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Wireless LAN optimization for HQ audio conferencing with an Application layer Protocol (WHAP)

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Wireless LAN optimization for HQ audio conferencing with an Application layer Protocol (WHAP)

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The under-signed hereby certify that the thesis entitled "**Wireless LAN optimization for HQ audio conferencing with an Application layer Protocol (WHAP)**" by **Preethipradha Elavarasu** fulfills the requirements for the degree of **Master of Science**, and recommend to the Faculty of Electrical Engineering, Mathematics and Computer Science for the acceptance of the thesis.

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Abstract

Wireless Local Area Networks (WLAN) are being used extensively in today's world for all types of data transfer applications. The use of unlicensed ISM band for this purpose, makes the network prone to bandwidth deficiency as the frequency is being shared by users transmitting all types of traffic. This poses a great challenge in providing QoS (Quality of Service) for real-time high quality voice transmissions. In the presence of heavy interference, the channel access mechanism in the standard acts as an obstacle by adding additional contention delay to the time-sensitive traffic, degrading the QoS greatly. In order to address this issue, IEEE 802.11e amendment was released in the year 2005 which is still in use today, providing contention based Enhanced Distributed Channel Access(EDCA) and contention free HCF (Hybrid Coordinator Function) Controlled Channel Access (HCCA) for real-time traffic. However, HCCA mechanism which is responsible in keeping the contention delay low, loses its purpose when there is interference from another HCCA network. In addition, HCCA remains an optional feature till today and majority of the Wi-Fi chipsets do not have this feature implemented in them.

In order to provide a better QoS with the available channel access mechanism, my thesis contributes an application layer protocol called WHAP that schedules real-time high quality packet transmissions by the stations. In addition, an optimized WHAP network is derived from the stations working with optimized parameters (packet size, datarate) and coexist with an interfering network occupying up to 50% channel bandwidth. The QoS performance of a such a network is then analysed using a network simulation tool OPNET and compared with the QoS performance provided by the state-of-art Wi-Fi standard. The results confirm that, a much better QoS performance can be achieved with WHAP enabled systems.

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1 Introduction

Wireless LAN is widely being used for various applications at different places like airports, restaurants, offices etc. It provides similar functions as that of traditional wired LAN along with flexibility in terms of scalability, mobility, and ease in network setup. Since wireless medium is shared and demand for bandwidth is high, it is a challenging to provide expected performance especially in case of real-time traffic. In order to tackle this, the WMM (Wi-Fi Multimedia) feature was introduced where the real-time traffic get special attention time sensitivity. Current technologies provide data speed of up to 1Gbps [15] compromising the reliability when it comes to coexisting with interference from overlapping BSS [13]. This again poses a challenge when it comes to transmitting high bandwidth traffic like real time voice. Table 1.1 lists the standards that are in use along with their modulation, data rate and range.

Apart from the standards that are listed in the Table 1.1, there are several amendments that are included in each of the succeeding standards in order to provide their improved features. Two of such amendments are IEEE 802.11e [1] and IEEE 802.11s [12] which have been included in all the standards released after the year 2005 and 2011 respectively, providing the WMM feature. This means that IEEE 802.11n incorporates IEEE 802.11e amendment that provides mandatory Enhanced Distributed Channel Access (EDCA) and optional Hybrid coordination function Controlled Channel Access (HCCA) for QoS whereas IEEE 802.11ac [15]incorporates IEEE 802.11s amendment that provides mandatory EDCA and optional Mesh coordination function Controlled Channel Access (MCCA). Both the standards are currently being used with only the mandatory QoS feature EDCA being implemented in the Wi-Fi chip sets.

802.11 standard	Release Year	Frequency (GHz)	Band width (MHz)	Spatial streams	Max. Datarate (Mbps)	Range indoor/ outdoor (meters)
97(1G)	1997	2.4	20	1	2	20/100
b(2G)	1999	2.4	20	1	11	35/140
a(3G)	1999	5	20	1	54	35/120
g(3G)	2003	2.4	20	1	54	38/140
n(4G)	2009	2.4/5	20	1-4	72.2-288.9	70/250
			40		150-600	
ac(5G)	2012	5	20	1-8	87.6-700.8	90/N/a
			40		200-1600	
			80		433.3-3464	
			160		866.7-6930	

Table 1.1: Overview of Wi-Fi standards

1.1 Background

In general, a group of 802.11 compliant devices operating within a certain range form a wireless network and is called a Basic Service Set (BSS). This area can range from less than 100 to several hundred meters.

1.1.1 Types of Network

The basic building block of an 802.11 network is the BSS and can be of two types, the Independent Network (IBSS) and Infrastructure Network/ Extended Service Set (EBSS).

Infrastructure Network (EBSS)

All the communication between the nodes in an infrastructure network is done via the access point as shown in Fig. 1.1. If one mobile station in an infrastructure BSS needs to communicate with another mobile station, which is also a part of the same network, then the communication will take two hops i.e., first from the source station to the access point and then from the access point to the destination station. Thus, the AP (Access Point) receives, buffers and relays data between the stations in the WLAN and the wired network. This type of network takes advantage of the AP's high power to cover wide range and since the network topology does not vary as much as in ad-hoc network, the throughput and range is almost constant thus keeping the delay jitter very low [3]. This is an important feature in real-time data transmission.

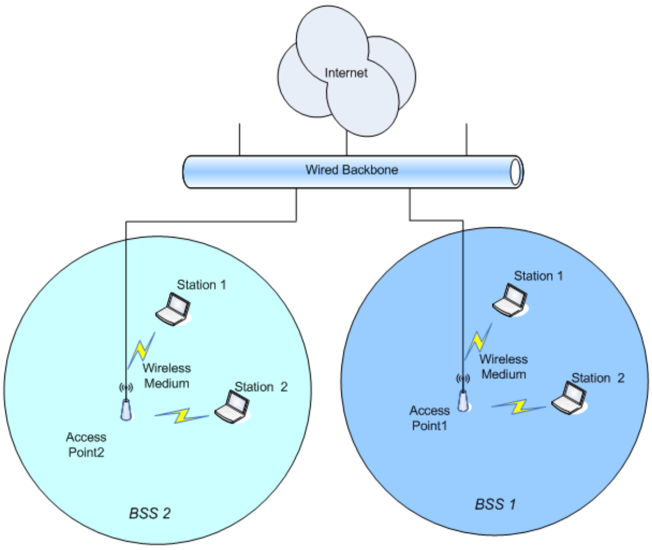


Figure 1.1: Infrastructure Network

Independent Network (IBSS)

In IBSS, otherwise called an adhoc network, stations communicate directly with each other (peer to peer). Since this type of network does not require a central control, it is easier to set up and is used especially in a small or temporary network [18]. A minimum of two stations in 802.11 network form the smallest IBSS. The network topology keeps changing and so does the range, throughput and delay. Fig. 1.2 shows an example IBSS network.

1.1.2 Network components

Any 802.11 network in general has four major physical components as in Fig. 1.1 and are discussed below.

Wired Backbone: This component plays a major role in providing internet access to all connecting access points and thus forms a large network. Routing of frames from source access point (sender) from BSS1 to destination access point (receiver) in BSS2 happens via a wired backbone.

Access Point: An AP is a device that connects wireless communication devices together to form a wireless network. The AP usually connects to a wired network, and can relay data between wireless devices and wired devices.

Wireless Medium: To transmit frames from one station to another station, the standard acts as an interface between the wireless medium and the communicating stations as there is a need for a common protocol to exist between these stations for communication. For this purpose, IEEE defined several physical layers and an architecture that allows these physical layers

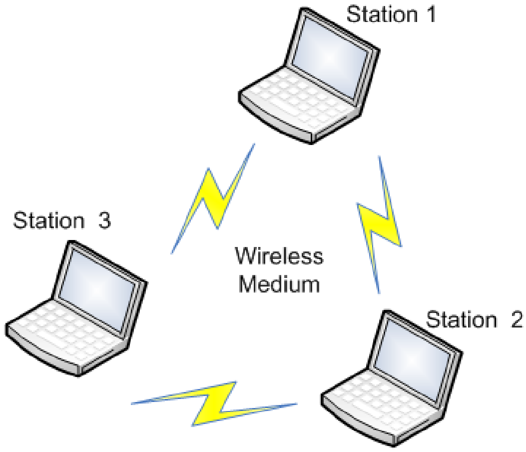


Figure 1.2: Ad-hoc based network

to communicate to the MAC layer i.e., the 802.11 standard. The details of various architectures can be found in [14].

Stations: Networks are built to transfer data between stations. Stations are computing devices with wireless network interface. Depending on the 802.11 standard as listed the Table 1.1 in use and the technique of how the data is transmitted into the wireless network plays a very important role in the performance of the stations. This is the main theme of this thesis.

1.2 Motivation and Challenges

QoS in general refers to the capability of a network to provide prioritized service to selected network traffic. This traffic includes time sensitive voice or multimedia (Wi-Fi multimedia WMM) and is addressed by IEEE 802.11e/s amendments as mentioned in Section 1.1. IEEE 802.11e provides the QoS by implementing two enhanced medium access mechanisms using Hybrid Coordination function (HCF) i.e. contention based EDCA and controlled access HCCA and this is included in 4G standard whereas, IEEE 802.11s provides QoS by implementing Mesh Coordination Function (MCF) i.e. contention based EDCA and controlled access MCCA and this is included in 5G standard. The current available Wi-Fi chipsets using 802.11n/ac however have implemented only contention based EDCA to provide the prioritized QoS data transmission leaving out controlled access HCCA/MCCA that provides parameterized/bounded value QoS. This reduces the performance of the real-time traffic when compared to what was promised in the standard.

This leaves us with the only possibility of altering the packet transmission technique that originates from the higher layer in order to improve the overall performance of real-time traffic transmission over WLAN.

1.3 Thesis Outline and Contribution

The goal of this research is to recommend optimum values for the high quality audio conferencing system parameters that communicates via a standard Wi-Fi network. Including the transmission delay of 5ms, an overall end to end delay of 10 ms is to be achieved. The end to end delay here denotes the time for the packet to be sent from the application layer to the wireless medium and received successfully by the receiving station. In order to achieve this value, an application layer protocol called WHAP is designed which schedules the packets sent such that it uses the Wi-Fi medium efficiently to provide the best of quality achievable on a standard Wi-Fi platform.

This thesis is organized with Chapter 2 discussing the basic concepts and details of the 802.11e included 802.11n standard helping one to understand the best performance that the current standard can offer. Chapter 3 discusses different policies in designing the WHAP protocol and concludes with a version which in combination with the optimized system parameters provides an optimized performance close to target performance and Chapter 4 shows the implementation of this protocol using a network simulator OPNET in each of the client station forming the discussion system with comparison of theoretical and practical performance discussed. Chapter 5 concludes the system performance results along with recommendations for future research.

1.4 Related work

In [18] specific discussion on the performance of adhoc network over infrastructure based network is shown with results having a better performance in a small network similar to the system constraint discussed in this thesis. [4] discusses specifically the voice traffic over mobile adhoc network and analysis of results show the packet loss rate to be high compared to the infrastructure based network. An overall view on the infrastructure based and adhoc based network for various traffic types can be found in [3]. In [5], HCCA model has been implemented in OPNET version 11.5 for the purpose of achieving performance in large network where as this thesis work is based on HCCA model for latest OPNET version 16.1 with maximum of 4 stations transmitting audio packets simultaneously from the application layer. The results obtained from implementation of HCCA are interpreted for a scenario that is configured with a audio conferencing system working under different interference condition. This helps to understand the best performance provided by the state-of-art technique for the system under consideration. In [9], an analytical HCCA model is design and compared with the mandatory EDCA channel access scheme with results showing a high utilization efficiency of HCCA compared to EDCA especially under heavy load conditions. [16, 17] explain the details of HCCA scheduler for TxOP allocation to the stations whereas [2, 7] show in specific the QoS performance for multimedia traffic using HCCA. The above mentioned studies deal with HCCA mechanism to improve QoS for real-time traffic but since it is considered an optional feature, most of the Wi-Fi chipsets do not implement this feature. [11, 6] describes the mesh network QoS performance in interference scenario and its challenges however, this plays a role only in an adhoc based network whereas, this thesis works with audio traffic transmission on a infrastructure based BSS. [8] gives a clear idea on performance of QoS in 802.11e EDCA vs. DCF considering parameters that are not optimized whereas, in this thesis work, optimum values for system transmitting audio traffic is proposed for the efficient working of the system.

2 QoS performance in WLAN

In order to understand the performance provided by the WMM feature, it is important to discuss in detail the MAC enhancements provided by the 802.11e/s amendments and its role in providing QoS. For this purpose, this chapter includes the details of the basic medium access mechanisms and the enhancements that are built on top of it. In addition, it also includes implementing these enhanced techniques on to a audio conferencing system simulated using a network simulator OPNET that include the following requirements.

System requirements

- An infrastructure based wireless system (Fig. 1.1) with all the clients participating having their clock synchronized.
- The audio samples are not compressed and they are sampled such that they have 768000bits per second i.e., sampling frequency of 48KHz with each sample representing 16bits = 48000∗16=768kbits per second.
- The stations operate under optimum values of the parameters as recommended in this thesis. Optimized parameters include packet size, data speed and other protocol based parameters.
- 802.11n/ac standard compliant, as it is aimed to co-exist with any kind of device using Wi-Fi channel.
- A maximum of 4 stations are allowed to transmit a packet from the application layer simultaneously.
- Co-existing BSS is restricted to use not more than 50% of the channel bandwidth.

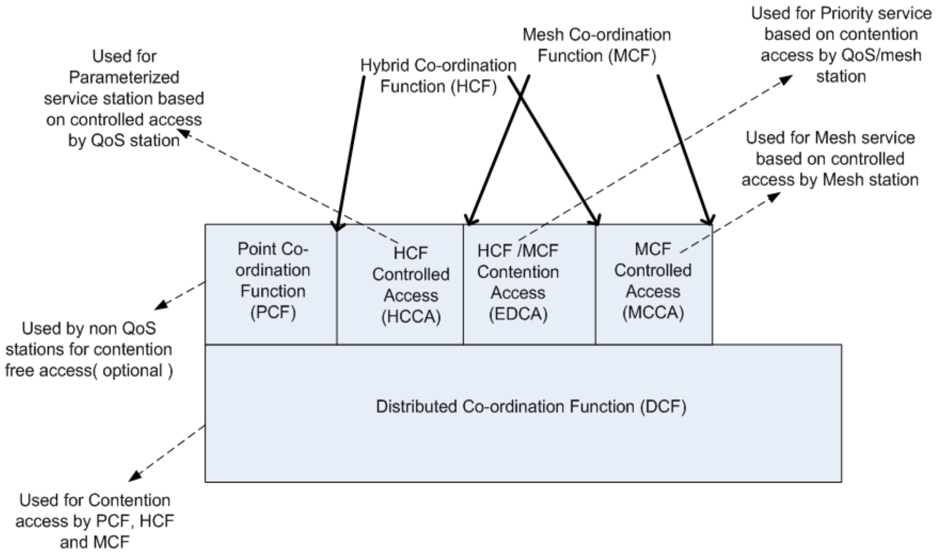


Figure 2.1: MAC Functions

The performance of such a system is analysed using network simulation tool (OPNET) and the results show the QoS performance provided by the existing standard functions.

2.1 MAC Sublayer Architecture

The 802.11 standard provides different access coordination functions as shown in the Fig. 2.2. Point Coordination Function (PCF), Hybrid Co-ordination Function (HCF) and the Mesh Coordination Function (MCF) services are included in this layer and are provided via Distributed Coordination Function services (DCF).

2.1.1 Distributed Coordination Function (DCF)

This is the basic access method of any IEEE 802.11 MAC and follows a technique known as carrier sense multiple accesses with collision avoidance (CSMA/CA). For a station (STA) to transmit, it first senses the medium to determine if another STA is transmitting. If the channel is sensed free, then the STA transmits. The CSMA/CA algorithm has a certain mandatory DCF Inter Frame Space (DIFS) between the contiguous frame sequences (Fig. 2.2). Once the channel is sensed busy for this inter frame space time, the STA defers by choosing a random backoff interval. This interval value starts decrementing when the channel is idle. When it becomes zero, the frame transmission resumes.

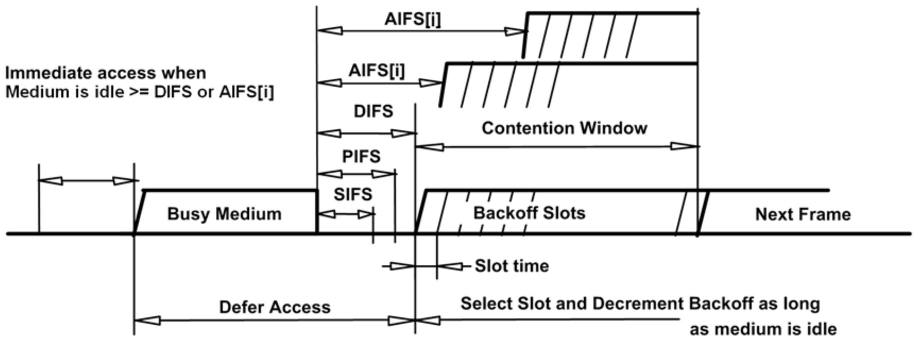


Figure 2.2: Channel access mechanism

The backoff value chosen depends on the number of attempts to send a frame i.e., when a frame is sent for a first time, an acknowledgement is expected. If no acknowledgement for the frame sent is received by the sender, the frame is retransmitted after waiting for time chosen randomly between [0,CW], where the CW value is chosen based in values shown in Figure 2.3. Basically DCF is meant for best effort traffic and there is no support for QoS in this coordination function.

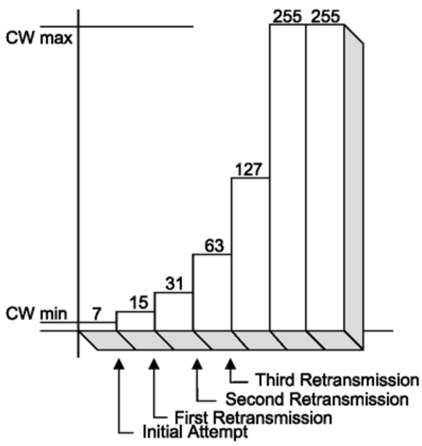


Figure 2.3: Exponential increase of contention window

2.1.2 Point Coordination Function(PCF)

This coordination function supports real-time traffic services and is built on top of DCF. It is supported only by infrastructure network. PCF uses poll based contention free mechanism where the AP acts as the point coordinator. When PCF is enabled in a system, channel is accessed every beacon interval with the interval having both contention free period and contention period (Figure 2.4). In the contention free period (CFP), the frames are transmitted after a PCF inter frame space time(PIFS) that is shorter than DIFS getting priority in their transmissions when compared to best effort traffic. This method of access also reduces the media access delay to minimum and constant value. However, it is not completely true when coexisting with an overlapping PCF enabled BSS as the two PCF enabled network contend for the channel. In addition, the probability of collision becomes high and in such a case there is retransmission with the backoff time chosen, losing its control over the medium to the coexisting BSS.

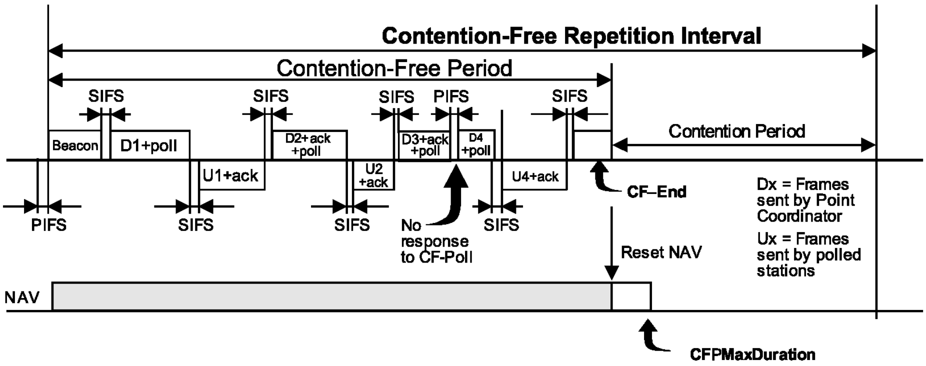


Figure 2.4: PCF Frame transfer mechanism

2.1.3 Hybrid coordination Function(HCF)

Among various 802.11 task groups, 802.11e task group has worked on QoS support in WLAN, for the real-time data transmission and released an IEEE 802.11e amendment in the year 2005. It enhances the DCF and PCF by introducing Hybrid Co-ordination Function (HCF) which comprises contention based Enhanced Distributed Channel Access (EDCA) providing prioritized QoS and HCF controlled Channel Access (HCCA) providing parameterized QoS (QoS with bounded values of parameters). These functions are built on DCF as shown in the Fig. 2.1. Only the QoS enabled stations i.e. the stations that use the 802.11e feature can use this these functions.
QoS enabled stations (QSTA) that obtain access to the medium are not allowed to use it longer than the specified limit known as TxOP (Transmission opportunity). Each TxOP is defined as the time duration in which the associated station delivers its frames. While operating in EDCA mode, the EDCA TxOP is used which prevents the low rate stations from using up all the channel time as in DCF wheras in HCCA mode the HCCA-TxOP is used which is otherwise called polled TxOP.

