STATUS OF THESIS

| Title of thesis | VOIP WITH ADAPTIVE RATE<br>IN MULTI- TRANSMISSION RATE WIRELESS LANS |
|---|---|

I, HANIYEH KAZEMITABAR

hereby allows my thesis to be placed at the Information Resource Center (IRC) of Universiti Teknologi PETRONAS (UTP) with the following conditions:

1. The thesis becomes the property of UTP
2. The IRC of UTP may make copies of the thesis for academic purposes only.
3. This thesis is classified as

☐ Confidential

☑ Non-confidential

If this thesis is confidential, please state the reason:

___

___

___

The contents of the thesis will remain confidential for ___________ years.

Remarks on disclosure:

___

___

___

Endorsed by

___________________________ ___________________________

Signature of Author Signature of Supervisor

Permanent Address:

No. 8, Sarsabz Deadend, Sazmane Aab Street,Sheikh FazlolAllah Highway, 1454643381, Tehran, IRAN

Date: ___________________________

Name of Supervisor:

Assoc. Prof. Dr. Abas Md. Said

Date: ___________________________

UNIVERSITI TEKNOLOGI PETRONAS

VOIP WITH ADAPTIVE RATE IN MULTI- TRANSMISSION RATE WIRELESS LANS

by

HANIYEH KAZEMITABAR

The undersigned certify that they have read, and recommend to the Postgraduate Studies Programme for acceptance of this thesis for the fulfillment of the requirements for the degree stated.

Signature: ________________________________

Main Supervisor: Assoc. Prof. Dr. Abas Md. Said

Signature: ________________________________

Co-Supervisor: ________________________________

Signature: ________________________________

Head of Department: Dr. Jafreezal B. Jaafar

Date: ________________________________

VOIP WITH ADAPTIVE RATE IN MULTI- TRANSMISSION RATE WIRELESS LANS

by

HANIYEH KAZEMITABAR

A Thesis

Submitted to the Postgraduate Studies Programme

as a Requirement for the Degree of

MASTER OF SCIENCE

COMPUTER & INFORMATION SCINCES

UNIVERSITI TEKNOLOGI PETRONAS

BANDAR SERI ISKANDAR,

PERAK

FEBRUARY 2013

DECLARATION OF THESIS

Title of thesis: VOIP WITH ADAPTIVE RATE IN MULTI- TRANSMISSION RATE WIRELESS LANS

I, HANIYEH KAZEMITABAR

hereby declare that the thesis is based on my original work except for quotations and citations, which have been duly acknowledged. I also declare that it has not been previously or concurrently submitted for any other degree at UTP or other institutions.

Witnessed by

| | |
|---|---|
| Signature of Author | Signature of Supervisor |
| Permanent Address: | Name of Supervisor: |
| No. 8, Sarsabz Deadend, Sazmane Aab Street,Sheikh FazlolAllah Highway, 1454643381, Tehran, IRAN | Assoc. Prof. Dr. Abas Md. Said |
| Date: | Date: |

# DEDICATION

To my beloved husband and my supportive family members.

ACKNOWLEDGEMENTS

"In the name of God, the most Beneficent, the most Merciful"

First and foremost I wish to express my sincere appreciation to my advisor and supervisor Associate Professor Dr. Abas Md. Said who has helped me technically and spiritually. I would also like to thank my beloved husband, my dear family and my friends whom without their encouragement and moral support this thesis work would not be possible. Not to forget my colleagues and all fellow postgraduate students, I thank you sincerely. Further I am grateful to Universiti Teknologi PETRONAS (UTP) for providing the education facilities and the financial support.

# ABSTRACT

“Voice over Internet Protocol (VoIP)” is a popular communication technology that plays a vital role in term of cost reduction and flexibility. However, like any emerging technology, there are still some issues with VoIP, namely providing good Quality of Service (QoS), capacity consideration and providing security. This study focuses on the QoS issue of VoIP, specifically in “Wireless Local Area Networks (WLAN)”.

IEEE 802.11 is the most popular standard of wireless LANs and it offers different transmission rates for wireless channels. Different transmission rates are associated with varying available bandwidth that shall influence the transmission of VoIP traffic.

The purpose of this study is to introduce a new algorithm that adjusts the speech-coding rate of VoIP according to the transmission rate of the wireless channels to prevent channel congestion and calls drop. In this study, different codecs and different packet sizes were examined to identify the best adaptation process. Several simulation scenarios were studied to determine the best adaptation instant. The Real-time Transport Control Protocol - Extended Report (RTCP-XR) was used to monitor the link’s status to identify the speech quality degradation.

Applying the adaptive algorithm for VoIP in the wireless nodes shows a huge improvement in quality of speech and effective bandwidth utilization in comparison to the VoIP with no adaptation algorithm. In addition, a comparative study between the proposed algorithm and current related algorithms shows the proposed algorithm is more accurate and more effective in terms of execution time, adaptation cost, and adaptation process for the transmission of voice over WLANs.

ABSTRAK

Suara melalui Protokol Internet (VoIP) merupakan satu kaedah komunikasi popular yang memainkan peranan penting dalam mengurangkan kos dan fleksibiliti. Walau bagaimanapun, teknologi VOIP ini masih mempunyai beberapa isu yang harus ditangani dari segi penyediaaan Kualiti Perkhidmatan (QoS) yang baik, pertimbangan kapasiti dan penyediaaan keselamatan. Kajian ini memberi tumpuan tentang isu QoS VoIP, khusus untuk Rangkaian Kawasan Tempatan tanapaWayer (WLAN).

IEEE 802.11 adalah piawaian yang paling popular untuk LAN tanapaWayer dan digunakan dalam kebanyakan infrastruktur sistem VoIP. Ia menawarkan kadar penghantaran yang berbeza yang dikaitkan dengan pelbagai jalur lebar yang dapat mempengaruhi penghantaran trafik VoIP.

Tujuan kajian ini adalah untuk memperkenalkan satu modul algoritma baru untuk melaraskan kadar pertuturan pengekodan mengikut kadar penghantaran saluran wayarles untuk mengelakkan kesesakan dan penurunan panggilan.

Walaupun sudah terdapat beberapa kajian dalam bidang ini, kaedah penyesuaian, kelajuan pelaksanaan dan masa penyesuaian masih perlu diperbaiki. Oleh itu, dalam kajian ini algoritma dan saiz paket yang berbeza telah dikaji untuk mengenal pasti proses adaptasi yang terbaik. Tambahan pula, beberapa proses simulasi telah dijalankan untuk menentukan penyesuaian masa yang terbaik. Khusus, algoritma yang dicadangkan adalah faktor-faktor yang telah dikenal pasti dan tambahan, ia menggunakan paket masa nyata Protokol Kawalan Pengangkutan - Laporan Terluas (RTCP-XR) untuk memantau pautan dan mengenal pasti degradasi kualiti ucapan.

Hasil daripada penggunaan algoritma ini, kawalan kadar penyesuaian pada nod wayarles menunjukkan peningkatan besar dalam kualiti pertuturan dan penggunaan lebar jolur yang berkesan berbanding dengan sistem bukan-penyesuaian.

Di samping itu, perbandingan telah dilakukan diantara algoritma yang sedia ada dan algoritma yang dicadangkan, dan hasil kajian menunjukkan bahawa algoritma baru ini adalah lebih tepat dan berkesan dari segi masa pelaksanaan, kos penyesuaian dan proses penyesuaian untuk penghantaran suara melalui WLAN.

TABLE OF CONTENT

LIST OF TABLES

LIST OF FIGURES

## LIST OF ABBREVIATION

| | |
|---|---|
| AIMD | Additive Increase Multiplicative Decrease |
| AMR | Adaptive Multi-Rate codec |
| AP | Access Point |
| BER | Bit Error Rate |
| BSS | Basic Service Set |
| CAC | Call Admission Control |
| CODEC | Coder-Decoder |
| ESS | Extended Service Set |
| FEC | Forward Error Correction |
| FPP | Frame Per Packet |
| GSM | Global System for Mobile Communications |
| IEEE | Institute of Electrical and Electronics Engineers |
| IntServ | Integrated Services |
| ISP | Internet Service Provider |
| LA | Link Adaptation |
| MAC | Media Access Control |
| MGW | Media Gateway |
| MOS | Mean Opinion Score |
| MOS-CQ | MOS-Conversational Quality |
| MOS-LQ | MOS-Listening Quality |
| PER | Packet Error Rate |
| PESQ | Perceptual Evaluation of Speech Quality |

| | |
|---|---|
| PHY | Physical layer of the OSI model |
| PLR | Packet Loss Ratio |
| PSQM | Perceptual Speech Quality Measure |
| PSTN | Public Switched Telephone Networks |
| QoE | Quality of Experience |
| QoS | Quality of Service |
| RFC | Requests for Comment |
| RR | Receiver Report |
| RTCP | RTP Control Protocol |
| RTCP-XR | RTP Control Protocol-Extended Report |
| RTP | Real-time Transport Protocol |
| RTT | Round Trip Time |
| SIP | Session Initiation Protocol |
| SNR | Signal to Noise Ratio |
| SR | Sender Report |
| TCP | Transmission Control Protocol |
| TFRC | TCP-Friendly Rate Control |
| UDP | User Datagram Protocol |
| UMTS | Universal Mobile Telecommunication System |
| VoIP | Voice over Internet Protocol |
| WLAN | Wireless Local Area Network |
| WSTA | Wireless Station |

# CHAPTER 1

# INTRODUCTION

## 1.1 VoIP Overview

We have entered into a new communication era and people look for a cheap communication method, easily accessible everywhere, with no time limits and capable of integrating services.

Unlike the traditional phone network that also known as Public Switched Telephone Network (PSTN) in which *analog voice* phone calls are transported through circuit-switched based networks [1], Voice over Internet Protocol (VoIP) refers to the process of transporting *digital voice* over packet-switched networks such as Internet. It has been verified that VoIP is a main competitor to the PSTN in terms of cost reduction, integrating of service, versatility and efficiency [2]. Therefore, VoIP has achieved wide acceptance.

There are two main VoIP tools: First is PC-based application such as VoIP software or “Soft Phones” and the second is “IP Phones” that look just like normal phones but instead of having the standard phone connectors, they have an Ethernet connector in order to connect to a packet-switched network.

It should be mentioned that “IP Telephony” term which sometimes is used instead of VoIP which refers to the transmission of voice, video and other real-time media over IP networks and it is not restricted to “voice” [3].

## 1.2 VoIP Protocols

VoIP protocols simply allow two or more devices to send and receive the real-time speech traffic. Generally, VoIP architectures are partitioned into two main components: *signaling* and *media transfer*.

*Signaling* covers the concepts such as endpoint naming, addressing and parameter negotiation. Depending on the architecture, Quality of Service (QoS) also can be a component of the signaling protocol (such as IntServ) [4].

*The media transfer* typically includes a simpler protocol for encapsulating data. Media transfer usually supports multiple codecs and security [4]. A popular media transfer protocol for the transport of real-time streams is RTP [5]. The services provided by RTP include time reconstruction, loss detection, security and content identification. RTP protocol comprises RTCP, which can control some RTP parameters between the communication endpoints [6].

Furthermore, VoIP uses UDP/IP instead of TCP/IP. Due to the acknowledgment and re-transmission features of TCP [7], this protocol is not suitable for real-time applications as it leads to excessive delay in communication [8]. Conversely, UDP does not have these features, so it has less delay of communication but on the other hand, it provides unreliable connectionless delivery service. However, to rectify this imperfection, RTP in conjunction with UDP, provides end-to-end network transport functions for transmitting real-time application. RTP does not have a resource reservation, thereby it does not guarantee quality of service. However, its companion protocol RTCP allows monitoring of the link [6]. Figure 1.1 illustrates the protocol stacks for IEEE 802.11 wireless networks (gathered from [9], [10], [11] and [12]).

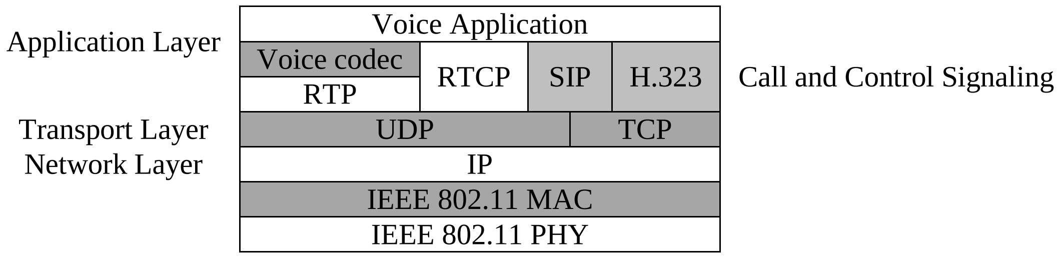



Figure 1.1: VoIP over IEEE 802.11 protocol architecture.

### 1.3 Speech Quality Measurements

Figure 1.2 demonstrates the quality measurement methods for speech that can be categorized into subjective and objective methods [11].

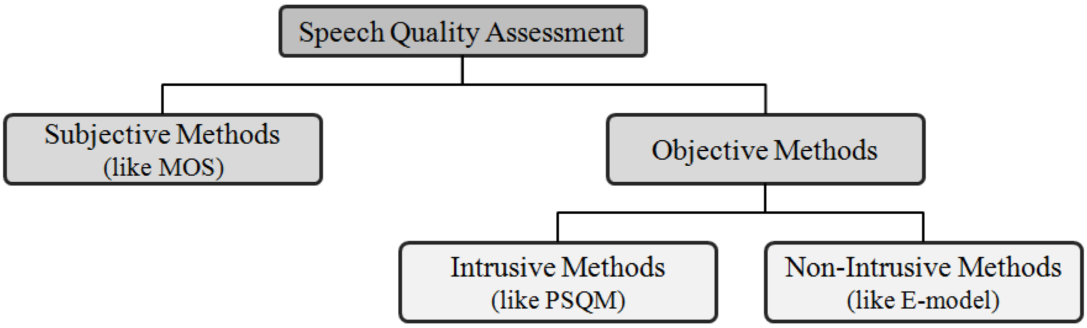



Figure 1.2: VoIP quality measurement methods.

Subjective methods are achieved by the average user perception by asking a group of listeners and provide them a limited response choice. ITU-T in Recommendation P.800 presented Mean Opinion Score (MOS) [13] which is based on user perception and it has a range from 1 (poor) to 5 (excellent) to demonstrate speech quality subjectively. However, subjective MOS is time consuming, costly, un-reliable and cannot be used for large scale or long-term voice quality monitoring [14]. This has led to the use of objective methods in voice quality measurement.

