Analysis of a Multi-channel Wireless MAC Protocol for Access Point Based Networks

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Abstract

We present the analysis of a multi-channel MAC protocol for wireless networks. We consider wireless networks in PMP (Point to Multi-point) configurations, with a base station (access point) and a number of clients associated with it. We focus on contention based operation of nodes in such networks using a multichannel MAC protocol, adapted from [4]. Using a slotted-time model, we derive expressions for the service time distribution of the packets in the network and also derive approximate expressions for queuing delay under Poisson traffic, when the contention window is of fixed size. We further extend this analysis to incorporate exponential backoff of the contention window size. Simulations are performed using a multichannel MAC developed for NS-2. We note that the analysis and simulation match well for moderate to high values of traffic intensity. Our analysis enables us to capture the impact of various system parameters such as frame time, number of clients, contention duration etc. on queuing delay.

This analysis can be extended to IEEE 802.11 DCF in access point mode of operation and contention based IEEE 802.16 based networks.

1 Introduction

Most wireless MAC standards specify multi-channel operation, for example IEEE 802.11 and IEEE 802.16 which is OFDM/OFDMA based. Multi-channel operation utilizes multiple orthogonal frequencies for communication between nodes in a network. Using multiple channels, higher network throughput can be attained as multiple transmissions can take place concurrently. As a result, throughput and delay for a multi-channel network show considerable improvement over networks utilizing a single channel.

IEEE 802.11 standard for wireless LAN [1] provides multiple channels available for use. The IEEE 802.11b physical layer (PHY) has 14 channels, 5MHz apart in frequency. However, to be totally non-overlapping, the frequency spacing must be at least 30MHz. So channels 1, 6 and 11 are typically used for communication in current implementations, and thus we have 3 channels for use. IEEE 802.11a provides 12 channels, 8 in the lower part of the band for indoor use and 4 in the upper part for outdoor use. However,the MAC protocol of IEEE 802.11 Distributed Coordinate Function (DCF) is designed for sharing a single channel between hosts. Moreover, the trans-receivers can either transmit or receive on a single channel at any instance of time. This gives rise the additional problem of *Multi-Channel Hidden Terminal* (situation when transmitting and receiving nodes are on separate channels, [4]). Hence, a single channel MAC does not work as efficiently in a scenario where nodes can switch channels dynamically.

In the IEEE 802.16 standard ([12]), of the three different PHYs specified in the standard, OFDM multi-access (OFDMA) is likely to emerge as the most preferred PHY supporting all usage models, which is multiple carrier based. One of the important reasons is due to the superior performance in multi-path fading channels. While IEEE 802.16 primarily defines a contention free MAC, there are some instances where contention based operation is used. For example, initial ranging for network entry and bandwidth request are contention based.

There have been a number of multi-channel MAC protocols proposed in the literature, refer [4] or [10] and the references therein for more details. While some analysis has been done for a multi-channel MAC in [10], we focus on a different protocol. We analyze in detail a contention based multi-channel MAC protocol, adapted from a protocol proposed earlier in the literature, in [4]. Our analysis is for uplink traffic, but can be easily adapted to downlink scenarios when a similar protocol is used. In this paper, we concentrate on point-to-multipoint (PMP, or access point based) configuration of clients around a base station. The remainder of the paper is structured as follows. In Section 3 we describe the multi-channel MAC protocol used. In Section 4, we use a slotted-time model to derive expressions for the service time distribution for the protocol described in Section 3. In Section 4 we derive expressions for the delay seen by Poisson traffic arriving at the queue of a node. In Section 4, expressions are derived for delay when binary exponential backoff is used by the clients. Section 5 presents simulation results based on the multichannel MAC protocol implemented in NS-2. The simulation results are compared the analytical results while varying various system parameters. Finally, we present the conclusions drawn and avenues for further exploration that result from this work.

2 Related Work

The use of multiple channels in wireless networks leads to increased throughput and reduced delay. A number of protocols/schemes have been proposed to exploit this feature. One class of such protocols divide the available channels in two classes - control and data channels. Control channels are used to exchange network control information while data channels are used for data transfer [5], [6]. In [5], the authors present an analysis of their protocol. However, it was not pursued in detail while [6] presented only experimental results.

A large body of work has also concentrated on the use of specialized trans-receivers and/or use of multiple trans-receivers that allow a node to either scan all the available channels concurrently, for example in [8], [6]. The use of multiple trans-receivers solves many complex problems with a simple solution. However, such an approach is not compatible with current wireless technology that operates with nodes having a single trans-receiver with half-duplex capability. As a result such an approach, though simple, is not cost effective. In [6], [8], the authors present comparisons of their scheme with known protocols based on simulations and experiments. No comprehensive analysis was performed.

