



Department of Electronic and Computer Engineering

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THESIS

**Impact of New WLAN P2P Technology on Existing Large Scale
WLAN Deployments**

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ABSTRACT

Wireless Local Area Networks (WLANs) are becoming the de facto last hop technology. Since its inception, the expected reliability and bandwidth offered by a WLAN has dramatically increased. This is evident by examining a sample of applications that run today over residential and commercial-grade WLAN networks such as VoIP and P2P applications, the introduction of which is a strenuous addition to an already congested radio frequency spectrum.

In this work, we design and deploy a large scale WLAN in the OPNET Modeler simulation environment. We perform frequency assignment and AP placement to satisfy VoIP Quality of Service (QoS) requirements with respect to performance considerations and then follow a microcell deployment strategy to provide thicker coverage. After the simulation framework is built, we introduce pairs of P2P links and observe their impact on the performance of the overlaid network.

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LIST OF ACRONYMS

WLAN	Wireless Local Area Network
P2P	Peer To Peer
BSS	Basic Service Set
AP	Access Point
DS	Distribution System
ESS	Extended Service Set
IBSS	Independent Basic Service Set
MAC	Medium Access Control
LLC	Logical Link Control
PDU	Protocol Data Unit
CSMA/CA	Carrier Sense Multiple Access with Collision Avoidance
DCF	Distributed Coordination Function
PCF	Point Coordination Function
DIFS	Distributed Inter Frame Space
SIFS	Short Inter Frame Space
PIFS	PCF Inter Frame Space
EIFS	Extended Inter Frame Space
RTS	Request To Send
CTS	Clear To Send
NAV	Network Allocation Vector
RSSI	Received Signal Strength Indicator
HCF	Hybrid coordination function
RTS/CTS	Request To Send / Clear-To-Send
RF	Radio Frequency
PS-Poll	Power Save – Poll Procedure
MIMO	Multiple Input Multiple Output
SDM	Space Division Multiplexing
FSM	Finite State Machine

MSDU	MAC Service Data Unit
MPDU	MAC Protocol Data Unit
VoWLAN	Voice Over WLAN
HA	High Availability
PSTN	Public Switched Telephone Network
DSP	Digital Signal Processor
ITU	International Telecommunication Union
ToS	Type of Service
HT-STA	High Throughput Station
HT-GF PLCP	High Throughput – Greenfield PLCP Header
DSCP	Differentiated Service Code Point

Chapter 1: Introduction to Wireless Local Area Networks

1.1. General Description

The principal application of Wireless Local Area Networks (WLAN) systems is to provide network connectivity to mobile and portable devices. Most modern WLANs are based on IEEE 802.11 standards, marketed under the Wi-Fi brand name. In the past, WLANs were mainly used to transport low-bandwidth, data-application traffic such as e-mail or web-browsing. Currently, with the expansion of WLANs into vertical (such as retail, finance, and education) and enterprise environments, there is a growing interest for effectively supporting a high density of users and a broad range of high-end applications, such as voice and video [2].

1.2. Benefits of WLANs

The emergence and continual growth of WLANs are being driven by the need to lower the costs associated with network infrastructures and to support mobile networking applications that offer gains in processing efficiency, accuracy, and lower business costs. Some of the benefits of WLANs are the following [5].

Mobility: Many employers require their employees to be mobile in an effort to increase efficiency. Inventory clerks, healthcare workers, policemen, and emergency care specialists are ideal candidates to benefit from wireless mobility.

Installation in Difficult-to-Wire Areas: The implementation of wireless networks offers many tangible cost savings when performing installations in difficult-to-wire areas.

Increased Reliability: A problem inherent to wired networks is downtime because of cable faults. The use of less cable in WLANs reduces the downtime of the network and the costs associated with replacing cables.

Reduced Installation Time: The deployment of wireless networks greatly reduces the need for cable installation, making the network available for use much sooner.

Long-Term Cost Savings: The advantage of wireless networking is again based on the lack of cable; you can move the network connection by just relocating an employee's PC or IP phone.

1.3. Types of WLANs

1.3.1. Infrastructure Mode

An infrastructure WLAN is what most companies, public hotspots, and homeowners implement. An infrastructure WLAN offers a means to extend a wired network. It is based on a cellular architecture where the system is subdivided into cells. Each cell, called **Basic Service Set (BSS)**, is controlled by a Base Station, called **Access Point (AP)**. Although a wireless LAN may be formed by a single cell, with a single Access Point, most installations will be formed by several cells, where the Access Points are connected through some kind of backbone called **Distribution System (DS)**. This backbone is typically Ethernet and can be wireless itself; it allows users to communicate with other wireless users, and with servers and network applications connecting to the distribution system. The whole interconnected Wireless LAN, including the different cells, their respective Access Points and the Distribution System, is seen as a single 802 network to the upper layers of the OSI model and is known in the Standard as **Extended Service Set (ESS)** [4].

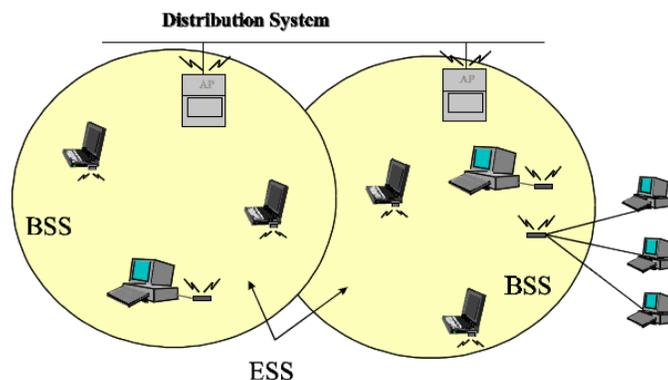


Figure 1: A Typical 802.11 WLAN

1.3.2. Ad Hoc Mode

Ad hoc or peer-to-peer WLANs only require 802.11 client radios in the client devices that connect to the network. Because there is no access point or WLAN controller and the stations are within range of each other, data transmitted by a particular source station travels directly to the applicable destination station. The rationale behind the ad hoc form of networking is to enable users to spontaneously set up WLANs. Access points are not necessary, which makes peer-to-peer networks easy to set up and take down. On the other hand, the absence of an access point in an ad hoc network means that an ad hoc WLAN must take on more of the MAC layer responsibilities. The 802.11 standard also refers to an ad hoc WLAN as an independent basic service set (IBSS) [5].

1.4. The 802.11 Standard

1.4.1. Introduction

WLANs provide functionality that corresponds to the physical layer (Layer 1) and the data link layer (Layer 2) of the OSI reference model. Figure 2 illustrates the logical architecture of 802.11. 802.11 WLANs define a portion of the data link layer – the MAC sublayer – which interacts with the IEEE 802.2 logical link control (LLC) layer. There are several different 802.11 physical layers, such as 802.11, 802.11b, 802.11g, and 802.11n. Some of the 802.11 physical layers, such as the original 802.11 direct-sequence and frequency-hopping spread spectrum, are not shown in Figure 2 [5].

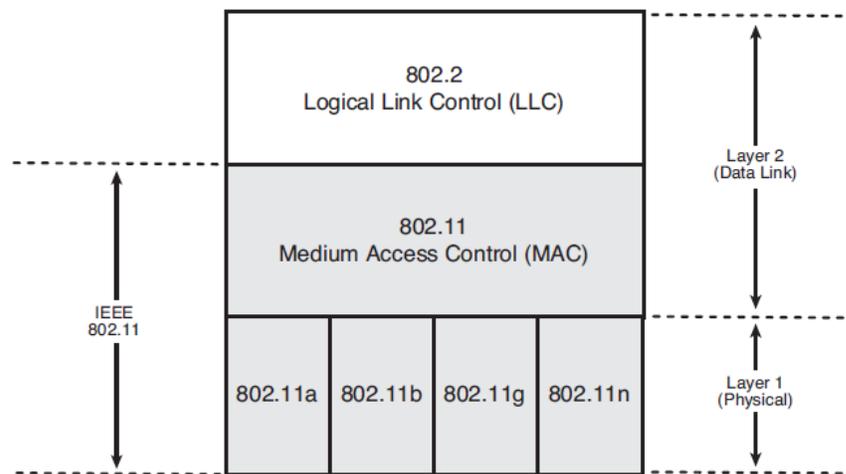


Figure 2: A Wireless LAN Provides Functions Related to Data Link and Physical Layers of a Network’s Architecture

1.4.2. IEEE 802.2 Logical Link Control

The Logical Link Control provides addressing and data link control, and is independent of the topology, transmission medium, and medium access control technique chosen. Higher layers, such as TCP/IP, pass user data down to the LLC expecting error-free transmission across the network. The LLC in turn appends a control header, creating an LLC protocol data unit (PDU). The LLC uses the control information in the operation of the LLC protocol. Before transmission, the LLC PDU is handed down through the MAC service access point to the MAC layer, which appends control information at the beginning and end of the packet, forming a MAC frame. The control information in the frame is needed for the operation of the MAC protocol [5].

1.4.3. IEEE 802.11 MAC Layer

The 802.11 MAC sublayer resides within the data link layer and enables multiple client devices to share the common air transmission medium via a carrier sense protocol. In general, the MAC layer manages and maintains communications between 802.11 stations (client radios and access points) by coordinating access to a shared radio channel and using protocols that enhance communications over a wireless medium. The MAC layer provides the following main functions [5]:

- **Data delivery:** The delivery of data between stations is the primary function of the MAC layer. This includes medium access, exchange of data frames, error recovery, frame aggregation, fragmentation, encryption, and multicasting.
- **Connectivity:** Before an 802.11 station can send and receive data, it must connect to the network. This includes scanning for available networks and authenticating and associating with the network.
- **Timing and synchronization:** Because of the sharing of the wireless medium, 802.11 stations must adhere to strict timing and synchronization rules.
- **Power management:** 802.11 power-save functions enable mobile 802.11 client devices to conserve power and lengthen battery longevity.

1.4.3.1. Access Methods

The MAC Layer defines different access methods, the Distributed Coordination Function, the Point Coordination Function and the Hybrid Coordination Function.

Distributed coordination function (DCF): The basic access mechanism, called the **Distributed Coordination Function**, uses a combination of both physical and virtual carrier-sense mechanisms to determine whether the medium is busy or idle. If both physical and virtual mechanisms indicate an idle medium, the station can transmit data. If not, the station must wait. Figure 3 illustrates the DCF process. The PHY layer provides a physical means of sensing the channel. The result of the physical channel assessment from the PHY layer is sent to the MAC layer as part of the information in deciding the status of the channel.

The MAC layer carries out the virtual carrier-sense protocol based on reservation information found in the Duration field of all frames. This information announces (to all other stations) a station's impending use of the medium. The MAC layer monitors the Duration field in all MAC frames and places this information in the station's NAV (network allocation vector) if the value is greater than the current NAV value. The NAV operates like a timer, starting with a value equal to the Duration field value of the last frame

transmission sensed on the medium and counting down to zero. As a condition to accessing the medium, the MAC layer checks the value of its NAV. If the NAV equals zero and the PHY layer indicates a clear channel (that is, received signal strength indication (RSSI) is below a specific threshold), the station can transmit a frame. Just before sending a frame, a station calculates the amount of time necessary to send the frame based on the frame's length and data rate. The station places a value representing this time in the Duration field in the header of the frame. This process reserves the medium for the sending station because the Duration field causes the MAC layers in other stations to hold off transmissions until the sending station is done sending its frame.

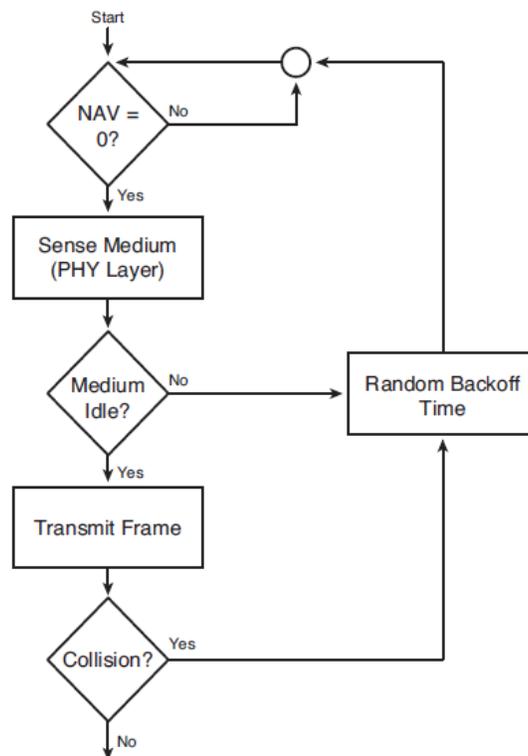


Figure 3: Flowchart Illustrating the Operation of the Distributed Coordination Function

An important aspect of the DCF is a random back off that a station must wait if it detects a busy medium. If the channel is in use, the station must wait a random period of time before attempting to access the medium. Thus, if the PHY layer indicates that the medium is not clear (that is, RSSI threshold is above a specific threshold), the MAC layer implements a backoff algorithm, regardless of the status of the NAV. This avoids the probability of collisions among stations waiting to transmit. The period of time immediately following a busy medium is when the highest probability of collisions occurs, especially under high utilization. Multiple stations may be waiting for the medium to become idle and will attempt to

transmit at the same time. Once the medium is idle, a random backoff time defers each station's transmission for a different amount of time, minimizing the probability that stations will collide.

In addition to implementing a backoff for busy medium indications, stations use the same backoff mechanism when retransmitting frames due to transmission errors. Under low utilization, stations are not forced to wait very long before transmitting their frame. If the utilization of the network is high, the protocol holds stations back for longer period of times to avoid the probability of multiple simultaneous transmissions [5].

Point coordination function (PCF): The PCF access method is part of the original 802.11 standard, but it implements a polling function that determines when a specific client station can transmit. As a result, PCF provides synchronous (predictable) access to the medium, which supports time-bounded delivery of information. Very few if any vendors implement PCF [5].

Hybrid coordination function (HCF): The HCF access method, which is based on PCF, provides time-bounded delivery of information [5].

1.4.3.2. MAC Frame Types

The MAC layer uses a variety of frame types, each having a particular purpose. The IEEE 802.11 specification divides MAC frames into three broad categories [4].

- **Data Frames:** Which are used for data transmission
- **Control Frames:** Which are used to control access to the medium (e.g. RTS, CTS, and ACK)
- **Management Frames:** Which are frames that are transmitted in the same manner as data frames to exchange management information, but are not forwarded to upper layers (e.g. beacon frames).

Each frame type is subdivided into different Subtypes, according to their specific function.

1.4.3.3. RTS/CTS

An optional feature, which reduces the probability of two stations colliding because they cannot hear each other, is the RTS/CTS (request-to-send / clear-to-send) function. If you enable RTS/CTS on a particular station, it will refrain from sending an 802.11 data frame until the station completes a RTS/CTS handshake with another station.

A station wanting to transmit a packet first transmits a short control packet called Request To Send (**RTS**), which includes the source, destination, and the duration of the following transaction (i.e. the packet and the respective **ACK**). The destination station responds, if the medium is free, with a response control

Packet called Clear to Send (**CTS**), which includes the same duration information. All stations receiving either the RTS and/or the CTS set their **Virtual Carrier Sense** indicator for the given duration.

This mechanism reduces the probability of a collision on the receiver area by a station that is “hidden” from the transmitter to the short duration of the RTS transmission because the station hears the CTS and “reserves” the medium as busy until the end of the transaction. The duration information on the RTS also protects the transmitter area from collisions during the ACK (from stations that are out of range of the acknowledging station).

It should also be noted that, due to the fact that the RTS and CTS are short frames, the mechanism also reduces the overhead of collisions, since these are recognized faster than if the whole packet was to be transmitted. This is true if the packet is significantly bigger than the RTS, so the standard allows for short packets to be transmitted without the RTS/CTS transaction. This is controlled per station by a parameter called **RTS Threshold** [4].

1.4.3.4. Connectivity

A client radio must first connect to an 802.11 network prior to sending any data, thus it needs to determine which 802.11 networks are within range. In order to do this a scanning function is implemented either in passive or active mode.

Passive scanning: The client station tunes to each radio frequency (RF) channel, listens for a period of time, and monitors 802.11 beacon frames coming from access point mode in infrastructure or other client stations when in ad-hoc mode. The client station records the signal strength of the beacon frame and continues the scan of other channels. After scanning each of the RF channels, the client station makes a decision on which access point to associate with. In general, the client station chooses the access point that had sent the beacon having the strongest signal. Some client radio vendors may include other parameters, such as noise levels and utilization, when making this decision.

Active scanning: Active scanning involves the client station sending broadcast 802.11 Probe request frames on each RF channel (or subset of channels). Access points (infrastructure networks) and client stations (ad hoc networks) within range will respond to the probe request frame by sending a probe response frame. The station sending the probes will wait on each RF channel for a period of time to receive the corresponding probe response frames coming from. Active scanning enables client stations to receive information about nearby access points in a timely manner, without waiting for beacons; however, active scanning generates additional traffic on the network (that is, probe and response frames). Similar to

passive scanning, the radio card uses the signal strength and possibly other information corresponding to the probe response frame to make decisions on which access point to associate with.

1.4.3.5. The Authentication Process

Once the station has located an Access Point, and decides to join its BSS, it goes through the authentication process. This is the interchange of information between the AP and the station, where each side proves the knowledge of a given password [4].

1.4.3.6. The Association Process

After authentication is successful, the client station must associate with the access point to complete the connection process. Association is necessary to synchronize the client station and access point with important information, such as beacon interval and supported data rates. A station is capable of transmitting and receiving data frames only after the association process is completed [5].

1.4.3.7. Timing and Synchronization

The 802.11 specification defines several spacing intervals that govern when a station can transmit certain types of frames and provide different priorities. Furthermore, stations need to keep synchronization, which is necessary for keeping hopping synchronized and other functions like Power Saving.

1.4.3.8. Inter Frame Spaces (IFS)

SIFS: The Short IFS is the shortest of the interframe spaces, providing the highest priority level by allowing some frames to access the medium before others. The following frames use the SIFS interval: ACK frame, CTS frame, the second or subsequent MAC Service Data Unit (MSDU) of a fragment burst.

PIFS: The PCF is the interval that stations operating under the point coordination function use to gain access to the medium. This provides priority over frames sent by the distributed coordination function.

DIFS: All stations operating according to the distributed coordination function use the DCF IFS interval for transmitting data frames and management frames.

EIFS: All DCF-based stations use the Extended IFS (EIFS) rather than DIFS to defer sending a frame if a transmission results in a bad reception of a frame. This interval provides enough time for the receiving station to send an ACK frame. The EIFS helps prevent transmitting over the ACK frame sent by hidden nodes because the node cannot decode the Duration field and set the NAV properly [5].

1.4.3.9. Keeping Synchronization

On an infrastructure BSS, synchronization is achieved by all the stations updating their clocks according to the AP's clock, using the following mechanism: The AP periodically transmits frames called **Beacon Frames**. These frames contain the value of the AP's clock at the moment of transmission (note that this is the moment when transmission actually occurs, and not when it is put in the queue for transmission. Since the Beacon Frame is transmitted using CSMA rules, transmission may be delayed significantly). The receiving stations check the value of their clocks at the moment the signal is received, and correct it to keep in synchronization with the AP's clock. This prevents clock drifting which could cause loss of synch after a few hours of operation.

1.4.3.10. Power Management

The 802.11 standard defines procedures that can be used to implement power management during inactive periods [1].

Wakeup Procedure: There are two reasons for the station to wake up: to transmit pending data or to retrieve buffered data from an AP. Waking up to transmit data is a straightforward operation, driven by the station. The decision to wake up and receive data is also made by the station after monitoring its pending data bit in a periodic beacon frame sent out by its AP. Following such a transmission, the station remains active so the AP can send any buffered downlink frames afterward.

Sleep Procedure: A STA in the active mode needs to complete a successful station initiated frame exchange sequence with the PS (Power Save) bit set to sleep to transition into the sleep mode. Following this operation, the AP buffers all the downlink frames to this station.

Power Save – Poll Procedure (PS-Poll): Instead of waiting for the AP to transmit the buffered downlink frames, a station in sleep mode can solicit an immediate delivery from its AP by using a PS – Poll frame. Upon receiving this PS-Poll, the AP can immediately send one buffered downlink frame (immediate data response) or simply send an acknowledgement message and response with a buffered data frame later (delayed data response). For the immediate data response case, the station can stay in the sleep state after finishing this frame exchange since there is no need for the station to transition to the active state given that the AP can only send a buffered downlink frame in response to the PS-poll from the station. For the delayed data response case, the station has to transition to the active state until receiving a downlink frame from the AP.

1.4.4. IEEE 802.11 Physical Layer

The physical (PHY) layer provides for the transmission of bits through a communications channel. This includes all specifications needed for defining the physical composition of the signals that are sent through the wireless network. For example, the 802.11 PHY layer defines modulation types, frequencies, and signal synchronization procedures. The 802.11 standard includes a different PHY layer specification for each of the WLAN types (that is, 802.11b, 802.11g, and 802.11n). All these PHY layers share common MAC sublayer functions [5].

1.4.5. Related Amendments

The 802.11 family consists of a series of over – the – air modulation techniques that use the same basic protocol. The most popular are those defined by the 802.11b and 802.11g protocols, which are amendments to the original standard. Other standards in the family (c–f, h, j) are service amendments and extensions or corrections to the previous specifications. In the following table, the most popular amendments and their characteristics are presented [3].

802.11 Amendments						
802.11 Amendment	Release	Band (GHz)	Bandwidth (MHz)	Data rate per stream (Mbps)	Allowable MIMO streams	Modulation
-	1997	2.4	20	1, 2	1	DSSS FHSS
a	1999	5	20	6, 9, 12, 18, 24, 36, 48, 54	1	OFDM
b	1999	2.4	20	5.5, 11	1	DSSS
g	2003	2.4	20	6, 9, 12, 18, 24, 36, 48, 54	1	DSSS FHSS
n	2009	2.4/5	20 40	7.2, 14.4, 21.7, 28.9, 43.3, 57.8, 65, 72.2 15, 30, 45, 60, 90, 120, 135, 150	4	OFDM

Table 1: 802.11 Amendments

1.4.6. IEEE 802.11n

802.11n, the amendment that is used in our experiments, is an enhancement to the IEEE 802.11 wireless network standard that includes many new features to increase transmission speeds, range and reliability compared to 802.11a/b/g. The enhancements translate to 300 Mbps of raw data throughput and/or double the range compared to current 802.11a/g technology.

802.11n can operate on the 2.4 GHz and/or 5 GHz bands and it provides backwards compatibility by supporting simultaneous 802.11a and 802.11n (802.11n/a) at 5 GHz and/or simultaneous 802.11b, 802.11g and 802.11n (802.11n/b/g) at 2.4 GHz. However, in Greenfield deployments an 802.11n AP can be configured to not be backward compatible with 802.11a or 802.11b/g. For simultaneous 2.4 GHz and 5 GHz support, two radios must be available on the AP. The 802.11n standard makes use of several innovative technologies to improve the range, throughput, and reliability of wireless LANs [18].

The three primary innovations are [6],[7]:

Multiple Input Multiple Output (MIMO) technology: The heart of the new 802.11n standard is a technique called MIMO, or Multiple Input - Multiple Output. On the transmission side, MIMO uses Space Division Multiplexing (SDM) to transmit multiple data streams on the same frequency but over different spatial channels, increasing the transmission power and data that can be sent over the air. MIMO on the receiving end allows multiple signals to be combined, increasing signal strength and eliminating the effects of multipath fading. While previous wireless technologies had problems dealing with multipath signal reflections and attenuation, MIMO actually uses these reflections to increase the range and reliability over the wireless coverage area.

Packet Aggregation: Packet aggregation increases efficiency by aggregating multiple packets of application data into a single transmission frame. 802.11n networks can send multiple data packets with the fixed overhead cost of just a single frame. Packet aggregation is beneficial for data applications such as file transfers. However, real-time applications (e.g. voice) do not benefit from packet aggregation. For voice, it is better to minimize the number of “packed” packets to reduce latency and eliminate jitter contentions.

Channel Bonding (40 MHz Channels): Normal 802.11a/g only supports 20 MHz channels that can carry a maximum of 54 Mbps of raw data per channel. 802.11n increases that throughput to 150 Mbps per channel, but also employs a technique called channel bonding to combine two adjacent 20 MHz channels

into a single 40 MHz channel. The technique more than doubles the channel bandwidth to 300 Mbps throughput. Channel bonding is most effective in the 5 GHz frequency given the far greater number of available channels, while the 2.4 GHz frequency has only three non-overlapping 20 MHz channels. Therefore, bonding two 20 MHz channels uses two thirds of the total frequency capacity in the 2.4 GHz spectrum.

1.5. WLAN Implications

When designing and supporting a WLAN, potential implications may occur. Some of them are security vulnerabilities, radio signal interference, impacts of multipath propagation, roaming issues, battery limitations and installation issues.

The major potential implication that we are interested in is the radio signal interference, as the introduction of P2P applications on a WLAN is expected to be a strenuous addition to an already congested radio frequency spectrum that may result in serious degradation of the performance of the overlaid WLAN. [5]

1.5.1. Radio Signal Interference

Radio signal interference involves the presence of unwanted, interfering radio signals that disrupt normal WLAN operations. Radio signal interference does not abide by the 802.11 protocols, so the interfering signal might start abruptly while a legitimate 802.11 station is in the process of transmitting a packet. If this occurs, the destination station will receive the packet with errors and not reply to the source station with an acknowledgement. In return, the source station will attempt retransmitting the packet, adding overhead on the network. Radio signal interference causes wireless clients and access points to hold off transmitting, which causes delay and lower throughput. This resulting decrease in performance can make browsing websites and downloading files sluggish and most importantly it can severely limit the number of active voice users. Also, in cases where interfering signals are strong enough the wireless clients might not be able to access the WLAN at all for an indefinite period of time.

There are various sources of radio signal interference. Some of them are:

- **Cordless Phone Interference**
- **Microwave Oven Interference**
- **Bluetooth Interference**
- **Neighboring Wireless LAN Interference**

With the growing proliferation of WLANs, it is likely that neighboring wireless networks will exist in the area where an organization installs the WLAN. In most cases, the neighboring networks will implement a version of 802.11 (802.11b, 802.11g, or 802.11n) that is compatible with a user's network. The advantage of this is that both the user's network and the neighboring one coordinate access to the common medium. Interference between different WLANs occurs when the access points within range of each other are set to the same RF channel. As a result, a radio cell from one network overlaps with a radio cell of the other network. In this case, with no data traffic occurring within each of the overlapping cells, there is insignificant interference. The only interference is resulting from the periodic transmission of 802.11 access point beacons and other management frames. When client devices in both cells begin to actively send and receive data over the WLAN, the client devices will share access of the medium based on the 802.11 medium access control. They will share the capacity of the RF spectrum, which reduces the potential performance that the networks can deliver. Because the interactivity between the client devices does not result in a significant increase in retransmissions similar to interference from other sources such as microwave ovens (because the client devices are sharing the medium politely), the performance degradation might not be noticeable as compared to a client device operating from one of the cells while the neighboring cell has no activity. Thus, interference from neighboring networks is only a problem as utilization increases.

Chapter 2: Designing a WLAN Deployment

2.1. Performance Considerations

When designing a WLAN with requirements such as higher data rates, support for a high density of users and bandwidth – demanding applications, several factors should be taken into consideration in order to achieve the expected performance. These are [10]:

2.1.1 Range and Performance

Range and performance are indirectly proportional. By increasing design elements like transmit frequency or data rate, the performance will increase but range will decrease if all the other design parameters and environment are constant.

The AP client radius can be extended further by supporting lower bit rates, but this is not recommended for the following reasons:

- Lowering the bit rate extends the AP client radius, but also increases the client co-channel interference radius.
- The lower bit rates reduce the overall call cell capacity, as lower bit rate packets consume more time, and transmit fewer packets.

In other cases, changing design elements, such as transmit power and antenna gain, will increase performance and range [4], [10].

2.1.2 Throughput and Data Rate

The data rate of a signal is based on the time it takes to send information and overhead data bits when transmitting. Therefore, the aggregate data rate (throughput) is actually much lower because of delays between transmissions. Data rate mostly affects the delay performance of a WLAN. The higher the data rate, the lower the delay when sending data from one point to another. As a result, higher data rates can increase the capability to support a larger number of users. Data rate alone is not a good measure for the performance of the system, though. Actual throughput is a much better indicator of the performance of a WLAN because it provides an indication of the time it takes to send information. Throughput is the flow of information over time. It is important to not confuse throughput with 802.11 data rate, which is the

speed that the data bits in individual 802.11 data frames are sent. Because there is idle time between 802.11 data frames and retransmission due to noise, the throughput is always less than the data rate. Throughput, however, provides a more accurate representation of the delays that users experience. The reason is that it is possible to have a very high 802.11 data rate and still have throughput that is relatively low. This can occur when the user is close to an access point but there are many other users actively accessing the WLAN from the same access point or if there is substantial radio signal interference present. In these cases, the WLAN carries the information at a much slower rate [10].

2.1.3 Transmit Power Settings

For a constant range, increasing the transmit power of an 802.11 radio increases performance in the outward direction. As the transmit power increases, communications at a particular location will be possible at greater data rates. The basis for this is that increasing transmit power improves the SNR at a particular location, which allows the receiving radio to decode signals at higher data rates. This increase in data rate applies to the communications in only one direction, which is the outward path relative to the radio with increased transmit power [10].

2.1.4 Signal Coverage

As the basis for providing good performance, it is important to have adequate signal coverage throughout the required coverage areas. In areas that have weak signal coverage, the signal level and corresponding SNR will be relatively low. The 802.11 radios might still be able to decode the signals and successfully communicate, but the data rate may be fairly low.

2.1.5 Radio Signal Interference

By reducing radio signal interference, it is possible to increase performance at specific points within the coverage area. A reduction of noise levels will raise the SNR at points throughout the coverage area. This enables the access points and client radios to successfully decode 802.11 signals at higher data rates. [10]

2.2 WLAN Radio Frequency Planning

A total of 14 channels are defined in the IEEE 802.11b channel set. Each channel is 22 MHz wide, but the channel separation is only 5 MHz. This leads to channel overlap such that signals from neighboring channels can interfere with each other. In a 14-channel DS system there are only three non-overlapping and thus, non-interfering channels: 1, 6, and 11—each with 25 MHz of separation. This channel spacing governs the use and allocation of channels in a multi-AP environment, such as an office or campus. APs are usually deployed in a cellular fashion within an enterprise, where adjacent APs are allocated non-overlapping channels.

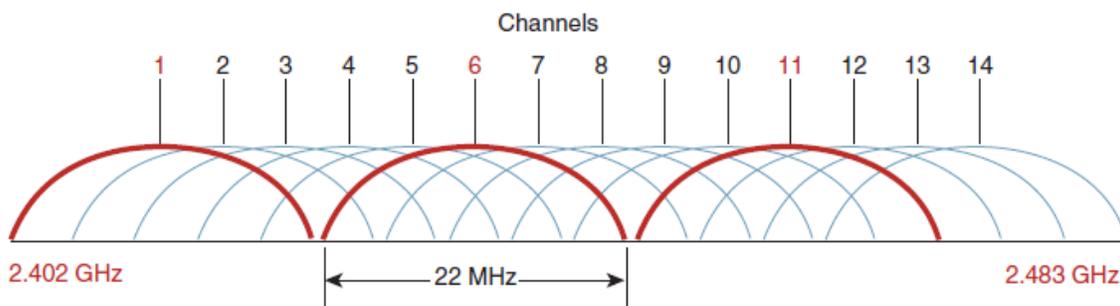


Figure 4: 2.4GHz Channel Allocations

This presents a challenge when trying to deploy APs, and ensure that an AP on the same channel cannot see the signal from an AP using the same channel. As mentioned before, the AP coverage radius changes with the client bit rates supported, and the boundary created by this radius is often considered the AP's boundary. The reality is somewhat more complicated because the AP influences the WLAN RF environment around it for a much greater distance than just the bit-rate boundary [10].

2.2.1 Voice over WLAN

An important factor that greatly affects the RF planning, design and implementation of a WLAN is the applications which are going to be accommodated. The issue of capacity planning is critical when supporting a voice network with high Quality of Service (QoS). In order to meet the delay and jitter requirements of VoIP applications and the increased mobility of Voice Over WLAN (VoWLAN), handset users place demands on connection quality and coverage that are beyond those of a typical WLAN data deployment. One area unique to the VoWLAN availability is RF coverage High Availability (HA)—providing RF coverage that is not dependent upon a single WLAN radio. The primary mechanism for providing RF HA is cell boundary overlap.

2.2.2 20 Percent Cell Overlap

The recommended AP cell overlap for VoWLAN deployments at 2.4GHz, by the Cisco VoWLAN Design Guide, is 20 percent. The purpose of the 20 percent overlap is to ensure that a VoWLAN handset can detect and connect to alternative APs, when it is close to the cell boundary, to provide thicker coverage and guarantee high rates even at the periphery of the coverage of the WLAN APs. This should allow a VoWLAN client to change AP associations with a minimum of interruption to a call, by minimizing the amount of data rate shifting and retransmission at a cell boundary for a given VoWLAN client.

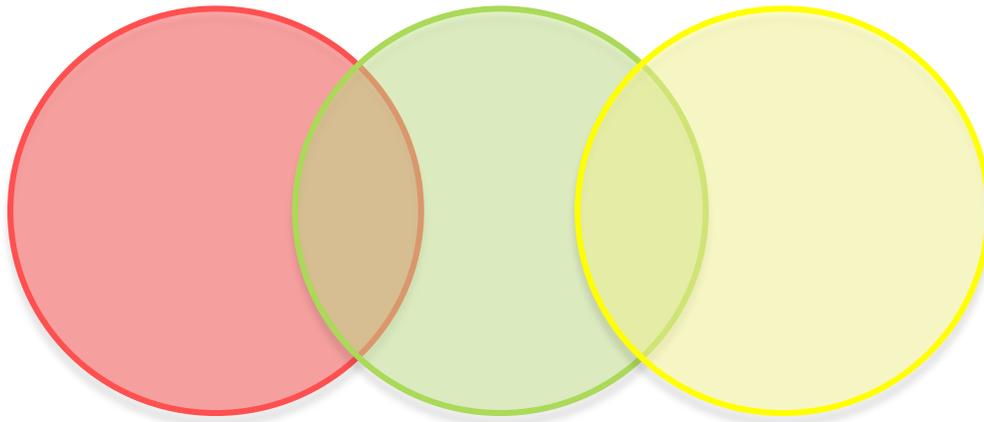


Figure 5: APs with 20 Percent Overlap

It is not an effective strategy to reduce the overlap in order to reduce co-channel interference, as users satisfaction can be greatly affected by poor roaming performance. In contrast, call capacity can be addressed by proper planning and design. [10]

2.2.3 Communication Distance

The maximum communication distance between two WLAN nodes is a function of three parameters: the transmission power of the sending node, the path-loss propagation model, and the reception power threshold (receiver sensitivity) of the receiving node. Based on the configured values of these parameters, we can model WLAN networks in which the communication distance is more than 300 meters. The IEEE 802.11 standard limits the distance between WLAN nodes to 300 meters and it can reach 400 meters for the IEEE 802.11n amendment [13].

The available transmit power settings in common Wi-Fi devices ranges from 10 mW to 100 mW while the maximum power setting will vary according to individual country regulations. The receiver sensitivity depends on the 802.11 amendment, the data rate and the MIMO mode and varies from -94dBm to -45.3 dBm (0.0000000004 mW to 0.0000295121 mW) [11],[12].

2.3 WLAN Deployment

The AP layout within a building depends greatly upon the construction and shape, as well as the WLAN coverage requirements. For our deployment we assume that the environment being modeled is an open rectangular area. We want to cover this area using the default transmission power of an AP in the simulation environment which is 0.005 Watt or 6.98 dBm. A typical access point layout begins by determining a minimum data rate that will be required throughout the area to be covered. We assume that the access points to be placed provide 52 Mbps in all locations. Therefore we set up a single cell with only one client and one AP, we configure their data rate to 52 Mbps and the power to 0.005W in order to measure the radius of the cell. This radius, which is the maximum distance from the AP that the client can transmit and receive traffic at 52 Mbps, is 372m.

We should note that no limits are imposed upon the value of the transmit power of the stations and APs in OPNET. The other important parameter that determines the radius of the cell, the received power threshold (receiver sensitivity) value of the radio receiver for arriving WLAN packets, is also set to the default value 0.0000000003 mW or -95 dBm. Packets with a power less than the threshold are not sensed and decoded by the receiver.

2.3.1. Access Point Placement – 9 APs

The 20 percent overlap requirement means that APs are spaced closer together than the (2x372) meters distance suggested by the cell boundary. If d is the distance between the centers of two circles of radius 1 and we solve for an area of 20 percent overlap between them, we get a d value of 1.374 or 511 meters between APs for our 0.005 Watt boundary.

The area of overlap between two circles for a standard radius of 1 is given by [17].

Using 9 APs we can provide complete coverage for an area of 1437 x 1045 m². We deploy the APs in a cellular fashion, where adjacent APs are allocated on non-overlapping channels. The deployment is presented in Figure 6 where the colors, red, green, and yellow represent channels 1, 6 and 11 respectively.

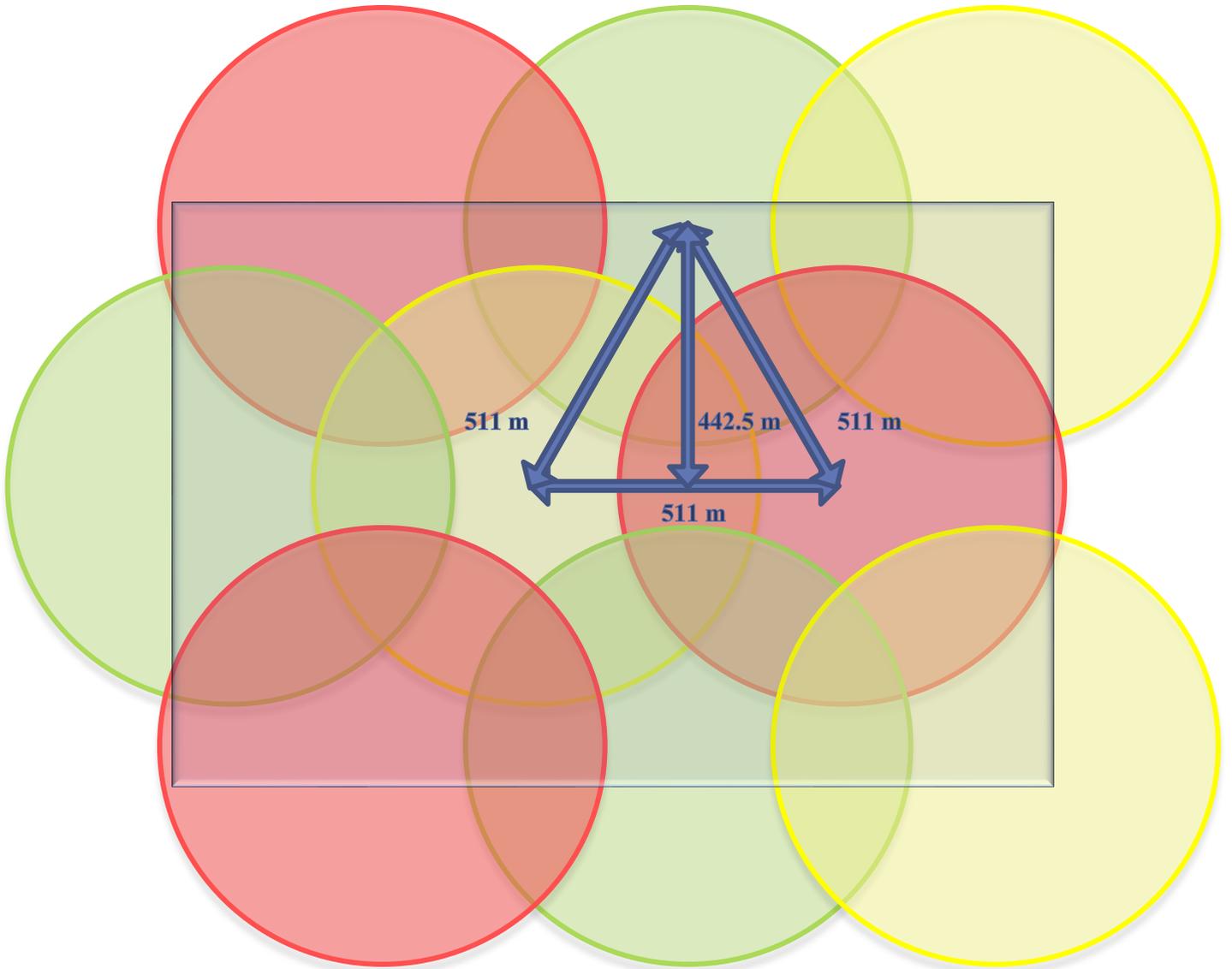


Figure 6: 9 AP Placement

Radius: 372 m

Length: $(0.9 * 372) + 511 + 511 + 0.213 * 372 = 1437$ m

Height: $(0.2 * 372) + 442.5 + 442.5 + 0.2 * 372 = 1045$ m

The exact BSS IDs, channels and coordinates of the APs used in our simulation are:

BSS ID	Channel	X – Coordinate	Y – Coordinate
1	1	0	0
2	6	511	0
3	11	1022	0
4	6	-255	-442.5
5	11	255	-442.5
6	1	766	-442.5
7	1	0	-885
8	6	511	-885
9	11	1022	-885

Table 2: Coordinates, BSS IDs and Channel Configuration of the APs Used in the 9 AP WLAN Deployment

For the sake of comparison we also present in Figure7 how we could cover the exact same area with only 6 APs. Here, the distance between access points on the same channel is even smaller, increasing the probability of co-channel interference.

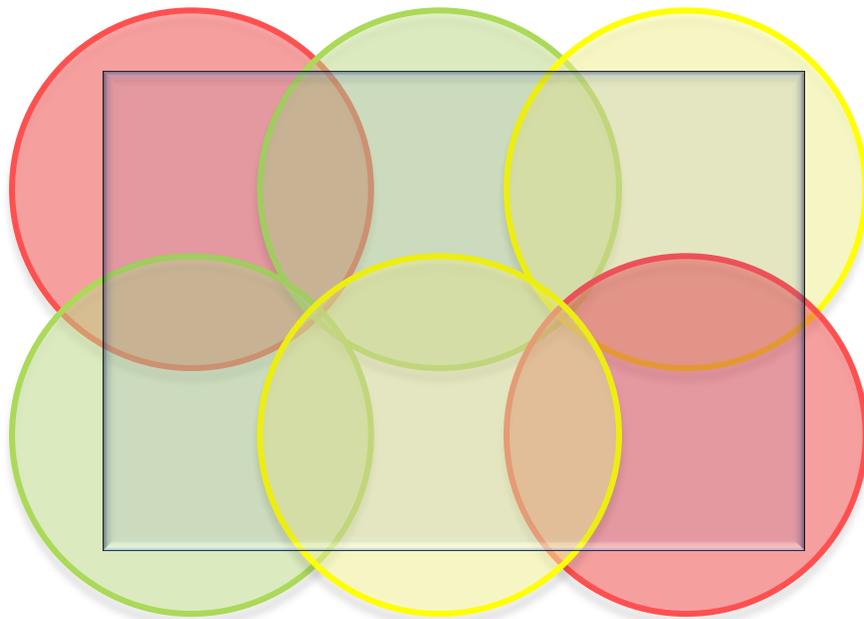


Figure 7: Alternative 9 AP Placement

2.3.2 Access Point Placement – 16 APs

2.3.2.1 The Microcell Deployment Strategy

In order to improve the performance of our WLAN deployment we consider using a high density of access points. This calls for turning down the transmit power of all access points and client radios, to force access points to be much closer together and avoid interfering with each other. This microcell architecture makes the physical area collision domains smaller than with conventional access point density. The microcell architecture leads to higher performance because there are fewer client devices connecting to each access point. This allows each client device radio to consume a greater percentage of the access point's capacity and avoid collisions with other radios. Therefore, performance can be significantly better.

An issue with using higher access point density to improve performance is that it leads to a great number of access points. Of course this means that the deployment will cost more.

In this deployment we want to cover the same rectangular area with 16 APs. Therefore we will use four APs in each row and the deployment will have four rows. The 20 percent overlap requirement means that the APs now should have a 378 meters distance from each other to cover the same length, thus the cell radius now should be 275 meters. Again, we assume that the access points to be placed provide 52 Mbps in all locations. Therefore we set up a single cell with only one client and one AP, we configure their data rate to 52 Mbps and we reduce the power of the AP until the radius of the cell reduces to 275 meters. This power is 0.003 W or 4.8 dBm.

Using 16 APs, we can provide complete coverage for a maximum area of 1439 x 1091 m². We should point out that in the 9 AP WLAN deployment we could cover an area of 1437 x 1045 m², so the same area can be covered. In our 16 AP deployment we also place the APs in a cellular fashion, where adjacent APs are allocated on non-overlapping channels. The deployment is presented in Figure 8 where the colors, red, green, and yellow represent channels 1, 6 and 11 respectively.

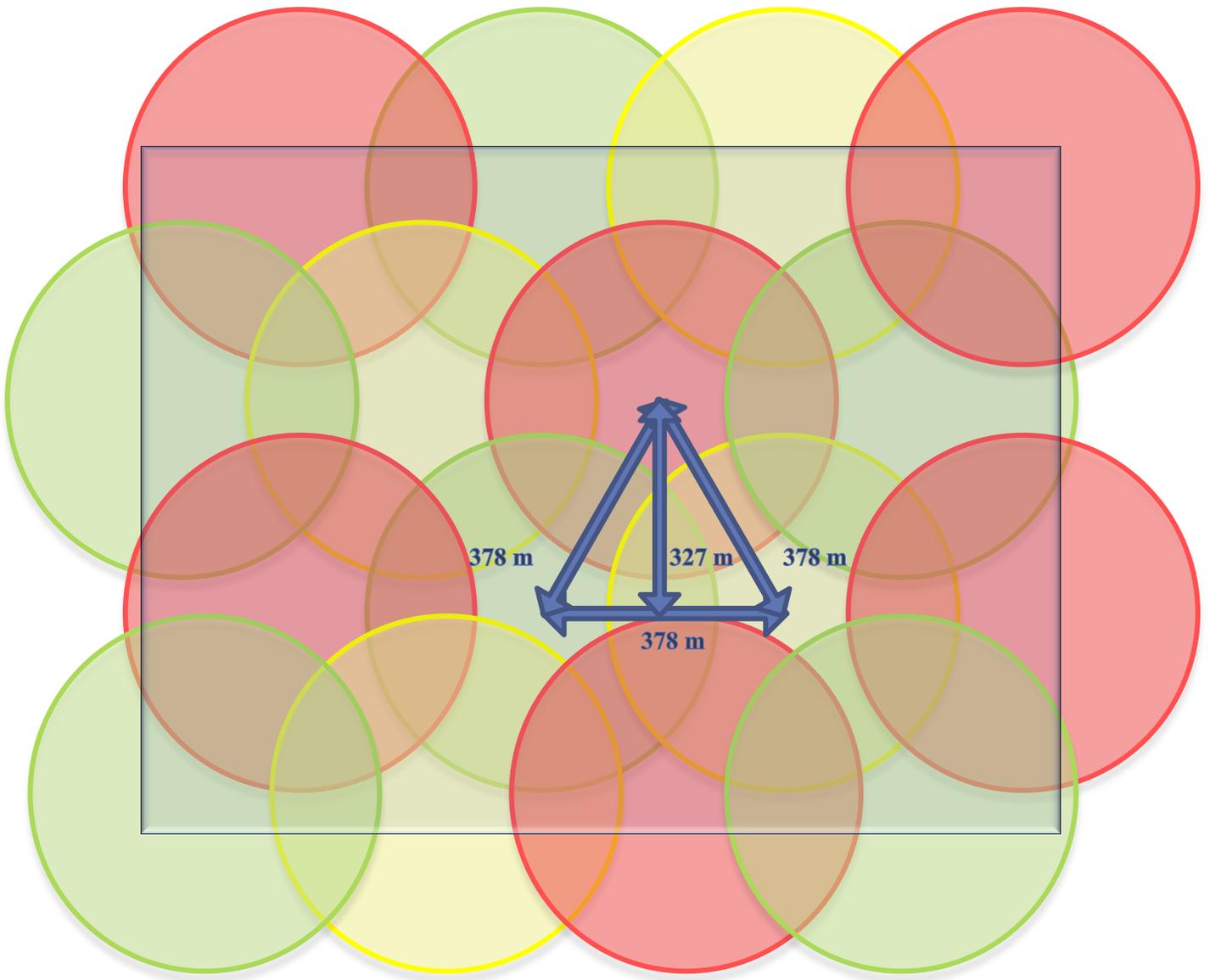


Figure 8: 16 AP Placement

Radius: 275 m

Length: $(0.9 * 275) + 378 + 378 + 378 + (0.213 * 275) = 1439$ m

Height: $(0.2 * 275) + 327 + 327 + 327 + (0.2 * 275) = 1091$ m

The exact BSS IDs, channels and coordinates of the APs used in the 16 AP simulation are:

BSS ID	Channel	X – Coordinate	Y – Coordinate
1	1	0	0
2	6	378	0
3	11	756	0
4	1	1134	0
5	6	-189	-327
6	11	189	-327
7	1	567	-327
8	6	945	-327
9	1	0	-654
10	6	378	-654
11	11	756	-654
12	1	1134	-654
13	6	-189	-981
14	11	189	-981
15	1	567	-981
16	6	945	-981

Table 3: Coordinates, BSS IDs and Channel Configuration of the APs Used in the 16 AP WLAN Deployment

Chapter 3: Opnet

3.1. What is Opnet Modeler?

Opnet Modeler is an environment for network modeling and simulation, allowing the design and study of communication networks, devices, protocols, and applications with flexibility and scalability. Modeler's object-oriented modeling approach and graphical editors mirror the structure of actual networks and network components. Modeler supports a variety of network types and technologies [8].

3.2. Opnet's Editors

OPNET Modeler is based on a series of hierarchical editors that directly parallel the structure of real networks, equipment, and protocols.

Project Editor: The Project Editor graphically represents the topology of a communications network. Networks consist of node and link objects, configurable via dialog boxes. There are two options to build a network in Opnet, simply by dragging and dropping nodes and links from the editor's object palettes to build the network, or by using import and rapid object deployment features. Also, there is the capability to use objects from OPNET's extensive Model Library, or to customize palettes to contain preferred node and link models. The Project Editor provides geographical context, with physical characteristics reflected appropriately in simulation of both wired and mobile/wireless networks [8].

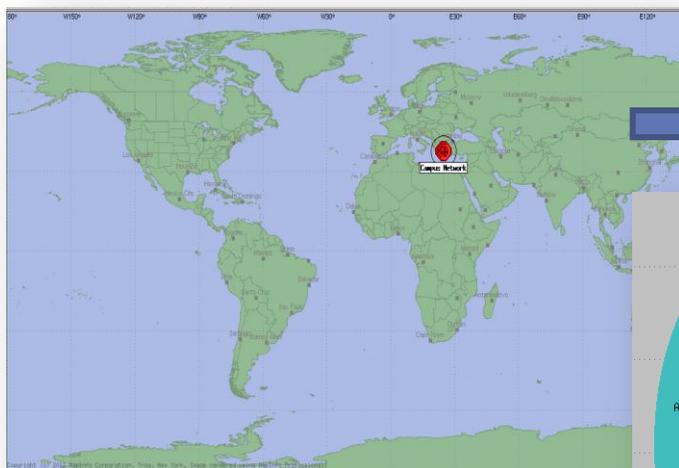


Figure 9: Geographical Position of the Simulated Network Topology

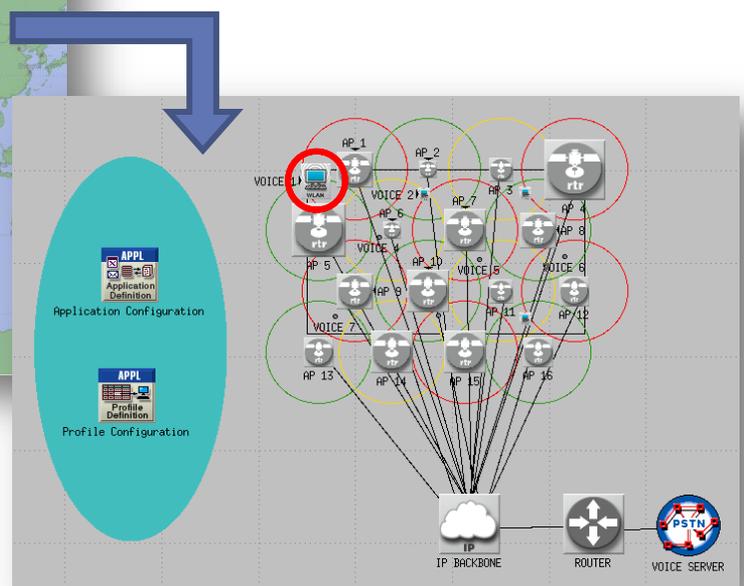


Figure 10: The Simulated Network Model Topology

Node Editor: The Node Editor captures the architecture of a network device or system by depicting the flow of data between functional elements, called "modules." Each module can generate, send, and receive packets from other modules to perform its function within the node. Modules typically represent applications, protocol layers, algorithms and physical resources, such as buffers, ports, and buses. Modules are assigned process models (developed in the Process Editor) to achieve any required behavior.