Objective methods are based either on comparison between the real speech signal and a reference signal or they are based on mathematical models that can measure physical quantities of the network parameter such as delay and packet loss [15]. Objective methods comprise two techniques for quality measurement:

Intrusive methods that generally inject a test signal into the network to compare the original signal and output signal and normally these methods are applied at the time of development of VoIP systems [16], while non- intrusive methods execute on real-time traffic without any reference signal and perform prediction directly from network impairment parameters such as jitter, delay and packet loss [17].

Technically, MOS value can also be achieved by objective methods like PESQ method or E-model [18]. E-model is a calculative method which has brought all impairment factors that have an effect on a voice quality into a single factor (*R-factor*) as an output which is convertible to MOS scale [19]. The values that are needed to calculate *R* can be received by RTCP packet information.

Furthermore, E-Model is able to track changes in quality and estimate instantaneous quality [20]. The scale of R is typically from 50 to 100, where anything below 50 is clearly unacceptable and 93.2 is a default value of R with all parameter values set to default [18]. MOS and E-model will be discussed in section 3.2.3 with more details.

## 1.4 VoIP Components

In order to transmit analog voice over packet-based network the voice must be sampled, quantized, encoded, compressed (not compulsory), and then encapsulated in a VoIP packet [21]. These steps are performed on the source side. The packets are then transported through the IP network to destination to retrieve the analog voice from the packets.

The principal components of a VoIP system which covers the end-to-end transmission of voice, are CODEC (Coder-Decoder), Packetizer and the playout buffer. According to ITU-T G.1020 Recommendation [10], Figure 1.3 demonstrated the process of VoIP transmissions.

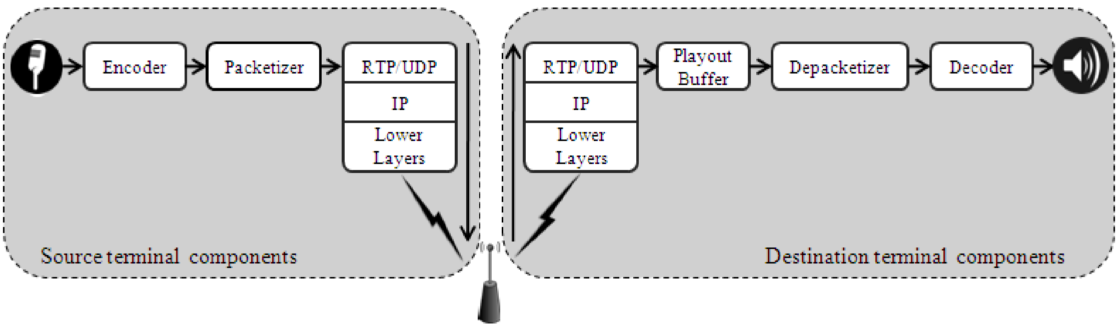



Figure 1.3: Components and transmission process of VoIP over WLAN.

### 1.4.1 CODEC (Coder-Decoder)

Voice codec or coder-decoder is the algorithm that is executed by sender and receiver to enable digital lines to carry the 'analog voice' by converting it to the digital format. In addition, it may provide compression methods to save the network bandwidth.

Different codecs have different bit-rates[1], packet length, speech quality, algorithmic delay, complexity and robustness against background noise. The most common voice codecs are G.711, G.723, G.726, G.728, and G.729. There are some studies that have analyzed the performance of different codecs on QoS [6], [22], [23], [12], [24]. Speech codecs are divided into three categories; waveform codecs, source codecs and hybrid codecs.

**Waveform codecs** produce high bit-rates and result in very good speech quality at the receiver. Since they take low processing effort, they have a lower delay that is the main concern of real-time applications. From the other side, they need large bandwidth. ITU-T G.711 codec [25] is from the waveform codec category.

**Source codecs** use the mathematical model generally with linear multi-parameter filter to generate the speech synthetically from human voice. In addition, they are able to predict human speech samples based on previous samples. When these two features come together, the overall bandwidth required for speech can be greatly saved [26]. Thus, the codecs of this category need very low bit-rates that results in low bandwidth consumption. However, because of their complicated algorithms they need much more processing which leads to the additional delay.

**Hybrid codecs** are the combination of the two previous categories that give fine speech quality with average output rates and fair bandwidth consumption. ITU-T G.729 codec [27] is in this category.

[1] Bit-rate is the number of bits per unit of time required to get samples of analog speech to encode it to digital format. Bit-rate

### 1.4.2 VoIP Packetization

The next component is the packetizer. Packetizer divides encoded voice into packets. Each packet requires a header, and each header adds 40 bytes to the packet [21]. So, the packetization interval directly affects bandwidth consumption and speech quality. Small packets associated with more redundant data that leads to high bandwidth consumption for header transmission so they would not use bandwidth efficiently. Besides, large packets require a greater period for samples to be gathered and filled in the payload of a single packet so it can lead to more delay. Later, in methodology chapter this will be discussed in more detail.

### 1.4.3 Playout Buffer

The playout buffer (jitter buffer), is another component of VoIP which is on the receiving side. It is used to rearrange packets according to the schedule of their playout time to mitigate the delay-jitter.

## 1.5 Overview of WLANs (IEEE 802.11 Architecture)

Wireless technology uses radio frequency or Infrared instead of wire or cable. Due to characteristics such as mobility, simplicity, scalability, edibility and cost effectiveness, Wireless LANs (WLANs) are replaced with wired LANs in home, office and other public places. Consequently, Voice is also run over WLANs (VoWLANs).

Simply, a wireless network can be configured in Ad-Hoc mode and infrastructure mode; earlier is peer to peer (P2P) network and each node can directly communicate with all other nodes while later the transmission between nodes pass through central point called an Access Point (AP) [28]. AP can be presented as a gateway to connect wireless and wired portion of the network.

In the infrastructure mode the term Basic Service Set (BSS) is used for a wireless network containing only a single wireless access point. While the vast majority of small offices and home networks fall into the BSS category, in larger offices, a single access point may not provide coverage for all wireless stations so it is better to use multiple access points. Such a network is called an Extended Service Set (ESS). Figure 1.4 shows infrastructure of both BSS and ESS mode.

The Institute of IEEE is responsible for setting the standards for LANs and 802.11 workgroup inside IEEE is tasked to develop standards for wireless LANs. There are several 802.11 protocols operate in different frequency bands such as: 802.11a, 802.11b, 802.11g and 802.11n. Characters such as “a”, “b”, “g” or “n” have been given beside 802.11 to categorize this standard  to even more specific tasks [28].

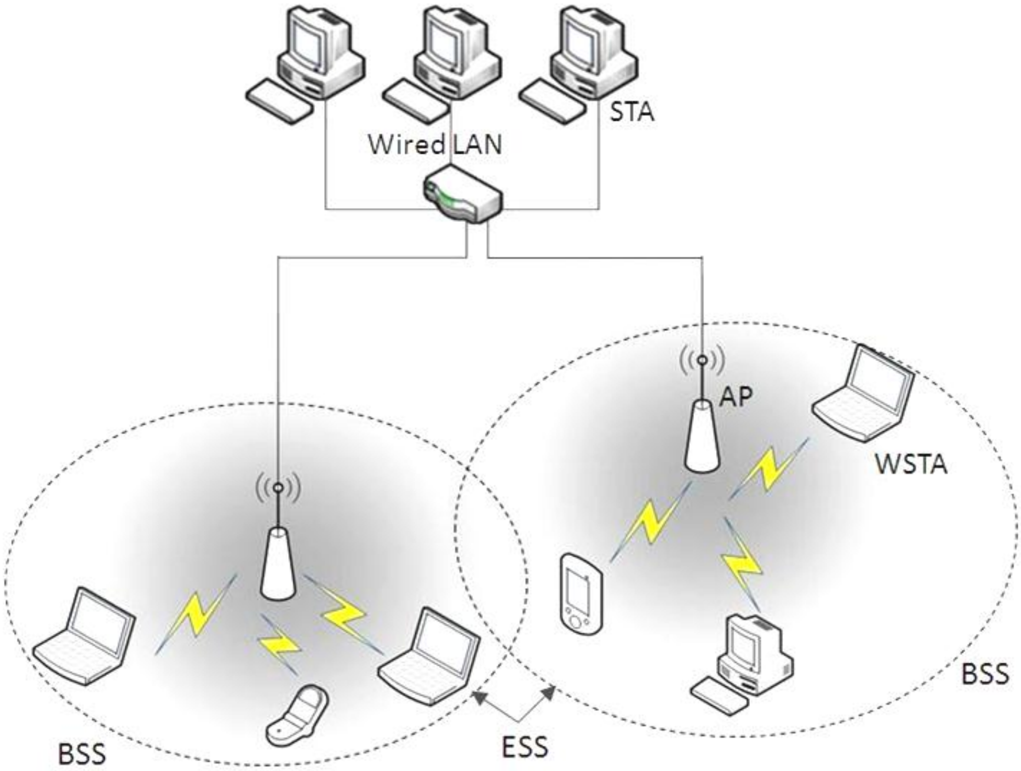



Figure 1.4: WLAN BSS and ESS infrastructure.

## 1.6 Motivation

Although, VoIP has been studied for around 20 years it still suffers from some problems in term of QoS requirements. Moreover, due to the fast expansion of WLANs, VoIP QoS over these networks are significant issue to be considered. Quality issues of VoIP are due to the mismatch between VoIP flow and the network condition. The QoS issues in Wireless LANs are more critical, because signals between AP and wireless stations are affected by distance, atmospheric conditions and presence of physical obstacles. Therefore, channel conditions change dynamically during communication that offers different transmission rate.

The capacity of the wireless channel in terms of the number of supported calls is variable. These fluctuations can affect the quality of speech in VoWLANs. Consequently, acceptable quality requirements of VoIP such as delay, jitter and packet loss, cannot be guaranteed over Wireless links.

Some requirements have to be met in order to provide satisfactory quality for VoIP. The voice stream must not suffer from a delay higher than 150 ms [29] and packet loss should be less than 3% [15]. In addition, voice codecs should be chosen carefully, since every codec has different features, they can affect the perceived speech quality. So finding the best VoIP coding parameter based on network condition is critical.

From another perspective, IP a is a satisfactory protocol for data traffic but it is not designed for real-time traffic like VoIP, because it is based on the 'best effort' principle and packets are not guaranteed to be delivered to the receiving side. Therefore, some methods should be applied to guarantee the delivery of packets to the other end. This is the motivation for seeking an algorithm to guarantee the delivery of voice over IP networks. Consequently, this research is to investigate these two controversial topics and introduce some mechanisms to improve the quality of voice over IP networks especially for WLANs.

## 1.7 Research Problem

IEEE 802.11 standard for wireless LAN presents multi-rate transmission feature. That is PHYICAL layer has *multiple data transfer rate* capability. According to this standard, the role of Link Adaptation (LA) function is to select an appropriate transmission rate from a set of possible rates based on wireless link conditions. In IEEE 802.11b the set of possible transmission rates are: 1, 2, 5.5, and 11 Mbps as shown in Figure 1.5 (Figure demonstrates based on [30], [31], [32]).

Technically, different transmission rates cause different network capacities and different capacities lead to the unsteady network for VoIP calls. For example, when the wireless station moves far from the Access Point (AP) due to signal fading, PHY layer in the transmitter adapts its transmission rate to the lower rate. In such a situation, available bandwidth is deceased and it may lead to shortage of required capacity to carrying the VoIP calls which may cause congestion. Congestion is not an ignorable issue, because it will lead to delay and in the more severe situation it will lead to packet loss.

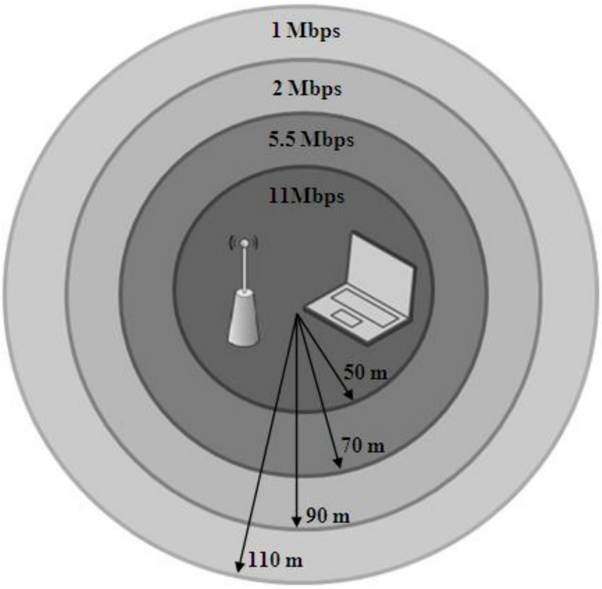



Figure 1.5: Transmission rates in IEEE 802.11b based on the average distance from AP.

As mentioned earlier, the VoIP output rate is related to codec type and its frame size. So, by changing these two parameters, the output rate of VoIP can be changed too. Table 1.1 tabulates the characteristics of two most famous codecs. G.711 generates a digitized voice with a rate of 64 Kbps, on the other hand G.729 generates digitized voice with the rate of 8Kbps which is one-eighth of G.711 codec output rate.

Table 1.1: Characteristic of two well-known codecs.

| Codec | Bit-Rate (Kbps) | MOS | Quality |
|---|---|---|---|
| G.711 | 64 | 4.1 | Excellent |
| G.729 | 8 | 3.9 | Good |

In this matter, lower bit-rate codec like G.729 uses a higher compression rate, leading to lower utilization of the bandwidth, so, they support more calls in such a fixed amount of bandwidth, unlike higher bit-rate codec like G.711. Figure 1.6 demonstrates an example of codec consumption in an assumed 100 Kbps bandwidth.

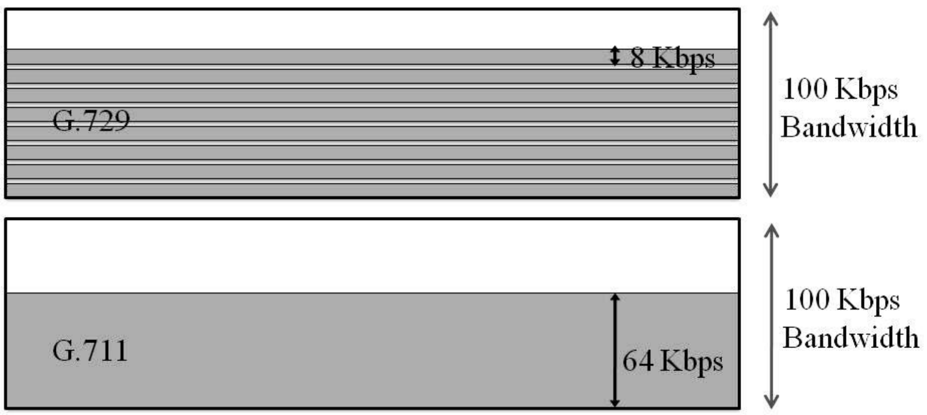



Figure 1.6: The example of codec bandwidth consumption.