Priority	UP(user priority)	802.1D designation	AC	AIFSN	Designation	CW min	CW max
LOW	1	BK	AC_BK	7	Background	15	1023
	2	(spare)	AC_BK		Background		
	0	BE	AC_BE	3	Besteffort	15	1023
	3	EE	AC_BE		Besteffort		
	4	CL	AC_VI	2	Video	7	15
	5	VI	AC_VI		Video		
	6	VO	AC_VO	2	Audio	3	7
HIGH	7	NC	AC_VO		Audio		

Table 2.1: User Priority to Access Category mappings

HCF Contention Based Channel Access (EDCA)

EDCA provides a differentiated access to the channel with four different Access Categories (AC). Before a packet is forwarded to MAC layer, these packets are mapped into their corresponding ACs based on the User Priority (UP). Based on the mapped value, the inter frame space time known as Arbitration IFS (AIFS) is set along with respective backoff contention window for each traffic class (TC) to provide QoS. Above details are understood from the Table 2.1. Figure 2.5 shows the mapping procedure in AC's.

Note that here the backoff countdown starts only after waiting AIFS[AC] interval which is derived from AIFSN[AC] (AIFS[AC] = AIFSN[AC]$*$ Slot Time + SIFS Time) whose values are shown in the Table 2.1. All the mentioned IFS times are from the standard [15]. In an infrastructure BSS these values are advertised by the AP in their Beacon or Probe response frames. The main purpose of WMM feature is to protect the higher priority traffic from the lower priority ones. But there may also be cases where a traffic in certain class must be protected from other traffic belonging to the same class. In order to provide this kind of service, admission control feature is made available but as an optional feature.

Admission Control EDCA

AP advertises the admission control details to the associated clients in the beacon or probe response frame. For the stations that do not support the admission control, transmit frames by setting the traffic class to low priority AC for which there is no admission control needed. For the stations supporting admission control, prior to transmission of actual traffic, an ADDTS (ADD Traffic Stream) Request is sent from the station (Fig. 2.6) that has the TSPEC (Traffic Specification) parameter for that particular application that needs to be transmitted. TSPEC includes mean and peak data

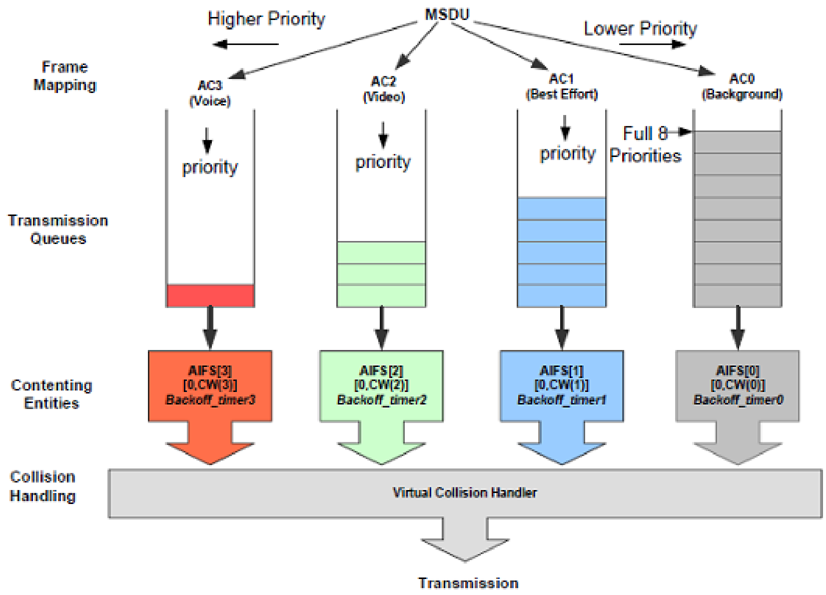


Figure 2.5: Access category mapping in EDCA

rate, mean and maximum frame size (Fig. 2.7). AP always responds with an ADDTS Response informing the acceptance or denial of the request. If accepted, the AP calculates the 'medium time' duration value as mentioned in [15] and sends this value along with the ADDTS response frame. From this the QSTA updates the *admitted_time* and *used_time* to keep track of the time it is using the medium. Thus if the *used_time* is greater than the *admitted_time* then the AC cannot transmit anymore and a new request is sent [15].

HCF Controlled Channel Access (HCCA)

As a part of 802.11e, HCCA provides controlled medium access which does the contention free transmission as an optional feature. Each superframe (beacon interval) is divided into contention period and contention free period similar to PCF except that the Contention Free Period (CFP) can be initiated anywhere within the contention period (CP) (Fig. 2.8). The controlled access period (CAP) provided by Hybrid Coordinator (HC) similar to Point Coordinator (PC) in PCF and resides in the 802.11e AP. The CAP during the contention period is allowed for the purpose of real-time data transfer and is initiated after channel is sensed free for PIFS time. There is limit in the duration of CAP within a superframe so as to accommodate contention based EDCA transmissions. Since PIFS is shorter than DIFS and

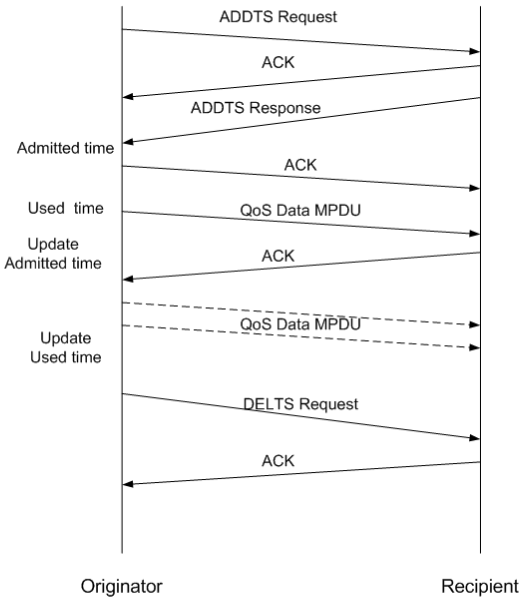


Figure 2.6: Admission control setup sequence

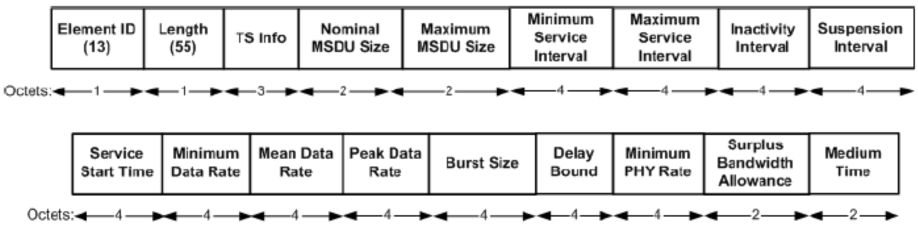


Figure 2.7: TSPEC fields

AIFS, HCCA frames are always given priority over EDCA frames (Fig. 2.8).

Admission Control for HCCA

In case of HCCA, initiation of admission control functions are similar to that of EDCA (Figure. 2.6). TSPEC contains all the information about the TS as in Fig. 2.7. Unlike in EDCA, the scheduler residing at AP calculates the MSI (Maximum Service Interval) that determines the duration between start of successive TxOP's for the stations transmitting QoS traffic. This TxOP value is calculated by the HC scheduler using the Eqn. 2.1 and Eqn. 2.2.

- First the scheduler calculates minimum of the MSI's values of all admitted TS. Then a number is chosen equal to or less than the minimum MSI such that it is a sub multiple of the beacon interval. This number is the scheduled Service Interval (SI) for all the admitted streams.

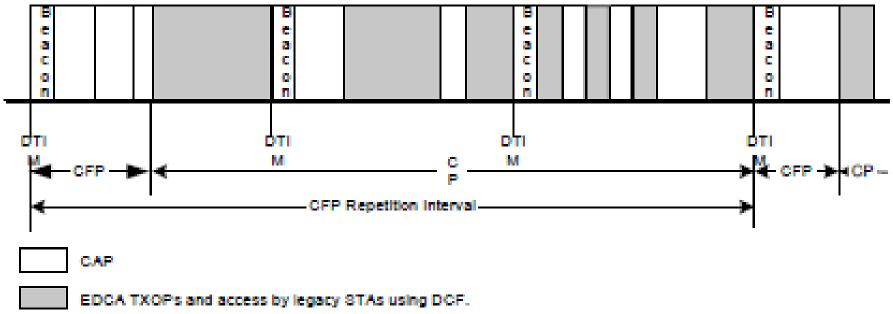


Figure 2.8: CAP/CP/CFP periods working

- Secondly the TxOP duration for the chosen SI is calculated as follows. Let N be the number of arriving frames in one SI for every stream i of a QSTA.

$$N_i = \lceil \frac{(SI * p_i)}{L_i} \rceil \quad i = 1, \ldots, n \tag{2.1}$$

where, SI is the scheduled service interval that is chosen as the minimum submultiple of beacon interval that is lower than the MSI requested by the station. p_i is the mean data rate, L_i is the payload size. Then using the N_i value obtained from Eqn. 2.1 the scheduler calculates the TxOP duration as the maximum value got by Eqn. 2.2

$$TxOP_j = \sum_{i=1}^{n} max[(\frac{(N_i * L_i)}{Ri}) + O; (\frac{M_i}{R_i}) + O] \tag{2.2}$$

where R_i is the PHY transmission rate, O is the overhead (MAC header + PHY header) and M_i is the maximum allowed MAC frame size (2304 bytes). In the upcoming subsections, the scheduler mechanism discussed here is implemented in the stations forming a real-time wireless transmission system in OPNET and the performance obtained is analysed.

2.1.4 Mesh coordination Function(MCF)

IEEE 802.11s task group introduced the QoS feature in mesh networking by including Mesh Coordination Function (MCF) [12] that is used only in an adhoc based network supported by 5G Wi-Fi devices. MCF includes contention based (EDCA) similar to HCF and contention free channel access (MCCA) as an optional feature. The main purpose of this function is to provide QoS for a network with multi hop. The details of EDCA are already discussed in the Section 2.1.1. The MCCA optional feature is not discussed in detail in this thesis as our focus is to analyse performance in a single hop network as it is similar to the real-time audio conference system under discussion. However, it is important to note that since MCCA is an optional feature, none of the state of art Wi-Fi chips implement this feature, thus again bringing our focus onto the available mandatory feature EDCA.

2.1.5 Additional QoS features

The frame format given in the Figure 2.9 shows fields that are responsible for providing QoS. These frames are supported only by QoS enabled stations.

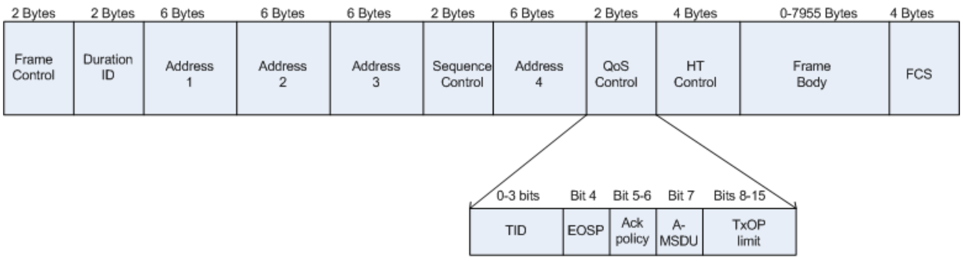


Figure 2.9: 802.11n MAC Frameformat

QoS Control: This field of the frame is of interest as we need to know what it can offer with delivering real-time audio transport. As shown in the figure, this field comprises of five subfields that are used based on the sender i.e. from Hybrid Coordinator (HC) as in the figure or by non-AP station.

TID subfield: It helps in identifying the Traffic category to which the particular MSDU belongs and the value can vary from 0-7 that determines the user priority (UP).

EOSP subfield: This field is of 1 bit size and is used only by the HC to inform the end of the service period (SP). The detail of service period calculation is discussed in the upcoming subsections.

Ack Policy subfield: This field is of 2 bits size and gives us an option to choose the type of acknowledgement expected from the station to which the particular MSDU is being delivered. The four types of Ack Policy offered are Normal Ack, No Ack, No explicit acknowledgement and Block Ack.

TXOP Limit subfield: This field as the name indicates, it specifies the EDCA TxOP time limit in the frames sent from QAP (QoS AP) in the QoS BSS.

2.2 Performance evaluation

A simulated audio conference system setup is configured with WLAN stations that have the above mentioned functions enabled in them. Using this, the QoS performance provided by the WMM feature can be evaluated.

2.2.1 Brief Introduction to OPNET

OPNET (Optimized Network Evaluation Tool) is a discrete event simulation tool which helps in modeling and evaluating the performance of various communication networks. It consists of variety of protocols along with different Wi-Fi standard models included in its library [10]. It works with a hierarchy of editors as discussed below.

Project Editor

The basic functions of this editor are to create the desired network model from the library which has custom node models. Choose the desired statistics that are necessary for performance evaluation and run the simulation. All the other editors are accessible from project editor itself. An optimized discussion system that is being discussed is designed and shown in Fig. 2.10.

Node Editor

It defines the entire communication flow behavior of the network objects that are used in the above Fig. 2.10. Example of the station involved in our discussion system is shown in Fig. 2.10.

Process Editor

It defines the function of the node model with Finite state machines which is nothing but combinations of various states and transitions that are executed with C++ code blocks. Example of the source type used in the nodes forming the discussion system is shown in Fig. 2.10.

2.2.2 Simulation setup

The project editor section of the Figure 2.10 shows an infrastructure based network forming a audio conferencing system where 4 stations are configured to transmit audio traffic simultaneously. In addition, these stations are simulated to work with both EDCA and HCCA. In order to evaluate the performance the following parameters are considered,

- **Interference utilization:** An interfering network is configured to coexist in the same channel as that of the audio conferencing network. This parameter denotes the percentage of the medium utilized by the interference in every packet interval of the audio transmitting stations.

- **Media access delay:** The time taken for the frames to be forwarded from MAC queue to its physical layer. This basically is the contention delay denoting the wait time of the frame when the channel is sensed busy.

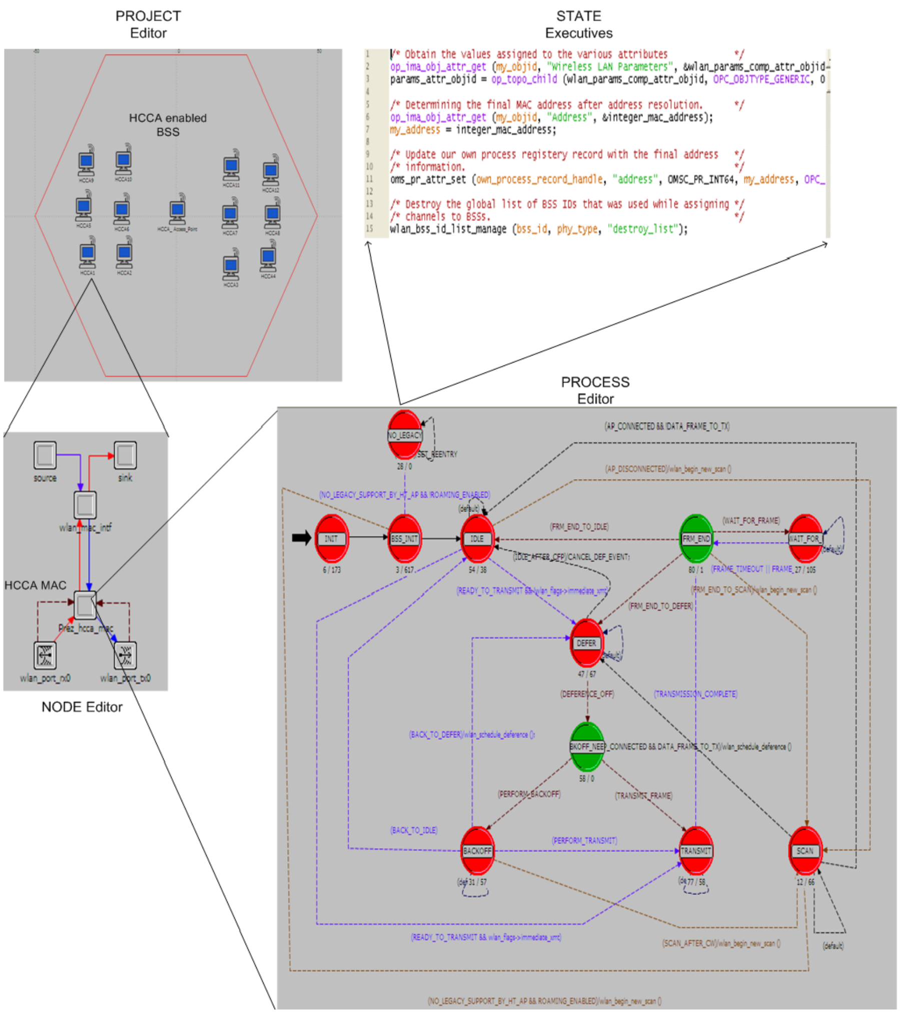


Figure 2.10: OPNET editors

- **Throughput:** The total number of bits that are received successfully across the stations and forwarded by the MAC to the higher layers for further process.

- **Number of retransmissions:** The number of times a frame is retransmitted by the source station due to lack of acknowledgement from the destination station.

Table 2.2 lists the values of parameters that are configured in each station involved in the simulated audio conference system. The reason behind choosing these values are already discussed in the Section 2.1.1.

PHY specifications	MAC specifications	APP specifications
802.11n HT 2.4GHz	HCCA manual implementation	Simple source
Guard Interval (GI) = 800ns	Max.SI = 10ms	p_i = 768000 bits
Spatial stream=1	AC = Voice	L_i = 15360 bits
R = 39 Mbps	Ack policy = Normal ACK	PI (Packet interval) = 20ms

Table 2.2: Parameters set at each station

EDCA result analysis:

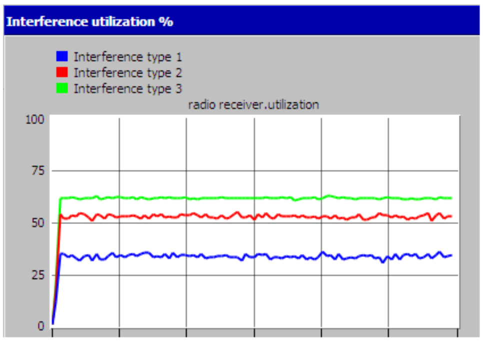
In OPNET, EDCA function was already implemented in WLAN station nodes and the results shown in the Figure 2.11 reflect the QoS performance provided by stations enabled with EDCA only.

Fig. 2.11(a) shows the different interference levels that were configured to coexist with the audio conference system and Fig. 2.11(b) shows the media access delay for each frame with respective interference utilization percentage. The coexisting interference was configured to transmit packets at interval equal to the packet interval of the EDCA audio traffic transmitting stations. As expected, for increase in interference percentage, an increase in media access delay is reflected. Fig. 2.11(c) shows the number of retransmissions that are sent in order to achieve the throughput shown in the Fig. 2.11(d). From the above results, it can be concluded that the performance achieved with EDCA implemented stations forming a audio conference system is not close to the QoS performance required by the system. An overall view of the EDCA delay performance with varying data rate and interference can be seen in the Figure 2.13.

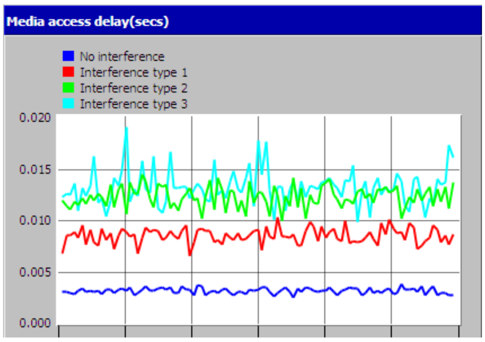
HCCA result analysis:

In OPNET, HCCA function is not provided by OPNET as it is an optional feature in the Wi-Fi standard. Hence, the HCCA function that was implemented manually in [5] is altered to support OPNET version 16.1. The results shown in the Fig. 2.12 show the QoS performance provided by stations enabled with HCCA only.