Recent publications have focused on using multiple channels in networks where nodes have a single trans-receiver with half-duplex capability. Due to this limitation, *hidden terminal* and *exposed terminal* problems have to be addressed. The presence of multiple channels introduces a new problem called *multiple channel hidden terminal problem* [4]. A popular approach to counter these problems is to use CSMA protocols for transmission on a single channel and allowing a node to ”reserve” a channel for a specific duration in the future, for example the protocols described in [5], [4], [10] and [11] use the reservation approach. In the queuing analysis in our paper is for a variant of the protocol proposed in [4]. The reasons for choosing the protocol in [4] over those in [10] and [11] are as follows. The RICH-DP (Receiver-Initiated Channel Hopping-Dual Polling) protocol proposed in [10] is receiver-initiated and makes stronger assumptions on the channel-hopping frequency. Moreover, a centralized (PMP) network employing RICH-DP, the central access point will need to poll each node in the network, thereby adding considerable overhead. The protocol described in [11] assume presence of a fixed control channel on which the nodes contend for the available channels. Moreover the underlying assumption in [11] is that the nodes can monitor all the available channels for any transmissions. This assumption is not generally true for all networks. In [5], busy nodes need to transmit a busy tone on a default/control channel while transmitting data. Again, this requirement requires the nodes in the network to be able to tranmsmit on two channels at the same time. Hence, the protocols in [5] and [11] cannot be applied to all networks in general.

On the other hand, the protocol in [4], MMAC (Multi-channel MAC) do not suffer from most of these problems, and can be easily adopted to IEEE 802.11 and IEEE 802.16 based networks.

We also note that the authors in [4] presented only experimental results. As a result, it would be interesting to investigate how different system parameters affect network performance. Our main focus here is to arrive at a simple expression that would allow easy prediction of delay as system parameters vary. This can assist system designers/operators to fine tune the performance of the network.

3 System Model

In this section, we present our proposed scheme. We make the following assumptions regarding the network:-

- There are M channels available. Each channel has equal Bandwidth.
- The access point can receive data on multiple channels simultaneously. This is a reasonable assumption since the access point can be a more specialized higher end device compared to the simpler clients that it serves.
- The channels are orthogonal. i.e. transmissions on a channel do not interfere transmissions on other channels. Here a channel may represent a code or a frequency band.
- Each network node, including the base station is equipped with a single radio trans-receiver capable of performing in a half duplex mode. Hence, each node can either transmit or receive a signal on a single channel at any point of time. The nodes can, however, switch to different channels dynamically.
- The network is synchronized. Again this assumption is not too restrictive to the networks as synchronization can be achieved through internal or external means. Some synchronization schemes have been discussed in [1], [2] and [3]

In the remainder of the paper we use the notations in Table 1 for the associated quantities:

Notation	Related Quantity
M	Number of Channels
N	Number of Nodes
T_F	Frame Length
S_A	Number of Contention Slots
S_D	Number of Data Slots
T_A	Contention Duration
T_D	Data Duration

Table 1. Notations Used in Analysis

We denote interval length as T_A and length of data interval as T_D. It is easy to observe that $T_F = T_A + T_D$. The data interval is divided into slots. A node can transmit data if it gains access to a particular time slot in the data interval on a particular channel. Each frame starts with the contention period. During the contention interval, all nodes and the base station switch to a default channel that is known beforehand. Each node waits for a random time uniformly distributed over $[0, CW_{min}]$ where CW_{min} is the size of the initial congestion window in slots. Note that the contention duration T_A is divided into a number S_A of contention slots, of equal length. As a node gains access to the channel, it sends a request to the base station for data transmission/reception. The base station and the node engage in channel negotiation as described later. At the end of channel negotiation, the node and the base station agree on the channel and a time slot in the data interval. Here we digress from the scheme in [4]. Our scheme allows the node and the base station to complete the channel negotiation before any other node attempts to transmit to the base station. Since all the non-transmitting nodes are listening on the channel, they can deduce when the negotiation is completed. This helps in reducing the number of collisions in the contention interval, hence reducing time wasted in retransmissions. This helps in allowing a larger number of nodes to negotiate for a channel and hence allowing the network to utilize available channels optimally. As a channel negotiation is completed, all the nodes update their *allocation table*. Each node has an *allocation table* to keep track of the data slot allocation on each channel. When the data interval begins, each node that has successfully engaged in channel negotiation switches to the negotiated channel and goes into a listen mode. Just as the negotiated time slot begins, the node listens on the negotiated channel for time interval DIFS to ensure that the channel is free before transmitting. Note that each node has a unique time slot and a channel in the data interval. Hence, no other node will be transmitting on the same channel and slot. A time interval of DIFS is allowed to compensate for small synchronization errors. Data transmission in each slot proceeds according to the existing MAC protocol (IEEE 802.11,802.16).