The problem arises when these fix properties of codec come in multi-rate wireless LANs which channels change frequently. If the VoIP calls use constant bit-rate codecs they provide fix output rates independent of network condition, VoIP flow cannot be adapted based on channel conditions. Therefore, in the case of lower transmission rate, if VoIP codec continues encoding/decoding of speech with a higher rate it causes congestion. However in same situations lower bit-rate codec due to having lower bandwidth consumption can relieve the congestion.

Beside codec, choosing different payload sizes also affects transmission efficiency. Small packets utilize channel bandwidth inefficiently because of a higher ratio of protocol overhead to the real data. So, bigger packet size may effect on lower bandwidth consumption.

Therefore, an alternative to meet VoIP requirement for multi-rate WLANs is to use the “adaptive-rate control” algorithm to reduce the effect of mismatch between VoIP traffic and network capacity. Adaptive-rate control algorithm for VoIP can adjust the VoIP transmission according to network conditions to optimize VoIP performance.

In fact, the idea of enhancing real-time applications with an adaptive-rate control was famed from early 2000 by [33]. Adaptive algorithms may work with voice codec switching and/or packet size switching at the sender side and/or work with adaptive playout buffer length of the receiver side [33], [34].

In spite of several previous researches on adaptive rate control algorithm [30, 34, 42-49], there are still some deficiencies in the quality monitoring, adaptation benchmarks, a proper adaptation process, execution time and some other factors. Consequently, a more efficient algorithm is required to address the limitation and deficiency of previous algorithms, which will be discussed in chapter 2.

## 1.8 Research Objectives

The ultimate goal of this research is to propose a new adaptive-rate control algorithm for VoIP systems that control the coding parameters based on WLAN link conditions and keep the QoS metrics in the acceptable range. This algorithm enables the network to monitor the link quality effectively and measure the quality in real-time. It will evaluate different adaptation process with the combination of codec and packet size. Moreover, it will rectify some imperfections of previous algorithms. The explicit objectives of this research are as follows:

1. To use an enhanced statistical index to estimate network state for finding congestion.

2. To investigate the right adaptation time based on transmission rate variation and the communication link quality monitoring.

3. To reduce the algorithm execution time by using most recent standards in the adaptation process.

4. To find the most appropriate adaptation process by focusing on voice coding parameter.

5. To evaluate and validate the performance of the proposed algorithm in contrast to existing schemes in the literature.

## 1.9 Research Scope

Since 802.11b is the most popular wireless LAN (WLAN) standard in research level, in this study the evaluation was performed over 802.11b networks in the BSS infrastructure ( see section 1.5).

- G.711 and G.729 codecs were studied in this research due to their popularity.
- This research only addresses the multi-rate problem of WLANs that are posed to VoIP and no other traffic like video or data will be considered.
- This research concerns only the situation of current calls, therefore there is no consideration of Call Admission Control (CAC).
- Other scope of concerned is the congestion problem caused by transmission rate reduction (i.e. no concern on transmission rate increment).
- OPNET 11.5 modeler is used for simulation models.

## 1.10 Research Methodology and Activities

As stated before, the function of LA is to choose different transmission rates for wireless channel based on the channel error rate that leads to different capacity for VoIP traffic and affects the quality of speech. Therefore, capacity estimation is one of the required steps to be developed for the main simulation model.

In this research, the adaptive-rate control algorithm is to adjust the VoIP transmission according to network conditions. Therefore, in order to determine the network conditions some experimental scenarios will be conducted to determine the best adaptation index among various quality factors.

As mentioned earlier, when the transmission rate of a wireless channel is reduced, switching to the lower bit-rate codec and/or bigger packet size can reduce the load of VoIP traffic to prevent the congestion. Therefore, some simulations are needed to show which adaptation process is more effective to mitigate the congestion problem with consideration of codec and packet size adaptation sequence. The research objectives are achievable by considering the following research methodology and activities:

- Conducting an extensive literature survey in detail on adaptive rate control algorithms for general IP networks and wireless LANs in detail.
- Determining the essential components of adaptive-rate control algorithm.
- Developing a model for the OPNET modeler to verify the algorithm.
- Finding the capacity for each transmission rate of 802.11b based on the different codecs and different packet sizes.
- Finding the effect of transmission rate changes on the fix number of calls and monitor QoS parameter like delay, packet loss, and MOS.
- Examining different codec and packet size adaptation method for different situations.
- Designing and development of a new adaptive rate control algorithm.
- Conducting a comparative performance analysis of algorithm with other well-known algorithms.

## 1.11 Contribution

In the proposed adaptive-rate control algorithm, we considered the combination of different constant bit-rate codec and different packet size. While many previous studies, only considered one of these parameters.

Furthermore, unlike previous works, adaptation is not necessary after each transmission rate reduction and most of the time when the transmission rate falls down to the lower rate, the system can sustain the call with the current coding parameters. So, in this study some quality factors like delay and jitter are included to find the right adaptation time. Despite previous works, this algorithm tries to find the congestion based on delay variation/jitter beside MAC monitoring thus it has less burden of adaptation. In addition, delay variation/jitter enables the algorithm to predict the congestion faster and as a result, react faster.

However, since voice is real-time and the algorithm should be executed fast, our algorithm does not include complicated or time-consuming calculation contrasting the algorithms in [35], [20] and it used RTCP-XR packet as a new feedback technique which provides a VoIP metrics block to evaluate the network situation. Moreover, the most appropriate method of codec and packet size adaptation has been proposed for different network conditions.

The contributions of this thesis also include performance evaluation of adaptive rate algorithms. It is expected that our adaptive VoIP rate algorithm works more effectively because it possesses the desirable properties, which include fast response, aggressiveness, and fair bandwidth allocation.

## 1.12 Structure of Thesis

This thesis tackles the problem of multi- transmission rate that affect VoIP traffic. This work introduces an adaptive rate VoIP to optimize the performance of speech packet transmission under any given network condition, and deliver a better performance than un-adaptive VoIP. The thesis has been structured as follows,

**Chapter 2** reviews related studies in detail and discusses the issue of the current algorithms.

**Chapter 3** describes the research method and the proposed algorithm components. The focus is on the technical properties of the algorithm, including finding the right adaptation time, quality measurement factors, codec and packet size adaptation processes, real implementation requirements, and capacity estimating of WLAN channel in different transmission rate.

**Chapter 4** contains a simulation model to evaluate the performance of the adaptive approach based on the main quality factors and comparison of the proposed adaptive rate control algorithm with non-adaptive algorithm and with the latest related work to validate the new proposed algorithm.

**Chapter 5** provides research conclusion and summarizing of the main findings of this research. It also describes the achieved objectives, benefits of using this algorithm as well as its limitations. Future work, proposed as last part followed by a list of references.

## 1.13 Chapter Summary

This chapter presented the background and motivation of the thesis. Furthermore, research problems, technical objectives and methodologies are discussed. This chapter comprises the scope of research, research contributions and organization of this thesis. The main purpose of this project is to propose an algorithm that can be used in VoIP software or VoIP gateway to obtain good voice quality by making a proper decision for adapting the voice codec parameters based on network conditions.

# CHAPTER 2

# BACKGROUND AND RELATED RESEARCHES

## 2.1 Chapter Overview

VoIP is one of the most important technologies in the communication world. Although after 20 years of research on VoIP, some of VoIP problems are still remaining. Especially, during the past decade with the wireless technologies fast growth, many researches have motivated to divert their focus from Wired-LAN to Wireless-LAN. VoIP over WLAN faces many challenges due to the vulnerable nature of wireless network. Moreover, issues like providing QoS at a good level, dedicating channel capacity for calls and having secure calls are more difficult than wired LAN. Therefore, VoIP over WLAN (VoWLAN) remains a challenging research area. Capacity and quality of VoIP are two important issues that need to be resolved before commercial deployment of VoIP and both issues are found to be dependent to the voice codecs [36].

This research has focused on the problem of multi-rate WLANs that poses a challenge to VoIP traffic due to providing different channel capacity. This chapter discusses the background of related researches that have proposed algorithm for VoIP adaptation with respect to network channel conditions. The adaptation methods and adaptable parameters considered in previous researches will be reviewed in detail. This chapter will categorize related algorithms according to the adaptation parameters, adaptation policies and methods, and the way of dealing with the multi-rate effect and congested channel.

A limited number of researches in the area of adaptive VoIP traffic for multi-rate WLANs are available which all have demonstrated that their methods provide better communications quality than ordinary fixed-rate VoIP. There are different techniques to handle the problem of multi-rate WLANs for speech traffic. On one hand, there are the rate adaptation techniques on the sender that can adjust the VoIP output rate based on the link conditions (Figure 2.1 and Table 2.1). In this group, coding adaptation can adjust codec bit-rate (compression rate) and packetization interval. On the other hand, adjusting the playout buffer length (jitter buffer size) based on network conditions can be implemented on the receiver's side to control the input rate of VoIP.

Some studies have been focused on the jitter buffer adaptation technique [37], [38], [39] and [40] while this study will focus on the first technique which adjusts the VoIP output rate based on the network conditions as it is more flexible and extendable technique.

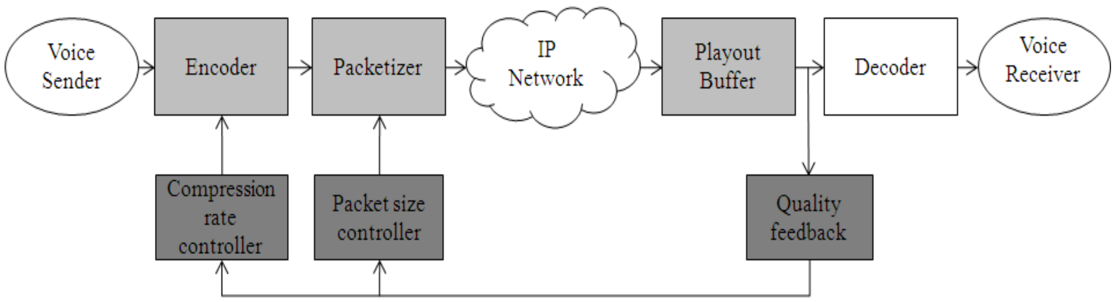



Figure 2.1: Rate-Adaptation mechanism.

Table 2.1: Different adaptation techniques.

| Adaptation Techniques | | |
|---|---|---|
| Sender Side (Controlling VoIP output-rate) | | Receiver Side (Controlling VoIP input-rate) |
| Coding bit-rate | Packetization Interval | Jitter (Playout) Buffer Size |

As shown in Figure 2.1, all adaptation methods need some statistical and qualitative-based feedback from the link for estimation of the network condition especially to determine congestion or quality degradation. Some of these methods will be discussed in the following.

The idea of end-to-end feedback control can be found in [41] and [42] in which their analytical model was to find congestion based on *delay feedback*. Round Trip Time *(RTT)* that declare delay to some extent, was used in [43] along with *packet loss rate*. The adaptive algorithm proposed in [33] is based on *delay and packet loss* measurement which are gained by RTCP packets.

Quality feedback method in [44] and [45] is the packet loss rate. The algorithms in [33], [46] and [47] used packet loss partially beside other indicator. Sfairopoulou et al. at the early stage of their work in [48] and Ngamwonwattana in his thesis [49] performed the adaptation based on moving average thresholds of delay and packet loss. Kawata and Yamada have made the decision of adaptation based on the *presence and absence of ACK* which can identify Frame Error Rate (FER) [50]. Other factors can also be considered as a congestion indicator like *phase-jitter* [51] and *media access delay* [52]. *E-model* (a method for estimating the expected voice quality) is used in [53], [54], [26] and [35]. Table 2.2 tabulated some different quality feedbacks used in previous works.

Table 2.2: Different quality indices used by other researchers.

| Finding Adaptation Time | | | | | | | |
|---|---|---|---|---|---|---|---|
| Delay | RTT | Media Access Delay | PLR | ACK | E-Model | FER | Phase-jitter |

The output rate of VoIP can be adjusted by two main categories namely, *adjusting the coding rate* and *adjusting the packet size*. Generally, the procedure of codec adaptation methods is based on this policy the if the adaptation algorithm finds the network congested, codec is switched to the lower bitrate to decrease the bandwidth consumption, and in the opposite situation higher bitrate codecs are chosen. Codec-based adaptation can be further categorized into two groups. The first group includes a set of fix bitrate codecs and the input rate is adapted by exchanging among different codecs of this set. In the second group, adaptive multi-rate codec like AMR [55] or Speex [56] which comprised the built-in variable bitrate are employed.

The study in [57] showed that when the network is highly loaded or congested, in most of the cases, G.711 has the highest PLR followed by G.729 and then G.723. Therefore, the order of codec switching is very important.

Table 2.3 shows different codec with their coding bitrate and their perceived quality for speech. Table 2.4 shows different bitrate and the quality of each coding bitrate of an AMR codec.

Table 2.3: Bit-rate and speech output quality for some codecs [26].

| Codec | Bit-rate (Kbps) | Speech Quality (MOS) |
|---|---|---|
| G.711 | 64 | 4.1 |
| G.726 | 32 | 3.58 |
| G.729 | 8 | 3.7 |
| G.723.1 | 5.3 | 3.6 |

Table 2.4: AMR codec modes, bit-rates and quality of speech [58].

| AMR mode | Bit-Rate (Kbps) | Speech Quality (MOS) |
|---|---|---|
| 0 | 4.75 | 2.6 |
| 1 | 5.15 | 2.7 |
| 2 | 5.9 | 3 |
| 3 | 6.7 | 3.1 |
| 4 | 7.4 | 3.2 |
| 5 | 7.95 | 3.2 |
| 6 | 10.2 | 3.5 |
| 7 | 12.2 | 3.6 |

AMR adjusts the speech rate based on signal to noise ratio (SNR) and bit error rate (BER). This scheme does not reduce the effect of delay impairment but the scheme of switching between several codecs can be established upon other benchmarks (rather than SNR and BER) to elevate the speech quality.

The significant surveys in [59] and [60] reviewed and compared different adaptation methods. Follow on to that, in this chapter, adaptive rate control algorithms are classified into four main categories based on their respective adaptation method.

There are those studies that proposed to change the input rate of VoIP based on the state in wireless link by changing packet size. However, other studies proposed to change the codec. There are also some other works that have taken both proposals into account and in the last category, some other factors have been taken into account. Figure 2.2 shows these divisions.