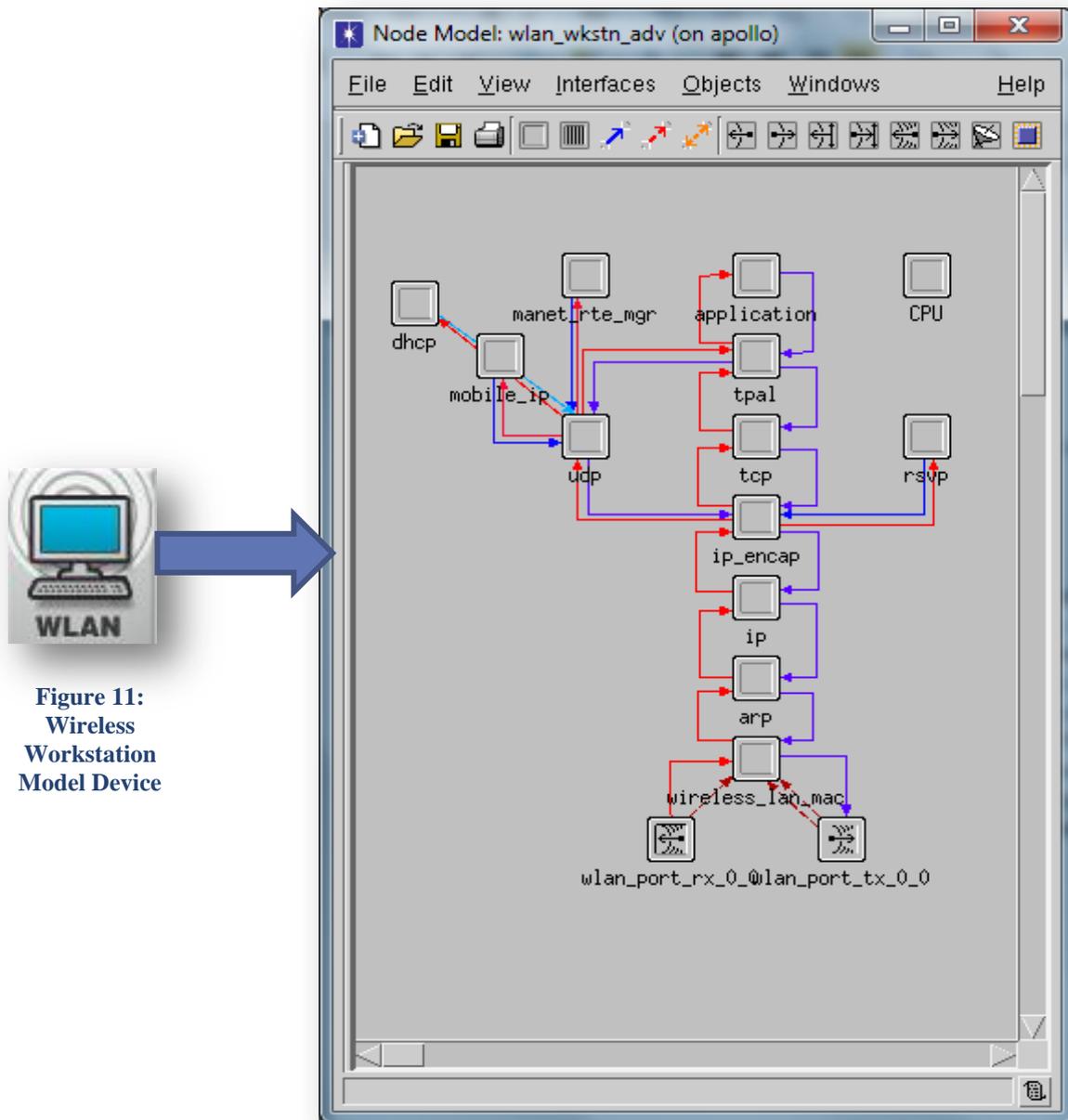


Figure 11:
Wireless
Workstation
Model Device

Figure 12: Node Model of the Wireless Workstation Model Device

Process Editor: The Process Editor uses a powerful finite state machine (FSM) approach to support specification, at any level of detail, of protocols, resources, applications, algorithms, and queuing policies. States and transitions graphically define the progression of a process in response to events. Each state of a process model contains C/C++ code, supported by an extensive library of functions designed for protocol programming. Each FSM can define private state variables and can make calls to code in user-provided libraries. FSMs are dynamic and can be spawned (by other FSMs) during simulation in response to specific events. Dynamic FSMs dramatically simplify the specification of protocols that manage a scalable number of resources or sessions, such as TCP or ATM. Opnet Modeler allows to use the Process Editor to develop entirely new process models. In our implementation we will only use the models in the OPNET Technologies' Model Library as a starting point [8].



Figure 13: Application Layer Module of the Wireless Workstation Model Device

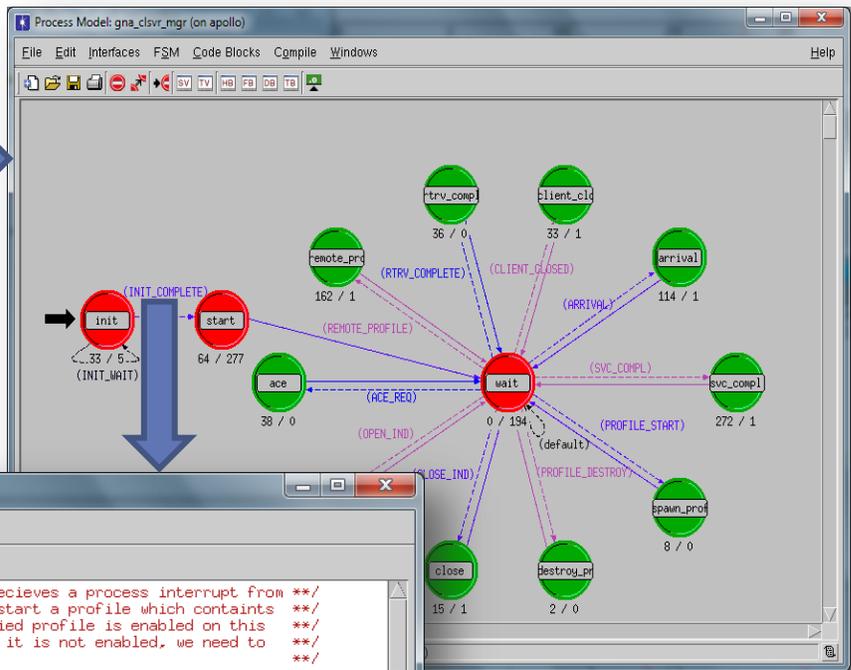


Figure 14: Process Model of the Application Layer Module

```

1  /** This state is entered when the server receives a process interrupt from */
2  /** a remote custom manager, telling it to start a profile which contains */
3  /** custom application in it. If the specified profile is enabled on this */
4  /** server, then we do nothing, however, if it is not enabled, we need to */
5  /** start it for one profile duration. */
6
7  /* Get the ici associated with the process interrupt.*/
8  ici_ptr = op_intrpt_ici ();
9
10 /* Get the profile name and the end time of the profile */
11 if ((ici_ptr == OPC_NIL) ||
12     (op_ici_attr_get (ici_ptr, "Profile Name", &profile_name) == OPC_COMPCODE_FAILURE)
13     (op_ici_attr_get (ici_ptr, "End Time", &remote_end_time) == OPC_COMPCODE_FAILURE)
14     (op_ici_attr_get (ici_ptr, "application repetition index", &repetition_index) == OI
15     (op_ici_attr_get (ici_ptr, "profile repetition index", &profile_repetition_index) :
16     (op_ici_attr_get (ici_ptr, "application row index", &appl_row_index) == OPC_COMPCODE
17     gna_clsivr_mgr_warn( "Could not get info from ICI");
18
19 /* Destroy ici */
20 if (ici_ptr != OPC_NIL)
21     op_ici_destroy (ici_ptr);
22
23 /* Check if the remotely specified profile is already enabled */
24 /* on this node. */
25 if (active_profiles_lptr != OPC_NIL)
26     {
27     active_size = op_prg_list_size (active_profiles_lptr);
28     for (i = 0; i < active_size; i++)
29     {

```

Figure 15: Open Model C/C++ Source Code

3.3. Opnet Wireless Module

The OPNET Wireless module extends the functionality of OPNET's core products with high-fidelity modeling, simulation, and analysis of wireless networks. The Wireless module also integrates OPNET's full protocol stack modeling capability, including MAC, routing, higher layer protocols, and applications, with the ability to model all aspects of wireless transmissions [9].

3.4. Modeling the Network

3.4.1. The Network Topology for deploying VoIP

At this initial phase a number of issues pertaining to the deployment of VoIP need to be investigated. We have conducted the radio frequency planning and determined the positions of the APs in the 9 and 16 AP WLAN deployments, the exact coordinates are presented in Table 2 and Table 3. The APs are connected to an internet backbone router via optical fibers whose speed is 622Mbps. The 802.11n amendment that we use can support up to 600Mbps data rate, thus the network should be able to support future traffic growth. When introducing a network service such as VoIP one has to characterize the additional components or devices that should be added in the topology in order to support it. For deploying VoIP, a VoIP Gateway is required to handle the external calls to/from the Public Switched Telephone Network (PSTN) which we connect to the Internet backbone through a switch via an Ethernet connection operating at 1 Gbps. The VoIP Gateway is modeled as an Ethernet workstation, since we are interested only in collecting statistics inside the network, and it supports one underlying Ethernet connection at a maximum connection of 1 Gbps. We also need typical wireless PC workstations that are VoIP enabled. The applications deployed are generated and configured by the "Application" and "Profile" configuration nodes respectively. The 9 and 16 AP WLAN simulation topologies are presented in Figure 16 and Figure 17 with one wireless workstation per cell and in Figures 18 and 19 with fifteen wireless workstations per cell. The exact models used are presented in Table 4.

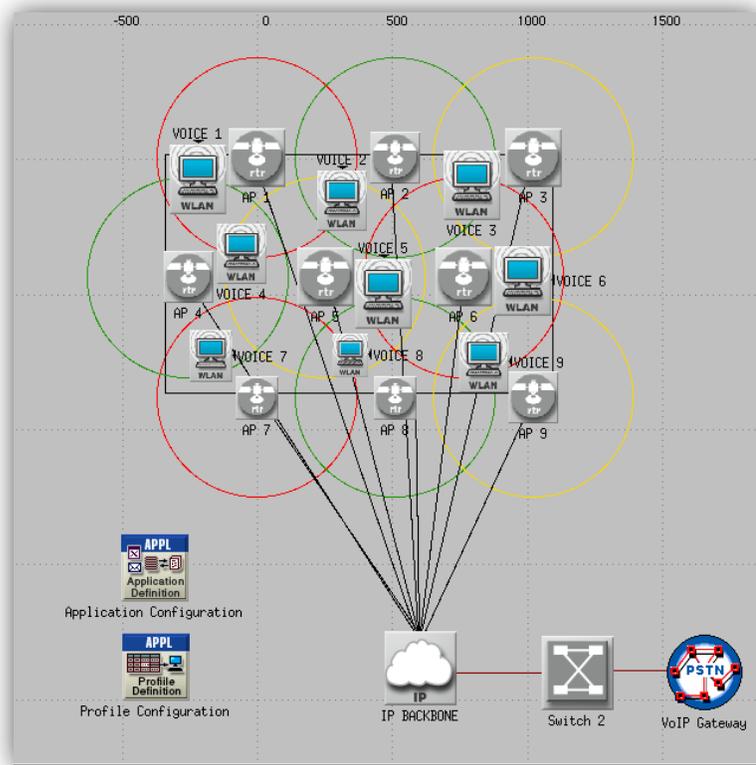


Figure 16: 9 AP WLAN Simulation Topology with One Wireless Workstation Per Cell

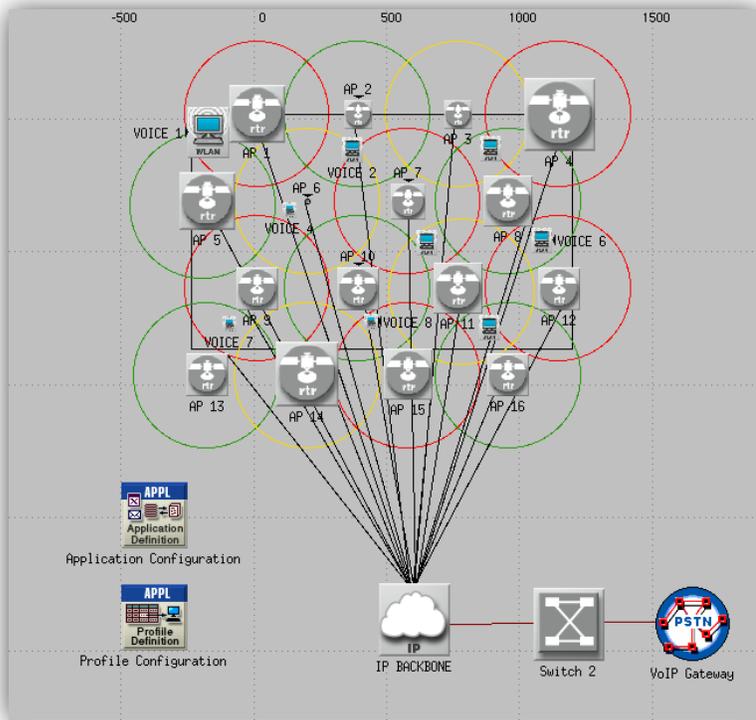


Figure 17: 16 AP WLAN Simulation Topology with One Wireless Workstation Per Cell

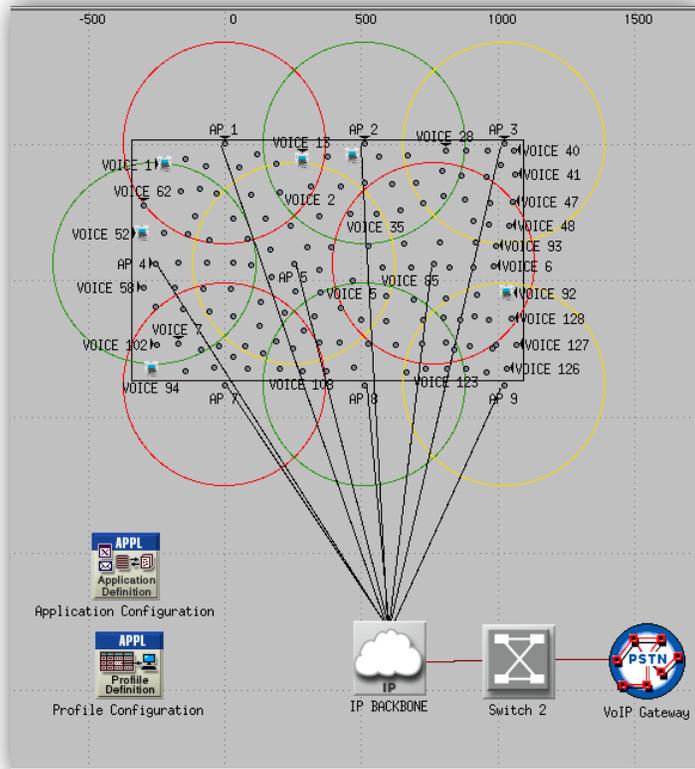


Figure 18: 9 AP WLAN Simulation Topology with Fifteen Wireless Workstations Per Cell

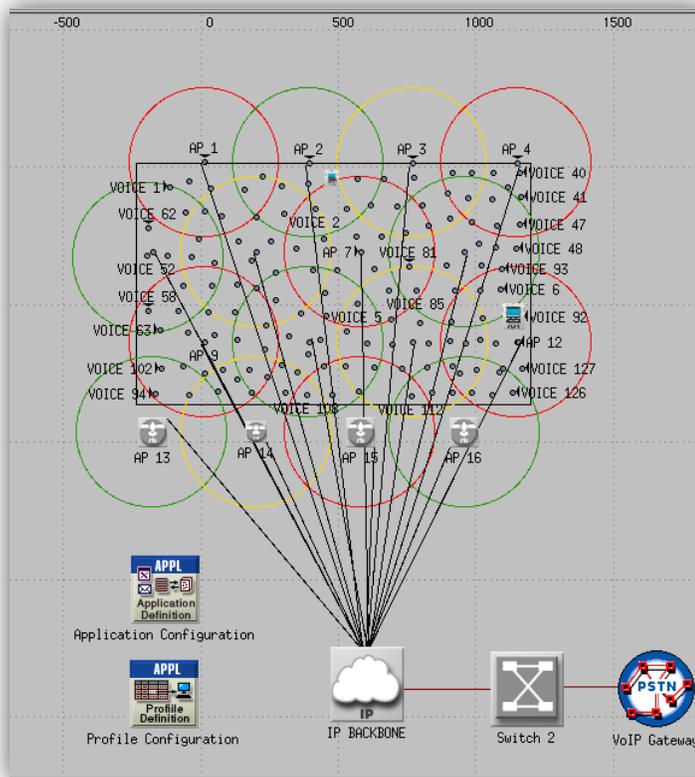


Figure 19: 16 AP WLAN Simulation Topology with Fifteen Wireless Workstations Per Cell

3.4.2. Enhancing the Network Topology to Support Additional Applications

In order to make our WLANs support HTTP, E-mail, Database, FTP, Print and Videoconferencing traffic we have to add some extra devices in our network. We need HTTP, E-mail, Database, FTP and Printer servers as well as a Videoconferencing traffic gateway.

The FTP, Database and Printer Servers are connected to a switch (Switch 1), which we connect to the Internet backbone via an Ethernet connection operating at 1 Gbps. The HTTP, E-mail servers and the Videoconferencing traffic gateway are connected to a second switch (Switch 3). Again, we connect the switch to the Internet backbone via an Ethernet connection operating at 1 Gbps. Remember that there is a third switch (Switch 2) connecting the VoIP Gateway to the IP backbone. It is more reliable and fault-tolerant not to connect all the devices to the same switch in order to eliminate problems that stem from a single point of failure. For example, if Switch 2 fails, only external calls to/from the network will be affected. All the servers are modeled as Ethernet servers and the Videoconferencing traffic gateway is modeled as an Ethernet workstation. The applications will be defined and configured using the “Application” and “Profile” configuration models.

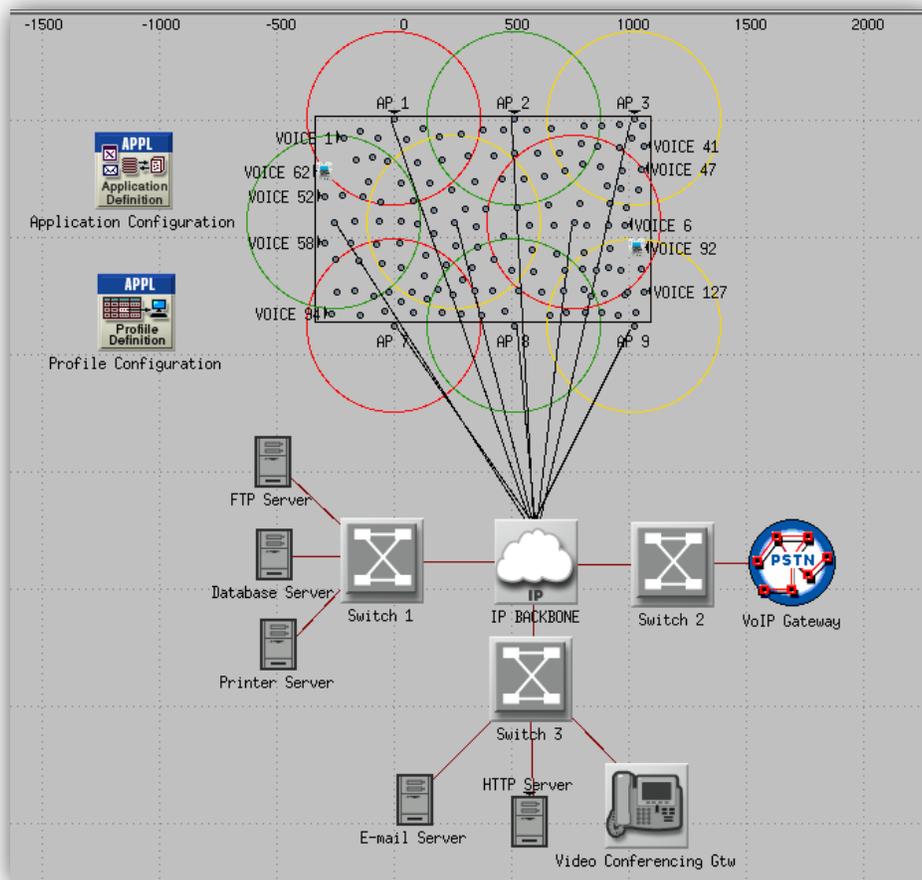


Figure 20: 9 AP WLAN Simulation Topology with Fifteen Wireless Workstations Per Cell Supporting FTP, Database, Print, E-mail, HTTP and Videoconferencing Applications

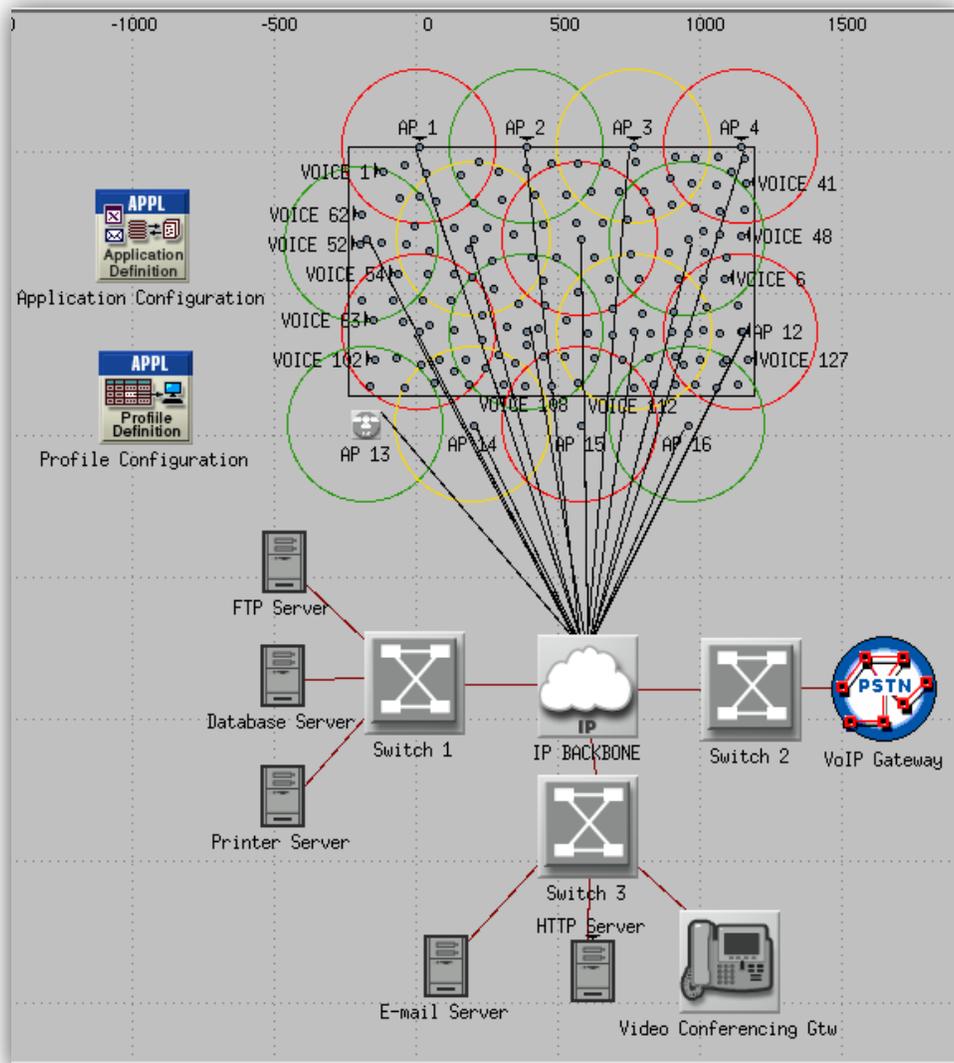


Figure 21: 16 AP WLAN Simulation Topology with Fifteen Wireless Workstations Per Cell Supporting FTP, Database, Print, E-mail, HTTP and Videoconferencing Applications

3.4.3. Introducing Peer – To – Peer Links to the Network Topology

Section 1.3.2 describes the operation of ad hoc or Peer-To-Peer (P2P) WLANs. In the initial network topology we introduce P2P links, in the form of mobile phone devices that are within range of each other. There is no access point or WLAN controller. The data transmitted by a particular source station travels directly to the applicable destination station. The mobile phone devices exchange IP Telephony VoIP calls and they can ‘hear’ each other on the channel that we assign to them. Each P2P link has its own unique BSS ID. In the simulation topology presented in Figure 22, we add three links in the 9 AP WLAN deployment. They are all added in cells which operate on channel 6 and 5 simultaneous calls are added on each link. In Figure 23, the P2P links are added in cells which operate on channel 1 and 8 simultaneous calls are added on each link. Of course a mobile phone cannot support 5 or 8 calls simultaneously. But the purpose of these experiments is to show that the limiting factor in our network is the channel and not the AP capacity, so we focus on the capacity of the channels and the impact of P2P links on them.

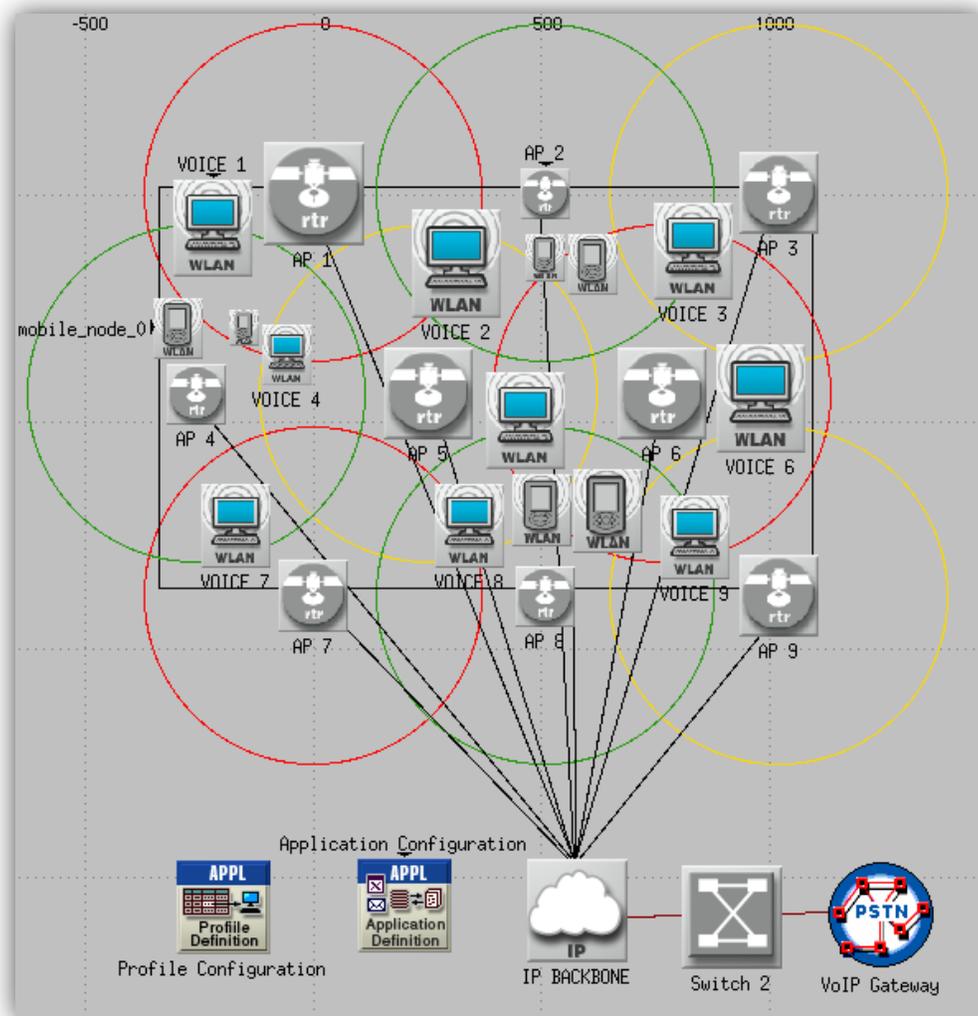


Figure 22: 9 AP WLAN Simulation Topology with P2P links on Channel 6

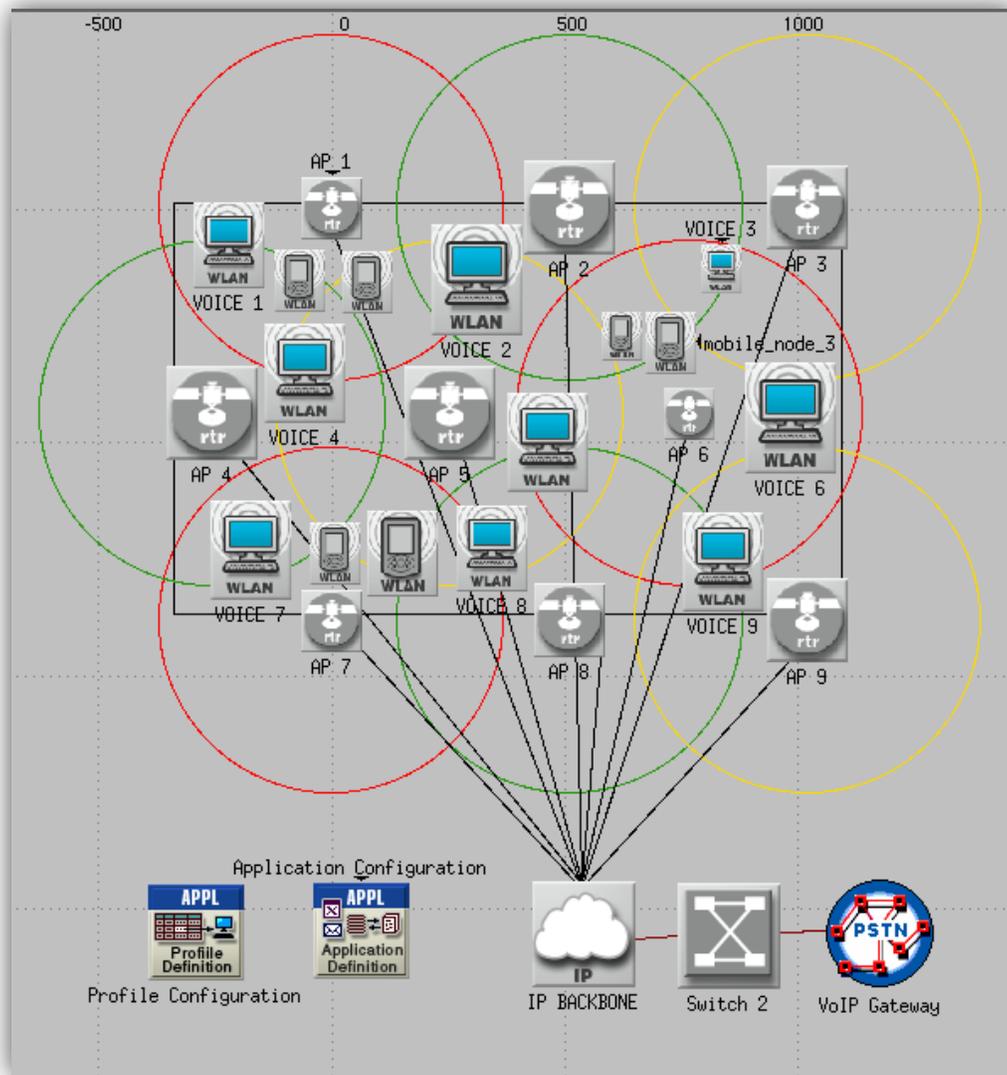


Figure 23: 9 AP WLAN Simulation Topology with P2P links on Channel 1

3.4.4. Models Used in the WLAN deployment

Model Icon	Model Description
	<p>The "wlan_wkstn_adv" node model represents a wireless workstation with client-server applications running over TCP/IP and UDP/IP.</p>
	<p>The "wlan_ethernet_slip4_router" is a Wireless LAN based router with one ethernet and 4 SLIP interfaces. (The Serial Line Internet Protocol (SLIP) is an encapsulation of the Internet Protocol designed to work over serial ports and modem connections.)</p>
	<p>The "ethernet16_switch" node model represents a switch supporting up to 16 Ethernet interfaces.</p>
	<p>The "router_slip64_dc_39_upgrade" router device supports 128 SLIP interfaces and it can be upgraded by adding extra Ethernet ports. It is used as the backbone router in our deployment. Backbone routers need to be powerful enough to handle information on the Internet backbone.</p>



This is the model that represents the VoIP Gateway – a sink for our all the external calls of the deployment. The "ethernet_wkstn" node model represents a workstation with client – server applications running over TCP/IP and UDP/IP. The workstation supports one underlying Ethernet connection at 10 Mbps, 100 Mbps, or 1000 Mbps.



This is the model that represents the Videoconferencing traffic gateway – a sink for all videoconference applications of the deployment. The "ethernet_wkstn" node model represents a workstation with client – server applications running over TCP/IP and UDP/IP. The workstation supports one underlying Ethernet connection at 10 Mbps, 100 Mbps, or 1000 Mbps.



The "ethernet_server" model represents a server node with server applications running over TCP/IP and UDP/IP. This node supports one underlying Ethernet connection at 10 Mbps, 100 Mbps, or 1 Gbps. The operational speed is determined by the connected link's data rate.



The "wlan_wkstn_adv" node model can also represent a mobile phone device operating in ad-hoc mode. Two "wlan_wkstn_adv" node models communicating in ad-hoc mode form a P2P link.



The "Application Configuration" node is used for the following specifications:

"Application Specification": Specifies applications using available application types. In the process of creating new applications we can specify a name and the corresponding description of the application.

"Voice Encoder Schemes": We can specify encoder parameters for each of the encoder schemes used for generating Voice Traffic in the network.



The "Profile Configuration" node can be used to create user profiles. These user profiles can then be specified on different nodes in the network to generate application layer traffic.

The applications defined in the "Application Configuration" objects are used by this object to configure profiles by specifying the behavior of the previously configured applications. Therefore, we create applications using the "Application Configuration" model before using this object.

We can specify the traffic patterns followed by the applications as well as the configured profiles.



The 1000BaseX duplex link represents an Ethernet connection operating at 1 Gbps and it is used to connect the IP backbone to the switch and the switch to the VoIP Gateway.



The "ppp_adv" point-to-point link connects two nodes with serial interfaces at a selectable data rate. This link connects the APs to the Internet backbone its speed is 622Mbps and the physical medium it represents is an optical fiber.

Table 4: OPNET Models Used in the WLAN Deployment

3.4.3 Opet Wireless LAN Parameters

The following WLAN parameters are included in all models that have a WLAN interface, such as Wireless LAN Workstations, Wireless LAN Servers and Routers with a WLAN interface (APs). These attributes are grouped under Wireless LAN Parameters and are presented in Table 5, along with the values used in our deployments.

Node Attributes



Figure 24:
Wireless
Workstation
Model Device

Frequency Hopping
Direct Sequence
Infra Red
OFDM (802.11a)
Extended Rate PHY (802.11g)
HT PHY 2.4GHz (802.11n)
HT PHY 5.0GHz (802.11n)

Figure 25: Available Options for
the Physical Characteristics
Parameters

Attribute	Value
Wireless LAN	
Wireless LAN MAC Address	Auto Assigned
Wireless LAN Parameters	(...)
BSS Identifier	1
Access Point Functionality	Disabled
Physical Characteristics	HT PHY 2.4GHz (802.11n)
Data Rate (bps)	52 Mbps (base) / 480 Mbps (max)
Channel Settings	Auto Assigned
Transmit Power (W)	0.0035
Packet Reception-Power Thre...	-95
Rts Threshold (bytes)	None
Fragmentation Threshold (byt...	None
CTS-to-self Option	Enabled
Short Retry Limit	7
Long Retry Limit	4
AP Beacon Interval (secs)	0.02
Max Receive Lifetime (secs)	0.5
Buffer Size (bits)	infinity
Roaming Capability	Disabled
Large Packet Processing	Drop
PCF Parameters	Disabled
HCF Parameters	Default
High Throughput Parameters	(...)
Number of Spatial Streams	1
Guard Interval	Short (400ns)
Greenfield Operation	Disabled
40MHz Operation Paramete...	Disabled
AP Specific Parameters	Default

Figure 26: Wireless LAN Parameters for Devices with a Wireless LAN Interface

Wireless Node Attribute	Description	Value
BSS Identifier	It identifies the BSS to which a WLAN MAC belongs.	1-9 , 1-16
Access Point Functionality	Enables or disables access point operation in the node.	Enabled for APs, Disabled for clients
Physical Characteristics	The options for the physical layer configuration in the IEEE 802.11 standard are depicted in Figure 19.	802.11n, 2.4 Ghz
Data rate	Specifies the data rate that will be used by the MAC for the transmission of the data frames via physical layer.	Variable
Channel Settings	The channel numbers correspond to the channels available in the 2.4 GHz ISM band specified in IEEE 802.11.	1,6,11
Transmit Power	Specifies the transmit power of the station in Watts. No limits are imposed upon the value of this attribute.	Variable
Packet Reception Power Threshold	<p>Defines the received power threshold (receiver sensitivity) value of the radio receiver in dBm for arriving WLAN packets. Packets with a power less than the threshold are not sensed and decoded by the receiver. The status of such packets is set to noise; they cause interference and possibly bit errors if they collide with valid packets at the receiver.</p> <p>The packets whose received power is higher than the threshold are considered as valid packets. They are sensed by the MAC and they can be received successfully unless they get bit errors due to interference, background noise and/or colliding with other valid packets.</p>	-95 dBm

RTS Threshold	Specifies the threshold for performing RTS/CTS frame exchange preceding the transmission of the data frame. For the data packets whose size together with the WLAN MAC header (which itself is 40 bytes) exceeds the threshold, completion of a successful RTS/CTS frame exchange is required before the transmission of the actual packet over the radio channel. If the value is set to "None", RTS/CTS exchange is not used for any frame regardless of its size.	None
Fragmentation Threshold	Specifies the fragmentation threshold in bytes as well as the size of the fragments except the last fragment. Any data packet received from a higher layer with a size greater than this threshold will be divided into fragments. Special value "None" indicates that fragmentation will not be used for the transmission of any higher layer data packet regardless of its size.	None
CTS – To – Self Option	Enables the optional CTS – to – Self protection mechanism that is specified in the IEEE 802.11g-2003 standard. Hence, the attribute is used only by the MACs that operate in 802.11g mode.	Disabled
Short Retry Limit	Specifies the maximum number of transmission attempts for the frames whose size are less than or equal to RTS Threshold. If RTS/CTS frame exchange is not enabled, this attribute applies to all frames. Frames that could not be transmitted after this number of attempts are discarded by the MAC.	7
Long Retry Limit	Specifies the maximum number of transmission attempts for the frames whose size are greater than the Rts Threshold (provided that RTS/CTS frame exchange is enabled). Frames that could not be transmitted after this number of attempts are discarded by the MAC.	4
Beacon Interval	Specifies the Target Beacon Transmission Time. In other words, the AP will try to transmit Beacon frames with a period specified by the value of this attribute. The actual time when the Beacon is transmitted by the AP can be slightly off since it may need to wait for the shared medium to become available for this transmission.	0.02 sec

Maximum Receive Lifetime	Specifies the maximum time allowed for the reassembly of a fragmented MSDU. MSDUs that can not be reassembled within this time will be discarded and their reassembly processes will be terminated.	0.5 sec
Buffer Size:	Specifies the maximum size of the higher layer data buffer in bits. Once the buffer limit is reached, the data packets arrived from higher layer will be discarded until some packets are removed from the buffer so that the buffer has some free space to store these new packets. We set this attribute to infinity so that the packets are discarded only due to retry threshold exceedance. This attribute determines the number of spatial streams in the STA (see below).	Infinite
Roaming Capability	The roaming capability permits the STA to scan for other APs and switch APs when the signal from the connected AP becomes weak.	Disabled
Large Packet Processing	The value of this attribute is used if MAC receives a higher layer packet whose size is larger than the maximum allowed data size (2304 bytes/18432 bits). If the value is set to "Drop" then these large packets are dropped and a simulation log message is written. If it is set to "Fragment" then these packets are accepted only if fragmentation is enabled.	Drop
Number of Spatial Streams	Spatial stream is a stream of bits or modulation symbols that may be transmitted over multiple spatial dimensions that are created by the use of multiple antennae at both ends of a communications link. Configuring the number of spatial streams to be greater than one, increases the transmission data rate from the base data rate. It is assumed that the receiving HT-STA (High Throughput Station) will always be able to decode data transmitted using 1 to 4 spatial streams irrespective of the number of spatial streams configured at the receiving HT-STA.	1

Guard Interval	This attribute allows the user to configure the guard interval value for a HT STA. The regular GI value is 800ns. Setting the attribute to "Short (GI)" increases the data rate from the base value as specified in the IEEE 802.11n 2009 standard. By default with HT-PHY, the STA will use short GI (400ns).	400 ns
Greenfield Operation	This attribute enables or disables Greenfield operation (see 1.4.6) in a HT-STA. If the Greenfield operation is enabled, then the HT-STA can use HT-GF PLCP (High Throughput – Greenfield PLCP Header) header for data frames when communicating with another greenfield capable HT-STA.	Disabled
40 MHz Operation Parameters	This attribute allows the user to specify whether this 11n capable MAC (HT-STA) supports 40Mhz operation defined as 20/40 MHz operation in the IEEE 802.11n standard, and to configure the attributes of 20/40 MHz operation	Disabled
AP Specific Parameters	This compound attribute contains parameters specifically for a high throughput (HT) access point (AP). It is determined by the attribute Legacy Support, which takes the value “Full Support” by default. Full Support" attribute setting is fully backward compatible with all the legacy technologies (802.11a, 802.11b and 802.11g) supported by 802.11n. The access point will not restrict association of the legacy STAs.	Default (Full Support)

Table 5: Wireless LAN Parameters for Devices with a Wireless LAN Interface and Values Used for the Simulations

3.4.4. Configuring the components

The APs: The main parameter to be set in the APs’ configuration is the forwarding rate. The forwarding rate is the number of packets that are processed by the forwarding processor in one second and in our network its value is 50000 packets/sec.

The Internet Backbone Router: The forwarding rate in this node is doubled, set to 100000 packets/sec.

The switches: Similarly, the main parameter to be set in the switch is the switching speed, which is the rate at which packets are switched from the switch processor to the appropriate output port. This is dependent on the speed of the switch processor and in our case is 500000 packets/sec.

3.5 Modeling the traffic

Modeling Applications in OPNET

OPNET Modeler provides standard built – in models for software applications such as VoIP, Video, HTTP, E-mail and FTP which can be easily configured or altered to simulate applications.

By “Applications” from now on, we will refer to HTTP, E-mail, FTP, Database Query and Print requests.

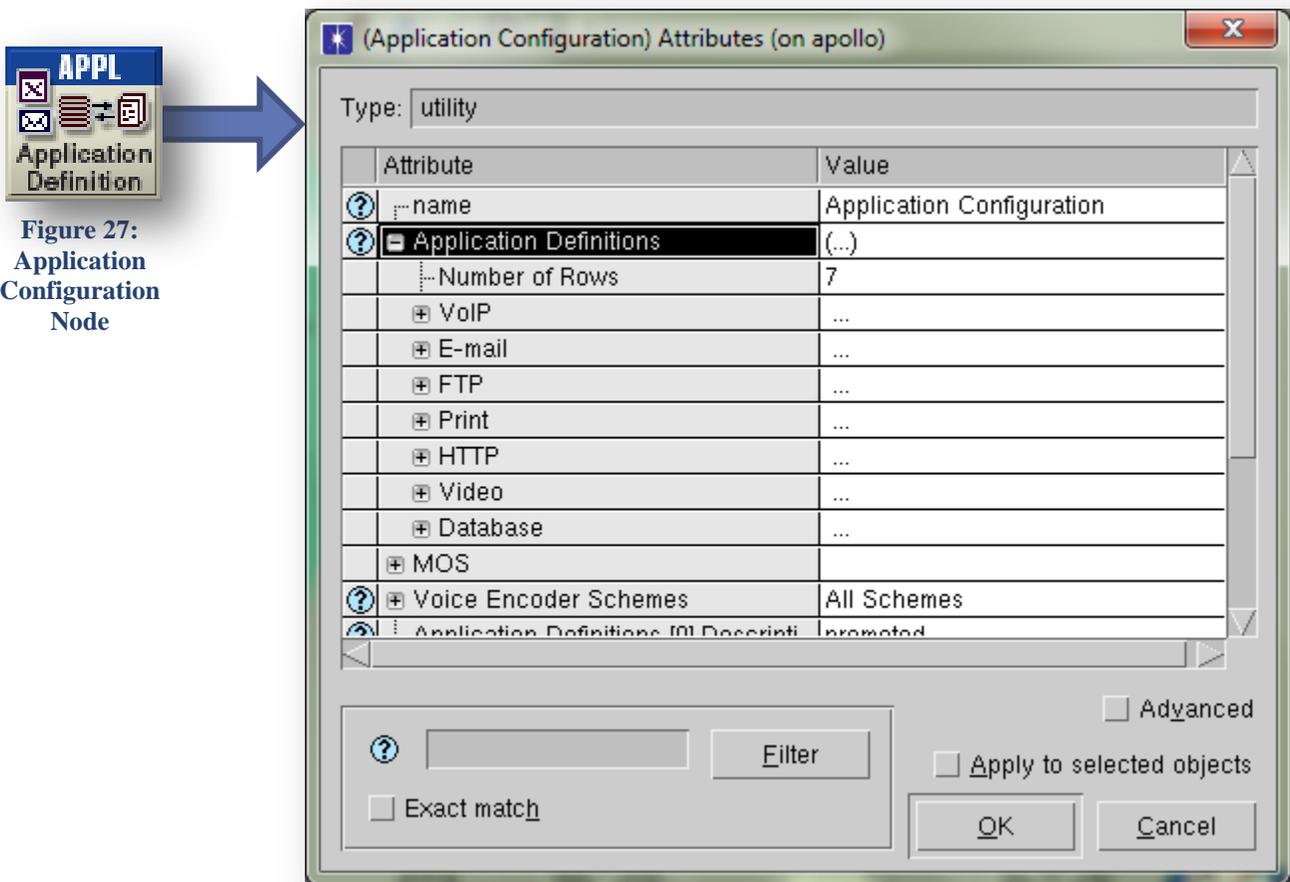


Figure 27: Application Configuration Node

Figure 28: Standard Built-in Models for Software Applications – VoIP, E-mail, FTP, Print, HTTP, Video, Database

3.5.1 VoIP Traffic Characteristics

Firstly we are going to introduce voice traffic in our WLAN deployments. We will deploy two different types of predefined voice calls in OPNET, IP Telephony and PCM Quality calls. The traffic characteristics of the calls introduced are:

Codec Information					
	Codec	Bit Rate (Kbps)	Codec Sample Size (Bytes)	Codec Sample Interval (ms)	Mean Opinion Score (MOS)
PCM Quality	G.711	64	80	10	4.1
IP Telephony	G.729a	8	10	10	3.92

Table 6: Codec Information for the PCM Quality and the IP Telephony Voice Calls

Bandwidth Calculations				
	Voice Payload Size (Bytes)	Voice Payload Size (ms)	Packets Per Second (PPS)	Required Bandwidth per Call (Kbps)
PCM Quality	80	10	100	115.2
IP Telephony	10	10	100	59.2

Table 7: Bandwidth Calculations for the PCM Quality and the IP Telephony Voice Calls

The explanation of these terms follows in Table 8. [15]

Codec Bit Rate (Kbps)	<ul style="list-style-type: none"> Based on the codec, this is the number of bits per second that need to be transmitted to deliver a voice call. $codec\ bit\ rate = \frac{codec\ sample\ size}{codec\ sample\ interval}$
Codec Sample Size (Bytes)	<ul style="list-style-type: none"> Based on the codec, this is the number of bytes captured by the Digital Signal Processor (DSP) at each codec sample interval.
Codec Sample Interval (ms)	<ul style="list-style-type: none"> This is the sample interval at which the codec operates. For example, the G.729 coder operates on sample intervals of 10 ms, corresponding to 10 bytes (80 bits) per sample at a bit rate of 8 Kbps.
MOS	<ul style="list-style-type: none"> MOS is a system of grading the voice quality of telephone connections. With MOS, a wide range of listeners judge the quality of a voice sample on a scale of one (bad) to five (excellent). The scores are averaged to provide the MOS for the codec.
Voice Payload Size (Bytes)	<ul style="list-style-type: none"> The voice payload size represents the number of bytes (or bits) that are filled into a packet. The voice payload size must be a multiple of the codec sample size. For example, G.729 packets can use 10, 20, 30, 40, 50, or 60 bytes of voice payload size.
Voice Payload Size (ms)	<ul style="list-style-type: none"> The voice payload size can also be represented in terms of the codec samples. For example, a G.729 voice payload size of 10ms represents a voice payload of 10 bytes $\frac{(10\ bytes * 8)}{10\ ms} = 8\ Kbps$
PPS	<ul style="list-style-type: none"> PPS represents the number of packets that need to be transmitted every second in order to deliver the codec bit rate. For example, for a G.729 call with voice payload size per packet of 20 bytes (160 bits), 50 packets need to be transmitted every second $PPS = \frac{codec\ bit\ rate}{voice\ payload\ size}$

Table 8: Explanation of Codec Information and Bandwidth Calculation Related Terms

3.5.1.1. PCM Quality

The required bandwidth for a single VoIP call, one direction, is 64 kbps. G.711 codec samples 10 ms of voice per packet. Therefore, 100 such packets need to be transmitted per second.

With every packet of size 80 bytes or 640 bits, headers of additional protocol layers are added. These headers include PREAMBLE + PLCP HEADER +MAC HEADER of sizes 144 + 48 + 320 bits, respectively. Therefore, a total of 1152 bits, need to be transmitted 100 times per second (100 PPS), or 115.2 kbps, in one direction.

3.5.1.2. IP Telephony

The required bandwidth for a single VoIP call, one direction, is 8 kbps. G.711 codec samples 10 ms of voice per packet. Therefore, 100 such packets need to be transmitted per second.

With every packet of size 10 bytes or 80 bits, headers of additional protocol layers are added. These headers include PREAMBLE + PLCP HEADER +MAC HEADER of sizes 144 + 48 + 320 bits, respectively. Therefore, a total of 592 bits, need to be transmitted 100 times per second (100 PPS), or 59.2 kbps, in one direction.

3.5.1.3. Voice Capacity

When deploying a VoWLAN system, one of the most important issues is to determine the network voice capacity, in terms of the maximum number of simultaneous voice connections that can be supported in the IEEE 802.11 WLAN. When more voice connections join the network, the service rates of the APs and mobile stations decrease non-linearly because more collisions may occur.

It is pointed out that when the number of voice connections approaches the voice capacity, one more VoIP connection may cause the AP to be unstable, and thus jeopardize the performance of all downlink voice traffic. In addition, the experimental and analytical results reveal that the voice capacity is a function of the system parameters, including transmission rate, voice packet payload length (dependent on the codec used), and packetization interval. [2]

3.5.1.4. Voice Quality of Service

VoIP quality is closely related to three factors: packet end-to-end delay, delay jitter (delay variation), and packet loss. These are the three parameters that determine whether a VoIP has acceptable performance or not. A two-way conversation is very sensitive to delay and the International Telecommunication Union

(ITU) has recommended that one-way end-to-end delay should be no greater than 150 ms for good voice quality, and up to 400 ms can be considered acceptable but not desirable. Connectivity through a geostationary satellite falls into this category: above 400 msec, it is unacceptable. For VoWLAN applications, delay includes codec delay, packetization delay, and network delay in both the WLAN and the backbone networks. Delay jitter, or delay variation, mainly due to network dynamics, has even more negative effects on voice quality than that of delay. Since the WLAN is presumably the bottleneck, delay jitter in the WLAN is the dominant part. Packet loss is also a major source of impairment in VoIP systems. There are two sources of packet losses: (1) network packet losses, mainly due to network congestion (router buffer overflow), link failures and rerouting, transmission errors; and (2) discarded packet losses for packets that experienced excessive delay. [2] In our simulation, we have eliminated packet losses due to router buffer overflows, since we assume that we have infinite buffers in the APs and the wireless workstations and there is no occurrence of link failures.

3.5.2 FTP Traffic Characteristics

For the FTP and the following applications OPNET recommends modeling application behaviors with the exponential distribution, which assumes that requests are independent of each other. We configured the OPNET simulation to have inter-request and interarrival times modeled using the exponential distribution.

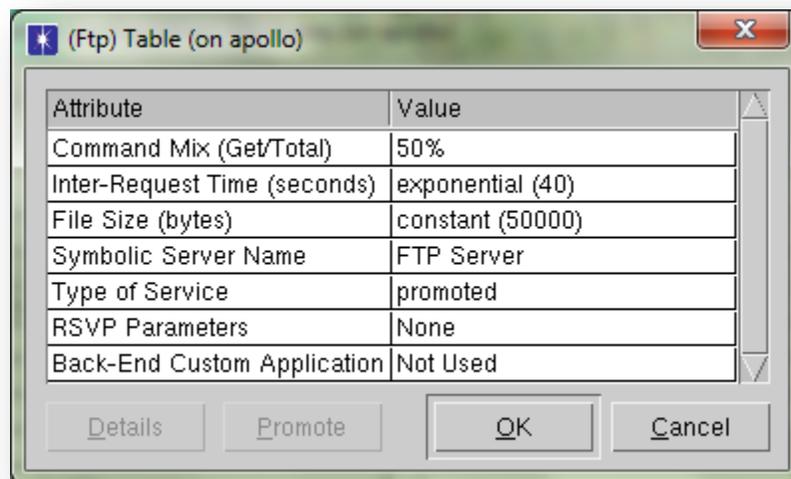


Figure 29: FTP Application Configuration – Traffic Parameters

The model allows for a custom configuration of FTP applications. The Command Mix parameter specifies the ratio between the number of get commands and the total number of FTP requests. For example, when the Command Mix parameter is set to its default value of 50%, half of the FTP requests will be to get (download) data and the other half to put (upload) data. The Inter-Request Time parameter is the time in seconds between consecutive FTP requests. The File Size defines the size in bytes of the FTP file to be transferred.

