

Research Article

Modified Isochronous Coordination Function for Enhancement of VoIP Call Capacity over IEEE 802.11 WLAN

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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. In this paper, we propose a 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.

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1. INTRODUCTION

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). 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 is based on random access of channel that is best suited for nonreal-time traffic, that is, bursty traffic, and PCF mode is based on polling mechanism that is best suited for real-time traffic. However, most of the early devices do not support PCF mode.

During early years of WLAN deployment, mostly the DCF mode was supported in WLAN devices, but in recent years, the importance of the PCF mode is being recognized and now the PCF mode is also being supported [12–14] in

new devices like laptops, personal digital assistants (PDAs). The DCF mode is based on carrier sense multiple access with collision avoidance (CSMA/CA). The timing diagram of DCF scheme is depicted in Figure 1.

In the DCF mode [1], 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 the same time) before transmitting.

In the PCF mode [1] 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 frame is divided in two parts: contention-free period (CFP) 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 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(a).

As shown in Figure 2(b), 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 only for polling and ACK, and hence it is wasted. Here, in Figure 2(b), stations P3 and P4 do 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 [11], modified PCF [6], adaptive PCF [8], 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 [9] to improve the capacity.

2. ICF OVERVIEW

Isochronous coordination function was introduced to handle constant bit-rate real-time traffic, especially voice traffic. It aims to provide a dynamic time division multiple access (TDMA-) like service for transporting voice packets efficiently [9]. The ICF-poll frame includes a status vector (SV), which is a string of polling bits, one for each admitted voice station. These polling bits are assigned to each station at the time of connection setup with the AP. In each ICF cycle, voice stations transmit in assigned time slots, as shown in Figure 3. 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. 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 corresponding to some lost packets can be reconstructed from the received voice packets. Studies have shown that in order to provide acceptable quality of service, the lost packet number should not be greater than 1% [5] of the total number of packets sent by a particular station. This characteristic of voice traffic is exploited by the ICF technique which provides a mechanism to trade off delay with packet loss. 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

TABLE 1

Codec	Packet duration (ms)	Payload size (bytes)
G.711	20	160
G.726	20	160
G.723.1	30	20/24
GSM (13.2 kbps)	20	33

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 when the polling queue is updated for the next superframe. This is done to ensure that consecutive 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) [6] 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:

$$N_{\max} = \frac{T_p}{2T_t}, \quad (1)$$

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. The reason for multiplying T_t by 2 is that the voice communication is full-duplex. N_{\max} can be higher if we account for the fact that normally we do not have voice data in both directions simultaneously. T_p depends upon the codec used in the VoIP client. [10, Table 1] lists these values for typical codecs.

A. 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:

$$N_{\max} = \frac{0.5(T_{\text{CFP}} - T_B - T_{\text{CE}})}{(T_v + T_p + 2T_{\text{SIFS}})}, \quad (2)$$

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

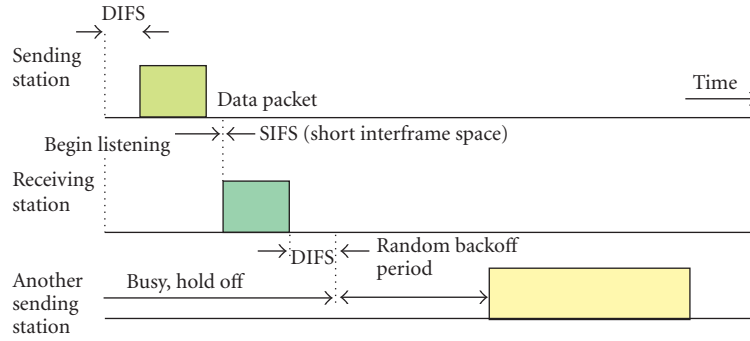


FIGURE 1: IEEE 802.11 DCF Scheme.