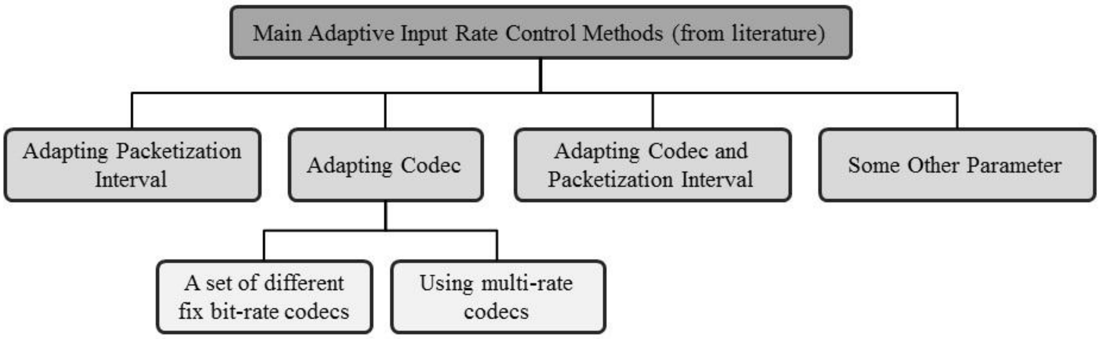



Figure 2.2: Categories of VoIP output rate adaptation approaches from literature.

Each of the categories in Figure 2.2 is explained and discussed with the related studies in the following sections.

## 2.2 Packetization-Based Approach

Packetization is an essential parameter that must be specified at the time of developing a VoIP system. It determines how many voice frames should be allocated into each packet [49]. Predetermine and fix packetization results in non-optimal performance. Regarding this fact that 802.11b network is not able to carry a large number of VoIP calls. Smaller frame size limits the maximum number of VoIP calls as well, so by use of a larger payload number of VoIP calls can be increased. However, delay and loss need to be considered [61].

This fact is also extendable to other networks like, Universal Mobile Telecommunication system (UMTS). Cao and Gregory evaluated the performance of VoIP in terms of end-to-end delay using different codec and different frame size for UMTS networks [12]. Their results demonstrated that when frame size increases, end-to-end delay increases too. In the other scenario, they assume a fix frame size and they have increased the number of frames per packet that showed with more number of frames in each packet end-to-end delay also increased.

Therefore, larger payloads have more transmission efficiency in term of bandwidth consumption but it requires a longer period to gather and to collect more audio into a single packet which will take longer time by packetizer and cause more end-to-end delay. On the other hand, small payload size is more tolerant to packet loss and has better voice quality at un-congested network condition.

Oouchi et al. in [62] investigated the effect of different packet length for VoIP network based on an experimental study in a real environment. Their study has evaluated appropriate packet length in various network conditions using two criteria, the first criterion is transmission efficiency and the second one is voice quality in term of jitter and packet loss rate. Their results have shown, longer voice payload size has lower bandwidth utilization due to less header redundancy but shorter voice payload size results in lower tolerance to packet loss. To achieve an algorithm that has good voice quality and high transmission efficiency they proposed to use variable packet length to adapt the packet length based amount of bandwidth and quality requirement.

Research in [63] has proved to achieve a better VoIP quality in congested channel in WLAN. They have shown packet size adaptation is enough to relieve the congestion while remaining on the same codec. Wang et al. [64] presented an algorithm based on the same concept that only packet size adaptation is enough. They have adjusted the number of frames per packet based on the values of delay and MOS as two quality indices.

The proposed method in [49] is based on constant bit-rate coder which uses different packetizations to tune the bandwidth. To evaluate the performance of the algorithm, they have measured one-way end-to-end delay and packet loss. They have shown that packetization approach can help to reduce delays and packet loss rate.

In one of the latest literature [65], the idea of adapting the payload size was proposed not only for congestion time but rather, before a user joins a VoIP session, the QoS standards would be negotiated to start the session with the optimum packet size for the codec. In their algorithm, network congestion is estimated using packet loss and delay jitter values gained from the RTCP Receiver Report (RR). If the delay increases due to network congestion or joining of a new user to the network, then payload size is decreased by reducing the number of frames per packet. Otherwise, the packet payload size is increased.

## 2.3 Codec Rate Based Approach

As mentioned before, fixed-rate voice coders generate a constant output bit-rate with no consideration of network condition. Therefore if the network is unable to afford enough bandwidth for traffic, it causes delay and packet loss. Generally, rate-adaptive approach has better results compared to non-adaptive (fixed-rate) approach. Since the adaptive VoIP application does not need a strict partitioning of the bandwidth, it is more efficient to utilize the network bandwidth in comparison with non-adaptive VoIP, also this flexible portioning prevents link congestion.

While UDP does not have any congestion control mechanism, the adaptive algorithm in [33] has proposed to use the famous congestion control scheme which is "additive increase-multiplicative decrease (AIMD)". For further information, TCP Friendly Rate Control (TFRC) mechanism for controlling congestion can be classified into three main categories: equation-based mechanisms, window-based mechanisms and additive increase, multiplicative decrease mechanisms. Window-based mechanisms use a congestion window to control transmission rate. The other one is equation-based mechanisms, also known as TFRC. In TFRC scheme sender uses equation-based scheme to regulate the sending rate of TCP connection based on Round Trip Time (RTT) and packet loss rate. According to [32] and [66] TFRC is a common rate control mechanism for wired networks. However, this algorithm is good for TCP flow congestion control.

In [33], AIMD scheme has been used as a congestion control mechanism and the source-based control algorithm allows some sources to adjust their bit-rate. They have focused on voice coders to find the most appropriate coder bit-rate based on network condition.

They quantify the performance of adaptive VoIP in terms of main QoS parameters namely packet loss rate and delay. Furthermore, they have compared their algorithm with few constant bit-rate codec. Their results showed an adaptive VoIP system acts better than non-adaptive VoIP system. However, their algorithm is not based on multi-rate characteristic of IEEE 802.11 standard but it considered dynamic traffic in the network.

An adaptive VoIP algorithm for heterogeneous wired/wireless environment has been studied by Trad et al. [43]. They have assumed weird segment is congestion free and they only considered last-hop from wired to wireless which is AP as bottleneck segment and run the algorithm on AP to transcode voice flows based on estimation of channel congestion.

In addition, they have proposed Vegas-Like Audio Rate Control algorithm in that coding adaptation take place based on link conditions that can be obtained from the RTCP receiver report statistic including packet loss rate and RTT between AP and wireless station. Thus, gateway keeps tracking of minimum RTT value. Every RTT in each instance should be compared with the default value to justify the network state.

Finally, they have compared their work with TFRC. Although TCP-Friendly rate control scheme can reduce congestion but as mentioned earlier, it is more appropriate for wired networks.

The first weakness of this algorithm is related to the thresholds of the adaptation process. The values for max and min thresholds are estimated based on the number of audio packets transmitted by the voice flow during a round trip time. Therefore, in real voice flow, these max and min threshold are varying all the time and they cannot be assumed fix. Although they ran the algorithm for different max and min threshold but these values have been assumed fix in the algorithm. In addition, despite the fact that RTT and loss rate are adequate indicators of the speech quality level, but later studies have used some other measurements as well to make adaptation more precise. Furthermore, they considered AP as a bottleneck for the traffic congestion since it connects the wired part of network to the wireless part. However, any wireless node can cause congestion. Moreover, their adaptation mechanism changes the coding rate of all the incoming calls, while in many cases, adaptation is not necessary for all the calls and only some of the calls need to adapt their codec to rectify the congestion.

Two algorithms that already discussed in this section focused on the adaptive VoIP mechanism. The first one has studied all IP networks and the second one has studied the connection of wired/wireless networks. However, adaptive QoS mechanisms with a special focus on multi-rate WLAN environment was investigated by Sfairopoulou et al. in [48]. They categorized their algorithm into three phases: monitoring, adaptation, and recovery phase.

In the first phase, they used RTCP sander and receiver report to collect the information from the link to monitor the link quality. For this purpose, they have defined a threshold for QoS parameter such as delay, packet loss and jitter by using Moving Average Function. So, if the values gained by RTCP packet is out of the threshold or if MAC layer information declares a rate change, then the next phase (adaptation) is commenced. In adaptation phase to have a more accurate estimation of link they have used RTCP packet in the shorter period (less than 5 seconds). Then, new calling parameters with the next SIP re-invite message in the current session will be set. After that in the recovery phase, algorithm monitors the network to check whether the rate change is effective or not.

In their simulation, there are two types of flow in the calls; either fast (5Mbps) or slow (1Mbps). They have changed some VoIP flows from fast to slow and they have shown overall throughput is reduced. Then they have changed the codec of reduced transmission rate flows *manually* to the lower bitrate codec. However, they did not use QoS parameter’s threshold which they calculated by moving average function also the information from RTCP packet.

Later, in the studies presented in [26], [67] and [68] they have enhanced the work in [48] to improve the monitoring phase by monitoring MAC and RTCP together. As a result, when a rate change is perceived in the MAC layer, *the transmission rate change alarm* triggered immediately and then the *RTCP packets* announce the QoS situation. So, based on the MAC and RTCP information, the algorithm decides whether to change codec or not.

In this solution, only some of the nodes need to adapt their codecs. First, the calls that have faced with the lower transmission rate adapt their codec to the lower bit-rate codec. Then, if congestion still remains high in the network, other calls also adapt their codec to reduce the load of the network.

Their algorithm is fast because there is no complex calculation and monitoring phase is perfect in term of determining transmission rate alteration and quality estimation. However, their algorithm would have been improved by implementing different frame size based on network condition as well as codec adaptation.

They have also evaluated two different modes of applying the location of adaptation algorithm: centralized and distributed. Centralized is implemented on AP and AP is responsible in charge of monitoring all calls while distributed is implemented on each node and each node is in charge of monitoring and adapting its own codec. They have found that in the centralized mode, AP as a central device has better estimation of voice quality. However, centralized algorithms require employing specific network equipment such as non-standard APs [51].

Another work that has addressed the effect of frequent dynamic change in bandwidth and presented an adaptive QoS technique based on codec switching was proposed in [47]. However, the multi-rate problem is not directly pointed out in this work. The codecs been observed in this work includes G.711, G.726 and Speex. Since Speex is a variable rate codec, it has different bit-rates from 2.1 to 26.4, so the algorithm should be extended to different bit-rates of this codec, but they have not taken all the specific bit-rates of this codec into consideration. As multi-rate codecs like Speex have their own adaptation mechanism, consequently it is better to consider a set of constant bit-rate codecs instead of having a combination of constant and variable bit-rate codecs.

In addition, Packet Loss Rate (PLR) has been implemented as their indicator of congestion and they have used RTP's serial number to calculate the number of lost packets. If PLR exceeds the defined threshold, it shows the quality degradation, so current codec in the algorithm switches to a lower bit-rate codec to reduce the required bandwidth of that call, otherwise it switches to a higher bit-rate codec. PLR is calculated and sent from receiver to the sender but the algorithm would have been more convincing if it had consider delay to have a faster reaction, because according to [3] packet loss is the effect on severe congestion.

They have also proposed a new packet structure, which adds redundant data to the packets in order to mitigate the high packet loss rate. However, increasing the bandwidth consumption and more delay that is required to extract the real data from the redundant data, are the disadvantages of this method.

In the category of codec adaptation schemes, Costa and Nunes [69] came up with the new idea of adapting the transport layer protocol which is done by switching between UDP and TCP during the high network congestion. Although their results have shown that in the saturated network condition switching from UDP to TCP provides better voice quality. However, all the studies discussed so far that have reviewed UDP and TCP shows UDP is the best protocol for real-time application including VoIP. The drawbacks of using TCP for real-time application is that TCP has *retransmissions feature* in order to provide more reliable packet delivery which imposes an undetermined delay in reception of information [26]. In addition, TCP has large overhead of acknowledging each packet that takes more bandwidth than UDP. Furthermore, during congestion, UDP transmit at a steady rate, while TCP stops transmission. Since, UDP achieves tiny higher performance compared to TCP [70], it will be better to use UDP with adding some congestion controlling mechanism over it.

### 2.4 Codec Bit-Rate and Packet size Based Approach

A considerable amount of literatures has been published on adaptation technique using both codec and packet size to address link fluctuations. One of the major studies that addresses the link adaptation is discussed in [51]. This work has studied a distributed way or terminal-oriented scheme that operates by adapting the voice codec of a handset to deal with the LA problem in WLAN. They have considered the terminal-oriented scheme because according to their literature review, most of the schemes that address the LA problem are centralized on AP, so they require employing specific network equipment such as non-standard APs. While their scheme is terminal based and it does not require modifications of network components such as AP and router. In their scheme, codec adaptation contains two parameters: codec type and packetization interval.

While preliminary works, mostly have used the comparison between downlink delays and losses before and after LA, here in [51] the authors claimed that the extensive increase in delay is a good indicator of network congestion but it is hard to estimate in the real-world deployment. So, they came up with easier available factor named phase-jitter to detect congestion.

As described, phase-jitter is the difference between the real packet arrival time and the expected packet arrival time. Any increase in the phase-jitter could be a sign of congestion. Therefore, if the transmission rate of a mobile node is reduced and phase-jitter is increased, this means adaptation is required. However in the recent RTCP information packet the better estimation of quality factors is provided.

They have analyzed channel occupancy time in 802.11b network for the purpose of performing adaptation mechanism. Channel occupancy time is defined as the time that a wireless station occupies a channel for a call during its residence in the channel [71]. Different codecs and packet intervals can affect on different channel occupancy time. So in adaptation phase, codec and packet interval are chosen based on nearest channel occupancy time which was used before LA.

One limitation that needs to be considered is that the process of searching and finding of the most proper codec based on channel occupancy time in the lookup table is a time consuming process which may affects real-time VoIP. Furthermore, in their approach, the codec is adapted only in the call faced to LA. However if the congestion remains in the network some other calls may also need to adapt their codecs. In their most recent study [72], they have also added CAC on the endpoints.

One of the attractive feature in this work is that they have shown all LAs do not cause the capacity to be exceeded thereby all the LAs do not result in WLAN to become congested. In such a this case, the calls that experience LA consequently sense a slight downlink access delay but they still perceive satisfactory quality even after LA.

Lack of bandwidth for VoIP calls have been considered by [52] which follows the work in [51]. They did not consider multi-rate effect and LA functionality but their approach is to perform codec adaptation based on the channel occupancy time of VoIP calls. However, unlike [51] which codec switching is compulsory [52] proposed a scalable speech only when rate changing occurs. This algorithm change only the codec of node who suffers rate changed. The distinguished feature in this work is that different bit-rates of the G.729.1 codec were used for adaptive process. Meaning that, they have used different bit-rates of one codec instead of exchanging codecs. There are 12 possible bit-rates for G.729.1 from 8 to 32 kbps [73], but as they only worked on G.729.1, calls are failing to perceive the best speech quality that provided by G.711 even for non-congested channel.

