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Analysis and Performance Evaluation of IEEE 802.11 WLAN

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Abstract--**With fast deployment of wireless local area networks VoIP over IEEE 802.11 wireless local area network (WLAN) is growing very fast and is providing a cost effective alternative for voice communications. WLANs were initially set up to handle bursty nonreal time type of data traffic. Therefore, the wireless access protocols initially defined are not suitable for voice traffic. Subsequently, updates in the standard have been made to provision for QoS requirements of data, especially the real time traffic of the type voice and video. Despite these updates, however, transmitting voice traffic over WLAN does not utilize the available bandwidth (BW) efficiently, and the number of simultaneous calls supported in practice is significantly lower than what the BW figures would suggest. Several modifications have been proposed to improve the call capacity, and recently isochronous coordination function (ICF) was introduced to mitigate the problem of low call capacity. The proposed modified ICF which further improves the performance in terms of the call capacity. The proposed scheme uses multiplexing and multicasting in the downlink to substantially increase the call capacity.**

Keywords – Voice over internet protocol (VoIP), IEEE802.11, wireless local area network (WLAN), quality of service (QoS), isochronous coordination function (ICF).

1. INTRODUCTION

Wireless local area networks (WLAN) hold the promise of VoIP over WLAN is becoming a very attractive solution for wireless voice communications. One of the reasons for the huge interest in VoWLAN is the potential of the WLANs to bypass the local loop of the traditional telephone system (PSTN).

Voice over Internet Protocol (VoIP) is a method used in a data networks and broadband internet to establish voice calls. This is implemented by converting the analog voice calls into digital formats that can be transmitted through the internet or intranet. Voice signals will be transferred through a packet-switched network instead of being transmitted through a dedicated circuit switched voice lines. The calls can therefore enter into a well-connected IP network directly through WLAN. The other reason is that WLANs are widely available and easy to deploy. This technology uses the existing packet-switched data network for transporting the packets and provides a low-cost alternative to the traditional telephone system. Wireless LAN standard 802.11 specifies two modes for wireless channel access. These are distributed coordination function (DCF) [1] and point coordination function (PCF) [1].

DCF mode [1] is based on random access of channel that is best suited for nonreal-time traffic, that is, bursty traffic. The DCF mode is based on carrier sense multiple accesses with collision avoidance (CSMA/CA). The timing diagram of DCF scheme is depicted in Figure 1. In the DCF mode control to the access of channel is distributed among all the stations. The DCF access method is based on the CSMA/CA principle in which a host, wishing to transmit, senses the channel to check if it is free. On finding the channel free, the host waits for a random amount of time (to avoid two hosts starting transmission at

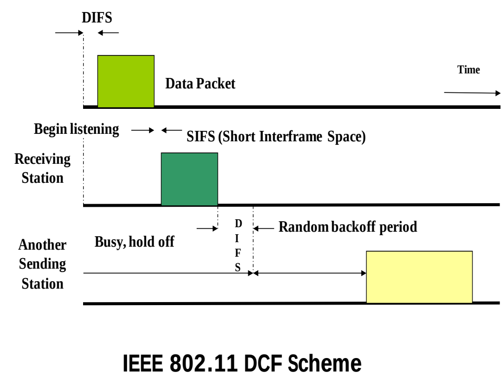

Figure 1: IEEE 802.11 DCF Scheme

the same time) before transmitting.