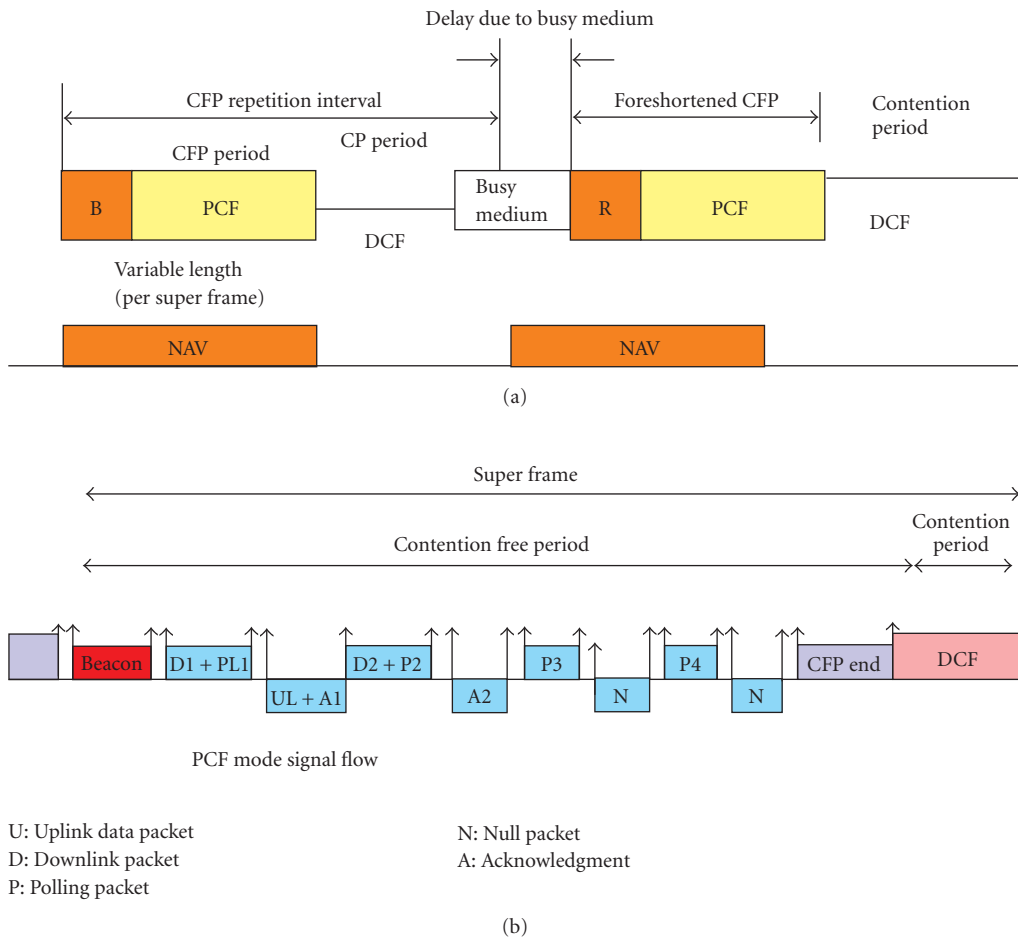


FIGURE 2: (a) Basic PCF Mode of operation; (b) flow of signals and data in PCF Mode.

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, polling the STA for uplink data in the frame in which downlink data for that STA is transmitted is not an efficient method of polling as it will result in unnecessary polls.

