed scheme it is noticed that, when the 12th session is added, the proposed mechanism handled any extra delayed packets in any station. Therefore, the scheme could treat with the 12th sessions without any problem. By increasing the number of sessions, the proposed scheme handled 21 sessions with a packet loss rate below 1%. Also by increasing the number of samples in merging process, the maximum capacity increased as shown in table 4.

Scheme / Capacity	IEEE 802.11	Proposed scheme with			
	1 sample	2 samples	3 samples	4 samples	5 samples
In analysis (session)	11.2	21.9	32	41.6	50.8
In simulation (session)	11	21	31	41	50

Table 4: Simulation capacity of ordinary IEEE 802.11b versus the proposed scheme with several merging sizes.

5.2 Delay performance

Beside the capacity analysis, the access delay was simulated, which is very important in VoIP packets to provide a good voice quality. The access delay of a VoIP packet is the time from when the packet is generated until it leaves the interface card. From an end-to-end viewpoint, it is essential for the local delay to be small, so that the overall end-to-end delay of a VoIP stream can achieve a good quality of service.

As a reference for delay investigations in this paper, a requirement was set that the average access delay of the downlink or uplink VoIP packets should not be more than 30 ms. This allows sample delay in the backbone network for an end-to-end delay budget of 150 ms as illustrated in [5, 20]. Figure 4 shows the average access delay for each session in case of the ordinary transmission, and in case of the proposed scheme with several packet lengths.

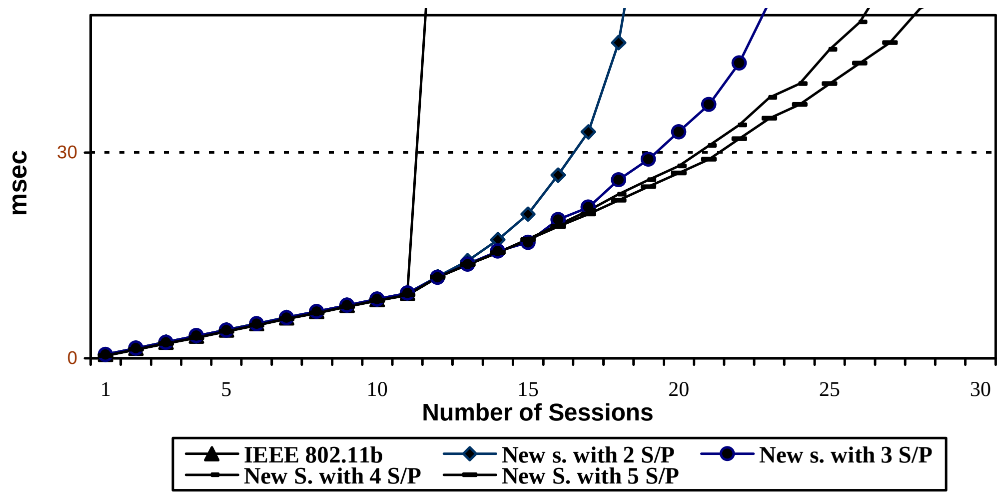

Figure 4: Access delay (ms) for ordinary IEEE 802.11b vs. the proposed scheme.

From access delay, it is noticed that, in ordinary transmission the delay still less than 30 ms as the number of sessions don't exceed the full capacity of 11 sessions. By increasing the number of sessions, the delay dramatically increased to unacceptable value. In the proposed scheme the access delay is much better due to decreasing the overhead of the total packets for each session. Also it is noticed that, the 30 ms access delay constrain cannot be achieved in the full capacity in all cases of the proposed scheme. The new full capacity due to the 30 ms constrain illustrated in table 5.

	IEEE 802.11	Proposed scheme with			
Scheme		2 samples	3 samples	4 samples	5 samples
Capacity (session)	11	16	19	20	21

Table 5: capacity after 30 ms constrain.

5.3 Jitter performance

The size of jitter buffer is a very critical performance in the RTP because any packet exceeds the jitter buffering delay it consider as a packet loss, therefore the size of the jitter buffer must set to the maximum delay jitter. Generally, the jitter buffer is ranging from 20 ms to 80 ms. Jitter buffer of 20 ms is considered low buffering size because it can drop many packets, on the other hand 80 ms buffer is a low QoS performance, therefore many products set the jitter buffer to be 40 ms [22] as it is set in simulation. Delay jitter is measured in each case of simulation. To determine the packet loss rate due to jitter effect, a histogram of a random session is measured for a random 10 second of simulation time as shown in figure 5.