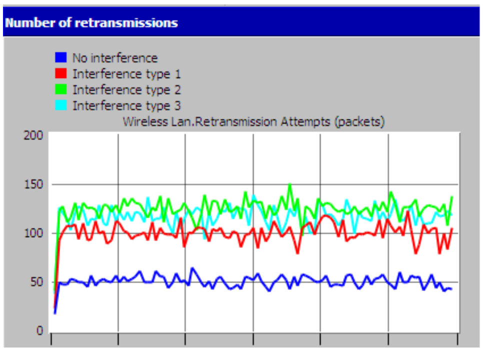
Fig. 2.12(a) shows the different interference utilization percentage that were configured to coexist with the audio conference system formed by HCCA stations. Note that here, the interference is configured to be from non-HCCA stations. Fig. 2.11(b) shows the media access delay for each frame with respective interference type. The main purpose of HCCA as discussed in above sections is to provide contention free access i.e., to have constant media access delay. This is reflected in the results. Since the AP



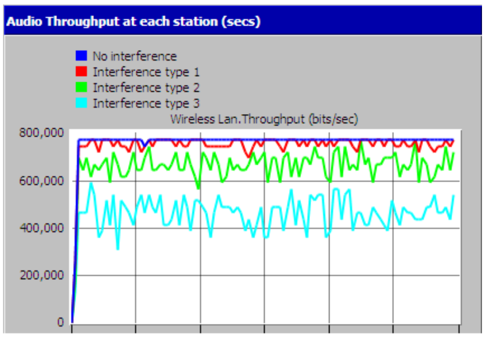
(a) Interference utilization %



(b) Media access delay

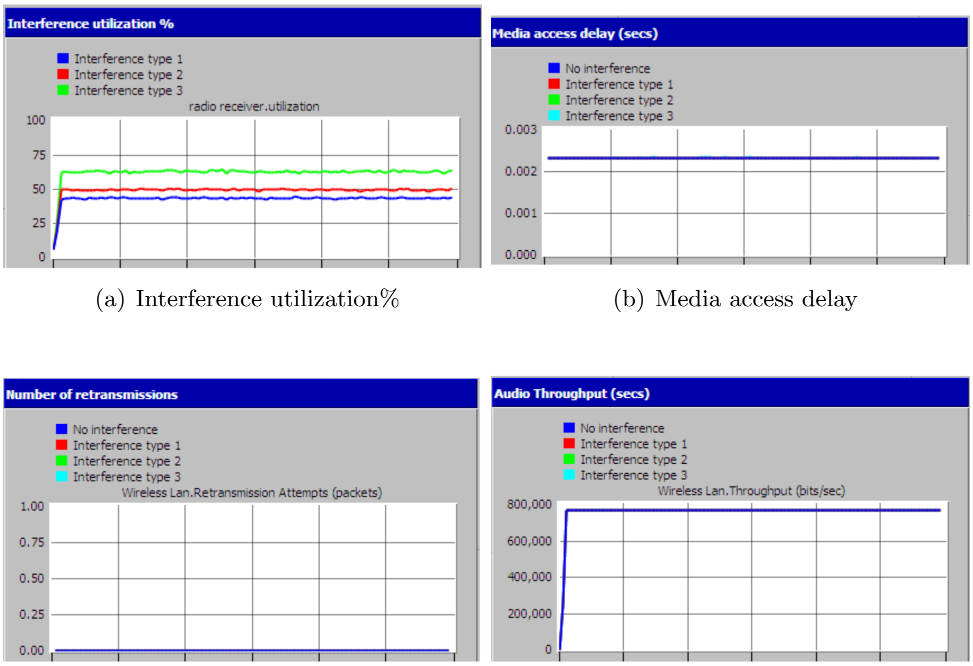


(c) Number of retransmissions



(d) Audio throughput

Figure 2.11: OPNET results showing the performance of EDCA



(a) Interference utilization%

(b) Media access delay

(c) Number of retransmissions

(d) Audio throughput

Figure 2.12: OPNET results showing the performance of HCCA

polls each of the station for audio packet transmissions, the retransmission of the packet is not necessary and the throughput of 100 percent has to be achieved. Fig. 2.11(c) shows the number of retransmissions that are sent in order to achieve the throughput shown in the Fig. 2.11(d) confining to theory. Thus, it can concluded that the performance achieved with HCCA implemented onto stations forming a audio conference system is very close to the required performance. But there are certain disadvantages with implementing HCCA onto stations.

- In case of interference from another HCCA enabled network, the whole of HCCA idea is not valid and there is contention for medium between the two HCCA networks.
- Another major factor acts as an obstacle for HCCA improvement is that it is an optional feature in the standard and is not implemented in most of the Wi-Fi chipsets.

2.3 Conclusion

Though EDCAs different AC promises priority for real-time traffic delivery, the media access delay is high in case of interference especially when all the stations are using the same AC. In case of the audio conference system that is simulated, all the stations use same audio AC and in case of various interference utilization, the overall delay performance seen in the Fig. 2.13 is greater than 5ms (target). Whereas, in HCCA, due to AP poll mechanism we are able to achieve a value close to the targeted delay of 5ms (Fig. 2.12(c)) but, due to the disadvantages mentioned above, this is not an option that can be used.

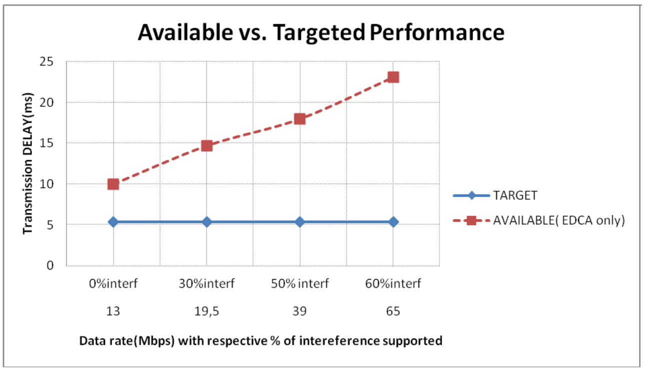


Figure 2.13: Comparison of delay performance with supported co-existing BSS interference

Hence,from all the above discussion and analysis, one understands that feasible option to improve the QoS performance in the presence of up to 50% interference is to make changes in the application layer so as to make the maximum use of the EDCA function available at the lower MAC layer.

3 WHAT design

From Chapter 2, it is understood that one is left with a system that works with an EDCA based MAC and with this method of channel access, it is impossible to provide the performance targeted with up to 50% interference (Fig. 2.13). Hence, an application-level protocol named WHAP is designed to work on top of the available EDCA MAC. The stations that implement WHAP are then optimized (optimum values of system parameters like packet size, data rate) with an interference tolerance of atleast about 50%. This chapter further discusses the design of WHAP keeping the performance requirement in mind.

3.1 Brief Description

One of the goals of the system considered in this thesis, is that it can allow up to 4 stations to transmit real-time audio data simultaneously from the application layer. From discussions in Chapter 2 one understands that when the stations start transmissions at the same time, they contend and wait till the medium is free (medium access delay). This not only leads to increase in the overall delay but also causes self interference due to delayed transmission. In order to keep the delay to a minimum, it is necessary that the stations do not start transmission at the same time, rather have a scheduled start time so that the medium access delay is kept zero. As shown in the Fig. 3.1, each packet interval (PI) is divided into certain number of slots that depends on the slot duration which in turn depends on the packet interval time. Each slot is allocated to a station for its transmission and no other stations start their transmission during that time.

In order to achieve this scheduled behavior, there are certain requirements expected from the system and are discussed below.

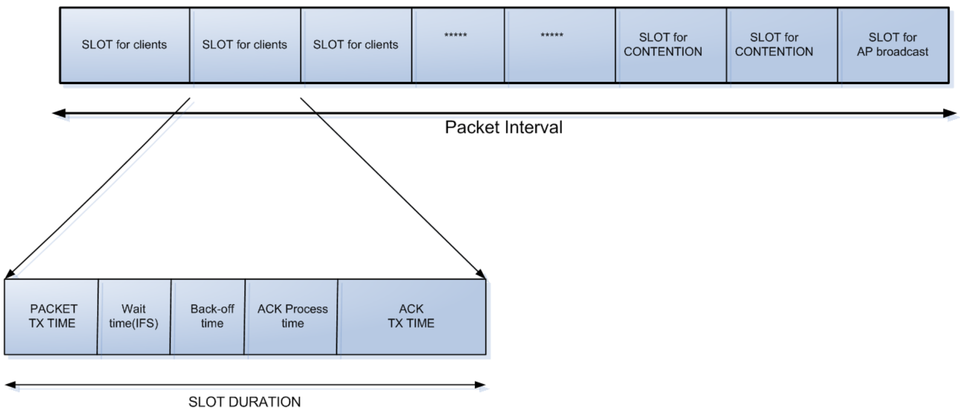


Figure 3.1: Packet schedule with slot duration with ACK

3.1.1 Design Requirements

There are certain requirements from the system that are expected to be accommodated when designing the protocol and are listed below.

- An access point with 'n' client devices associated forming a audio conferencing discussion system with 4 clients transmitting upstream audio packets and AP broadcasting an downstream audio packet simultaneously.
- We need to keep the self interference to minimum in order to avoid media access delay (similar to HCCA scheduling) and hence individual slots (start time) need to be known to the clients so that they do not start their transmission at the same time. This is to be done without involving poll messages from AP as in HCCA and by this way we not only reduce the medium access delay to zero but also avoid additional traffic transmission delay.
- Slot duration value should be chosen such that it can accommodate one packet transmission time + waiting time + backoff time + ACK time (if used).
- Within the certain packet interval time, 4 client packets upstream + 1 AP packet downstream each with 'x' slots for retransmission + control traffic from other associated clients with 'y' slots for this traffic are all to be accommodated.
- Further, number of slots allocated for contention should accommodate connection requests, keep alive messages from clients and beacon messages, connection response, synchronization messages from AP. Since these messages are non-realtime, they use low priority AC and contend for the channel access using EDCA procedure.

3.1.2 Assumptions

The following conditions are assumed to be working prior to the start of the packet transmission i.e., start of slot duration (Fig. 3.2).

- AP is ON and all the clients taking part in the discussion are all associated.
- Each client has a client ID (IP address) and the active clients receive respective slot ID's for their transmission via the connection response message from the AP. Thus, clients know their respective slot ID (S_i) that is addressed to them (using client ID).
- The clients also receive the 'slot duration value (S_d)' from the AP along with the slot ID information and calculate their start time (S_t) by themselves using the Eqn. 3.1.

$$S_t = ((S_d * S_i) - S_d) \tag{3.1}$$

Here, S_d is the slot duration and the calculation of this value is given by the Eqn. 3.3.

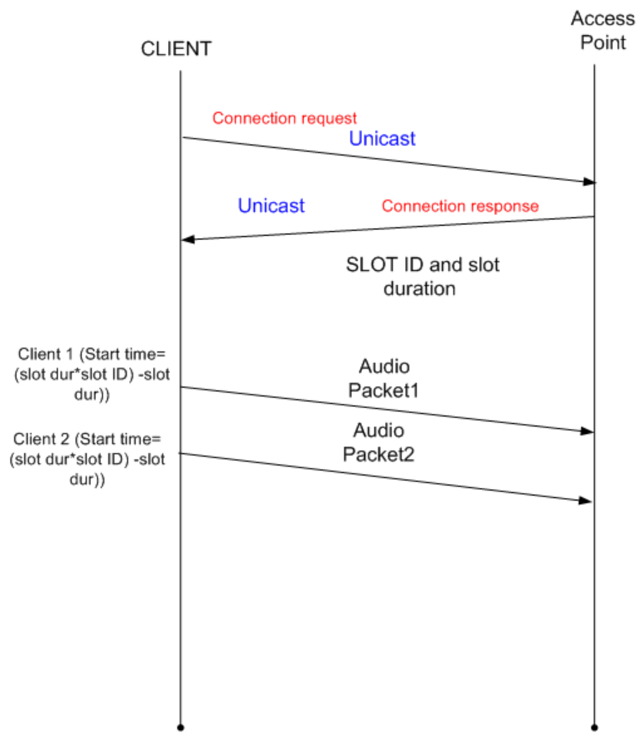


Figure 3.2: Sequence diagram of initial assumed connection setup

3.1.3 Design Constraints

There are some features that are implemented at the MAC that are responsible for reliable frame transmissions. These features play an important role in the entire performance of the system and are considered as a major factor in the design of the protocol. They are,

- **Acknowledgement policy:** As discussed in Chapter 2, an acknowledgement for each packet transmission promises a reliable communication. However, in case of time sensitive traffic transmission this degrades the QoS performance. For this purpose, MAC level ACK policy is set to 'No ACK'. However, there is a possibility of using different ACK policies from application layer and it is important to consider the delay performance of system when this is implemented.
- **Application level retransmissions:** Since, 'NO ACK' policy is chosen at the MAC level, there are no MAC level retransmissions. There can however be retransmissions sent from application level. The performance of the system under such a case is to be analysed in detail.
- **Freezing time:** This denotes the time that the backoff value stops decrementing due the channel being busy with interference. This time plays a very important role in the system performance and needs to be discussed in detail with different interference scenarios.
- **Network scalability:** The total number of stations that can take part in the audio conferencing discussion system needs to be considered. This mainly depends on the control traffic that keeps the station associated to the AP. A detailed analysis of the time occupied by this traffic and the number of stations that can be accommodated is to be done.

All the above mentioned constraints are discussed in detail in the upcoming sections and an optimized system is proposed with theoretical performance results.

3.2 Application level ACK policies

As discussed, it is important to consider different ACK policies that can be implemented on the system from the application layer. This is because, No ACK policy that is chosen at the MAC layer makes the communication less reliable. However, the time taken in processing the ACK and transmission of this message affects the system performance to a certain level. This section briefs the parameters of WHAP protocol that are involved in case of using each of the ACK policy.

3.2.1 Using Acknowledgements for every packet

Figure 3.3 shows how this ACK policy can be accommodated into the WHAP basic scheduling design discussed in the Section 3.1.

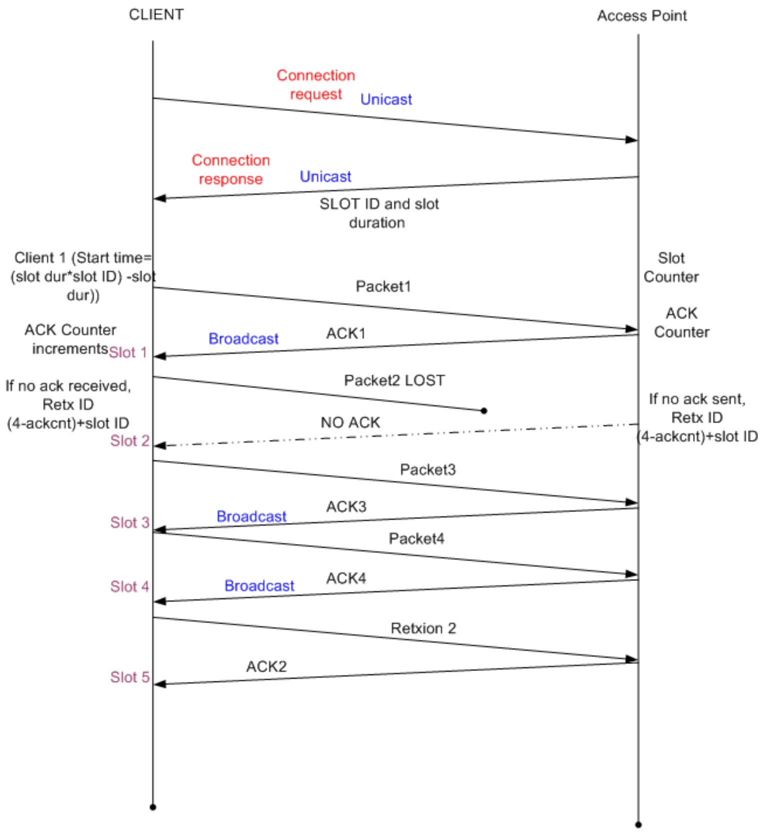


Figure 3.3: Sequence diagram of system working with ACK for every packet policy

For each packet sent from the client to AP, ACK counter (A_c) starts on all the clients and the AP. This counter is incremented by 1 for each successful ACK received and sent. This is important because, in case of packet or ACK loss, there needs to be certain number of retransmissions and start time of them also needs to be calculated by the client themselves. For this purpose, a retransmission slot ID (R_i) calculation (Eqn. 3.2) is carried out by the client that expects an ACK to allocate itself a retransmission packet slot and to calculate the start time as in Equation 3.1. In order for an ACK to be generated from the application layer, there is certain process time involved for the message to traverse through the OSI layers, which also needs to be included.

$$R_i = ((N_a - A_c) + CS_i) \tag{3.2}$$

Where, CS_i denotes the current slot ID that is being used and N_a denotes the number of active stations that are transmitting audio packets. Note that, the ACK counter at AP increments when an ACK is sent whereas, at the clients the increment happens when an ACK is received. Hence the ACK considered here is sent one way from AP to the client.

Fig. 3.1 shows the basic WHAP schedule mechanism along with details that are to be considered when calculating the duration of each slot (Eqn. 3.3).

3.2.2 Using Block ACK

In this policy, instead of AP sending ACK for every packet received, a separate slot can be allocated (by the AP itself) to transmit one ACK packet that acknowledges all the four packets received by the AP. In this case, the block ACK is sent after all expected packets are received within a certain time and the block ACK should also be received by all the clients within certain time. One can already predict that this provides improved performance over ACK for every packet, as the slot duration is kept shorter (Fig. 3.4. But an extra slot is used for sending this block ACK and in case of unpredictable interference, allocation of a specific slot for the purpose of block ACK might not be so meaningful as there may occur retransmissions due to block ACK timeout at the clients.

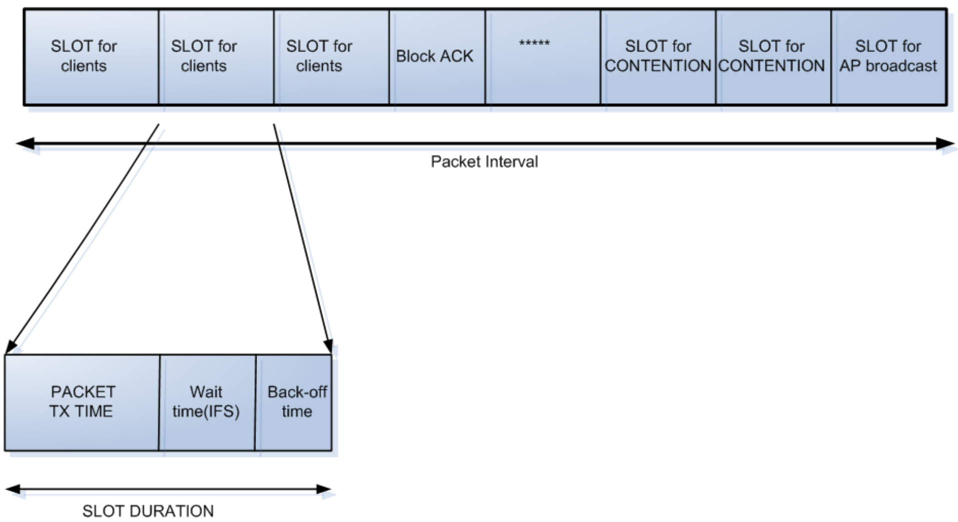


Figure 3.4: Packet schedule with slot duration based on block ACK policy

3.2.3 No ACK policy

As the name indicates there are no ACK packets are sent by the AP's application layer. This will obviously reduce the reliability of the network as there is no guarantee that the packets sent are received by the AP. However, there is a possibility to retransmit a packet from the application layer, and details on how this is done is discussed in detail in Chapter 4.

3.3 Retransmissions

In order to have a reliable communication, there are retransmissions that are sent from the application layer. This technique is incorporated into the stations using WHAP i.e., the retransmission decision is taken by the clients themselves based on the queue size of the MAC which holds the previously transmitted packet. In case of interference, there is a possibility that the 1st packet sent by the client is queued at MAC for certain time. When the MAC queue is empty, a trigger is sent to the application level and depending on the time difference between sending a packet and receiving the trigger, the application decides whether to send a retransmission or not (Fig. 4.1).

Before discussing the details of this policy, there is a need to understand if this policy can provide the performance that is expected. Before discussing the details of the decision making (Section 4.1) , it is first important to theoretically analyse the transmission time taken for this purpose and the number of retransmissions ('x') that can be accommodated. Eqn. 3.5 gives an idea on how the utilization of the stations can be calculated and using this, the performance of the system with various number of retransmissions can be calculated. Figure 3.5 shows the utilization factor for a case where ACK for every packet was chosen with a process time of 50 micro seconds. It is clear from the Fig. 3.5 that the utilization factor is $> 50\%$ for 1 retransmission sent every packet. Hence, we chose 1 retransmission for every packet and then work on bring the utilization $\leq 50\%$.

3.4 WHAP Parameters

All the ACK policies that are discussed in the previous sections need to analysed to see if they can provide performance that is close to the required performance as shown in the Fig. 2.13. For this purpose, each of the ACK policy's are considered to work along with the WHAP basic scheduler scheme and the parameters(S_d , U) involved are analysed. This not only provides the understanding of performance provided but also in choosing optimum values of the those parameters. The calculations discussed below give the details describing these parameters.

Slot Duration: The slot duration value is chosen such that it includes

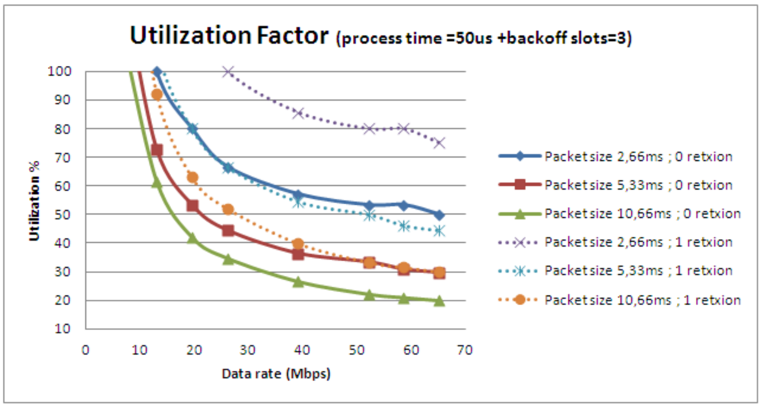


Figure 3.5: Utilization % for various number of retransmissions

total time given to each station for transmission of one successful packet (Fig. 3.1 and Fig. 3.4).

$$S_d = P_{tx} + W_t + b_i^j + (P_t + A_{tx}) \tag{3.3}$$

The values involved in the slot duration calculation is discussed below.

- *Transmission time for packet (P_{tx}):* For more accurate calculation of this transmission time, the calculation described in 802.11-2012 standard is used [15].
- *Waiting time (W_t):* DIFS + SIFS are included and these values equal to the standard values (Table 3.3).
- *ACK process time(P_t):* Since the ACK is sent from the application layer, there is certain amount of time taken for the trigger of ACK message to be processed once a packet is received successfully by the AP. This time varies from 10 microseconds to few hundred micro seconds. This process time needs to be included in the slot duration based on the type of ACK policy being used.
- *Transmission time for ACK (A_{tx}):* Similar to time calculation for the packet.
- *Backoff time (b_i^j):* The random backoff values are chosen from "CW_{min}" and they keep increasing exponentially for every retransmission attempt. Since we use NO ACK policy at MAC level, the retransmissions are NIL. This means that backoff is chosen as between $[0, CW_{min}]$. The default EDCA contention window values for the 802.11g/a/n PHY in a QoS BSS are defined as in Table 2.1. This means that maximum of (3*9us) = 27us needs to be added to the slot duration as audio transmission is considered in the system discussed in this thesis.