So, CFP is further divided into uplink transmission period and downlink transmission period. In uplink period, CF-pollable STAs are polled according to the polling list management scheme implemented in AP. If assigned uplink transmission period is not fully utilized by the stations in the uplink polling list, the remaining duration is utilized for sending downlink voice data to STAs which do not appear in the downlink polling list. Downlink traffic is transmitted using

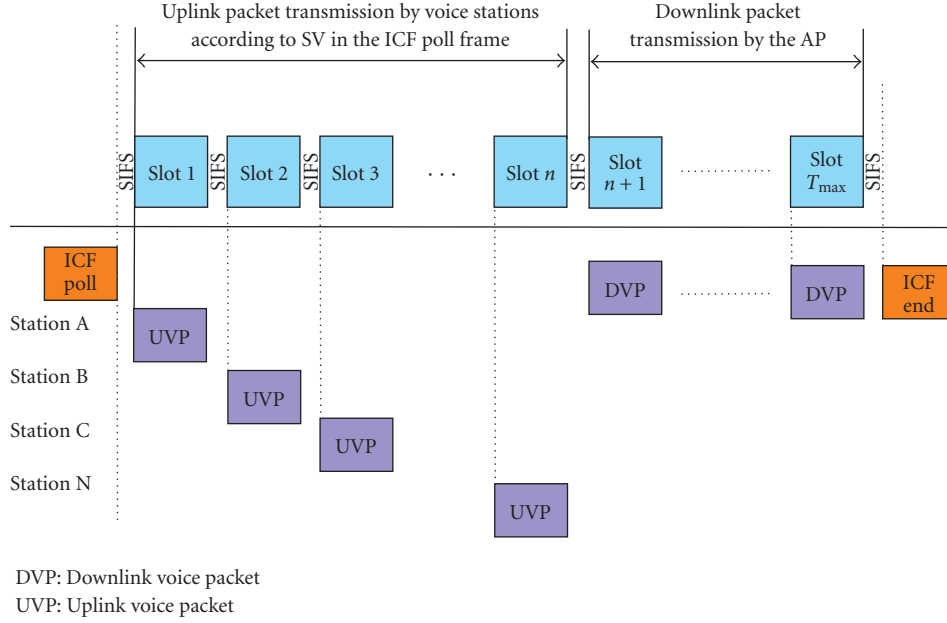


FIGURE 3: Isochronous coordination function [9].

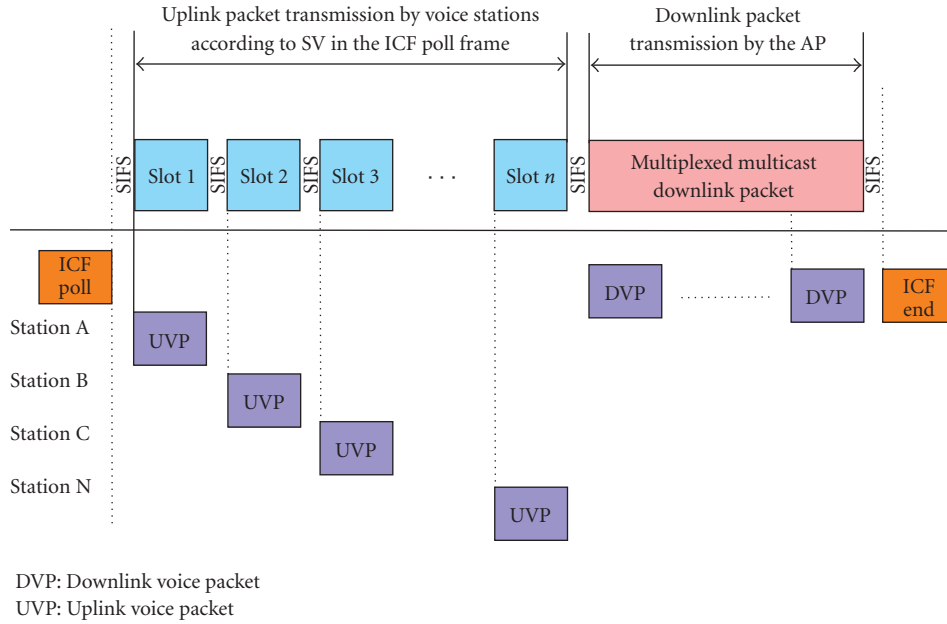


FIGURE 4: Modified ICF Scheme.

