Analysis of VoIP Traffic with Multiple Packet Transfer

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***Abstract*— Voice over Internet Protocol (VoIP) is a form of voice communication. Today Wireless LANs are providing a cost effective alternative for the voice communication. Initially the wireless access protocols are not suitable for the voice communication. Subsequently, new techniques are developed for the real time voice communication. In voice communication the delay is an important factor. In this paper a new technique is proposed for the voice over IEEE 802.11 WLANs. The proposed technique is compared with Modified ICF.**

***Keywords*— WLAN, ICF, MICF, delay, time slots, Buffer, 3-Buffered Packet ICF.**

I. Introduction

Voice over Internet Protocol (VoIP) is one of the most important communication technologies in the world of voice communication. VoIP is simply a method to make phone calls through the Internet. VoIP transmit packets via packet switched network in which voice packets may take the most efficient path in the network [1]. VoIP over WLAN is becoming a very efficient solution for wireless voice communications. The WLANs bypass the traditional PSTN system. The calls enter into a well connected IP network directly through WLAN [2]. Wireless LAN standard IEEE 802.11 specifies two method of wireless channel access. These are Point Coordination Function and Distributed Coordination Function [2].

DCF mode is based on random access of channel. It is best for non real-time traffic that is bursty traffic. The DCF mode works on the principle of carrier sense multiple accesses with collision avoidance (CSMA/CA) in which a host wishing to transmit a packet senses the channel to check whether it is free or not. On finding the channel free, the host waits for a random amount of time before transmitting. This waiting time avoids two hosts to start the transmission of packet at the same time [3].

In the PCF mode of operation, the access of the wireless channel is centralized by a polling-based protocol controlled by the Point Coordinator (PC). The access points (APs) generally act as PCs. The PCF mode provides contention-free service to the wireless stations. In PCF mode, frame is divided into two part CFP and CP. The PC sends a beacon frame, which indicates the start of the contention-free period. Beacon frame contains the list of pollable stations and other polling management information. The CFP is repeated after a fixed interval. After sending the beacon frame, the PC starts polling all stations one by one in the sequence indicated in the beacon. In CFP, if the PC has a data packet to send to a station, it sends the polling packet piggy backed on the data packet and if the PC does not have any data to send, then it sends only a polling packet [3].

The polled station responds by sending the uplink ACK packet and piggybacks any uplink data on the ACK packet. If polled station does not have data to send in the uplink, then it only sends a null packet in response to the poll by PC. In this scheme some of the bandwidth is used for the polling and ACK. That means it is wasted. If other stations may not have any uplink and downlink data, but even then the PC polls these stations resulting in wastage of bandwidth. These limitations of the basic PCF mode limit the number of simultaneous VoIP calls [3].

To improve the call capacity different techniques like PCF with Signalling Scheme [4], Modified PCF [5], Dynamic PCF [6], ICF [7] etc. were developed.

This paper is organized in five sections. In the section II Modified ICF technique is discussed. Section III describes about the new proposed scheme. Section IV is performance analysis in which the results are discussed. Section V gives the conclusion.

II. Modified ICF Technique

The Modified ICF [3] proposed in which the concept of M-M scheme is used in the downlink stream. In MICF access point combines the data from several downlink streams into a single larger downlink packet. This will reduce the overhead from multiple VoIP packets to of a single packet. This scheme also saves the SIFS intervals between the two adjacent time slots. The Modified ICF scheme saves large amount of MAC and PHY layer overheads by transmitting a single large packet rather than multiple smaller packets with their individual overheads. The saved bandwidth can be used for supporting additional stations, thereby increasing the capacity.

In this scheme, at the start of an ICF cycle, the uplink stations will send the packets according to the entries in the SV. When all uplink transmission is complete, the AP will sense that the channel is free for SIFS time interval and finding it free, it will transmit the downlink voice traffic. The downlink traffic goes through a MUX that replaces the RTP, UDP and IP headers of each voice packet with a compressed miniheader of 2 bytes which combines multiple packets into a single multiplexed packet then multicasts this multiplexed packet to the WLAN through the AP using a multicast IP address. The payload of each VoIP packet is preceded by a mini header in which an identification ID used to identify the session of VoIP packet. All STAs will receive the multicast packets. The extraction of packets from multiplexed packet is performed by a DEMUX at the receiver [3].