As the algorithm in [51] studied AP as a bottleneck for network congestion, algorithm in [52] also track this concept but it is different from the work in [51] in monitoring and adaptation phase. The monitoring module in [52] is on the AP and scalable codec controller module is compounded in a wireless nodes.

Furthermore, the method of finding congestion in [52] is based on *media access delay* time for each packet. This factor is recorded when the packet is sent through physical layer in any of the stations. From the other side, monitoring module on the AP observes the media access delay time on the MAC layer and sends this factor to the wireless station by a "beacon frame"[2]. After that, with comparing media access delay time in wireless station and media access delay time in AP congestion can be distinguished. This method needs some modification in the standard beacon frame. In addition, extra time might be taken to send the beacon frame to the station if the network is congested, moreover having other quality factor beside media access delay help to precise the congestion detection which later will be discussed in our methodology chapter.

The comprehensive adaptive speech management for VoIP proposed by [20] has considered fluctuation in general network not specially for WLAN. In this study to control the quality factors, authors used the E-model. After each talk-spurt (non-stop part of speech between two silent gaps), E-model is measured as an instantaneous quality.

---

[2] In IEEE 802.11 WLANs, beacon frame is a periodically transmitted frame for management purposes. It contains the AP's clock on the exact transmission time (not included queue time) also other factor like beacon interval, timestamp, Service Set Identifier (SSID), supported rates, parameter sets, capability information and Traffic Indication Map (TIM) which can be received by other nodes.

However, one of the drawbacks of this method for multi-rate wireless LAN which transmission rate change frequently is that the quality of speech needs to be observed even less than each talk-spurt duration. Besides, since the instantaneous quality is not adequate to make the adaptation decision they came up with the integral perceptual quality factor as a mean of instantaneous quality from the beginning of the call.

Their proposed method performs the adaptation process according the result of Q(M)-Q(T), where QM is the maximum quality level under the given set of speech encoding parameters and QT is the integral speech quality. First limitation of their algorithm is that QT has complex and time-consuming calculation. Second, the threshold for QI that is the instantaneous quality level sets based on some assumptions that can be varied in other assumption. Consequently, in the real network which situation changes over time, it is hard to get threshold for this parameter. Third, to find the most fitting encoding parameter for adaptation, too many comparisons and condition can be caused long adaptation time, which affects real-time voice.

Later, in the similar approach with Sfairopoulou et al. [26], Tuysuz et al. [35], and Alshakhsi et al. [74], they have proposed to use the same factors namely MAC and RTCP for monitoring phase. In addition, Tuysuz et al. [35] have added capacity estimation to their algorithm as a third threshold. Even though, calculating the capacity is a good checkpoint, it is complicated and time consuming calculations that lead to delay of decision in real-time voice.

In the adaptation phase, they also have categorized the packet loss problem into two categories i. e. due to congestion or due to error prone channel. If the packet loss were due to congestion, the frame size of the calls would change to bigger size. Else, the algorithm would change the frame size of other calls. They also have amended their algorithm with eliminating capacity estimation and adding Call Admission Control (CAC) module [75] and later in [76] they have added adaptive jitter buffer module.

Another work which considered adaptive VoIP for multi-rate feature of wireless and LA function has been presented in [50] by Kawata et al. Their topology included the remote wired station (STA) in one side and wireless station (WSTA) in the other side.

During transmission of voice from WSTA to STA via AP, the voice application can adapt coding rate and voice packet size for incoming packet based on the current status of the PHY layer. For modeling fluctuation in wireless link they have used different SNR samples. "ACK messages" is also used for estimation of current link quality. Based on the presence and absence of ACK, the algorithm adjusts encoding rate and packetization interval for the sender. This method of adaptation is used by [77] as well. However, ACK as an adaptation index is failing to give precise quality feedback, by using RTCP packet a better estimation of link condition and quality of voice can be provided.

Also, in their simulation evaluation [77], a set of fixed combined coding rate and packetization interval is assumed for each specific transmission rate. *For example*, in transmission rate of 5.5 Mbps, encoding rate=64 kbps comes with packetization interval=40 millisecond. However, this work would have been more persuasive if the algorithm chooses a different set of codec and packet size dynamically or even separately, (only codec or only packet size) depend on link situation. This is because sometimes only changing codec is enough to mitigate the network congestion. In addition, a media gateway[3] (MGW) function that they implemented in the AP on their method imposes an additional delay to the algorithm. This work has been criticized by [59] pointing out that this proposal is expensive in term of processing power and inefficient transcoding process.

## 2.5 Variable Codec / Other Approaches

AMR is a multi-encoding rate codec with eight modes (Table 2.4), that adaptation decision and switching between rates is based on channel interference. ETSI and its partner 3GPP initially developed this codec for wireless networks and now it is widely used in GSM and UMTS. There are some works that studied AMR for source rate adaptation like [53] and [79].

[3] The media gateway convertor or translator of media available in one type of network like PSTN to the format required in another type of network Like IP network [78] F. Cuervo, N. Greene, A. Rayhan, C. Huitema, B. Rosen, and J. Segers, " Megaco Protocol Version - RFC 3015," ed: Internet Engineering Task Force (IETF), 2000.

A combination of adaptive sender rate and priority marking was presented in [79] to develop QoS control technique. The method of speech quality measurement is the objective MOS [80], instead of individual network impairment measurement such as delay or packet loss. To perform priority marking, some of the speech frames are more important in the flow and they need more priority in comparison to others. Therefore, during congestion frames that marked as higher priorities have more chance to deliver to the receiver side whereas others have a more drop probability. The priority marking in user's level can be applied to the adaptive VoIP algorithm. However, this technique only guarantees the quality of users with high priority.

Along with [79], authors of [81] also proposed the similar session based scheme for active calls in the network. They proposed a method to prevent the congestion by restricting the user with lower priority to the low bit-rate codec during the CAC to keep the acceptable quality of high priority sessions. However, we will consider the method in that all calls have the same priority.

In addition to speech coding parameters (bit-rate and frame size), the size of playout buffer and the amount of data used for Forward Error Correction (FEC) mechanism also can be adapted based on the state of the network. The combination of these parameters was proposed in [82] beside security consideration. They proposed to use audio watermarking techniques in the adaptive algorithm to transfer the information about network conditions to the receiving side instead of using RTCP packets.

Audio watermarking is a technique that special signals embedded in digital audio. This technique is difficult to design and develop. One of the reasons is due to sensitivity of human ear to the embedding noise, while voice sensors have some imperfection and they are not very sensitive [83]. Therefore, it is better to use RTCP packet information for quality measurement.

The QoS control strategy in the [84] is based on congestion and priority consideration. The main goal of this work is to propose a new network layer QoS control scheme for the adapting VoIP based on different priority queues.

The level of congestion in [84] determined by PLR and three categories are assumed for network congestion: minor, general and severe congestion. Under severe congestion (PLR $\geq$10%) traffic monitoring frequency is every 8sec., congestion between 3% and 9% is considered as a moderate or general congestion and traffic monitoring frequency is every 4sec. also for PLR in the range of 1% to 2% monitoring is every 2 Sec. The classification of network congestion in this work is remarkable.

## 2.6 Literature Review Summary

Table 2.5: Summary of related works from literature review.

| Study | Adaptation Index | Method | Parameter Consideration |
|---|---|---|---|
| Oouchi et al. (2002) [62] | Transmission efficiency- jitter and Packet Loss Rate (PLR) | Variable voice payload length | Packet size |
| Mahlo et al (2005) [63] | TFRC scheme (Round Trip Time (RTT) and PLR | Changing the packet rate is sufficient WLAN | Packet size |
| Boonchai (2007) [49] | One way end-to-end delay and PLR | Tuning Packetization intervals in constant bitrate codecs | Packet size |
| Wang et al. (2010) [64] | Delay-MOS | Adjusting number of frames per packet | Packet size |
| Mukhopadhyay (2011) [65] | Delay jitter-PLR | Adjusting the payload size | Packet size |
| Barberis et al. (2001) [33] | Delay and packet loss (PL) from RTCP packet | Decrease input rate; gradually for delay≥ threshold and sharply for PL | Input rate |
| Trad et al. (2004) [43] | RTT, PLR | Changing the coding rate incoming flows | Codec + Forward error correction |
| Sfairopoulo (2006) [48] | Moving Average function for delay, PL, jitter +MAC layer info | Changing coding rate for low rate flows | Codec |
| Sfairopoulo (2007-2008) [67], [68], [26] | RTCP Quality factors, R-factor and MAC layer info | Codec adaptation for some calls including low-rate flows | Codec |
| Chunxia (2009) [47] | PLR Calculated by RTP's Serial number | Codec adaptation Fix and variable bitrate codecs | Codec |
| Costa et al. (2009) [69] | Delay Codec QoS threshold | Codec switching and UDP/TCP switching | Codec + Transport layer protocols |
| McGovern [51], [72] (2006, 2011) | Phase-jitter (Real arrival time-expected arrival time) | Channel occupancy time to be replaced by nearest one | Codec and packetization interval |

| Study | Adaptation Index | Method | Parameter Consideration |
|---|---|---|---|
| Ki-jong et al. [52] (2008) | Media access delay | Monitoring module in AP works crosslayerly with a codec controller module in stations | Codec and packetization interval (Only G.729.1 bit-rates) |
| Myakotnykh [85], [34] (2008, 2009) | E-model<br>Instant quality<br>Integral quality | Adapting packet size then codec (depends on the situation) | Codec and packetization interval |
| Tüysüz et al. (2010)[35] | MAC layer info<br>RTCP Quality factors<br>R-factor<br>Capacity estimation | Categorizing PL and switching to proper packet size and codec switching | Codec and packetization interval |
| Tüysüz et al. (2010) [76] | MAC layer info<br>RTCP Quality factors<br>R-factor<br>Adaptive jitter buffer | Categorizing PL and switching to proper packet size and codec switching | Codec and packetization interval |
| Qiao (2004)[79] | Objective MOS | Coding rate adaptation<br>Higher priority packets pass during congestion | Variable bitrate codec and Priority marking in the user level |
| Mazurczyk (2007) [82] | Watermarking<br>MOS scale | Adaptation of speech codec (bitrate-frame size) And Adaptation of the playout buffer size and amount of data used for Forward Error Correction | Coding rate<br>Playout buffer size<br>FEC data length + security |
| Liu et al. (2011) [84] | PLR using SNMP | Different type of priority queues | Different categories of congestion |

## 2.7 Chapter Summary

This chapter has presented the background of adaptive rate VoIP studies. These studies have been classified into four categories based on their adaptation mechanism. In each category the most related studies have been discussed and their parameters and mechanisms investigated in detail to verify their advantages and disadvantages. Based on the classification, some of the studies have considered only codec adaptation while some others considered only packet size adaptation. The proposed algorithm in this study will consider both approaches to exploit their benefits.

One of the main disadvantages of the algorithms that used different codec as their adaptation method is high cost including paying a license fee for each codec and/or upgrading the current hardware such as routers or gateways with the new codecs. Another disadvantage is that switching from the higher bit-rate codec to the lower bit-rate codec has the obvious effect in speech quality perceived by end users.

The algorithms that have studied adaptive packet size as their adaptation method have lower costs in term of paying extra fees and also the modification of the current equipment are not required. They are more transparent to the end user since the codec does not change [49]. Furthermore, packet adaptation can adapt the output rate while codec is fixed. Hence, it has the benefit of using high bit-rate codecs and at the same time adapts the output rate. However, most of codecs support 10-40 ms audio in each RTP packet thereby the level of rate adaptation using packet size is restricted. Besides, the packetization delay that is posed by the larger packet size limits this approach for the severe congestion, where delay is high and larger packet size adds extra delay.

The basic idea gathered from the above discussion is that both approaches have some advantages and disadvantages. In order to minimize the disadvantages of codec adaptation (high cost) packet size adaptation is proposed for low to, moderate congestion and codec adaptation will be done only in severe congestion where packet size adaptation is fail to reduce the output rate remarkably.

As mentioned above our proposed algorithm considers codec and packet size adaptation to obtain the benefits of both approaches and reduce the drawbacks of them. Yet, in the area of research with consideration of both parameters, there are some researches that have focused only on LAN networks or they have not focused on multi-rate effect that causes by the IEEE 802.11 standard. Furthermore, in some others, their adaptation method needs to be improved to reduce the adaptation cost and/or their calculation parts need to be minimized and/or the right adaptation time need to be determined.

This study tries to address the issues found in previous studies. This research focuses on the multi-rate effect which caused by LA in IEEE 802.11WLANs. Additionally, in the adaptation process this research tries to find the balance between codec and packet size adaptation. Furthermore, for finding the right adaptation time the focus of this study is on RTCP-XR that is including the set of VoIP quality metrics and network condition information. This research is the foremost work that uses RTCP-XR in its algorithm to reduce the calculation part of previous researches and to get more information that is accurate for adaptation process.

In this study, the method of finding the right adaptation time is to monitor VoIP stream beside delay jitter (as an instant quality metric) and MOS (as result of the general quality metric) which are the best parameters among all the parameters chosen by earlier researches. This method is similar to [26] and [35] but there are two main problems in these two studies. First, they used older RTCP and they need to calculate the R-factor. However, in RTCP-XR VoIP metrics block, R is provided, so calculation part can be omitted which can affect the faster execution of algorithm. Second, when the transmission rate falls to the lower rate, previous algorithms trigger the codec adaptation immediately, while it is not necessary to adapt the codec in every transmission rate changes, and sometimes the system can sustain these changes which will be investigated later.

# CHAPTER 3

# METHODOLOGY

## 3.1 Chapter Overview

The goal of this study is introducing an adaptive VoIP output rate algorithm for WLANs, and all of the steps to achieve this goal have illustrated as a methodical sequence in Figure 3.1. In this dissertation, the methodology chapter is not only illustrates the research flow but also answers the essential research questions which are required to achieve a real implementation of VoIP rate adaptive algorithm.

This chapter comprises the designed simulation model and the steps used for different simulation scenarios to identify the best quality factors needed for adaptation instance. In addition, different adaptation methods (codec and/or packet size adaptation) will be examined to achieve the best balance between codec and packet size adaptation for the adaptation procedure of the proposed algorithm. Then based on the two above procedures (finding the best adaptation instance and the best adaptation process) the new VoIP rate control algorithm will be proposed. Finally, evaluating the algorithm, which is the last part of methodology, will be done in the result chapter.