Synchronization in the network allows us to divide the time scale into slots. As each node has information regarding the slot it has to transmit/receive, wasteful re-transmissions to gain access to the channel are obviated. Since, the contention and data intervals are limited, the number of re-transmissions and size of backoff window would be limited by T_A and T_D. Hence, it is desirable to cut down the number of collisions (hence, re-transmissions) and keep the size of congestion window to a nominal level.

3.1 Channel negotiation

We present the channel negotiation protocol between the node and base station in this section. Due to the centralized nature of the network, we again digress from [4]. As all requests are directed to the base station, the base station has complete information regarding the channel and slot allocation in the network. We make use of this observation to come up with a simple negotiation procedure.

Consider a node wanting to engage in data exchange with the base station.Depending on the trans-receiver characteristics and allocation table, node generates a list of *preferred channels* that may be used for data transmission & reception. The node gains access to the default channel during the contention interval (by selecting a slot at random) and transmits a request that contains the preferred channels list. Note that other nodes would not transmit until the negotiation is complete, hence packet loss due to collision is reduced. The base station, when it receives the request, compares the list of the preferred channels with it's own allocation table. If a slot is available on one of the channels, the base station replies to the request with the designated slot and channel. If none of the channels is available, the base station can designate any of the available channels and slots and transmit to the node. The node, upon reception of the message from the base station, can either agree or decline to use the channel. The node then sends a message appropriately. Finally the base station sends an acknowledgment to the node and updates it's allocation table. The acknowledgment from the base station contains the information about the negotiated channel and slot (if a channel is agreed upon). This allows other nodes in the network to update their respective allocation tables.

Note that a base station can also reject a request from the node if no slots are available. In that case, or when node rejects the channel, the allocation table is not updated. Again allocation of a slot in addition to a channel reduces the channel access attempts during the data interval unlike [4] where nodes again contest for channel in the data interval.

4 Analysis

In this section, we present a worst case analysis of the delay experienced by a packet in the network when transmitted from a node to the base station. Since the ATIM window length is limited, it is reasonable to assume that the number of re-transmissions allowed for a packet, in event of a collision, is limited. Note that a node waits for a random time between 0 and CW_{min} before contending. In this section, we present analysis for the case when no re-transmissions are allowed (or $CW_{min} =$ ATIM window length). We follow this simple case with an extension to approximate for the case when k re-transmissions are allowed.

For the analysis, we assume that each node in the network has a single trans-receiver with half-duplex capability while the access point can transmit/receive on multiple channels simultaneously. The network nodes can, however, change their channel of communication dynamically. Figure 1 depicts the frame structure on a single channel to illustrate some parameters used in the analysis.

It can be observed that $T_r \geq 4T_p$ as four packets are required to complete a channel contention in the ATIM window. We divide the ATIM window and data window in slots. Each slot in the ATIM window is of length T_r. Hence, a node can complete the channel contention in exactly one slot. ATIM window has S_A

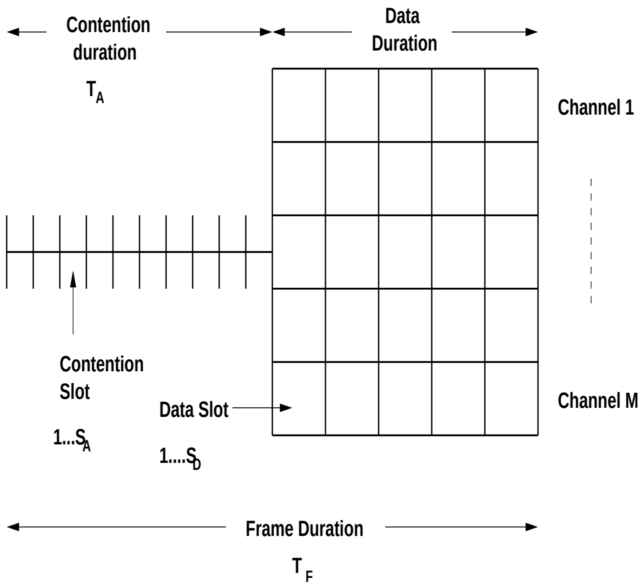

Figure 1. Slot structure

slots where $S_A = \frac{T_A}{T_r}$. The data window is divided into $S_D = \frac{T_D}{T_d}$ slots. Further we assume that channels are assigned randomly. Note that channels can also be assigned in a deterministic fashion by the access point in a deterministic fashion. Since there may be several criteria to assign channels, we assume a uniformly random model for simplicity and to ensure that all channels are equally occupied on an average.