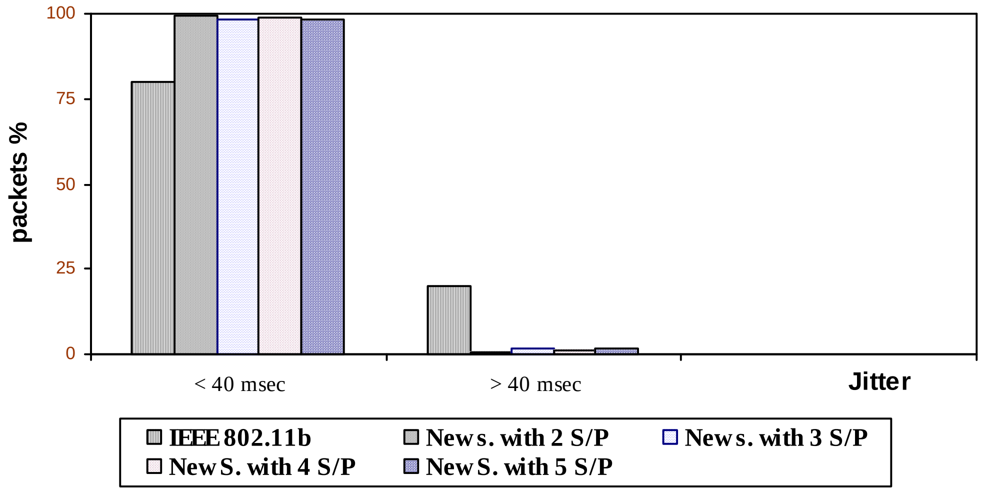

Figure 5: a jitter histogram of a random session in case of 14 sessions in the WLAN

The histogram is measured in each case of simulation until the packet loss rate due to jitter buffer effect reaches 1%. It is noticed that, the network can handle just 14 sessions in case of 40 ms jitter buffer. The lower of the capacity due to jitter effect comes from the variation of delay before and after any aggregated packet which will be at maximum. This short coming with any aggregation scheme is expected due to the variation of delay after and before the aggregation process. The final results of the proposed scheme after considering the three constrains are shown in table 6.

Scheme	IEEE 802.11	Proposed scheme with 2 samples	3 samples	4 samples	5 samples
Max Capacity (sessions)	11	14	14	14	14
Mean jitter (ms)	9.8	14.1	14.7	14.9	15.1
Access delay (ms)	10.3	16.2	15.8	15.5	15.3
Packet loss rate (%)	0.3 %	0.4 %	0.7 %	0.8 %	0.9 %

Table 6: maximum network capacity after considering the three constrains.

6. VoIP coexisting with TCP interference traffic

In this section, the proposed scheme is tested under a different traffics situation. In WLANs, VoIP may coexist with traffic from other applications. This traffic is mostly transported using TCP. When VoIP interfere with TCP there will be two problems, the first occurs at the AP for downlink streams and the second occurs when traffic at different nodes contend to access the WLAN. In most access points, all downlink traffic shares a common first in first out queue. In this case, VoIP packets intermix with TCP packets in the AP buffer, leading to the typical UDP/TCP competition problem, as noted in [23]. Therefore, in this research, a priority dual queue (PDQ) is used in the AP to decrease the effect of the TCP flow on the voice streams. On the other hand, the TCP generates two ways traffic in the WLAN. After the sender sends a TCP data packets, the receiver must be reply by sending a TCP ACK. In the WLAN, both TCP data and TCP ACK are treated as data frames.

Although the payload of TCP ACK is small, the transmission of TCP ACK can consume a considerable amount of bandwidth due to the header and other overheads. The final result of mixing VoIP and TCP on the same WLAN is decreasing the maximum number of calls can be handled. In this section, such a case is considered because it is a very common problem need to be solved, and also another challenge for the proposed scheme. The PDQ is used in which voice packets are given priority over the TCP packets within the AP buffer by limiting the number of TCP packets in the non-real time queue to be equal to 1 Mbps, or 86 packets per second. The payload of the TCP data is set to be 1460 byte. The topology of the simulation is shown in figure 6.

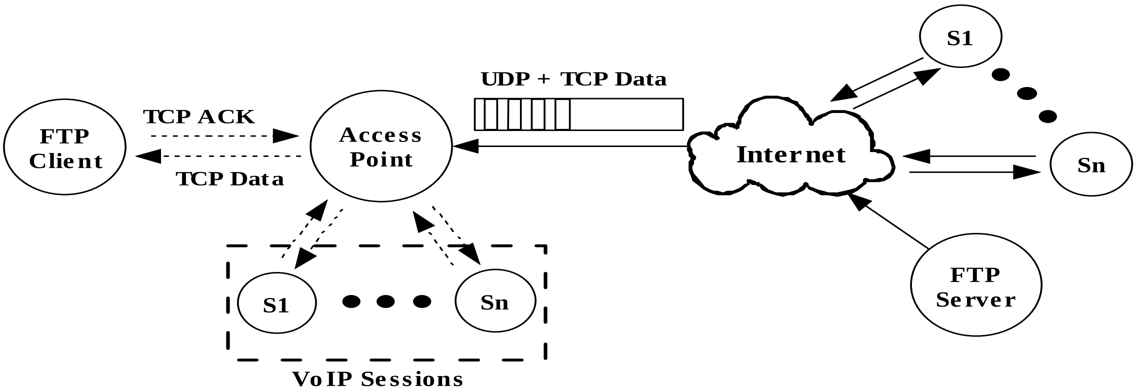

Figure 6: Simulation topology when a TCP client coexists in a VoWLAN.