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 serve as PCs. The PCF mode provides contention-free service to the wireless stations. In PCF mode, a

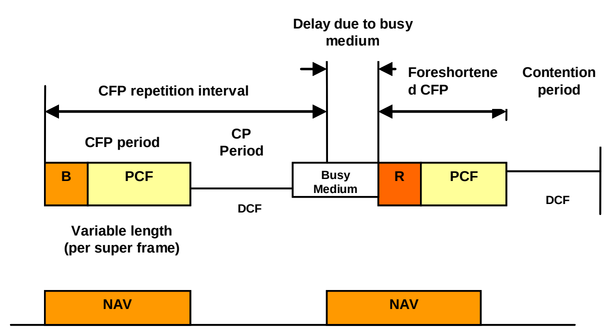

Figure 2: Basic PCF Mode of operation

frame is divided into two part CFP and CF. The PC indicates the start of the contention-free period by sending a beacon frame that contains the list of pollable stations and other polling management information. The CFP is repeated after a fixed interval. The CFP and CP together constitute a superframe whose structure is shown in Figure 2. After sending the beacon, the PC starts polling stations one by one in the order 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 frame is and contention period (CF). The PC indicates the start of the contention-free period by sending a beacon frame that contains the list of pollable sta and other polling management information. The CFP is repeated after a fixed interval. The CFP and CP together constitute a superframe whose structure is shown in Figure 3. After sending the beacon, the PC starts polling stations one by one in the order indicated in the beacon. In CFP, if the PC has a data packet to send to a station, it sends the polling packet piggybacked on the data packet, and if the PC does not have any data to send, then it sends only a polling packet. 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 just 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, and

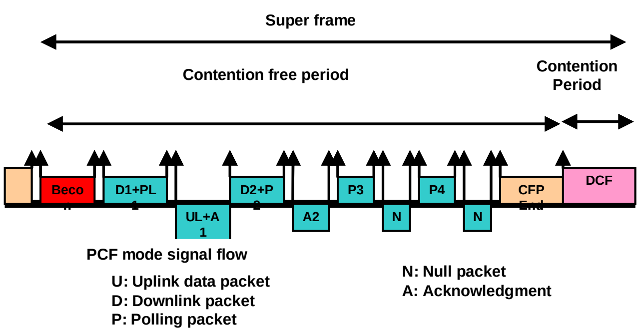

Figure 3: Flow of signals and data in PCF Mode

hence it is wasted. Here other stations may not have any uplink and downlink data, but even then the PC polls these stations resulting in wastage of bandwidth. These drawbacks of the basic PCF mode limit the number of simultaneous VoIP calls.

There are several proposals given by various authors, like dynamic PCF [6], modified PCF [10], adaptive PCF and so forth, which improve call capacity. These proposals seek to overcome the call capacity deficiencies of the PCF mode of operation, thereby providing capability to the WLAN network to accommodate a larger number of simultaneous VoIP calls. One of the proposed techniques introduces a new modified multiple access mechanism termed as isochronous coordination function to improve the capacity.

2. ICF

The main aim of the Isocronous Coordination System is to provide a dynamic time division multiple accesses (TDMA)-like service for transporting voice packets efficiently [2]. In general, the AP may initiate an ICF cycle whenever necessary (e.g., periodically) during the optional CFP or during the CP. At the beginning of an ICF cycle, the AP of a BSS broadcasts an ICF-poll frame. Included in the ICF-poll frame is a status vector (SV), which is essentially a string of polling bits, one for each admitted voice station. At the time of connection setup with the AP each admitted voice station is assigned a polling bit. The polling bit will be reused by another admitted voice station when the current connection is terminated. In each ICF cycle, voice stations transmit in assigned time-slots, as shown in Fig.4. To retain the channel throughout the ICF cycle, consecutive time-slots are separated by an SIFS period. Based on its polling position and the status of other stations, as indicated by the SV in the ICF-poll frame, an active station determines its time slot (if any) in the ICF cycle. In the SV, a "1" polling bit indicates that the corresponding station may transmit a voice packet in the current cycle, and vice versa.