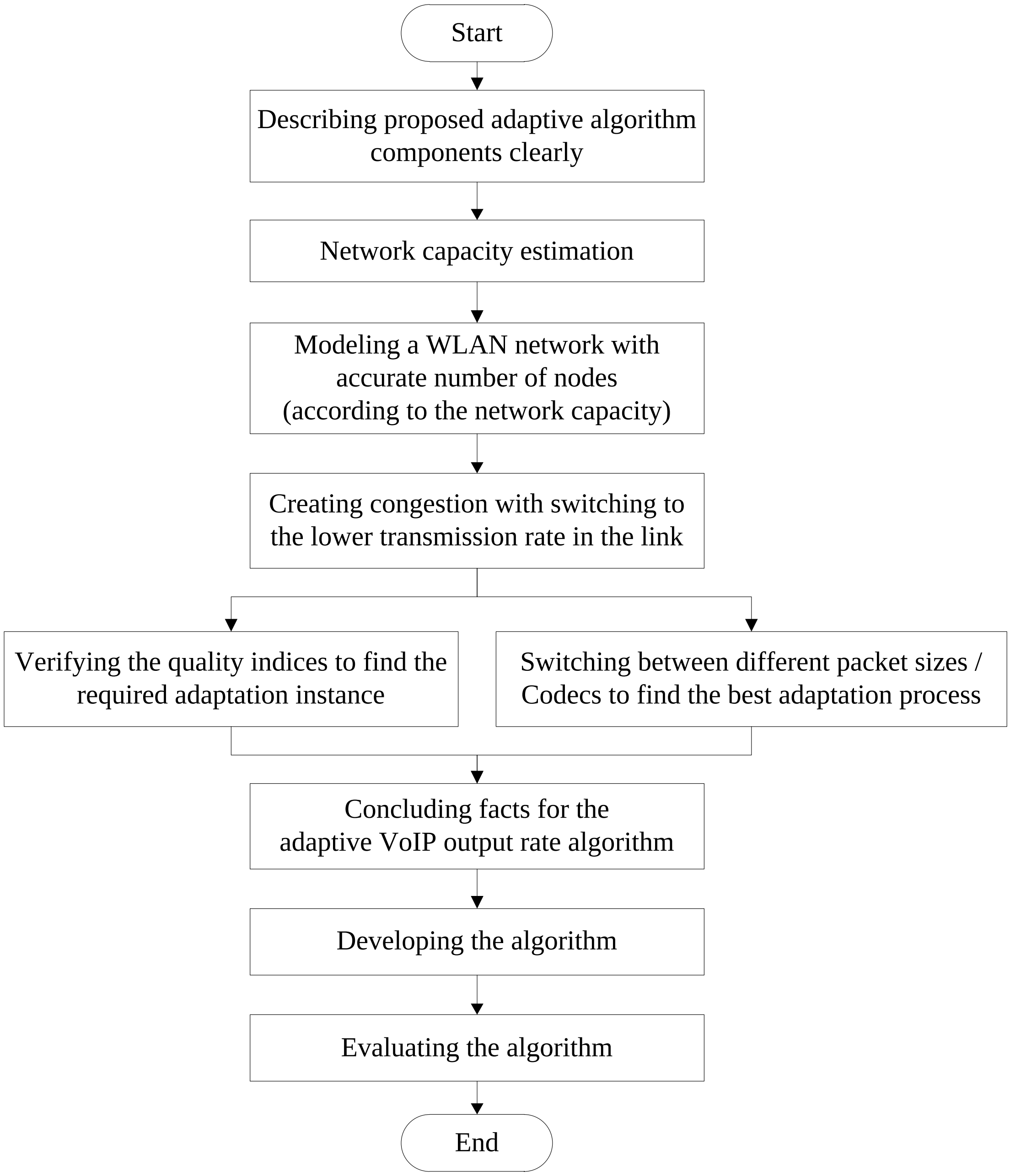



Figure 3.1: Methodology flowchart.

## 3.2 Answering the Essential Research Questions

As mentioned in the previous chapters, a multi-rate wireless LAN does not guarantee available bandwidth for transmission of real-time data, and channel condition is changed frequently. On the other hand, VoIP uses UDP as a transmission layer protocol that does not provide any congestion control mechanism. So, some mechanisms should be applied to control the load of the network during network saturation. Usually, ‘application layer’ and ‘lower layer’ work together with the cross layer approach to implement such a mechanism. That means, by these mechanisms, application layer parameters may adapt according to the link condition in the physical layer. The main component of these kinds of adaptive process will be discussed in the following.

Adaptive algorithm for VoIP can be included mechanism for adjusting parameters as below [82], [34], [76] :

- Codec (output rate and voice frame size).
- Playout buffer size (jitter buffer size).
- Amount of generating redundant data for Forward Error Correction (FEC) mechanism.

The proposed algorithm in this study uses two adaptive parameters, namely coding rate and payload size to manage the speech output rate according to each transmission rate. Therefore, this algorithm enables an application to reduce the load of the traffic in the network to prevent congestion which consequently can lead to lower speech quality and higher call dropping or blocking rate.

The main concerns of this algorithm is as follows;

1. When to make the decision for adaptation?
2. How to monitor or measure the speech quality in real-time?
3. How to control coding parameters based on specific criteria?
4. How to apply new parameters in the call?
5. Where to apply this adaptation scheme?

### 3.2.1 When to Make the Decision of Adaptation

When wireless channel keeps the transmission rate at a fix rate, applying the adaptation mechanism is not necessary. However, if the transmission falls to a lower rate congestion may happen and so adaptation may be required. Therefore, this question should be separated into two sub-parts:

- How to detect changes in transmission rate?
- How to determine congestion?

#### *3.2.1.1 How to Detect Transmission Rate Changes*

Identifying changes in transmission rate is based on MAC layer information. Through monitoring of the changes in VoIP media stream, if any alarm related to changing transmission rate (either falling down or rising up) is triggered, the algorithm will start to take action.

#### *3.2.1.2 How to Determine Congestion*

Basically, UDP does not provide congestion control mechanism, so RTP is used over UDP (Figure1.1) to rectify the UDP's limitation. RTP has appended a sequence number and timestamp to the frames that can detect packet loss and can allocate proper speech frames playback at the receiver side. Furthermore, timestamp enables system to calculate delay and jitter for QoS measurement purposes. The RTP packet has the following format [86], [87]:

<table>
<tr><td>Version =2</td><td>Padding</td><td>Extension</td><td>CSRC Count</td><td>Marker</td><td>Payload Type</td><td>Sequence number</td></tr>
<tr><td colspan="7">Timestamp</td></tr>
<tr><td colspan="7">Synchronization source (SSRC) identifier</td></tr>
<tr><td colspan="7">Contributing source (CSRC_1) identifier</td></tr>
<tr><td colspan="7">…</td></tr>
<tr><td colspan="7">Contributing source (CSRC_n) identifier</td></tr>
<tr><td colspan="7">P A Y L O A D</td></tr>
</table>

Figure 3.2: RTP packet format.

Furthermore, RTCP Associated with RTP provides feedback on the quality of the data transmission. This is related to the flow and congestion control function of UDP transport protocols. RTCP does not transport any streams itself since it is a controlling packet over UDP.

RTCP feedbacks can be used to control adaptive encoding rate without even receiving the real data so it can be used for network monitoring. As RTCP packets contain information for quality of service monitoring, adaptive algorithm can use RTCP packet parameters to identify congestion during calls.

RTP provides QoS feedbacks by sending network statistics and link information every 5 seconds and the traffic does not go above 5% of total traffic. These feedbacks are obtained from the RTCP Sender Report (SR) and Receiver Reports (RR). SR and RR exchange information on number of transmitted packets, number of lost packets, end-to-end delay and delay jitter and multicast it to the network. For instance RR can be used by sender or by other receivers or by a third-party monitor, based on the statistics of this report, sender can modify its transmissions as shown in Figure 3.3.

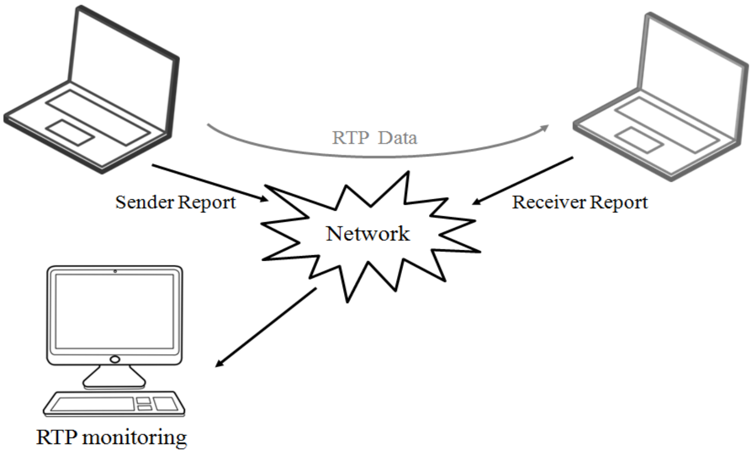



Figure 3.3: RTP data and report flow.

The format of the RTCP packets is similar to RTP packets. Each RTCP packet starts with a header similar to RTP packets. SR and RR formats are demonstrated in the following Figures 3.4 and 3.5:

| V=2 | P | Report Count | PT=SR=200 | Length |
|---|---|---|---|---|
| SSRC of the sender | | | | |
| NTP timestamp (MSB) | | | | |
| NTP timestamp (LSB) | | | | |
| RTP timestamp | | | | |
| Sender's packet count | | | | |
| Sender's octet count | | | | |
| First reception report block (SSRC_1) | | | | |
| … | | | | |
| Last reception report block (SSRC_n) | | | | |

Figure 3.4: Format of Sender Report (SR).

| V=2 | P | RC | PT=RR=201 | Length |
|---|---|---|---|---|
| SSRC of the sender | | | | |
| SSRC of the first source | | | | |
| Fraction lost | | Cumulative number of packets lost | | |
| Extended highest sequence number received | | | | |
| Inter-arrival jitter estimate | | | | |
| Last sender report timestamp (LSR) | | | | |
| Delay since last sender report (DLSR) | | | | |
| … | | | | |
| Last reception report block (SSRC_n) | | | | |
| Profile-Specific extensions | | | | |

Figure 3.5: Format of Receiver Report (RR).

RR and SR have the similar format except two differences. First, the packet type field filled with different constant (200 for SR and 201 for RR) and second, the five words of sender information including the NTP and RTP timestamps and sender's packet counts and octet counts have been removed [86]. The information provided by these packets can be used to determine QoS factors and congestion level.

### 3.2.2 How to Monitor or Measure the Speech Quality in the Real-Time

SR and RR provide some fields that allow quality assessment for multimedia application. Here in this section the speech quality metric has been discussed. Generally, the difference between the last two reports received can be used to estimate the recent quality of the communication. For example, to calculate the packet loss rate in a time period, first and last reception reports of this period are used and the difference of the cumulative number of packets lost field in the RTCP packet gives the number lost during that time period. The ratio of these two fields is the packet loss fraction over that time period.

According to RFC [86] packet lost ratio fraction can gain by the formula below:

$$Fraction\ Lost = \frac{N_{PL}}{N_{PR}} \tag{3.1}$$

where:

$N_{PL}$ is the number of packets lost and $N_{PR}$ is the number of packets expected to receive and

$$N_{PR} = SN_H - SN_1 \tag{3.2}$$

where:

$SN_H$ is highest sequence number received in an RTP data packet and $SN_1$ is the initial sequence number.

Furthermore, the receiver reports can be used to calculate the Round-Trip Delay (RTD) between sender and receiver. The RR includes the LSR (time of receipt of the most recent RTCP packet from the last SR) and DLSR (delay since last SR) fields that allow the sender to calculate the round-trip delay directly based on the formula in (3.3):

$$RTD = RR_{Received} - SR_{Sent} - delay \text{ [88]} \tag{3.3}$$

where :

$RR_{Received}$ is the time that source received the most recent RTCP report,

$SR_{Sent}$ is the timestamp of last Sender Report (LSR field in Figure 3.5) and

$delay$ is the delay since last SR report (DLSR field in Figure 3.5).

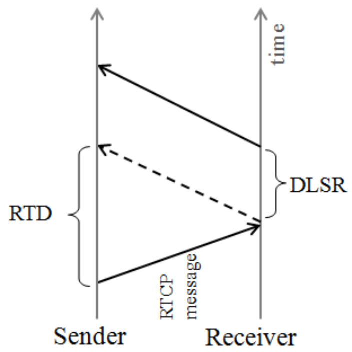



Figure 3.6: Timing diagram for calculation of round trip delay.

Following the RTCP, RTCP Extended Report (RTCP-XR) defined in RFC 3611 [89]. XR packets report statistic information beyond that already contained in the standard RTCP's sender report or receiver report. Also, in RTCP-XR the frequency is not strictly to 5 second instead it is “on-demand”. For example it includes comprehensive jitter statistics information as follows:

<table>
<tr><td>BT=6</td><td>L</td><td>D</td><td>J</td><td>ToH</td><td>Rsvd.</td><td colspan="2">Block length=9</td></tr>
<tr><td colspan="8">SSRC of source</td></tr>
<tr><td colspan="6">Begin_seq</td><td colspan="2">End_seq</td></tr>
<tr><td colspan="8">Lost_packets</td></tr>
<tr><td colspan="8">Dup_packets</td></tr>
<tr><td colspan="8">Min_jitter[4]</td></tr>
<tr><td colspan="8">Max_jitter</td></tr>
<tr><td colspan="8">Mean_jitter</td></tr>
<tr><td colspan="8">Dev_jitter</td></tr>
<tr><td>Min_ttl_or hl</td><td colspan="3">Max_ttl_or hl</td><td colspan="2">Mean_ttl_or hl</td><td>Min_ttl_or hl</td><td>dev_ttl_or hl</td></tr>
</table>

Figure 3.7: The Statistics Summary Report Block.

In addition, XR is useful for reporting the VoIP quality information to end user terminals. It has a VoIP metrics report block that contains quality metrics such as packet loss, delay, and jitter (Figure 3.8) to identify the factors that cause degradation in QoS. This block is also able to report whether packets are lost due to IP channel fluctuation or due to discarding in the jitter buffer. Moreover the RTCP-XR, offers the voice quality information given by the E-Model and MOS to facilitate feedback of quality in the network [14].

[4] min_jitter: The minimum relative transit time between two packets in the above sequence number interval. All jitter values are measured as the difference between a packet's RTP timestamp and the reporter's clock at the time of arrival, measured in the same units.

| | | | |
|---|---|---|---|
| BT=7 | reserved | block length=8 | |
| SSRC of source | | | |
| loss rate | discard rate | burst density | gap density |
| burst duration | | gap duration | |
| round trip delay | | end system delay | |
| signal level | noise level | RERL | Gmin |
| R factor | Ext. R factor | MOS-LQ | MOS-CQ |
| RX config | reserved | JB nominal | |
| JB maximum | | JB abs max | |

Figure 3.8: VoIP Metrics Report Block.

