

Voice Call Capacity Model for Hybrid Multi-Channel Protocol over Multi-Hop Multi-Channel Multi-Radio Wireless Mesh Networks

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Abstract—Multi-channel multi-radio wireless networks improve network performance by enabling more simultaneous transmissions. Among multi-channel multi-radio protocols, the hybrid multi-channel approach has been shown to be efficient in providing higher system throughput.

Voice over internet protocol (VoIP) service is expected to be widely supported in wireless networks, though its capacity has not been studied in relation to the emerging hybrid multi-channel protocol over multi-channel multi-radio wireless mesh networks (MCMR WMNs). To enabling VoIP service over MCMR WMNs, the call capacity model is essential. This is because the accuracy of the call admission control highly depends on how well the call capacity over the multi-hop MCMR WMN is inferred.

In this paper, we introduce a new voice call capacity model of hybrid multi-channel protocol on multi-hop MCMR WMNs. Both experimental and simulation results demonstrate that the proposed call capacity model accurately estimates the voice call capacity on a multi-hop MCMR WMN.

I. INTRODUCTION

Voice over Internet Protocol (VoIP) services have been proliferated, and the widespread wireless LAN (WLAN) supports VoIP services in homes and offices [1]. For example, Skype, one of the most popular VoIP applications, is used by over 521 million users worldwide [2]. However, the WLAN has some limitations, such as small service coverage and difficulty of deployment [3]. To resolve these limitations, the emerging wireless mesh network (WMN) is currently attracting.

The WMN appends a routing functionality to each mesh node, thereby eliminating the wired backbone, and enabling it to be deployed easily. But WMNs based on single channel single radio IEEE 802.11 has the capacity limitation problem. To mitigate this capacity limitation problem of a single channel single radio WMN, each node in the WMN is equipped with multiple radios tuned to non-overlapping channels. This allows that a node can effectively transmit and receive simultaneously. Many multi-channel multi-radio (MCMR) protocols have been proposed to boost the capacity of the networks. The main issue of MCMR protocols is to coordinate nodes for assigning the set of channels to the radios in a node. According to the coordination of radios and channels, these MCMR protocols can be categorized as three approaches, namely common

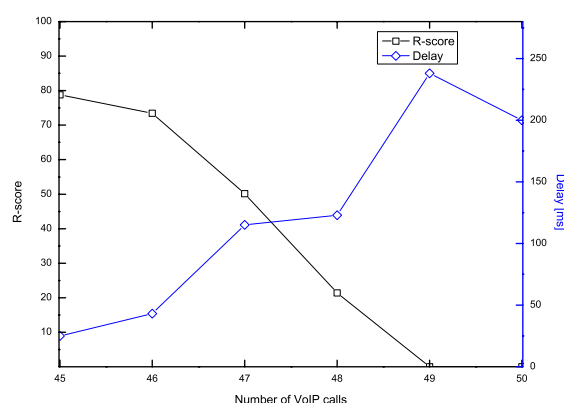


Fig. 1. Experimental results of hybrid multi-channel protocol with G.729 voice codec (sample period of 20 ms) on a 2-hop multi-channel multi-radio wireless mesh network (IEEE 802.11a at data rate of 54 Mbps). The call capacity threshold is reached delay increases dramatically and R-score goes down.

control channel approach [4], static channel approach [5], [6], and hybrid channel approach [7].

In the common control channel approach, a sender negotiates the timing and the data channel to use with the intended receiver via the dedicated control radio tuned to the common control channel before transmitting packets. The static channel approach requires a careful assignment of channels to its radios to achieve network connectivity based on significant topology knowledge. In the hybrid channel approach, each node can switch across channels for achieving network connectivity during multi-channel operation. Among these approaches, the hybrid multi-channel approach has been shown to be efficient in providing higher system throughput [7]. So, we select the hybrid multi-channel protocol (HMCP) as a baseline protocol in this paper. HMCP leverages the network performance, however, supporting delay sensitive realtime VoIP applications over WMNs is still challenging. To measure performance of HMCP over an MCMR WMN, we build the testbed operating