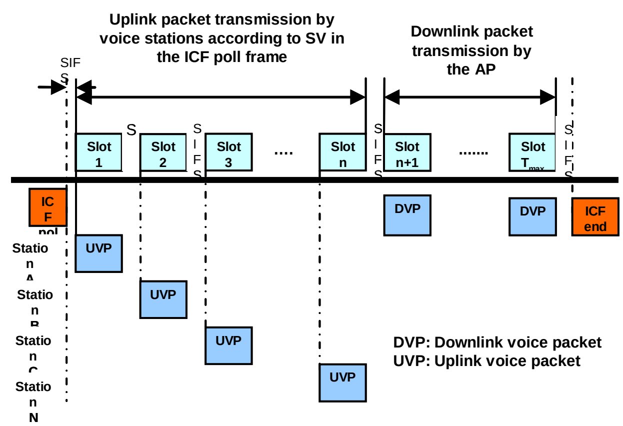

Figure 4: Isochronous coordination function

This scheme aims to exploit voice traffic correlation to obtain a tradeoff between call capacity and loss ratio. Voice traffic is correlated to some extent and therefore voice data orresponding to some lost packets can be reconstructed from the received voice packets. ICF uses fixed-size time slots for scheduling traffic and this type of scheduling mirrors isochronous traffic pattern exactly. However, fixed-size packet implies that speech frame can no longer be buffered and it has to be dropped if a time slot is not made available to a particular station in a given superframe. The procedure for slot allocation is such that it maximizes the number of users supported while ensuring that the packet loss for any user is not greater than 1%.

Due to the limited number of time slots in an ICF cycle, all stations may not be polled, so an efficient polling list management is implemented by using cyclic polling queue [9]. Due to the time-sensitive but loss-tolerable nature of voice, the unpolled stations (which do not get time slot in ICF cycle for transmission) drop one packet. When such a packet drop takes place, then this particular station is provided higher priority in slot allocation packet loss is kept to a minimum. Thus, the cyclic polling queue management ensures fair polling of active voice stations and seeks to minimize consecutive packet losses.

3. CAPACITY ANALYSIS

IEEE 802.11 capacity analysis

A constant bit-rate (CBR) [10] VoIP client generates one VoIP packet every packetization interval. Therefore, the number of packets that can be sent during one packetization interval is the maximum number of calls that can be supported. The capacity of VoIP can be calculated as follows:

$$N_{\max} = \frac{T_P}{2T_t} \tag{1}$$

where $N_{\max}$ is the maximum number of calls, T_p is the packetization interval, and T_t is the time for sending one packet of voice. T_p depends upon the codec used in the VoIP client.

VoIP capacity of PCF

To avoid delay, VoIP station needs to be polled every packetization interval, which means that CFP cannot be more than the packetization interval. Therefore, $N_{\max}$ is the maximum number of stations that can be polled in CFP, which can be calculated as follows:

$$N_{\max} = \frac{0.5(T_{CFP} - T_B - T_{CE})}{(T_v + T_P + 2T_{SIFS})} \tag{2}$$

where T_{CFP}, T_B, T_{CE}, , $T_{v,}$ T_P and T_{SIFS} are the durations of contention-free period, beacon frame, contention-free period end frame, transmission time for voice packet, transmission time for polling frame, and short interframe space (SIFS) period, respectively. Ordinarily, in voice communication, uplink and downlink stations do not transmit voice packets simultaneously Therefore, $\mathrm{N}_{\max}$ can be calculated using following equation:

$$N_{\max} = \frac{0.5(T_{CFP} - T_B - T_{CE})}{(2T_P)} = \frac{(T_{CFP} - T_B - T_{CE})}{2(T_v + T_{SIFS})} \tag{3}$$

Here, T_{p} is the transmission times for polling frame.

oIP capacity of ICF

If we compare the time required for sending the voice traffic and the polling frame, it becomes apparent that polling each STA individually constitutes a very large overhead. This procedure becomes even more inefficient when some stations do not have voice packet to send (here a polling frame is sent and a null frame is sent as response; either of these packets does not carry any useful traffic). Calculation shows that only one additional STA can be polled when three STAs do not have voice traffic to transmit.

