Service Integration Multiple Access (SIMA)

A Protocol for Supporting Voice and Data in Wireless LANs

Apichan Kanjanavapastit[1,2] and Hassan Mehrpour[2]
[1]*Department of Telecommunication Engineering, Mahanakorn University of Technology, Bangkok 10530, THAILAND. E-mail: apichan@ee.unsw.edu.au*

[2]*School of Electrical Engineering and Telecommunications, University of New South Wales, Sydney 2052, AUSTRALIA. E-mail: h.mehrpour@unsw.edu.au.*

Abstract: *We propose a wireless medium access control (MAC) protocol based on IEEE 802.11 called Service Integration Multiple Access (SIMA) for supporting voice and data transmissions over wireless local area networks (WLANs). This proposed protocol is a hybrid protocol between time division multiple access (TDMA) and carrier sense multiple access with collision avoidance (CSMA/CA). Since voice traffic generated by some standardized encoding schemes is classified as constant-bit-rate (CBR) traffic, we use TDMA protocol to support the transmission of the CBR voice traffic. On the other hand, CSMA/CA is still used to support the transmission of data traffic. The maximum number of voice calls in the same local area network when using various standardized voice encoding schemes is computed. In addition, the total transmission delay between terminals is discovered.*

Key words: IEEE 802.11, TDMA, Voice.

1. INTRODUCTION

The research area in real-time traffic transmission over wireless local area networks (WLANs) has been much interest in the past few years. One of the key drivers of this research field is the deployment of the IEEE 802.11 wireless LAN standard [1]. Real-time voice traffic can be transmitted in the IEEE 802.11 network by using point coordination function (PCF) in which polling technique is used to support the voice transmission. However, the transmission of data traffic uses distributed coordination function (DCF), which actually is a protocol named carrier sense multiple access with

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collision avoidance (CSMA/CA). Many studies of voice transmission over IEEE 802.11 have been performed. The effect of overhead for voice frame on the number of possible voice conversations has been studied in [2]. Discovery of various voice payload sizes and effect of polling scheme have been performed in [3]. In addition, an analysis of voice transmission in IEEE 802.11 has been carried on in [4].

Normally, some standardized voice encoding schemes always generate fixed-length voice packets in certain interval. The voice traffic generated by these schemes can be classified as constant-bit-rate (CBR) voice traffic. The CBR voice traffic transmitted using polling technique as used in IEEE 802.11 may lead to channel inefficiency resulted from large overheads of the polling packets. Although packing a number of voice packets generated within an interval into one large-sized packet as proposed in [4] can reduce the inefficiency, voice quality might be degraded due to a chunk of combined voice packets lost in a wireless medium. In addition, combining CBR voice traffic generated within a period might increase delay and delay jitter. Therefore, we introduce a wireless medium access control (MAC) protocol named Service Integration Multiple Access (SIMA), which is based on IEEE 802.11, for supporting CBR voice traffic generated from a standardized voice encoding scheme. We use time division multiple access (TDMA) to establish a circuit-switched connection-oriented service for the CBR voice traffic. CSMA/CA, however, is still used since it supports the transmission of data traffic. Therefore, in this paper, we only concentrate on the performance of voice transmission in SIMA. To discover the effect of various standardized voice encoding schemes, we analyze the maximum number of voice calls in the same local network when using G.729 and two encoding rates of G.723.1. We also compute the total delay to transmit voice packets in two directions within the same local network.

The paper is organized as follows. Section 2 presents SIMA frame structure and access method. Section 3 describes delay components in voice encoding scheme. Analytical results of the protocol with different voice encoding schemes are provided in Section 4. Conclusion is presented in Section 5.

2. SERVICE INTEGRATION MULTIPLE ACCESS

2.1 Frame structure

To support voice and data traffics, channel transmission time of SIMA is separated into two periods: contention-free period (CFP) and contention period (CP), as IEEE 802.11. These two periods can be grouped into superframe structure as shown in Figure 1. During a CFP, voice traffic is transmitted using TDMA. Thus channel transmission time in a CFP is divided into contiguous time slots. The number of TDMA slots per CFP is set to a certain number; the length of every CFP is fixed. Like IEEE 802.11, CSMA/CA protocol is still used to support data transmission in a CP. Some studies use voice activity detection (VAD) to improve channel efficiency by allowing other terminals to use temporarily idle TDMA slots during silent periods. However, we did not consider the exploitation of the VAD in that way because it is difficult for the other terminals to detect the silent intervals especially in wireless environment. The other terminals may misinterpret the status of a temporarily unused slot; this results to packet collision in that slot. On the other hand, we designed that a terminal who has established a voice call has to use its own TDMA slot if it has data traffic to transmit. Hence, the data traffic of that terminal must be fragmented into small data packets in which the lengths of the packets are equal to the payload size of the voice traffic. This can improve channel efficiency and also reduce the congestion of data traffic in a CP.