Lastly, the Symbolic Server Name and the Type of Service (ToS) attributes are common to this and the following applications and define the name of the server that will accommodate the traffic and the priority that the traffic has respectively. The ToS attribute has been “promoted” for all applications. In OPNET terminology this means that we do not predefine its value in the application’s attribute table but we can configure it at a higher level. Therefore, it becomes a variable for our simulation and we can set it when we will run the experiment.

3.5.3 Database Traffic Characteristics

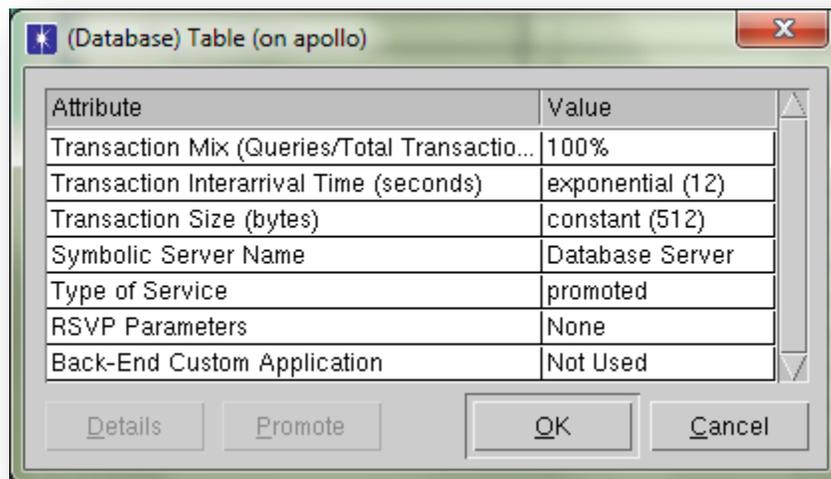


Figure 30: Database Application Configuration – Traffic Parameters

The database application configuration starts with the Transaction Mix attribute, which denotes the percentage of database query transactions to the total transactions. The remaining percent of the transactions are database entry transactions (in our case zero). The Interarrival Time parameter is the time in seconds between consecutive Database queries and the Transaction Size defines the size in bytes of the database transaction request.

3.5.4 Printer Traffic Characteristics

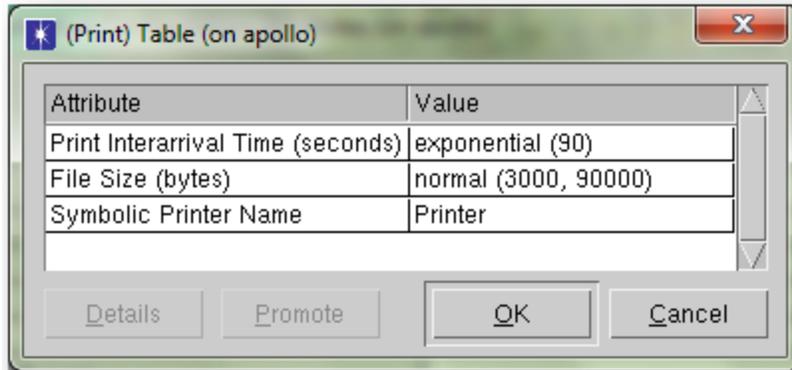


Figure 31: Print Application Configuration – Traffic Parameters

In the Print application configuration table, we can specify the Print Interarrival Time which is the amount of time in seconds between consecutive print jobs sent to the Printer server. The File Size defines the size of the print request in bytes, and here it adopts the predefined File Size configuration for a Text File print request in OPNET, which follows the normal distribution, with a mean outcome between 3 KB and 9 KB.

3.5.5 E-mail Traffic Characteristics

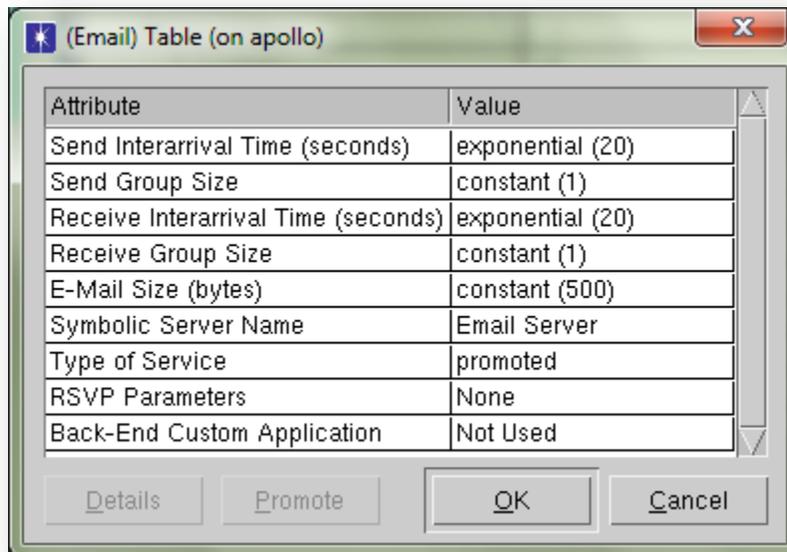


Figure 32: E-mail Application Configuration – Traffic Parameters

In the E-mail configuration table we can specify the Send and Receive Interarrival Times, which stand for the amount of time in seconds between consecutive sent and receive operations. The Send and Receive Group Size determines the number of e-mails messages per single sent or receive operation, while the E-Mail Size defines the size of the e-mail message in bytes.

3.5.6 HTTP Traffic Characteristics

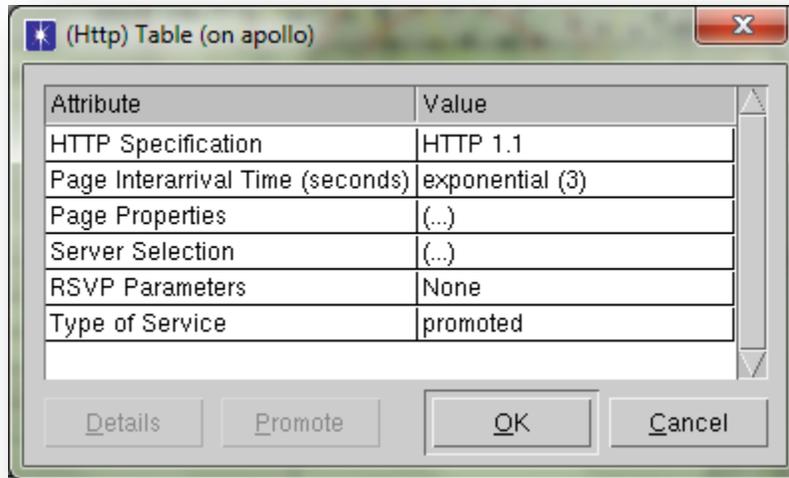


Figure 33: HTTP Application Configuration – Traffic Parameters

To configure a web application, the application named HTTP should be selected from the list of built-in models. OPNET allows configuring web applications via parameters such as: HTTP Specification which defines the version of HTTP protocol, Page Interarrival Time which specifies time in seconds between consecutive webpage downloads, and Page Properties (presented in Figure 32).

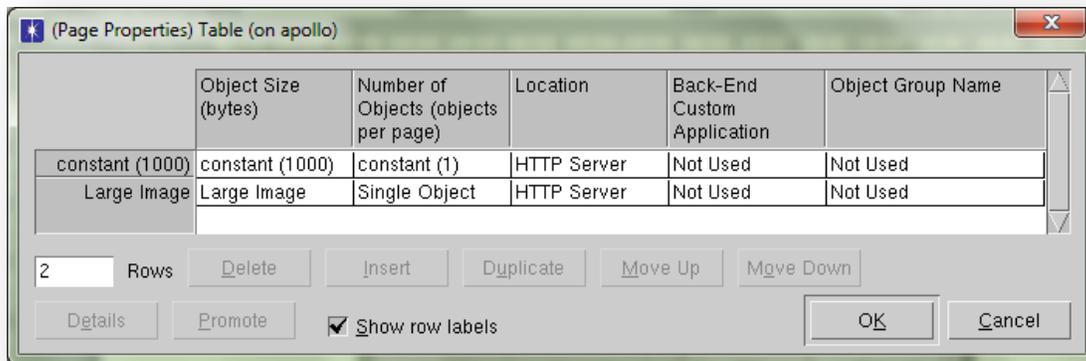


Figure 34: HTTP Page Properties Details

In our case, every HTTP Page contains two objects. A large image whose size is determined according to the uniform distribution and is between 2 to 10 MB (Figure 33) and a second object which has a constant size of 1 MB and could represent the text of the HTTP Page.

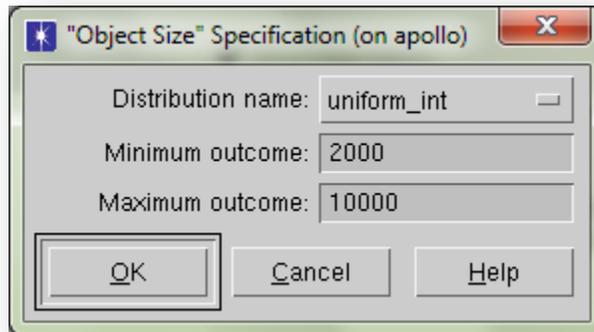


Figure 35: Large Image - Object Size Specification

3.5.7 Videoconference Traffic Characteristics

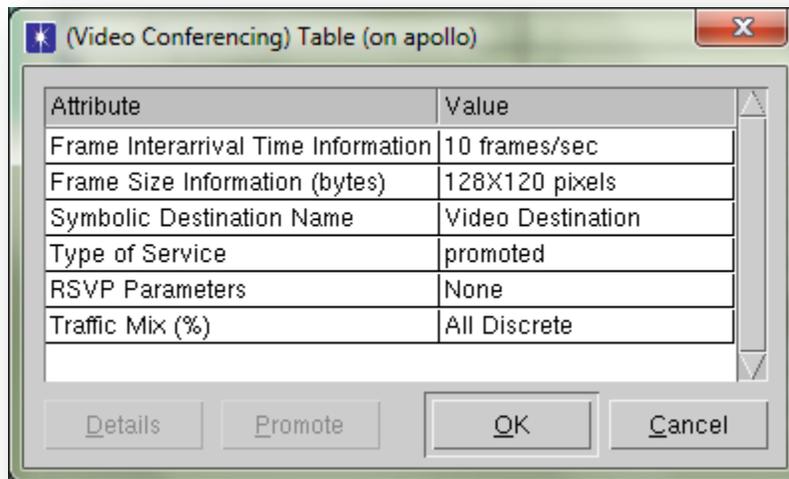


Figure 36: Videoconferencing Application Configuration – Traffic Parameters

In order to model the video application, again we deploy a predefined video application in OPNET, the Low Resolution Video application. This video application models a videoconference application which generates 10 frames/sec. Each frame has a size of 128X120 pixels and 9 bits/pixel. The Traffic Mix

parameter specifies whether the traffic is generated as pure discrete or pure background or part discrete/part background. The percentage specified is the part of the background traffic. Here the traffic is pure discrete.

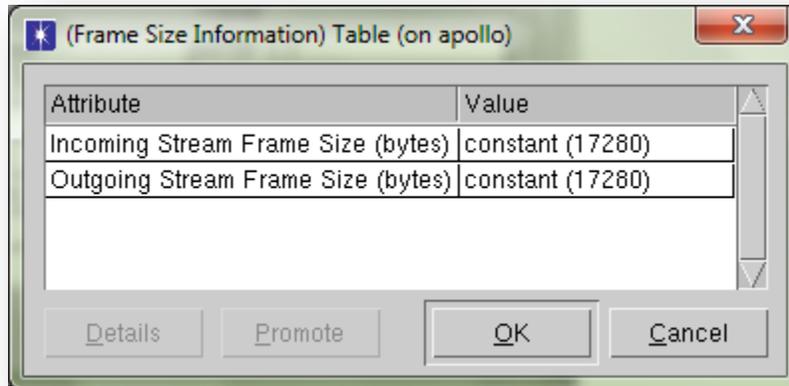


Figure 37: Videoconferencing Frame Size Configuration

The incoming and outgoing frame sizes are 17280 bytes.

$$frame\ size = \frac{128 * 120\ pixels}{frame} * \frac{9\ bits}{pixel} = 138240\ bits\ or\ 17280\ bytes$$

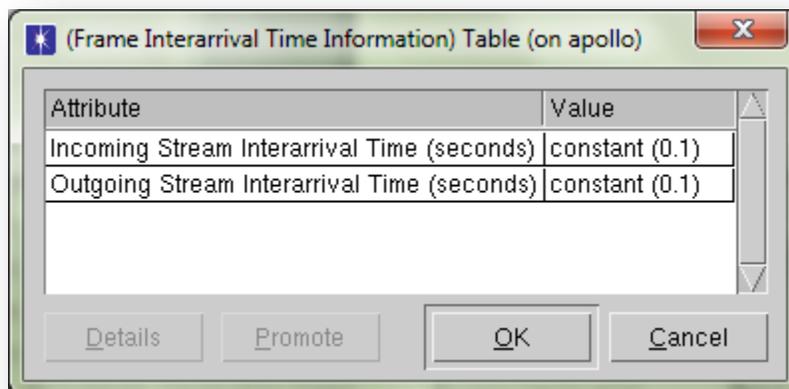


Figure 38: Videoconferencing Frame Interarrival Time configuration

3.6 Simulations

As we have mentioned previously, our goal is to determine the network capacity for VoIP, i.e., the maximum number of calls that the WLAN can support while maintaining VoIP QoS.

This can be done by adding calls incrementally to the network while monitoring the bounds for VoIP delay and network capacity or bandwidth. When any of these bounds is reached, the maximum number of calls can then be known. Note that the bound for network bandwidth is reflected in seeing packets being lost or observing a mismatch between VoIP traffic being sent and received. [16]

3.6.1 Simulation Scenarios

3.6.1.1 Determining the network capacity

We run the following scenarios for the 9 AP and the 16 AP WLAN deployments, for 6.5, 13, 19.5, 36, 52 and 65 Mbps data rates. We add IP Telephony and PCM Quality voice calls, in order to determine the network capacity for VoIP:

- 1) In the first scenario we deploy one wireless workstation per cell. We add 30 calls per workstation every 3 seconds and the simulation's duration is 240 seconds. This is the maximum duration for which we could run the experiment due to memory issues.
- 2) We remember that as more voice connections join the network, the service rates of the APs and mobile stations decrease because more collisions may occur. So we observe the maximum calls that each cell can support in the first scenario and, with respect to the results. Again, the simulation's duration is 240 seconds.
- 3) In the third scenario, we keep previous configuration, adding 15 wireless workstations per cell and deploying 1 call per workstation every 3 seconds. We also add the applications presented in 3.5. Here the simulation's duration is 190 seconds.
- 4) We add videoconference traffic on top of the applications in every one of the 15 wireless workstations per cell. The simulation's duration is 190 seconds.
- 5) Peer to Peer (P2P) links are added in the first scenario, in order to observe their impact on the number of the voice call connections. The simulation's duration is 200 seconds.

3.6.2 Simulation Statistics

Before we run the simulations, OPNET has to be configured to obtain results for the network components that we choose. These components include workstations, routers, and links.

In OPNET we can collect global and object statistics. If we want to collect a statistic from the entire network we choose the global statistic. Object or node statistics refer to specific devices.

Global Statistics

Voice

Statistic		Description
Traffic Sent	packets/sec	Average number of bytes or packets per second submitted to the transport layers by all voice applications in the network.
	bytes/sec	
Traffic Received	packets/sec	Average number of bytes or packets per second forwarded to all Voice applications by the transport layers in the network.
	bytes/sec	
Packet End – To – End Delay	sec	<p>The total voice packet delay, called "analog-to-analog" or "mouth-to-ear" delay:</p> $\text{delay} = \text{network_delay} + \text{encoding_delay} + \text{decoding_delay} + \text{compression_delay} + \text{decompression_delay},$ <p>where:</p> <p>Network delay is the time at which the sender node gave the packet to RTP to the time the receiver got it from RTP.</p> <p>Encoding delay (on the sender node) is computed from the encoder scheme.</p> <p>Decoding delay (on the receiver node) is assumed to be equal to the encoding delay.</p> <p>Compression and Decompression delays come from the corresponding attributes in the Voice application configuration.</p> <p>This statistic records data from all the nodes in the network.</p>

DB Query		
DB Query Response Time	sec	Time elapsed between sending a request and receiving the response packet. Measured from the time when the Database Query Application sends a request to the server to the time it receives a response packet. Every response packet sent from a server to a Database Query application is included in this statistic.
E-mail		
E-mail Download Response Time	sec	Time elapsed between sending request for emails and receiving emails from email server in the network. This time includes signaling delay for the connection setup.
FTP		
FTP Download Response Time	sec	Time elapsed between sending a request and receiving the response packet. Measured from the time a client application sends a request to the server to the time it receives a response packet. Every response packet sent from a server to an FTP application is included in this statistic.
HTTP		
HTTP Page Response Time	sec	Specifies time required to retrieve the entire page with all the contained inline objects.
WLAN		
Throughput	bits/sec	Represents the total number of bits (in bits/sec) forwarded from wireless LAN layers to higher layers in all WLAN nodes of the network.

Table 9: Description of Global Statistics Collected

Node Statistics

Statistic		Description
Forwarding Memory Queue Size	bytes	<p>This statistic represents the current amount of forwarding memory used by packets, expressed in bytes.</p> <p>The forwarding memory holds packets awaiting processing, currently being processed, and, in the case of centralized processed, processed packets awaiting to sent out on an appropriate interface.</p>
Forwarding Memory Queuing Delay	bytes	<p>This statistic represents instantaneous measurements of packet waiting times in the forwarding memory.</p> <p>Packets remain in the forwarding memory until sent to an appropriate interface if in centralized mode or sent to an appropriate slot if in distributed mode.</p>
CPU Utilization		<p>This statistic reports the utilization, in percentage (%), of the 'Simple CPU'. The 'Simple CPU' is used to model the IP packet forwarding delays and application processing delays.</p> <p>This statistic measures the utilization of central CPU only. It does not measure the utilization of CPUs used for IP slot processing.</p>

Table 10: Description of Node Statistics Collected

Chapter 4: Simulation Results

We present some indicative results of our work in Sections 4.1 – 4.4 and we provide comments on them in Section 4.5.

4.1. 9 AP WLAN Deployment - Voice Traffic Sent in the Application Layer for data rates 6.5, 13, 19.5, 39, 65 Mbps

4.1.1. Only VoIP Calls Added

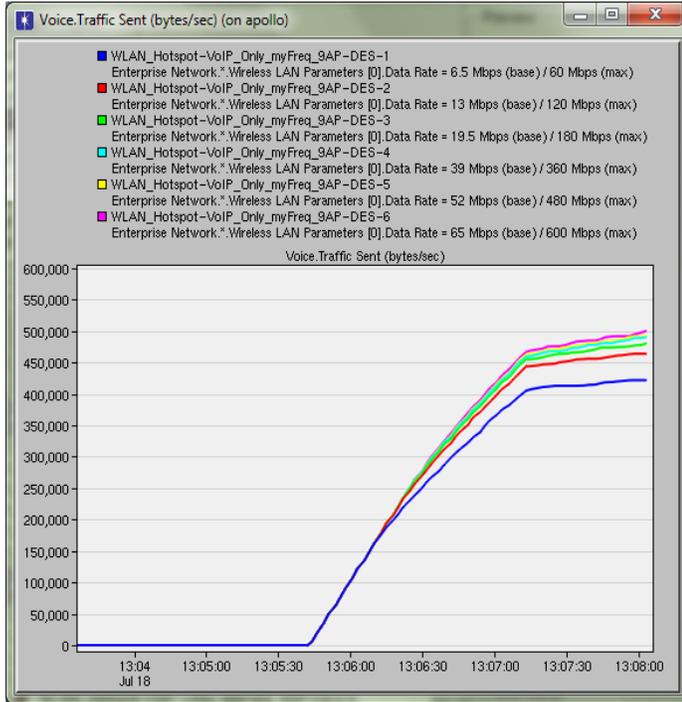


Figure 39: Voice Traffic Sent, 1 workstation per cell, 30 calls per workstation, IP Telephony Call



Figure 40: Voice Traffic Sent, 1 workstation per cell, 30 calls per workstation, PCM Quality Call

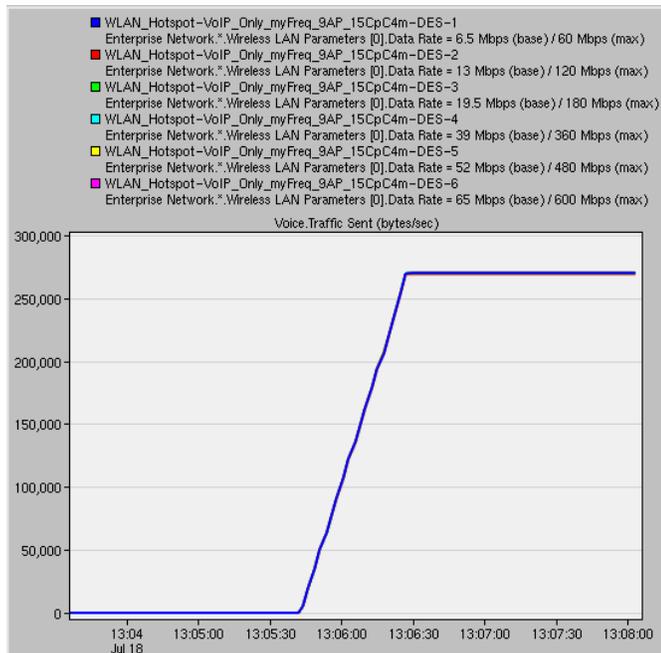


Figure 42: Voice Traffic Sent, 15 workstations per cell, 1 call per workstation, IP Telephony Call

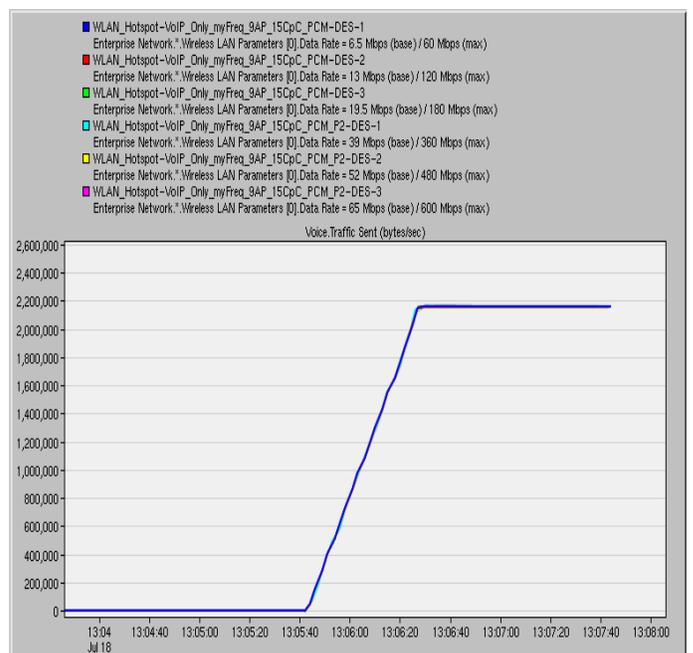


Figure 41: Voice Traffic Sent, 15 workstations per cell, 1 call per workstation, PCM Quality Call

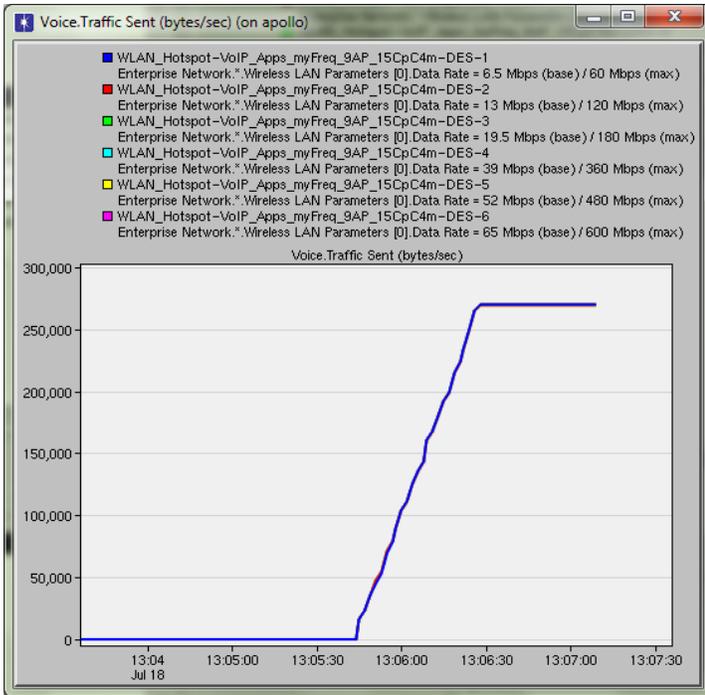


Figure 43: Voice Traffic Sent, IP Telephony Call, Applications added in the simulation

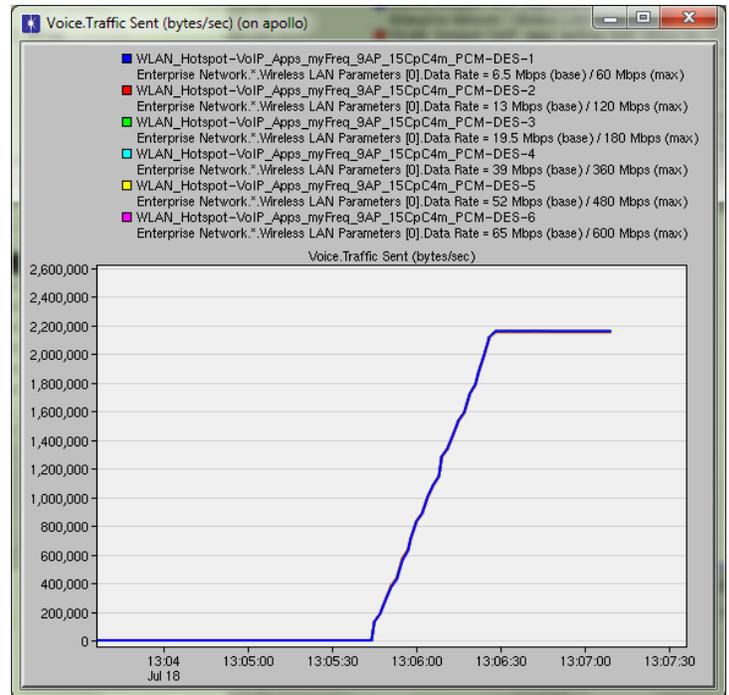


Figure 44: Voice Traffic Sent, PCM Quality Call, Applications added in the simulation

4.1.3. VoIP Calls, Applications and Video added

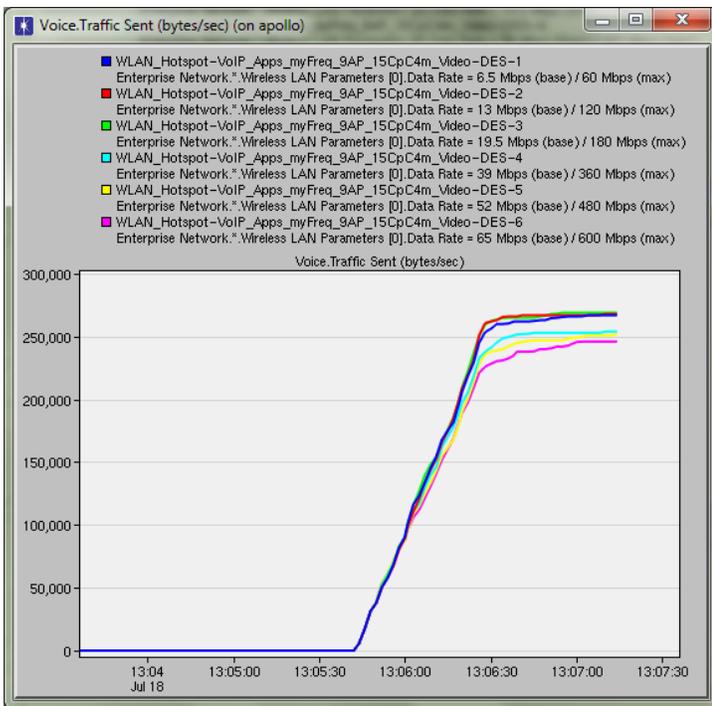


Figure 46: Voice Traffic Sent, IP Telephony Call, Video and Applications added in the simulation



Figure 45: Voice Traffic Sent, PCM Quality Call, Video and Applications added in the simulation

4.2. 9 AP WLAN Deployment-Voice Traffic Received in the Application Layer for data rates 6.5, 13, 19.5, 39, 65 Mbps

4.2.1. Only VoIP Calls Added



Figure 48: Voice Traffic Received, 1 workstation per cell, 30 calls per workstation, IP Telephony Call

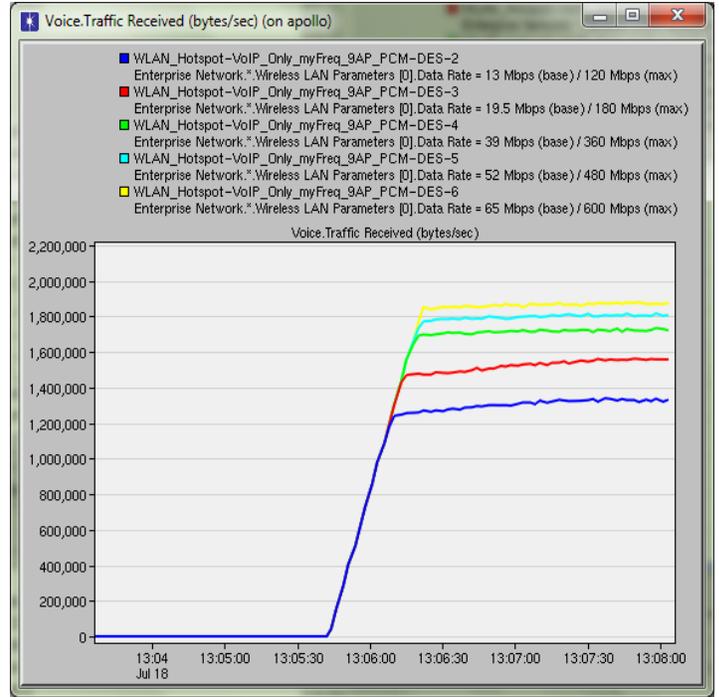


Figure 47: Voice Traffic Received, 1 workstation per cell, 30 calls per workstation, PCM Quality Call

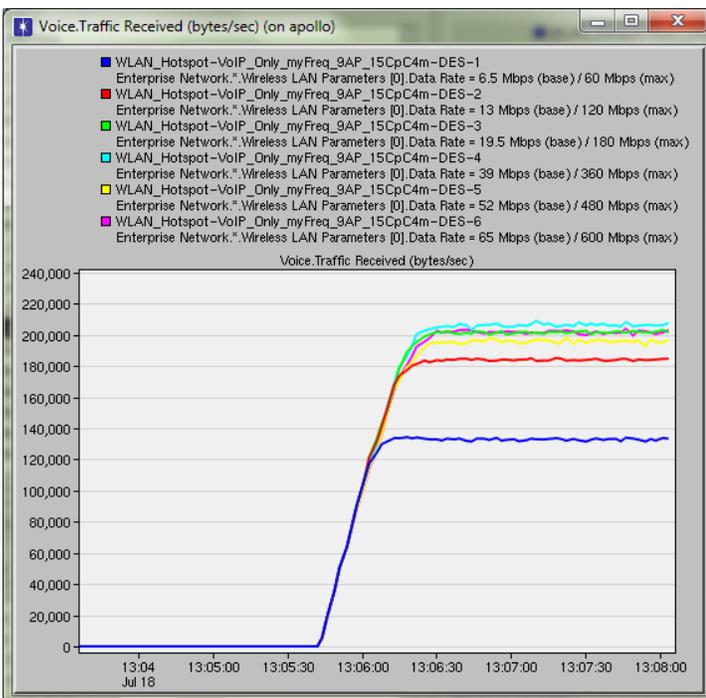


Figure 49: Voice Traffic Received, 15 workstations per cell, 1 call per workstation, IP Telephony Call

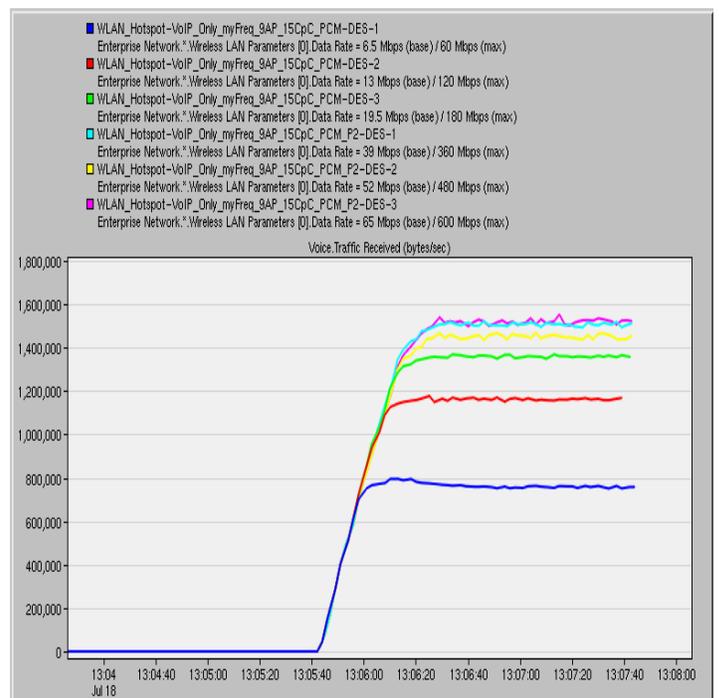


Figure 50: Voice Traffic Received, 15 workstations per cell, 1 call per workstation, PCM Quality Call

4.2.2. VoIP Calls and Applications added

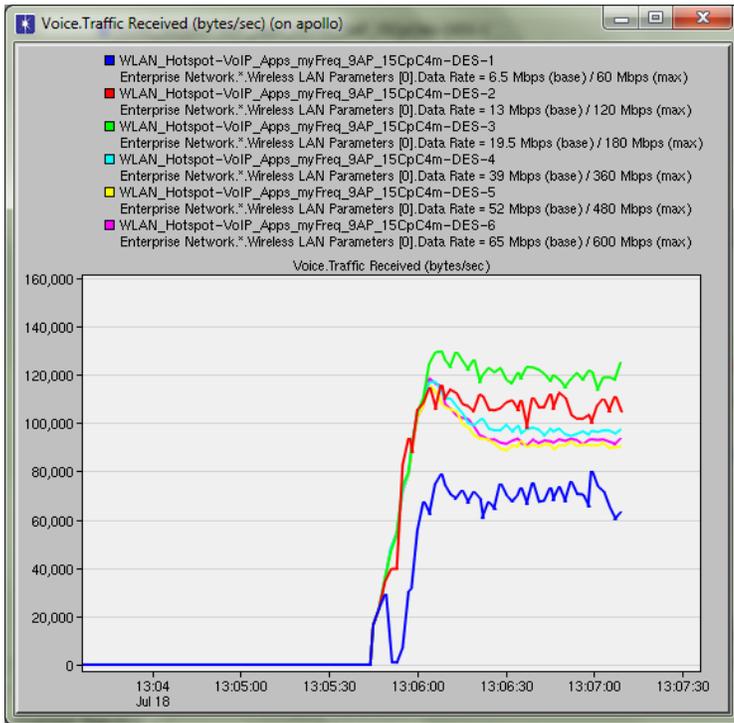


Figure 52: Voice Traffic Received, IP Telephony Call, Applications added in the simulation

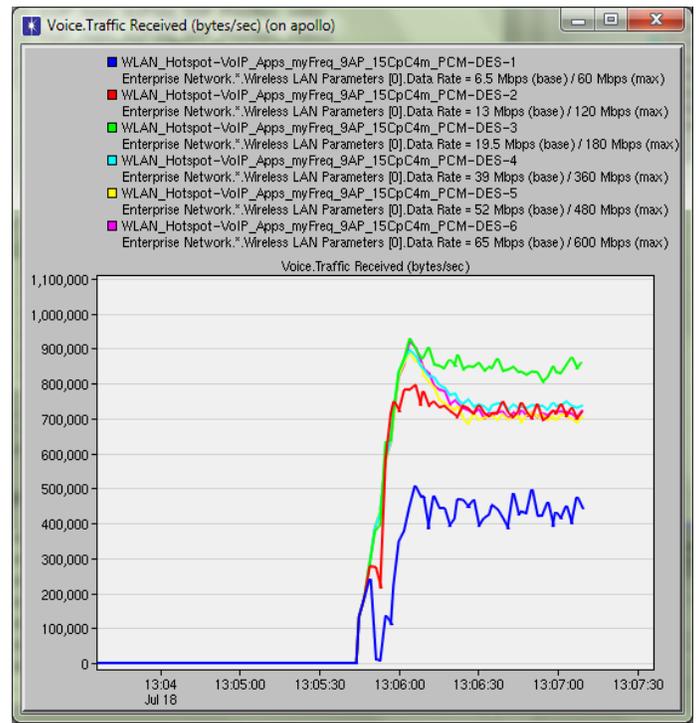


Figure 51: Voice Traffic Received, PCM Quality Call, Applications added in the simulation

4.2.3. VoIP Calls, Applications and Video added

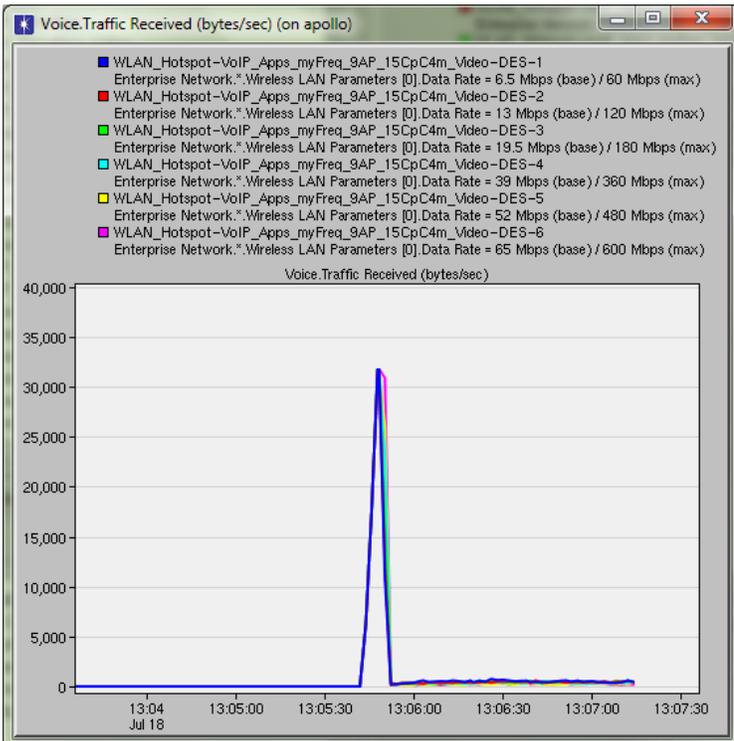


Figure 53: Voice Traffic Received, IP Telephony Call, Video Applications added in the simulation

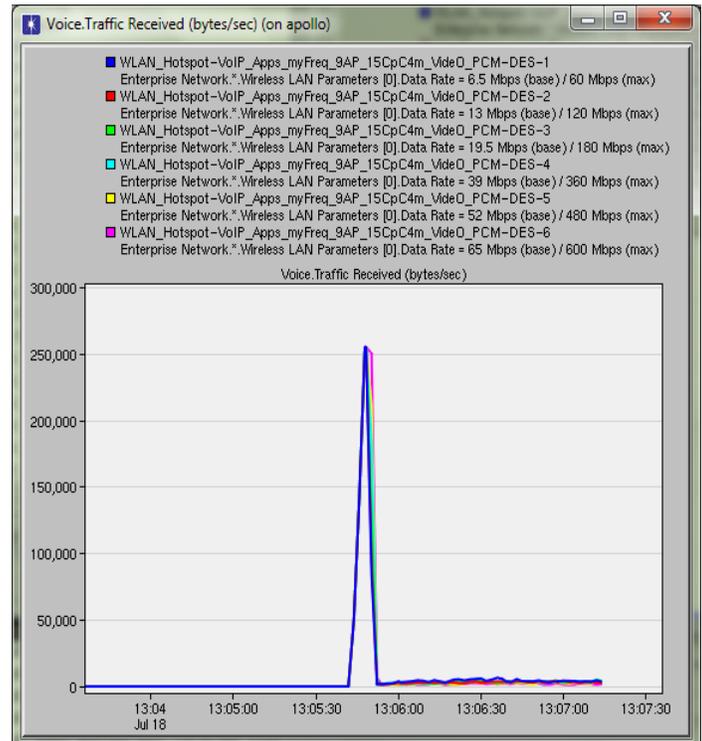


Figure 54: Voice Traffic Received, PCM Quality Call, Video and Applications added in the simulation

4.3. 16 AP WLAN Deployment - Voice Traffic Sent in the Application Layer for data rates 6.5, 13, 19.5, 39, 65 Mbps

4.3.1. Only VoIP Calls Added

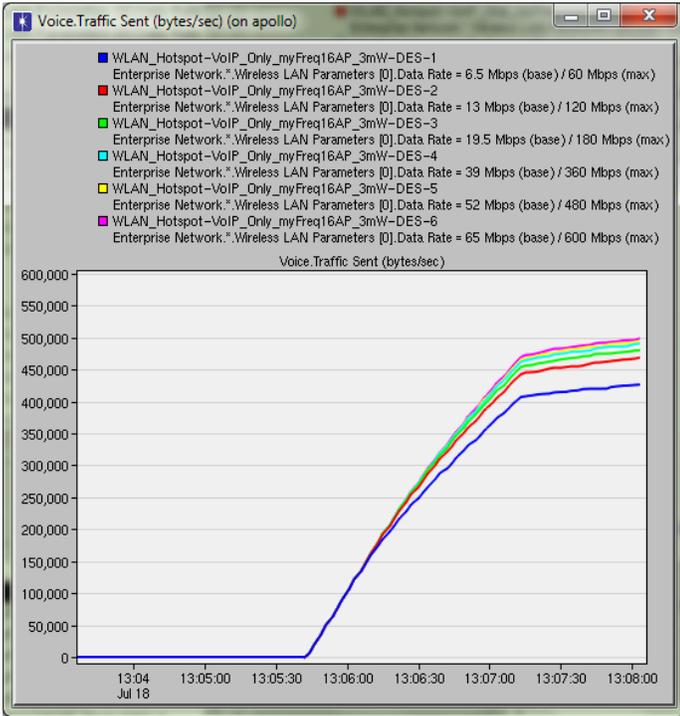


Figure 55: Voice Traffic Sent, 1 workstation per cell, 30 calls per workstation, IP Telephony Call



Figure 56: Voice Traffic Sent, 1 workstation per cell, 30 calls per workstation, PCM Quality Call

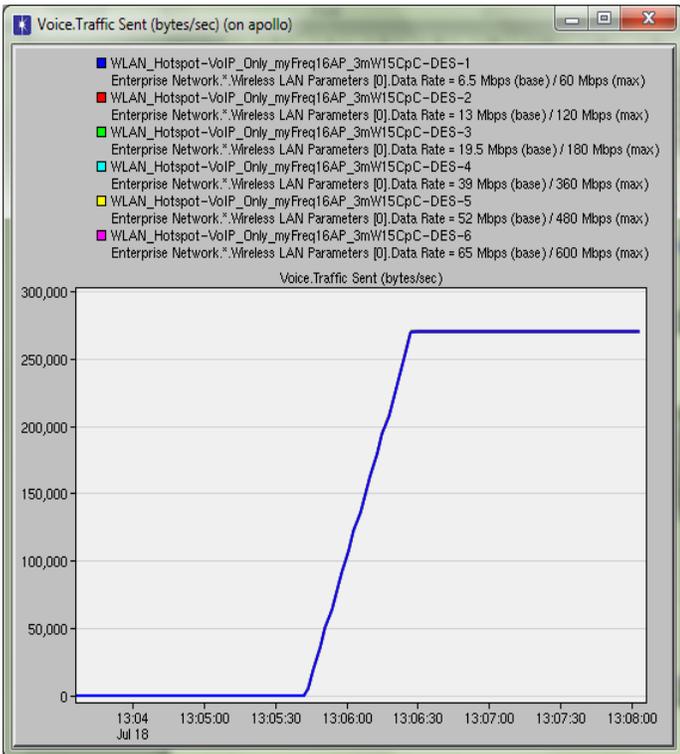


Figure 57: Voice Traffic Sent, 15 workstations per cell, 1 call per workstation, IP Telephony Call

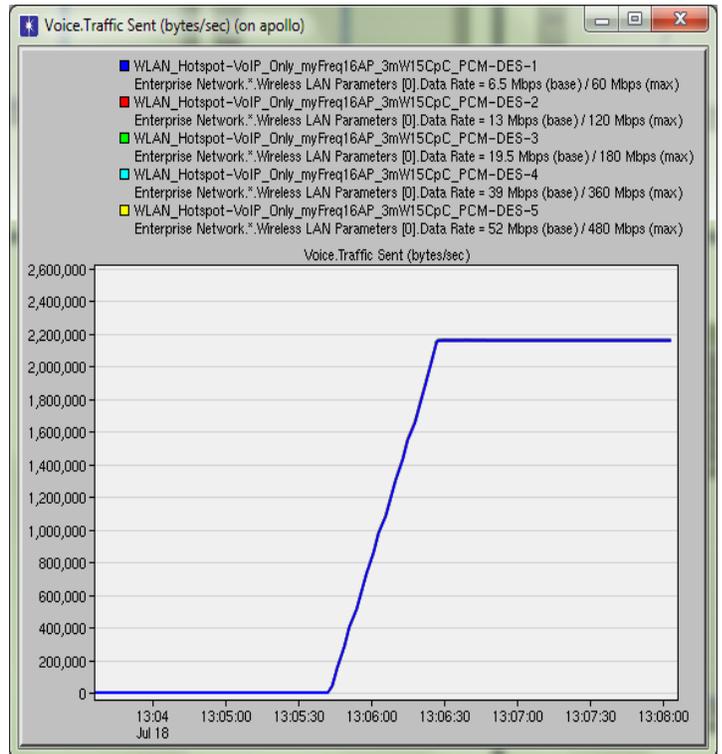


Figure 58: Voice Traffic Sent, 15 workstations per cell, 1 call per workstation, PCM Quality Call

4.3.2. VoIP Calls and Applications added

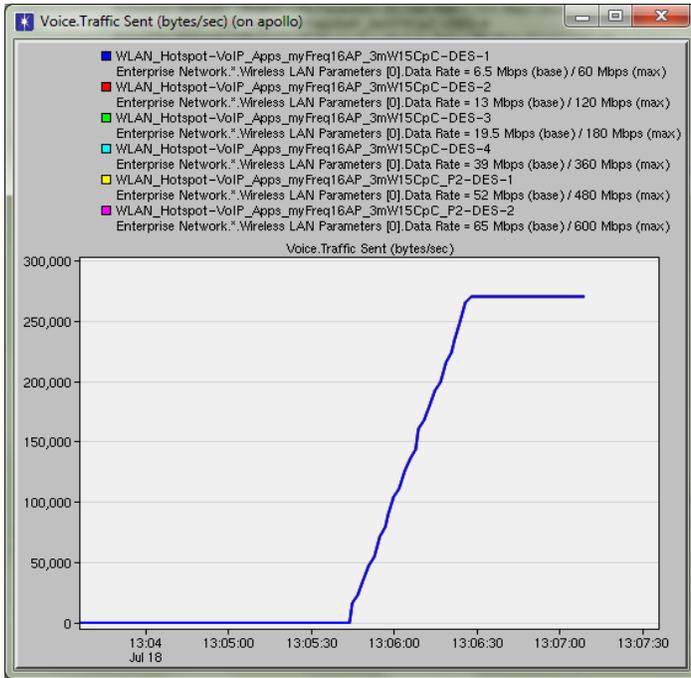


Figure 59: Voice Traffic Sent, IP Telephony Call, Applications added in the simulation

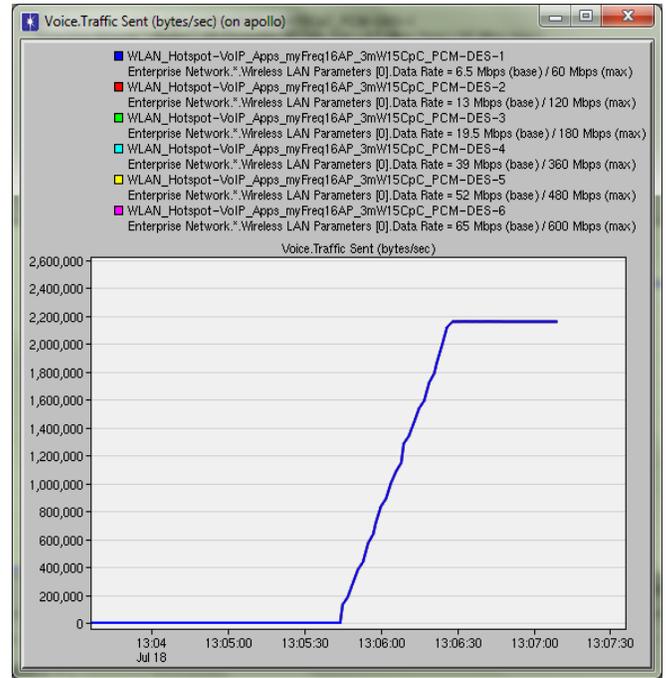


Figure 60: Voice Traffic Sent, PCM Quality Call, Applications added in the simulation

4.3.3. VoIP Calls, Applications and Video added

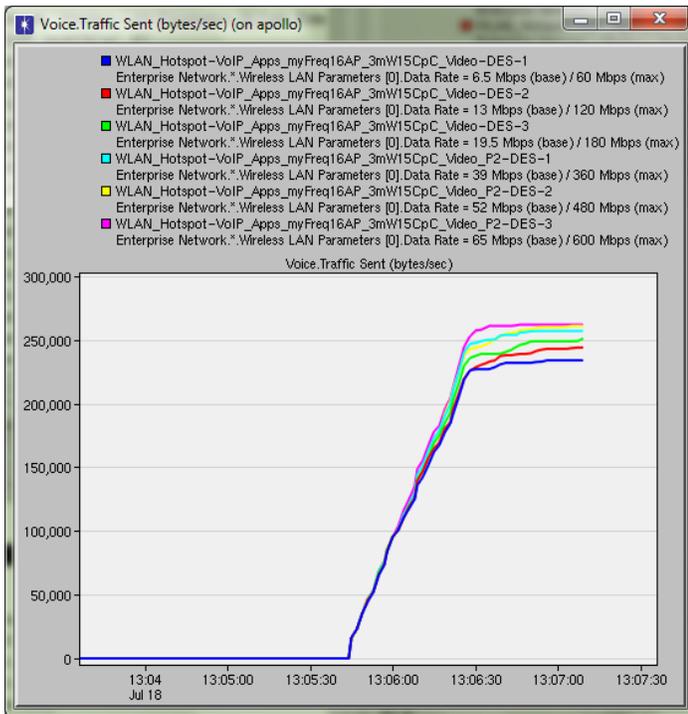


Figure 61: Voice Traffic Sent, IP Telephony Call, Applications and Video added in the simulation

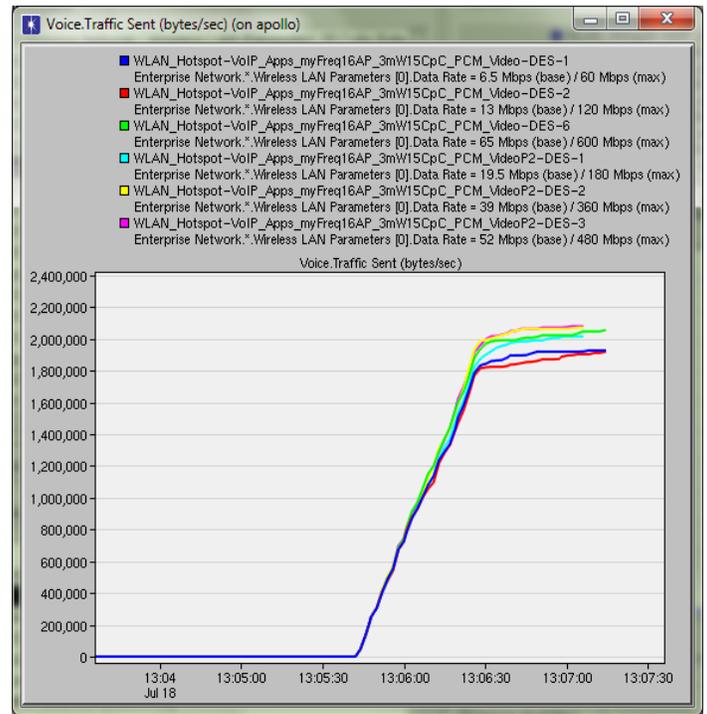


Figure 62: Voice Traffic Sent, PCM Quality Call, Applications and Video added in the simulation