In ICF mode, the transmission order of every STA is decided by the access point at the time of association. AP transmits the status vector in the beacon frame, and the STAs use this information to obtain their position in the transmission order. Using Figure 4 it is easy to obtain $N_{\max}$ as follows:

$$N_{\max} = \frac{(T_{CFP} - T_B - T_{CE})}{(T_v + T_{SIFS})} \tag{4}$$

Table 1: Lists the call capacities of the various schemes

	$\boldsymbol{N_{\max}}$	
CFP	**PCF**	**ICF**
15ms	15	30
17ms	17	34
19ms	19	39

4. MODIFIED ICF

The modification of the ICF scheme will results in enhancement of call capacity. In the previously proposed scheme (isochronous coordination function [9]), the downlink packets are sent using the same procedure as the one used for uplink packets. To improve the performance of ICF scheme, modified ICF (MICF) scheme was proposed for channel access. Here proposed multiplex-multicast (M-M) scheme [5] is to be used in downlink stream. This proposed modification exploits the fact that there is an opportunity with the access point to combine the data from several downlink streams into a single larger downlink packet. This will reduce the overhead from that of multiple VoIP packets to

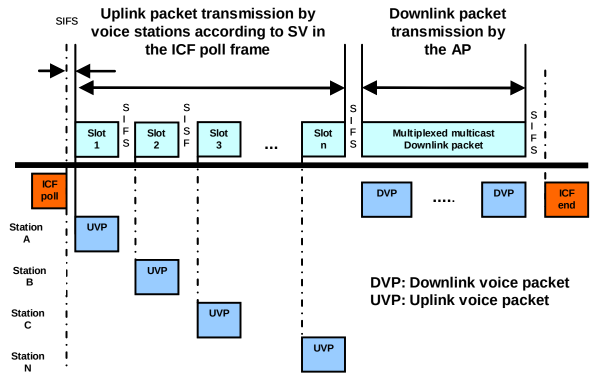

Figure 5: Modified ICF Scheme

that of a single packet (thereby resulting in better bandwidth utilization). This scheme also saves the SIFS intervals between the two adjacent time slots (downlink direction). The modified ICF scheme, as shown in Figure 5, 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 time

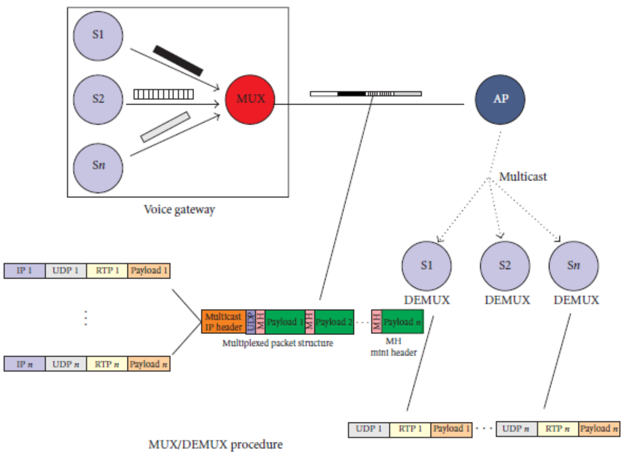

Figure 6: MUX / DEMUX procedure.

required for sending 3 downlink packets (data of three users) in the current ICF scheme can be used to send data of 8 users. The bandwidth thus saved can be used for supporting additional stations, thereby increasing the capacity.

In the modified 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 then it will transmit the downlink voice traffic. The downlink VoIP traffic goes through an MUX and replaces the RTP, UDP, and IP (combined header size of 40 bytes) headers of each voice packet with a compressed miniheader of 2 bytes, which combines multiple packets into a single multiplexed packet then multicasts the multiplexed packet (containing downlink voice traffic as per the entries in the SV) to the WLAN through the AP using a multicast IP address. The payload of each VoIP packet is preceded by a miniheader in which there is an identification ID used to identify the session of VoIP packet. All STAs will receive the multicast packets. The extraction is performed by a DEMUX at the receiver.