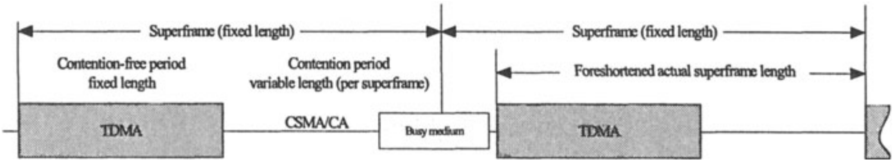

Figure 1. Superframe structure

To keep the delay time between two consecutive voice packets (delay jitter) from the same terminal within an acceptable range, the interval time of a superframe should be set according to the interarrival time of the voice packets depending on voice encoding scheme to be used in the protocol. Note that the interarrival time can be known from the total voice encoding delay as discussed in Section 3. Like IEEE 802.11, however, the medium may be busy at the end of a CP; this results to a foreshortened superframe length for the next period as shown in Figure 1. In this case, the quality of voice at a called terminal might be affected by the varied delay jitter.

2.2 Access protocol

Each terminal can establish a call by notifying a called terminal via an access point (AP) during a CP. After the called terminal has confirmed the call setup, the access point assigns an uplink TDMA slot and a downlink TDMA slot in a CFP to the caller terminal (and called terminal if they are in the same local network) if some slots are still available. On the other hand, voice calls are blocked if all TDMA slots are occupied. The TDMA slot length is equal to voice payload size depending on voice encoding scheme to be used plus some headers and a short interframe space time (T_{sifs}), as used in IEEE 802.11. Each terminal who has established a call can transmit voice packets in the assigned slot for future transmission until the call is terminated.

After assigned some slots to voice calls, the unused TDMA slot may be used to transmit data traffic by any terminal. Data transmissions in these available TDMA slots use slotted ALOHA random access protocol. A terminal sends a data packet in any of the available TDMA slots, which is randomly selected from a discrete uniform distribution, to a destination terminal. If the data packet is successfully received, the destination terminal transmits an acknowledgement packet back to the source terminal in the next CP. If the source terminal cannot receive the acknowledgment packet, it attempts to send the data packet again in the next CFP. Note that the successful source terminal cannot reserve the random selected slot in the subsequent CFP for subsequent data transmission.

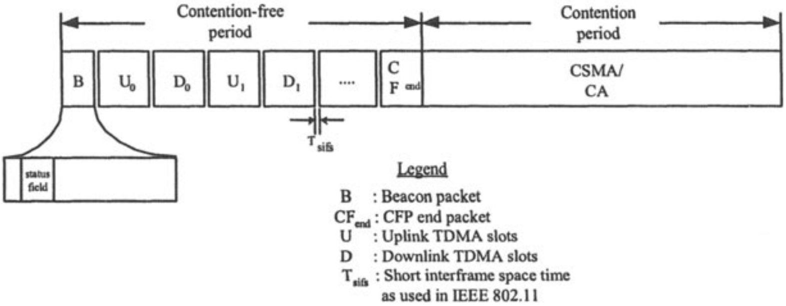

Figure 2. Operation of SIMA protocol

As illustrated in Figure 2, the access point broadcasts a beacon packet to begin a CFP. The use of beacon packet can eliminate the problem of shifting physical position of TDMA slots due to the medium being busy at the end of a CFP. This implies that the position of TDMA slots in SIMA is defined as a logical position instead of physical position. In addition, the beacon packet is transmitted to allow any terminal to locate the position and identify the status of the TDMA slots. In this way, any terminal can transmit data packets in the available TDMA slots as mentioned. The number of TDMA slots in a CFP and the status of TDMA slots are indicated in a status field as shown in the figure. We modified the ordinary beacon packet used in IEEE 802.11 by adding some extra bits to be the status field. Some bits out of the total extra bits are used to indicate the number of TDMA slots in a CFP while the remaining bits are used to specify the status of TDMA slots.