Both MOS types for Conversational Quality (MOS-CQ) and Listening Quality (MOS-LQ) are commonly used in VoIP applications. MOS-LQ measures the quality of speech for listening purposes only and it does not consider any of bidirectional effects, such as delay and echo. MOS-CQ considers listening quality in each direction, in addition to the bidirectional effects [89].

Since RTCP-XR provides the R-factor and objective MOS besides the other main QoS parameter like packet loss, delay and jitter. So, in order to provide more comprehensive information and faster reaction the proposed technique shall use the RTCP-XR packet.

### 3.2.3 How to Find the Adaptation Instance Criteria

In order to control the coding parameter according to the link condition in wireless link there is a need to measure the quality of speech. As mentioned earlier, RTCP-XR enables multimedia applications to have an almost accurate estimation of channel and network condition. Either E-Model or MOS can be used to track changes in quality and provide instantaneous quality monitoring.

The basic formula for the E-Model is below.

$$R\ Factor = Ro - Is - Id - Ie + A \tag{3.4}$$

where:

R Factor is the Overall network quality rating (ranges between 0 and 100.

Ro is the signal to noise ratio

Is is the impairments simultaneous to voice signal transmission

Id is impairments delayed after voice signal transmission and

Ie idthe effects of Equipment (e.g. codecs)

A is the advantage factor (attempts to account for caller expectations)

The best criteria to determine the adaptation time is when quality metrics pass their acceptable limit. The medium and high quality boundary for MOS and R (section 1.3) are shown below [90]:

$$MOS \geq 3.60\ fair\ quality \qquad\qquad R \geq 70.07\ fair\ quality$$
$$MOS \geq 4.03\ high\ quality \qquad\qquad R \geq 80.16\ high\ quality$$

As mentioned in chapter 1, R factor and MOS[5] are convertible to each other. This conversion is using the followed equivalence [18], [91], [53]:

$$MOS = \begin{cases} 1 & R \leq 0 \\ 1 + 0.035R + 0.000007R(R-60)(100-R) & 0 < R < 100 \\ 4.5 & R \geq 100 \end{cases} \tag{3.5}$$

Table 3.1 shows the range of R-factor and MOS from the user satisfaction point of view [53]:

Table 3.1: Equivalency of R-Factor and MOS [53].

| R-Factor | MOS | Perceived Quality by Users |
|---|---|---|
| 90-100 | 4.3-4.5 | Very Satisfactory |
| 80-90 | 4.0-4.3 | Satisfactory |
| 70-80 | 3.6-4.0 | Satisfactory for Some Users |
| 60-70 | 3.1-3.6 | Dissatisfactory for Many Users |
| 50-60 | 2.5-3.1 | Almost All Users Dissatisfied |
| 0-50 | 1.0-2.5 | Not Recommended |

Figure 3.9 shows the relation of MOS and R-factor. The darker area shows good quality and lighter area shows medium quality.

[5] Objective MOS

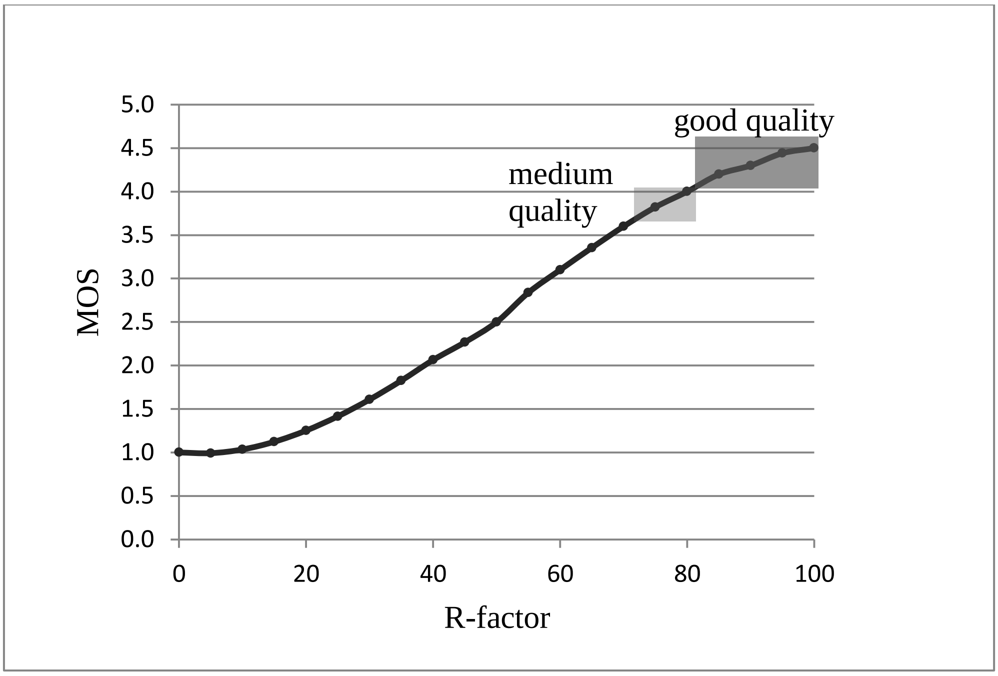



Figure 3.9: Relevancy of R factor and the MOS.

While both E-Model and MOS can be used to track changes in quality and provide instantaneous quality information, MOS will be used in this dissertation since it is the most accepted voice quality assessment method [90] and also OPNET modeler provide it in the voice application statistic. However, it is possible to map MOS to R in order to compare this work with most related ones.

Most related previous researches which used RTCP and MAC monitoring (namely [26] and [35]) have taken R as an index to determine the quality and congestion threshold; if the quality is below threshold (R<70) means the network is saturated (congested) so adaptation should be commenced to change the coding parameters. In this study, the proposed algorithm has used objective MOS for quality estimation of channel and network. The MOS below 3.6 shows the low speech quality and it means the network is already congested. Although MOS is a good index to show the link condition but it is the result of overall quality and it cannot predict the link congestion before it causes to severe speech quality degradation. Hence, the instant quality factors such as delay and jitter can be used to predict the congestion before it result in the low speech quality.

According to [33] work packet loss is a result of severe congestion and delay is the result of low to moderate congestion. The main point extracts from their work is that any reaction to packet loss would be late, so delay should be considered to detect congestion faster. Therefore, in the proposed algorithm, also jitter interval value obtained by RTCP-XR is used to keep track of congestion on the link.

From another perspective, the transmission rate reduction is the main sign to perform the coding adaptation that is the idea of most related previous works namely [26] and [35] which have used the MAC alarm for the fast adapting reaction to the transmission rate reduction. However, adaptation of codec right after the MAC alarm is not an optimal solution to deal with network congestion. Because every change in does not require codec adaptation, rather than that sometimes the system can sustain the current codec [51]. This is another reason that the proposed adaptation algorithm in this study uses the delay and jitter beside MAC and MOS.

The work in [51] also conducted a set of experiments to demonstrate that the effect of LA does not lead WLAN to become congested in all the cases. Their results have shown in many cases calls continue with an acceptable quality even after LA, but the call faced with LA experiences a very small increase in downlink access delay and the other calls that have unchanged transmission rates will go through a small increase of one way delay. The results of the work in [51] also showed that after WLAN capacity was exceeded, LA will lead to congestion. It means if LA causes the total network capacity to become full it is the time for adaptation because calls suffer from the unsatisfactory quality and very high access delay not only for the calls suffer from LA but for all calls. One of the conclusions from their results is when the capacity of a WLAN reaches near the limit, it causes a tiny increase in downlink delays.

Since exceeding the capacity limits leads to congestion, to develop an accurate model, network capacity should be identified. Therefore, upper bound of WLAN network capacity for different transmission rate and different type of call's parameters is estimated in section (3.3).

### 3.2.4 How to Control Coding Parameter Based on Specific Criteria

To answer this question firstly these two sub questions need to be answered:

- What is the effect of different codec (different compression rate) on VoIP quality?
- What is the effect of frame size variation on VoIP quality and bandwidth consumption?

#### *3.2.4.1 How Different Codecs Affect VoIP Quality?*

Choosing a proper codec for voice signals in wireless networks represents a challenge because it affects the voice quality and bandwidth consumption simultaneously [92]. The optimal codec for VoWLAN depends on factors such as bandwidth, number of calls, expectation quality for user and network policies.

Some of the codecs provide higher compression and as a result, lower utilization of bandwidth, so they are supporting more calls and on the other hand, some others provide lower compression and so higher bandwidth consumption and less number of calls [6]. From another point of view, higher compression codecs has lower bit-rate which means lower perceived quality. Therefore to choose the optimal codec for VoWLAN at the development time, it is important to consider which factor is more important; higher quality or minimum utilization of bandwidth.

Two main speech codecs namely G.711 with 64 kbps and G.729 with 8 kbps bit-rate are widely used. G.729 utilizes one eighth of the bandwidth compared to G.711, it means G.729 supports more calls but with lower quality. Table 3.2 tabulates the main properties of different codecs (from [15] and [93]).

Table 3.2: Main characteristics of well-known codecs.

| Codec | Compression Technique | Bit-Rate (Kbps) | Frame Length | Complexity MIPS | Encoding Delay(ms) | Loss Tolerance | Speech Quality (MOS) |
|---|---|---|---|---|---|---|---|
| G.711 | Pulse Code Modulation (PCM) | 64 | 0.125 | 0.1 | 0.13 (negligible) | 7-10 % | 4.2 |
| G.726 | Adaptive Differential PCM (ADPCM) | 16, 24, 32, or 40 | 0.125 | 12 | 0.4 | 5 % | At 40 kbps=4 |
| G.729 | Conjugate Structure Algebraic Code-Excited Linear Prediction (CS-ACELP) | 8 | 10 | 22<br>12 for G.729.A | about 25 | <2 % | 4.0 |
| G.723 .1 | Algebraic Code Excited Linear Prediction (ACELP) | 6.3 or 5.3 | 30 | 16 | about 67.5 | <1 % | 3.9/ 3.7 |

Figure 3.10 gives the results of the MOS versus different type of codecs when the number of calls is 20 (a crowded network). It is obvious that in the highly loaded network, G.729 provides better quality and it has the advantage of having a less average delay of communication. This study confirmed that G.729 provides an optimum quality for VoIP when number of calls are high [94].

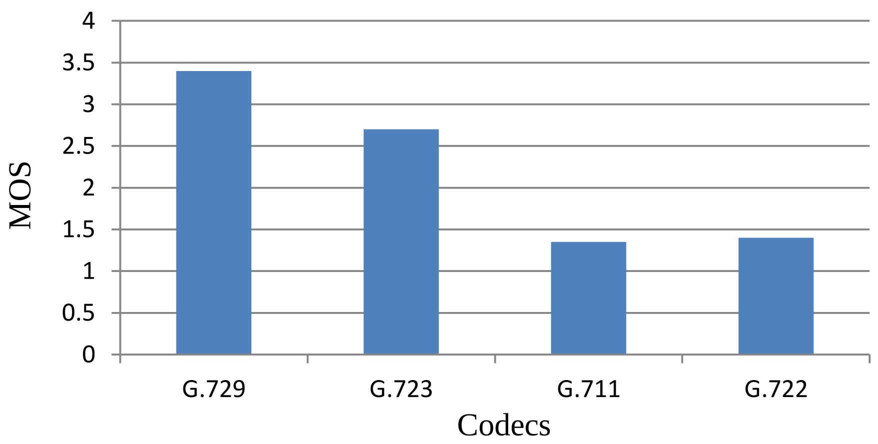



Figure 3.10: MOS versus codec for 20 generated calls [94].

According to Figure 3.10, G.729 and G.723 consume lower bandwidth and allow more calls. Unlike G.711, these two codecs are more robust to voice degradation at the traffic congestion time, because even with audio packet loss, the quality stays acceptable (greater than 3). However, when the network is not congested, G.711 provides the best perceived quality which is almost equal to PSTN network quality. Due to the popularity of G.711 and G.729 codec, they will be the focused codecs in this dissertation. It should be mentioned here that G.711 codec does not have licensing fee, so it can be used in VoIP applications freely. G.729 is a licensed codec, but most of the well-known VoIP phone and gateway have implemented this codec in their chipset, i.e. the licensing fee has already been absorbed by the manufacturer of the device [95], [96].

*3.2.4.2 How Different Frame Sizes Affect VoIP Quality and Bandwidth Consumption*

As we mentioned earlier besides codec, different payload sizes will also affect transmission efficiency including bandwidth utilization and delay. Hence, the amount of encoded voice that can be placed per packet should also be a factor to be considered. Figure 3.11 shows speech coders are generally frame-based [93].

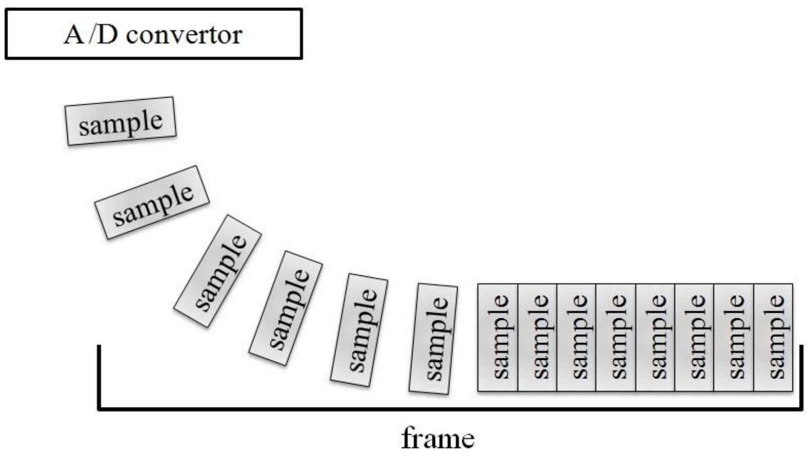



Figure 3.11: Frame based codec.

In the end-to-end delivery of speech packets, each packet requires a fixed overhead for (IP/UDP/RTP) headers to be added to the encoded speech packets. Size of this packet overhead is fixed and is the same for small and large packets. In view of the fact that WLAN protocol also adds a large overhead to the payload (Figure 3.12), the number of simultaneous supported calls is less than expected.

The in the bandwidth estimation, the size of packet is calculated based on below formula [92]:

$$\textit{Total packet size} = \textit{Layer 2 header} + (\textit{IP/UDP/RTP})\textit{header} + \textit{voice payload size} \tag{3.6}$$

| | | | | | |
|---|---|---|---|---|---|
| IEEE 802.11 PHY Header | IEEE 802.11 MAC Header | IP Header (20b) | UDP Header (8b) | RTP Header (12b) | Payload |

Figure 3.12: Packet format of VoIP over IEEE 802.11.