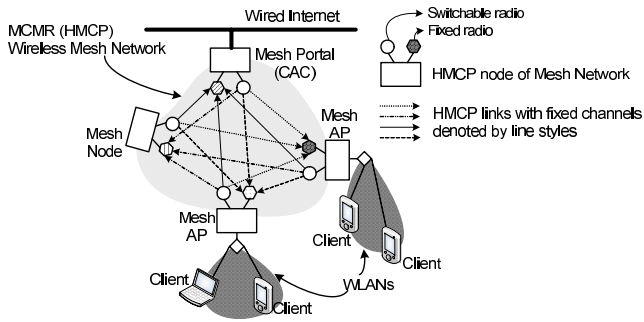


Fig. 2. Example of an MCMR WMN with HMCP

as HMCP in our building (our testbed described later in Section 4.1). Based on our testbed, we check a network delay and a voice call quality defined by R-score [8] on a 2-hop WMN. As shown in Figure 1, we notice that at the point the delay starts increasing suddenly around 46 calls, voice call quality also falls rapidly. This experiment demonstrates that a call admission control is needed to maintain a good call quality¹ over multi-hop MCMR WMNs. To maintain the network performance of an MCMR WMN stable and to provide good VoIP services, we must do call admission control with the accurate call capacity model. The accuracy of network quality management highly depends on how well the call capacity of the wireless network is inferred.

So far, several analytical models have been reported to estimate the voice call capacity of a wireless network. The authors of [1] and [9] proposed VoIP call capacity models of a WLAN considering QoS requirements in terms of the end-to-end delay and packet loss. In [10], an analytical model to estimate call capacity utilization with the offered VoIP calls in a single channel multi-hop WMN is presented. In [11], a call admission control scheme of a multi-rate multi-radio multi-channel network is reported, but this analysis is for static channel approach protocols [5], [6]. None of earlier studies theoretically analyzed the voice call capacity of HMCP over a multi-hop MCMR WMN. Thus, in this paper, we propose a call capacity model for HMCP in a multi-hop MCMR WMN and also validate this model with multi-hop scenarios via both testbed experiments and NS-2 simulations [12].

The reminder of the paper is organized as follows: In Section 2, we provide a system model for HMCP and VoIP over a multi-hop MCMR WMN. We develop our call capacity model of HMCP with considerations of packet collision, traffic overhead, and channel switching delay in Section 3. Section 4 shows the description of the testbed, experimental and simulation results. We conclude the paper in Section 5.

II. SYSTEM MODEL

As shown in Figure 2, mesh nodes are a basic element of a WMN, and they have a routing functionality to connect to each other. A mesh access point (AP) has an additional function to

¹In this paper, a call quality is defined by the R-score. A good call quality means that R-score is over 70 in this paper.

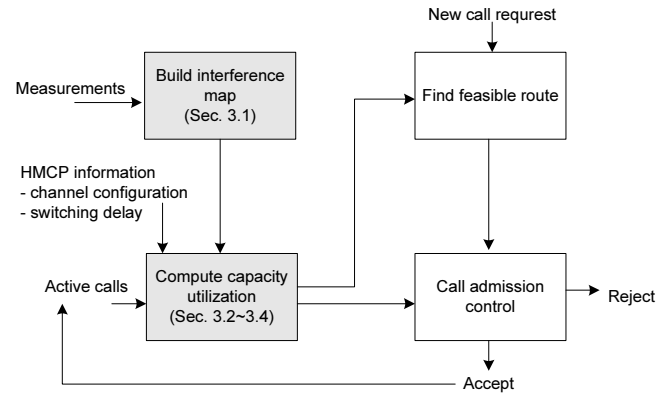


Fig. 3. The architectural block diagram proposed in this paper

the mesh node for serving clients. A mesh portal is also a gateway to the Internet. A call admission controller can be located in the mesh portal. The call admission controller uses a call capacity model to decide acceptance of the arriving call with a route between source and destination based on interference map and traffic load information. This centralized configuration enables to manage stable VoIP call management easily and many commercial WLAN and WMN platforms have followed a centralized manner [13]–[15].

In a typical scenario of VoIP communications on a WMN, a client in the WMN makes or receives a VoIP call. This call initiation starts from communicating with a session initiation protocol (SIP) server located in the mesh portal. The SIP server authorizes the arriving call and resolve call's QoS requirement information. Based on this QoS requirement, a call admission controller decides whether the arriving call is accept or not. Finally, the route for the arriving call is established on the WMN. At the call admission step, the call admission controller uses a call capacity utilization model to decide acceptance of the arriving call with a route between source and destination. The overall architectural block diagram for supporting VoIP calls over an MCMR WMN is presented in Figure 3. We touch only the gray blocks, which are basic components for supporting VoIP services. The route computation can be done to call capacity constraints based on the proposed call capacity model (Refer to [10]). Thus, we focus on a call capacity utilization model, which is used for deciding acceptance of the arriving call.