3. DELAY COMPONENTS IN VOICE COMPRESSION

To determine the superframe length of SIMA, delay components in voice compression must be discovered. In general, analog voice signal is digitized into pulse code modulation (PCM) signals by a PCM encoder. Then the PCM samples are passed to a compression algorithm, which analyzes the samples in a block format. The size of a block depends on the compression technique to be used. Normally, a processor is used to compress a block of PCM samples. Hence, there is a delay time called processing delay occurred by the operation of the processor. This delay varies with the processor speed. However, we considered processing to be fast enough so that processing delay could be ignored. In addition to the processing delay, since the compression algorithm must have some knowledge of the PCM samples in block N+1 to accurately reproduce sample in block N, an additional delay called lookahead delay could be occurred. Therefore, the encoding delay consists of the sum of the lookahead delay and the processing delay. After compressed, the encoded/compressed bit stream is packetized incurring a packetization delay. This delay is a function of the sample block size and the number of blocks placed in a single voice packet. Therefore, the total voice coding delay is the summation between the encoding and the packetization delays. Table 1 shows some voice encoding schemes and their parameters.

Table 1. Voice encoding standards

Standard	Coding type	Bit rate (kbps)	Block size (ms)	Look ahead (ms)	Packetization delay (ms)	Total delay (ms)	Payload (bytes)
G.711	PCM	64	0.125	0	20	20	160
G.726	ADPCM	32	0.125	0	20	20	80
G.729	CS-ACELP	8	10	5	20	25	’20
G.723.1	MP-MLQ	6.3	30	7.5	30	37.5	24
G.723.1	MP-ACELP	5.3	30	7.5	30	37.5	20

4. ANALYTICAL RESULTS

In this section, we analyzed the maximum number of voice calls in which the caller terminal and called terminal are in the same local area network. In addition, the total transmission delay between the terminals was computed.

Table 2. Parameters for analysis

Parameter	Symbol	Unit	Value
Duration of superframe	T_{SF}	ms	37.5(G.723.1) and 25(G.729)
Transmission rate	R	Mbps	2 (FHSS) and 11 (DSSS)
Transmission rate of PHY header	R_p	Mbps	1
Higher layer and MAC headers	H	bytes	57
Physical layer header	P	bytes	16(2Mbps) 24(11Mbps)
Max. number of calls	N_c	-	Computed in section 4.1
Min. value of CP	$T_{CP\text{-}min}$	ms	Computed in section 4.1
Max. size Service Data Unit	S_{maxSDU}	bytes	2304
Fragment threshold size	f	bytes	2304
Beacon size	B	bytes	73
CF-End size	CF_{end}	bytes	24
Voice payload size	V	bytes	24 and 20
SIFS interval	T_{sifs}	ms	0.028
A slot time	T_{slot}	ms	0.05
Time to handle a voice call	T_V	ms	Computed in section 4.1
Transmission time of RTS (20 bytes)	T_{rts}	ms	Computed in section 4.1
Transmission time of CTS (14 bytes)	T_{cts}	ms	Computed in section 4.1
Transmission time of ACK (14 bytes)	T_{ack}	ms	Computed in section 4.1

4.1 Maximum number of voice calls in the same network

We derived an equation to find the maximum number of calls in a CFP for two encoding schemes: G.729 and G.723.1. The parameters for this analysis are shown in Table 2. Since IEEE 802.11 states that the header of IEEE 802.11 physical layer must be transmitted at the rate of 1 Mbps, the transmission time for sending the physical layer header should be considered separately from the transmission time of the MAC information. We followed the parameters as used in [4] for the analysis except the beacon packet size, which was adapted to use with our protocol by adding extra 264 bits in a beacon packet to be the status field as mentioned in Section 2.2. Eight bits out of 264 bits are used to indicate the number of TDMA slots in a CFP. The remaining 256 bits are used to specify the status of each TDMA slot whether it is available or occupied. Moreover, we still used the same MAC header size, as stated in [4], for information transmission in the two periods.