4.4. 16 AP WLAN Deployment-Voice Traffic Received in the Application Layer for data rates 6.5, 13, 19.5, 39, 65 Mbps

4.4.1. Only VoIP Calls Added

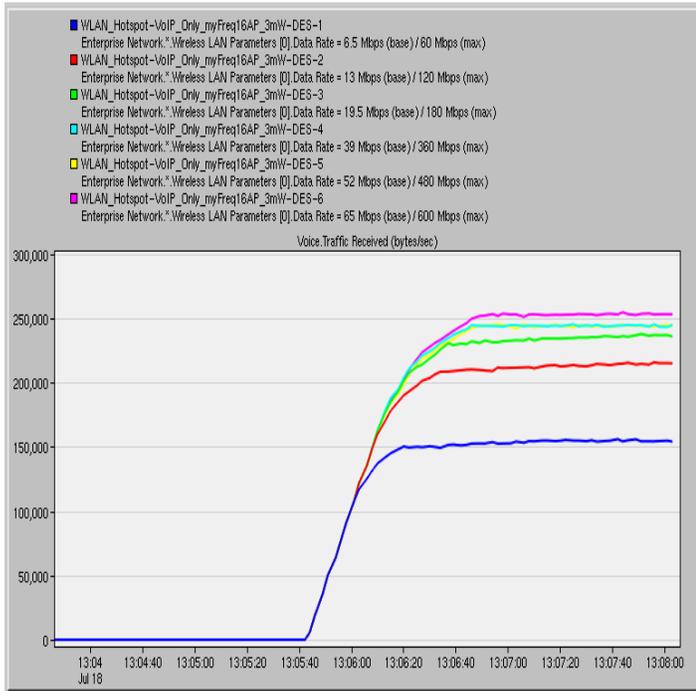


Figure 63: Voice Traffic Received, 1 workstation per cell, 30 calls per workstation, IP Telephony Call

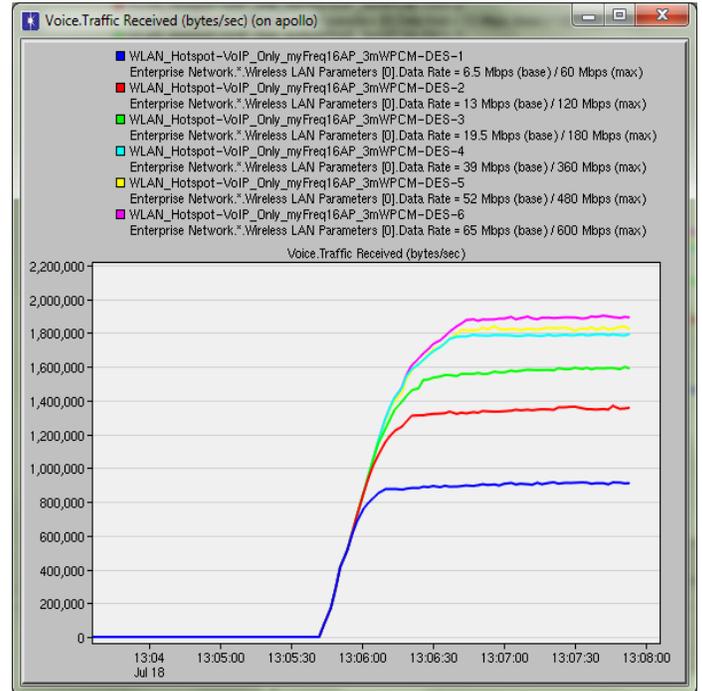


Figure 64: Voice Traffic Received, 1 workstation per cell, 30 calls per workstation, PCM Quality Call

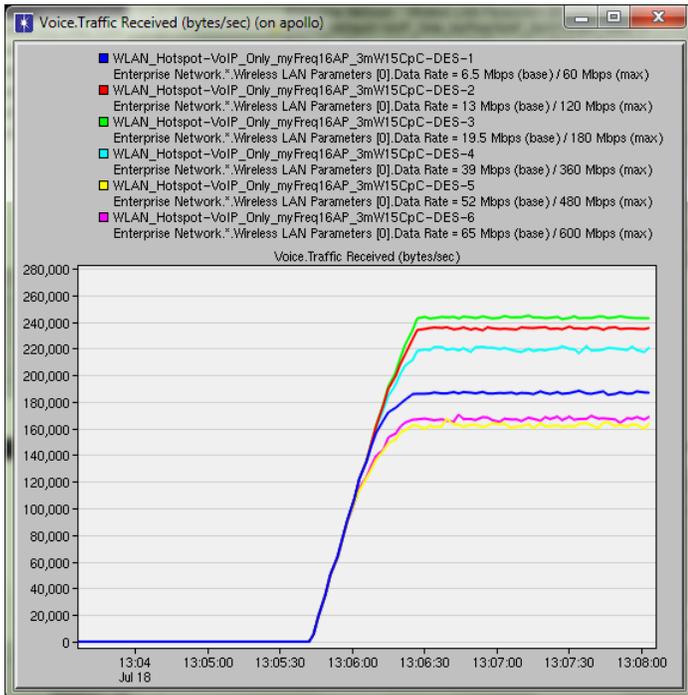


Figure 65: Voice Traffic Received, 15 workstations per cell, 1 call per workstation, IP Telephony Call

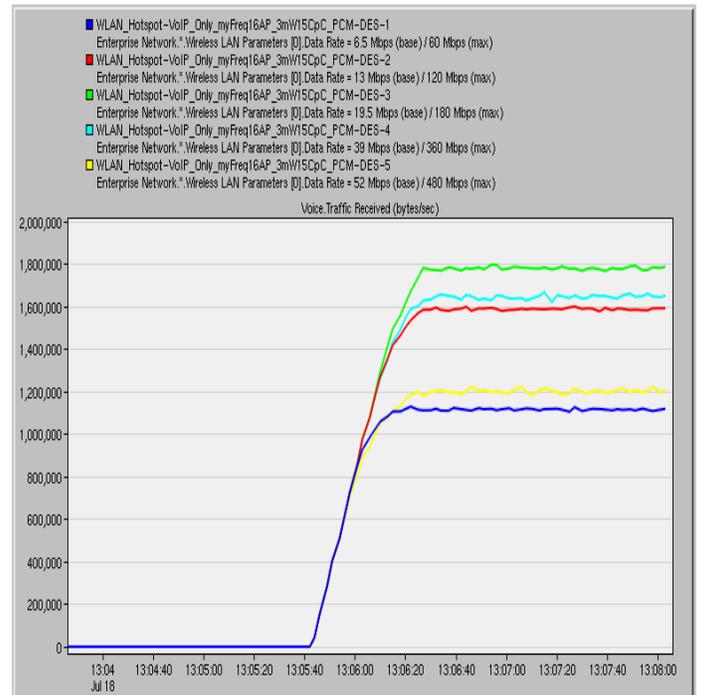


Figure 66: Voice Traffic Received, 15 workstations per cell, 1 call per workstation, PCM Quality Call

4.4.2. VoIP Calls and Applications added

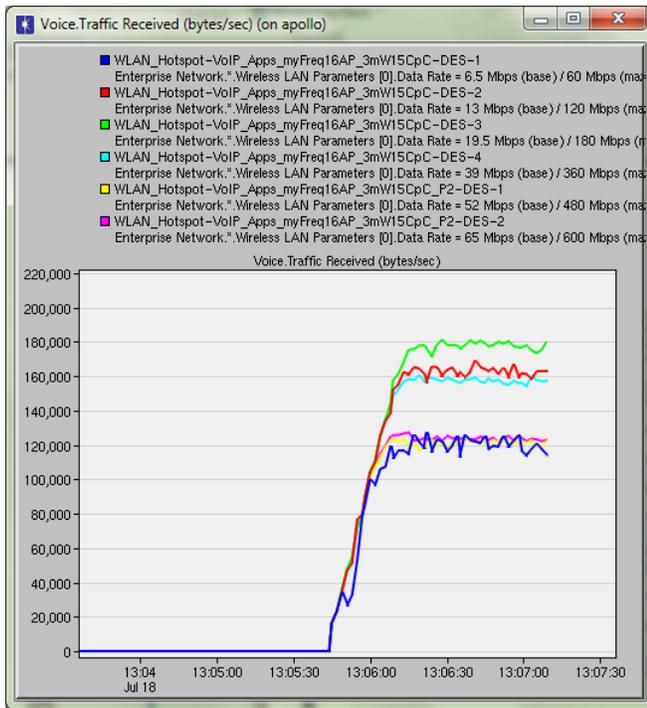


Figure 67: Voice Traffic Received, IP Telephony Call, Applications added in the simulation

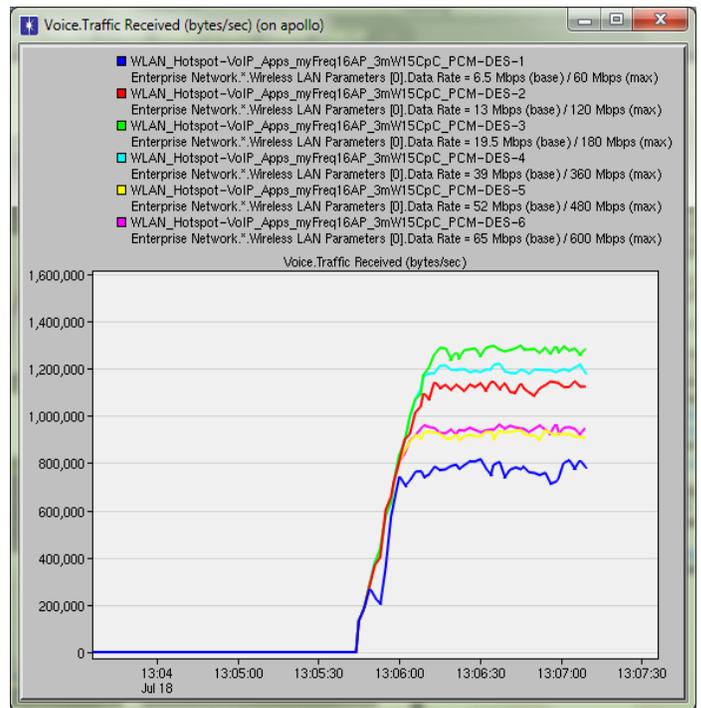


Figure 68: Voice Traffic Received, PCM Quality Call, Applications added in the simulation

4.4.3. VoIP Calls, Applications and Video added

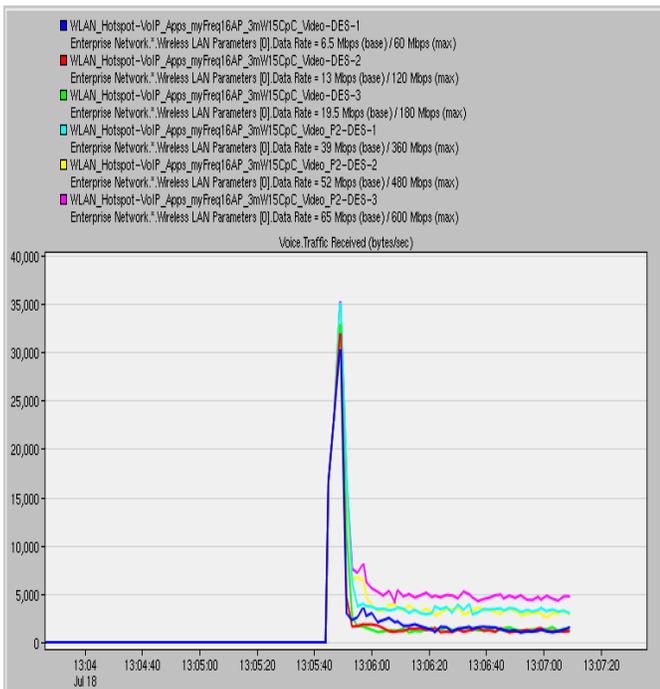


Figure 69: Voice Traffic Received, IP Telephony Call, Video and Applications added in the simulation

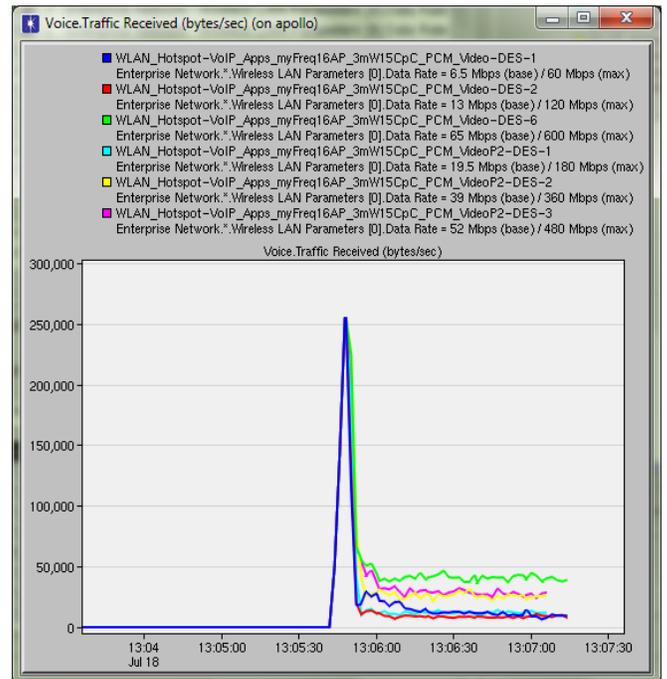


Figure 70: Voice Traffic Received, PCM Quality Call, Video and Applications added in the simulation

4.5. Comments

In order to interpret the traffic sent (bytes/sec), we should recall the definition of the statistic we collect as well as the traffic characteristics of the calls we periodically add. The “Traffic Sent” statistic presented in Sections 4.1, 4.2 shows the average number of bytes or packets per second submitted to the transport layers by all voice applications in the network; Hence, it represents all the traffic sent from the workstations to the VoIP Gateway and the traffic sent from the VoIP Gateway to the workstations respectively.

For the IP Telephony voice calls, the Voice Payload Size is 10 bytes and 100 pps are transmitted. So the average number of bytes per second submitted to the transport layer by the application layer for a call is 1000 bytes. For both directions, 2000 bytes/sec need to be sent for a single call.

Therefore, when we add 30 calls to a workstation and we have 9 workstations, 270 calls are added to the network. Thus, 540000 bytes/sec should be submitted to the transport layer by the application layer for 270 calls for both directions. Similarly, for 15 calls/cell, 270000 bytes/sec are required to be sent for 135 calls for both directions (Figure 42).

In the case of PCM Quality voice calls, the Voice Payload Size is 80 bytes and again 100 pps are transmitted. The average number of bytes per second submitted to the transport layer by the application layer for a call is 8000 bytes. For both directions, 16000 bytes/sec need to be sent for a single call.

Therefore, when we add 30 calls to every one of our 9 workstations, 4320000 bytes/sec should be submitted to the transport layer by the application layer for 270 calls for both directions. Similarly, for 15 calls/cell – 2160000 bytes/sec are required to be sent for 135 calls for both directions (Figure 41).

In Figures 39, 40, 55 and 56, where 30 calls are added to a workstation, for the 9 AP and the 16 AP WLAN deployments, we can observe that the aforementioned upper bounds are not reached and the traffic sent differs for various data rates.

As previously mentioned in 3.6, the bound for network bandwidth is reflected in observing a mismatch between VoIP traffic being sent and received. Following this pattern the Network Call Capacity is determined in Table 11 for the experiments where 30 Calls are added Per Workstation, and we have 1 Workstation Per Cell.

Network Capacity – Average Number of Calls Per Cell				
Data Rate (Mbps)	IP Telephony (G.729a)		PCM Quality (G.711)	
	9 AP WLAN	16 AP WLAN	9 AP WLAN	16 AP WLAN
6.5	8.05	8.33	-	6.25
13	11.1	11.66	8.68	9.02
19.5	12.2	13.3	10.41	10.41
36	13.2	13.6	11.08	12.15
52	13.3	13.6	12.5	12.5
65	13.8	13.8	12.84	12.84

Table 11: Network Call Capacity for 30 Calls added Per Workstation, 1 Workstation Per Cell

Here we should refer to Table 7 (page 43) and the bandwidth that each call requires. 59.2 Kbps for an IP Telephony call and 115.2 Kbps for a PCM quality call.

We observe that for lower data rates, fewer calls are supported.

- ✓ For 30 IP Telephony calls a total of 1,776 Mbps needs to be transmitted by each cell’s AP when the 30th call is added and 15,984 Mbps are submitted to the IP Backbone.
- ✓ For 30 PCM Quality calls a total of 3,456 Mbps needs to be transmitted by each cell’s AP when the 30th call is added and 31,104 Mbps are submitted to the IP Backbone.

In both cases, theoretically, the traffic to be transmitted can be handled by the network even by the lowest data rate of the 6.5Mbps and the links are underutilized. But by lowering the bit rate the AP client radius is extended, and the client co-channel interference radius increases. Hence, there is the possibility of retransmissions (especially when there is more than one workstation in a cell and collisions occur) and we

should not forget the demands for non-data traffic by the network such as acknowledgment (ACK) action management frames and beacon frames.

Furthermore, we notice that when we have a single workstation in a cell, the performance is almost the same for the 9 AP and the 16 AP deployment. This is expected since one workstation is still connecting to one AP, therefore we cannot see the benefits of the microcell deployment strategy, yet. We observe that the two deployments have similar performance, something that means that the microcell deployment strategy is successful (as we achieve the same maximum calls with a lower transmission power and more APs), and minor changes may be due to the different distances of the workstations from the APs. The workstations have the exact same coordinates in the 9 AP and the 16 AP WLAN deployments. The only thing that changes is the workstation's BSS ID, according to the cell it "falls in" each time, in order to associate with the correct AP.

It is also clear by comparing the Figures in Sections 4.2 and 4.4 that the number of calls supported does not significantly change according to the call type. Approximately 2-3 more calls can be accommodated using IP Telephony calls, despite the considerably lower payload of the 10 bytes, against the 80 bytes of the PCM Quality Calls (for a Codec Sample Interval equal to 10 ms for both call types, which is the standard G.711 (PCM Quality) Codec Sample Interval time).

This is because of the excessive overhead caused by packet headers and the incompressible protocol timings DIFS, SIFS. Reducing the coding rate (using G.729 – IP Telephony) and increasing the channel capacity (using higher data rate) has only a slight impact on VoIP call capacity. A more significant gain in capacity could be obtained by increasing the packet size. By doubling or tripling the packetization delay, the payload is increased in the same proportions leading to a much lower relative overhead. The voice capacity of an 11 Mbps WLAN is almost doubled when the Codec Sample Interval is increased to 20 ms [19].

Network Capacity - Average Number of Calls Per Cell				
Data Rate (Mbps)	IP Telephony (G.729a)		PCM Quality (G.711)	
	9 AP WLAN	16 AP WLAN	9 AP WLAN	16 AP WLAN
6.5	6.7	8.8	5.2	6.6
13	9.7	13	7.6	9.7
19.5	10.8	13.6	9.0	12.5
36	11.1	11.1	9.7	11.1
52	9.72	6.6	9.0	6.59
65	9.72	6.6	9.7	-

Table 12: Network Call Capacity for 1 Call added Per Workstation, 15 Workstations Per Cell

In the experiment with the 30 calls per workstation the maximum IP Telephony and PCM Quality calls supported are 14 and 13 respectively. As more voice connections join the network, the service rates of the APs and mobile stations decrease because more collisions may occur. This is also confirmed by the results shown in Table 12 as for 15 workstations in a cell with one call per workstation the maximum calls do not exceed 11 and 9.7 for the IP Telephony and PCM Quality call type in the 9 AP WLAN. Another significant observation is that when we don't have collisions the maximum number of calls is supported for the highest data rates (65 Mbps for both the 9 AP and 16 AP WLAN). On the other hand with collisions the maximum number of calls is supported for the 39 Mbps data rate in the 9 AP WLAN. A microcell architecture leads to higher performance because there are fewer client devices connecting to each access point. This allows each client device radio to consume a greater percentage of the access point's capacity and avoid collisions with other radios. Therefore, performance can be significantly higher. Table 12 shows that the maximum number of calls is increased from 10.8 to 13.6 for the IP Telephony and from 9.7 to 12.5 for the PCM Quality VoIP call in the 16 AP WLAN (where we have implemented this architecture), however this occurs for the 19.5 Mbps data rate, in the 16 AP WLAN case.

Regarding the results presented in Sections 4.1 – 4.4 for the cases where applications and videoconference are added, it is clear from Table 13 that the call capacity dramatically decreases. This applies to both the 9 AP and the 16 AP WLAN deployments when we add applications and videoconference videos on each one of the 15 workstations per cell. When we add applications in the 9 AP WLAN deployment many

delays, abnormalities and significant mismatches occur for the voice traffic sent and received, due to the much higher number of collisions.

The results are better and the plots are smoother for the 16 AP WLAN deployment, compared to the ones in the 9 AP WLAN deployment, thus we extract from Figures 67 – 68 and their comparison with Figures 59 – 60 the following average number of calls per cell. Notice that the call dropping is between 3-4 calls per cells when the applications are deployed. When we add the videoconference applications their bandwidth demands are so high that barely one call can be added in the WLAN for both the 9 AP and the 16 AP WLAN deployments. The **peak value** of calls per cell under the presence of videoconference (on top of the other applications) is also presented in Table 13.

Network Capacity - Average Number of Calls Per Cell						
Data Rate (Mbps)	IP Telephony (G.729a), 16 AP WLAN			PCM Quality (G.711), 16 AP WLAN		
	VoIP Only	VoIP And Applications	VoIP, Applications, and Video	VoIP Only	VoIP And Applications	VoIP, Applications, and Video
6.5	8.8	5.5	1.6	6.6	4.8	1.7
13	13	8.8	1.6	9.7	6.9	1.7
19.5	13.6	10	1.6	12.5	9	1.7
36	11.1	8.6	1.6	11.1	8.3	1.7
52	6.6	6.6	1.6	6.59	6.2	1.7
65	6.6	6.6	1.6	-	6.2	1.7

Table 13: Network Call Capacity for the 16 AP WLAN, under the presence of applications and videoconference, 1 Call added Per Workstation, 15 Workstations Per Cell

4.6. 9 AP WLAN Deployment - Voice Packet End To End Delay for data rates 6.5, 13, 19.5, 39, 65 Mbps

As mentioned earlier, a voice call is very sensitive to delay and the International Telecommunication Union (ITU) has recommended that one-way end-to-end delay should be no greater than 150 ms for good voice quality, and up to 400 ms can be considered acceptable but not desirable. In the following Figures, the statistic presented is the total voice packet delay and it records data from **all** the nodes in the network. Therefore, we can conclude that, based on these simulation results, the voice packet end to end delay has the acceptable value until the maximum number of voice calls that can be supported by the network is reached.

After that and as more calls, applications or videos are trying to access the medium and transmit the voice packet end-to-end delays increase, causing unacceptable delays.

4.6.1. Only VoIP Calls Added

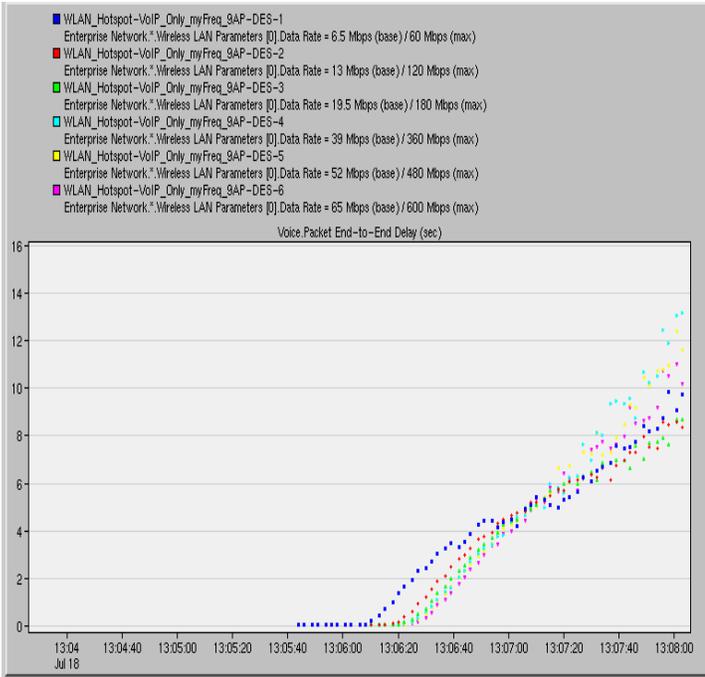


Figure 72: Voice Packet End To End Delay, 1 workstation per cell, 30 calls per workstation, IP Telephony Call

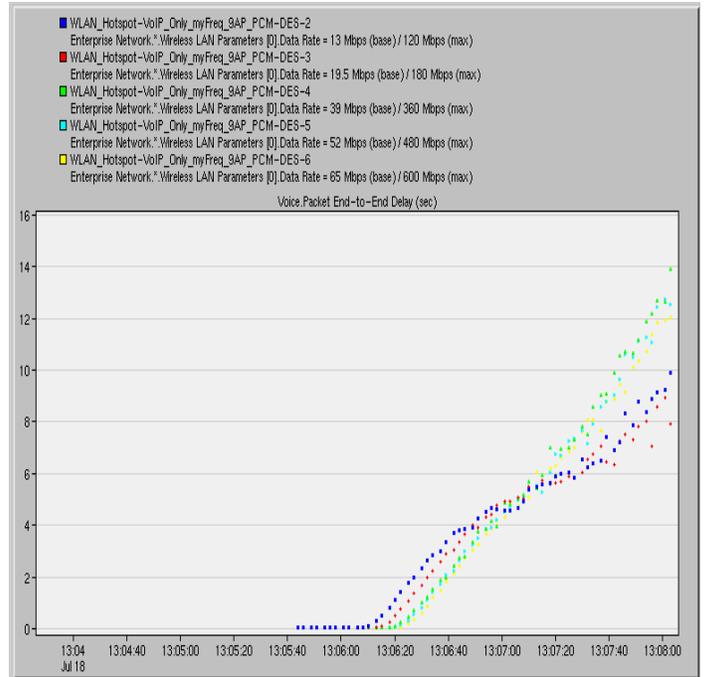


Figure 71: Voice Packet End To End Delay, 1 workstation per cell, 30 calls per workstation, PCM Quality Call

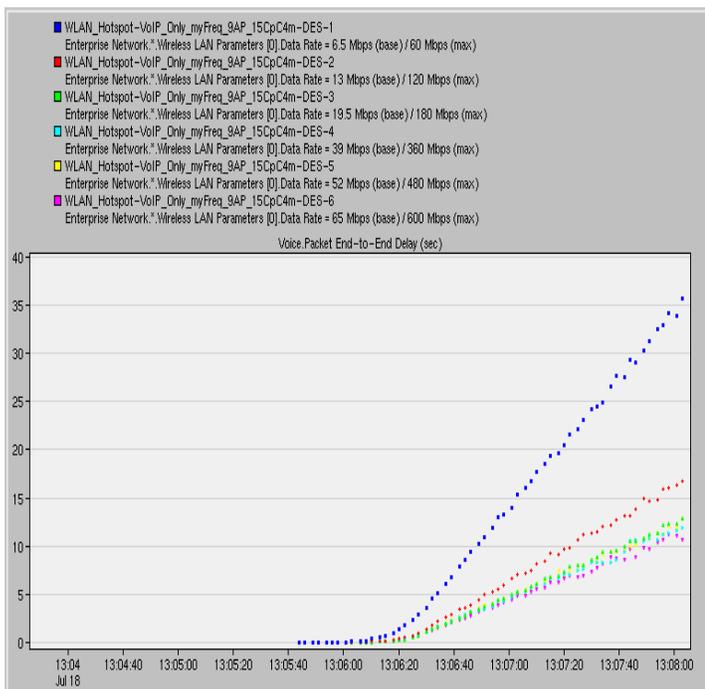


Figure 73: Voice Packet End To End Delay, 15 workstations per cell, 1 call per workstation, IP Telephony Call

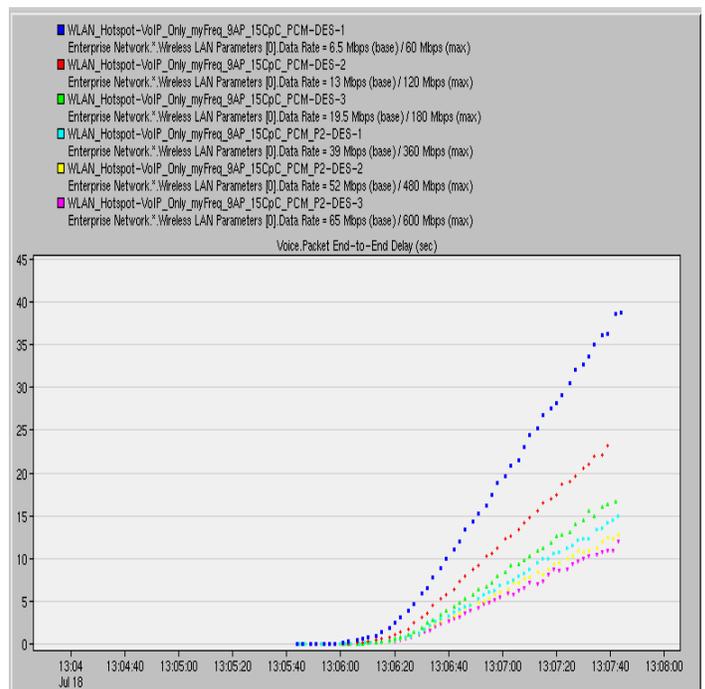


Figure 74: Voice Packet End To End Delay, 15 workstations per cell, 1 call per workstation, PCM Quality Call

4.6.2. VoIP Calls and Applications Added

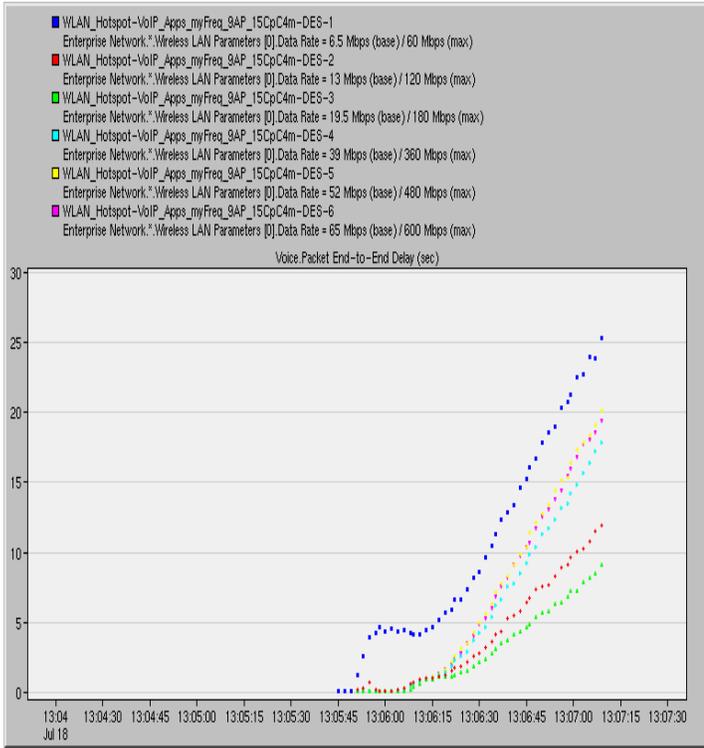


Figure 76: Voice Packet End To End Delay, IP Telephony Call, Applications added in the simulation

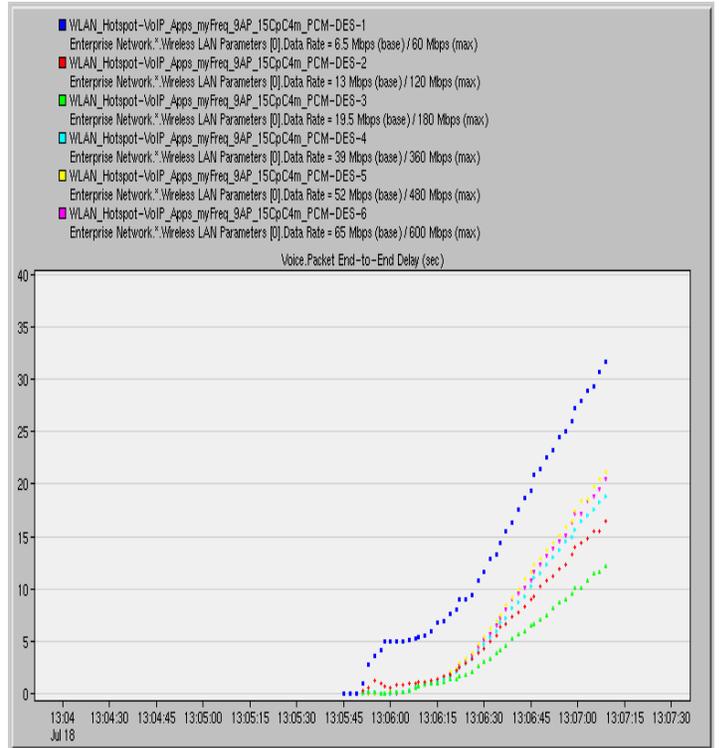


Figure 75: Voice Packet End To End Delay, PCM Quality Call, Applications added in the simulation

4.6.3. VoIP Calls, Applications and Videos Added

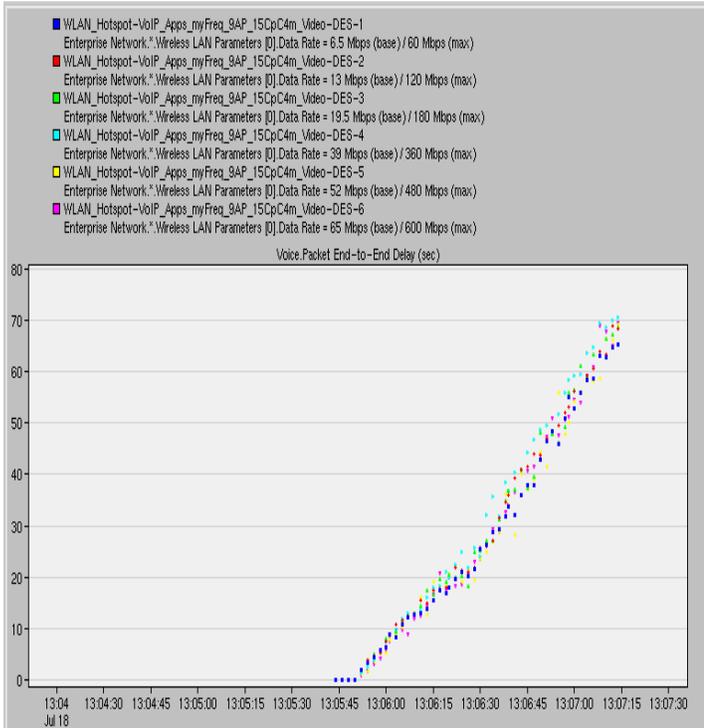


Figure 78: Voice Packet End To End Delay, IP Telephony Call, Video and Applications added in the simulation

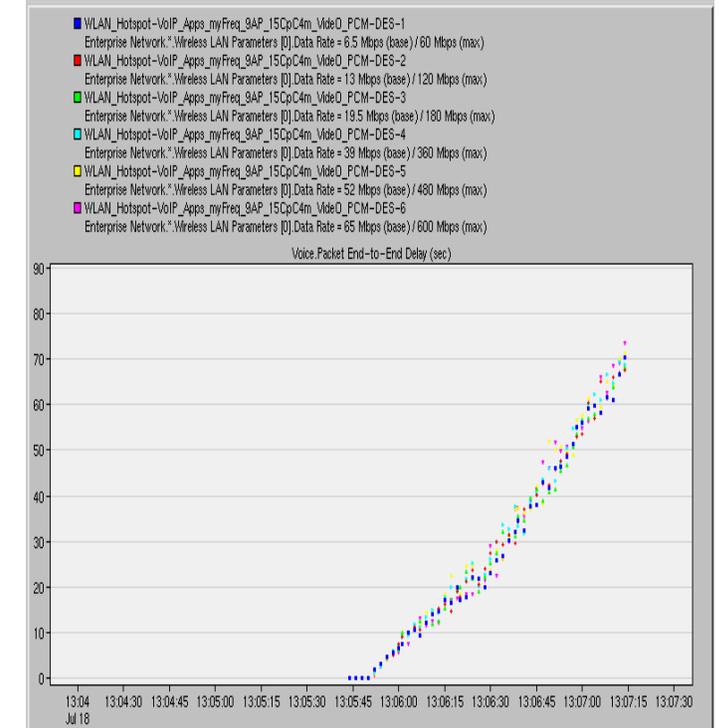


Figure 77: Voice Packet End To End Delay, PCM Quality Call, Video and Applications added in the simulation

4.7. 16 AP WLAN Deployment - Voice Packet End To End Delay for data rates 6.5, 13, 19.5, 39, 65 Mbps

4.7.1. Only VoIP Calls Added

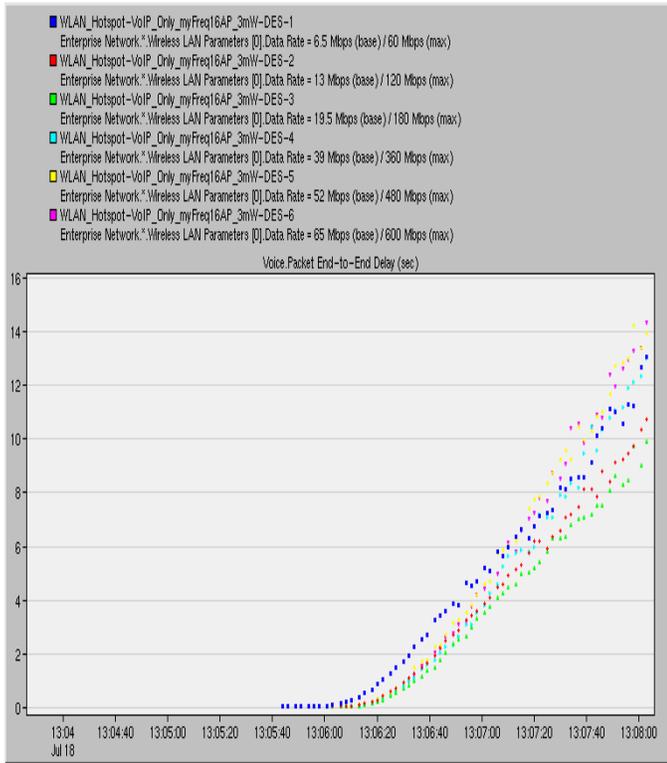


Figure 80: Voice Packet End To End Delay, 1 workstation per cell, 30 calls per workstation, IP Telephony Call

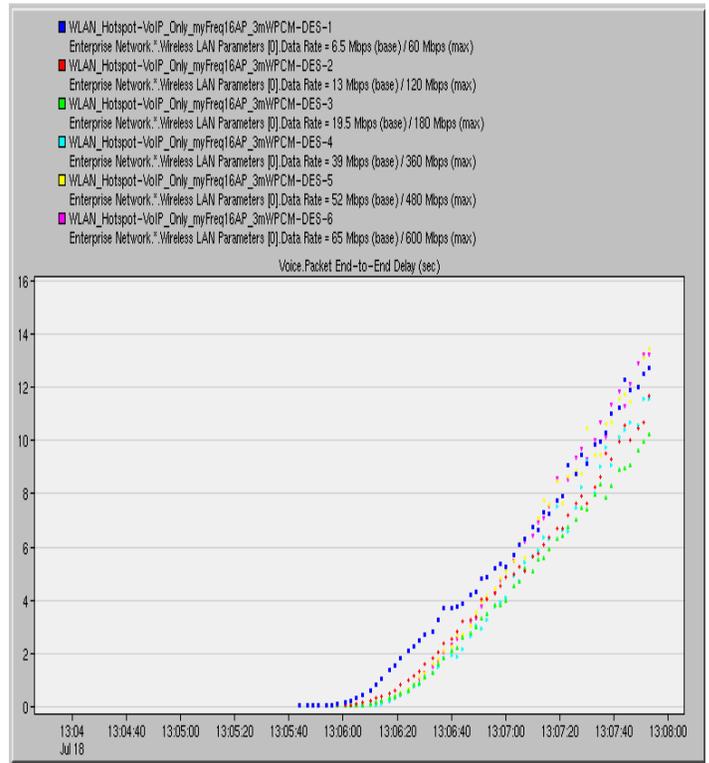


Figure 79: Voice Packet End To End Delay, 1 workstation per cell, 30 calls per workstation, PCM Quality Call

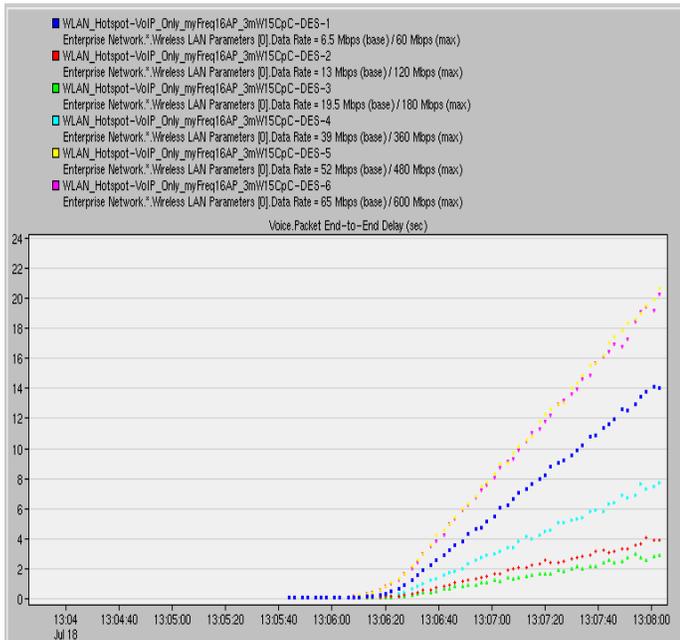


Figure 82: Voice Packet End To End Delay, 15 workstations per cell, 1 call per workstation, IP Telephony Call

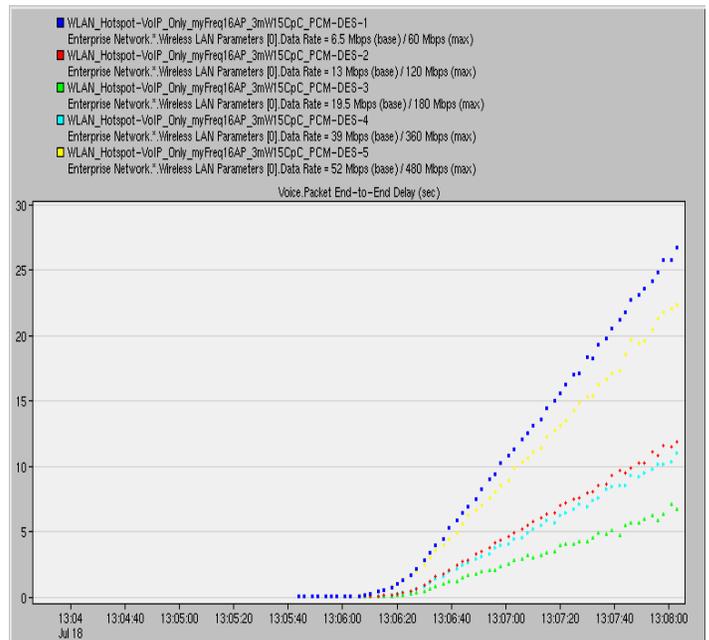


Figure 81: Voice Packet End To End Delay, 15 workstations per cell, 1 call per workstation, PCM Quality Call

4.7.2. VoIP Calls and Applications Added

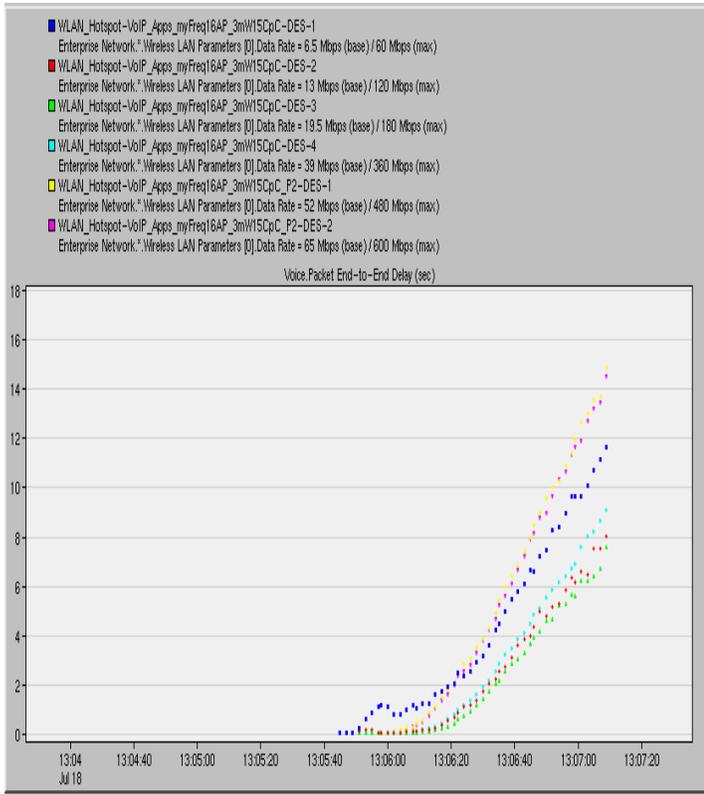


Figure 84: Voice Packet End To End Delay, IP Telephony Call, Applications added in the simulation

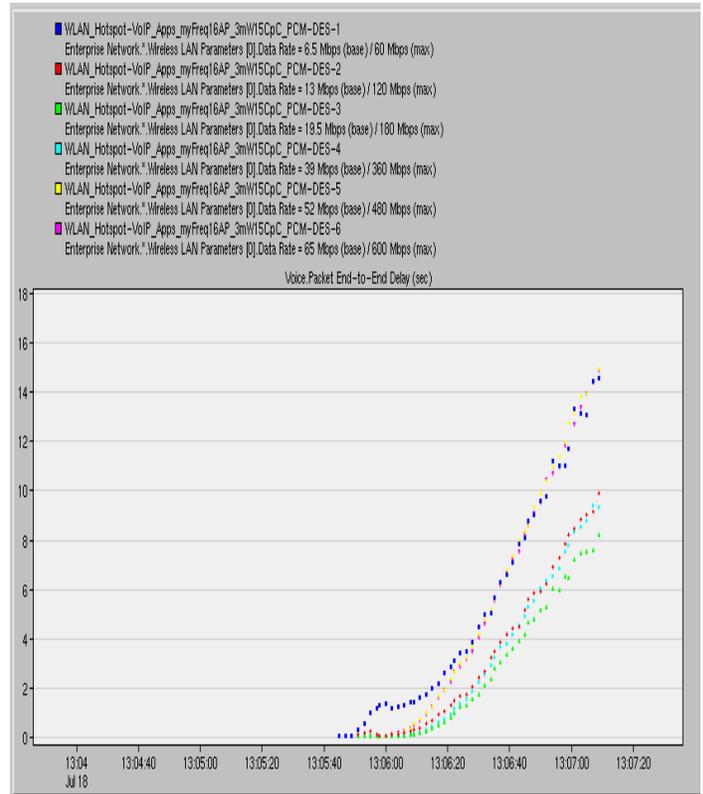


Figure 83: Voice Packet End To End Delay, PCM Quality Call, Applications added in the simulation

4.7.3. VoIP Calls, Applications and Videos Added

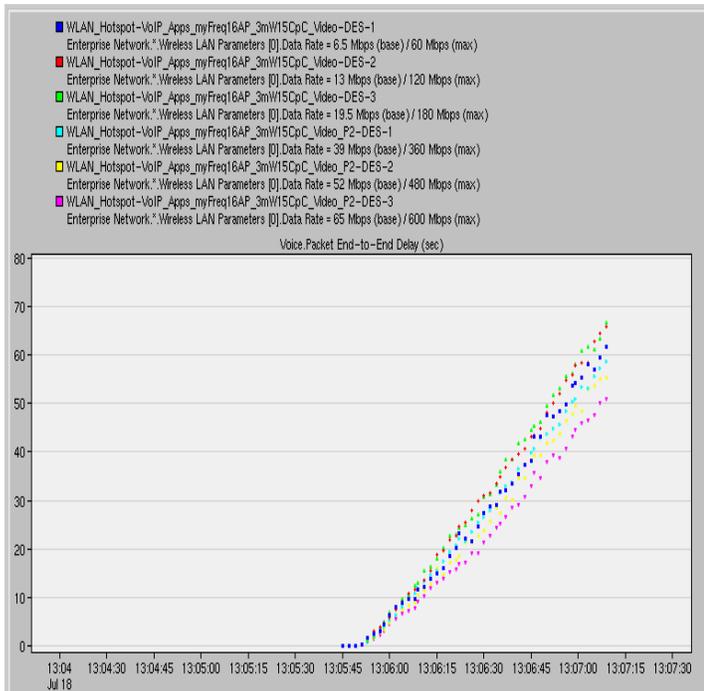


Figure 86: Voice Packet End To End Delay, IP Telephony Call, Video and Applications added in the simulation

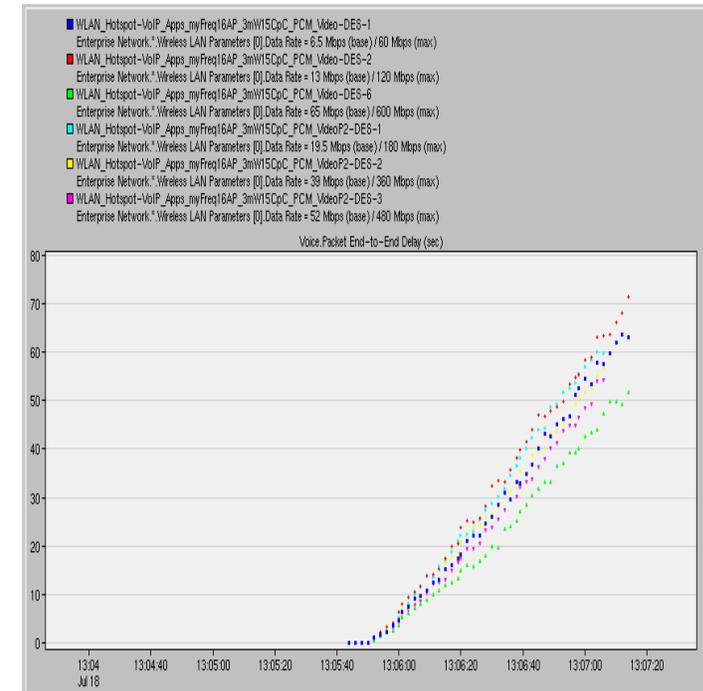


Figure 85: Voice Packet End To End Delay, PCM Quality Call, Video and Applications added in the simulation

4.8. 9 AP WLAN Deployment - Throughput for data rates 6.5, 13, 19.5, 39, 65 Mbps

The “Throughput” statistic represents the total number of bits/sec forwarded from wireless LAN MAC layers to higher layers in all the WLAN nodes of the network . Note that WLAN nodes are the APs and the workstations in our deployments, but not the VoIP Gateway, which is an Ethernet workstation or the IP backbone router and the switches.

So, for the deployments with VoIP applications only added in them, we can see this statistic as the (VoIP traffic received x 2) expressed in bits/sec, as the traffic passes through the APs. Also, the traffic forwarded by the WLAN MAC, contains the MAC Header, which is 320 bits.

For example, the Voice Traffic Received in the 9 AP WLAN deployment with 1 workstation per cell and 30 calls per workstation (IP Telephony Calls) the voice traffic received is 250000 bytes/sec for both directions, for 65 Mbps data rate. Thus, 125000 bytes/sec pass through the VoIP Gateway to the APs and from the APs to the workstations. VoIP Traffic of 125000 bytes/sec corresponds to 125 IP Telephony calls. If we calculate the bandwidth the 125 IP Telephony calls require including the MAC header we have:

$$[125 * (\text{MAC HEADER} + \text{IP Telephony payload size})] * 100 \text{ packets/sec} = \\ = [125 * (320 + 80)] * 100 = 5 \text{ Mbps},$$

5Mbps*2=10Mbps which is the equivalent result presented in Figure 88 (65 Mbps data rate)

Observe that when the applications are added on the experiments, the throughput does not always increase. On the contrary, it either remains stable or it decreases. When the applications are added they have the highest priority in the WLAN, their priority is set to Reserved and the VoIP calls’ priority is set to Best Effort. When the videoconference is added, its priority is set to Streaming Multimedia, which is lower than the applications’ (Reserved) priority and higher than the VoIP calls’ priority, but still as we can see from the throughput results it manages to dominate in the throughput results because of its bandwidth – hungry nature.