FIFO mode. Therefore, N_{\max} can be calculated using following equation:

$$N_{\max} = \frac{(T_{\text{CFP}} - T_B - T_{\text{CE}})}{(2T_p)} = \frac{(T_{\text{CFP}} - T_B - T_{\text{CE}})}{2(T_v + T_{\text{SIFS}})}. \quad (3)$$

Here, T_p is the transmission time for polling frame.

The parameter values listed in [10, Table 2] are for the G.711 codec, with voice traffic being modeled as Markov bistate [7].

B. VoIP 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.

TABLE 2

IEEE 802.11b parameters	Values
Data rates for data packets	11 Mbps
Data rates for control packets and PHY overheads	1 Mbps
MAC layer overheads	34 bytes
PHY layer overheads	24 bytes
Transmission time for TB	744 μ s
Tx time for null frame	464 μ s
Tx time for polling frame	464 μ s
Tx time for CF end frame	352 μ s
CFP contention-free period	15–19 ms
SIFS interval	10 μ s
Slot time interval	20 μ s
Probability of ON period	0.43
Probability of OFF period	0.57

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 3, we can easily obtain N_{\max} as follows:

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

4. MODIFIED ICF

In this section, we propose a modification of the ICF scheme which results in enhanced 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, we propose a modified ICF (MICF) scheme for channel access.

Here, we propose the multiplex-multicast (M-M) scheme [10] 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 that of a single packet (thereby resulting in better bandwidth utilization). This scheme also saves the time period corresponding to SIFS intervals between the two adjacent time slots (for data to be sent in the downlink direction). The modified ICF scheme, as shown in Figure 4, saves large amount of MAC and PHY layer overheads by transmitting a single large packet rather than multiple smaller packets with their individual overheads. As shown by the calculations later in this section, the time required for sending 3 downlink packets (and therefore 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 for the given cycle, the AP will sense that the channel is free for

SIFS time interval and then it will take the control of channel to transmit the downlink voice traffic. The downlink VoIP traffic first goes through an MUX in the voice gateway. The MUX 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 and their packets will be extracted by VoIP ID present in the miniheader. The extraction is performed by a DEMUX at the receiver. After retrieving the VoIP payload, the DEMUX then restores the original RTP header and necessary destination information and assembles the data into its original form before forwarding it to the VoIP application. The proposed modification is illustrated in Figure 5.

We now illustrate the saving in bandwidth that can be achieved using M-M scheme in downlink. The following 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 as the one used in the basic ICF:

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

The optimal payload size for the multiplexed downlink packet is chosen to be 1500 bytes (this choice of packet size is explained later in the section), and for a voice frame data size of 160 bytes (corresponding to the 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 [10]:

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

where payload = 160 bytes, $H_{\text{UDP}} = 8$ bytes, $H_{\text{UDP}} = 20$ bytes, $H_{\text{MAC}} = 34$ bytes, and $\text{OH}_{\text{Sender}} = \text{SIFS} + \text{PHY} = 202$ microseconds.

On substituting the values, we obtain T_{down} to be about 1200 microseconds. This duration corresponds to about 3 ICF time slot durations (refer to (4)).

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 [15–17], 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 (refer to [17, Figure 2]). In our simulation, we have multiplexed the data of 8 VoIP STAs (to achieve the optimum payload size of 1500 bytes). We can send more than one multiplexed packet of 1500 bytes