Each node in an MCMR WMN operates as HMCP. An HMCP node uses two radios to communicate with its neighboring nodes. As shown in Figure 2, one is a *fixed radio* to receive data from its neighboring nodes via its own fixed channel. The other is a *switchable radio* to transmit data to neighboring nodes via the neighboring nodes' fixed channel. Each HMCP node selects the two-hop least used channel as its fixed channel [7]. An HMCP node advertises its fixed channel information via broadcasting HELLO packets over all available channels and utilizes this information to lookup the fixed channel of the destined node of the packet. If an HMCP node has traffic for two or more neighbors with different

fixed channels, then channel switching occurs to transmit data. Current IEEE 802.11 chipset supports channel switching within 25~35 μ s [16]. In this paper, terms interfaces and radios are used interchangeably.

III. CALL CAPACITY UTILIZATION MODEL

From the simple experiment based on our MCMR WMN testbed, we notice followings. As shown in Figure 1, the average delay increases with load, but suddenly rises when the offered load reaches close to the call capacity. In order to meet the QoS for a given set of active VoIP calls, the load on each node in an MCMR WMN must meet a capacity bound. In order to ensure the above condition, we determine the call capacity utilization at every node for a given set of active calls.

We define the normalized capacity utilization² at node i , (c_i), which is the total bits/second traffic transmitted, received, heard, or channel switching at node i , normalized to the nominal link capacity. Throughout this paper, all voice traffic flows are bi-directional constant bit rate (CBR) traffic flows. The capacity analysis with considering data, video, and other elastic traffic types is our future work.

A. Interference Map

Modeling call capacity utilization starts from characterizing the interference between nodes and building an interference map. We use the *carrier sense factor* or *csf* to characterize the interference between a given pair of nodes [10]. For given nodes x and y , csf_x^y (carrier sense factor of x with y) is defined by

$$csf_x^y = \frac{x's \text{ actual transmission rate in presence of } y}{x's \text{ actual transmission rate without } y}, \quad (1)$$

where both x and y attempt to transmit data with the maximum possible rate. This csf_x^y has values between 0.5 and 1. If a value of csf_x^y is 0.5, then both x and y sense each other perfectly, and, if the value of csf_x^y is 1, then neither x and y can hear each other. To simplify our analysis, we assume that csf_x^y is not frequency selective, but this assumption can be easily relaxed by adding channel dimension ($csf_x^y(c)$) to characterize the frequency selective interference. The method for measuring *csf* was introduced in [10].

B. Original Offered Load

An original offered load (l) at each node is normalized by the call link capacity between nodes. The call link capacity model for a WLAN (single channel single radio environment) was reported in [1], and we modified it to model the single hop link capacity in an MCMR WMN. In a multi-channel multi-radio environment, the call link capacity between nodes can be expressed as

$$N_{x,y} = \left\lfloor \frac{1}{R[f_{x,y} \cdot (T_{voice.pkt}) + (T_{slot} \cdot CW_{min}/2)]} \right\rfloor, \quad (2)$$

where $T_{voice.pkt}$ is $T_{payload}$ + short interframe space (SIFS) + T_{ack} + distributed coordination function interframe space

²For brevity, we write “capacity utilization” for “normalized capacity utilization.”

(DIFS). $T_{payload}$ is the time to transmit voice data consisting of a physical layer convergence protocol (PLCP) header, a preamble, a MAC header, a frame check sequence, real-time transport protocol/user datagram protocol/IP headers, and a payload. T_{ack} is the required time for sending an acknowledgement (ACK) frame, and T_{slot} is a slot time. R is the number of packets generated by an encoder per second, and CW_{min} is the minimum contention window size. The value of $f_{x,y} \in \{1, 2\}$ is determined by the configuration of the channel assignment in an MCMR WMN. $f_{x,y}$ is 1 if node x 's fixed channel is different from node y 's fixed channel. Otherwise, $f_{x,y}$ is 2 because a VoIP call consists of two reverse-direction flows, and these flows interfere each other.