We assume that each node will contend in a frame with a probability p. We denote p as the activity of a node. The activity of a node can be related to the average arrival rate under Poisson assumption as will be described later.Consider a node i that is active in a frame and attempts to contend in slot s. Since $CW_{min} = S_A$ in slots. Let $P($slot s is selected for contention $) = P(s)$. Then we have:-

$$P(s) = \frac{1}{S_A} \tag{1}$$

we denote *nc* as the event when only one node selects a particular slot. We denote i_A as the event when node i is active and Nc as the intersection of these two events.Using the ALOHA approximation, the probability that only node i selects slot s for contention of channel m when n nodes are active in a frame is given as:-

$$P(nc|i_A, s, n, m) = (1 - P(s))^{n-1} \tag{2}$$

Let $P($ channel m is contented $) = P(m)$. Since we assume a uniformly random assignment of channels, we have $P(m) = \frac{1}{M}$. It can be easily observed that $P(i_A) = p$. Let $P(n$ nodes are active in a frame $) = P(n)$. Assuming a binomial distribution, we have:-

$$P(n|i_A) = \binom{N-1}{n-1} p^{n-1} (1-p)^{N-n} \tag{3}$$

As a result, unconditioning (2) from n, s and m we get:-

$$\begin{aligned}
P(nc|i_A, s, m) &= \sum_{n=1}^{N} P(n|i_A) P(nc|i_A, s, n, m) \\
P(nc|i_A, m) &= \sum_{s=1}^{S_A} P(s) P(nc|i_A, s, m) \\
P(nc, i_A|m) &= P(i_A) P(nc|i_A, m) \\
P(Nc, m) &= \frac{p}{M} \left(1 - \frac{p}{S_A}\right)^{N-1} \qquad (4)
\end{aligned}$$

If we ignore the probability of failed contention due to channel/noise conditions, then $P(nc, m)$ denotes the probability of successful contention of channel m. Since only a limited number of data slots are available for the nodes, a contention can fail if all the S_D data slots in the data window on the channel m have been assigned before the slot node i has selected for contention, assuming node i is contending for channel m. As a result, if we denote probability of successful contention as P_R, the event that no node other than i selects slot s as $R_{i,s}$ for contention of channel m then:-

$$\begin{aligned}
&\text{denote } P(Nc, m) = P_1 \\
&P_{R|R_{i,s}} = \sum_{j=0}^{S_{min}} \binom{s-1}{j} (P_1)^j (1 - P_1)^{s-1-j} \qquad (5) \\
&S_{min} = \min\{S_D - 1, s - 1\}
\end{aligned}$$

It can be observed that equation (5) denotes the probability that at most $S_D - 1$ successful contentions take place before slot s. Then, probability of successful contention P_R can be derived as :-

$$\begin{aligned}
P(R_{i,s}|i_A, n, m) &= (1 - P(s))^{n-1} \\
P_2 &= P_{R|R_{i,s}} P(R_{i,s}|i_A, n, m) \\
P_R &= P(i_A) \sum_{n=1}^{N} P(n|i_A) \sum_{s=1}^{S_A} P(s) \sum_{m=1}^{M} P(m) P_2 \qquad (6)
\end{aligned}$$

Hence, now if Y = number of frames taken by a packet to transmit, then we have :-

$$P(Y=y) = P_R(1-P_R)^{y-1} \tag{7}$$

Equation (7) gives us an approximate characterization of the service time in terms of number of frames y. If we denote X as the average service time in number of frames, the expression for average service time can be given as:-

$$E(X) = \frac{1}{P_R} - 0.5 + \frac{T_A}{T_F} \tag{8}$$

In order to get the queuing delay, we need to express activity p as a function of packet arrival rate and use that in the equations. If we assume that a node is active only if the queue at the node is not empty (or node has data to transmit), then by simple M/G/1 approximation, we can approximate $p = \lambda E(X)$. Here λ is the average packet arrival rate per frame. Hence activity p is approximated as the probability that the queue at the node is not empty.

In order to calculate average delay, we condition on the arrival seeing either an empty or a full queue. Assuming that the arriving packet sees N_{av} packets in the queue, where N_{av} is the long term average of number of packets in the queue, we can write

$$E[D_{qe}] = E(X) \tag{9}$$

Where $E(D_{qe})$ is the expected value of delay seen by the arriving packet when queue is empty. In the case where the queue is occupied, delay seen by the arrival under consideration will consist of it's own service time, the service time of the packets in the queue currently not being serviced, and the residual service time for the packet at the Head of Queue, hence we have

$$E[D_{qf}] = E(X) + (N_{av}-1)E(X) + R_{av} \tag{10}$$

Here, R_{av} is the average residual service time for the packet at the head of the queue. From the standard M/G/1 queue analysis, $R_{av} = \frac{\lambda E(X^2)}{2}$. Here $E(D_{qf})$ is the average delay seen by the packet when queue is not empty. By Little's law $N_{av} = D_q \lambda$ where D_q is the average delay seen by the packet. Then we have:-

$$D_q = (1-\rho)E(D_{qe}) + \rho(E(D_{qf})) \tag{11}$$

Using (9) and (10) can express the average queuing delay D_q in number of frames as :-

$$D_q = E(X)/(1+\rho) + \left(\frac{\rho}{1-(\rho)^2}\lambda E(X^2)\right) \tag{12}$$

4.1 Extension for Backoff

Extending the above analysis for backoff is difficult as the analysis presented in the previous subsection would have to be repeated for each congestion window (number of congestion windows = number of retransmissions allowed + 1). Note that if a channel contends successfully, it will not contend later. This leads to a $P(nc|s)$ expression that depends on the number of successful contentions. The number of successful contentions is again a variable.