A file transfer protocol (FTP) server is connected to the internet. It sends the downloading packets to a FTP client through an AP. Therefore, in the AP buffer, the VoIP packets intermix with TCP data packets. At the same time, TCP ACK packets sent from the FTP client will contend with VoIP uplink packets sent from all the VoIP clients. The proposed scheme is used with 2 samples in the merging process since it is the best characteristics. The throughput of the FTP client and the other voice clients in case of the original and the proposed scheme is shown in figure 7 and table 7.

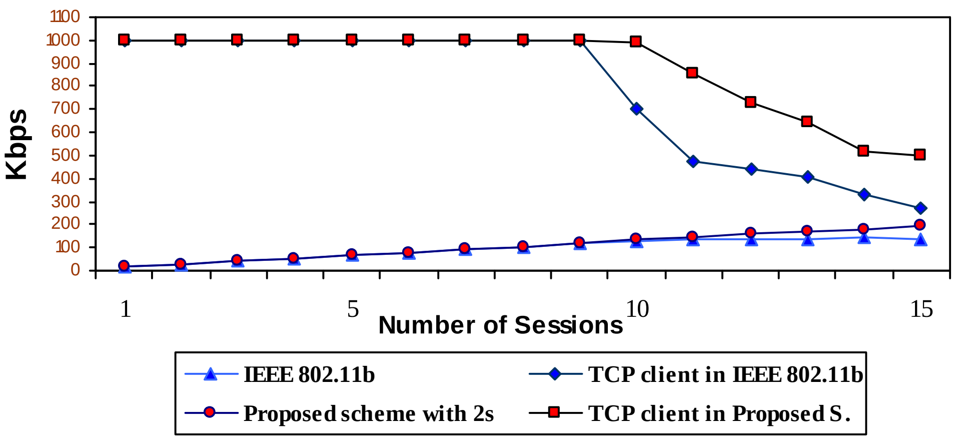

Figure 7: Throughput (Kbps) for original IEEE 802.11b protocol and the proposed scheme in case of coexisting a FTP client in the WLAN.

	Total sessions can be support	Access Delay	Mean jitter	Packet loss rate	TCP Throughput
Ordinary VoIP	8 sessions	10.8 ms	9.4 ms	0.4 %	1.00 Mbps
Proposed scheme	13 session	17.8 ms	14.6 ms	0.6 %	0.645 Mbps

Table 7: The results of the simulation considering CBR voice source with a TCP client.

The ordinary VoWLAN protocol can support only 8 voice sessions with the full throughput of the TCP client. On the other hand, the proposed scheme can support 13 voice sessions with a throughput of the TCP client a little more than the half of the required throughput. That result mean, when the TCP client share the transmission medium of the voice clients, the capacity of the voice network decreased. In case of the ordinary protocol which can not deal with any delayed packet, the total number of sessions decreased from 11 to 8 sessions, but the throughput of the TCP client still at maximum. On the other hand, the proposed scheme could deal with 13 sessions with acceptable access delay and packet loss rate. The problem in case of the proposed scheme is the throughput of the TCP client, which decreased to only the half of the required throughput. From the voice view point, the degradation of the data throughput to that value is not a big problem because the data throughput is not constrained by time as voice. Therefore the user can stay a little longer to receive his file. But in case of the voice transmission the time for sending and receiving voice packets is very important to the quality of the voice conversation and can not be negotiable.

8. Conclusion

In this paper, two critical problems in VoIP over WLAN were investigated, to overcome the problem of the delayed voice packets that VoIP calls suffer on the infrastructure IEEE 802.11 wireless LANs. And the unacceptable VoIP performance in the presence of coexisting traffic from other applications. The proposed scheme aggregates any delayed voice sample in the next voice packet. There are three constrains are set to determine the total capacity with acceptance QoS. The packet loss rate doesn't exceed 1 %, the access delay less than 30 msec and the packet loss due to jitter effect doesn't exceed 1 %. The proposed scheme with those constrains can overcome the delayed voice packets problem and increase the capacity by 27 %. Also, it can improve the performance of voice calls when VoIP exists in a network that carries TCP interference traffic. The proposed scheme can work as an extension to the voice codecs. Therefore, it doesn't need any changes in the IEEE 802.11 protocol. Also, the proposed scheme can work with any other schemes that don't need any changes in the upstream traffic such as [5, 13].

9. REFERENCES

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