The original offered load for VoIP calls at node i to node k using fixed channel c is expressed as

$$l_{i,k}(c) = \frac{n_{i,k}}{f_{i,k} \cdot N_{i,k}}, \quad (3)$$

where $n_{i,k}$ is the number of calls through the link from node i to node k using fixed channel c . For instance, for a two-hop call through A-B-C nodes, the original offered loads on A and C are $1/(f_{A,B} \cdot N_{A,B})$ and $1/(f_{C,B} \cdot N_{C,B})$ respectively. However, the original offered load on the intermediate node is $1/(f_{B,A} \cdot N_{B,A}) + 1/(f_{B,C} \cdot N_{B,C})$ due to forwarding traffic in both directions. Thus, the total original offered load at node i can be represented by

$$l_i = \sum_c \sum_k l_{i,k}(c) = \sum_c l_i(c), \quad (4)$$

where $c \in \{ \text{available channels} \}$ and $k \in \{ \text{neighbors of } i \}$.

C. Actual Traffic Load

Each node attempts to transmit the amount of an original load. However, the node can transmit extra traffic due to retransmissions by packet collisions. For VoIP traffic, the request-to-send/clear-to-send (RTS/CTS) mechanism is typically disabled because the VoIP payload size is small, and RTS/CTS is an overhead. For example, the payload size for G.729 codec is 20 bytes with a 20ms sample interval. Collisions occur at a receiver if the receiver has an interferer in its carrier sensing range, which is outside the carrier sensing range of the transmitter (hidden terminal problem). To model this hidden terminal problem, we can justify conditions for the hidden terminal problem. Node i with an intended receiver k , and node j are hidden from each other if $csf_i^j = 1$ and $csf_k^j < 1$. The proportion of node j 's traffic that reached node k is $2 \cdot (1 - csf_k^j)$. Thus, we approximate the probability of packet collisions at the receiver k with the transmitter i and the interferer j to be $l_{i,k}(c) \cdot l_j(c) \cdot 2 \cdot (1 - csf_k^j)$. The probability that a collision occurs twice or more for the same packet is relatively small [9], so we approximate the number of retransmissions of the MAC frame as 1. Finally, an actual traffic load (t) on the given channel c , which consists of the original offered load and extra traffic due to retransmissions, can be expressed as:

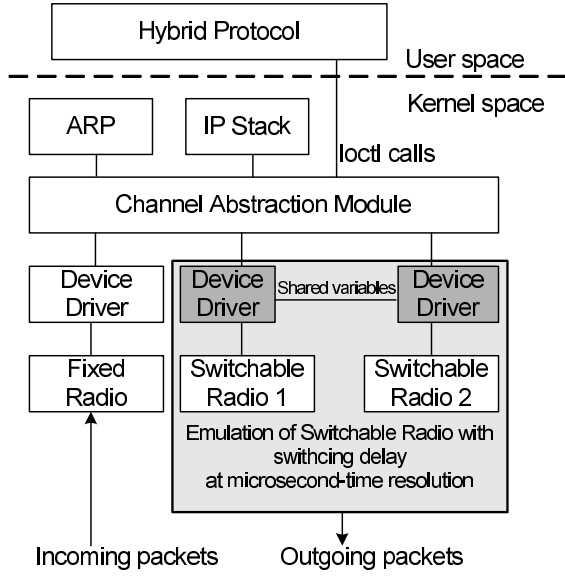


Fig. 4. System architecture of HMCP with microsecond-time switching delay (modified component in gray)

$$t_i(c) = \sum_k l_{i,k}(c) \cdot (1 + \sum_j l_j(c) \cdot 2 \cdot (1 - csf_k^j)), \quad (5)$$

where $k \in \{ \text{neighbors of } i \}$, and j satisfies $csf_i^j = 1$ and $csf_k^j < 1$. Total actual traffic load at node i can be presented as $t_i = \sum_c t_i(c)$.

D. Traffic Overheard and Switching Delay

The remaining components of the call capacity utilization at a node are traffic overheard and switching delay. During this time, the node cannot transmit data to its neighbors.

In a single channel environment, node i must hear a $2 \cdot (1 - csf_i^j)$ fraction of traffic from node j , where they satisfy $0.5 < csf_i^j < 1$. In a multi-channel environment, each node transmits data over the neighbor's fixed channels with channel switching. The fraction of time for staying on each channel at the node ($\frac{t_i(c)}{t_i}$) is proportional to the actual traffic load on each channel; thus, we can model the time for traffic overheard at node i on the given channel c as

$$o_i(c) = \sum_{i \neq j} t_j(c) \cdot 2 \cdot (1 - csf_i^j) \cdot \frac{t_i(c)}{t_i}, \quad (6)$$

where node i and node j satisfy $csf_i^j < 1$.