Utilization Factor:

This parameter gives the percentage of time within a packet interval that is being used for the packet transmission (real and non-real time traffic) of WHAP enabled BSS audio conferencing system under discussion. There are certain parameters used for this calculation.

- Number of active audio transmissions(N_a) that can occur simultaneously varies from 1 to 4. AP also transmits a packet downstream and thus maximum of 5 audio packet transmissions are allowed to occur simultaneously.
- Number of retransmissions(N_r) allowed for each packet (both client and AP) is 1 as discussed in Section 3.3.
- Time allocated for EDCA contention based access = 2 slots. This value decides on the scaling of the audio conference system.
- Total number of slots needed (S_n) in order to accommodate all the above discussed traffic is 12 slots.
- Total number of slots available (S_a) depends on the as in the Eqn. 3.4

$$S_a = PI/S_d \tag{3.4}$$

Using the above parameters, utilization factor is calculated as shown in the Eqn. 3.5.

$$U = (S_n/S_a) * 100 \quad [\%] \tag{3.5}$$

These parameters are influenced by the ACK policy chosen by the stations involved and the following sections discuss this issue in detail.

3.5 Theoretical Analysis

The concepts briefed in Section 3.2 are discussed in detail along with the analysis of their performance and optimum values for the best system performance are found.

3.5.1 Detailed analysis when using ACK for each packet

When a packet is received successfully by the AP, an application layer ACK is sent and when the packet is not received successfully, there is no ack sent.

For each successful ACK sent by the AP or received by the client stations, an ACK counter(A_c) keeps on incrementing at both AP and clients. When the ACK1 or packet1 information is sent by the AP or client1 respectively

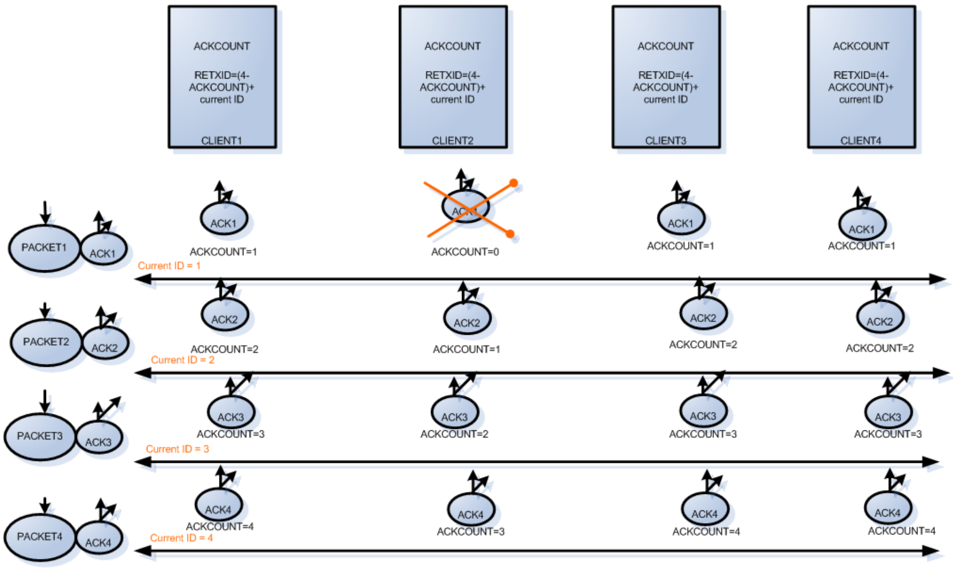


Figure 3.6: Detailed working of STA enabled with WHAP + ACK for every packet

are lost due to interference, client1 sends a retransmission and a slot gets allocated for this purpose (R_i) from Eqn. 3.2). In case of ACK broadcast message being lost, the clients update their A_c accordingly. This can cause the stations to choose the same slot for their retransmissions. A detailed behaviour of system under these conditions is discussed below.

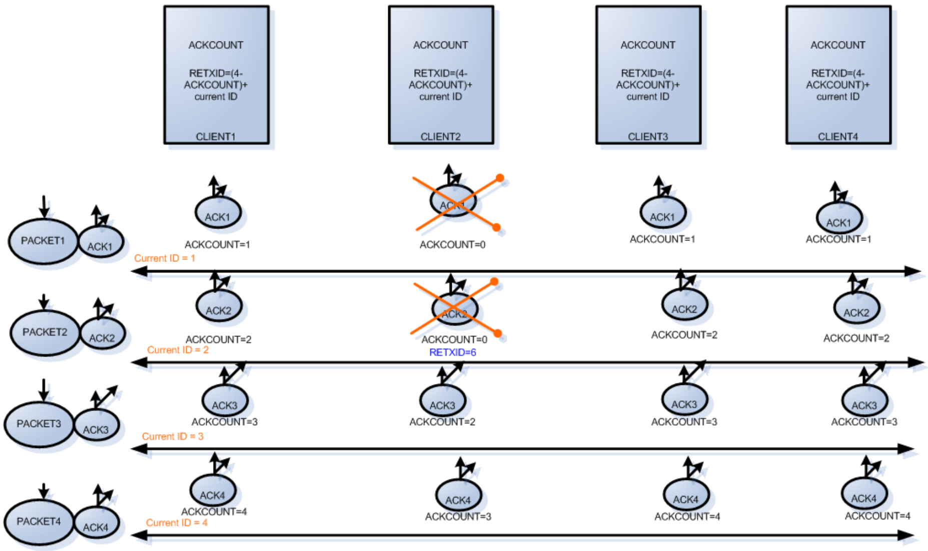
Type 1: Packet not acknowledged due to ACK LOSS

CASE 1: Figure 3.6 shows the details of the system behavior under this case. ACK1 supposed to be for client1 packet, is received by client1 but not by the client2. Hence, client2 ACK counter is not incremented. This does not affect the system performance much as far as client 2 or the other clients do not calculate R_i as far as ACK2 for packet2 sent by it is lost.

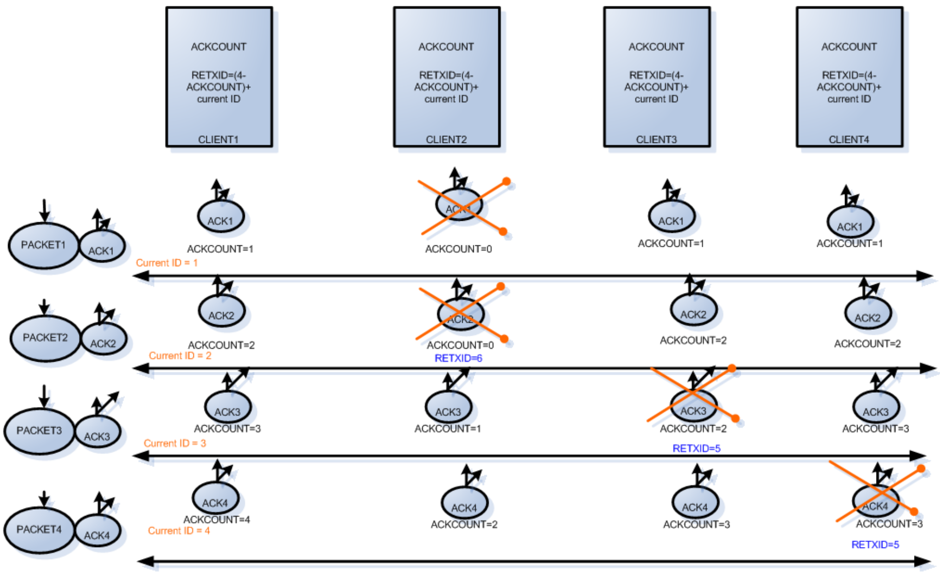
CASE 2a: Similar to case 1, if ACK2 that is supposed to acknowledge client2 packet not received by client2 but received by others. Then, the R_i of client2 = slot6 from Equation 3.2 shown in Figure 3.7.

CASE 2b: As the scenario of case 2a proceeds, there occurs a collision of retransmitted packets when the client 3's ACK3 is lost. This is because, client3 calculates its R_i = slot5 and when client 4's ACK 4 is lost, client 4 also calculates its R_i = slot5. Hence, there occurs a collision at slot5 as the start time for these clients are the same (Fig. 3.7). However, the AP has received all packets successfully and thus, in this case the only disadvantage is increase of utilization as the slot5 is used unnecessarily for retransmission.

Type 2: Packet not acknowledged due to PACKET LOSS

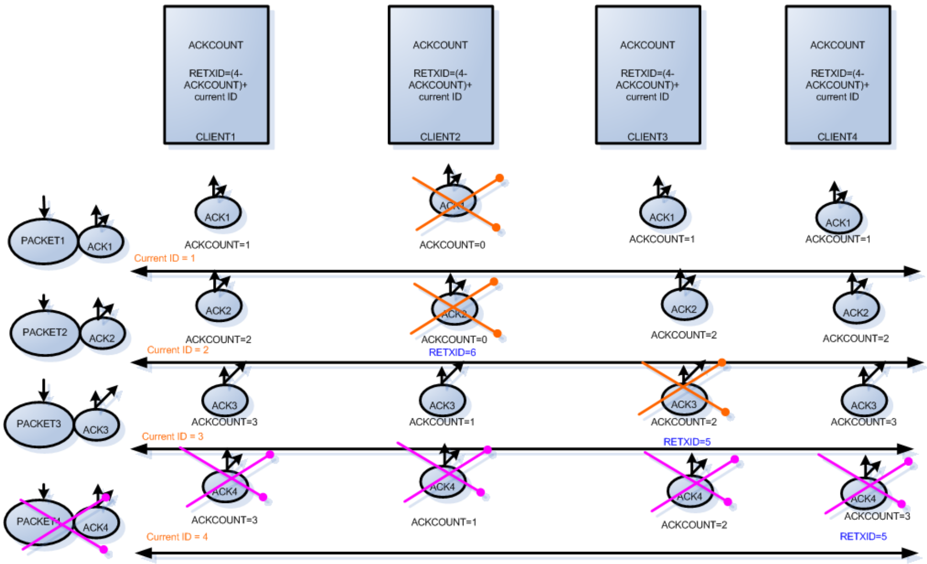


(a) ACK2 lost for client2

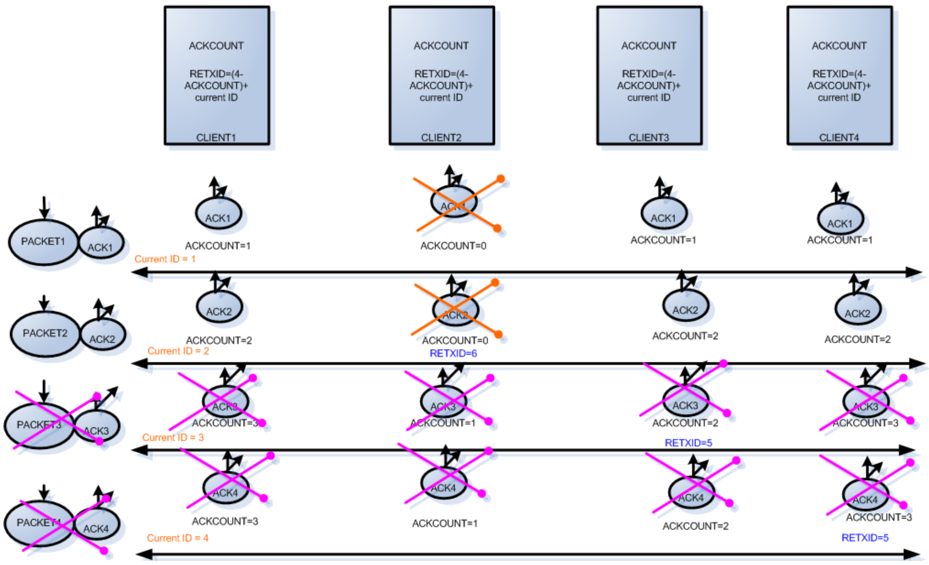


(b) ACK4 lost for client4

Figure 3.7: ACK2 lost for client2 and ACK4 lost for client4

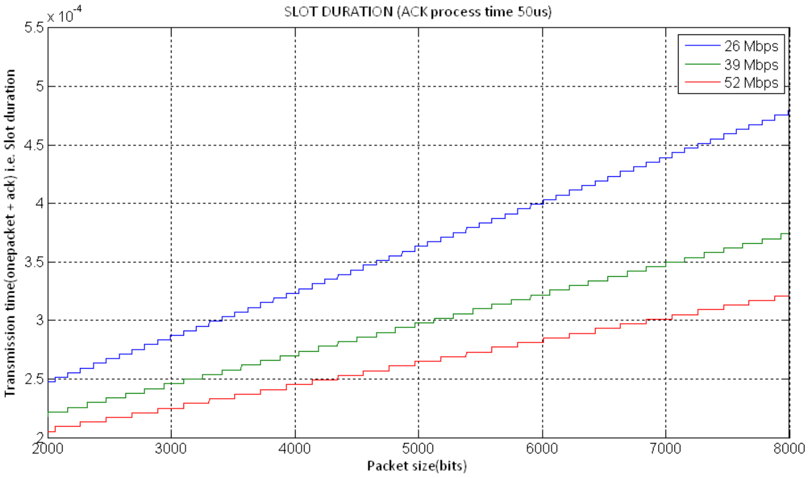


(a) PACKET4 lost



(b) PACKET3 and 4 lost

Figure 3.8: ACK2 lost for client2 and ACK4 lost for client4



(a) 50us

Figure 3.9: Slot duration for 50us process time

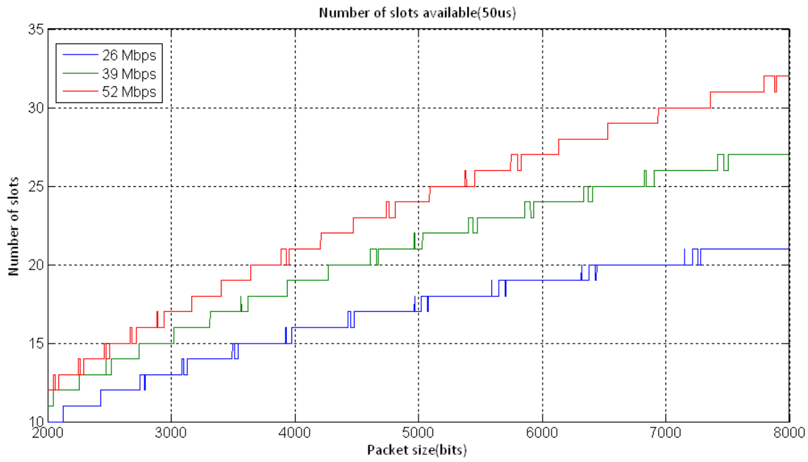
The system behaviour is similar to case 2 discussed above, except that the packets are lost due to interference and the retransmissions are not received successfully by the AP due to the fact that the R_i value calculated by the stations are the same. This decreases the performance to a greater extent than in type 1 as in type 1 only the ACK is lost unlike in type 2, the packets themselves are lost along with their retransmissions. Fig. 3.8 explains this case in detail.

Now that possible ACK loss cases and their effect on the WHAP system are discussed, it is necessary to analyze their performance by calculating the values of parameters (Section 3.3) involved.

Figure 3.9 shows the total slot duration values (Section 3.4) for various packet sizes that can be chosen. From the packet size (payload L_i), the transmission delay (PI) can be calculated as show in the Eqn. 3.6. Note that since there the ACK process time can vary depending on the processor load, two cases of 50us and 100us is chosen for performance study. Figure 3.9 shows values for 50 microseconds (us) and similar calculations are done also for 100 microseconds where the values listed in the figure is incremented by another 50 us.

$$PI = L_i/768000 \tag{3.6}$$

Figure 3.10 shows the number of slots available(S_a) for ACK process time of 50us, with which the utilization can be calculated as shown in Eqn. 3.5. Figure 3.11shows the same for ACK process time of 100us.



(a) 50us

Figure 3.10: Number of slots available when using ACK for every packet

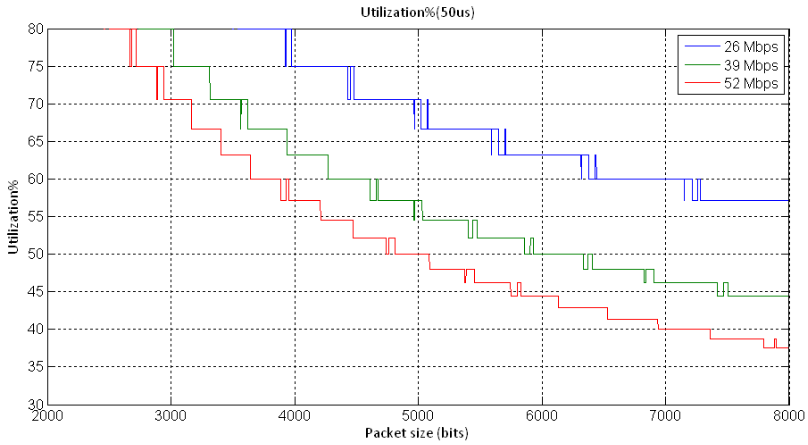
3.5.2 Detailed analysis when using Block ACK

As discussed in Section 3.2, under different packet loss scenarios, there is a possibility of losing the purpose of block ACK as there is a fixed slot allocated for this packet. Since the packets in the previous slots are lost, choosing a specific slot for block ACK might not work always. There is however an advantage with this method which is, the value of slot duration chosen becomes shorter (Fig. 3.12) as there is no ACK for each packet being sent. However, there needs to be a separate slot allocated for block ACK purpose. Thus, the calculation of the values of parameters is as follows.

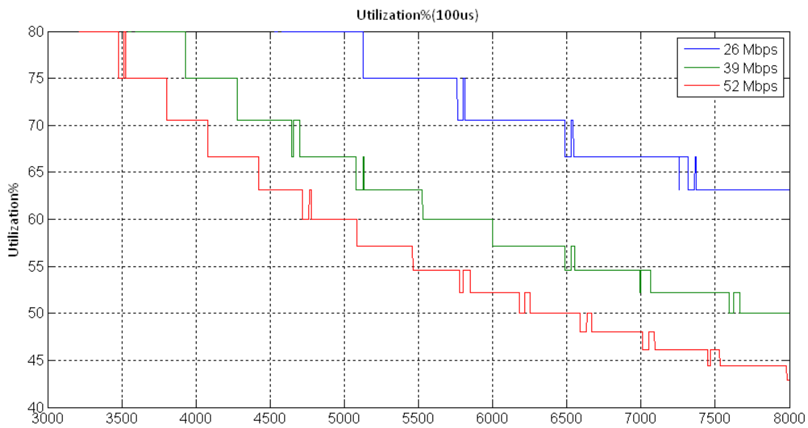
Figure 3.12 shows the slot duration values for various packet sizes for block ACK policy. Figure 3.13 shows the utilization factor that is calculated as in the Eqn. 3.5. Note that here, since there is a separate slot allocated for block ACK, the total number of slots needed = 13 (Section 3.4 for calculation).

3.5.3 Detailed analysis when using No ACK

Here, the slot duration value is equal to the values shown in the Figure 3.12. There is no separate slot allocated for ACK purpose hence the total number of slots remain equal to 12. Figure 3.14 shows the values of slots available and utilization factor.



(a) 50us



(b) 100us

Figure 3.11: Utilization %

3.6 Comparison and Conclusion

In the above graphs, we see a step wise increase in slot duration (S_d) and thus a step wise increase in utilization factor. The reason for step wise increase is that, there are certain 'x' number of bits supported by each OFDM symbol and for the packet sizes varying within this 'x' limit bits, the slot duration remains the same.

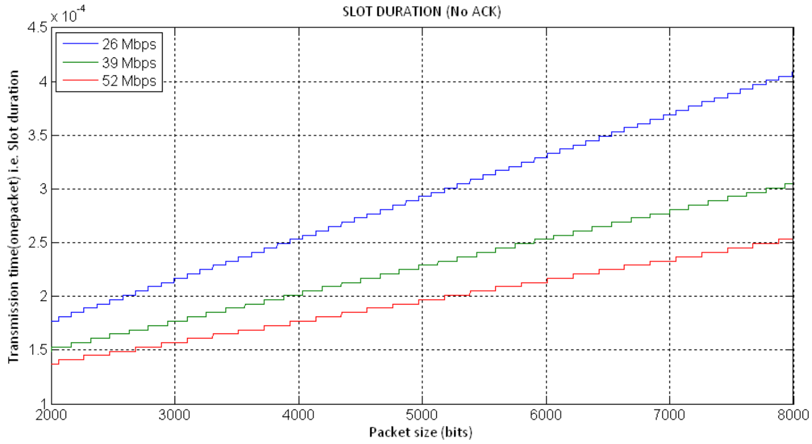


Figure 3.12: Slot duration without ACK and its waiting and process time

The optimum data rate is chosen close to the minimum (to provide reliable and long range transmission) that can provide the targeted performance and that is 39 Mbps. Table 3.1 shows the optimum values to be chosen for system working with respective ACK policies mentioned.

From the Table 3.1, we see that with block ACK policy and no ACK policy, we are close to the targeted delay of 10ms end to end delay (E_d = time taken for the packet to be sent from application layer of sender to the receiver). No ACK policy is chosen to be implemented as part of WHAP design. Note that, the delay value for this policy is calculated to be 9.06 ms with 50% utilization. But, it would be wise to choose packet size that provides us with utilization lesser than 50% giving a flexible margin for randomly behaving interference that may coexist with our system. Thus, choosing a packet interval of 5ms with packet size of 3840 bits is concluded as an optimum value and listed in Table 3.2. With the obtained optimum values, we proceed further with our discussion on how the WHAP enabled system works under different interference scenarios.