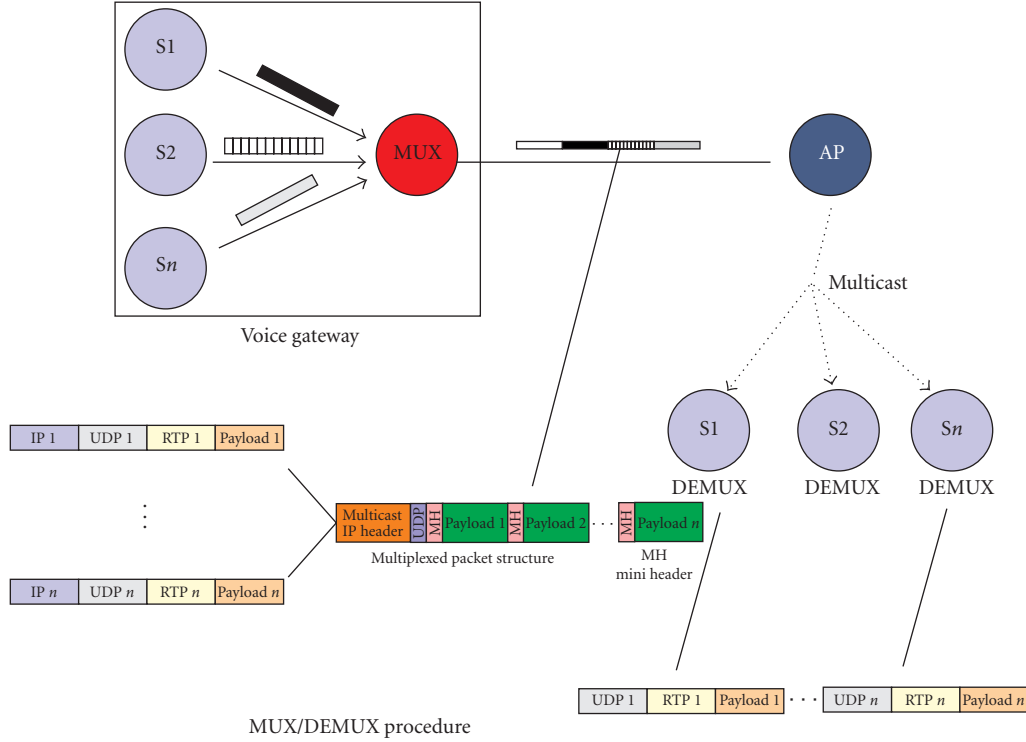


FIGURE 5: Mux/Demux procedure for MICE.

payload if more time slots are available in the CFP. Any packets remaining at the end of the CFP period will be dropped as in the ordinary ICF case.

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

This section presents some simulation results to compare the proposed MICE with the existing schemes. Using the information provided in tables and equations in the previous sections, the call capacity (number of simultaneous voice calls) for the different schemes has been calculated. Figure 4 shows a comparison between ICF, basic PCF, and MICE. In this simulation, the CFP is taken as 15 milliseconds and frame repetition interval as 20 milliseconds. Figure 6 represents loss ratio as a function of the number of simultaneous voice calls. As it is evident from the plot, the proposed MICE scheme has the lowest loss ratio for a given number of simultaneous calls (the region of interest is the one corresponding to loss ratio of 1% or less). A more important measure of the efficacy of scheme is the number of simultaneous calls while maintaining the QoS requirement. Generally, a loss ratio of

TABLE 3

	N_{\max} (simulated)		
	PCF	ICF	MICE
15 ms	6	30	39
17 ms	7	37	46
19 ms	10	42	53

less than 0.01 results in acceptable QoS. For this loss ratio, MICE can support a larger number of simultaneous calls as compared to basic ICF. Table 3 lists the call capacities of the various schemes for different values of CFP interval. Figure 7 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.

The simulation parameters used for the above results are briefly explained below.