We also consider the channel switching delay of HMCP. If a node has traffic to neighboring nodes with different fixed channels, then the node should tune its switchable radio's channel to its neighbor's fixed channel to transmit data. At this time, channel switching occurs, and so does delay. This channel switching delay with a high data rate network system is not negligible. For instance, the channel switching delay is $35 \mu s$, and T_{voice_pkt} is $37.96 \mu s$ in IEEE 802.11a with a data rate of 54Mbps, a G.729 codec, and a sample period of 20 ms.

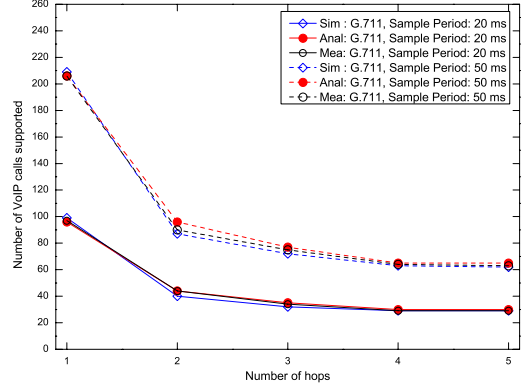


Fig. 5. Voice call capacity of HMCP with G.711 codec

As we mentioned above, the fraction of time for staying on each channel is proportional to the actual traffic load on each channel. Thus, the probability that the node did not stay on channel c can be denoted by $1 - \frac{t_i(c)}{t_i}$. Consequently, total time of switching delay at node i on channel c , $s_i(c)$, is expressed as:

$$s_i(c) = SW_{norm_delay} \cdot t_i(c) \cdot (1 - \frac{t_i(c)}{t_i}), \quad (7)$$

where SW_{norm_delay} is the normalized switching delay to the link capacity ($= \text{switching_delay} / [T_{voice_pkt} + (T_{slot} \cdot CW_{min}/2)]$).

By summing up all components of the call capacity utilization at node i , we can represent c_i as:

$$c_i = \sum_c t_i(c) + \sum_c o_i(c) + \sum_c s_i(c). \quad (8)$$

Based on (8), we can obtain the call capacity utilization at each node in an MCMR WMN. To check feasibility for supporting VoIP calls with the given route, the capacity utilization of each node, c_i , should be smaller than 1.

IV. PERFORMANCE EVALUATION

In this section, we validate the capacity utilization model of HMCP on multi-hop MCMR WMN scenarios using our testbed and NS-2 simulators [12].

A. Experimental Testbed and Simulator

To validate our call capacity model, we utilize an MCMR WMN testbed called Net-X, which is developed by the Wireless Networking Group at the University of Illinois at Urbana-Champaign (UIUC) [17]. Net-X is implemented by the daemon process in the user space and the device driver in the kernel space. In current Net-X system, however, the channel switching delay is millisecond-scale. The channel switching delay is determined by several factors such as, handling interrupts at the device level, tuning with the new channel. Actually, current IEEE 802.11 chipset provides tuning to the new channel within $25 \sim 35 \mu s$ [16]. In current Net-X system,

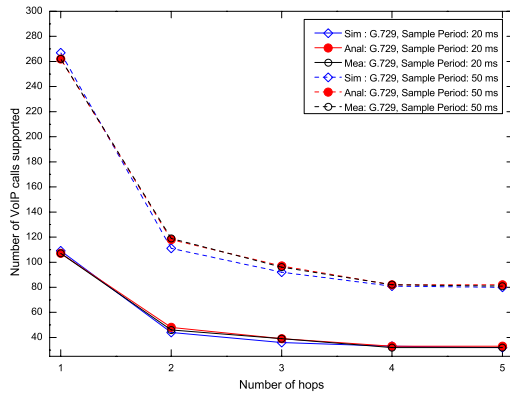


Fig. 6. Voice call capacity of HMCP with G.729 codec

controls of channel switching for HMCP is implemented at the device driver level. But if we implement these controls at the firmware level, the channel switching delay is significantly reduced theoretically. As shown in Figure 4, we emulate one switchable radio at microsecond-scale switching delay with two radios, since we have no right to modify firmware. The channel switching delay can be configured to the arbitrary value with this emulation technique. Consequently, each HMCP node has three radios consisted of one fixed radio and two radios for emulating for one switchable radio in our test bed.