3.7 Freezing Time

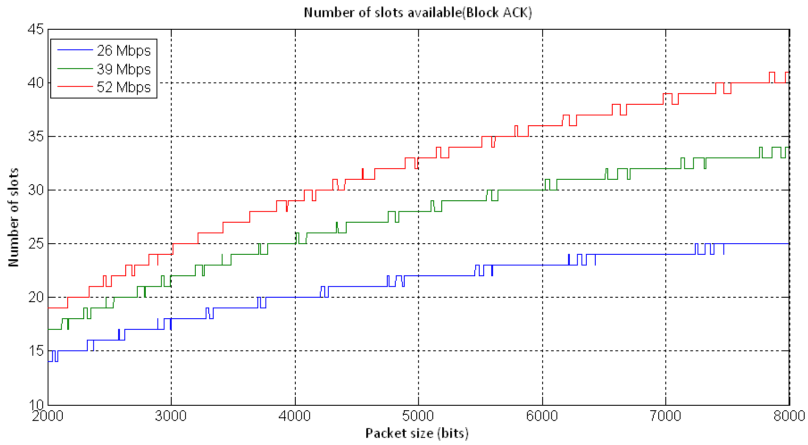
The above considered values may be true for perfect no interference conditions but in practice, there are interferences' that have to be taken into account. During interference, the channel is sensed busy and the backoff time considered in the above sections freezes without decrementing hence this freezing time i.e., time occupied by the interference needs to be taken into account. Figure 3.15 shows the different categories of interference that need to be considered.

Param	ACK				ACK				BlockACK				No ACK			
P_t	50us				100us				0us				0us			
DR	39(Mbps)				39(Mbps)				39(Mbps)				39(Mbps)			
S_d	0.29(ms)				0.37(ms)				0.205(ms)				0.185(ms)			
L_i	5900 bits				7650 bits				4094 bits				3480 bits			
PI	7.68ms				9.96ms				5.33ms				4.53ms			
E_d	15.36ms				19.92ms				10.66ms				9.06ms			
N_r	1				1				1				1			
N_a	4	3	2	1	4	3	2	1	4	3	2	1	4	3	2	1
S_a	24	24	24	24	24	24	24	24	26	26	26	26	24	24	24	24
S_n	12	10	8	6	12	10	8	6	13	11	9	7	12	10	8	6
$U\%$	50	38	30	20	50	38	30	20	50	42	35	27	50	38	30	20

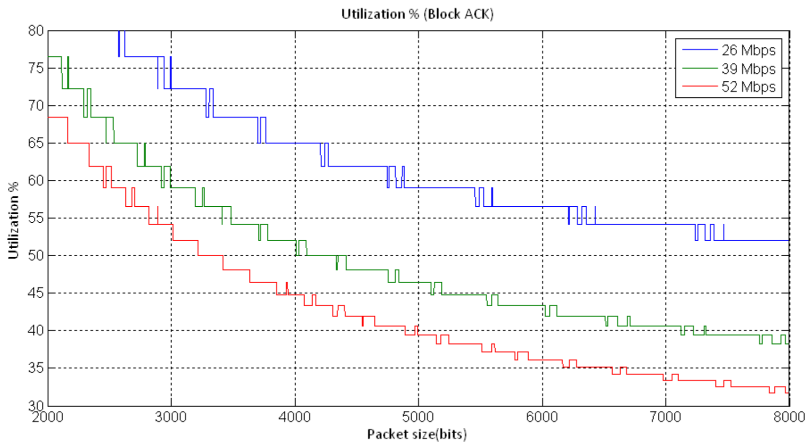
Table 3.1: Values of parameters involved for all ACK policy's

WHAP Parameters(No ACK policy)	Optimum value			
Data rate(DR)	39 Mbps			
Slot Duration(S_d)	0.20 ms			
Packet size(Payload)	3840 bits			
Packet interval(PI)	5 ms			
End DELAY(one-way)=(2*PI)	10 ms			
Retransmissions per station(N_r)	1			
Number of stations active(N_a)	4	3	2	1
Total number of slots available(S_a)	25	25	25	25
Total number of slots used(S_n)	12	10	8	6
Utilization %(U)	48	40	32	24

Table 3.2: Optimum values chosen for WHAP system



(a) Number of slots available

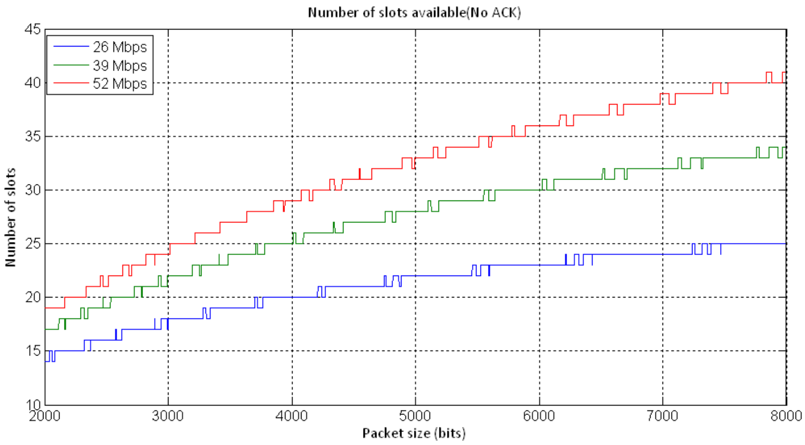


(b) Utilization%

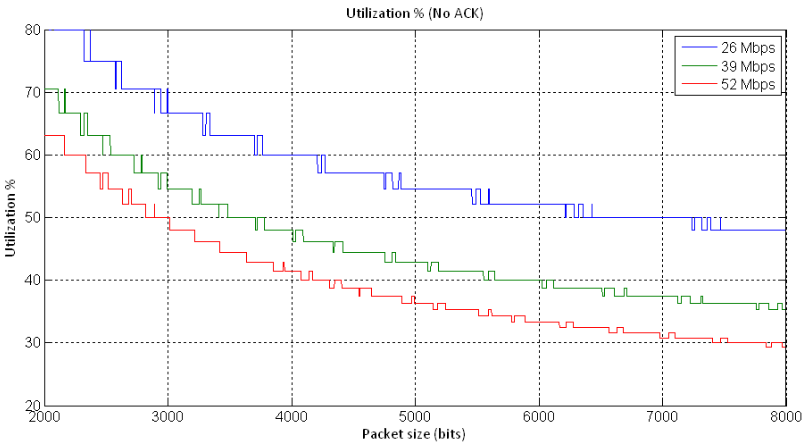
Figure 3.13: Number of slots available(S_a) and utilization % (Block ACK)

Jamming type interference includes the use of the frequency by the devices without a channel sense mechanism. In case of using 2.4 GHz frequency band, interference from bluetooth devices, microwaves represent jamming type interference. In 5 GHz frequency band radar signals represent jammer type interference.

Wi-Fi based inteference represents interference from Wi-Fi devices that include the mandatory the channel sense mechanism (Section 2.1.1). In or-



(a) Number of slots available



(b) Utilization%

Figure 3.14: Number of slots available(S_a) and utilization % (No ACK)

der to understand the type of interference behaviour with which the WHAP enabled audio conferencing BSS can coexist and provide the best performance, the packet sizes for respective data rates of the interfering network and its effect on the WHAP BSS is discussed.

In case interference, the freezing time can be related to the medium access delay time of the WHAP BSS. With the assumption of WHAP system coexisting with 50% interference (as targeted), we see that the maximum

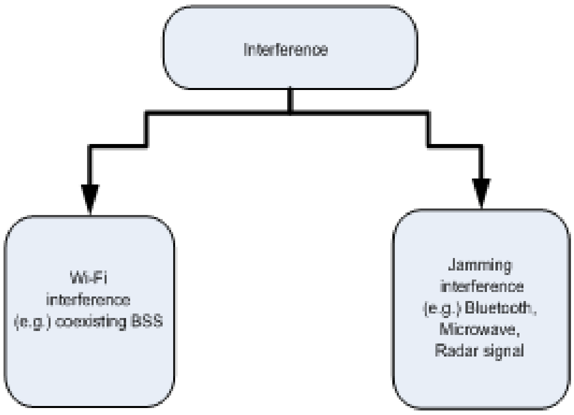


Figure 3.15: Interference categories

tolerable medium access delay of WHAP BSS station is $\leq$ 2.5ms (50% time) of one packet interval (5ms). From this analysis we can say that WHAP BSS stations can provide the best of QoS performance when the interferer follows the rule that the size of the packets transmitted from the stations of coexisting BSS should be such that it occupies only 50% of the medium every WHAP BSS packet interval.

A general idea on packet sizes from coexisting BSS that can be tolerated by WHAP stations are calculated and shown in the Figure 3.16. Parameters of coexisting BSS that affect the freezing time of WHAP BSS,

- Data rate: 1 Mbps to 65 Mbps.
- Packet size: 100 bytes to 8000 bytes.
- Start time: Can vary from slot n to slot n+25 within a packet interval of WHAP BSS.

Figure 3.16 shows the packet sizes for respective data rates of the coexisting BSS that correspond to packet loss that occur in WHAP BSS stations. Packet sizes of up to 8000 byte packet size (aggregated payload + header) from coexisting stations are considered.

3.8 Network Scalability

In the above discussions regarding the slot allocation, 2 slots have been allocated for the purpose of contention. With this time allocation, it is possible to predict theoretically the number of stations that can be associated within a WHAP BSS. The calculations used for this purpose are shown below.

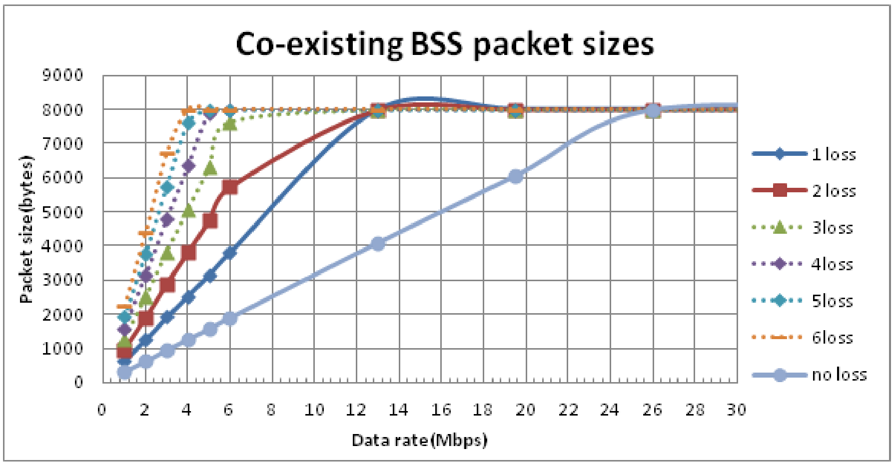


Figure 3.16: Packet sizes of co-existing BSS influencing the WHAP BSS system

Message type	Access Category	Waiting Time	Size (bits)	Data Rate (Mbps)	Txtime + Wait time (ms)	Frequ ency (Hz)
Beacon	NIL	SIFS(16us)	912	1	0.921	4
Sync	Audio	AIFS(34us)	128	39	0.074	10
				13	0.14	10
Control messages	Background	AIFS(79us)	64	39	0.083	4
Keep alive	NIL	DIFS(34us)	176	1	0.21	0.1

Table 3.3: Time occupied by the control traffic

We have allocated 2 slots (0.4 ms) for the contention (EDCA) access where the control traffic, keep-alive message, beacon message, sync message are all to be accommodated. Table 3.3 lists the total time needed to be allocated for these messages. Applying the same calculation as in the Section 3.4 which uses standard methods for transmission time calculation for various data rates, we get the values shown in the Table 3.3. The frequency of the messages considered are arbitrary values that are considered also in the OPNET simulated environment.

Every 5 ms we have 0.40ms for contention, Hence for 250 ms, we have 20 ms in total for contention out of which,

Case 1: when the synchronization message is transmitted at a data rate of 39 Mbps

- Sync messages = 2.5 → 2.5*0.074 ms = 0.185 ms used
- Beacon message= 1 → 1*0.912 ms =0.912 ms used
- Remaining time= 20 ms - 1.097 ms = 18.90 ms.
- Keep alive= 0.21 ms used every 10 seconds, so 0.21/40 = 0.0052 ms used for 250 ms by one station. Thus, for 'N' number of stations it is (N*0.0052 ms)
- 'N' stations with each having transmission time of 0.083 ms.

Thus a total of N = 213 messages can be accommodated in 250 ms, if they all do not start transmitting such that they all backoff with a value chosen from [0,15].

If they all start their transmission such that, they all backoff choosing the maximum of 15 slots then,

- 'X' stations occupy (X * 0.083ms)
- Total backoff time of (X * 0.135) where, backoff time = 15*0.009ms = 0.135ms can be chosen at the maximum by a station.

This leads us to conclude that a total of X = 82 messages (6.80ms + 11.07ms) can be sent if they start transmission at the same time (b_i^j is chosen from window [0,15] for the first transmission attempt). That is 41 stations send control messages upstream and the AP responds with 41 respective control messages.

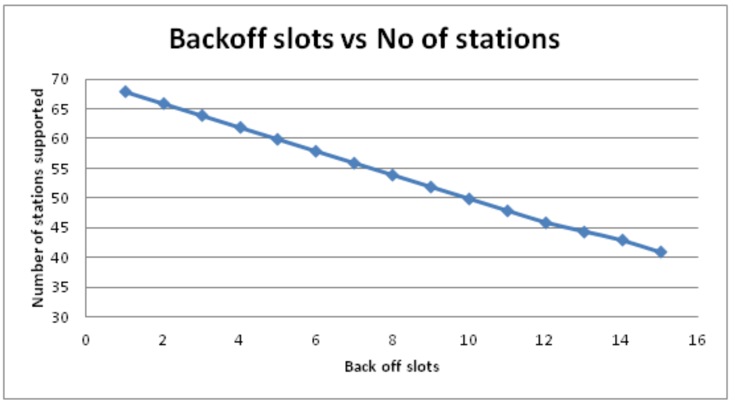


Figure 3.17: Number of stations supported based on backoff time

In the above discussions we have considered the worst case scenario of all the stations transmitting such that all of them chose the maximum backoff time. But practically each of the stations can chose values between [0,15] and Figure 3.17 shows the varying number of stations that can be supported based on the backoff time chosen by the stations.

Case 2: When the synchronization message is transmitted at a data rate of 13 Mbps

- Sync messages = 2.5 $\rightarrow$ 2.5*0.14ms = 0.35ms used
- Beacon message= 1 $\rightarrow$ 1*0.912 ms =0.912ms used
- Remaining time= 20ms - 1.2725ms = 18.73ms.
- Keep alive 0.21ms used every 10 seconds, so 0.21/40 = 0.0052 ms used for 250ms.
- 'N' stations with each having transmission time of 0.083 ms.

Thus a total of N = 212 messages can be accommodated in 250 ms, if they all do not start transmitting such that they all backoff. The difference is only one message and the rest of the calculations are similar to that stated above.

4 Analysis and Implementation

So far, we have discussed the basic WHAP protocol and behaviour of its scheduled transmission of traffic under certain interference conditions. But it is important to know the system behaviour in a more practical environment that not only includes interference conditions assumed in Chapter 3 but also interference that can random. For this purpose, this chapter first introduces the two important parameters of the interferer that affects the WHAP system performance greatly. The WHAP design is then implemented in the stations using OPNET. Simulation of the WHAP BSS (the entire audio conference system setup) under various interference conditions is analysed.

4.1 Application level Retransmissions

The start time of the interference affects the system greatly because each station decides to send a retransmission based on the channel conditions i.e., each station at its application layer forwards a packet to its MAC and waits for a message trigger that indicates that its MAC queue is empty (forwarding the packet to its PHY). At the application layer, the time difference between the time that the packet was sent (Ts) and the time that the trigger was initiated (Tr) is calculated. If the difference value is > 2.5 ms, then there is no retransmission sent otherwise there is an interrupt generated for retransmission to be sent. Figure 4.1 shows the details of retransmission decision procedure.

The main aim of this technique is to prevent self interference (happens only when the external interference utilization is $> 52\%$) due to retransmissions that are sent after a packet interval time such that, it involves occupancy in the consecutive packet interval. With this technique applied, we are able to reduce that self interference to 20% at the maximum (otherwise it will increase to 40%). This technique works well when the start

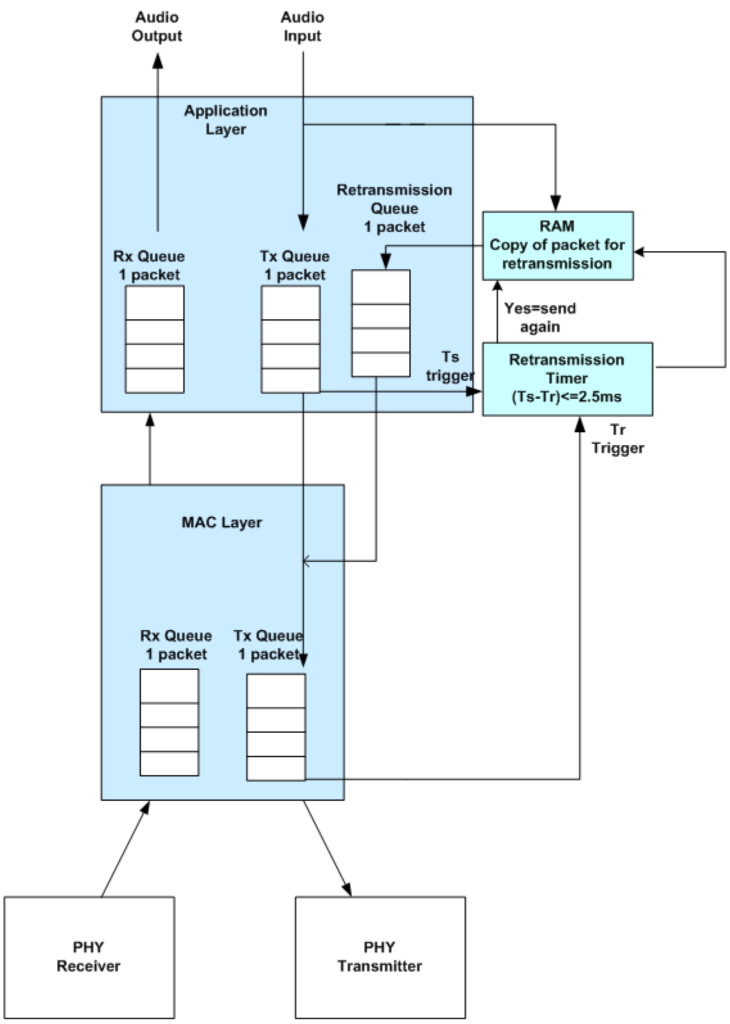


Figure 4.1: Retransmission decision at each station enabled with WHAP

time of the interference (> 52%) is exactly same as that of WHAP packet interval start time when the first packet from the first station is sent. When the interference (> 52%) start time is after the first 5 successful packet transmissions of WHAP station, then according to this technique, there are retransmissions sent for each of the stations. However, even in this case, the self interference remains as a maximum of 20%.

4.2 Interference utilization limit

A general view of the system behavior when the interference **does not** exceed the allowed limit (=52%) is discussed below (Fig. 4.2). Note that

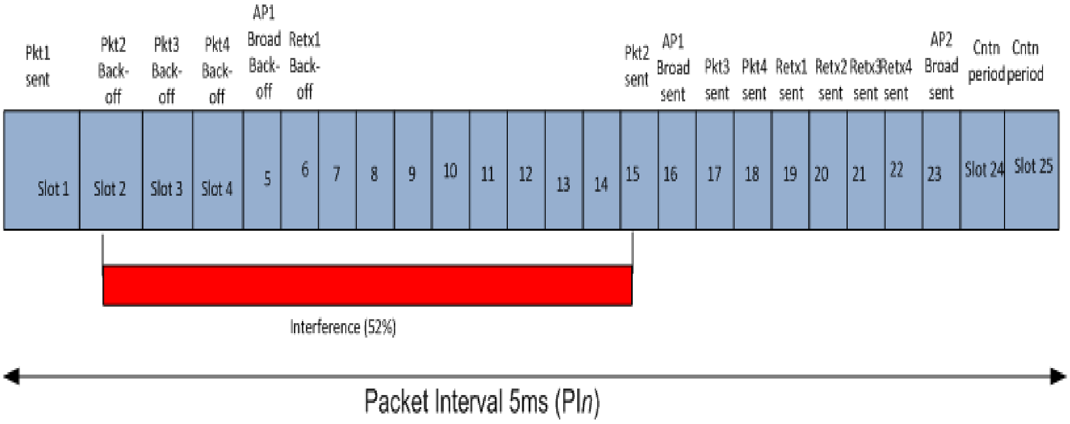


Figure 4.2: General working of WHAP enabled stations when interference is 52%

the start time of the interferer (worst case type:jammer is chosen here) in this case does not affect the WHAP system performance i.e., when the interferer occupies exactly 52% slots every packet interval (PI_n) we need to have a system with WHAP that can make use of 48% slots whatever may be the start time of the interferer.

Note that, for the stations that have their packets backed off, there occurs a possibility that the stations chose the same backoff time from the interval [0,3] (Tab. 2.1). Let, b_i^j denote the backoff value chosen by the i^{th} station for the j^{th} packet transmission, then this value is given by the Eqn. 4.1 where, $rand[0, CW]$ is chosen from a discrete uniform distribution of values from the window [0,3]. The distribution of the $rand[0, CW]$ is shown in the Eqn. 4.2 where, y denotes the probability of choosing a value from $1..N$.

$$b_i^j = rand[0, CW] \quad CW = 0, \ldots, 3 \tag{4.1}$$

In cases where the stations choose the same backoff time, there occurs packet collision and the throughput reduces due to loss of packet. However, since the interference is exactly = 52%, there are retransmissions sent that compensate for the loss.

$$y = f(x \mid N) = (1/N)_{I_{1,\ldots,N}(x)} \tag{4.2}$$

General Description

- We have 12 slots (48% slots in one PI_n) in total for our audio packet transmission.