To illustrate the difficulty, let us take a look at a simplified backoff case where nodes that have been unsuccessful in contention, try to contend only when the current congestion window expires. This results in tractable expressions and is sufficient to demonstrate the difficulties associated. Let k_i denote the number of nodes that have successfully contented for a channel after i attempts ($k_0 = 0$). Thus, if a total of n nodes are contending in a frame, only $n_i = n - \sum_{j=1}^{i} k_j$ nodes will be contending during the i'th congestion window.

Let us consider a case when only one retransmission is allowed. Hence we have 2 congestion windows of lengths CW_1, CW_2 in ATIM slots. Due to exponential backoff principle, we have $CW_2 = 2CW_1$. It is easy to observe that $CW_1 = \frac{S_D}{3}$. Let P_{T_1} be the probability of successful contention by a node during the first congestion window. This can be derived by replacing S_A by CW_1 in (6). Now, we assess a node j that attempts to contend again during the second congestion window, we denote this event as A_j Hence, the probability density of number of nodes that have successfully engaged in channel contention k_1 given a node j has not suceeded is given by:-

$$P_{k_1}(k_1|A_j) = \binom{n-1}{k_1}(P_{T_1})^{k_1}(1-P_{T_1})^{n-1-k_1} \tag{13}$$

Let $k_{i,m}$ = number of nodes that have engaged successfully for contention of channel m after i attempts. It is simple to observe that $P(A_j) = 1 - P_{T_1}$. Now when node j attempts to contend for a channel m in a slot s after the start of second congestion window, it will be able to successfully contend if no other $n_1 - 1 - k_1$ nodes attempt to contend in slot s and there is at least 1 data slot available in channel m before slot s in second congestion window.These conditions are expressed by probabilities $P(nc_2)$ and P_{S_2} respectively. If we denote $k_{2,m}(s)$ as the number of users who have successfully contented for channel m before slot s in the second congestion window.Then we have :-

$$P(nc_2) = P(nc|k_1, k_{1,m}, A_j, s, m)$$
$$P_{S_2} = P(k_{1,m} + k_{2,m}(s) \leq S_D - 1|A_j, s, k_1, k_{1,m})$$

where

$$P(nc|k_1, k_{1,m}, A_j, s, m) = \left(1 - \frac{1}{CW_2}\right)^{n-k_1-1} \tag{14}$$

$$P_{S_2|k_{1,m}} = \sum_{l=0}^{S_D-1-k_{1,m}} P(k_{2,m}(s) = l) \tag{15}$$

Since channels are assigned randomly, we can write

$$P(k_{1,m} = l|k_1, m) = \binom{k_1}{l} \left(\frac{1}{M}\right)^l \left(1 - \frac{1}{M}\right)^{k_1-l} \tag{16}$$

We can also approximate $P(k_{2,m}(s))$ as

$$P(k_{2,m}(s) = w) = \binom{n-1-k_1}{w} \times \left(\frac{P(nc_2)}{M}\right)^w \left(1 - \frac{P(nc_2)}{M}\right)^{n-1-k_1-w} \tag{17}$$

Let T_2 denote the event of successful contention in second congestion window.Then we have:-

$$P(R_2|k_1, k_{1,m}, s, A_j) = P_{S_2} P(nc_2)$$
$$P(R_2|k_1, k_{1,m}, A_j) = \sum_{m=1}^{M} \frac{P_{S_2} P(nc_2)}{M}$$
$$P(R_2|k_1, A_j) = \sum_{l=0}^{S_{min}} P(k_{1,m} = l) P(R_2|k_1, k_{1,m}, A_j)$$
$$S_{min} = \min(S_D - 1, k_1)$$
$$P(R_2|A_j) = \sum_{k_1=0}^{C_{min}} P(k_1) P(R_2|k_1, A_j)$$
$$C_{min} = \min(n-1, CW_1)$$

Hence, we get the Probability of a successful contention as $P_{T_1} + (1 - P_{T_1})(P(R_2|A_j))$. As can be observed, the complexity of the derivation with the above approach grows exponentially with the number of congestion windows. As a result, we use the following approximation for the case when we have more than one congestion window.