Each node of our testbed has 3.0 Ghz central processing unit, one ethernet card, and three IEEE 802.11 a/b/g Atheros chipset-based interface cards with 3Com Omni-directional Workgroup Bridge Antenna. We run Linux kernel 2.4.26 with MadWifi drivers 0.94 [18] to control interface cards. To generate realtime VoIP traffic and measure the network performance, we use a traffic measurement tool called D-TIG [19]. To get precise experimental results, all nodes in our testbed synchronized by network time protocol (NTP).

We also implemented HMCP based on NS-2 and patched a realistic packet capture model. Table I gives system parameters for theoretical analysis, experiments, and simulations. As shown in Table I, the sample period of a voice codec determines the number of generated packets per second (R) at an encoder. HMCP nodes are deployed as a linear topology to make multi-hop communications. HMCP nodes have different fixed channels after the channel assignment of HMCP in this network configuration [7]. After that, we analyzed the maximum number of calls supported with the required QoS level³ using our capacity utilization model, experiments and simulations. The hop count for VoIP calls varies from 1 to 5 on an MCMR-WMN.

³The required QoS level for a voice call in terms of R-score is over 70. To meet the required QoS level, the one-way delay should be less than 200ms and the packet loss rate should be less than 5.

B. Performance Results

We plot the performance results from the theoretical analysis, experiments and simulations. The result of the maximum number of calls supported for G.711 codec is in Figure 5. This figure demonstrates that our call capacity utilization model estimates the call capacity for G.711 codec with 20 and 50 ms sample periods quite well within 6% of the actual capacity obtained by experiments and simulations. The call capacity of HCMP with G.711 shapely decreases as the number of hops increases because each node contends for transmitting bi-directional traffic flows at each hop with channel switching. For G.729 codec, the proposed capacity utilization model predicts the call capacity within 4% of the capacity from both experiments and simulations, as shown in Figure 6. The performance results demonstrate that the proposed call capacity model provide a tight upper bound of the voice call capacity.

V. CONCLUSIONS AND FUTURE WORKS

In this paper, we addressed the challenge issue for providing VoIP service on MCMR WMNs. To maintain QoS of VoIP calls, we must do call admission control. The accuracy of the call admission control highly depends on how well the call capacity over the multi-hop MCMR WMN is inferred. We develop the voice call capacity model of HMCP, which is a representative hybrid multi-channel protocol on MCMR WMNs. We also validate the proposed capacity model via both experiments and simulations. The performance results show that our call capacity model estimate a tight upper bound of the actual call capacity obtained from experiments on our testbed.

Based on our call capacity model, we can check the feasibility for supporting VoIP calls with the given configurations of HMCP over MCMR WMNs. This means that we can find the feasible route of the new incoming call by checking the feasibility with possible routes. Proposing the efficient computation scheme for finding feasible route of the incoming call can be our future work (*find feasible route* and *call admission control blocks* in Figure 3). We also will address our call capacity model with considerations of heterogeneous traffics (i.e., data, video, and other elastic traffic) as our future work.

VI. ACKNOWLEDGMENTS

This research was supported by the MKE (The Ministry of Knowledge Economy), Korea, under the ITRC(Information Technology Research Center) support program supervised by the NIPA(National IT Industry Promotion Agency) (NIPA-2010-C1090-1031-0004). Also, this research was supported by Basic Science Research Program through the National Research Foundation of Korea(NRF) funded by the Ministry of Education, Science and Technology(2010-0008149).

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TABLE I
PARAMETERS FOR IEEE 802.11A, HMCP, AND VOICE CODECS

IEEE 802.11a			
Data rate	54Mbps	T_{slot}	$9\ \mu s$
SIFS	$16\ \mu s$	DIFS	$34\ \mu s$
CW_{min}	16	CW_{max}	1024
T_{ack}	$26.1\ \mu s$	$T_{payload}$	$35+(\text{payload}\cdot 8/54)\ \mu s$
TxPw	6 dBm	RxPw threshold	-65 dBm
CSPw	-91 dBm	Capture threshold	24.56 dB for 54Mbps
Path-loss model		Two-ray ground reflection model	
Hybrid Multi-Channel Protocol (HMCP)			
# of CHs	11	Switching delay	$35\ \mu s$ [16]
Channel assignment of HMCP		Two-hop least used channel [7]	
Frequently used voice codecs			
Voice codec (codec bit rate)		G.711 (64kbps)	G.729 (8kbps)
Sample period		Payload (byte)	Payload (byte)
20ms (R = 50)		160	20
50ms (R = 20)		400	50

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