- Each of the clients, tries to send its packet during its respective slot allocated to them.
- Due to interference, there occurs backoff at any or all of these slots. Since we can compensate up to 52% of interference, after 13 slots of interference, the system regains the channel and continues to transmit at the respective slots.
- A total of 8 slot times are used only for audio packets i.e., 1 slot for each active client(N_a) and 1 for retransmission which is decided by the station depending on the channel condition. AP broadcasts the packet to all the clients associated at the fixed slot (slot 5) and it retransmits it depending on the channel condition.
- Finally comes the contention period where the passive stations can access the channel to exchange control traffic packets. Since they are the low priority packets they wait the longest to get access to the channel. If the system works as predicted, they get 2 slot durations only for contention.

Thus for such a case, WHAP with no ACK policy works well and can provide the targeted performance. Here since the interferer does not exceed the 52% limit, the N_a send their packets and their retransmissions successfully.

4.3 Analysis with varying interference utilization

Though the allowed interference is calculated as 52% it is important to be able to predict the performance of WHAP systems when the interference increases beyond the 52%. The technique described in Section 4.1 helps preventing self interference in this kind of situation. Further, the scenarios considered in this section cover the worst case interference conditions to examine the performance of WHAP system.

When the interference occupies 100% slots in a PI_n, the system behaves as shown in Figure 4.3 with self interference being a maximum of 20%. There is drop of one packet from each station as this packet does not reach in time (causing self interference) for processing at AP. As discussed in Section 4.1, there is retransmission of the packet only when the MAC queue is empty within specified time limit and in such case, this condition is not satisfied and hence no retransmissions. Thus, in the upcoming packet interval (PI_{n+1}), only the backed off packets are sent which acts as self interference for PI_{n+1}. Here again the backoff time chosen by the stations can be the same and there can occur collision of packets. In that case, the self interference becomes minimum i.e., 4% (1 slot) and throughput is reduced,

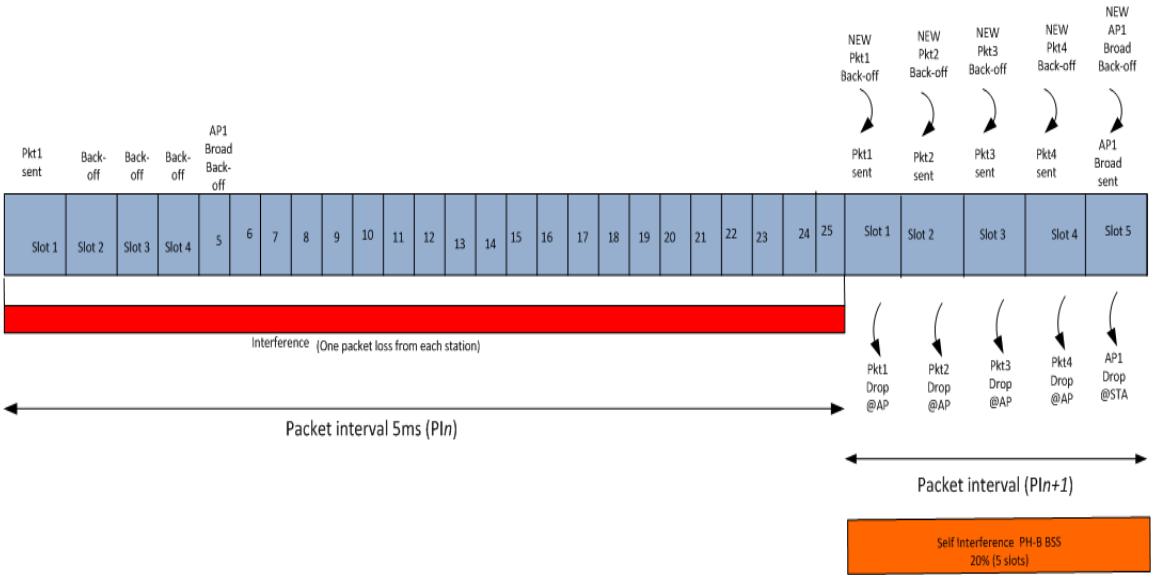


Figure 4.3: General working of WHAP enabled stations when interference is 100%

however depending on the interference level at PI_{n+1} the performance is decided. The scenarios discussed here assumes the worst case self interference of 20% with the stations not choosing the same backoff time.

1. There is increase in the utilization factor of the upcoming interval (PI_{n+1}) by 20% due to self interference but if there is no external interference in that interval, then WHAP becomes stable within that packet interval.
2. In the PI_{n+1} interval if there is 52% interference from the jammer and 20% interference from PI_n adds up to 72% interference in the PI_{n+1}. Now, WHAP system can recover from this excess interference depending on the start time of the external interferer in that interval PI_{n+1}. There are two scenarios considered to explain this condition as shown below.

4.3.1 Scenario (1): Start time of interference and WHAP stations are equal

If the start time is such that the stations in that packet interval (PI_{n+1}) do not send retransmissions (Fig. 4.4) then its utilization for that interval reduces to 28% (7/25 slots, as 5 slots needed for retransmission not used out of total 12 slots) and thus it can tolerate up to 72% interference (18/25 slots). Thus, AP receives packets with overall throughput 100% even with no

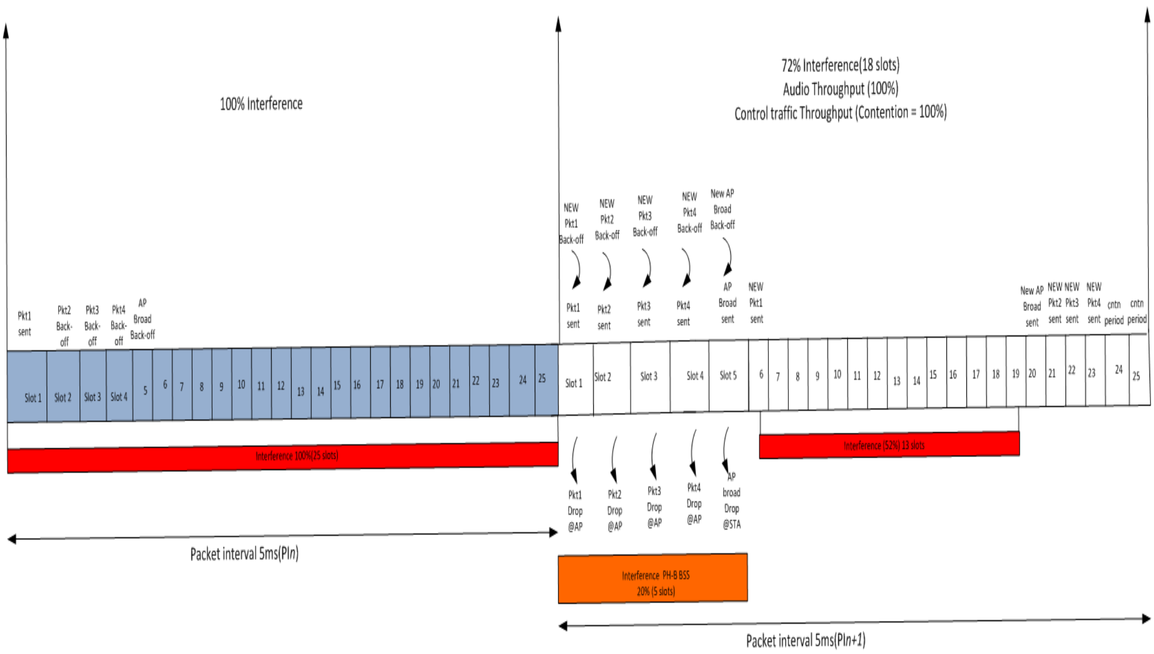


Figure 4.4: Working of WHAP enabled stations in scenario(1)

retransmission packet received from the N_a whereas, the associated clients also have a throughput of 100% even with AP not retransmitting either. This is considered as the best the system can provide and this is mentioned as ‘throughput’ in Fig. 4.4.

4.3.2 Scenario (2): Start time of interference and WHAP stations are not equal

If the start time is such that there are 4 retransmissions in the interval PI_{n+1} with the N_a utilization =48% in that interval, then the maximum tolerable interference utilization is 52%. But since the total interference utilization% is 72% (external + self), this causes reduction in throughput to less than 100% with self interference of 8% propagating to packet interval PI_{n+2} (Figure 4.5) in this case.

Considering a different version of this case as shown in the Fig. 4.5 where an increased interference utilization (>52) in PI_{n+1} interval is acting on the WHAP BSS stations. Here the WHAP system becomes stable only when it gets enough slots to transmit PI_n and the $PIn + 1$ buffered packets (Fig. 4.6). The self interference propagates to PI_{n+2} and the stability depends on the interference behavior in that interval.

Thus, from the above discussion, we can conclude that the when the interference utilization increases more than what the system was designed

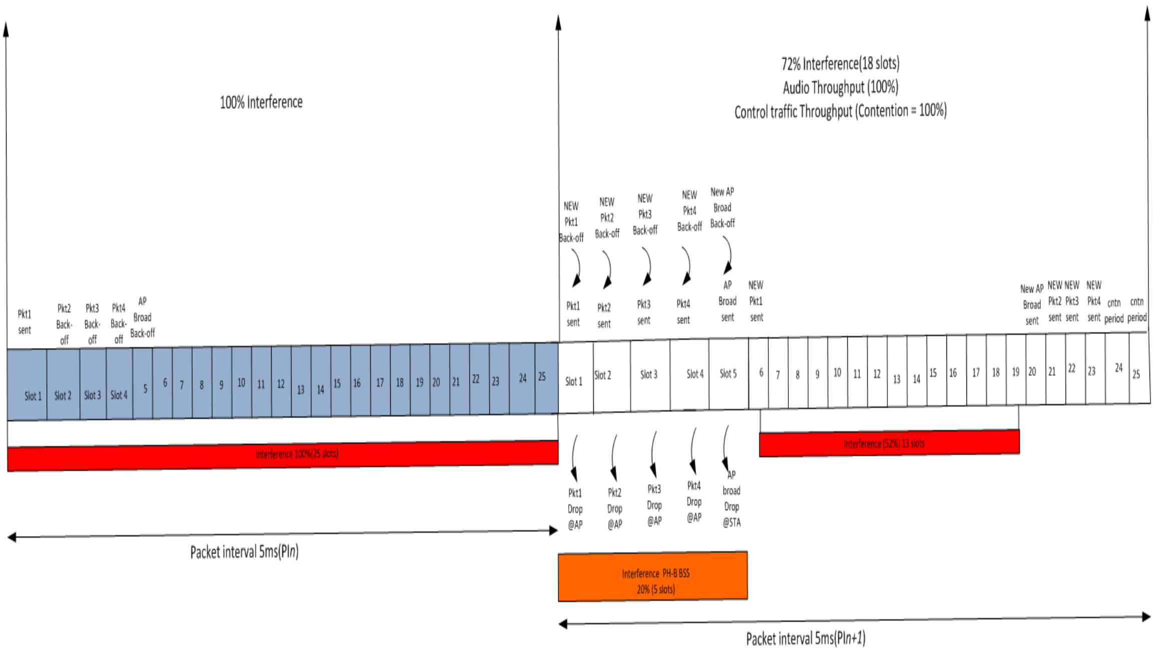


Figure 4.5: Working of WHAP enabled stations in scenario(2)

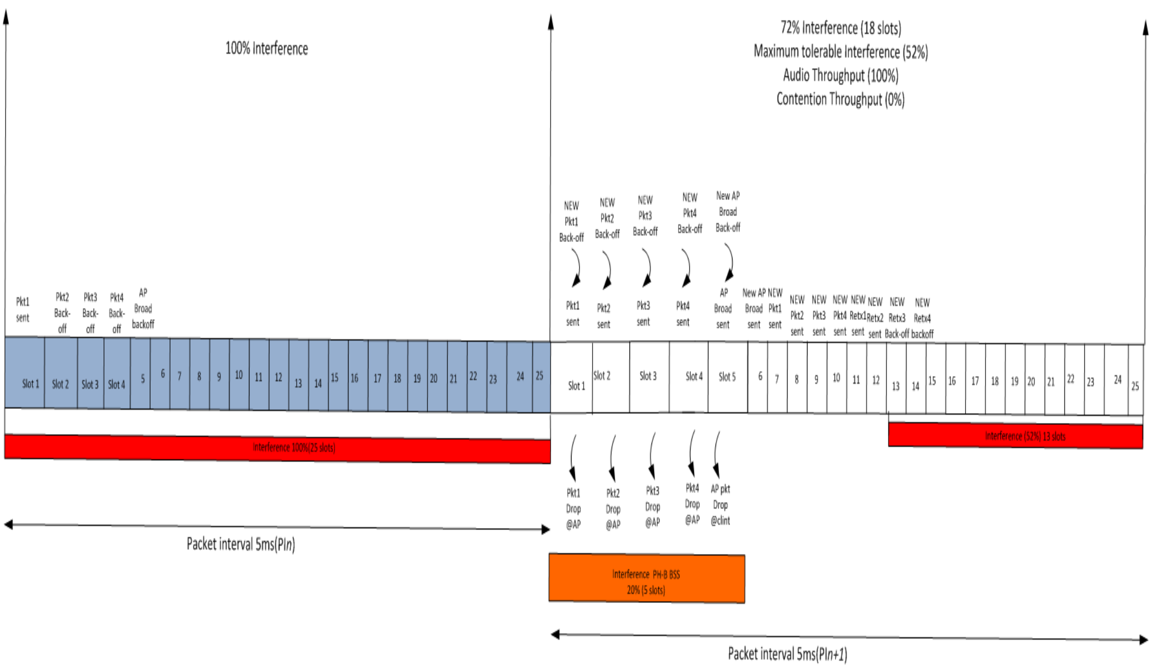


Figure 4.6: Working of WHAP enabled stations in scenario(2) extended version

for (=52%) the performance totally depends on the start time of the interferer with respect to the WHAP slot timing. In case of higher interference utilization, the WHAP performance is best when the start time of the interference is same as that of the start time of the packet interval (Fig. 4.4) and the same performance can be achieved for lower utilization % of interference when the start time is after first 5 successful transmissions (Fig.4.5). These scenarios need to be simulated in OPNET to see if we are able to achieve the throughput as predicted.

4.4 OPNET Implementation

Optimized Network Engineering tool (OPNET) helps analysing the performance of various protocol and network standards. In this work the focus is on 802.11e amendment. This section discusses the implementation of WHAP onto the wireless stations and analysing the performance of the entire audio conferencing BSS in the presence of various interferers.

Basic Hierarchy

As discussed in Section 2.2.1, the hierarchy of the editors along with the respective system configurations are shown Fig. 4.7 where the WHAP protocol is implemented in the source module of the node editor. This is because, application layer controls are from the this source module. In the editors shown, necessary changes are made in order to obtain the working of system with WHAP as discussed in Chapter 3. For example, the node editor in the Figure 4.7 shows the implementation of retransmission decision from the MAC queue information as discussed in Section 4.1 and a high quality audio conference system setup is simulated in OPNET (Project editor in Fig. 4.7) with parameters as mentioned in the Table 4.1.

4.5 OPNET Results

In order to analyse the performance of the WHAP enabled BSS, four parameters are measured from the simulated network using OPNET (Fig. 4.8). The parameters are,

(a)Medium Utilization [%] which gives the time occupied by the WHAP active and passive stations within the packet interval time, similar to calculations in Section 3.4 except that here, the waiting time of the stations are not included.

(b)Medium access Delay value representing the total of queuing and contention delay at the MAC layer. This value is similar to the value of the parameter considered in Section 4.1. The medium access delay

Bosch AP	**PHY specifi cations**	**MAC specifi cations**	**APP specifi cations**
	5GHz channel 36	802.11n HT 5 GHz	Packet size=3840 bits
	Guard inter-val(800ns)	QoS EDCA parameter set	Packet inter-val=5ms
	Spatial streams=1	QoS ACK pol-icy(NO ACK)	2 packets every 5 ms
	Data rate 39 Mbps	Beacon inter-val=200ms	Start time= 1.3ms
Bosch active STA	**PHY specifi cations**	**MAC specifi cations**	**APP specifi cations**
	5GHz channel 36	802.11n HT 5 GHz	Packet size=3840 bits
	Guard inter-val(800ns)	QoS EDCA parameter set	Packet inter-val=5ms
	Spatial streams=1	QoS ACK pol-icy(NO ACK)	2 packets every 5 ms
	Data rate 39 Mbps	Beacon inter-val=200ms	Start time= 0.1ms
Bosch passive STA	**PHY specifi cations**	**MAC specifi cations**	**APP specifi cations**
	5GHz channel 36	802.11n HT 5 GHz	Packet size=128 bits
	Guard inter-val(800ns)	QoS EDCA parameter set	Packet inter-val=5ms
	Spatial streams=1	QoS ACK pol-icy(NO ACK)	2 packets every 5 ms
	Data rate 39 Mbps	Beacon inter-val=200ms	Start time= 4.1ms

Table 4.1: Parameters considered in the OPNET simulated network

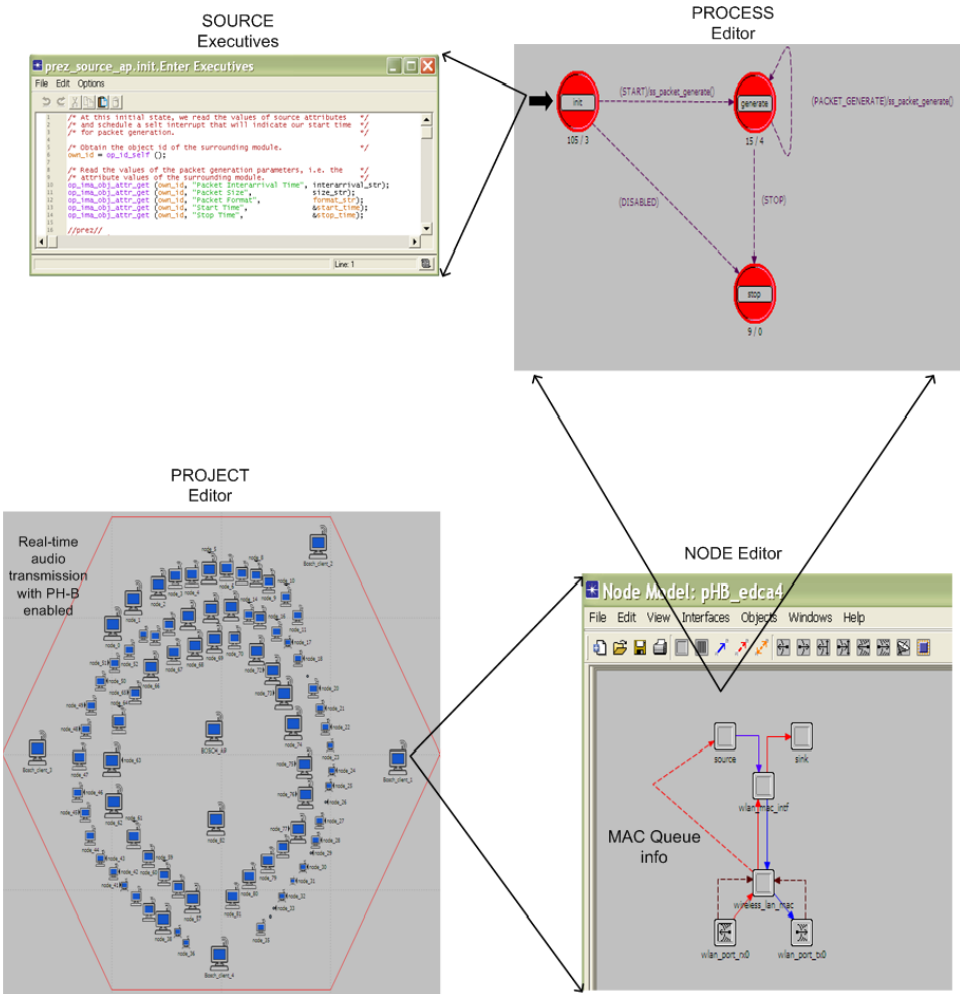


Figure 4.7: Basic architecture of OPNET network structure

for audio traffic should be not > 2.5 ms whereas, the medium access delay of the control traffic should preferably be ≤ 250 ms.

(c) **Audio throughput at each station** denotes the total number of bits that are received and forwarded to the higher layer successfully at each individual station. This is similar to the "throughput" value described in the Section 4.3.1.

(d) **Control traffic throughput at the AP** represents the total number of bits that are received and forwarded to the higher layer successfully at the AP. The reason for this measurement is that, since these are low priority yet important traffic sent by the passive stations, it is necessary to analyse the effect of interference on these kind of traffic too.

WHAP enabled BSS here means that, the network formed with 4 active audio traffic transmitting stations (Bosch clients in the Fig. 4.8) along with 82 passive stations transmitting one way upstream contention traffic to the AP. Note that all the stations have WHAP protocol implemented at their application layer hence the name WHAP enabled BSS. The project editor in the Fig. 4.7 represents this network.