P_R from (6) can be expressed as a function of S_A, S_D and $P(nc|i_A, s, n, m)$ from (2) as a function of n. Let

$$P(nc|i_A, s, n, m) = g(n)$$
$$P_R = f(S_A, S_D, g(n))$$

Let P_{T_i} denote the probability of successful contention in the i'th window. Hence if there are C congestion windows, the probability of successful contention $P_R(C)$ can be given as

$$P_R(C) = P_{T_1} + ... + P_{T_C} \prod_{i=1}^{C-1} (1 - P_{T_i}) \tag{18}$$

We approximate P_{T_i} as follows:-

$$\text{if } n - 1 \geq \sum_{j=0}^{i-1} E(k_i)$$
$$P_{T_i} = f(CW_i, S_D(i), g(n(i)) \tag{19}$$
$$\text{else}$$
$$P_{T_i} = 1$$
$$\text{where}$$
$$S_D(i) = S_D = \sum_{j=0}^{i-1} \frac{E(k_j)}{M} \tag{20}$$
$$n(i) = n - \sum_{j=0}^{i-1} E(k_j) \tag{21}$$
$$E(k_j) = CW_j P_{T_j} \tag{22}$$

We compare the numbers obtained from the above analysis with simulation values obtained and present them in the following section. Since we are assuming worst case scenario, it is expected that the simulation values will be less than the calculated numbers but should match the calculated values with appreciable accuracy.

Parameter	Value
Number of Channels (M)	3
Data Rate	10Mbps
Contention Slot Time (T_r)	0.2msec
Data Slot Time (T_d)	0.5msec

Table 2. Simulation parameters

5 Simulation Results

In this section we present simulation results to validate the analysis in the earlier sections. We implemented a multichannel PHY layer and a MAC layer simulating the protocol described in section 3. Each data point in each simulation is averaged over a sufficiently long period of time.

The simulation parameters are listed in Table 2

We note that analytical results match the simulation values. As a result, the performance of the network (delay) can be calculated accurately using the expressions obtained in the earlier sections.

Figure 2 compares simulation results and analysis for the average service time seen by a packet for T_F =5msec, λ =20 packets/sec. We observe that the simulation values closely match the analytical values.

Note that an increase in the number of contention slots (S_A) decreases the number of data slots available, for a fixed frame length. Increasing the number of contention slots available reduces the probability of collision in a given contention slot, but simultaneously reduces the number of data slots (S_D) available. Hence a larger number of successes in the contention period might not result in more packets being transmitted successfully due to too few data slots being available on the channels selected. Similarly, increasing the number of data slots reduces the number of contention slots. As a result even though there are enough data slots in the channel, there are too few successful contentions during the contention period. Both these cases result in higher delay as seen in Figure 2, at the two extreme points. Note that increasing the frame time does not alleviate this problem, since a larger frame time results in a significant amount of delay incurred by a packet arriving on average halfway through the current frame itself, in which case, it waits half the current frame, before it even starts the contention process.

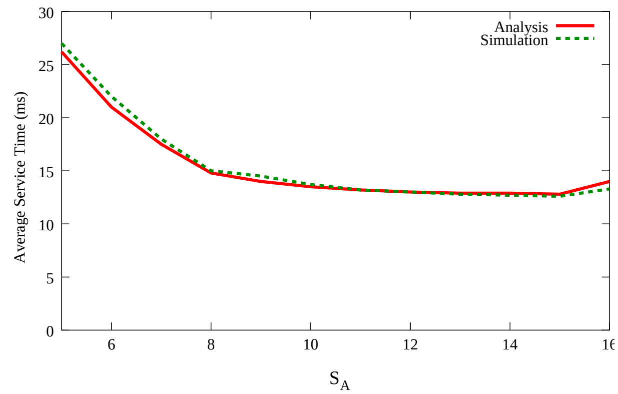

Figure 2. Average Service Time: Simulation v/s Analysis, T_F =5msec, λ =20 packets/sec

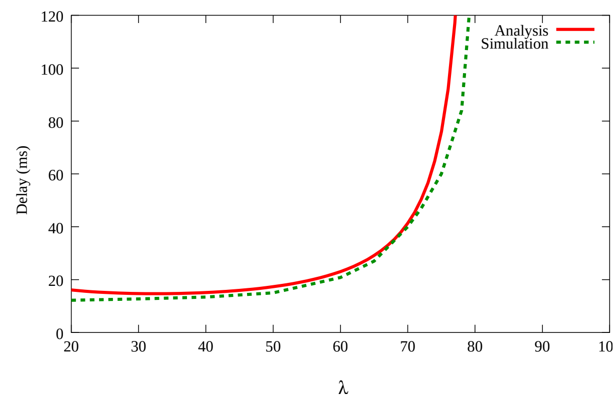

Figure 3. Queuing Delay v/s λ. T_F =5msec, S_A =10, n =10

Figures 3 and 4 compare the average queuing delay obtained from analysis and simulation for Poisson input traffic of varying intensity and for no backoff. The frame time is fixed at 5msec, and the number of contention slots are fixed at 10 in Figure 3, and 7 in Figure 4. As seen, the analysis follows the simulation results closely.