If the caller and the called terminals are in the same local network, the total number of TDMA slots for voice communication is four since the communication from a terminal is sent first to the access point and then from the access point to the other terminal. Time to handle a call in a CFP is

$$T_V = \frac{4\cdot(V+h)}{R} + \frac{4\cdot P}{R_p} + 4\cdot T_{sifs} \tag{1}$$

As [4], to find the maximum number of calls, the minimum period of CP, T_{cp}, is required. The time needed for a maximum packet size stretch, $T_{cp-stretch}$, is also included in the calculation of T_{cp}. Therefore,

$$T_{cp} = T_{cp-\min} + T_{cp-stretch}, \text{ where} \tag{2}$$

$$T_{cp-\min} = 2T_{sifs} + 2T_{slot} + 8T_{ack} + T_{\max} \tag{3}$$

$$T_{cp-stretch} = T_{rts} + T_{sifs} + T_{cts} + T_{sifs} + T_{\max} \tag{4}$$

$$T_{\max} = (m-1)\left(\left\lceil \frac{f+h}{R} \right\rceil + \frac{P}{R_p} + T_{ack} + 2T_{sifs}\right) + \frac{S_{\max SDU} - f(m-1) + h}{R} + \frac{P}{R_p} + T_{ack} + T_{sifs} \tag{5}$$

where $m = \lceil S_{\max SDU} / f \rceil$, $T_{ack} = T_{cts} = \frac{20 \cdot 8}{R} + \frac{P}{R_p}$, and $T_{cts} = \frac{14 \cdot 8}{R} + \frac{P}{R_p}$

A beacon packet and a CF_{end} packet are sent in a CFP, therefore the overhead of these packets, T_{oh}, is also required to determine the maximum number of calls. Note that we let the number of beacon packets generated by the access point to have one packet per superframe. As mentioned in Section 2.1, the superframe length should be equal to the total encoding delay of the encoding scheme to be used in the protocol. To find the maximum number of calls, we used the following equation.

$$N_c = \frac{T_{SF} - T_{cp} - T_{oh}}{T_V} \tag{6}$$

As seen in Figure 3, G.723.1 provides the maximum call higher than G.729. Also, the maximum call when using G.723.1 at the coding rate of 6.3 kbps and 5.3 kbps is not much different. However, quality of voice provided by an encoding scheme to be used should also be considered.

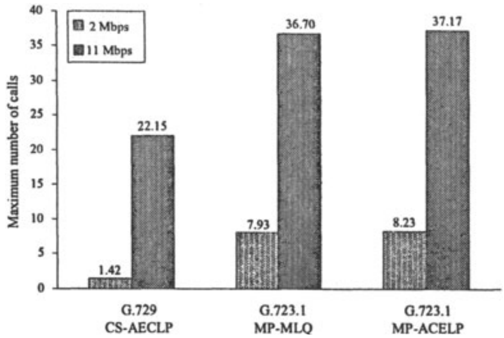

Figure 3. Maximum number of calls in various encoding schemes

4.2 Total delay of voice transmission

We computed the total delay of the voice transmission between two terminals (A and B) in the same network. When considering a typical voice packet generated, the delay suffered by this packet should include: (1) the time between its generation and the position of uplink slot, (2) the queuing time, (3) the transmission time, and (4) the propagation delay. The queuing time could be ignored because there is one voice packet per frame only. The propagation time was also neglected since the radio link is short. Therefore, the delay from terminal A to B (and vice versa) is:

$$\frac{T_{SF}}{2}+\frac{T_V}{2}\leq D_{A\leftrightarrow B}\leq\frac{T_{SF}}{2}+\frac{T_{cp-strech}}{2}+\frac{T_V}{2} \quad (7)$$

As seen, there will be no stretching period in the lower bound. Since the superframe is fixed, the average time of the generation time to the position of uplink slot is $T_{SF}/2$. Note that the position of the uplink slot was assigned in uniform distribution. Also, the position of downlink slot was assumed to locate next to the position of the uplink slot. When assuming there will be stretching period in the upper bound, the delay from the first component is $T_{SF}/2+T_{cp-strech}/2$. As [4], given the delay jitter, the maximum total delay for the two directions of the call is shown in equation (8). As seen, the smaller superframe length, the smaller total delay of the voice call.

$$TD_{A\leftrightarrow B}^{\max}=D_{A\leftrightarrow B}^{\max}+\left(D_{A\leftrightarrow B}^{\max}-D_{A\leftrightarrow B}^{\min}\right) \quad (8)$$

5. CONCLUSION

Service integration multiple access (SIMA) was proposed. Voice traffic generated by voice encoding scheme is transmitted using TDMA, while data transmission uses CSMA/CA. We analyzed the maximum call for some encoding schemes. We also computed total transmission delay of the call.

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