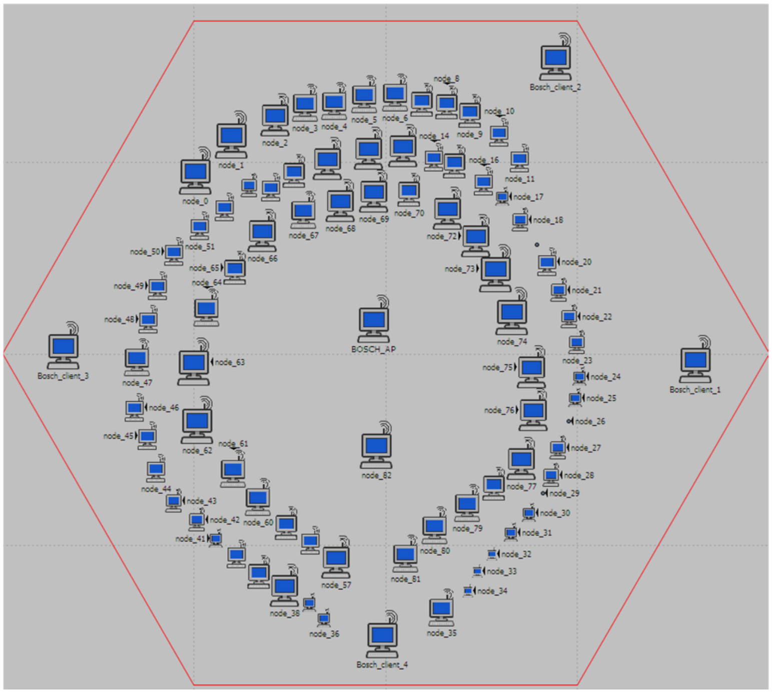


Figure 4.8: The real-time audio transmission system(WHAP BSS) with no interference

4.5.1 No interference case

For analysis of this case, the WHAP enabled BSS alone is considered without any interferer stations as shown in the Figure 4.8.

The performance values obtained from this case are used as a “reference” for all other performances obtained from different interference scenarios. This helps one understand that the WHAP BSS performance values in case of interference that are close to that of the reference values (without interference) are concluded as the best performance that can be obtained from the WHAP enabled audio conferencing system.

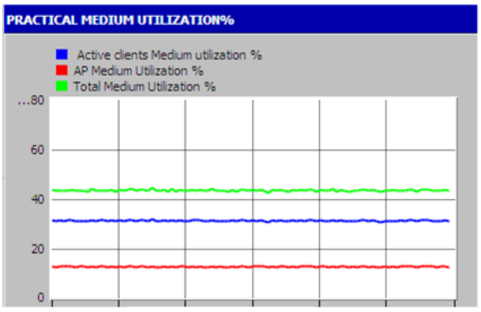
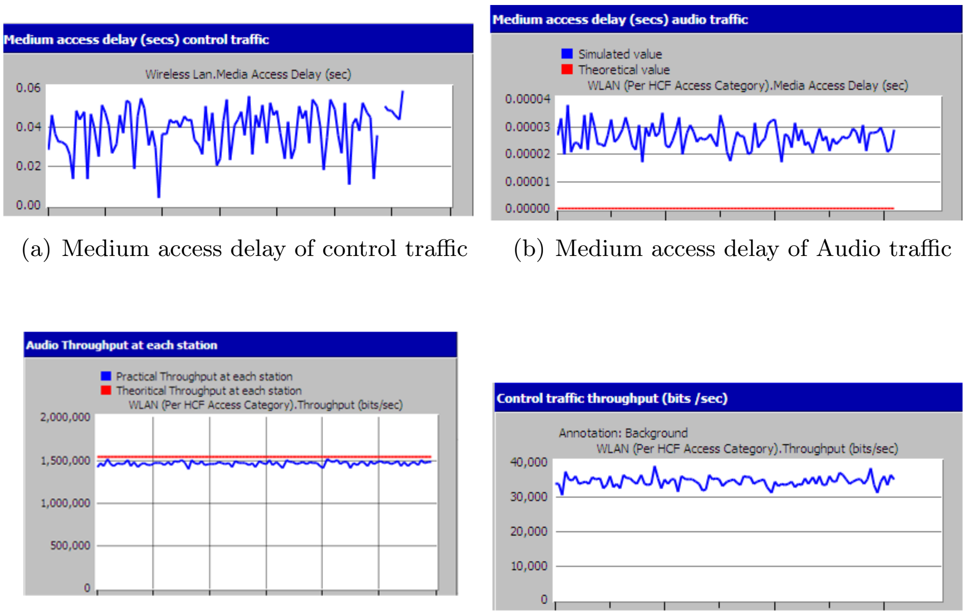


Figure 4.9: Total medium utilization %



(a) Medium access delay of control traffic (b) Medium access delay of Audio traffic

(c) Audio throughput at each station (d) Control traffic throughput

Figure 4.10: Performance of parameters in no interference case

Theory	OPNET Results(Fig. 4.9)
2 audio packets each station (8 slots = 8/25=**32%** utilization)	clients medium utilization = **32%** Includes the connection request
AP broadcasts 2 packets and used 2 slots(2 slots= **8%** utilized)	AP utilization= **12%** .This includes the response to connection response
Other stations send connection requests and thus their total utilization= 2 slots=**8%**	Total Medium utilization=**44%**
TOTAL UTILIZATION: **48%**	TOTAL UTILIZATION:**44%**

Table 4.2: Theoretical and Practical medium utilization values

Result Analysis:

Figure 4.9 shows the utilization efficiency percentage of the WHAP enabled BSS (see Tab. 4.2 for theoretical comparison). The difference in the utilization factor is because the simulated utilization calculation does not include the waiting time of the stations. Figure 4.10(b) shows a comparison of medium access delay expected to that obtained in simulation. Theoretically, this value is expected to be zero but the in simulation, this is not completely true due the random contention based control traffic transmissions by the passive stations. Fig. 4.10(a) shows the medium access delay for the control traffic (preferable value $\leq$ 250 ms).

Figure 4.10(c) shows both theoretical and simulated results of audio throughput at each of the stations. Since 2 audio packets (1 original + 1 retransmission) are being broadcasted by the AP, the theoretical value is 2*768000 = 1.5Mbps. The simulated results show a value very close to this. The deviation in the throughput is due to the contention based control traffic occupying the channel. Figure 4.10(d)shows the control traffic sent from the passive stations.

4.5.2 Interference type:JAMMER

The Figure 4.11 shows the WHAP BSS network coexisting with the jammer type interference i.e., using the same frequency channel as that of the WHAP BSS. For this case, two scenarios are simulated that are similar to the ones considered in Section 4.3. Since we have predicted theoretically the expected results, we shall compare them with the result obtained from the simulated environment.

Scenario (1) simulation:

During the simulation of this scenario, the interferer is assumed to start its transmission exactly at the same time as the first packet transmission

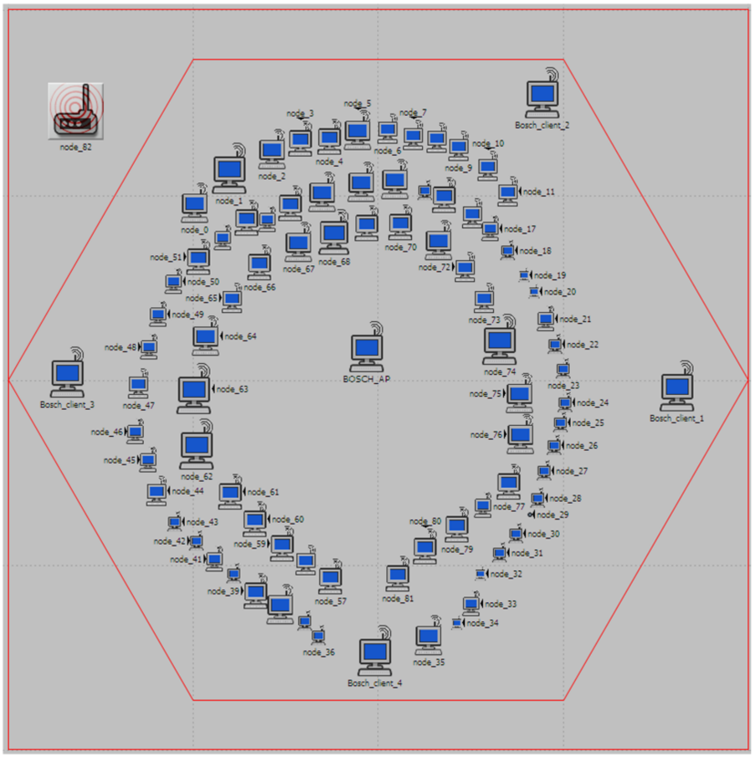


Figure 4.11: The whole audio conferencing system with WHAP enabled stations and JAMMER

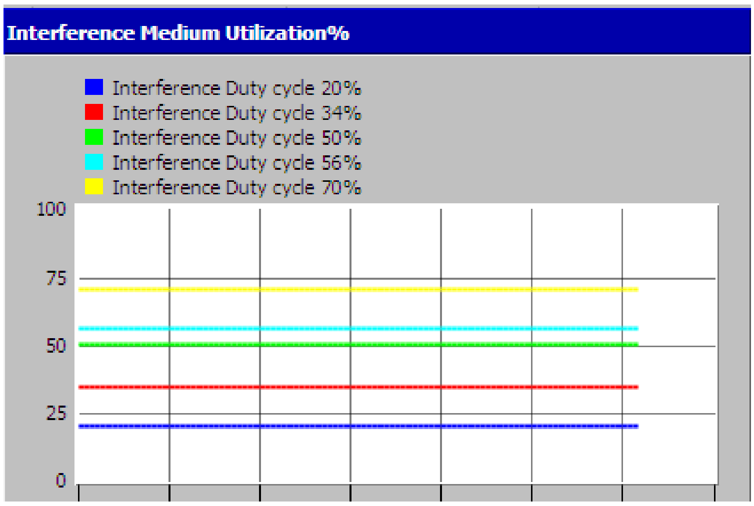
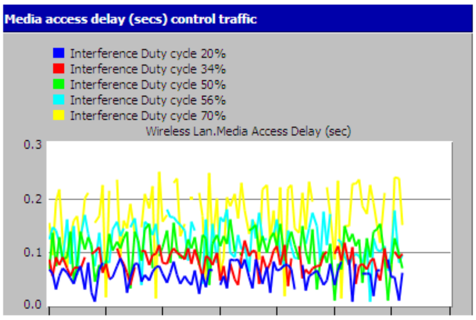
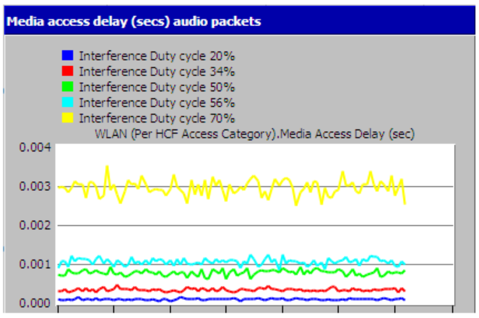


Figure 4.12: Jammer interference utilization %

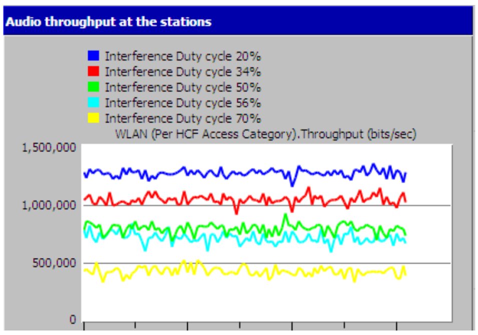
of the active WHAP station in the WHAP BSS. For this scenario, we theoretically discussed that for interference utilization greater than 52% that



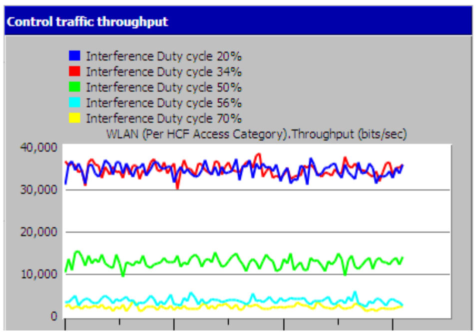
(a) Medium access delay of the control traffic



(b) Medium access delay of the audio traffic

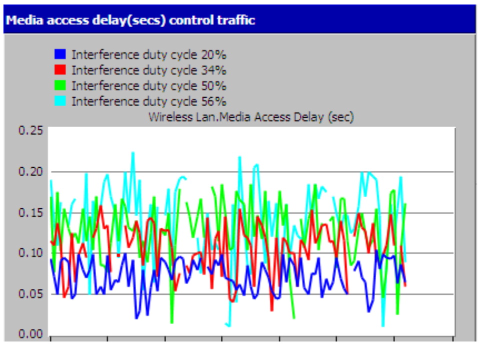


(c) Audio throughput at each station

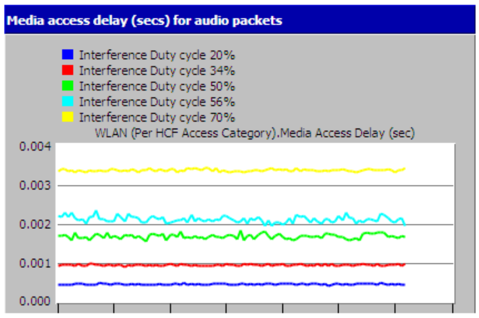


(d) Control traffic throughput

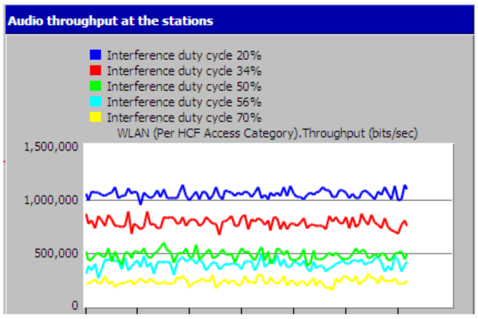
Figure 4.13: Performance of parameters with JAMMER:Scenario(1)



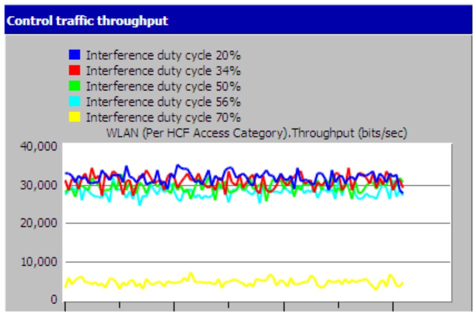
(a) Medium access delay of the control traffic



(b) Medium access delay of the audio traffic



(c) Audio throughput at each station



(d) Control traffic throughput

Figure 4.14: Performance of parameters with JAMMER:Scenario(2)

start their transmission such that there are no retransmissions (start time of interference exactly same as the start time of the WHAP BSS packet interval) by the WHAP active stations then, from the discussion in Section 4.3, theoretical interference tolerance = 72%. Interference tolerance here means that the throughput of 100% (one packet received every station) can still be achieved. For this case, the control traffic (contention) throughput is also 100% in theoretical calculation.

Result Analysis:

Figure 4.12 shows various utilization% of interferer being considered with Figure 4.13(b) showing the respective medium access delay values of each of the active WHAP stations and Figure 4.13(a) shows the medium access delay of each of the passive WHAP stations. From Figure 4.13(c) we can understand that in the simulated environment, an interference of 58% is tolerable (providing 100% throughput) and for all interference greater than 58% the audio throughput reduces drastically. In case of control traffic,

Fig. 4.13(d) the throughput in case of 58% interference remains close to 10%.

The significant difference in the interference utilization factor value calculated in theory and simulation is due to the fact that in OPNET, the simulation setup has the control traffic being sent without a scheduling. This causes a random behaviour due to random backoff of their frames. This random medium occupancy of the control traffic not only reduces the throughput of the audio traffic but also the control traffic at high interference utilization (72%). Note that, the jammer type interference has strict boundaries on the medium occupancy of the WHAP stations. This can be seen in also in upcoming simulations that have jammer type interference.

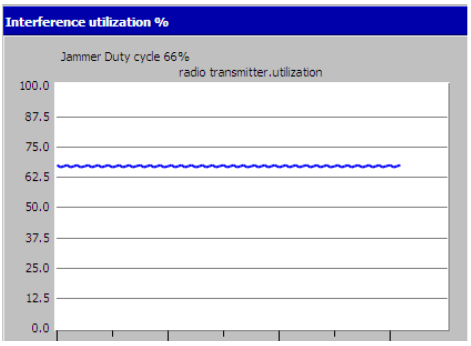


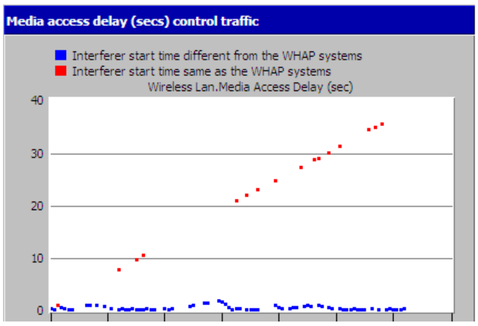
Figure 4.15: Jammer interference utilization % (non-sync)

Scenario(2) simulation:

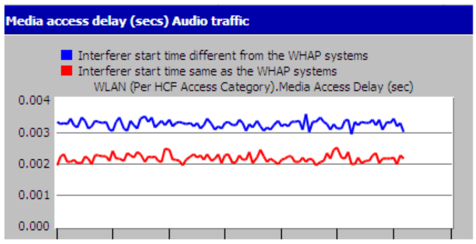
The simulation setup is similar to the one shown in the Figure 4.11. Theoretically for this scenario, we have discussed that for interference utilization % greater than 52% that start transmission such that there are retransmissions (start time of interference after transmission of 5 packets by the WHAP BSS packet) by the WHAP active stations. Then, from the discussion in Section 4.3, theoretical interference tolerance = 52%. Interference tolerance here means that the throughput of 100% (one packet received every station) is achieved.

Result Analysis:

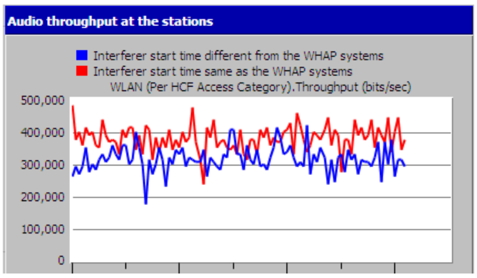
During the simulation of this scenario, the interferer is configured to start its transmission after first 5 packet transmission of the active WHAP station (4 + 1 AP packet) in the WHAP BSS. Figure 4.12 shows various duty cycle % of interferer being considered with Figure 4.14(b) showing the respective medium access delay values of each of the active WHAP stations and Figure 4.14(a) shows the medium access delay of each of the passive WHAP



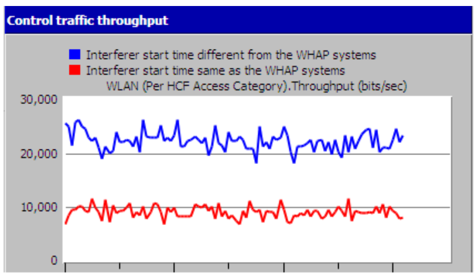
(a) Medium access delay of the control traffic



(b) Medium access delay of the audio traffic



(c) Audio throughput at each station



(d) Control traffic throughput

Figure 4.16: Performance of WHAP parameters with non-sync JAMMER

stations. In Figure 4.14(c) we can see that, in the simulated environment an interference of 34% is tolerable (providing 100% throughput) and for all interference greater than 34% the throughput reduces drastically. In case of control traffic, Figure 4.14(d) the throughput in case of 34% interference remains close to 100%. The control traffic here increases the medium utilization % of the WHAP stations as compared to what was calculated theoretically (0%). This explains the reason why the 100% throughput of the WHAP stations are achieved with lower interference utilization % than calculated theoretically.

4.5.3 Jammer interference that are not synchronized to WHAP stations:

In Section 4.5.2, the jammer interference was configured such that, the packet interval of WHAP stations and the jammer interference are same. In this case, the packet interval for the jammer interference is not synchronized with the WHAP stations. Fig. 4.15 shows the utilization of the jammer interference with respect to its packet interval. Figure 4.16 shows

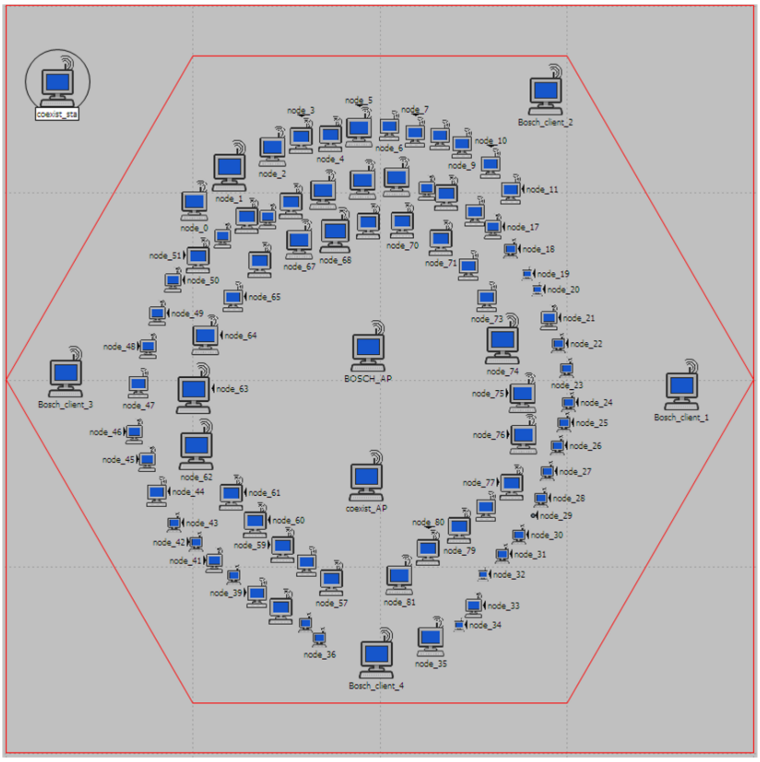


Figure 4.17: The whole audio conferencing system with WHAP enabled stations with Wi-Fi CBR interferer coexisting

the WHAP parameters for non synchronized interference. The performance however is similar to that discussed in the synchronized interference.