In Figures 5 and 6 we show the variation of queuing delay when the number of contention slots is varied from 5 to 15 for a 5msec frame, for 10 nodes. This is to illustrate the impact of the choice of number of contention slots on the delay seen. Note that in the case of queuing delay as considered in these graphs we expect to see effects similar to that seen in the Figure 2.

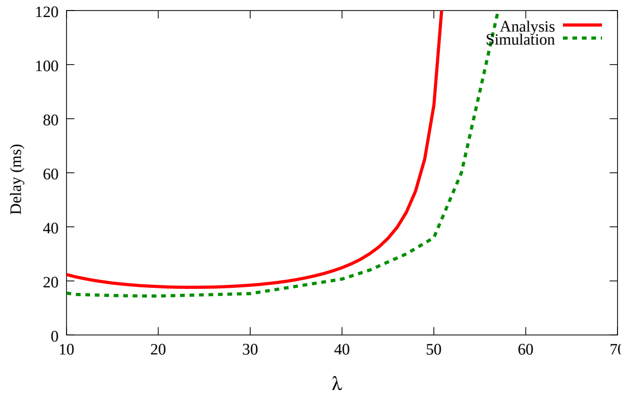

Figure 4. Queuing Delay v/s λ. T_F =5msec, S_A =7, n =10

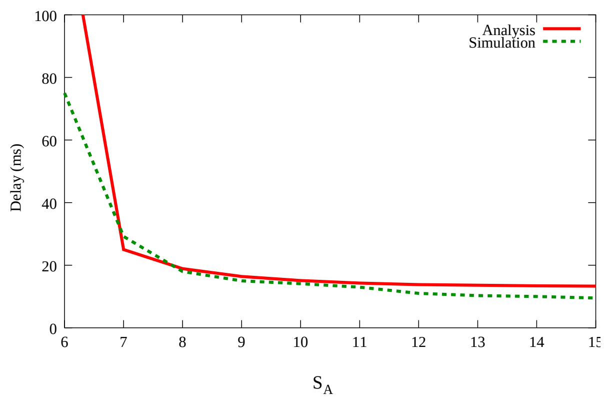

Figure 5. Queuing Delay v/s λ: Varying number of contention Slots, $\lambda = 40$, $n = 10$, T_F =5msec

Figures 7 and 8 show the variation of the average delay when backoff is used.

Figure 7 shows the variation of delay with λ for 20 nodes with a 10msec frame time. As is seen, the analysis and simulation match closely. Figure 8 shows simulation results for 30 nodes with backoff. Note that we observe much higher delays at lower values of λ, as compared to the values in Figure 3. This is because we have chosen larger frame times in the simulation which causes larger delay.

In Figure 9, we compare queuing delay as λ varies, for different values of S_A. The results in Figure 9 are obtained from analysis. We see that for lower values of S_A (4,5), smaller values of λ result in higher delays

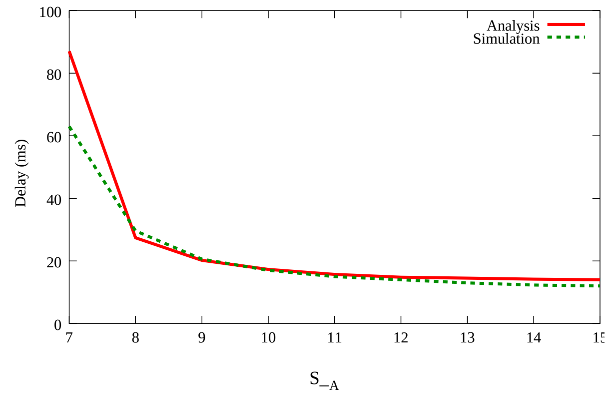

Figure 6. Comparison of Queuing Delay: Varying number of Slots, $\lambda = 50$, $n = 10$, T_F =5msec

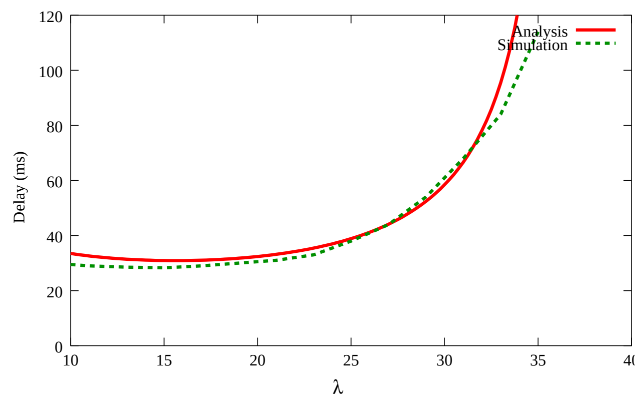

Figure 7. Queuing Delay v/s λ (with backoff): Varying number of Slots, $\lambda = 50$, $n = 20$, T_F =10msec

compared to larger values of S_A (11,12). This is due to stations colliding in the contention process. Note that as we increase S_A even further (18,20), performance degrades as delay increases for lower values of λ once again. In this case, successful contentions do not result in successful transmissions due to a smaller number of available data slots. Hence, there is an optimal setting of the S_A value, keeping other parameters constant.