4.8.1. Only VoIP Calls Added

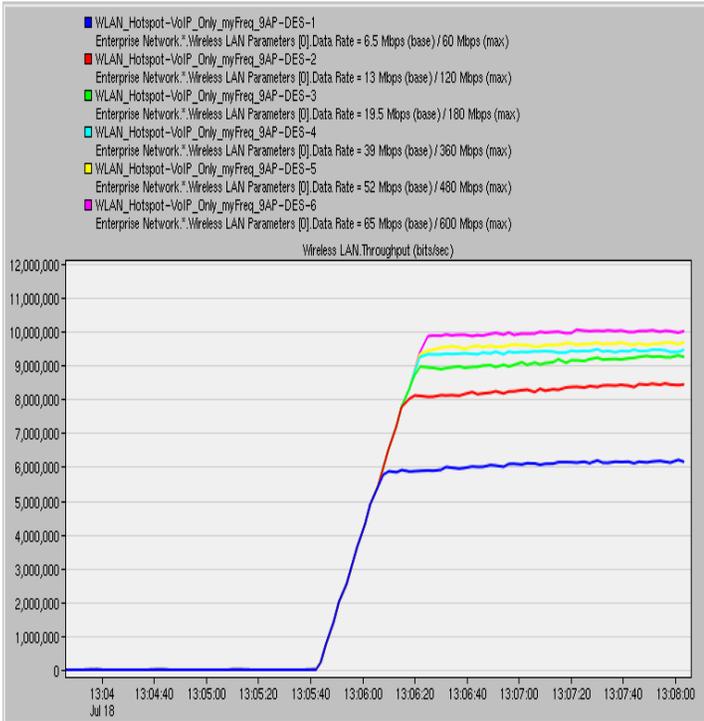


Figure 88: Throughput, 1 workstation per cell, 30 calls per workstation, IP Telephony Call

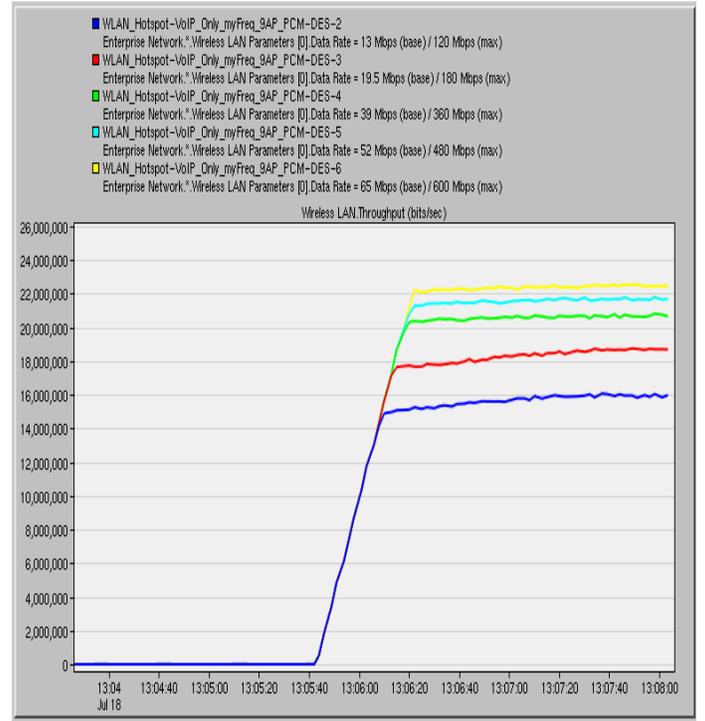


Figure 87: Throughput, 1 workstation per cell, 30 calls per workstation, PCM Quality Call

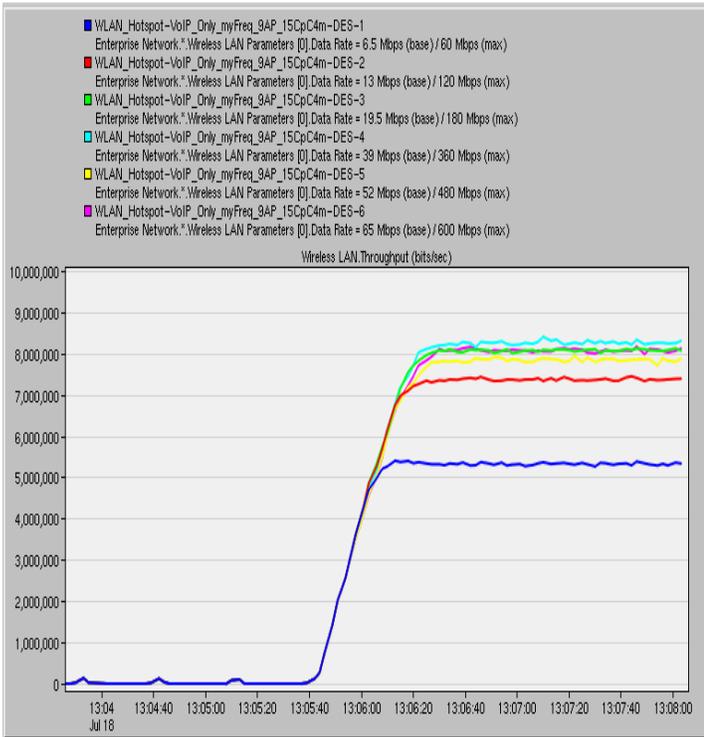


Figure 89: Throughput, 15 workstations per cell, 1 call per workstation, IP Telephony Call

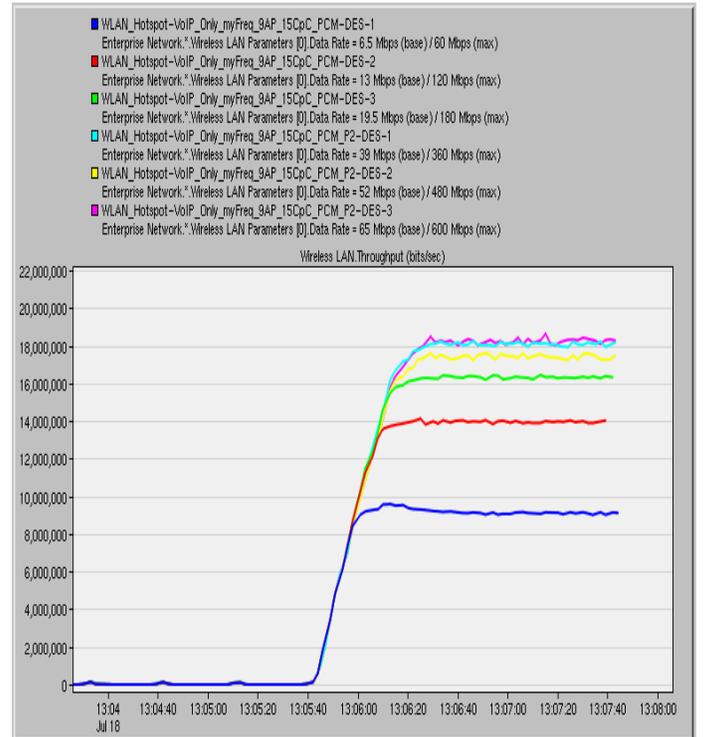


Figure 90: Throughput, 15 workstations per cell, 1 call per workstation, PCM Quality Call

4.8.2. VoIP Calls and Applications Added

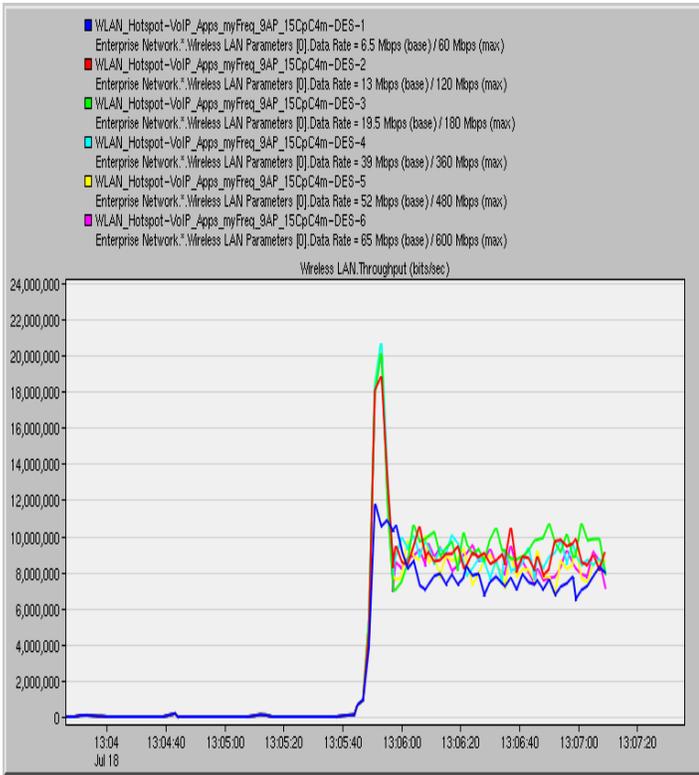


Figure 92: Throughput, IP Telephony Call, Applications added in the simulation

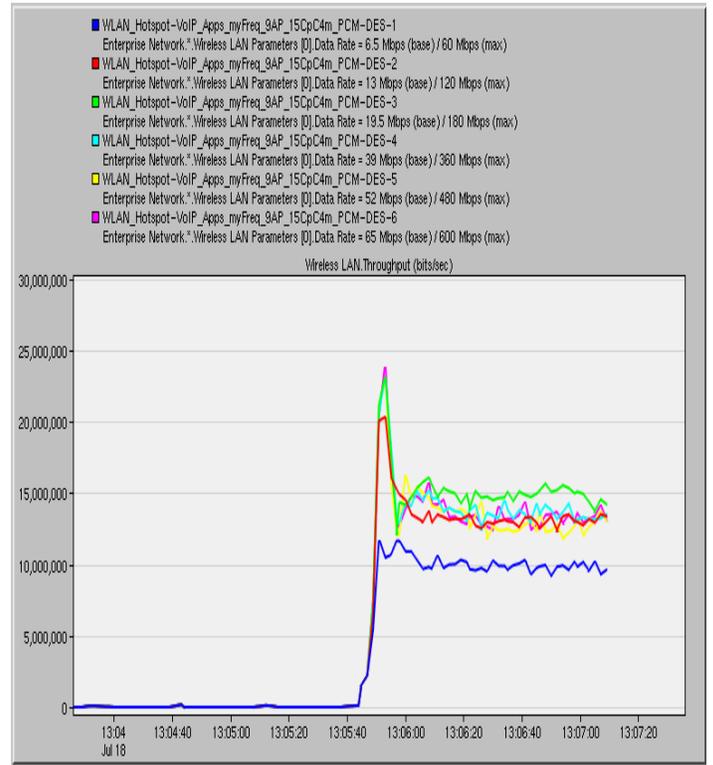


Figure 91: Throughput, PCM Quality Call, Applications added in the simulation

4.8.3. VoIP Calls, Applications and Videos Added

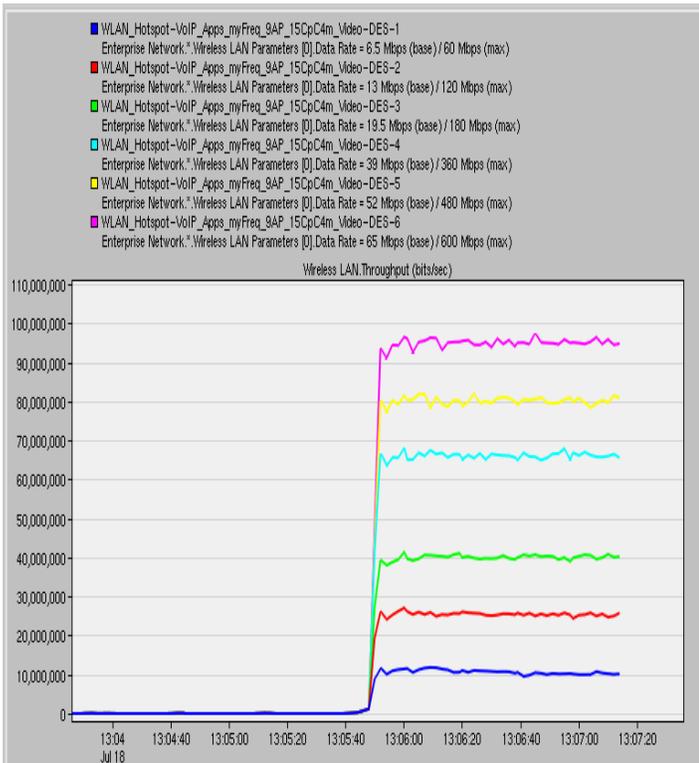


Figure 93: Throughput, IP Telephony Call, Video and Applications added in the simulation

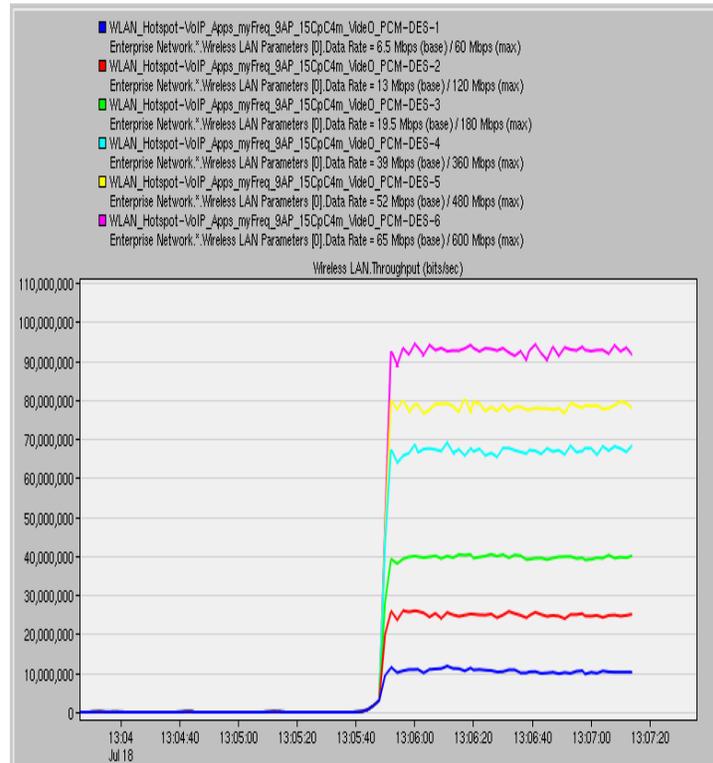


Figure 94: Throughput, PCM Quality Call, Video and Applications added in the simulation

4.9. 16 AP WLAN Deployment - Throughput for data rates 6.5, 13, 19.5, 39, 65 Mbps

4.9.1. Only VoIP Calls Added

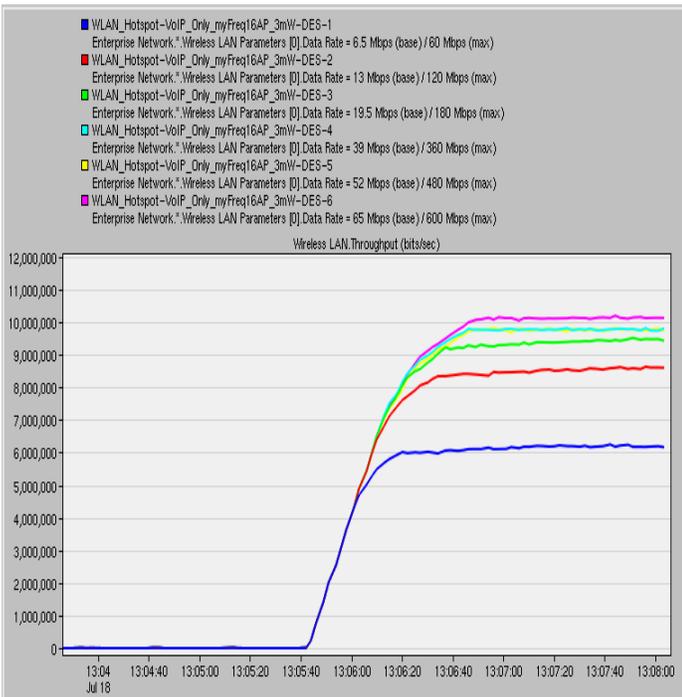


Figure 96: Throughput, 1 workstation per cell, 30 calls per workstation, IP Telephony Call

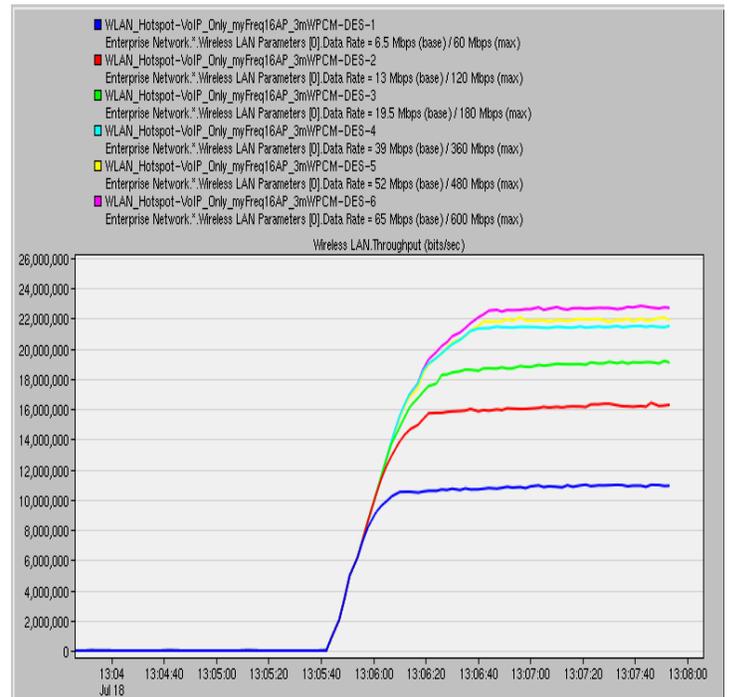


Figure 95: Throughput, 1 workstation per cell, 30 calls per workstation, PCM Quality Call

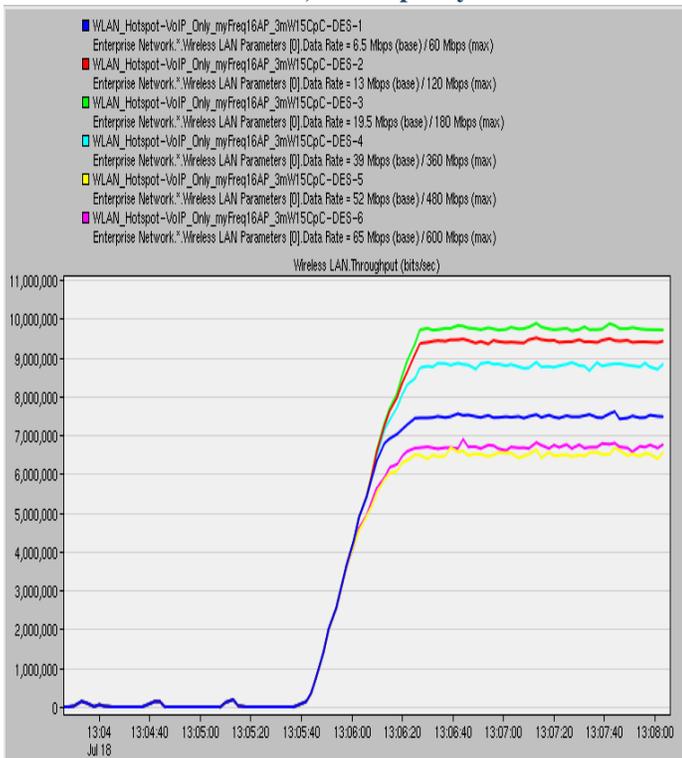


Figure 97: Throughput, 15 workstations per cell, 1 call per workstation, IP Telephony Call

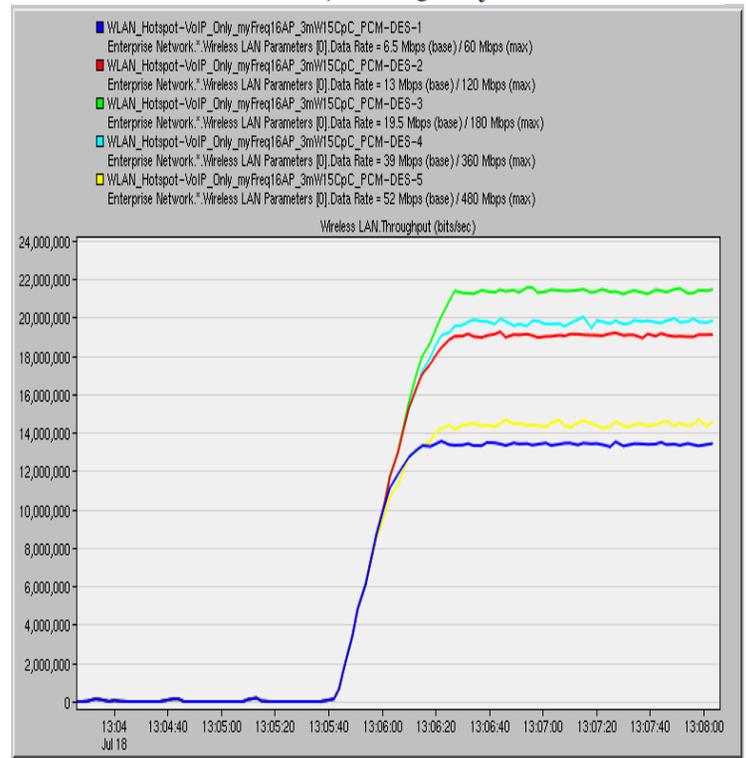


Figure 98: Throughput, 15 workstations per cell, 1 call per workstation, PCM Quality Call

4.9.2. VoIP Calls and Applications Added

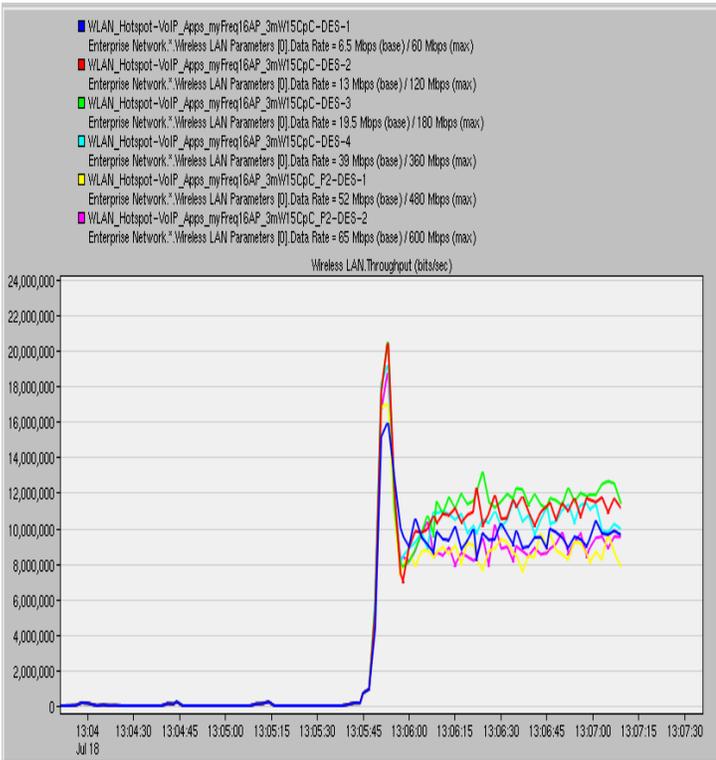


Figure 100: Throughput, IP Telephony Call, Applications added in the simulation

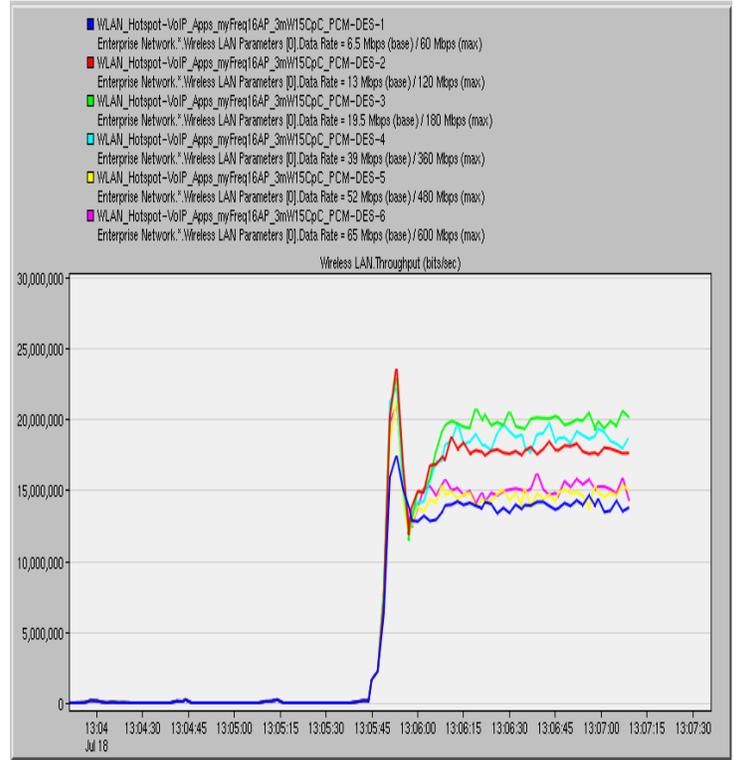


Figure 99: Throughput, PCM Quality Call, Applications added in the simulation

4.9.3. VoIP Calls, Applications and Videos Added

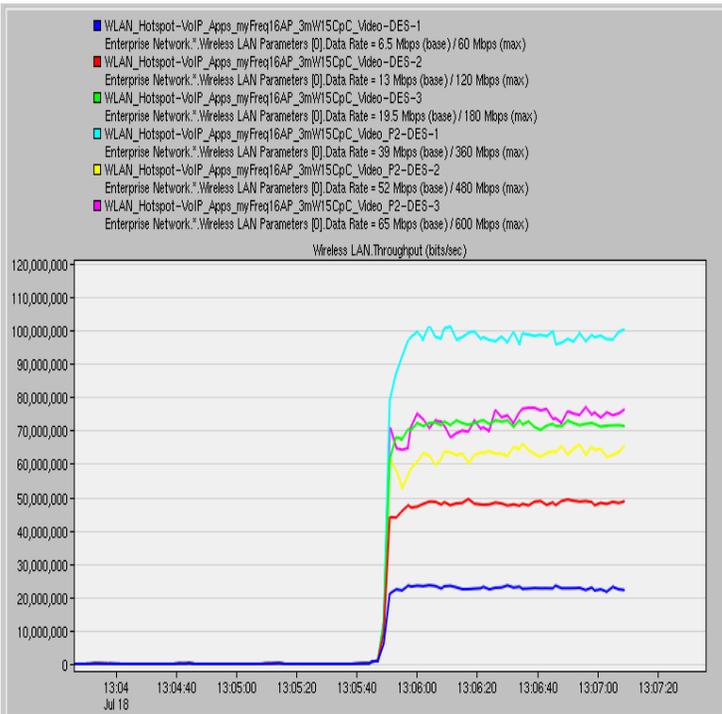


Figure 101: Throughput, IP Telephony Call, Video and Applications added in the simulation

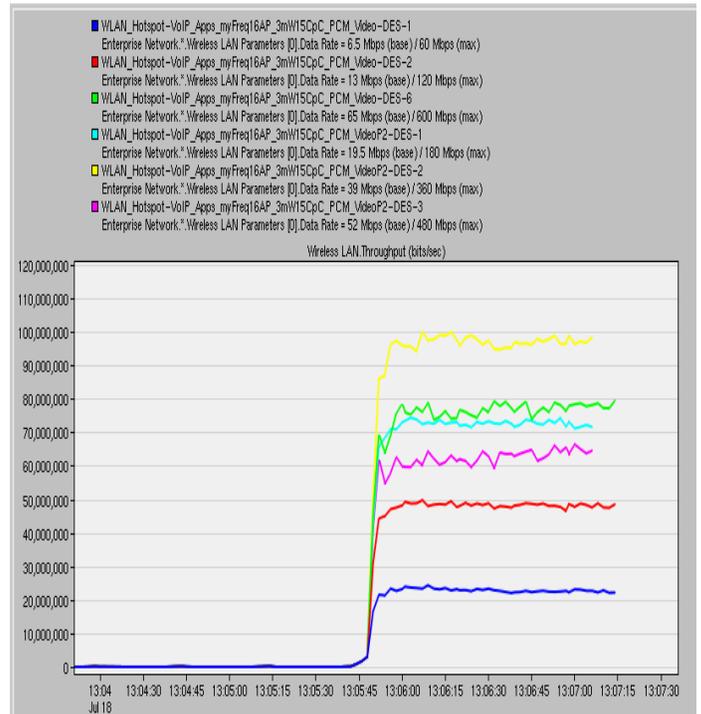


Figure 102: Throughput, PCM Quality Call, Video and Applications added in the simulation

4.10. AP WLAN Deployment – AP 1 Forwarding Memory Queue Size for data rates 6.5, 13, 19.5, 39, 65 Mbps

The APs are vital components for the WLAN networks under study. It is imperative to investigate their health, performance, and behavior. Some important key health indicators are shown in the results presented in Sections 4.10 – 4.15. The queue size, queuing delay and the CPU utilization are key performance indicators.

For these results, we have set the forwarding rate to 50000 packets/sec and we present indicative results only from AP 1.

We should recall that the APs (like the workstations) also have infinite higher layer data buffers in our configuration to avoid packet dropping due to buffer overflow. Hence, no data packets arriving from higher layers will be discarded, as the APs always have free space to store new packets, but we should not confuse this with the memory of the router’s forwarding queue which we observe and which has a size of 16 MB. This represents the memory used to store packets awaiting processing or currently processed by the forwarding CPU. In a centralized scheme, processed packets remain in that memory until sent out on an appropriate interface. When space is unavailable to buffer a packet, the incoming datagram is dropped. We can see the forwarding memory queue size increasing as more traffic is added on the network. The maximum Forwarding Memory Queue Size values that we find in our results for AP 1, are presented in Table 14 and never exceed 70 Bytes.

Maximum Forwarding Memory Queue Size (bytes)				
Call Type	IP Telephony		PCM Quality	
Deployment	9 AP WLAN	16 AP WLAN	9 AP WLAN	16 AP WLAN
VoIP (30 calls per cell)	1.75	8	3.5	8
VoIP (15 calls per cell)	1.4	3.5	2.5	5.5
VoIP and Apps(15 calls per cell)	3.5	5.25	7.5	10
VoIP, Apps and Video (15 calls per cell)	47	57	70	70

Table 14: Maximum Forwarding Memory Queue Size Used Over All Data Rates

The small forwarding memory queue sizes also explain the small forwarding memory queuing delay values, which never exceed 0.00005 seconds.

The “CPU utilization” statistic, which models the IP packet forwarding delays and application processing delays, never exceeds 10% as shown in all of the Figures in Sections 4.14 – 4.15. Thus, the saturation processing point 100% never occurs for the router.

The underutilization of both the CPU and the forwarding memory queue size strengthens the viewpoint that the channel capacity and the interference are the limiting factors in terms of the throughput received and not the number of the APs in a WLAN network.

4.10.1. Only VoIP Calls Added

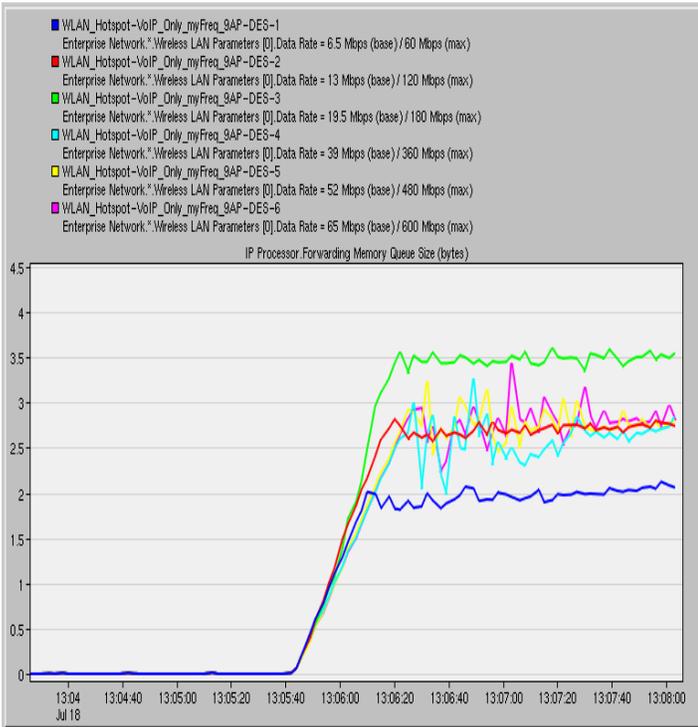


Figure 103: AP 1 Forwarding Memory Queue Size, 1 workstation per cell, 30 calls per workstation, IP Telephony Call

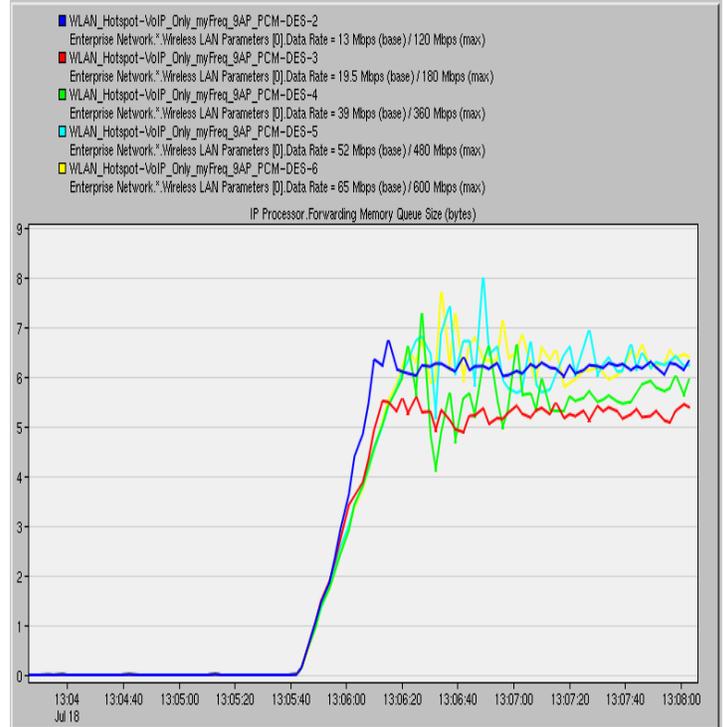


Figure 104: AP 1 Forwarding Memory Queue Size, 1 workstation per cell, 30 calls per workstation, PCM Quality Call

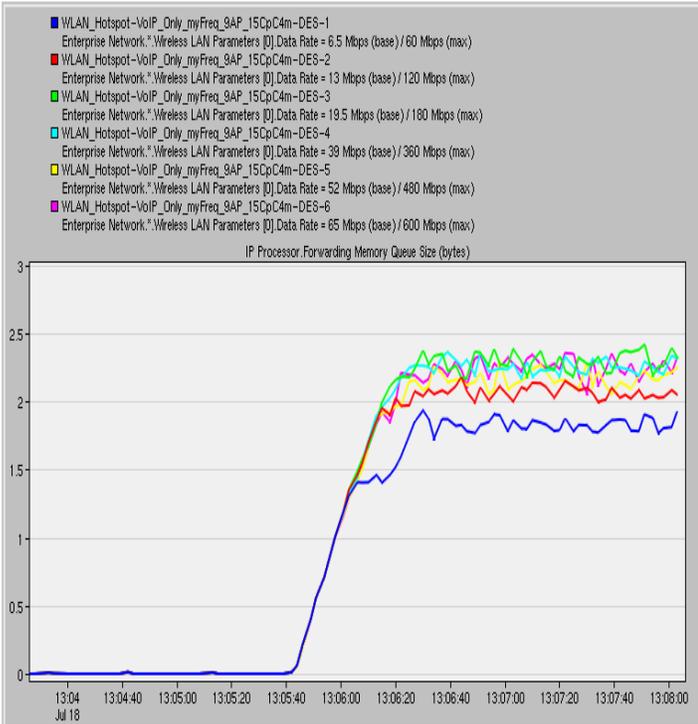


Figure 105: AP 1 Forwarding Memory Queue Size, 15 workstations per cell, 1 call per workstation, IP Telephony Call

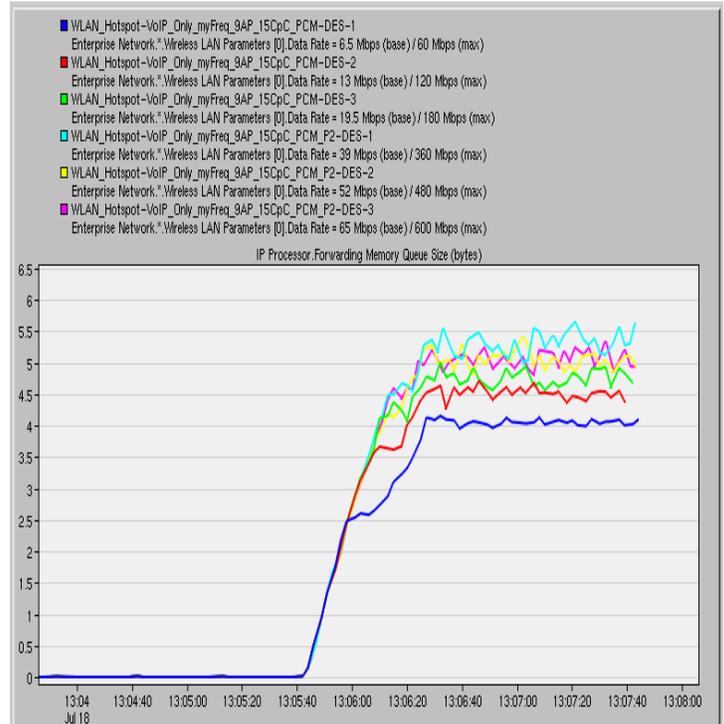


Figure 106: AP 1 Forwarding Memory Queue Size, 15 workstations per cell, 1 call per workstation, PCM Quality Call

4.10.2. VoIP Calls and Applications Added

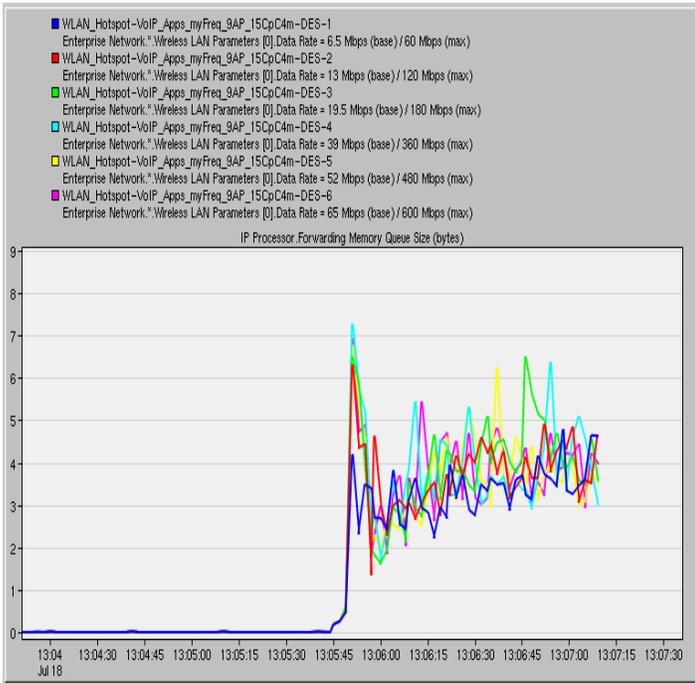


Figure 108: AP 1 Forwarding Memory Queue Size, IP Telephony Call, Applications added in the simulation

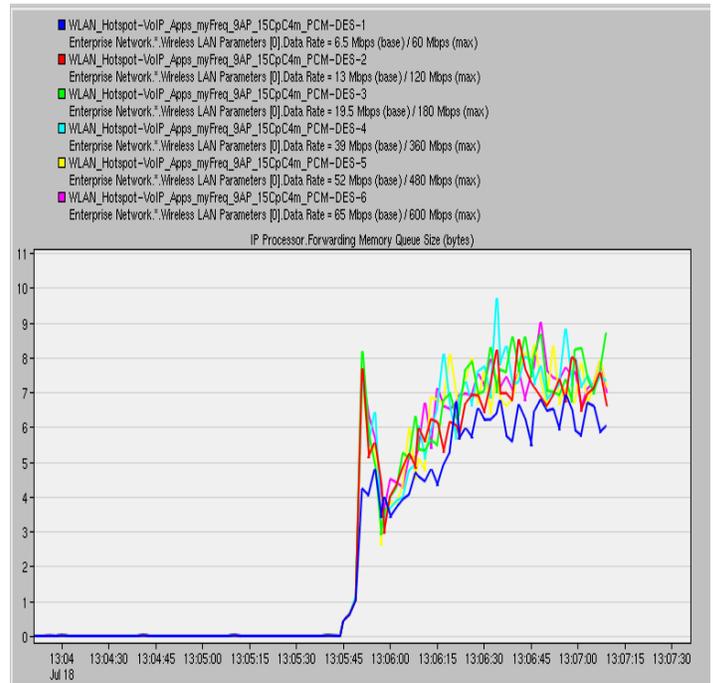


Figure 107: AP 1 Forwarding Memory Queue Size, PCM Quality Call, Applications added in the simulation

4.10.3. VoIP Calls, Applications and Videos Added

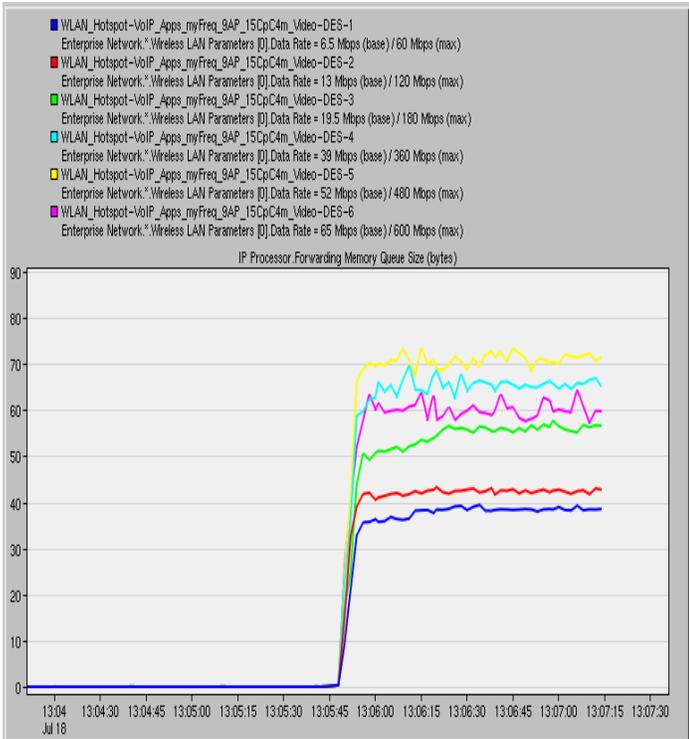


Figure 109: AP 1 Forwarding Memory Queue Size, IP Telephony Call, Video and Applications added in the simulation

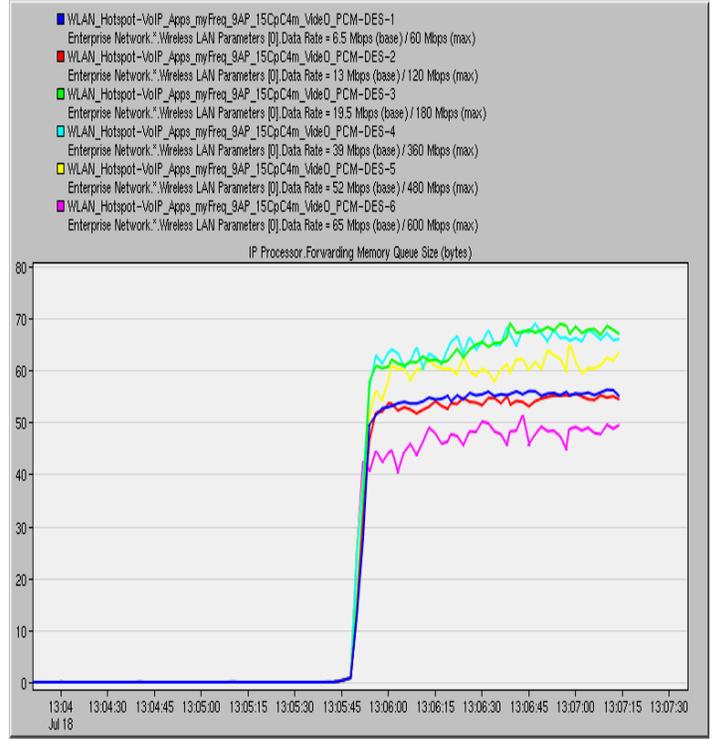


Figure 110: AP 1 Forwarding Memory Queue Size, PCM Quality Call, Video and Applications added in the simulation

4.11.16 AP WLAN Deployment – AP 1 Forwarding Memory Queue Size for data rates 6.5, 13, 19.5, 39, 65 Mbps

4.11.1. Only VoIP Calls Added

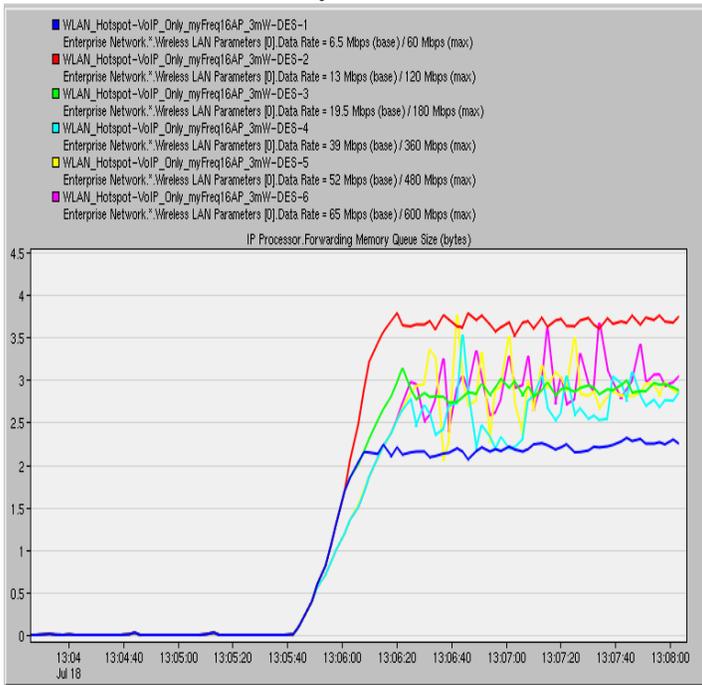


Figure 112: AP 1 Forwarding Memory Queue Size, 1 workstation per cell, 30 calls per workstation, IP Telephony Call

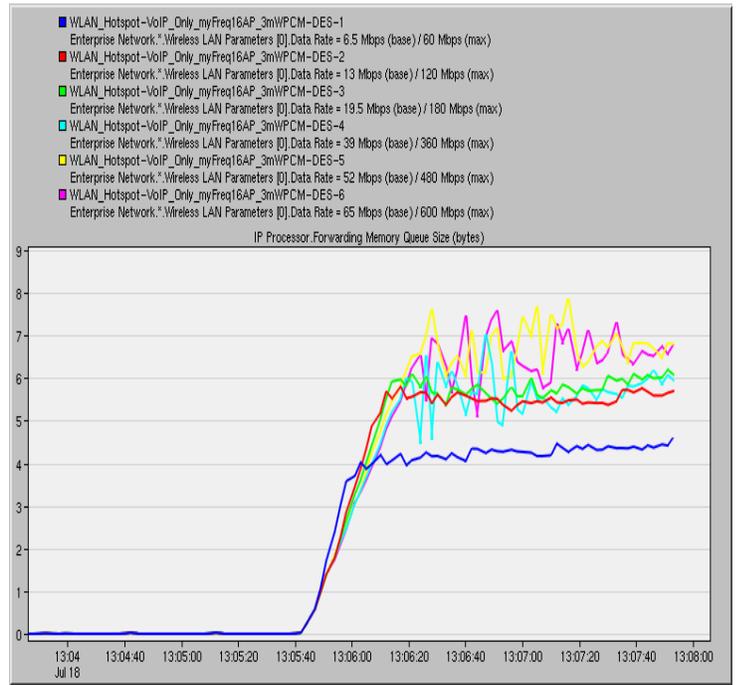


Figure 111: AP 1 Forwarding Memory Queue Size, 1 workstation per cell, 30 calls per workstation, PCM Quality Call

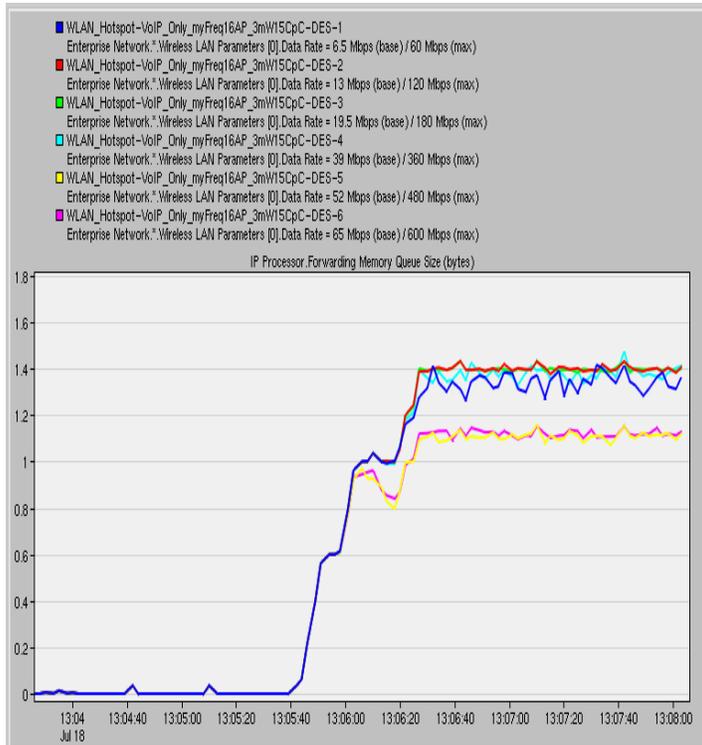


Figure 114: AP 1 Forwarding Memory Queue Size, 15 workstations per cell, 1 call per workstation, IP Telephony Call

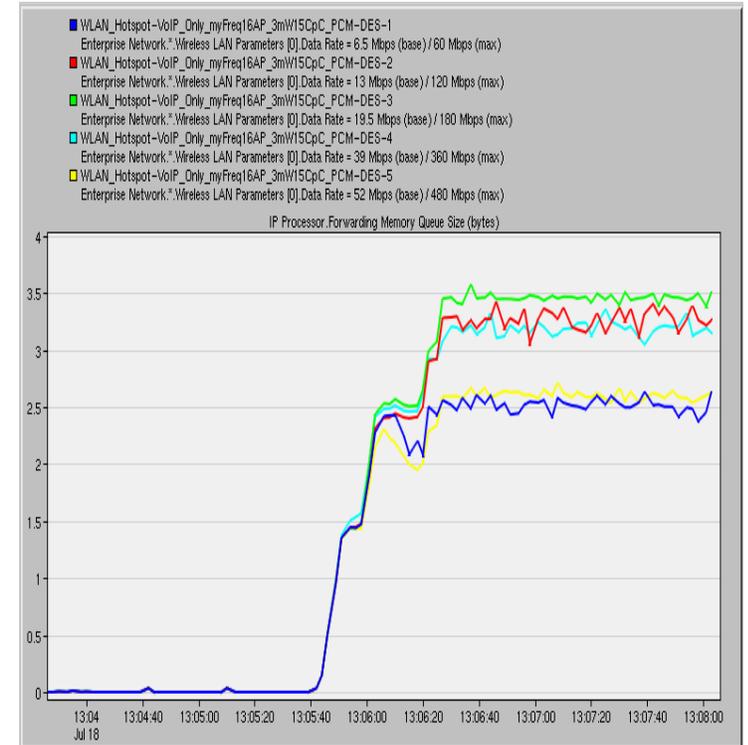


Figure 113: AP 1 Forwarding Memory Queue Size, 15 workstations per cell, 1 call per workstation, PCM Quality Call

4.11.2. VoIP Calls and Applications Added

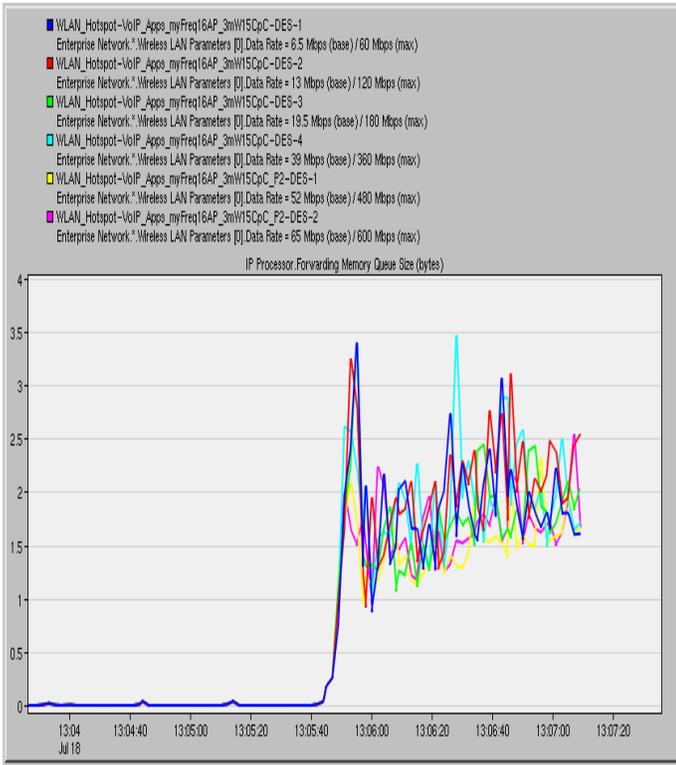


Figure 116: AP 1 Forwarding Memory Queue Size, IP Telephony Call, Applications added in the simulation