4.5.4 Interference type:Wi-Fi with Constant Bit rate traffic

In this case, the audio conferencing WHAP BSS network is simulated along with the Wi-Fi overlapping BSS stations (Fig. 4.17) that are configured such that they transmit traffic with the same packet interval as the WHAP stations. By varying the packet sizes of the interferer, different interference utilization % is introduced and the performance of the WHAP BSS is analysed. Here again, two scenarios are considered, scenario (1) similar to that discussed in Section 4.5.2, has the interference start time same as that of the transmit time of first WHAP station with performance shown in Fig. 4.19. Scenario (2) has the interference to start after first 5 transmissions of WHAP stations with performance shown in Figure 4.21.

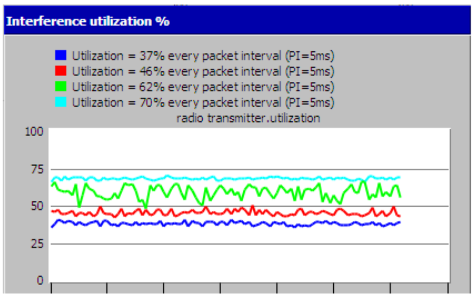
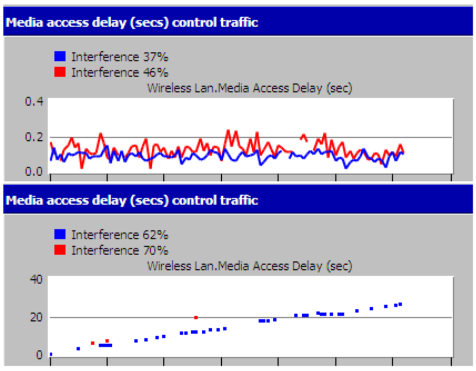


Figure 4.18: Wi-Fi interference utilization % for scenario(1)

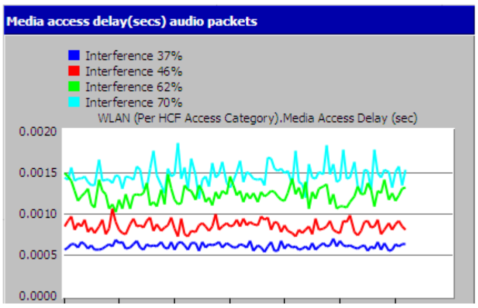
Result Analysis:

The analysis of results obtained from simulation of both scenario(1) and scenario(2) are done similar to the Section 3.5.2. From the analysis, one can see that, in case of Wi-Fi type interferer, the tolerable interference utilization for both the scenarios are found to be 52% equal to the theoretical targeted interference utilization%. Note that here, the theory and practical utilization % are same due to fact that, the interferer is Wi-Fi based system (with channel sense mechanism).

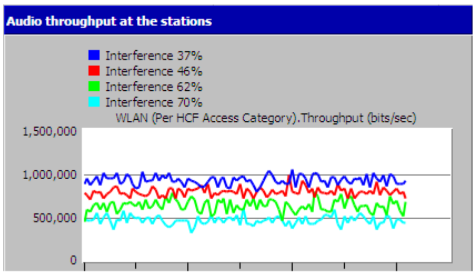
Fig. 4.22 shows the performance of the interference network. From the Fig. 4.22(b) one can see the throughput performance of the interfering network is very poor compared to the WHAP network and Fig. 4.22(a) shows the medium access delay for the audio traffic transmitted by the coexisting interference network. Note that, the interference network delay is higher than the WHAP stations because the interference network does not follow scheduling and have MAC level retransmissions.



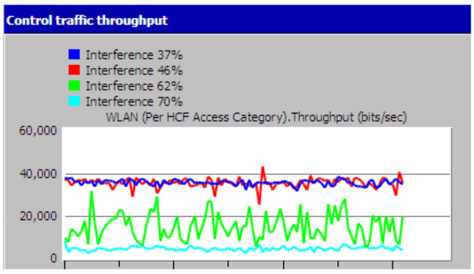
(a) Medium access delay of the control traffic



(b) Medium access delay of the audio traffic



(c) Audio throughput at each station



(d) Control traffic throughput

Figure 4.19: Performance of parameters with Wi-Fi CBR stations co-existing: Scenario(1)

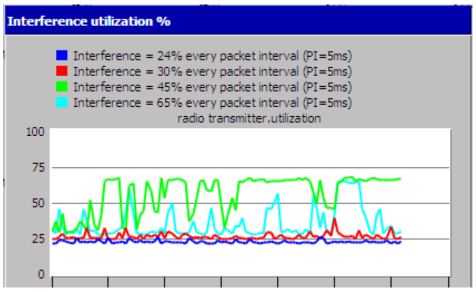


Figure 4.20: Wi-Fi interference utilization % for scenario(2)

4.5.5 Wi-Fi interference that are not synchronized to WHAP stations:

In Section 4.5.3, the Wi-Fi interference was configured such that, the packet interval of WHAP stations and the Wi-Fi interference are same. In this case, the packet interval for the Wi-Fi interference is not synchronized with the WHAP stations. Fig. 4.23 shows the utilization of the Wi-Fi interference with respect to its packet interval.

Figure 4.24 shows the WHAP parameters for non synchronized interference. The performance however is similar to that discussed in the synchronized interference.

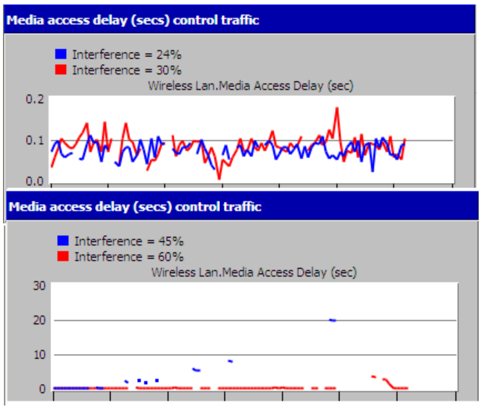
4.5.6 Interference type:Wi-Fi with Burst traffic

In this case, the audio conferencing WHAP system is simulated along with the Wi-Fi based overlapping BSS stations (Fig. 4.25) that are configured such that they transmit bursty traffic. Note that this type of interference shows more real-life interference scenario where the start time of the interferer transmission can be random. Here, the utilization factor shown in the Fig. 4.27(a) shows the transmission utilization of the WHAP BSS AP. This is because, effect of the burst traffic on the WHAP performance can be analysed.

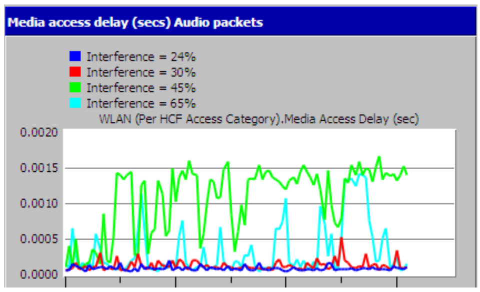
The Burst traffic configuration is varied as shown in the Figure 4.26 by which the 3 different scenario's (light, heavy and very heavy traffic) are configured and Figure 4.27 shows the performance of WHAP BSS in case of different utilization of interferer traffic.

Result Analysis:

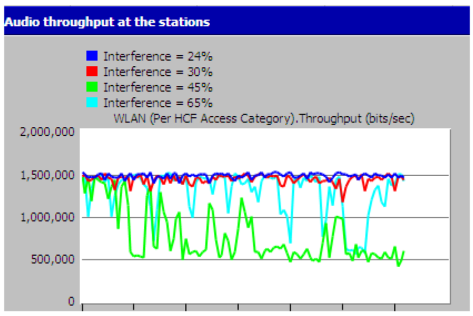
Since the interferer here is assumed to start its transmission after first 5 packet transmission of the active WHAP station (4 + 1 AP packet) in the WHAP BSS in Figure 4.27(a) we can see the variation in the utilization efficiency of the WHAP station indicating the start of the interference at the time of utilization drop for different interference types. Figure 4.27(b)



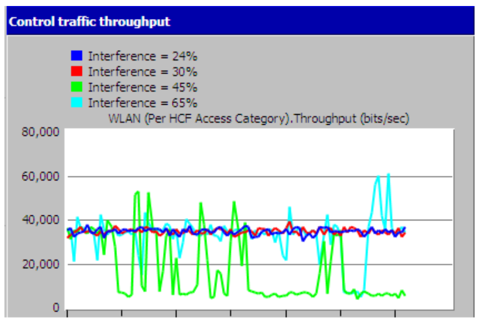
(a) Medium access delay for control traffic



(b) Medium access delay for audio traffic

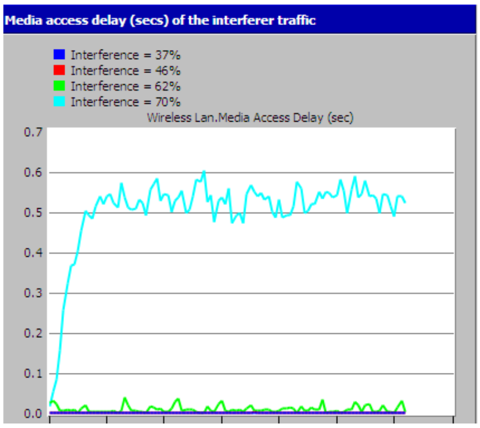


(c) Audio throughput at each station

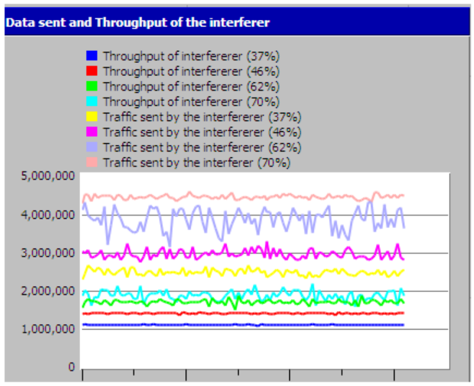


(d) Control traffic throughput

Figure 4.21: Performance of parameters with Wi-Fi CBR stations co-existing: Scenario(2)



(a) Medium access delay



(b) Throughput performance

Figure 4.22: Performance of the interference network

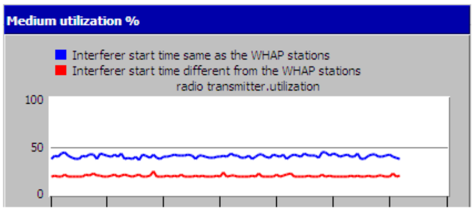
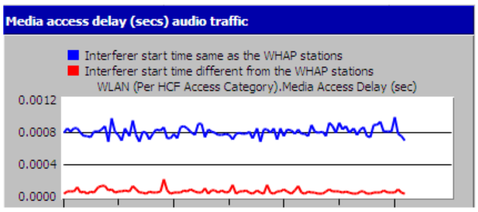
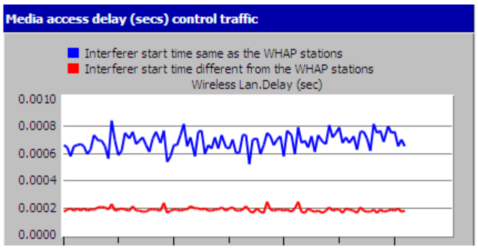


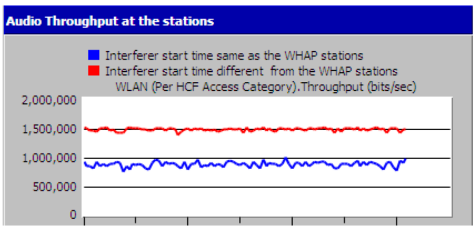
Figure 4.23: Wi-Fi interference utilization %(non-sync)

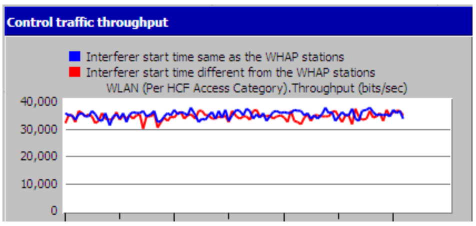
showing the respective medium access delay values of each of the active WHAP stations. From Figure 4.27(c) we can understand that in the simulated environment an heavy traffic interferer is tolerable (providing 100% throughput) and in case of control traffic, Figure 4.27(d) the throughput reduces to 50%. Due to the fact that, the interferer is an Wi-Fi based one, the theory and simulated utilization values are found similar as discussed in the previous sections.





(a) Medium access delay of the control traffic (b) Medium access delay of the audio traffic





(c) Audio throughput at each station (d) Control traffic throughput

Figure 4.24: Performance of WHAP parameters with non-sync Wi-Fi interference

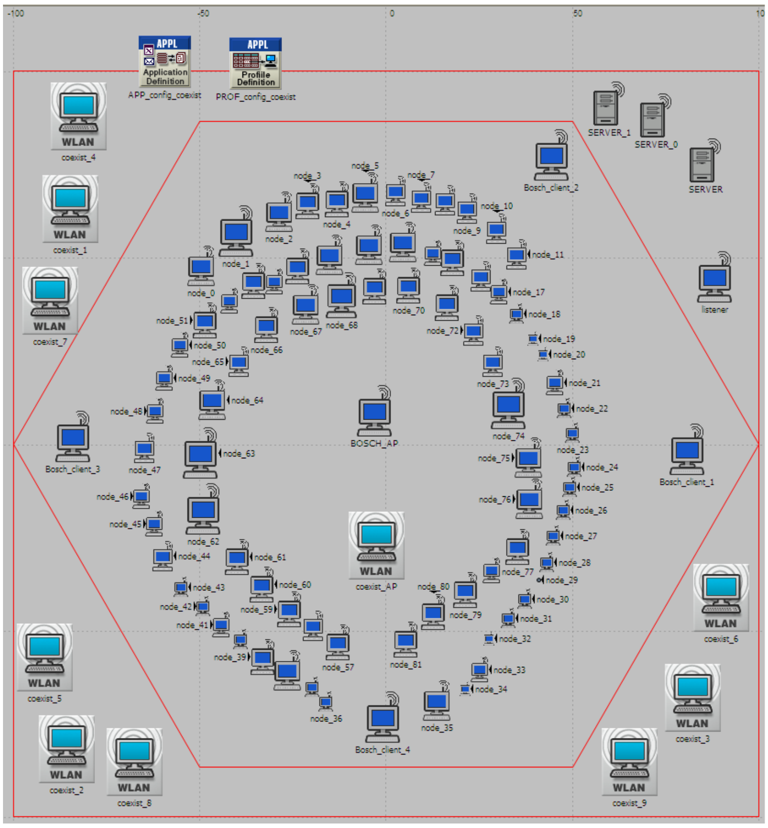


Figure 4.25: The whole audio conferencing system with WHAP enabled stations coexisting with Wi-Fi VBR interferer

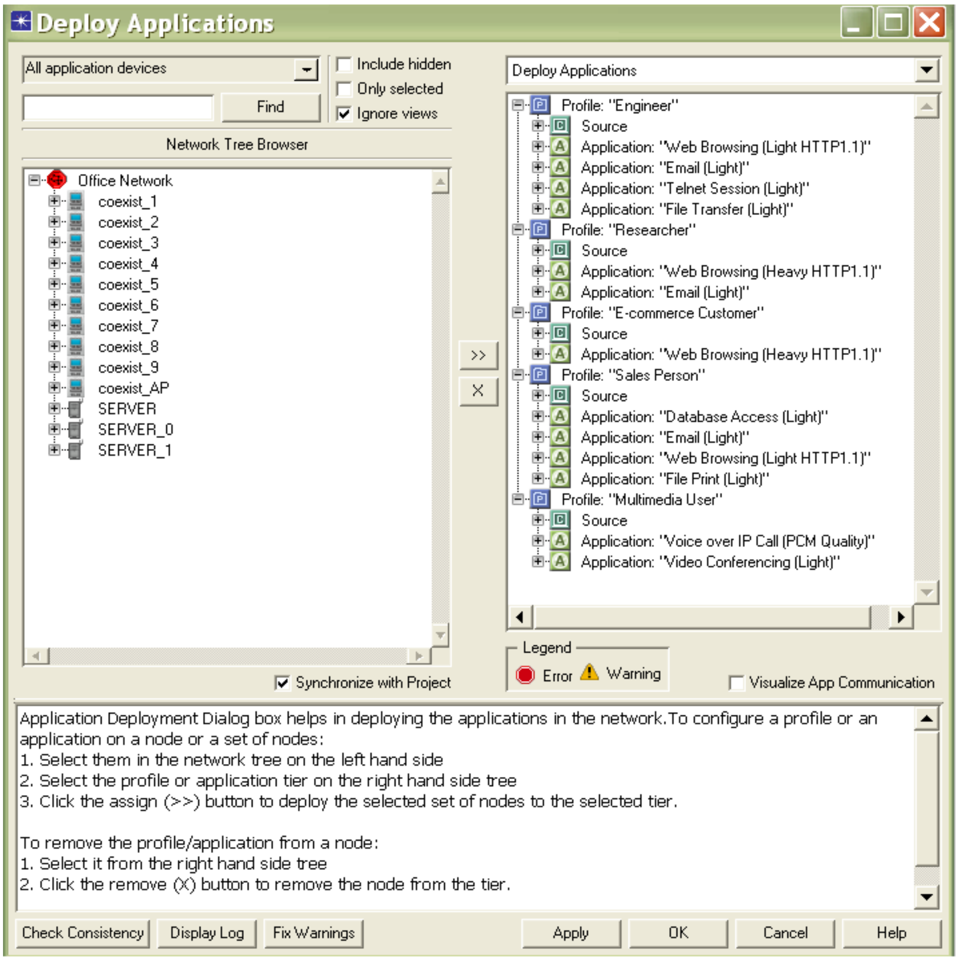
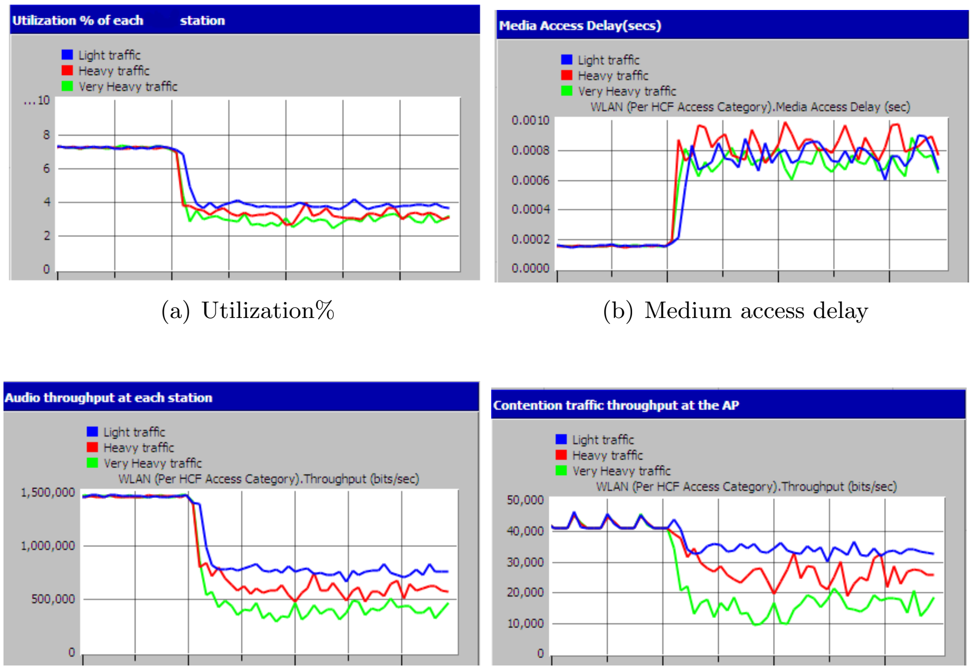


Figure 4.26: Wi-Fi based bursty interference configuration



(a) Utilization%

(b) Medium access delay

(c) Audio throughput at each station

(d) Control traffic throughput

Figure 4.27: Comparison of Performance of WHAP BSS at different intensities of interference%

5 Conclusion

The goal of this thesis was to optimize a audio conferencing system that communicate via the standard Wi-Fi network and to provide a maximum transmission delay of 5 ms (one way end to end delay of 10 ms) when co-existing with an interferer that use up to 50% of the channel bandwidth. First the QoS performance provided by the existing 5th generation Wi-Fi standard is analysed in detail. The performance provided by WMM feature in the standard did not satisfy the requirements of the audio conferencing system. This lead to designing a protocol at the application layer (WHAP) that provided scheduling and retransmission mechanism that made it possible to achieve the required performance. The WHAP based audio conferencing BSS was analysed under different interference scenarios both in theory and in simulated environment using network simulator OPNET. The results show the possibility to provide the required performance when co-existing with an interference of up to 50% as targeted. Finally, it can be concluded that the performance of the audio conferencing setup configured with WHAP enabled stations provide much better performance compared to the available state-of-art standard techniques.

Recommendation

In order to achieve the 5 ms delay performance using the standard Wi-Fi, I would recommend configuring the coexisting network such that they utilize not more than 50% of the medium. The parameters (data rate, packet size) that decide the medium utilization are discussed in this thesis. In the future work, the audio conference system with WHAP enabled stations must be setup in a practical environment. A more realistic performance analysis can be obtained since such an environment will include the packet loss due to physical factors like scattering, multipath, fading etc.

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