In Figure 10, we compare queuing delay as λ varies, for different values of M, the number of channels. We note that as the number of channels increase, there is a marked improvement in the delay characteristics initially, but we experience diminishing returns on delay

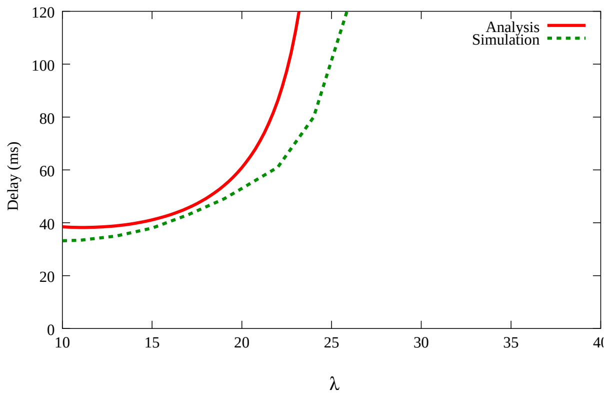

Figure 8. Queuing Delay v/s λ (with backoff): Varying number of slots, $\lambda = 50$, $n = 30$, T_F =20msec

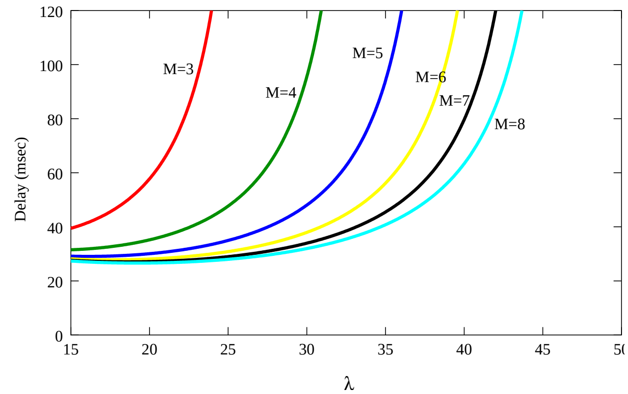

Figure 10. Queuing Delay v/s λ: Varying number of channels, M, T_F =10msec, T_A =8msec, N =20

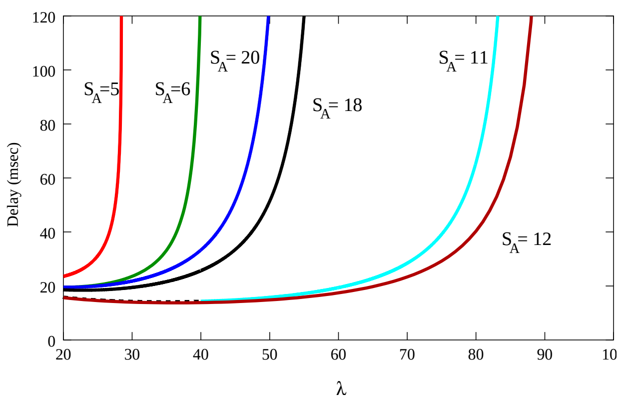

Figure 9. Queuing Delay v/s λ: Varying number of slots, $n = 10$, T_F =5msec

as more channels are added. Hence from a frequency provisioning perspective, there is a tradeoff between adding another channel and the obsserved improvement in delay.

6 Conclusion and Future Work

In this paper we present a detailed analysis of a distributed technique of channel access in multichannel PMP networks. We derive expressions for service time and average delay for packets in such networks. Extensive simulations are used to compare the analytical and simulation values, which match well. The extension of the analysis to the case where nodes contend with backoff also followed the simulation results closely. We can therefore predict the delay experienced by data in such a network fairly accurately using the derived expressions. We noted that the analysis does not accurately match the simulation values for very low values of λ. This behaviour is due to the overestimation of the variance of service time for low λ. This regime is however, not of significant interest, since it is intuitive that the delay seen by incoming packets will be very low, typically of the order of a frame time T_F.

We are currently investigating the above mentioned issue.

In general, we note that in some cases, for a particular choice of system parameters, backoff operation can show much better performance than having no backoff. However, the purpose of these simulations is to show the validity of the approximations made in the derivation of the backoff delay expressions, not to find the conditions under which the backoff operation exceeds the simple case in performance. This is an interesting avenue for future research. It would also be valuable to characterize the optimal operating point of the system as a function of the variables used in the analysis.

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