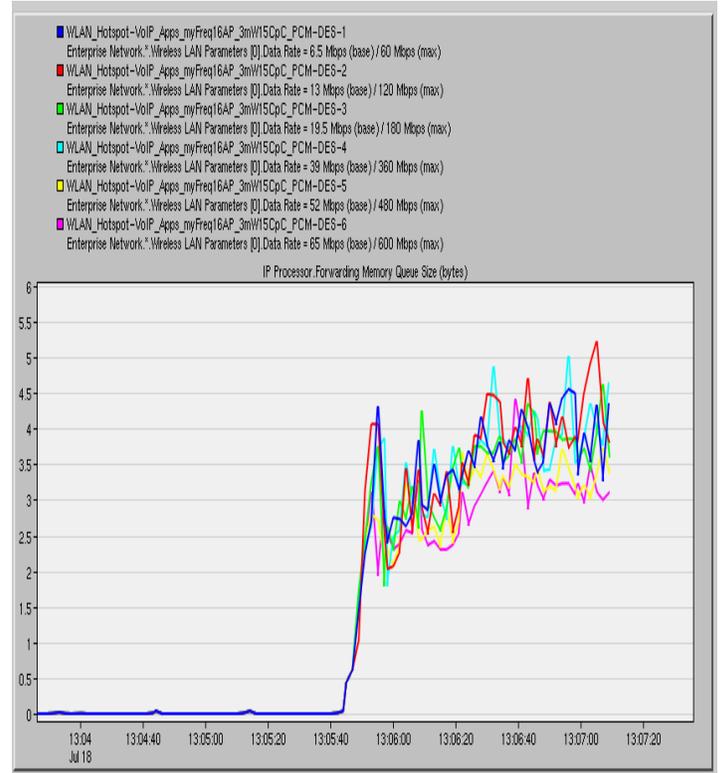


Figure 115: AP 1 Forwarding Memory Queue Size, PCM Quality Call, Applications added in the simulation

4.11.3. VoIP Calls, Applications and Videos Added

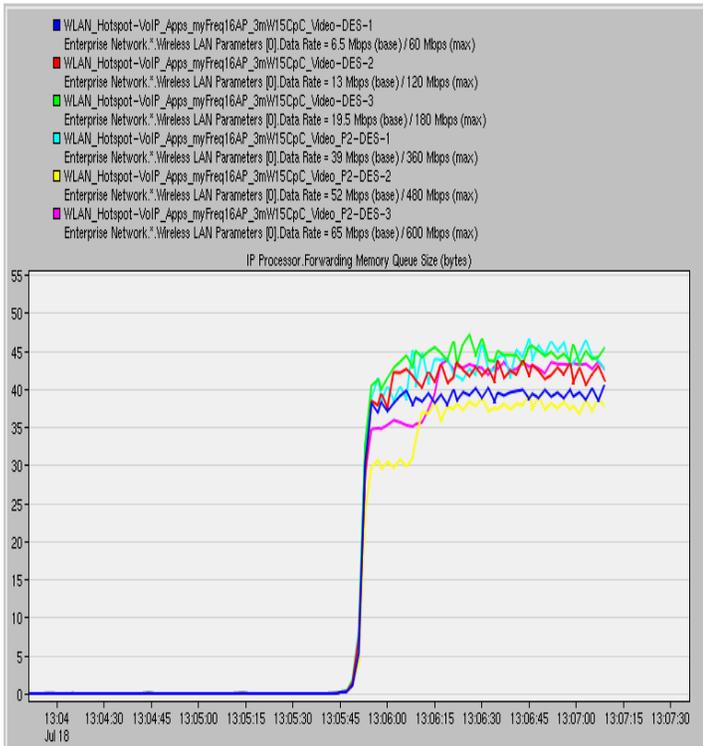


Figure 117: AP 1 Forwarding Memory Queue Size, IP Telephony Call, Video and Applications added in the simulation

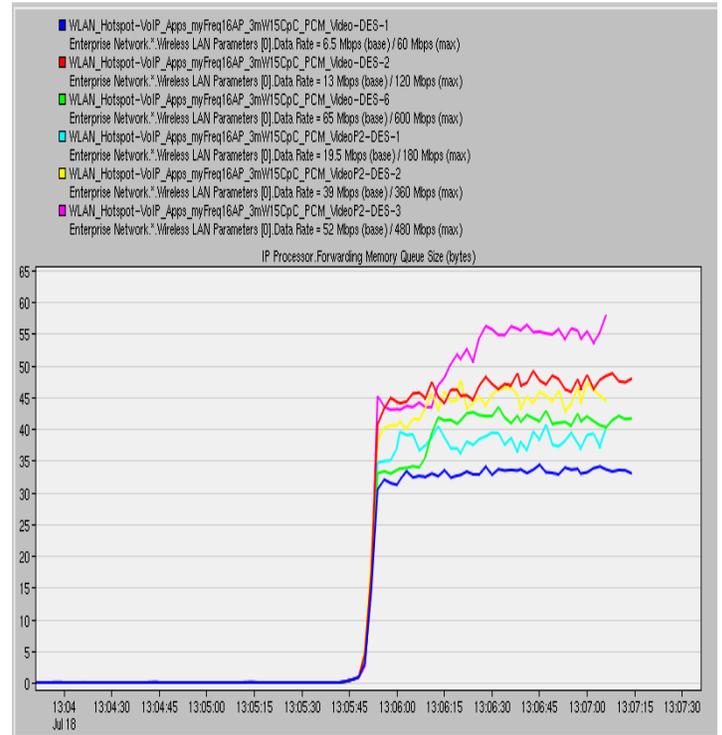


Figure 118: AP 1 Forwarding Memory Queue Size, PCM Quality Call, Video and Applications added in the simulation

4.12.9 AP WLAN Deployment – AP 1 Forwarding Memory Queuing Delay for data rates 6.5, 13, 19.5, 39, 65 Mbps

4.12.1. Only VoIP Calls Added

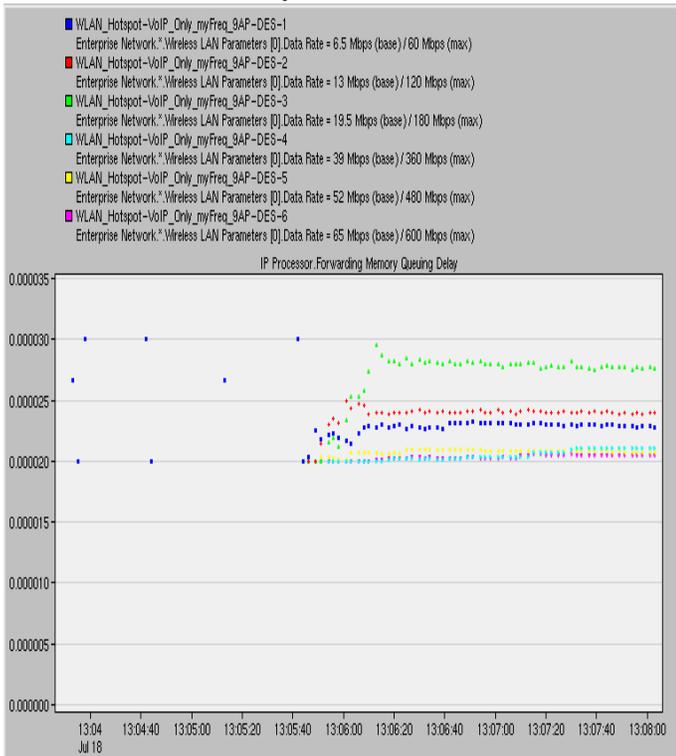


Figure 120: AP 1 Forwarding Memory Queuing Delay, 1 workstation per cell, 30 calls per workstation, IP Telephony Call

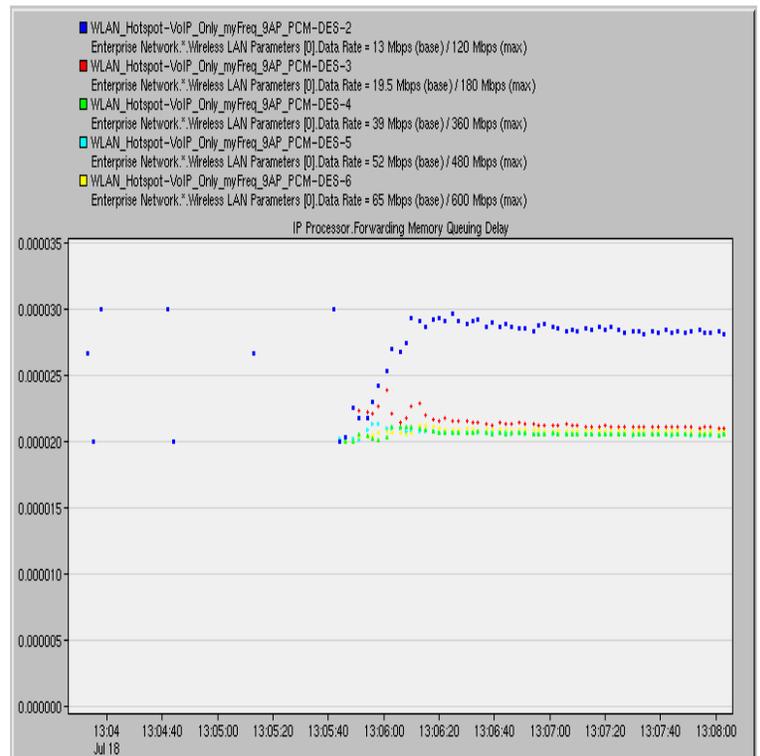


Figure 119: AP 1 Forwarding Memory Queuing Delay, 1 workstation per cell, 30 calls per workstation, PCM Quality Call

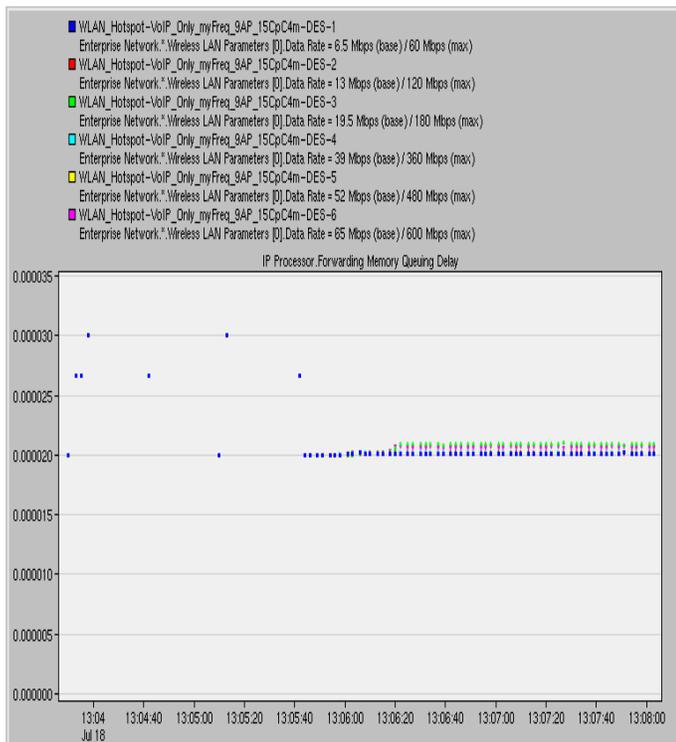


Figure 121: AP 1 Forwarding Memory Queue Size, 15 workstations per cell, 1 call per workstation, IP Telephony Call

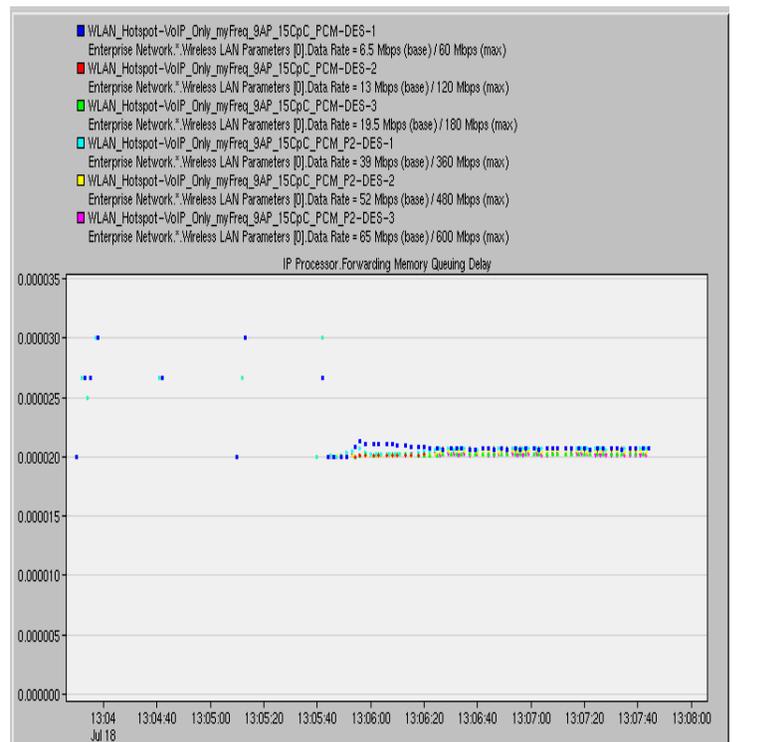


Figure 122: AP 1 Forwarding Memory Queue Size, 15 workstations per cell, 1 call per workstation, PCM Quality Call

4.12.2. VoIP Calls and Applications Added

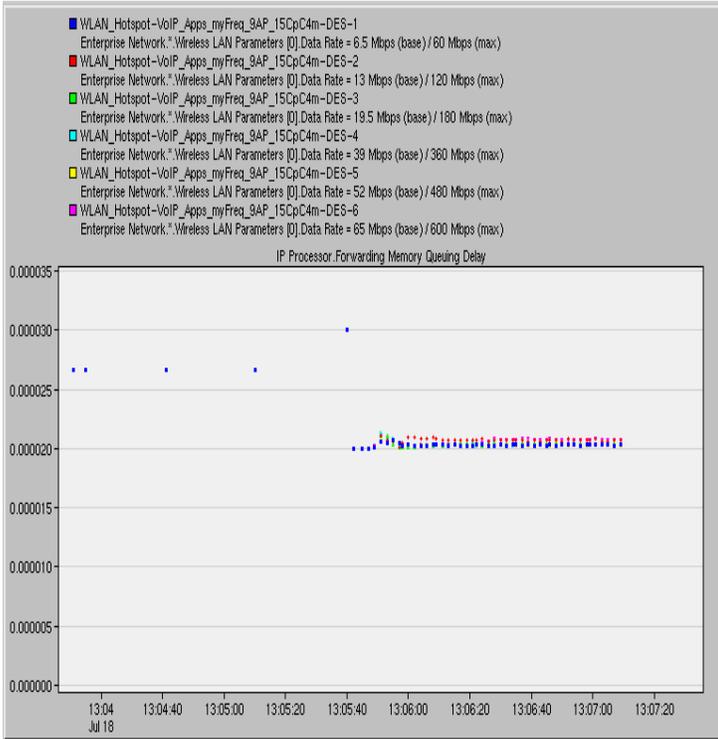


Figure 124: AP 1 Forwarding Memory Queuing Delay, IP Telephony Call, Applications added in the simulation

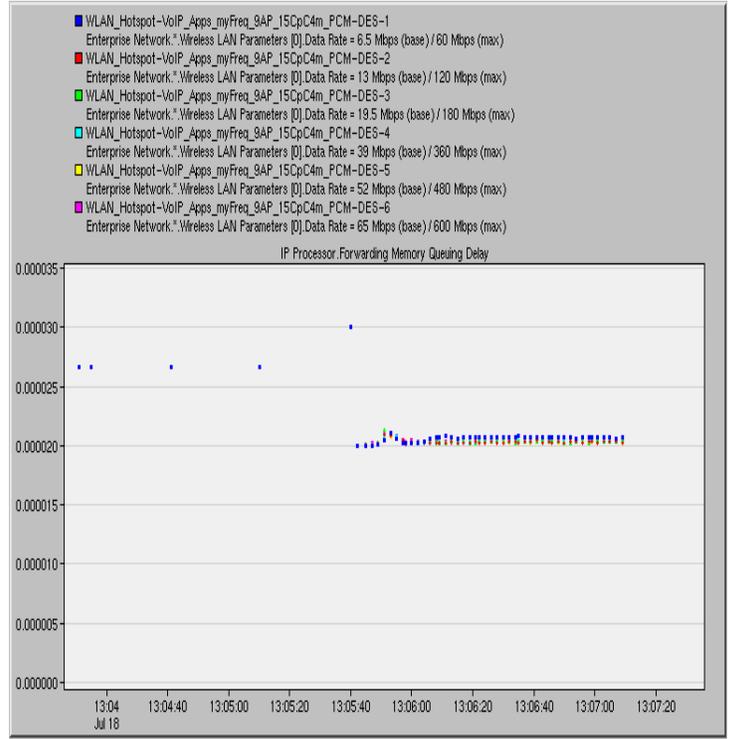


Figure 123: AP 1 Forwarding Memory Queuing Delay, PCM Quality Call, Applications added in the simulation

4.12.3. VoIP Calls, Applications and Videos Added

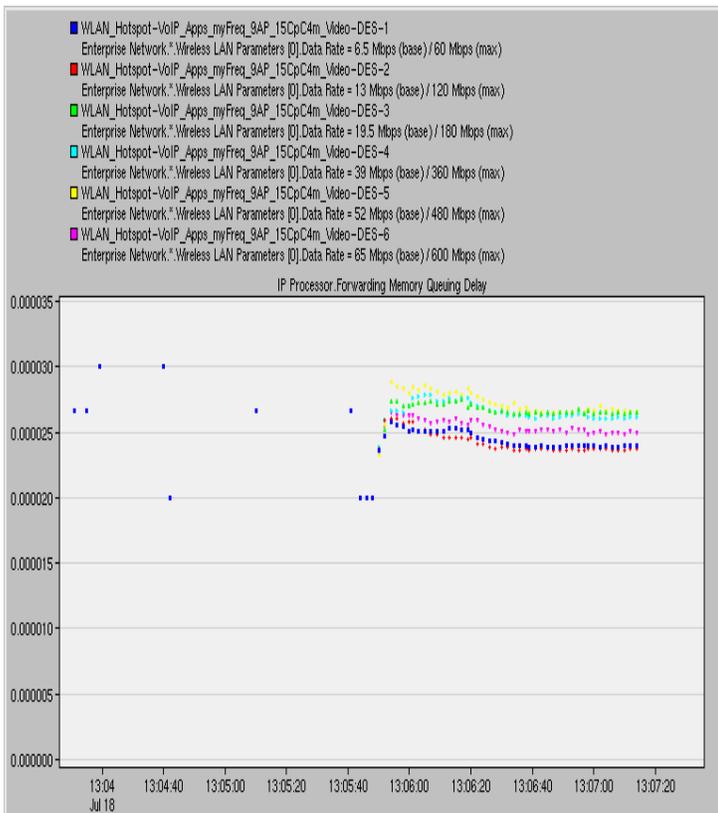


Figure 126: AP 1 Forwarding Memory Queuing Delay, IP Telephony Call, Video and Applications added in the simulation

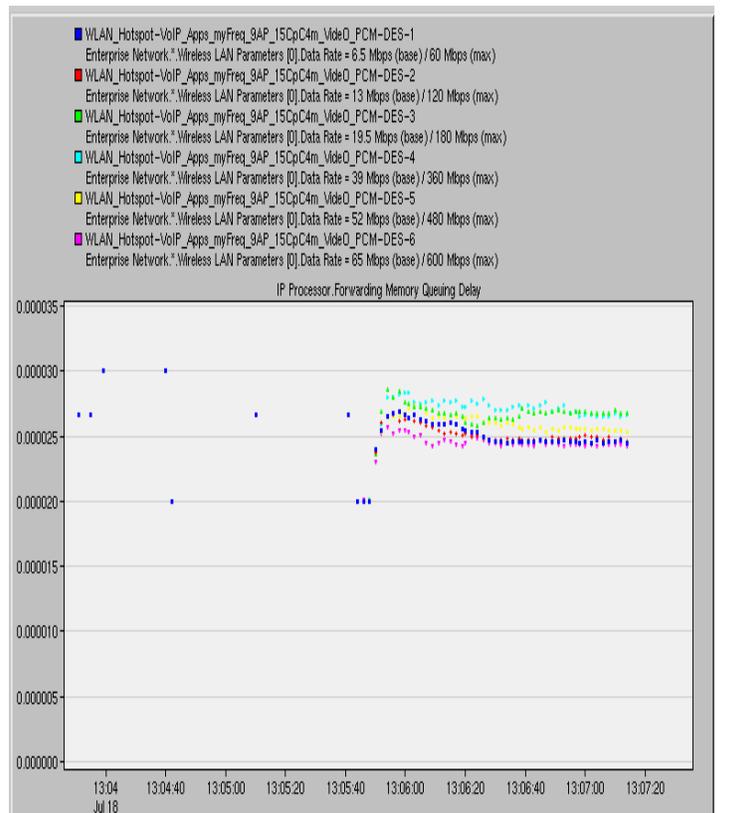


Figure 125: AP 1 Forwarding Memory Queuing Delay PCM Quality Call, Video and Applications added in the simulation

4.13.16 AP WLAN Deployment – AP 1 Forwarding Memory Queuing Delay for data rates 6.5, 13, 19.5, 39, 65 Mbps

4.13.1. Only VoIP Calls Added

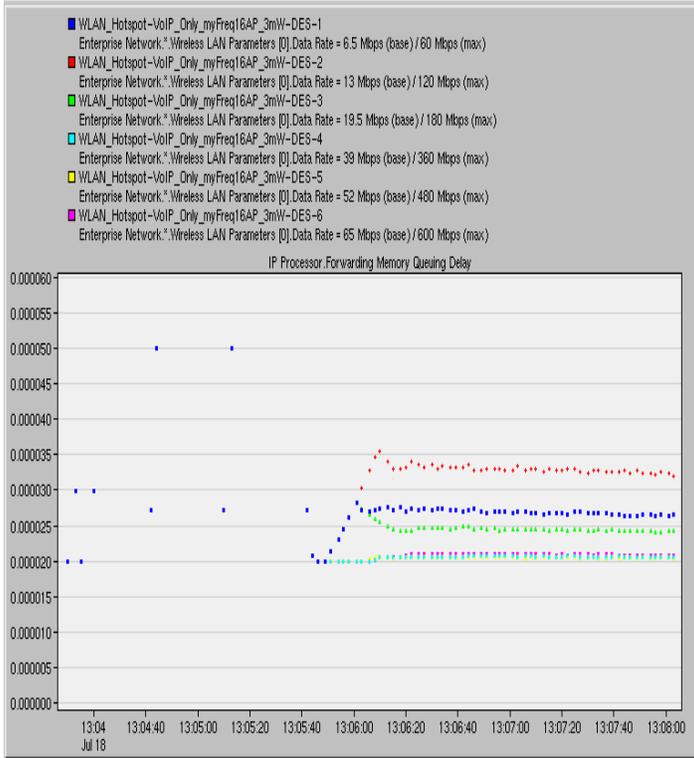


Figure 128: AP 1 Forwarding Memory Queuing Delay, 1 workstation per cell, 30 calls per workstation, IP Telephony Call

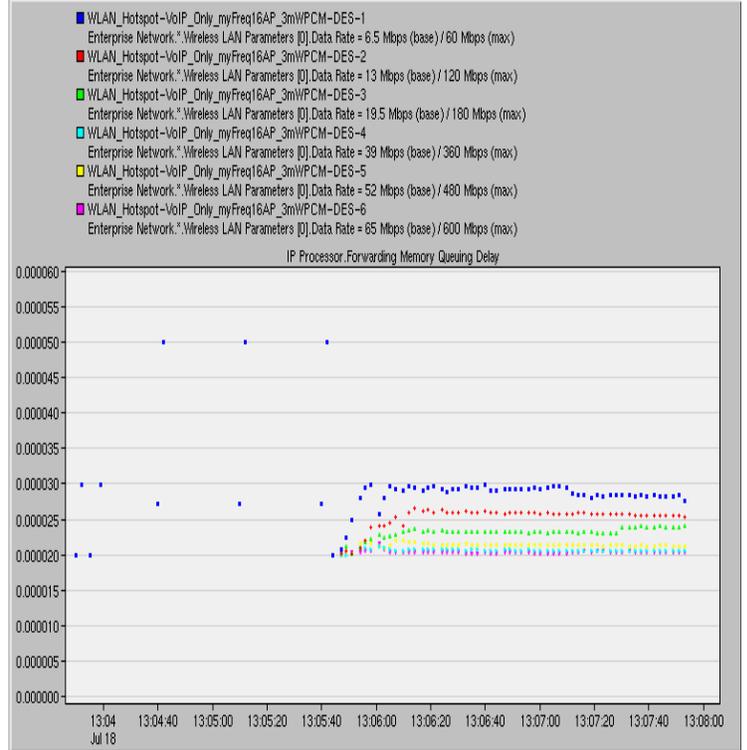


Figure 127: AP 1 Forwarding Memory Queuing Delay, 1 workstation per cell, 30 calls per workstation, PCM Quality Call

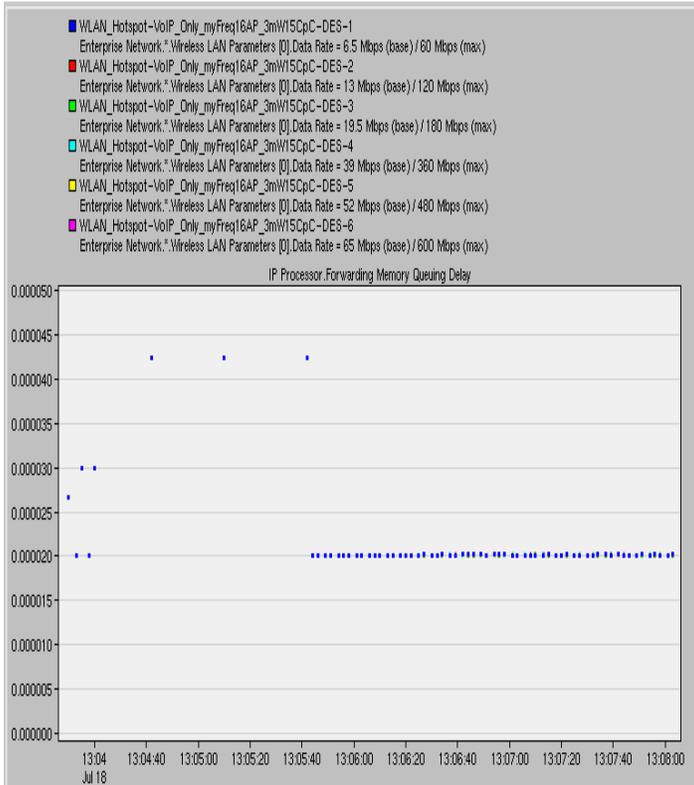


Figure 129: AP 1 Forwarding Memory Queuing Delay, 15 workstations per cell, 1 call per workstation, IP Telephony Call

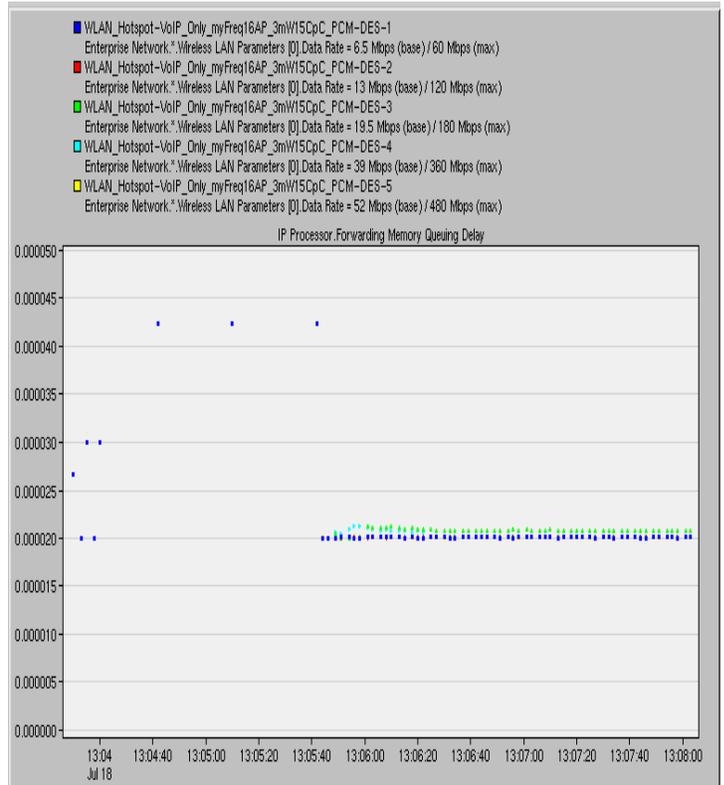


Figure 130: AP 1 Forwarding Memory Queuing Delay, 15 workstations per cell, 1 call per workstation, PCM Quality Call

4.13.2. VoIP Calls and Applications Added

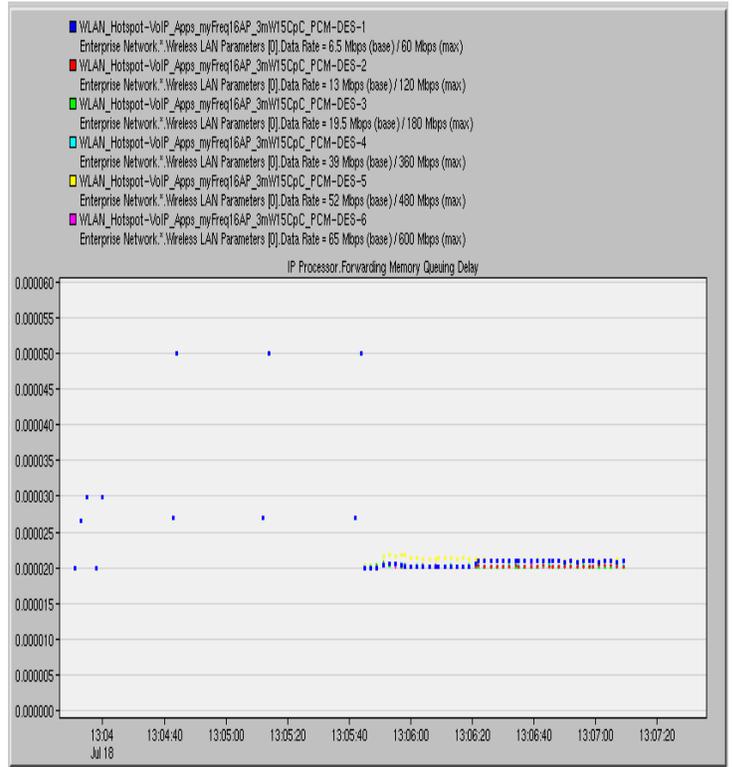
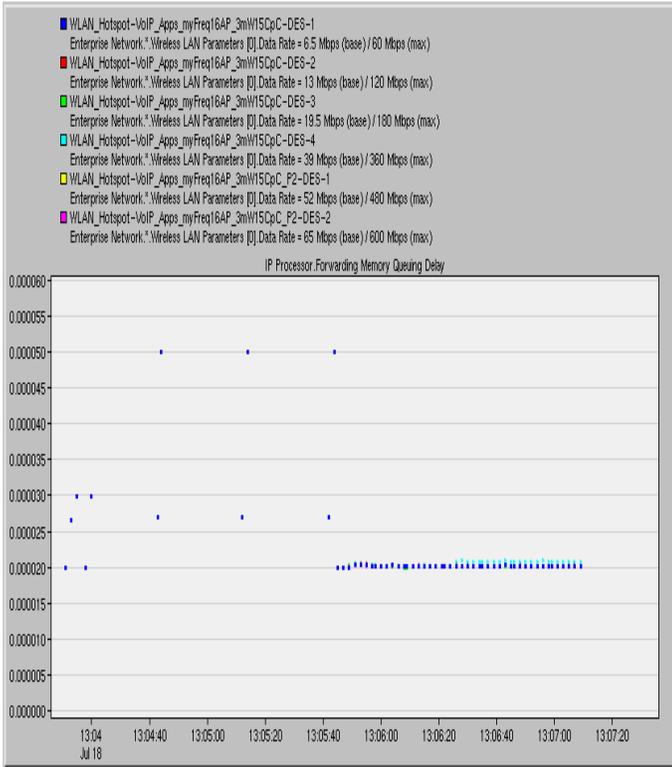


Figure 132: AP 1 Forwarding Memory Queuing Delay, IP Telephony Call, Applications added in the simulation

Figure 131: AP 1 Forwarding Memory Queuing Delay, PCM Quality Call, Applications added in the simulation

4.13.3. VoIP Calls, Applications and Videos Added

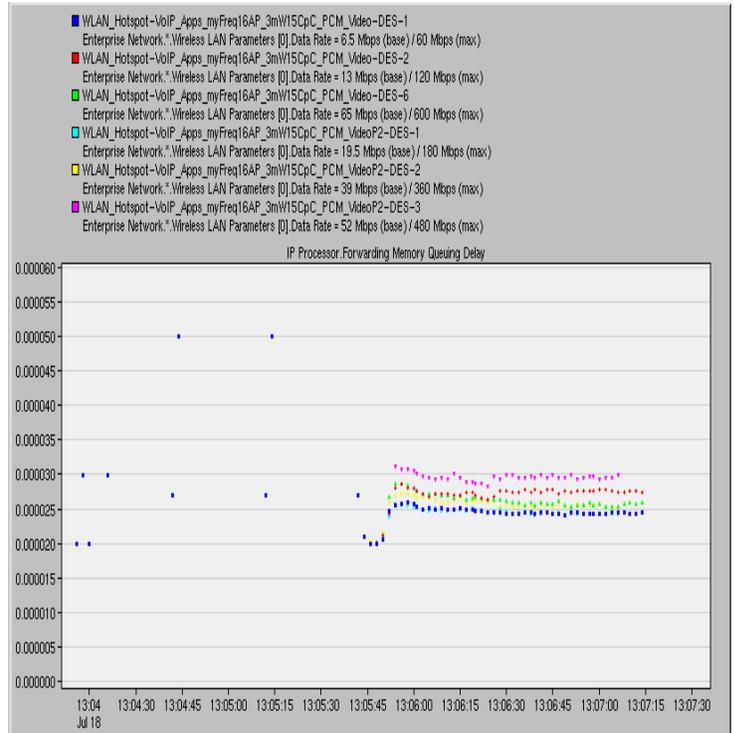
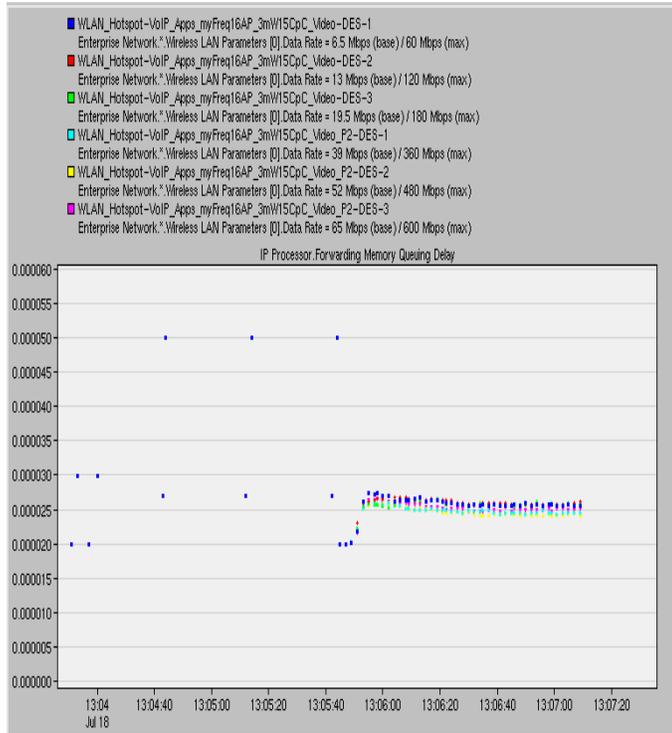


Figure 133: AP 1 Forwarding Memory Queuing Delay, IP Telephony Call, Video and Applications added in the simulation

Figure 134: AP 1 Forwarding Memory Queuing Delay, PCM Quality Call, Video and Applications added in the simulation

4.14.9 AP WLAN Deployment – AP 1 CPU Utilization for data rates 6.5, 13, 19.5, 39, 65 Mbps

4.14.1. Only VoIP Calls Added

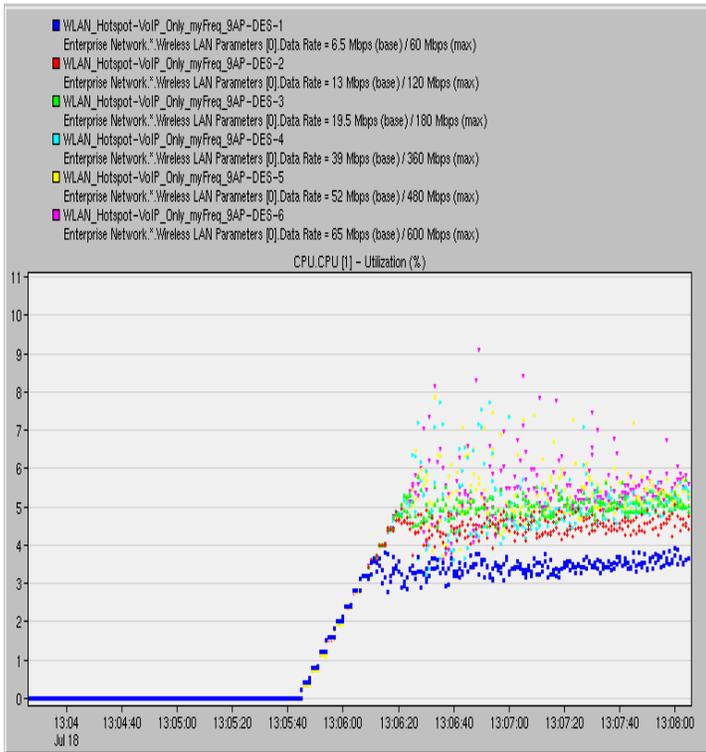


Figure 135: AP 1 CPU Utilization, 1 workstation per cell, 30 calls per workstation, IP Telephony Call

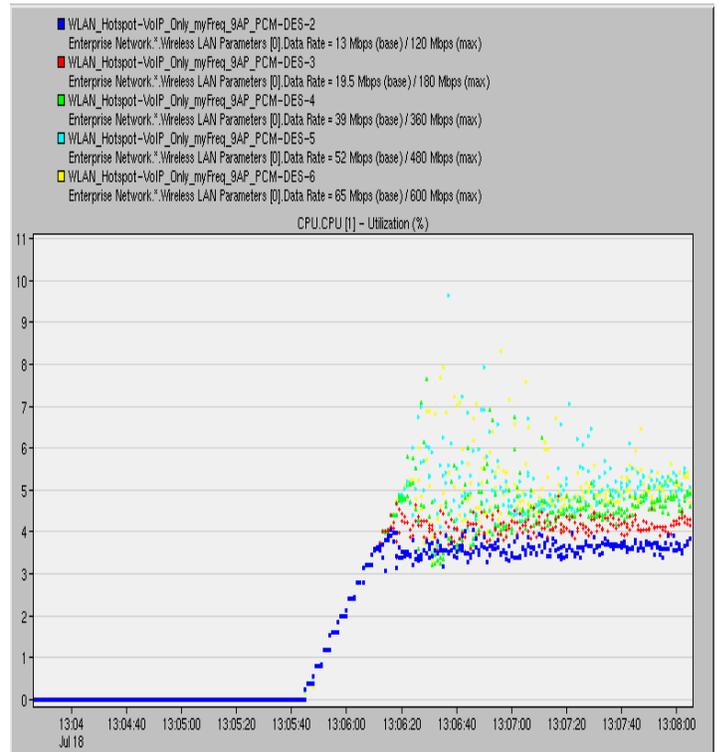


Figure 136: AP 1 CPU Utilization, 1 workstation per cell, 30 calls per workstation, PCM Quality Call

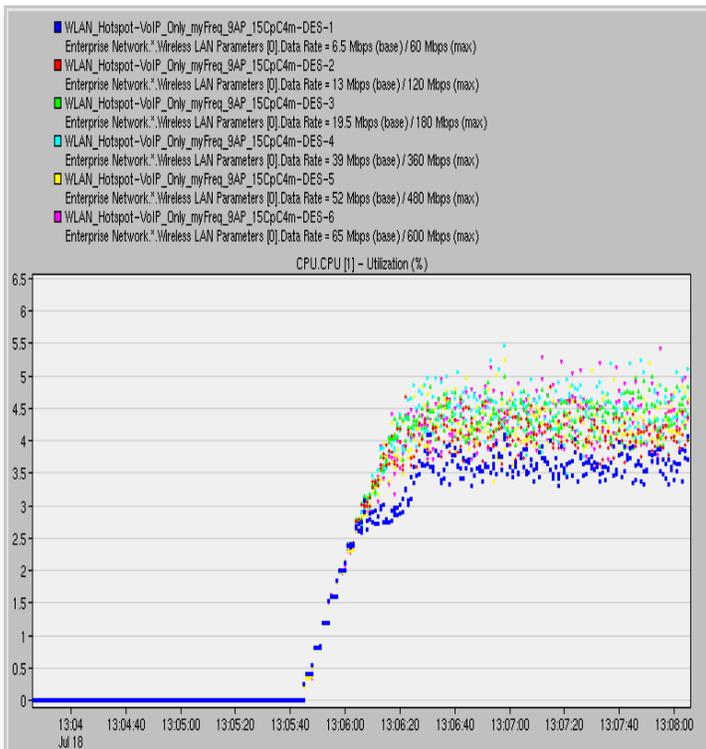


Figure 137: AP 1 CPU Utilization, 15 workstations per cell, 1 call per workstation, IP Telephony Call

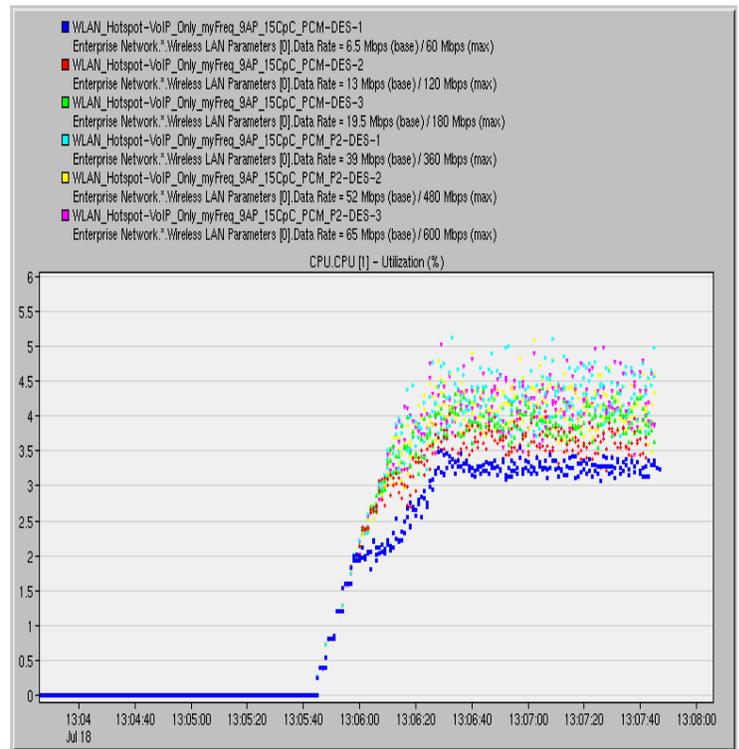


Figure 138: AP 1 CPU Utilization, 15 workstations per cell, 1 call per workstation, PCM Quality Call

4.14.2. VoIP Calls and Applications Added

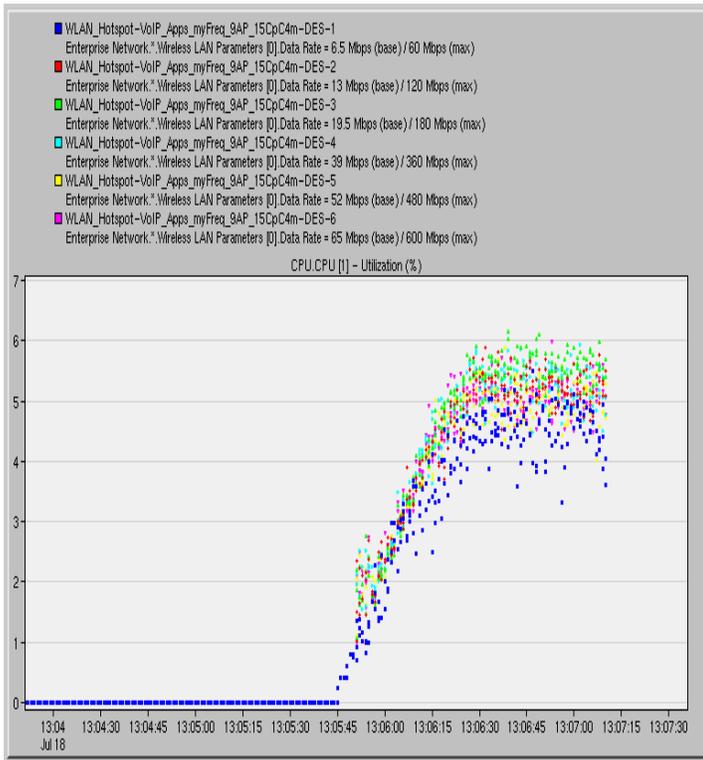


Figure 140: AP 1 CPU Utilization, IP Telephony Call, Applications added in the simulation

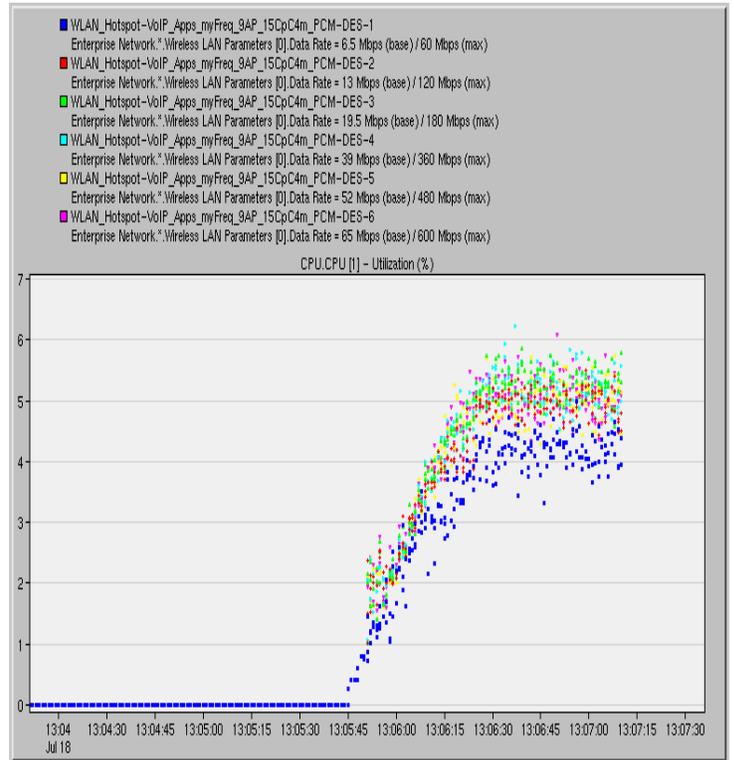


Figure 139: AP 1 CPU Utilization, PCM Quality Call, Applications added in the simulation

4.14.3. VoIP Calls, Applications and Videos Added

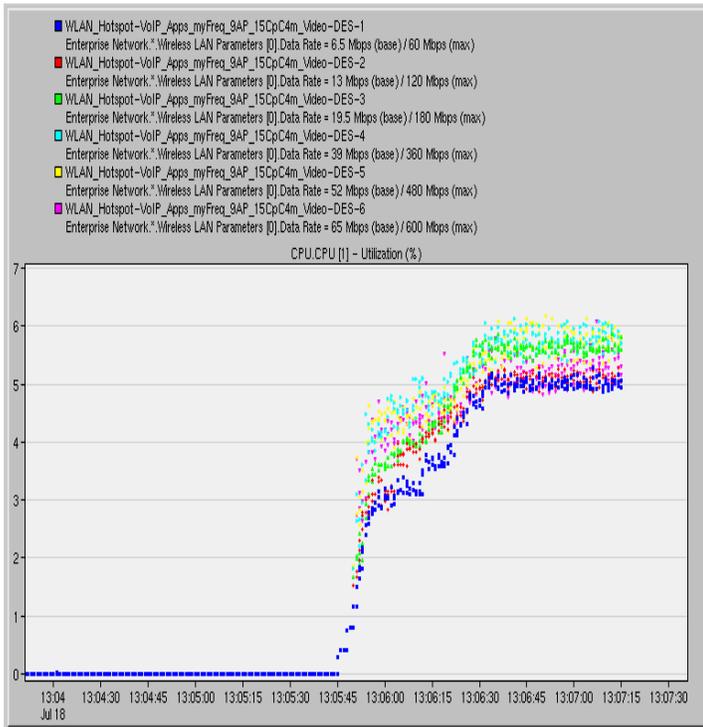


Figure 142: AP 1 CPU Utilization, IP Telephony Call, Video and Applications added in the simulation

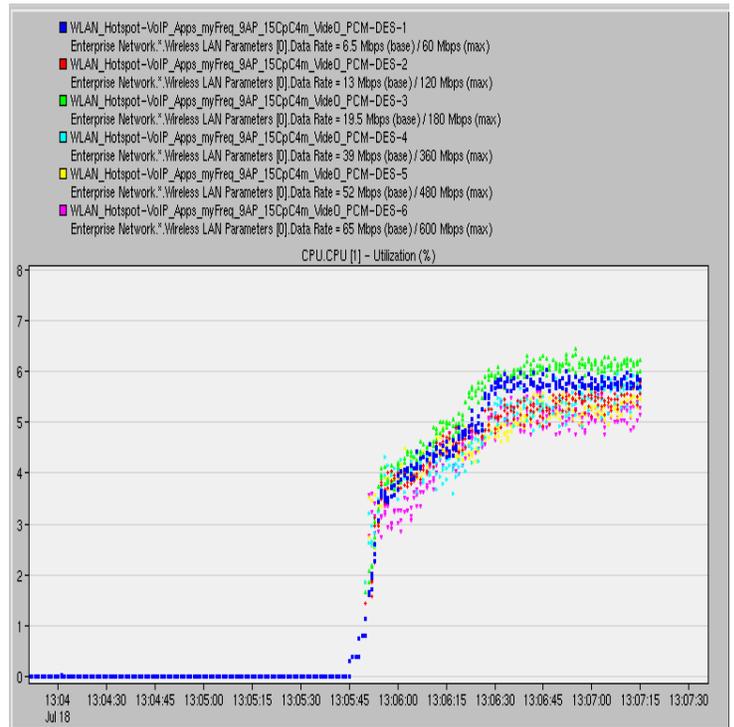


Figure 141: AP 1 CPU Utilization, PCM Quality Call, Video and Applications added in the simulation

4.15.16 AP WLAN Deployment – AP 1 CPU Utilization for data rates 6.5, 13, 19.5, 39, 65 Mbps

4.15.1. Only VoIP Calls Added

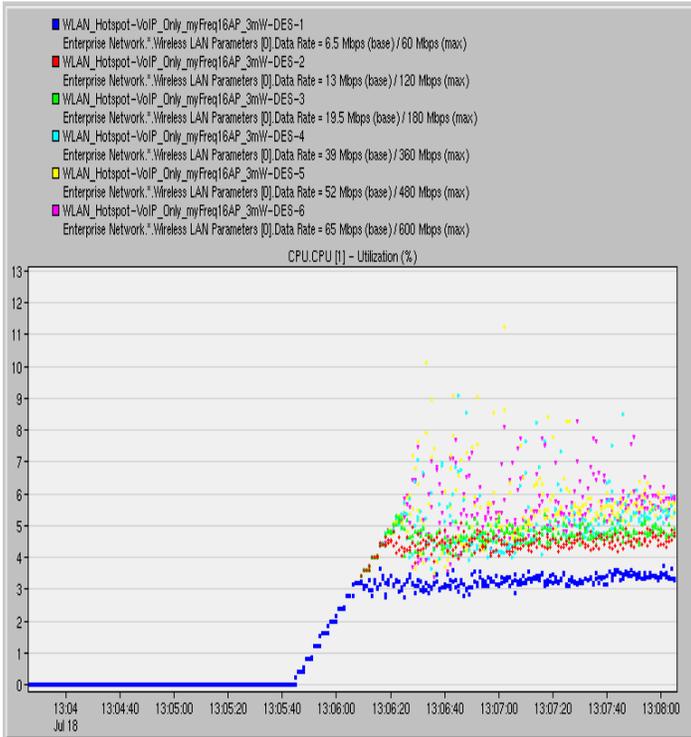


Figure 144: AP 1 CPU Utilization, 1 workstation per cell, 30 calls per workstation, IP Telephony Call

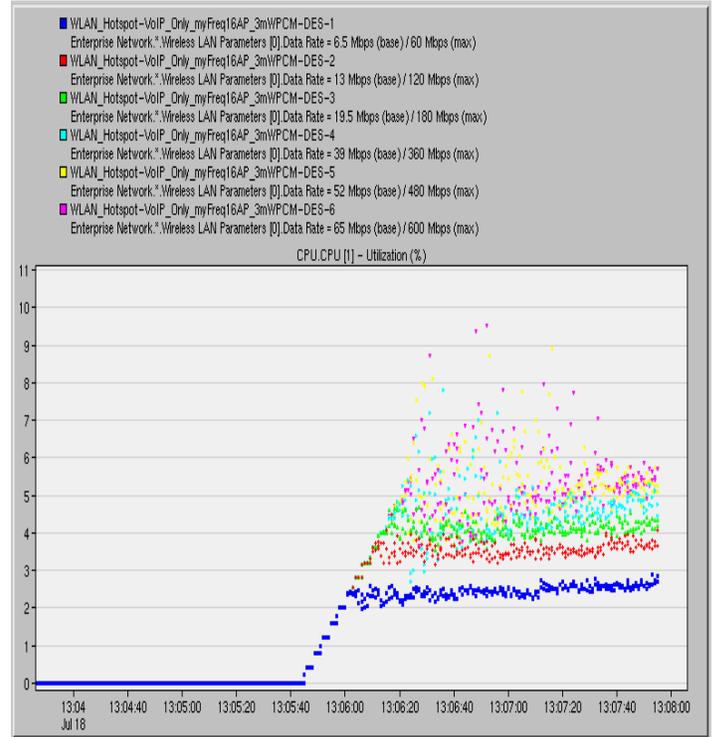


Figure 143: AP 1 CPU Utilization, 1 workstation per cell, 30 calls per workstation, PCM Quality Call