- (1) In these simulations, the G.711 codec has been assumed, and this results in corresponding payload size of 160 bytes (packetization interval of 20 milliseconds).
- (2) The superframe size is dependent on the packetization interval of the codec. The G.711 codec, however, does not constrain the packetization interval. We have chosen 20 milliseconds (which correspond to the packetization interval of a lot of popular codecs) as the

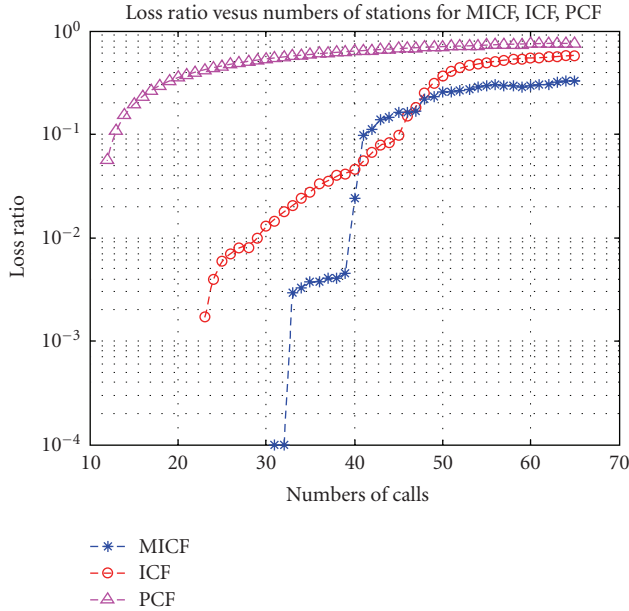


FIGURE 6: Comparison of PCF, ICF, and MICF for CFP = 15 milliseconds.

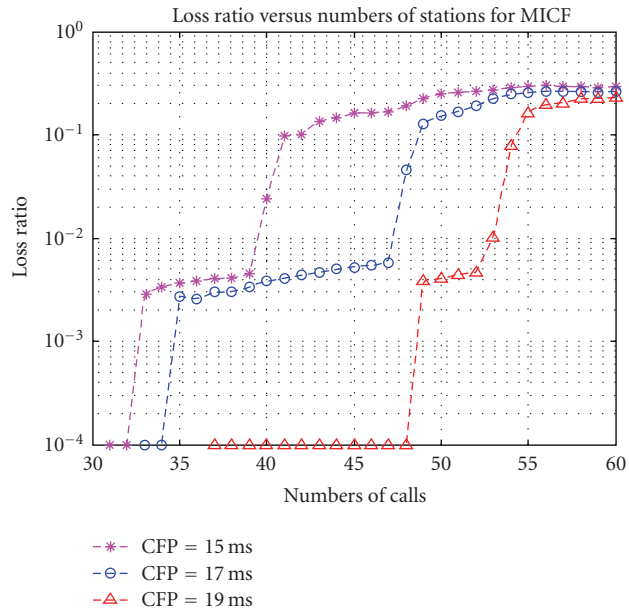


FIGURE 7: Effect of varying CFP period on MICF.

superframe size. In these simulations, CFP interval values varying from 15 milliseconds to 19 milliseconds have been used.

- (3) Critical values of loss ratio and delay for maintaining QoS have been assumed to be 1% [9] and 60 milliseconds, respectively. Consecutive packet loss has also been constrained to less than 1% for maintaining QoS.

Some assumptions made during the simulations are as follows.

- (1) Hidden terminal problem is assumed not to be present (needed in PCF simulations).
- (2) All stations are assumed to have PCF mode capability.
- (3) Network and stations have been assumed to have capability to handle multiplexed and multicast packets.
- (4) No stations are in power save mode.
- (5) The simulation assumes an 802.11b DSSS physical layer at the bottom of the protocol stack.
- (6) Traffic patterns are assumed to be the ones that correspond to the BSS having reached steady state.

6. CONCLUSIONS

This paper proposes a scheme for increasing call capacity of voice traffic. There is a difference between the electronic version and manuscript of Figure 5. Please check. The ICF technique which leads to a large call capacity has been modified to increase the call capacity further. The proposed MICF scheme improves the performance by further 30% (refer to Figure 6, where for loss ratio of 1%, the number of simultaneous calls for MICF is 39 as opposed to 30 for ICF; these numbers are also listed in Table 3). 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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