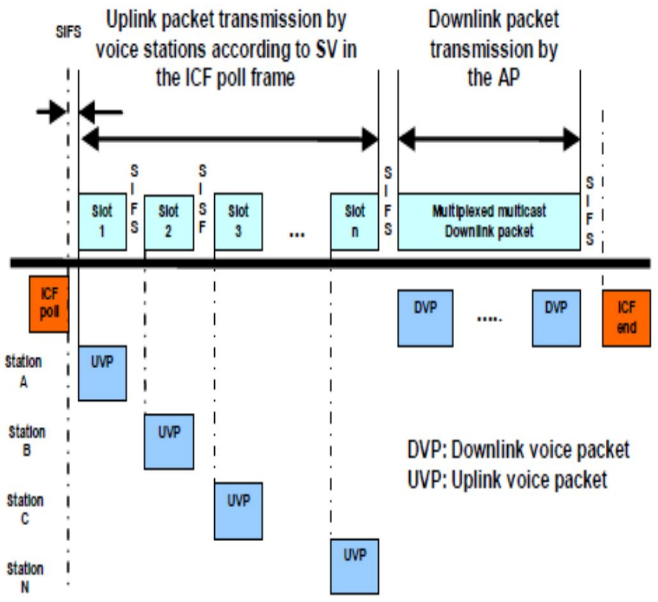

Fig. 1 Modified ICF [3]

Although the Modified ICF increases the call capacity but there are some problems associated with it. The uplink voice packets are sent one by one in normal way. There is no multiplexing scheme implemented at the uplink time.

On the other hand it increases time as the time required making the MUX packet where other headers are replaced with 2 byte header and at DEMUX side also time required to separate the voice packets and add their original headers. The 2 byte header reduce the size of packet and this packet can be sent in one time slot but MUX and DEMUX process increase the delay.

The Modified ICF removes the SIFS between the downlink packets because there is only one MUX packet. But one has to compromise with the delay. The preparation and retrieval of information from this packet takes time. The call capacity increases in the MICF but MUX and DEMUX process increase the delay.

III. 3-Buffered Packet ICF

A. 3-Buffered Packet ICF

In Modified ICF delay is created by the MUX and DEMUX. The packet size is also large. To overcome the drawbacks of Modified ICF a new scheme is proposed that is 3-Buffered Packet ICF which decreases the delay in WLAN. In this scheme the TDMA like service is used for the voice packet transmission.

TABLE 1
IEEE 802.11b Parameters and Values [2]

IEEE 802.11b standard Parameter	Values
Data Rate for data packet transfer	11Mbps
MAC layer overhead	34 byte
Phy layer header	24 byte
Transmission time of CF end (T_{ce})	352 µs
Transmission time of beacon packet (T_b)	744 µs
Transmission time of null frame	464 µs
Transmission time of polling frame	464 µs
Transmission time of Payload (T_v)	372.18 µs
SIFS interval (T_{sifs})	10 µs
Voice Packet Payload	160 byte
RTP+UDP+IP overhead	40 byte
Contention Free Period (T_{cfp})	15-19 ms

The CFP is divided in to the number of slots. Each slot has some space between the next time slots. This interval or space is called the SIFS. At the starting of ICF cycle the AP broadcast the ICF poll frame. The ICF poll frame contains status vector (SV) that contains the string of polling bits. At the time of connection establishment, the AP assigns the polling bit. A station sets it to send data, otherwise it is zero. Slots are allotted to each station. The station count the number of one's and find its time slot.

In the 3-Buffered Packet ICF scheme, each station has the buffer. Three packets are stored in the buffer. These packets are ready to transmit. The scheme is called 3- Buffered Packet ICF.

The AP assigns the time slot to the particular station as per the SV. Now the station transmits the buffered packet to the AP. In 3-Buffered Packet ICF, three packets are transmitted. Here the important point is that the packets are stored in the buffer. As the slot allotted, the packets are transmitted to the AP.

At the downlink stream, the packets are sending to the particular station by the AP. Three packets transmitted to the station in 3- Buffered Packet ICF by AP when the time slots allotted.

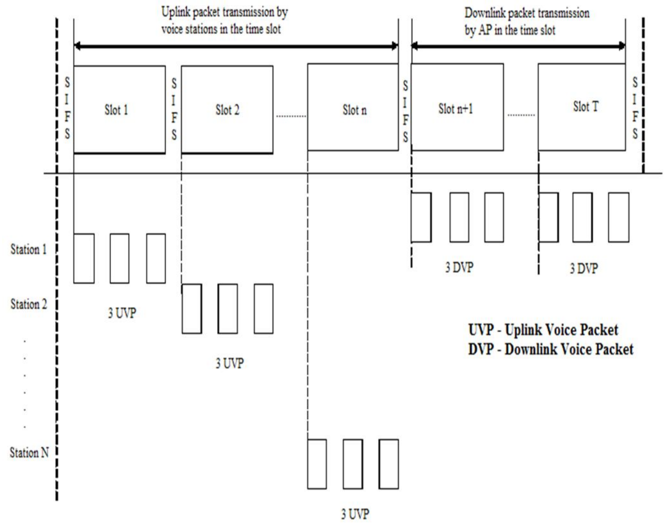

Fig.2 3-Buffered Packet ICF Scheme

B. Slot Calculation

1) Slots in ICF:

The time slots in ICF can be calculated as follows [3]:-

$$\text{No. of time slots in ICF} = \frac{(Tcfp - Tb - Tce)}{(Tv + Tsifs)}$$

Where T_{cfp} is the Contention Free Period, T_b is transmission time of beacon packet, T_{ce} is the transmission time of CF end, T_v is transmission time of voice payload and T_{sifs} is SIFS interval.