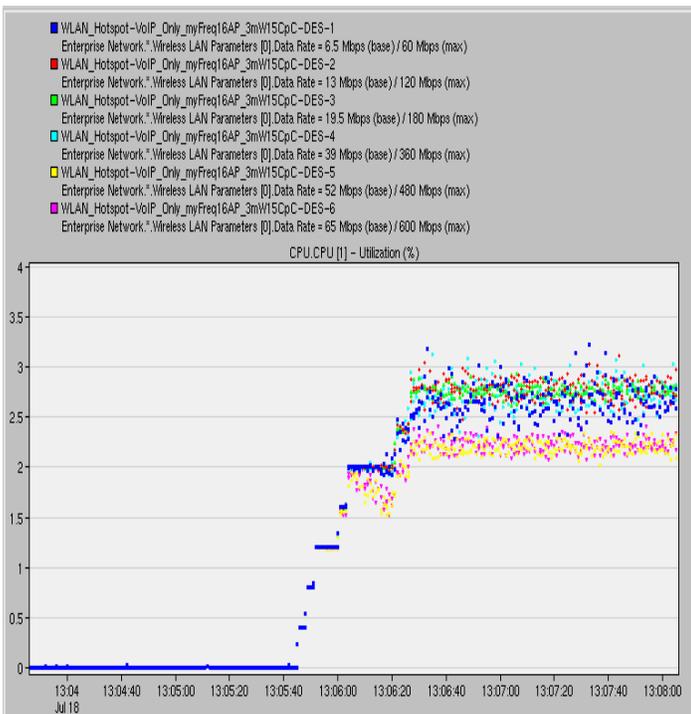


Figure 145: AP 1 CPU Utilization, 15 workstations per cell, 1 call per workstation, IP Telephony Call

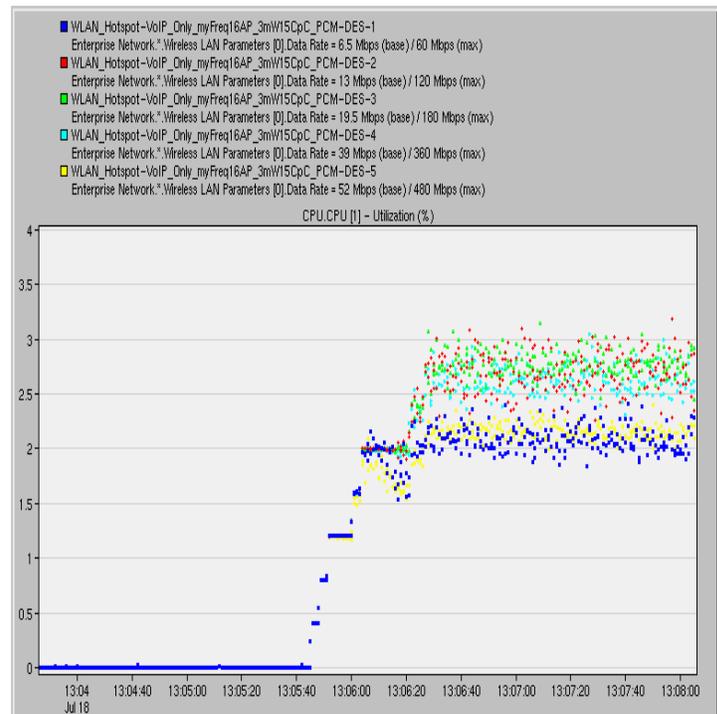


Figure 146: AP 1 CPU Utilization, 15 workstations per cell, 1 call per workstation, PCM Quality Call

4.15.2. VoIP Calls and Applications Added

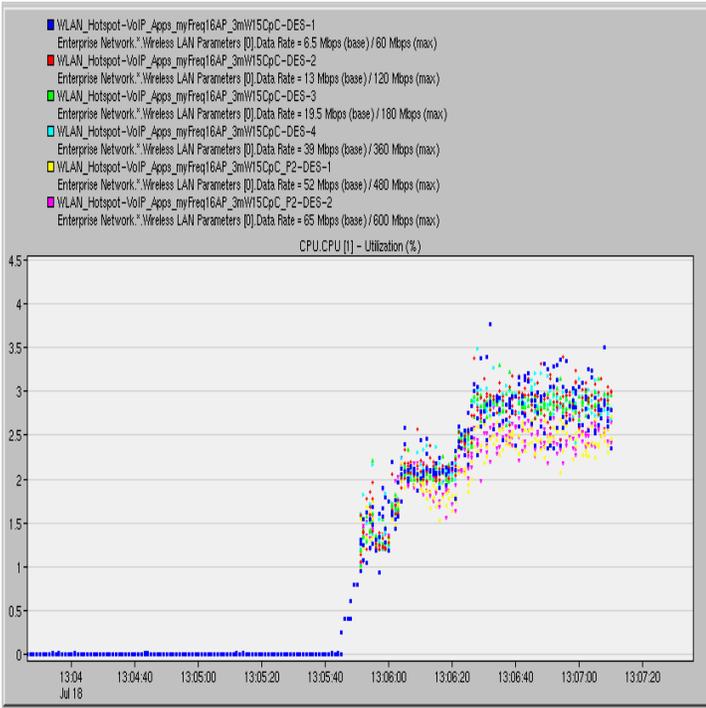


Figure 148: AP 1 CPU Utilization, IP Telephony Call, Applications added in the simulation

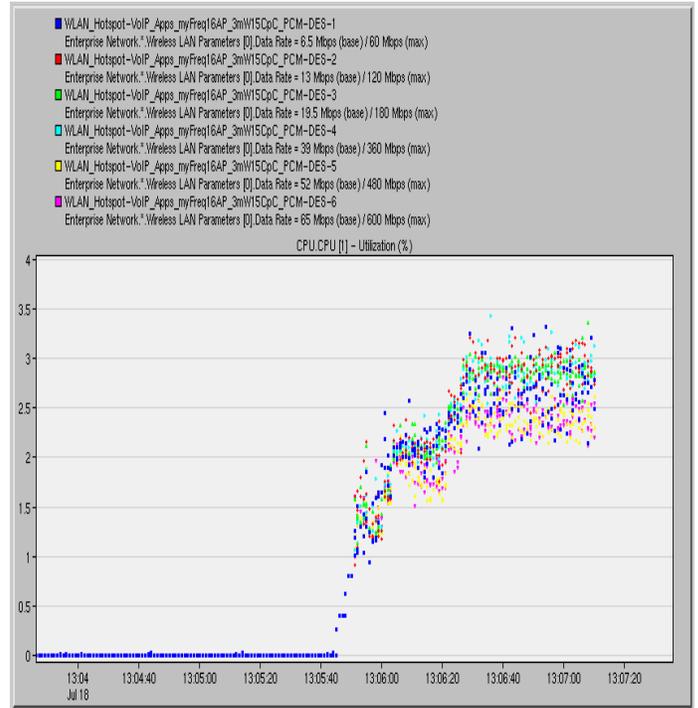


Figure 147: AP 1 CPU Utilization, PCM Quality Call, Applications added in the simulation

4.15.3. VoIP Calls, Applications and Videos Added

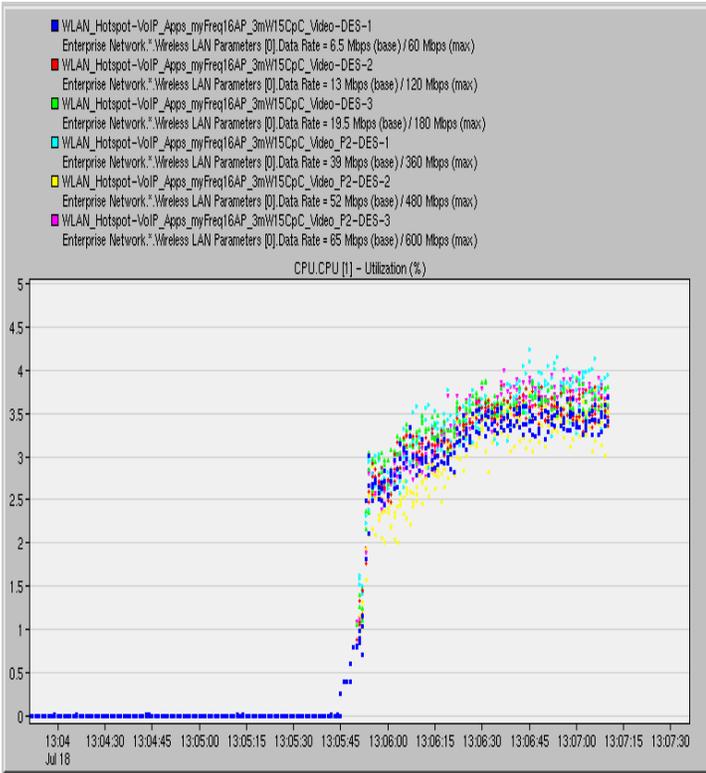


Figure 149: AP 1 CPU Utilization, IP Telephony Call, Video and Applications added in the simulation

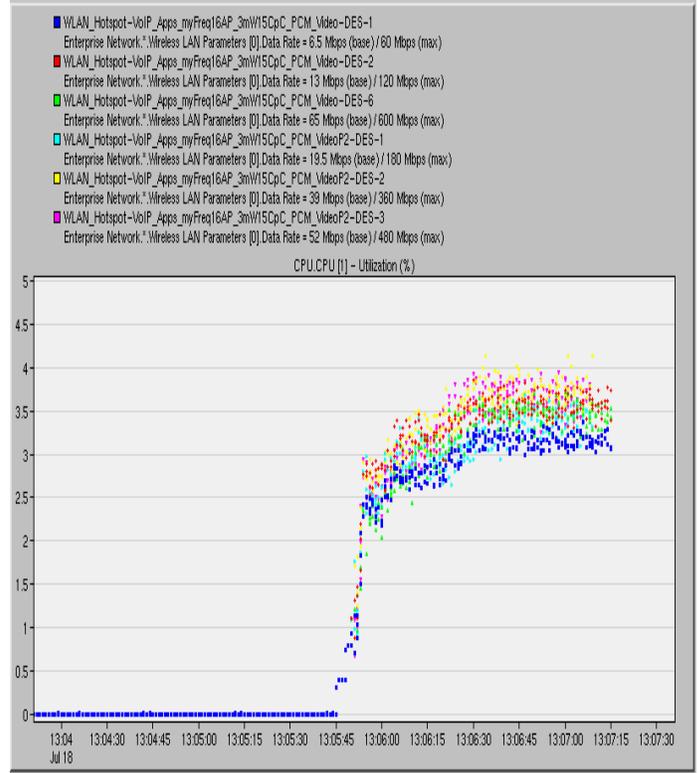


Figure 150: AP 1 CPU Utilization, PCM Quality Call, Video and Applications added in the simulation

4.16.9 AP WLAN Deployment – DB Query Response Time for data rates 6.5, 13, 19.5, 39, 65 Mbps

As shown in the Figures in Sections 4.16 – 4.23, all the application response times in our experiments follow the same pattern. For the 9 AP WLAN deployments, we have 15 workstations per cell and one call added in every workstation, every 3 seconds. The applications start when the first call is deployed, and they all run simultaneously. With VoIP calls and applications deployed together, the applications have small response times, very close to zero. However, higher response time values occur only for the 6.5 Mbps data rate. We have already explained that in this case the problem is related to interference, as the cell radius extends for small data rates.

For the 16 AP WLAN deployments with VoIP calls and applications deployed, again we have 15 workstations per cell and one call added in every workstation, every 3 seconds. The applications start when the first call is deployed, and they all run simultaneously. The applications also have small response times, very close to zero, but higher response time values occur only for the 52 and the 65 Mbps data rate. Here the problem is related to the high number of collisions that result in unacceptable delays, and to the fact that the transmission power is not high enough to support these data rates.

For both the 9 AP and the 16 WLAN deployments when videoconference is added on every node, the response time reaches very high values for all data rates, resulting in unsatisfying behavior due to the high bandwidth consumption by the videoconference sources.

4.16.1. VoIP Calls and Applications Added

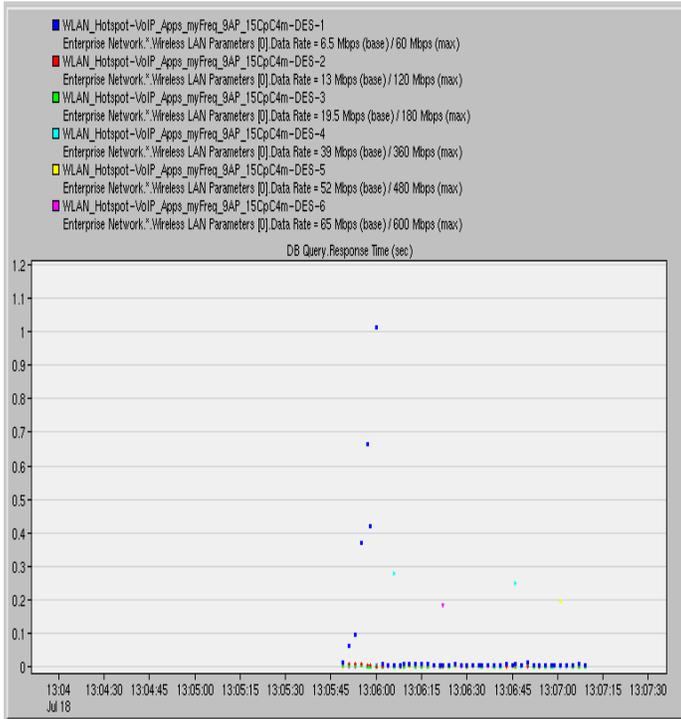


Figure 151: DB Response Time, IP Telephony Call, Applications added in the simulation

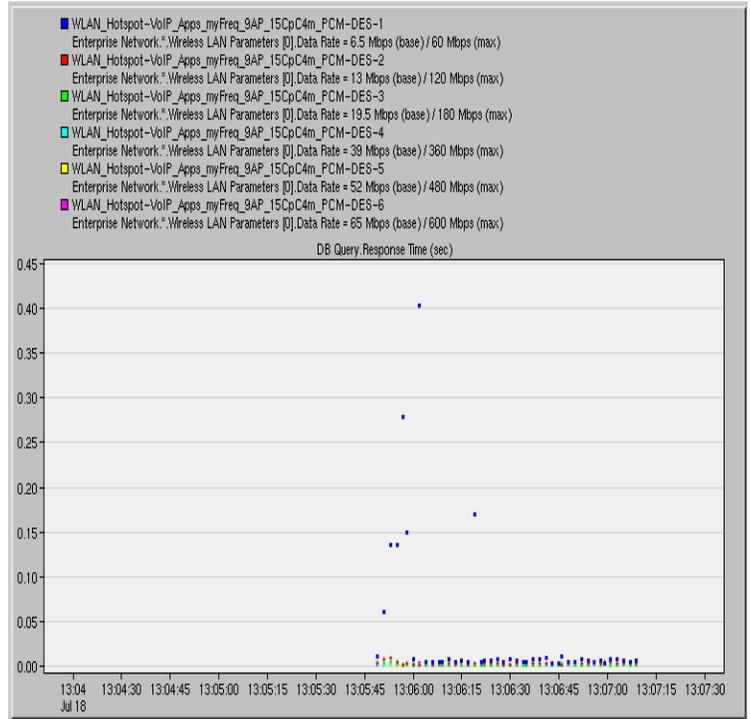


Figure 152: DB Response Time, PCM Quality Call, Applications added in the simulation

4.16.2. VoIP Calls, Applications and Videos Added

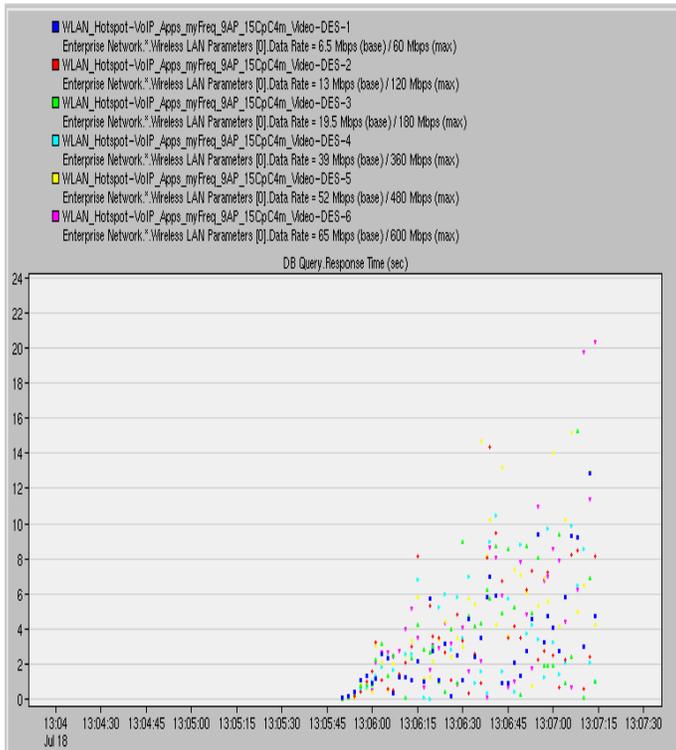


Figure 153: DB Response Time, IP Telephony Call, Video and Applications added in the simulation

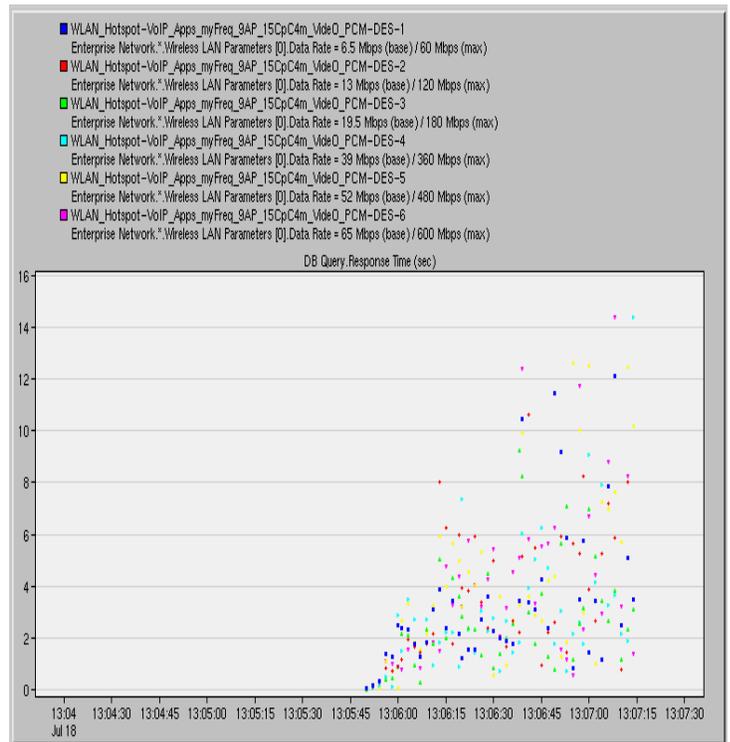


Figure 154: DB Response Time PCM Quality Call, Video and Applications added in the simulation

4.17.16 AP WLAN Deployment – DB Query Response Time for data rates 6.5, 13, 19.5, 39, 65 Mbps

4.17.1. VoIP Calls and Applications Added

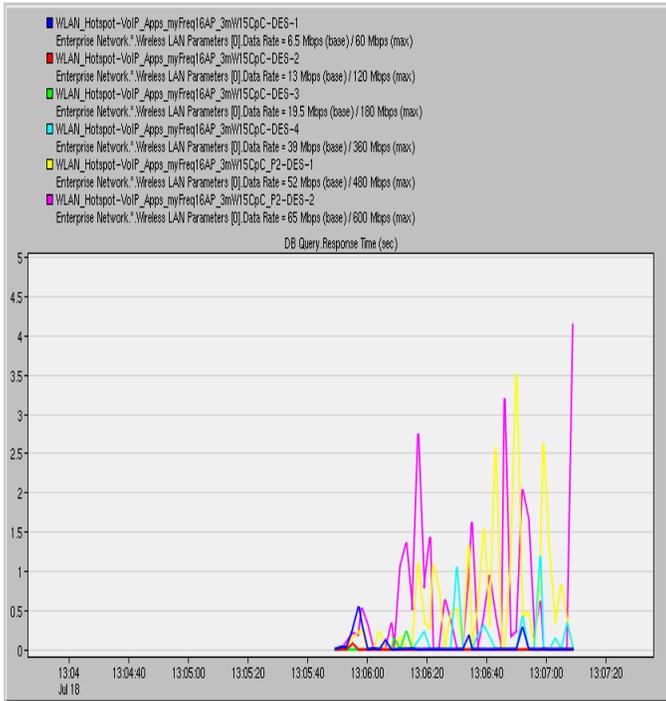


Figure 156: DB Response Time, IP Telephony Call, Applications added in the simulation

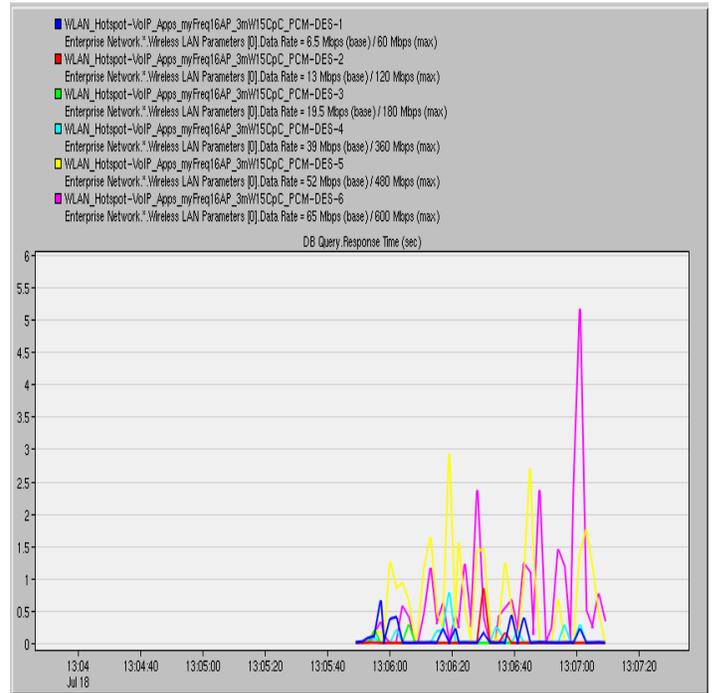


Figure 155: DB Response Time, PCM Quality Call, Applications added in the simulation

4.17.2. VoIP Calls, Applications and Videos Added

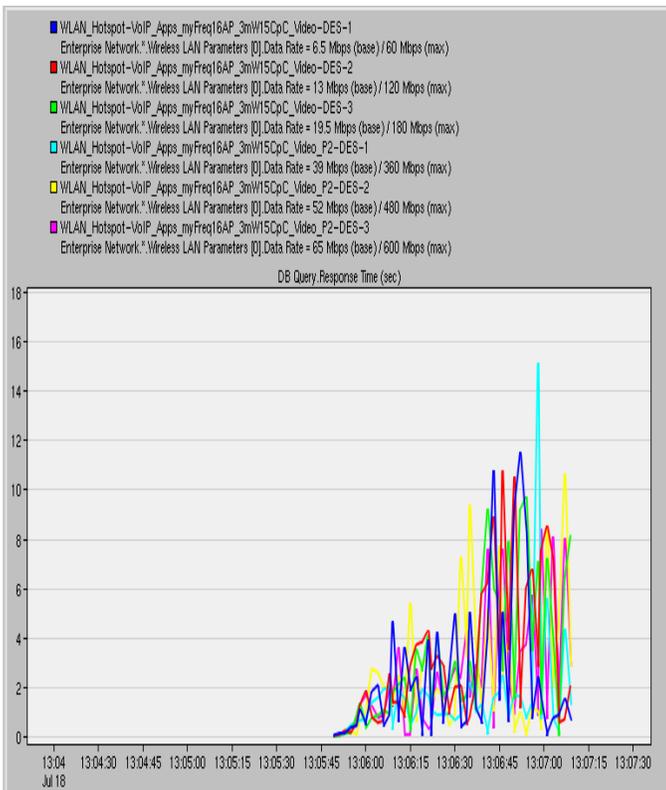


Figure 157: DB Response Time, IP Telephony Call, Video and Applications added in the simulation

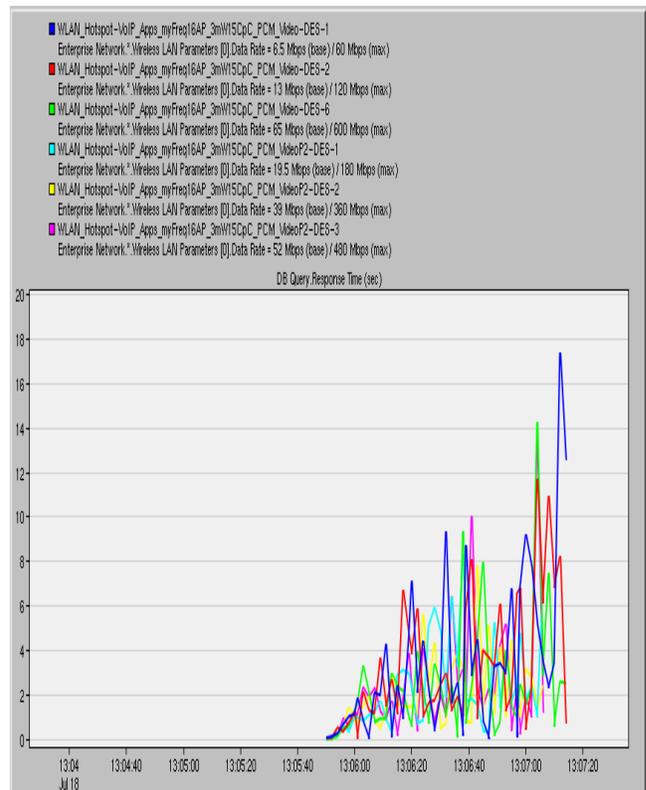


Figure 158: DB Response Time PCM Quality Call, Video and Applications added in the simulation

4.18.9 AP WLAN Deployment – E-mail Download Response Time for data rates 6.5, 13, 19.5, 39, 65 Mbps

4.18.1. VoIP Calls and Applications Added

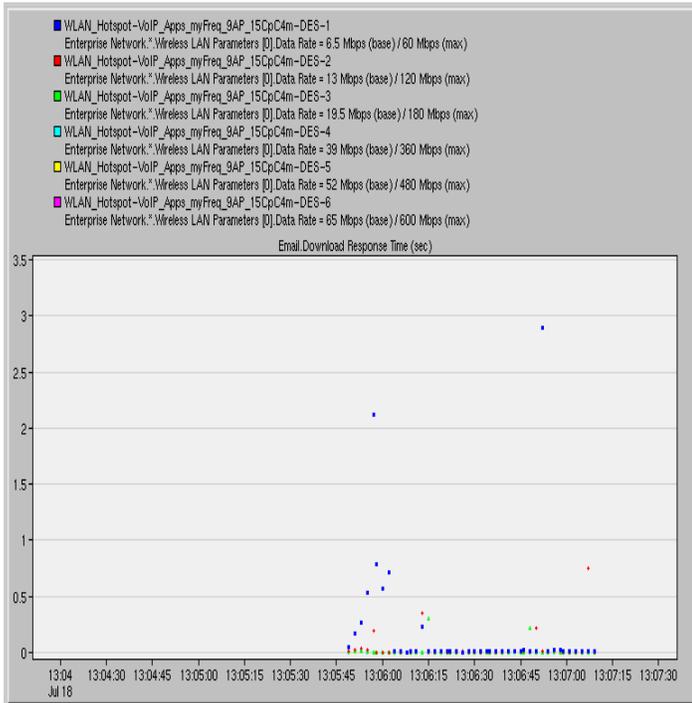


Figure 159: E-mail Download Response Time, IP Telephony Call, Applications added in the simulation

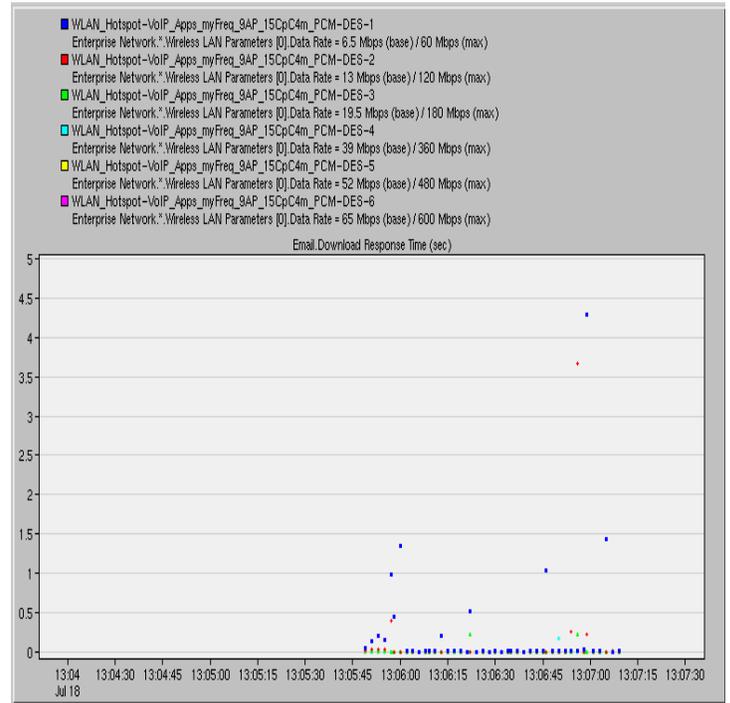


Figure 160: E-mail Download Response Time, PCM Quality Call, Applications added in the simulation

4.18.2. VoIP Calls, Applications and Videos Added

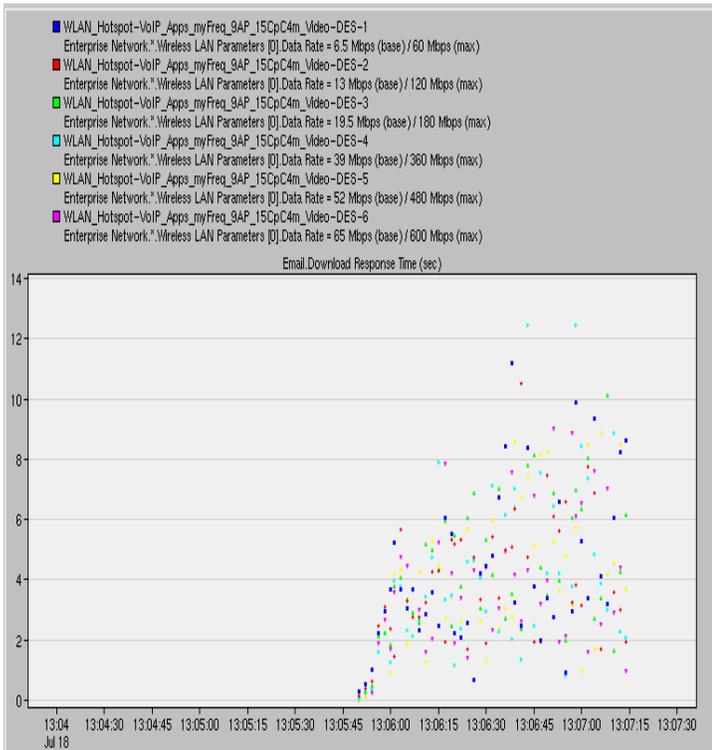


Figure 161: E-mail Download Response Time, IP Telephony Call, Video and Applications added in the simulation

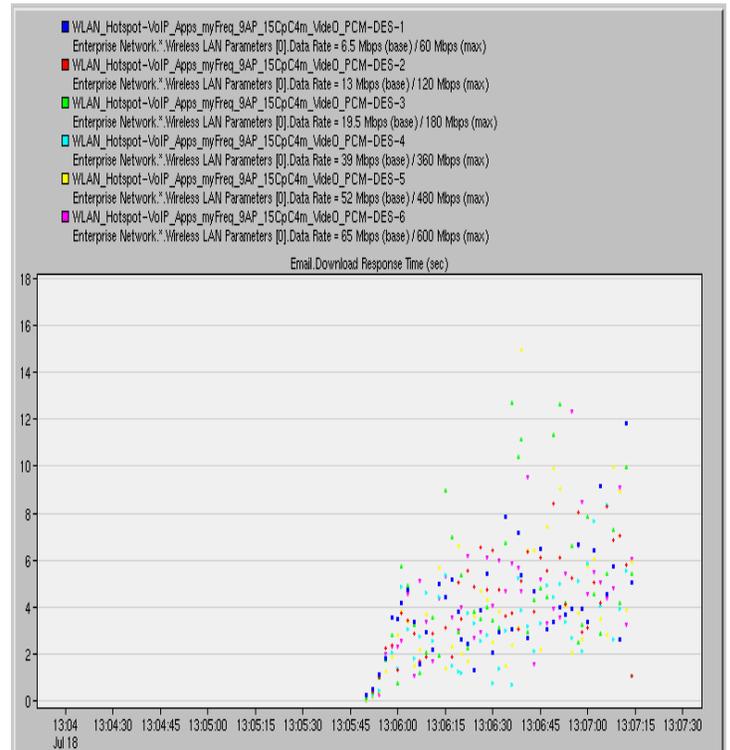


Figure 162: E-mail Download Response Time, PCM Quality Call, Video and Applications added in the simulation

4.19.16 AP WLAN Deployment – E-mail Download Response Time for data rates 6.5, 13, 19.5, 39, 65 Mbps

4.19.1. VoIP Calls and Applications Added

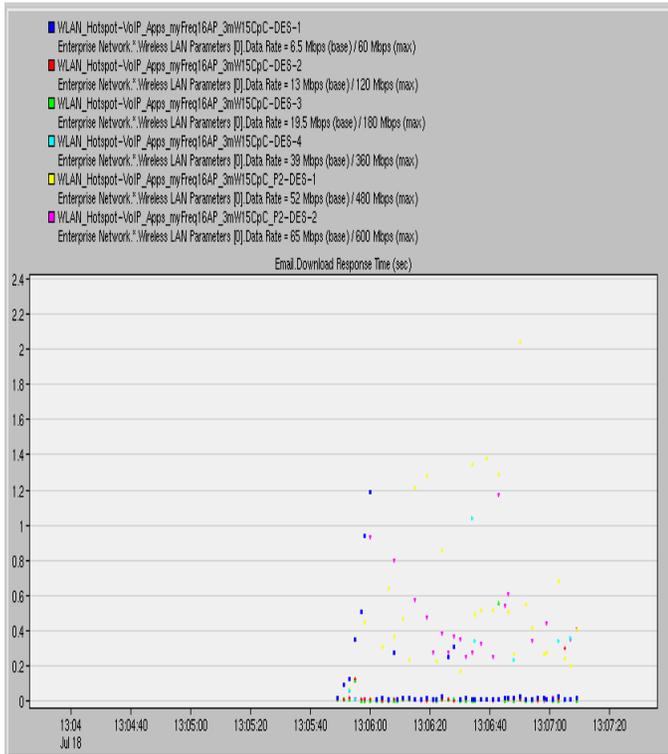


Figure 164: E-mail Download Response Time, IP Telephony Call, Applications added in the simulation

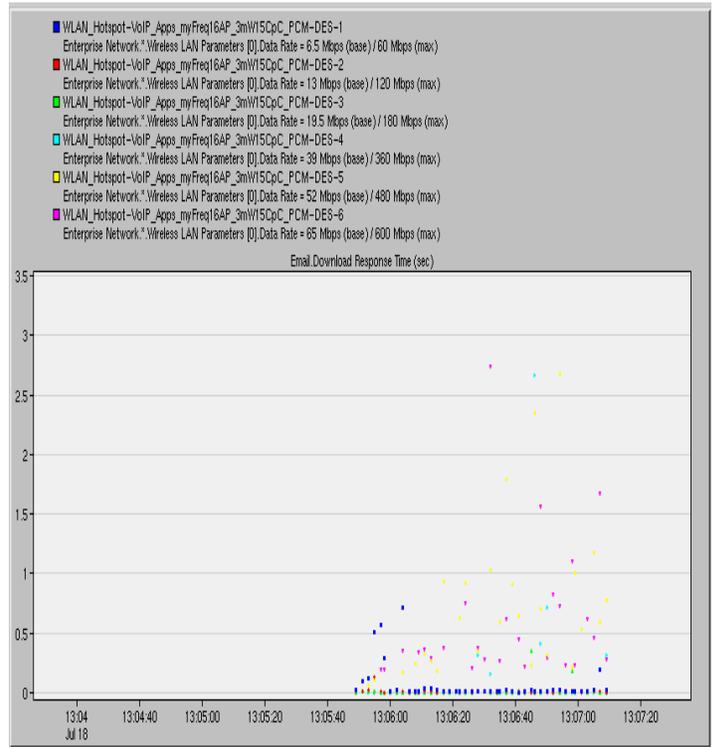


Figure 163: E-mail Download Response Time, PCM Quality Call, Applications added in the simulation

4.19.2. VoIP Calls, Applications and Videos Added

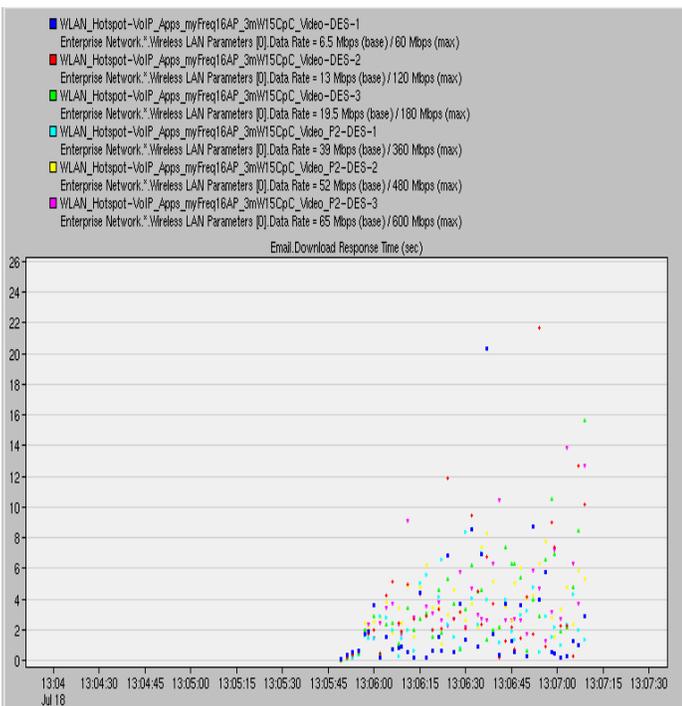


Figure 165: E-mail Download Response Time, IP Telephony Call, Video and Applications added in the simulation

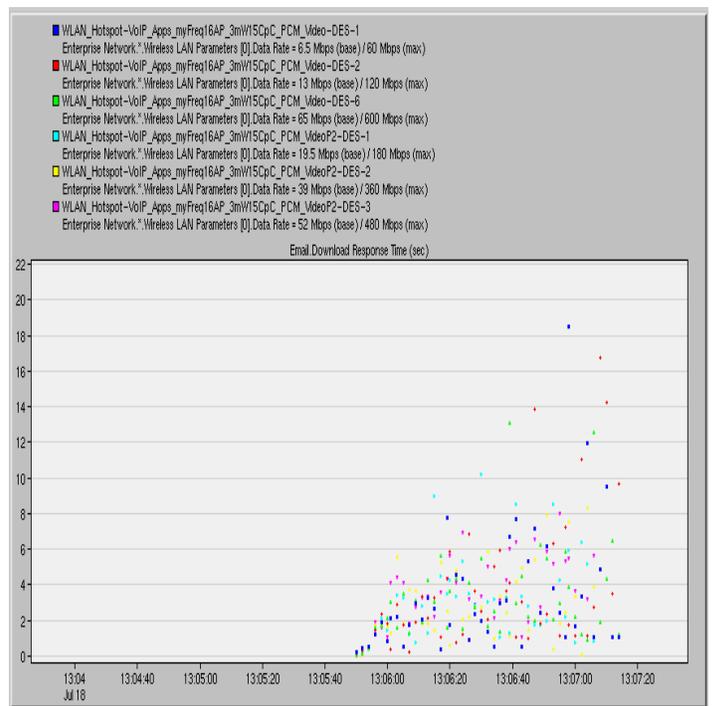


Figure 166: E-mail Download Response Time, PCM Quality Call, Video and Applications added in the simulation

4.20.9 AP WLAN Deployment – FTP Download Response Time for data rates 6.5, 13, 19.5, 39, 65 Mbps

4.20.1. VoIP Calls and Applications Added

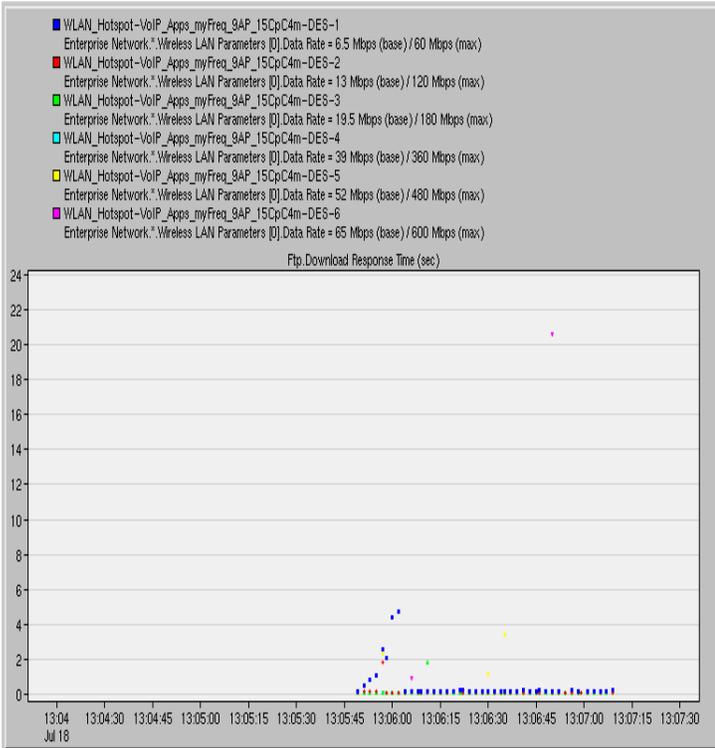


Figure 168: FTP Download Response Time, IP Telephony Call, Applications added in the simulation

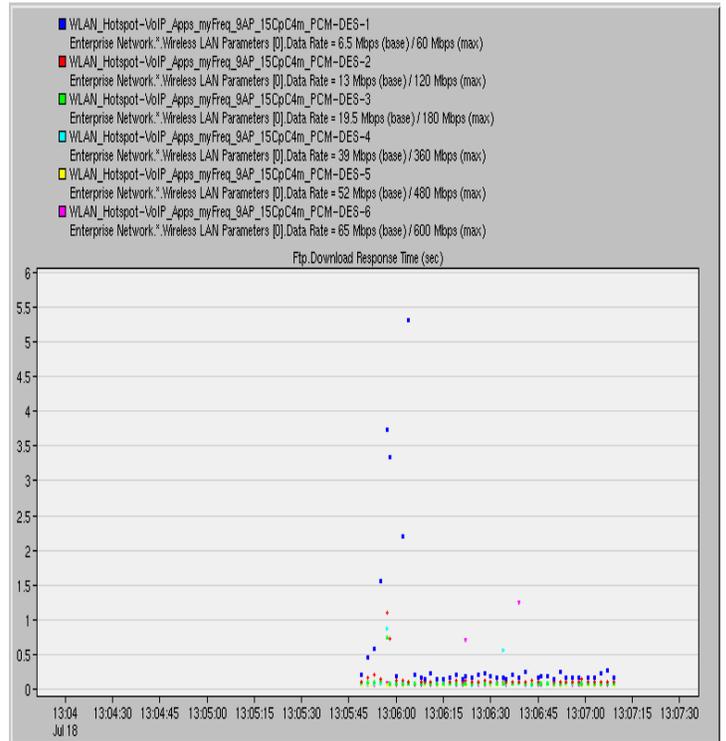


Figure 167: FTP Download Response Time, PCM Quality Call, Applications added in the simulation

4.20.2. VoIP Calls, Applications and Videos Added

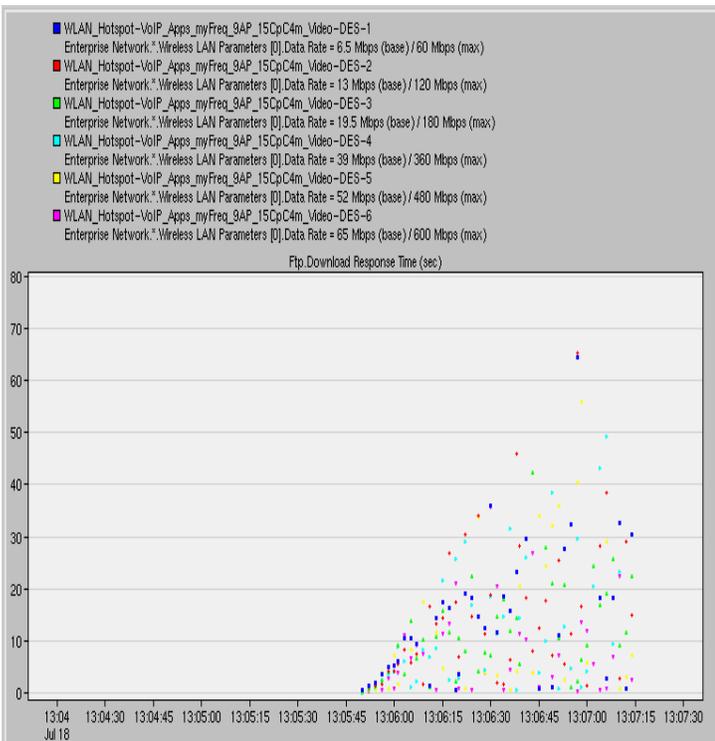


Figure 169: FTP Download Response Time, IP Telephony Call, Video and Applications added in the simulation

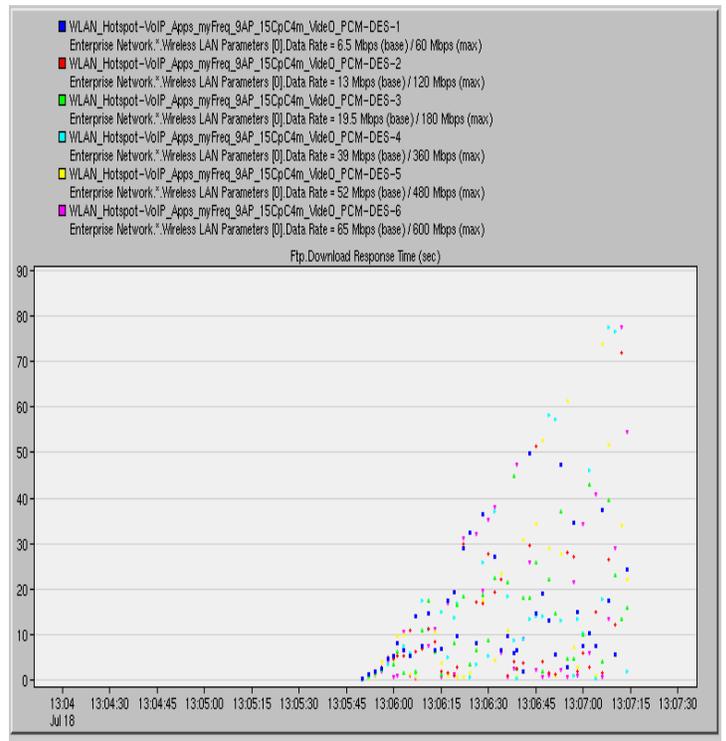


Figure 170: FTP Download Response Time, PCM Quality Call, Video and Applications added in the simulation

4.21.16 AP WLAN Deployment – FTP Download Response Time for data rates 6.5, 13, 19.5, 39, 65 Mbps

4.21.1. VoIP Calls and Applications Added

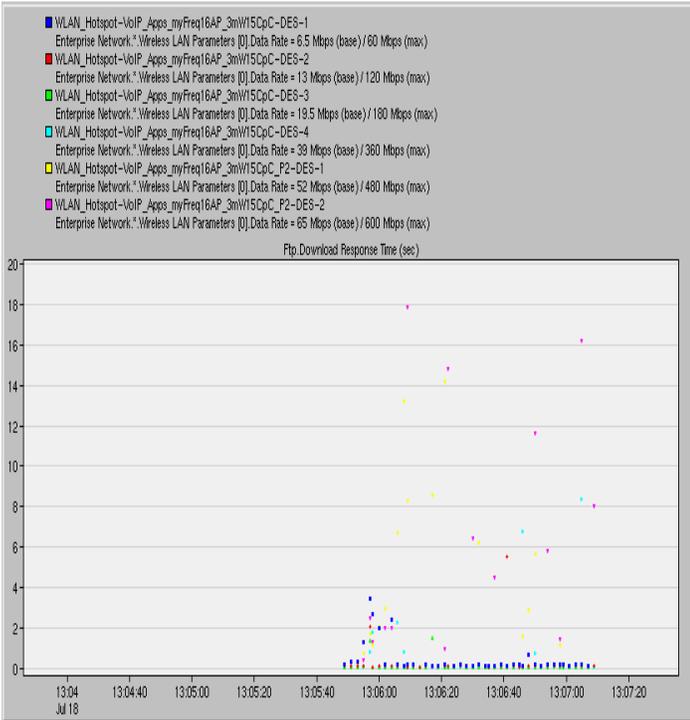


Figure 172: FTP Download Response Time, IP Telephony Call, Applications added in the simulation

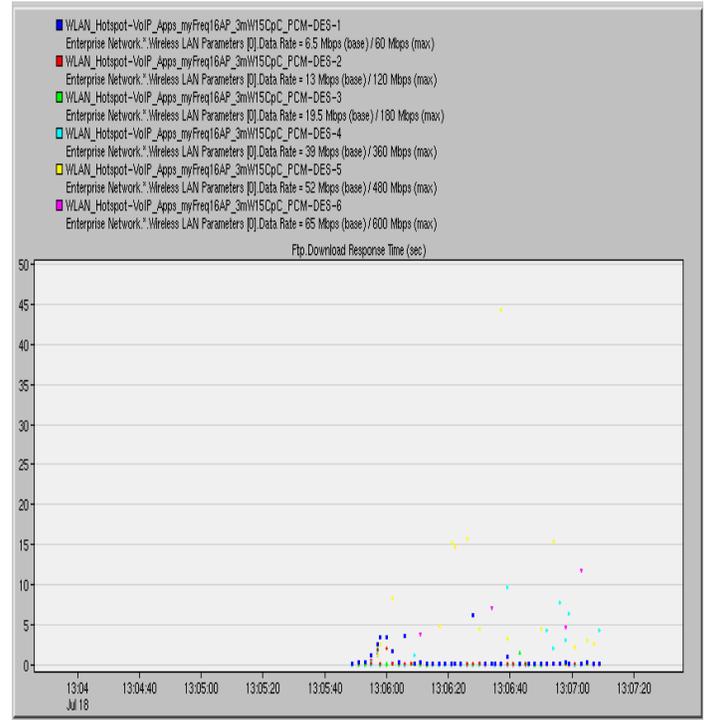


Figure 171: FTP Download Response Time, PCM Quality Call, Applications added in the simulation

4.21.2. VoIP Calls, Applications and Videos Added

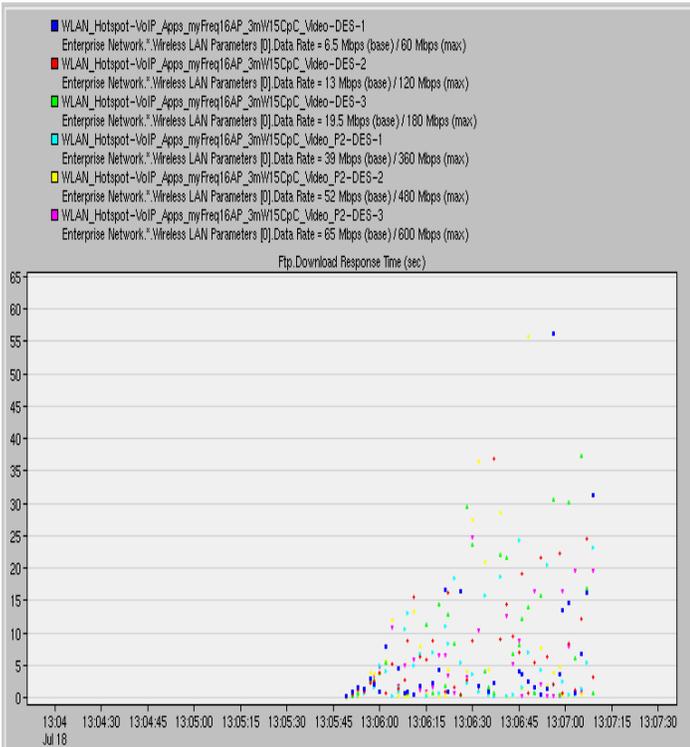


Figure 173: FTP Download Response Time, IP Telephony Call, Video and Applications added in the simulation