In M-M scheme which is used in Modified ICF it is illustrated that the saving in bandwidth that can be achieved using M-M scheme in downlink. The calculations show that 8 stations can receive their downlink VoIP packets in three ICF time slots using the MICF scheme (this takes 8 time slots in the basic ICF scheme). The time slots made available by using M-M scheme may be utilized to accommodate a larger number of uplink stations. The polling queue is maintained using the same algorithm used in the basic ICF:

$$\text{ICF time slot} = OH_{sender} + OH_{hdr} + \text{Payload} \quad (5)$$

The optimal payload size for the multiplexed downlink packet is chosen to be 1500 and for a voice frame data size of 160 bytes (for G.711 codec), this implies that multiplexing 8 stations results in an optimal packet size. The time duration T_{down} to send a multiplexed packet containing 8 voice frames can be obtained as follows:

$$T_{down} = 8/11*[(\text{payload}+2)*N+H_{udp}+H_{mac}]+OH_{sender} \quad (6)$$

where payload = 160 bytes, H_{UDP} = 8 bytes, H_{UDP} = 20 bytes, H_{MAC} = 34 bytes, and OH_{Sender} = SIFS + PHY = 202 microseconds.

On substituting the values, we obtain T_{down} to be about 1200 microseconds. This duration corresponds to about 3 ICF time slot durations. Multiplexing more stations will lead to greater saving in bandwidth, but it will result in an increase in the probability of packet loss because of increased packet size [3-4] and [7], and thus it will negate the gain achieved. There is tradeoff between packet size and packet loss rates. The payload size has been chosen to be 1500 bytes, as this payload size produces a good compromise between effective throughput and bandwidth gain due to larger payload size..

Implementation of the M-M scheme improves the voice capacity of the WLAN. However, on the other hand, this scheme introduces some complexity in form of MUX functionality at gateway and DEMUX functionality at the receiving station. The receiving stations have to demultiplex the received multiplexed multicast packet to extract the payload intended for them. This adds some processing delay; however, this delay is small and can be offset by choosing better (and costlier) hardware.

5. SIMULATION RESULTS

Using the particular values of IEEE 802.11b parameters equations, the call capacity (number of simultaneous voice calls) for the different schemes has been calculated. Figure 7 shows a comparison between ICF, basic PCF. Figure 8 shows a comparison between ICF, MICF. In this simulation, the CFP is taken as 15 milliseconds and frame repetition interval as 20 milliseconds. Figure 7 represents loss ratio as a function of the number of simultaneous voice calls.

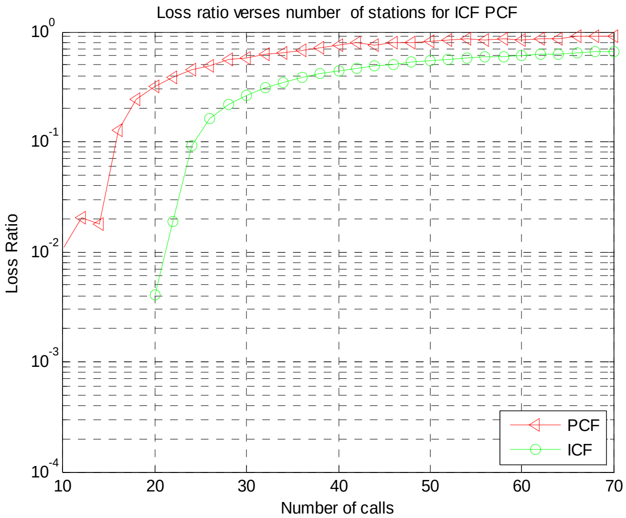

Figure 7: Comparison of PCF and ICF.