TABLE 2
NO. OF TIME SLOTS IN ICF FOR DIFFERENT VALUES OF T_{cfp}

T_{cfp}	No. of Time slots in ICF
15 ms	36
17 ms	41
19 ms	46

2) Slots in 3- Buffered Packet ICF:

In the 3- Buffered Packet ICF scheme, three voice packets are stored in the buffer of station. These three packets have its header. There is no common header of these three packets. These three packets are transmitted simultaneously in the channel access. These three packets are separated by $T_{sifs.}$ that means the slot size is equal to the size of three packets. So No. of Slots is given by

$$\text{No. of time slots in 3 Buffered Packet ICF} = \frac{(Tcfp - Tb - Tce)}{3\,(Tv + Tsifs)}$$

For the value of T_{cfp} = 15 ms, 17 ms and 19 ms, the value of time slots is 12, 13 and 15 respectively as shown in Table 3.

TABLE 3
NO. OF TIME SLOTS IN 3-BUFFERED PACKET ICF FOR DIFFERENT VALUES OF T_{cfp}

T_{cfp}	No. of Time slots in 3- Buffered Packet ICF
15 ms	12
17 ms	13
19 ms	15

IV. PERFORMANCE ANALYSIS

Performance analysis of the 3-Buffered Packet ICF is done in MATLAB 7.0.

In 3-Buffered Packet ICF scheme, for the value of T_{cfp} =15 ms, 17 ms and 19 ms, the results are obtained.

Case 1 when CFP = 15 ms then the number of time slot is 12. In this case 34 stations give the delay less than 10 ms and all other stations gives the delay less than 38 ms as shown in fig. 3. There is increase in the number of packets in one slot in the 3-Buffered Packet ICF and the maximum delay is 37.7 ms.

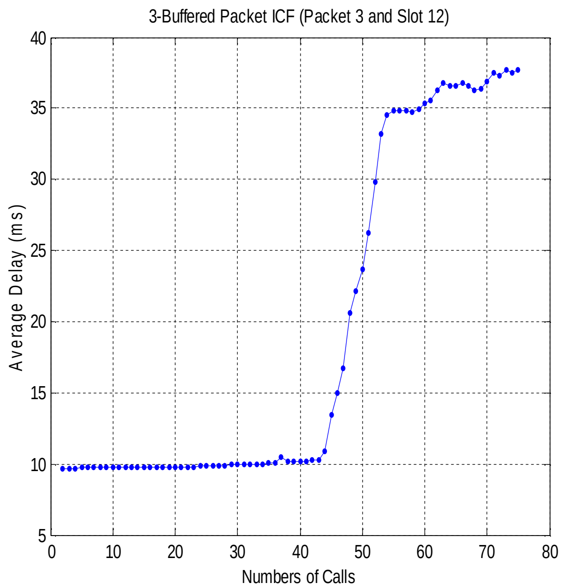

Fig. 3 3-Buffered Packet ICF scheme with CFP =15ms

Case 2 when CFP = 17 ms then the number of time slot is 13. In this case 29 stations give the delay less than 10 ms and all other stations gives the delay less than 35 ms as shown in fig. 4. As the size of CFP increases, the number of slots increases, which increase the number of calls and decrease the delay. The highest delay reduces from 37.7 ms in previous case to 34.5 ms.