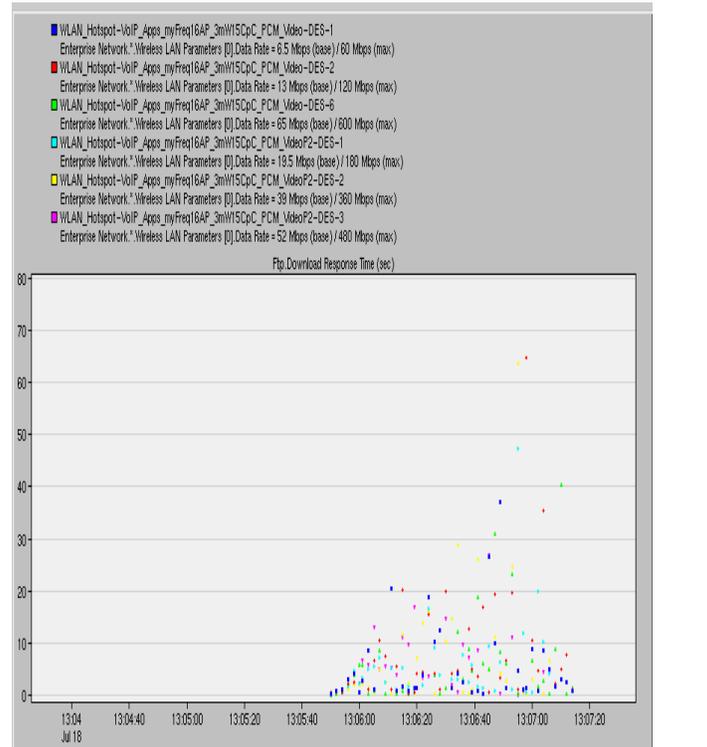


Figure 174: FTP Download Response Time, PCM Quality Call, Video and Applications added in the simulation

4.22.9 AP WLAN Deployment – HTTP Page Response Time for data rates 6.5, 13, 19.5, 39, 65 Mbps

4.22.1. VoIP Calls and Applications Added

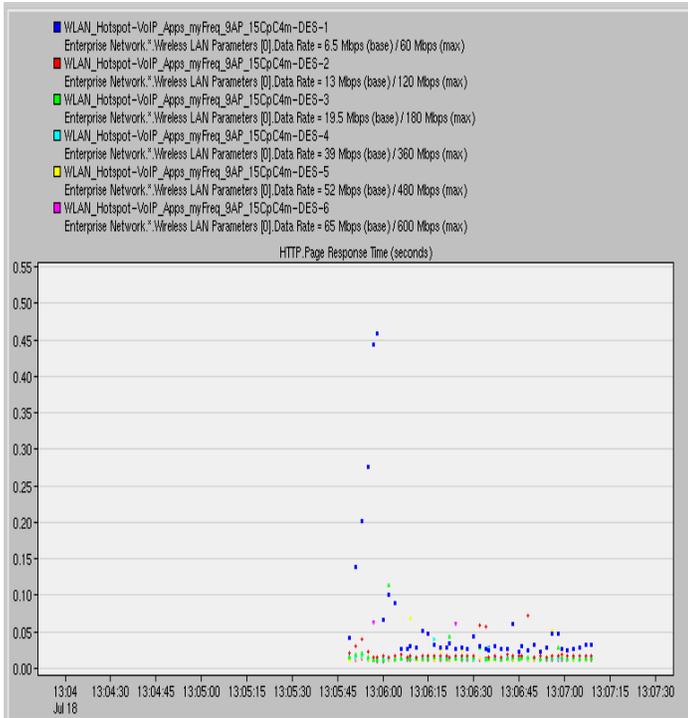


Figure 176: HTTP Page Response Time, IP Telephony Call, Applications added in the simulation

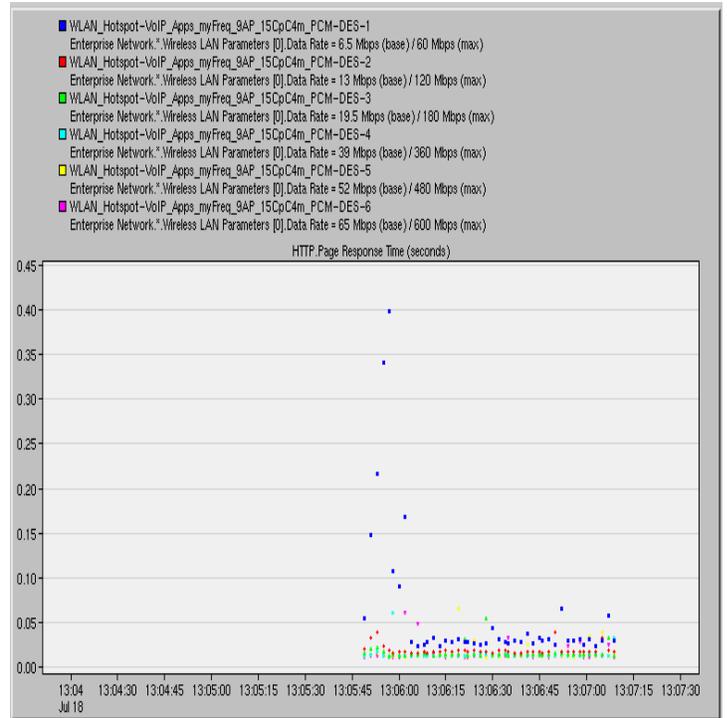


Figure 175: HTTP Page Response Time, PCM Quality Call, Applications added in the simulation

4.22.2. VoIP Calls, Applications and Videos Added

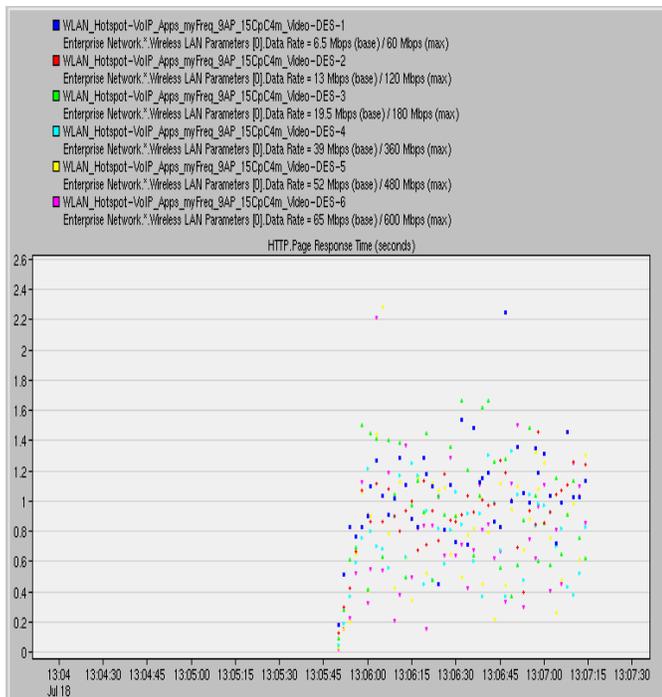


Figure 177: HTTP Page Response Time, IP Telephony Call, Video and Applications added in the simulation

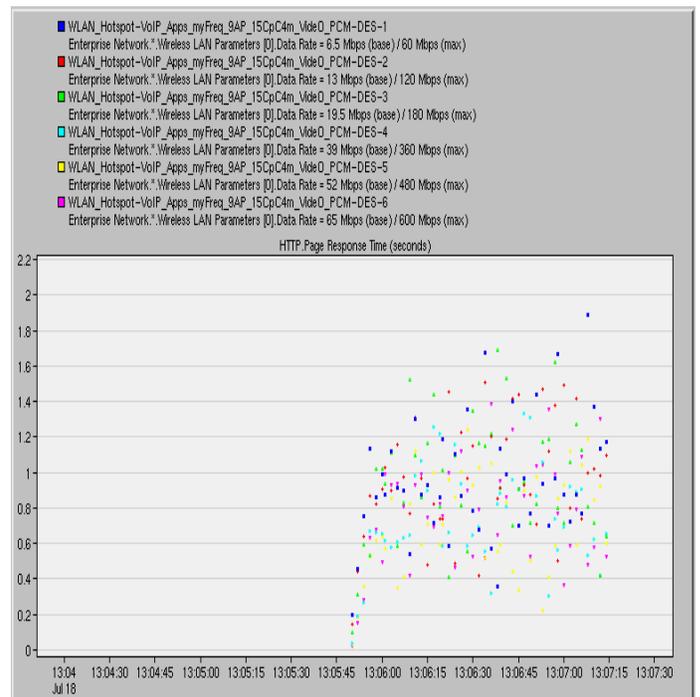


Figure 178: HTTP Page Response Time, PCM Quality Call, Video and Applications added in the simulation

4.23.16 AP WLAN Deployment – HTTP Page Response Time for data rates 6.5, 13, 19.5, 39, 65 Mbps

4.23.1. VoIP Calls and Applications Added

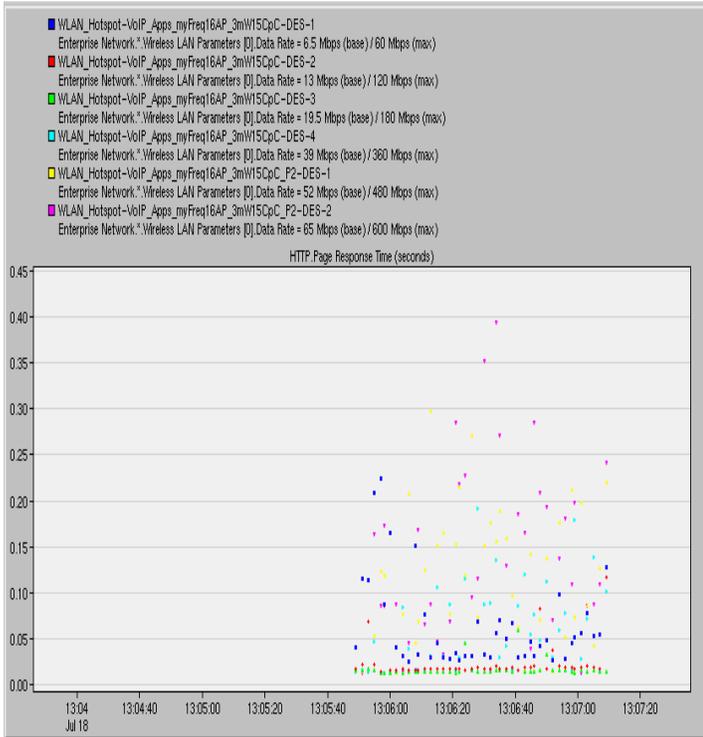


Figure 180: HTTP Page Response Time, IP Telephony Call, Applications added in the simulation

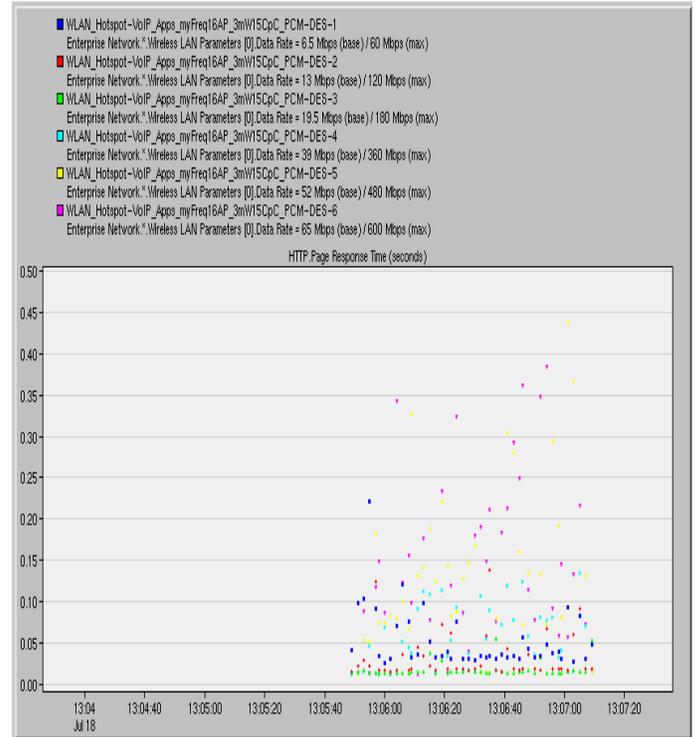


Figure 179: HTTP Page Response Time, PCM Quality Call, Applications added in the simulation

4.23.2. VoIP Calls, Applications and Videos Added

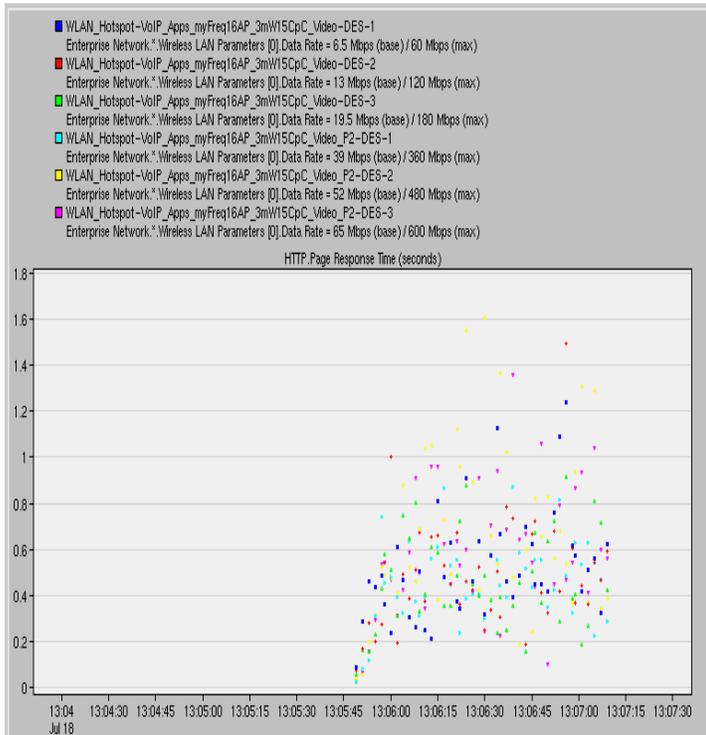


Figure 181: HTTP Page Response Time, IP Telephony Call, Video and Applications added in the simulation

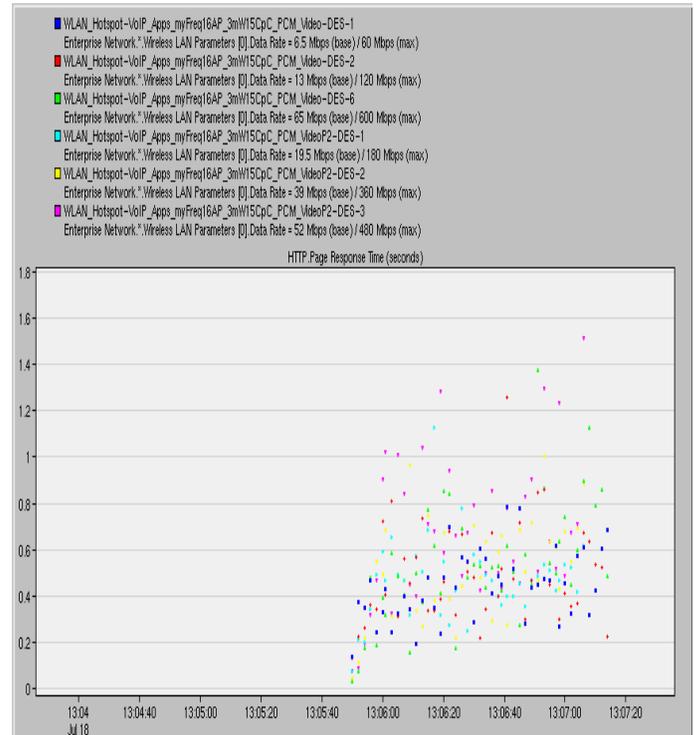


Figure 182: HTTP Page Response Time, PCM Quality Call, Video and Applications added in the simulation

4.24.9 AP WLAN Deployment , P2P Links Deployed –Global Statistics

For the 9 AP WLAN Deployment with P2P links deployed we first present some Global Statistics, i.e., the Voice Traffic Sent and Received, the Packet End - To - End Delay and the Throughput for the entire network. As we have explained how these results are calculated, we will only note that we can observe the global Traffic Received and the Throughput increasing as more calls are added to the P2P links.

4.24.1. Voice Traffic Sent

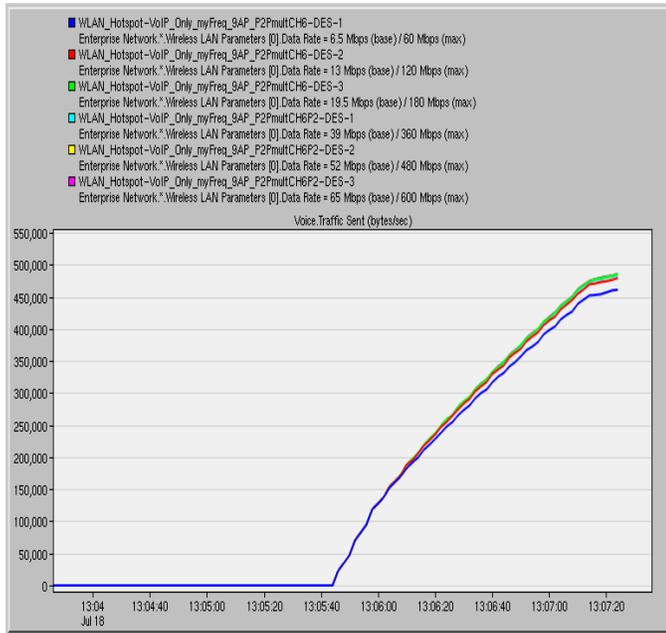


Figure 184: Global Traffic Sent – P2P Links Operating On Channel 6, 5 Calls Per P2P Link

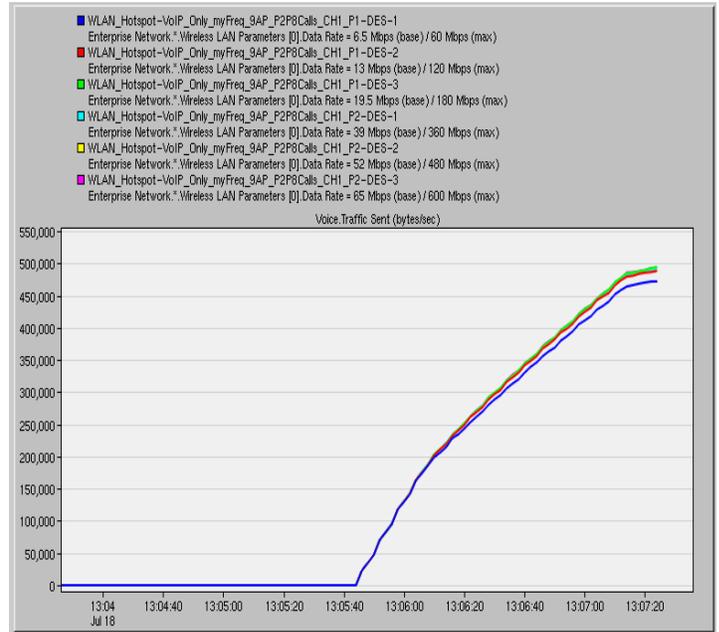


Figure 183: Global Traffic Sent – P2P Links Operating On Channel 1, 8 Calls Per P2P Link

4.24.2. Voice Traffic Received

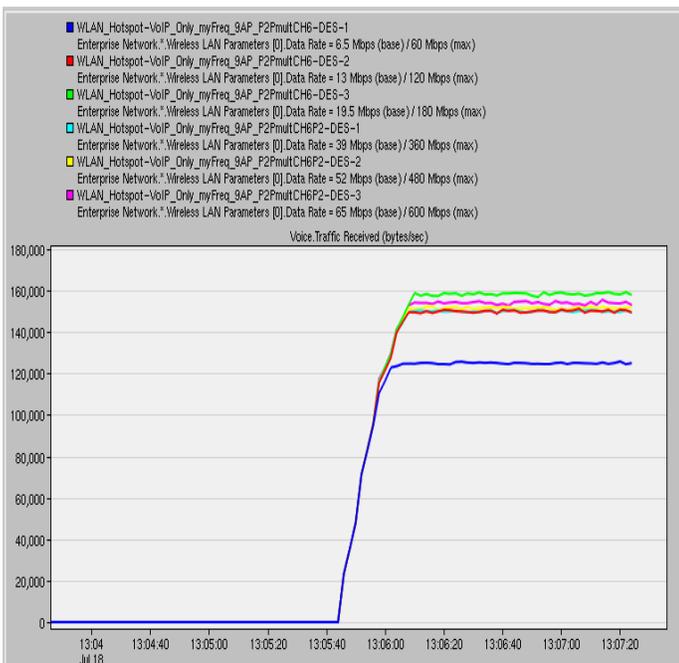


Figure 185: Global Traffic Received – P2P Links Operating On Channel 6, 5 Calls Per P2P Link

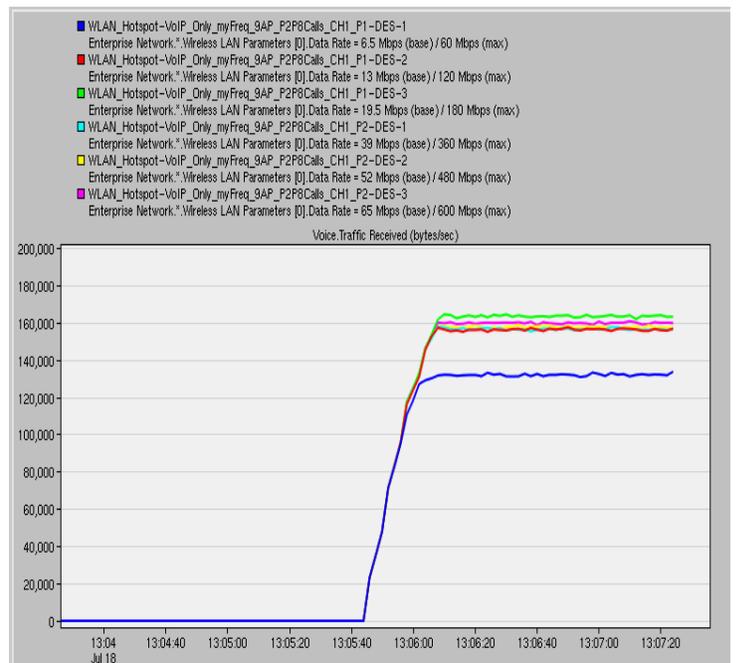


Figure 186: Global Traffic Received – P2P Links Operating On Channel 1, 8 Calls Per P2P Link

4.24.3. Voice Packet End – To – End Delay

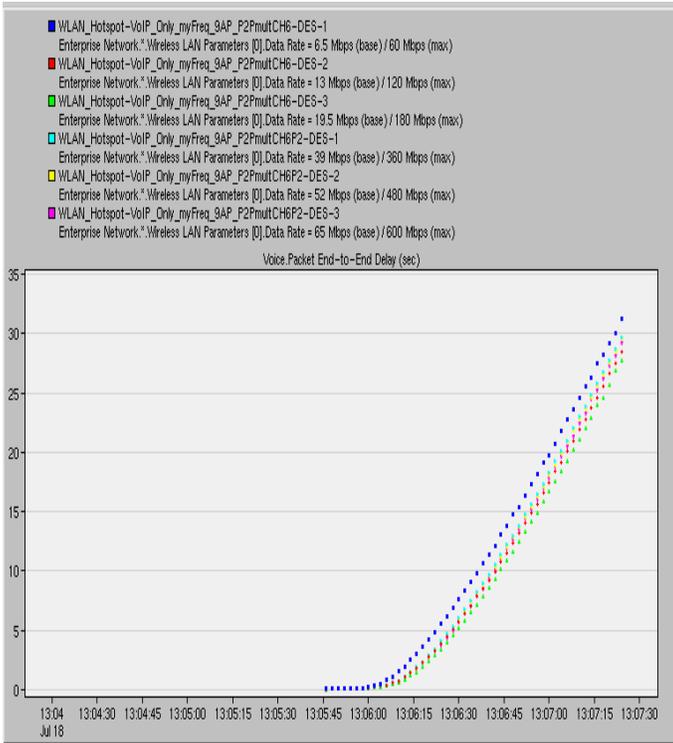


Figure 187: Global Voice Packet End –To – End Delay, P2P Links Operating On Channel 6, 5 Calls Per P2P Link

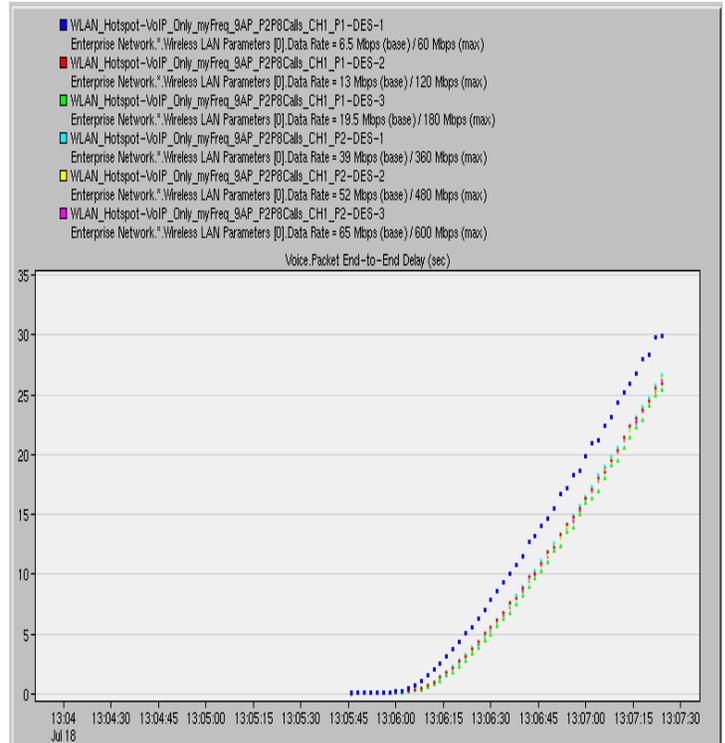


Figure 188: Global Voice Packet End –To – End Delay, P2P Links Operating On Channel 1, 8 Calls Per P2P Link

4.24.4. Throughput

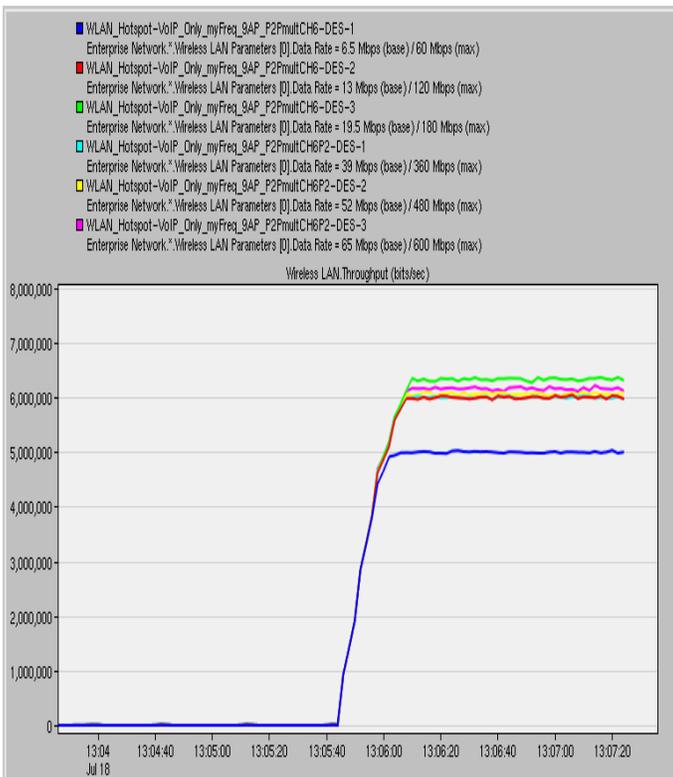


Figure 190: Global Throughput – P2P Links Operating On Channel 6, 5 Calls Per P2P Link

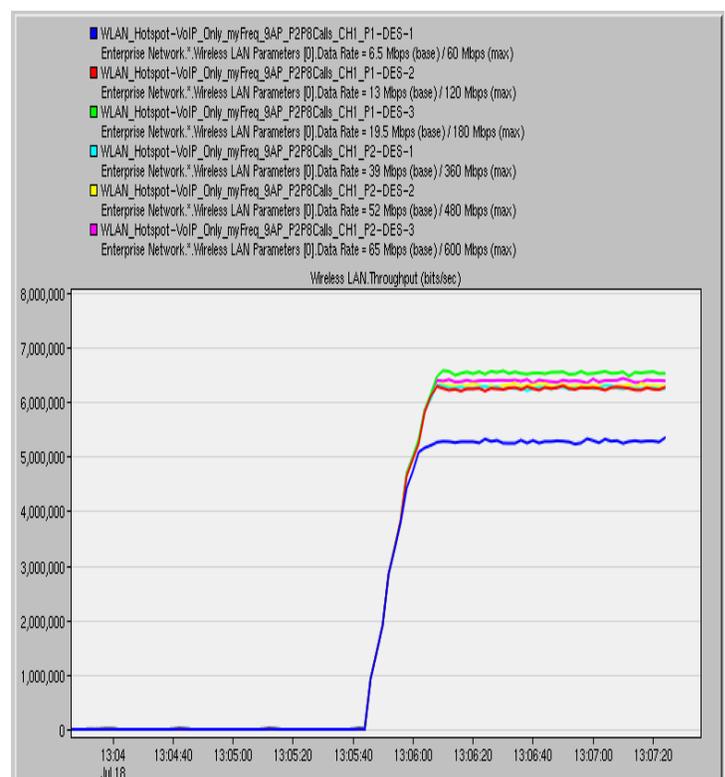


Figure 189: Global Throughput – P2P Links Operating On Channel 1, 8 Calls Per P2P Link

4.25. AP WLAN Deployment , P2P Links Deployed –Node Statistics

Here, we will mainly focus on the Node Statistics. We choose to examine the call capacity for the 65Mbps data rate as, from the global statistic Voice Traffic Received, it appears to show a satisfying behavior. Remember that on both the workstations and the mobile nodes IP Telephony VoIP calls are deployed. The calls on the mobile phones keep the Interactive Voice priority while the calls added on the existing workstations are configured to Best Effort so that we can see the effect of the mobile P2P phone calls on the voice capacity of the existing infrastructure.

In Figures 195 – 197, we observe that in the initial scenario, almost 14 IP Telephony Calls could be added on every workstation. When the P2P links are added the calls that can be accommodated by the workstations drop to 7-8. This can be explained by the change in priority from Interactive Voice to Best Effort for the workstations. However the workstations 2, 4, 8 operating on the same channel (Channel 6) as the P2P links, constitute an exception as they can accommodate only up to 4 calls. The same applies on the experiment where 8 calls are added on P2P links that operate on channel 1. Now call dropping occurs on the workstations 1, 6, 7 operating on the same channel (Channel 1). At the beginning of the experiment 4-5 calls can be added to P2P links but this traffic drops as more P2P VoIP calls are added.

4.25.4. Traffic Received From The P2P Links

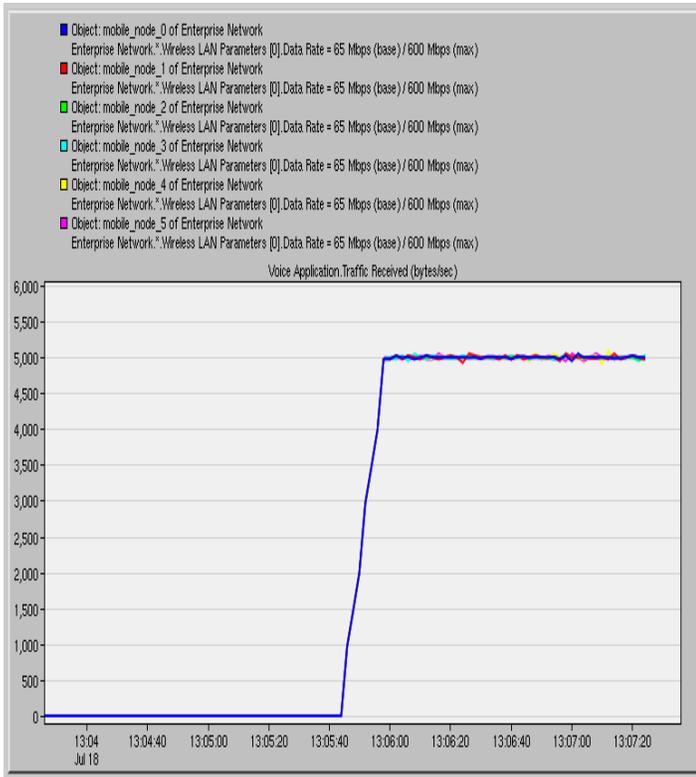


Figure 198a: Traffic Received From the P2P Links – P2P Links Operating On Channel 6, 5 Calls Per P2P Link

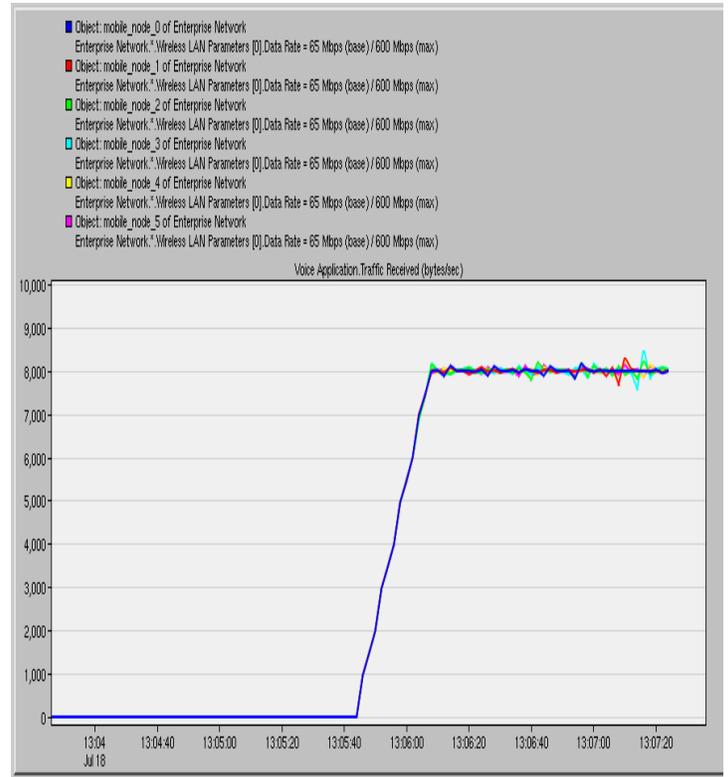


Figure 198b: Traffic Received From the P2P Links – P2P Links Operating On Channel 1, 8 Calls Per P2P Link

In Figures 198a and 198b, the voice traffic received from the P2P Links (all the mobile nodes) matches the voice traffic sent, thus all the P2P voice calls are added successfully, due to the Interactive Voice Priority they have. We can see minor differences that are more noticeable in the experiment with the 8 calls added – their impact on the quality can be interpreted by the following statistics.

4.25.5. MOS – Workstations

In Figures 199 – 201, the change in the workstations' priority can be seen by comparing the Mean Opinion Score (MOS) of the deployment before (Figure 199) and after (Figures 200 – 201) the change of VoIP traffic to Best Effort. Also, we can see that the MOS is even worse for a workstation when P2P links operate in the same channel as the workstation.

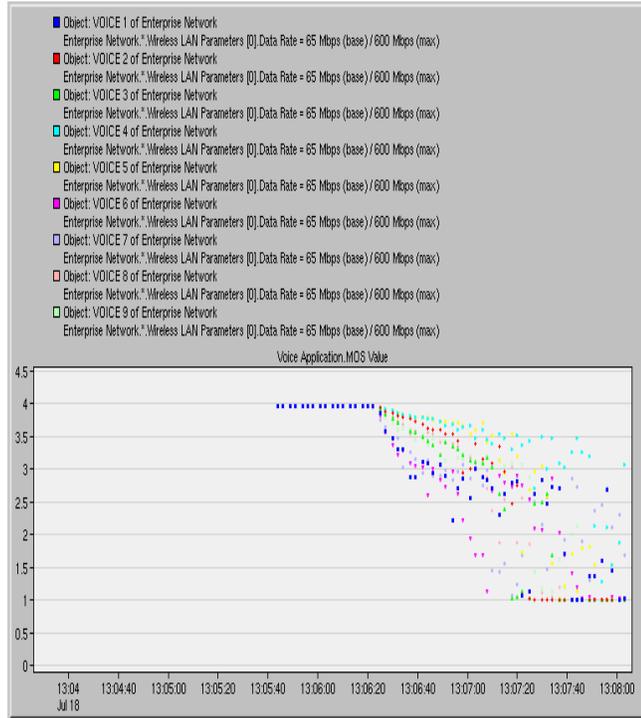


Figure 199: MOS For Each Workstation – No P2P Links

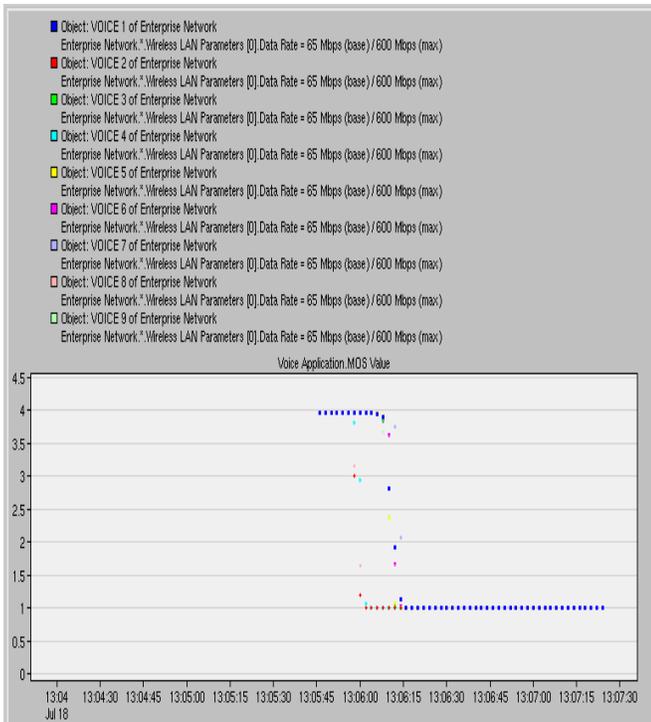


Figure 200: MOS For Each Workstation – P2P Links Operating On Channel 6, 5 Calls Per P2P Link

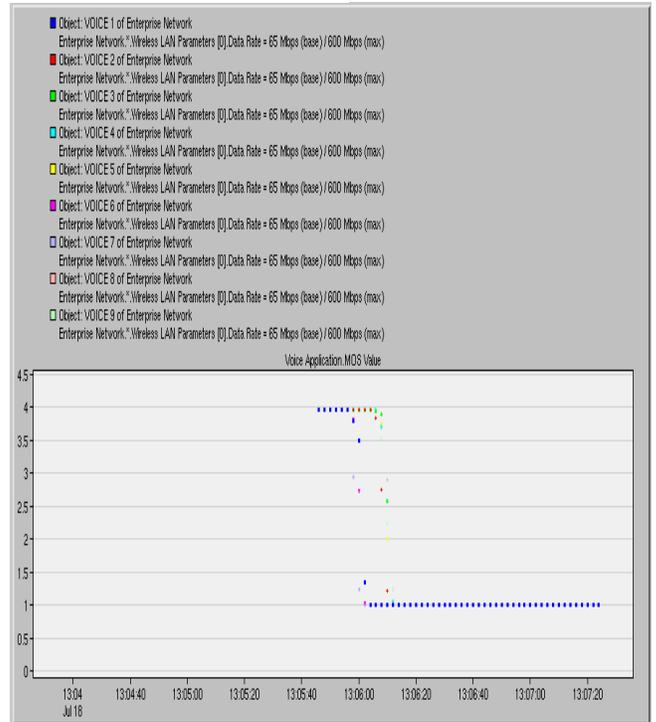


Figure 201: MOS For Each Workstation – P2P Links Operating On Channel 1, 8 Calls Per P2P Link

4.25.6. Packet End to End Delay – Workstations

By observing the voice packet end to end delay we can link the previous results for MOS with this statistic. Remember that a voice call is very sensitive to delay and the International Telecommunication Union (ITU) has recommended that one-way end-to-end delay should be no greater than 150 ms for good voice quality. Notice that the workstation 1, which is in cell 1 and operates on channel 1, while the P2P links on this experiment operate on channel 6, has a good MOS value of 4 as the packet end to end delay remains below 150 ms for 30 seconds. On the second, experiment the call lasts only for 15 seconds due to the impact of the P2P link on the same channel.

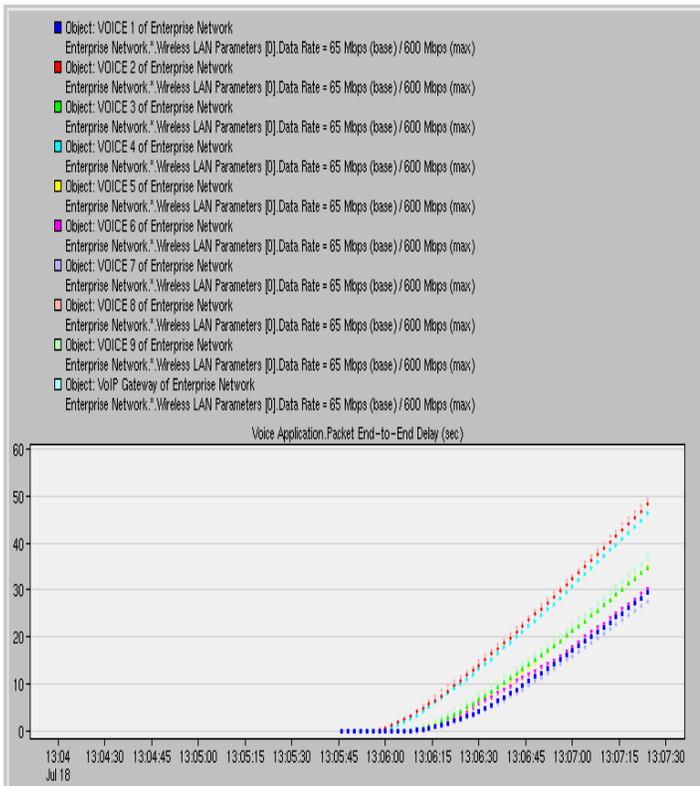


Figure 203: Packet End To End Delay For Each Workstation – P2P Links Operating On Channel 6, 5 Calls Per P2P Link

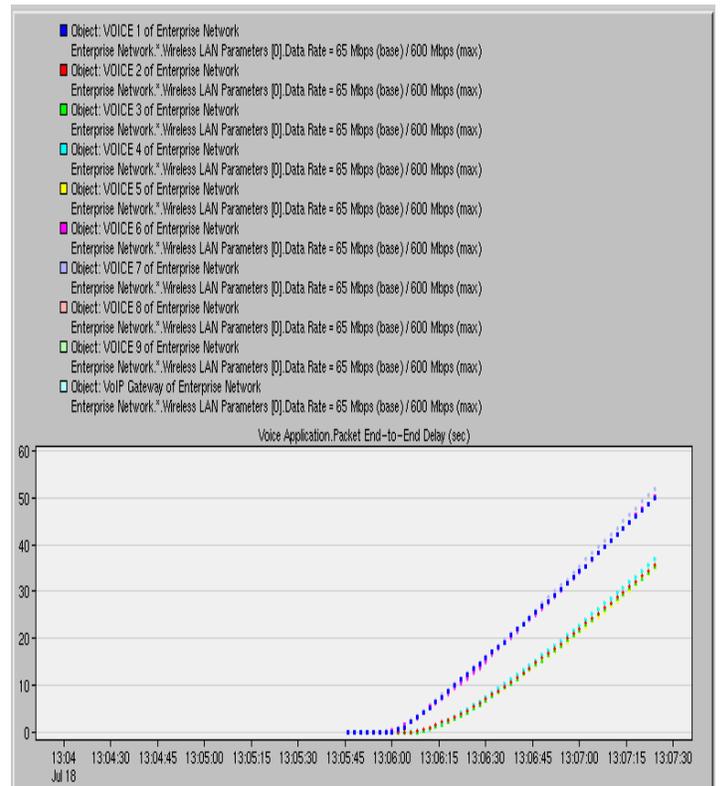


Figure 202: Packet End To End Delay For Each Workstation - P2P Links Operating On Channel 1, 8 Calls Per P2P Link

4.25.7. MOS – P2P Links

As shown in Figures 204 – 207 the MOS and Packet End To End Delay results are not disappointing for the P2P links as they were for the workstations. The reason is again the higher priority (Interactive Voice) offered to VoIP traffic in P2P links. In this case the MOS values do not drop below 3.9 and 3.4 when we add 5 calls and 8 calls respectively and they can reach up to 3.97 for the entire simulation period. The Packet End To End Delay results confirm this good performance as they do not exceed the 67 ms and the 80 ms. Notice that the more calls we add on the P2P links the worst are the results in the performance consideration statistics, MOS and Packet End To End Delay.

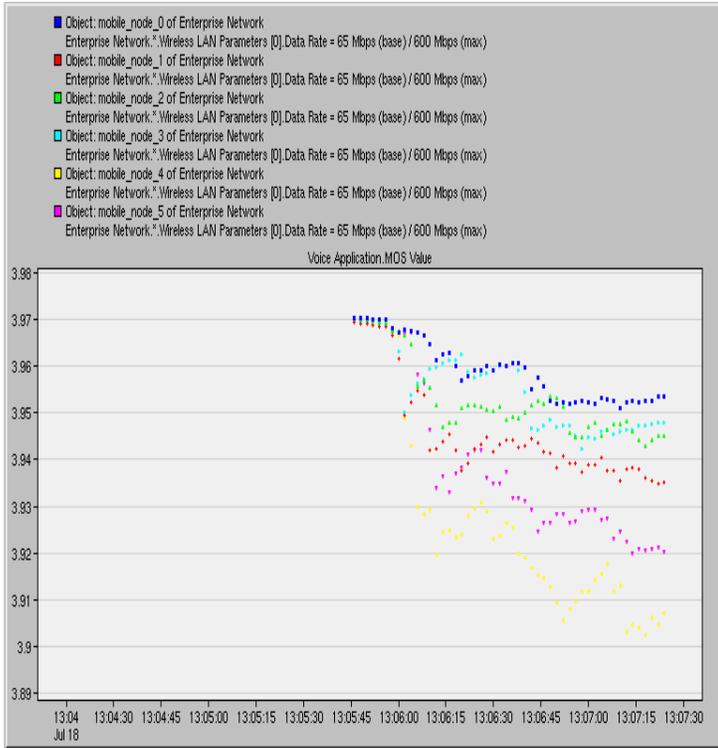


Figure 205: MOS For Each Mobile Node – P2P Links Operating On Channel 6, 5 Calls Per P2P Link

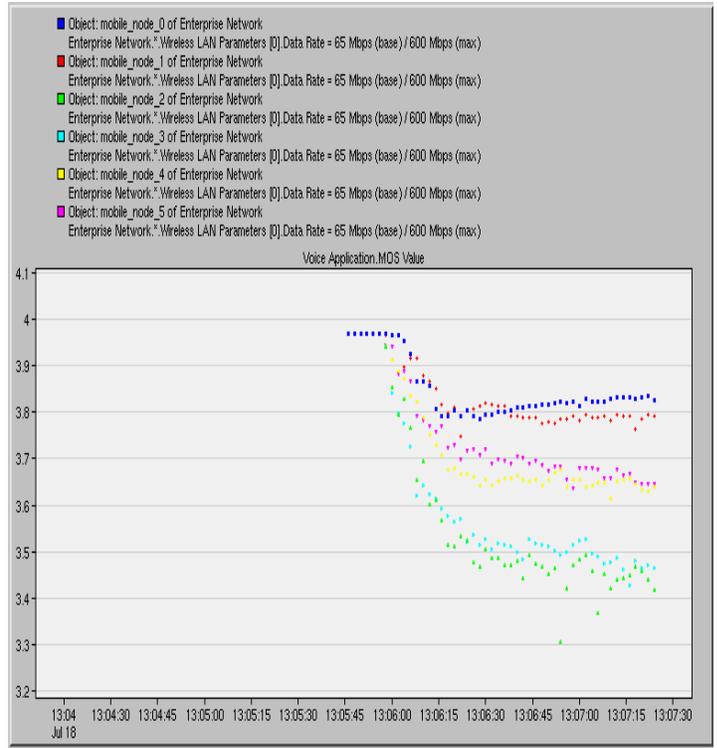


Figure 204: MOS For Each Mobile Node P2P Links Operating On Channel 1, 8 Calls Per P2P Link

4.25.8. Packet End to End Delay – P2P Links

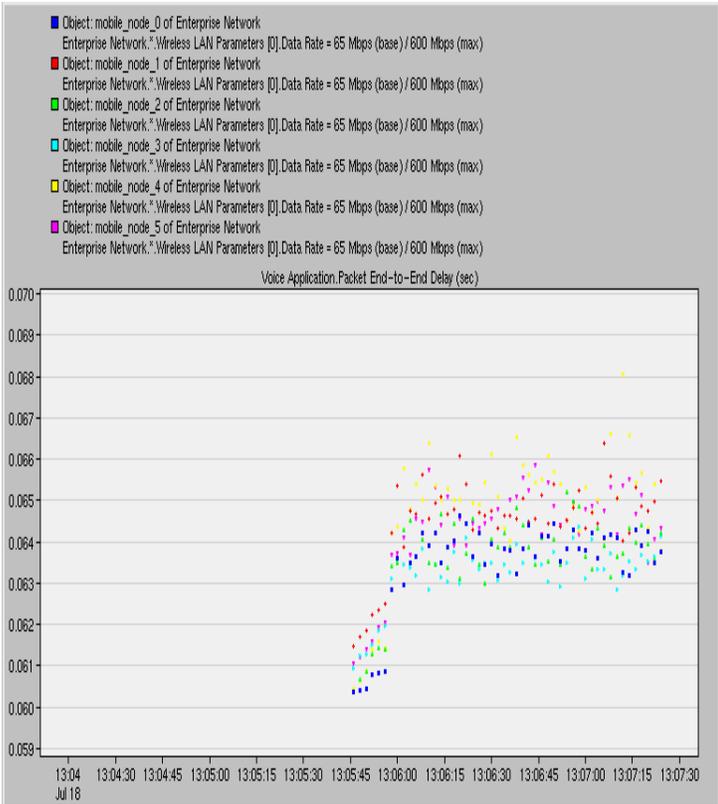


Figure 206: Packet End To End Delay For Each Mobile Node – P2P Links Operating On Channel 6, 5 Calls Per P2P Link

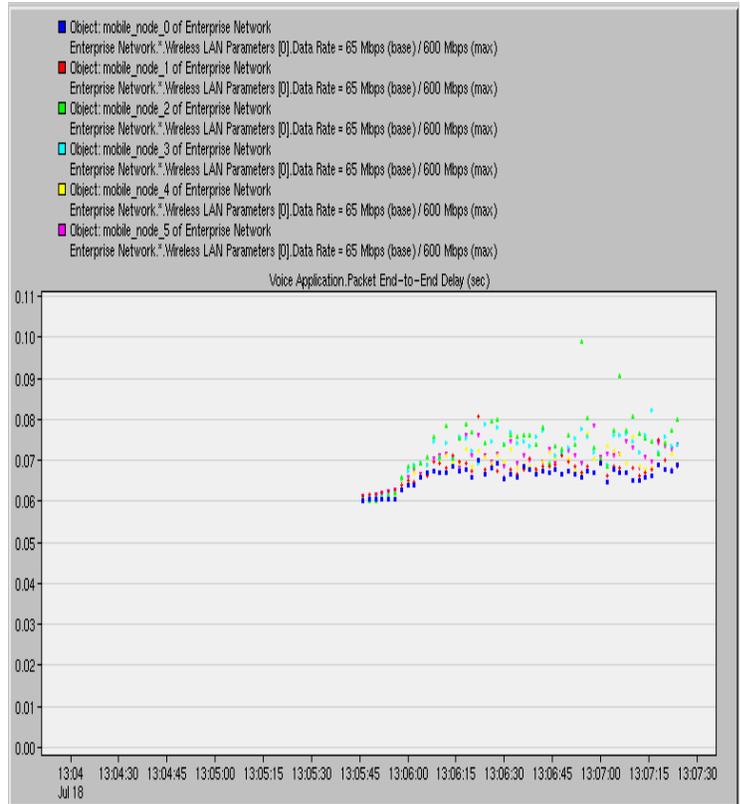


Figure 207: Packet End To End Delay For Each Mobile Node - P2P Links Operating On Channel 1, 8 Calls Per P2P Link

Chapter 5: Conclusions

This work presents a detailed approach on designing and deploying a large-scale WLAN deployment. It includes WLAN frequency planning for voice over WLAN, the study of a microcell deployment strategy and its simulation in OPNET – Modeler.

After the simulation framework is built, various applications and pairs of P2P links are added on the overlaid infrastructures. Via a large number of simulations, we observe that the addition of HTTP, FTP, E-mail, Database Query and Print applications reduces the number of VoIP calls that the network can support. The addition of video applications causes a serious degradation of the VoIP performance and for its deployment a new network design approach should be considered. An important conclusion of our study is that the channel capacity and the interference are the limiting factors in terms of the throughput received and not the number of the APs in a WLAN network.

Lastly, with the introduction of P2P links on the network, the number of VoIP also decreases, as their addition is proved strenuous for an already congested radio frequency spectrum.

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