Figure 8 shows a comparison between ICF, MICF. and Figure 9 shows a comparison between PCF,

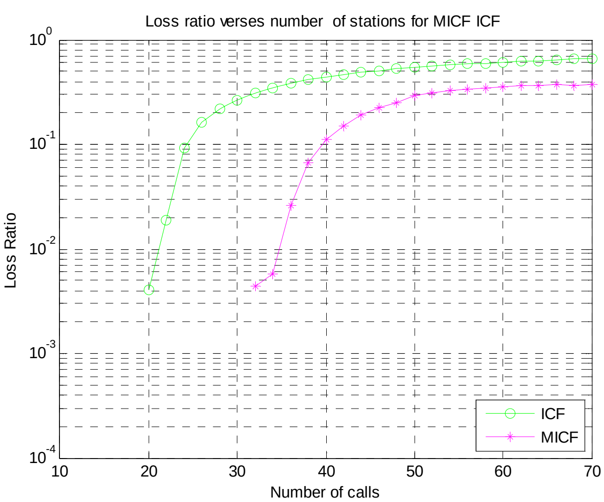

Figure 8: Comparison of ICF and MICF.

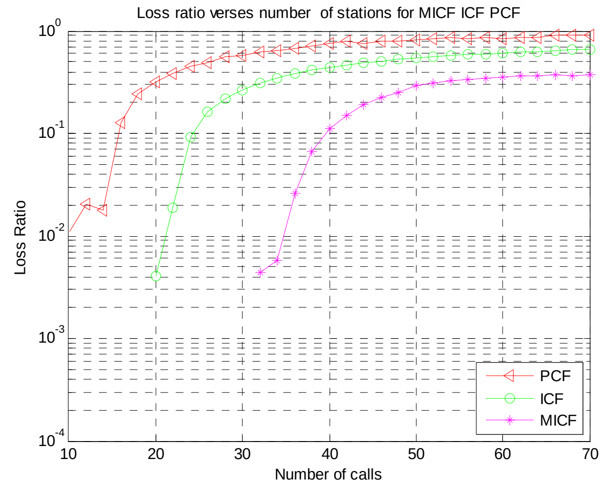

Figure 9: Comparison of PCF, ICF, and MICF for CFP = 15 milliseconds.

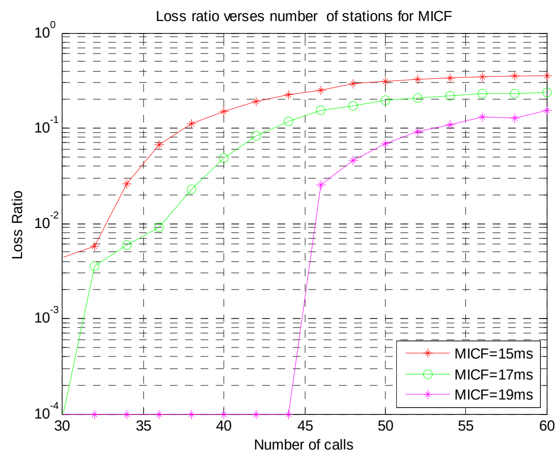

Figure 10: Effect of varying CFP period on MICF.

ICF, MICF. In this simulation, the CFP is again taken as 15 milliseconds and frame repetition interval as 20 milliseconds. Figure 10 shows that by increasing the CFP period, we can improve the call capacity, but this results in unfair distribution of bandwidth between real-time (in CFP) and nonreal-time (in CP) traffics. The choice of CFP period is therefore a compromise between call capacity and fair distribution between real-time and nonreal-time traffics.

6. CONCLUSION

VoIP over IEEE 802.11 wireless local area network (WLAN) is growing very fast and is providing a cost effective alternative for voice communications. Here proposes a scheme for increasing call capacity of voice traffic. The ICF technique which leads to a large call capacity has been modified to increase the call capacity further. The proposed scheme exploits the strength of the M-

M scheme and integrates it into the ICF technique resulting in a high call capacity procedure.

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