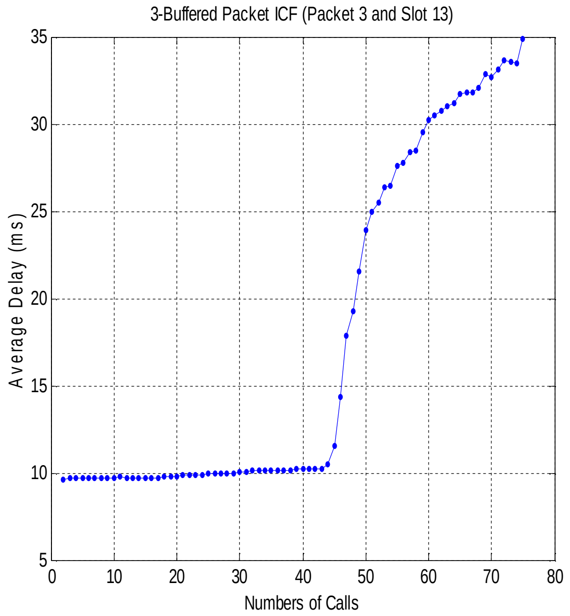

Fig.4 3-Buffered Packet ICF scheme with CFP =17ms

Case 3 when CFP = 19 ms then the number of time slot is 15. In this case 39 stations gives the delay less than 10 ms and all other stations gives the delay less than 34 ms as shown in fig. 5. As the size of CFP increases, the number of slots increases, which increase the number of calls and decrease the delay. The highest delay decreases from 34.5 ms (in previous case) to 33.7 ms.

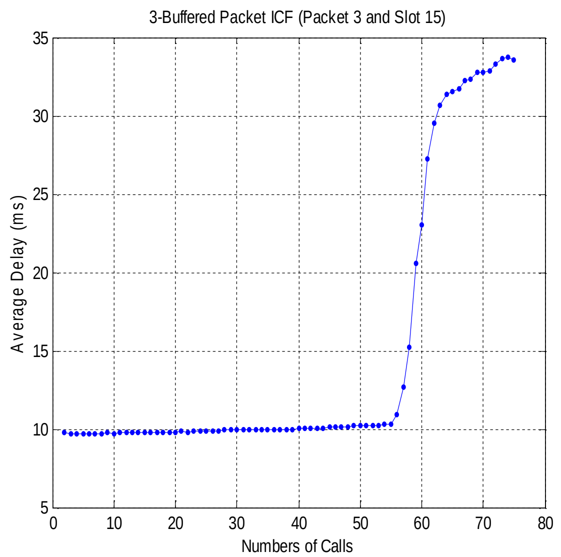

Fig.5 3-Buffered Packet ICF scheme with CFP =19ms

The variation in the delay is occurred due to the dynamic nature of the voice traffic. In the fig. 6, the graph between CFP and minimum and maximum delay for 3–Buffered Packet ICF is shown. It is identified that the maximum delay decreases as increase in the CFP. There is variation in minimum delay due to the dynamic behavior of traffic.

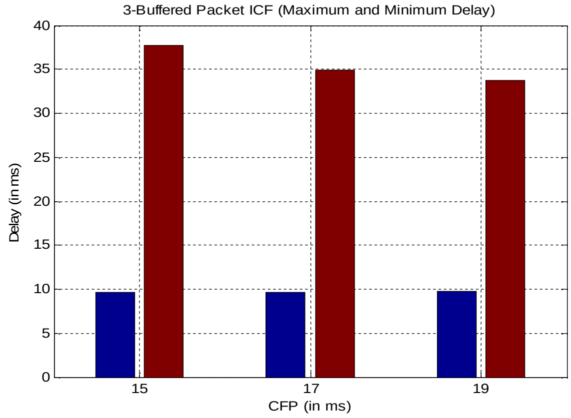

Fig.6 Graph between CFP and Delay (Max. and Min.) for 3-Buffered Packet ICF

In the Table 4, the number of calls shown on the basis of delay < 10 ms and delay < 40 ms for the 3-Buffered Packet ICF scheme. 3-Buffered Packet ICF gives the better result at CFP = 19 ms than CFP=15 ms.

TABLE 4
NUMBER OF CALLS WHEN DELAY < 10ms AND DELAY < 40 ms OF 3-BUFFERED PACKET ICF

CFP	No. of Calls	
	3-Buffered Packet ICF	
	Delay < 10 ms	Delay < 40ms
15 ms	34	75
17 ms	29	75
19 ms	39	75

The comparison of 3-Buffered Packet ICF with the Modified ICF is shown in the table 5. The proposed scheme gives the better results than the previous scheme.

TABLE 5
COMPARISON OF 3-BUFFERED PACKET ICF WITH MODIFIED ICF

CFP	No. of Calls at Delay < 40ms	
	3-Buffered Packet ICF	Modified ICF [3]
15 ms	75	33
17 ms	75	36
19 ms	75	46

V. CONCLUSION

In daily life large amount of real time and non real time data is generated. The voice is normally transferred using the

PSTN. VoIP is the newly developed method to transfer voice through WLAN. The voice traffic is increasing day by day.

Large amount of work has been done to improve the call capacity. In this work 3-Buffered Packet ICF is proposed to improve the call capacity at delay < 40 ms. The main concern is delay in the WLAN. In this work 40 ms delay is taken in to account. 3-Buffered Packet ICF provides better result than Modified ICF, because multiple (three) packets are transmitted with less delay.

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