If this large overhead added to the packet that contains only one frame, the amount of overhead could be larger than the real data size (Figure 3.13). Therefore, to maintain overhead lower than the data part in each packet, most of codecs support multiple frames in each packet [93] (Figure 3.14).

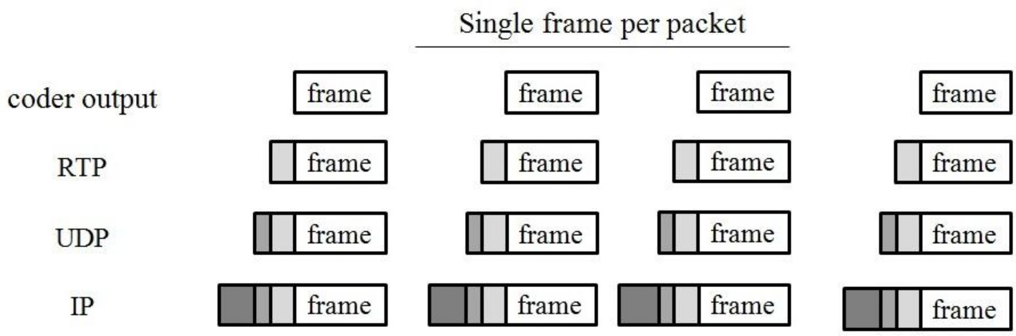



Figure 3.13: Single frame per packet.

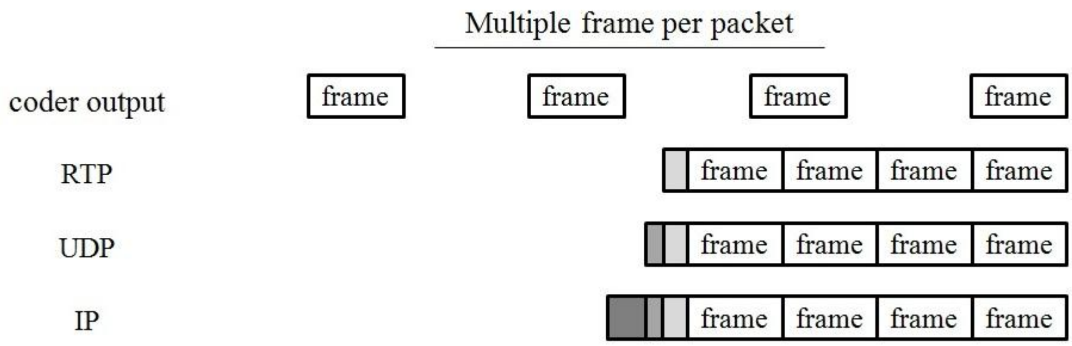



Figure 3.14: Multiple frame per packet.

Consequently, VoIP system with larger voice payload has higher transmission efficiency. From the other side, by use of larger payloads, more audio (i.e. a longer period of time) is required in packetizer to gather voice frames as a single packet which causes more end-to-end delay.

Number of frames per packet also has a direct effect on the bandwidth consumption. As an example the work in [97] studied wireless multi-hop environment to transmit different codec with varying number of frames per packet. Their results show for coding the voice signal with G.729 and 1 frame in each packet, 406 Kbps bandwidth is required while with 5 frames only 87 Kbps bandwidth is required. As well for G.711 with 1 frame per packet voice transmission requires 462 Kbps bandwidth while with 5 frames in each packet 143 Kbps is required (table 3.3).

Table 3.3: Relation of packet size and bandwidth consumption.

| Codec | Number of frame(s) per packet | Bandwidth consumption (Kbps) |
|---|---|---|
| G.711 | 1 | 462 |
| | 5 | 143 |
| G.729 | 1 | 406 |
| | 5 | 87 |

Although more frames per RTP packet leads to less required bandwidth and higher transmission efficiency but at the same time quality should be taken into consideration too. Since the larger packet size encounters with higher end-to-end delay and higher data loss.

Oouchi et al. [62] have investigated the above issue and they have demonstrated voice quality levels with different length of voice packets under various network conditions. They have shown VoIP systems with shorter frame length can reduce the packet loss for maintaining a good speech quality. The summary of their work tabulated in Table 3.4.

Table 3.4: Characteristics of speech frame length

| | Bandwidth consumption | Packet loss effect | Advantage |
|---|---|---|---|
| Long frame length | Low | Higher ratio of packet loss | High transmission efficiency |
| Short frame length | High | Tolerant to packet loss | Lower degradation in voice quality |

Researchers have found that it is better to use 10 to 30 ms of speech packet length to trade-off between network efficiency and delay [16]. However, most of the codecs support 10-40 ms audio in each RTP packet, and many commercial implementations of IP phones use a payload size of 20 or 30 ms [61]. The current payload size can be determined by sender information.

According to [34] in congested networks adaptation of voice payload size is more useful rather than codec adaptation, because it may offer better quality in comparison with adapting the codec to the higher compression codec. However, if the congestion still resists after packet size adaptation, higher compression codec can be helpful. The next section (3.2.5) will discuss how different codec and packet size adaptation can affect the quality and bandwidth.

### 3.2.5 How to Apply New Parameter in the Call

To apply the new parameter in the call, a signaling protocol in the application layer is required. H.323 [98] and SIP [99] are two basic protocol architectures for multimedia over IP. H.323 is recommended by ITU-T and Internet Engineering Task Force (IETF) developed SIP. H.323 is older than SIP and it can be found in many products and organizations that have employed VoIP technologies in earlier times. Although SIP is young, it is popular for its lower complexity.

The comparison between these two protocols can be found in [100] from complexity, extensibility, scalability and enabling features point of view, which demonstrate SIP is less complex and more scalable and extensible than H.323. In addition, the study in [101] has shown that in different queuing policies for severe traffic congestion, SIP handles more number of successful session establishments comparing to H.323. The further comparison between them can be found in [102] that is based on their features, services and inter-working with PSTNs. It illustrates H.323 standard is more particular in QoS and mobility. Furthermore, it shows SIP designed as a general transaction protocol for setup and tear down of the sessions. Besides, H.323 is more supportive in conjunction with PSTNs. Since the proposed model in this study is a WLAN environment (not including PSTN networks) thus to obtain the other advantages of SIP in supporting more number of successful session establishment during congestion and lower complexity, SIP is chosen to be used in this study.

A SIP signaling protocol in the application layer is used for establishing, manipulating and terminating sessions. The major functions of the SIP signaling protocol are: (1) to locate the resources or parties; (2) to invite to service sessions; and (3) to negotiate the service parameters [15]. SIP implements text-based request-response messages in client-server architecture. The INVITE request is the most important types of requests which is used to invite a user to a call. The ACK is acknowledgment from caller to callee to confirm the latest response reception. The BYE request is used to terminate the session between two parties. Besides, there are three other types of message; CANCEL request is to cancel any incomplete searching for a users but it does not tear down a current call, OPTIONS request is to query the capabilities of servers and REGISTER request is to register a user with a SIP server [15].

SIP can be used for call management such as adding, dropping, or transferring participants. Moreover, SIP enables *changing the features of a session while it is in progress*. It means, during the session, if the algorithm decides to change the characteristics of the media session includes new codec and frame size, this process will be accomplished by sending a ***re-INVITE*** request in order to ***re-negotiate*** the characteristics of the media session (Figure 3.15).

The re-INVITE request refers to the existing dialog, so the other party knows that it is to modify an existing session instead of establishing a new session. If the negotiation completes successfully two parties will start the session with the new parameters otherwise (e.g. when the new codec is not acceptable for one of the parties) the failure of the re-INVITE will happen but it does not cause the existing call to be dropped, rather it will continue with previous parameters. Thereby if the algorithm decides to drop the call it should be done by the algorithm itself [99]. It should be mentioned that RFC 3611 is independent from SIP, therefore a more recent standard (RFC 6035) allows RTCP-XR metrics to be reported through SIP.

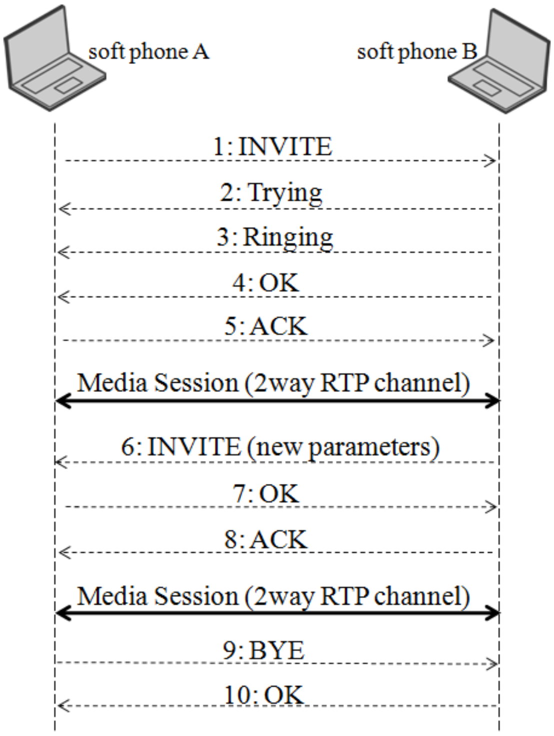



Figure 3.15: A SIP session messaging.

### 3.2.6 Where to Apply These Adaptation Schemes

There are two possible ways to implement a VoIP adaptive algorithm in the network; centralized (on access point) or distributed (on workstations). In the distributed mode (workstation-based), each station monitors its link condition and makes the decision of adaptation locally, while in the centralized mode AP is responsible to make the decision of adaptation.

In the ***centralized mode***, AP has the information of the entire network so it is a more optimized solution. But in this mode some modification is needed on the AP, therefore it demands the programmable APs which are expensive.

In the ***distributed mode,*** the stations monitor their physical layer condition, the processing load of the algorithm will be distributed among the stations rather than AP alone and it is easier to implement and faster in reaction [67]. Also, in practice VoIP are often capable of adjusting their multimedia quality and transmission rates on the workstation-basis, especially when software codes are used [46]. Consequently, in this study the distributed mode will be used.

### 3.3 Network Capacity Estimation

In accordance to the proposed research methodology flowchart (Figure 3.1), the first step is to estimate the capacity of the network including the number of wireless nodes and the number of calls assigned to these nodes to anticipate when congestion is going to happen.

In fact, congestion happens when the numbers of calls exceed the possible capacity. Exceeding the capacity limit causes quality degradation such as jitter, delay and packet loss. So, network capacity's upper limit is determined by the maximum number of possible calls while having the acceptable quality.

To find the maximum number of possible calls (network capacity upper limit) a simulation scenario is set up with two wireless workstations as a sender/receiver that are connected via an AP, and the number of calls is increased regularly between them. Whilst the overall quality of calls is acceptable means, there is still space for more new calls but when the quality of calls starts to degrade means the network is saturated and it cannot handle more calls, so the current number of calls is determined as its capacity. Flowchart of this approach is shown in Figure 3.16. This simulation approach should be repeated for all the transmission rates which are 1, 2, 5.5 and 11 for 802.11 b standard also different combinations of codec and packet size. The attribute of wireless stations (wireless nodes) tabulated in Table 3.5.

Table 3.5: The station attributes.

| **Attribute** | **Value** |
|---|---|
| Transmit Power (W) | 0.005 |
| Packet Reception-Power Threshold (dBm) | -95 |
| Max Receive Lifetime (Millisecond.) | 30 |
| Buffer Size (Byte) | 32000 |

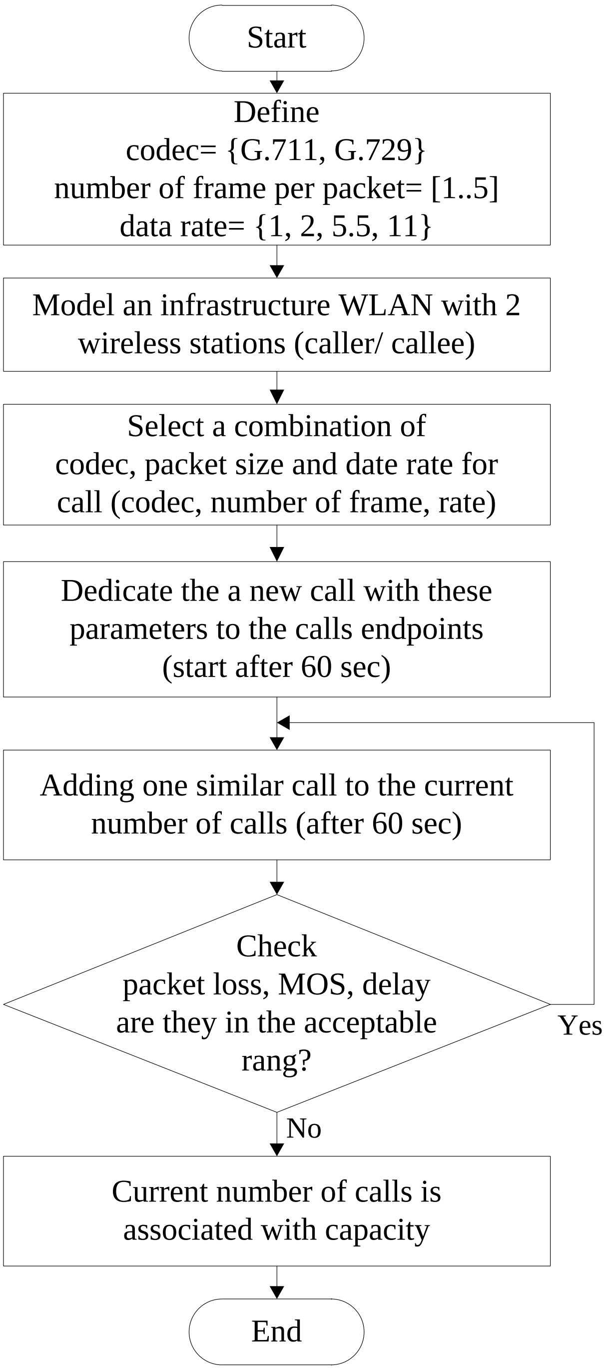



Figure 3.16: Capacity finding flowchart.

In OPNET, modeling of any application such as VoIP needs a configuration node named “Application Configuration” that is used to set up the parameters likes codec, size of packet and two endpoints of each call (Figure 3.17). Further, it is necessary to set “Profile Configuration” node for defining the application behavior such as application start time, repeatability of application and duration of simulation run (Figure 3.18).

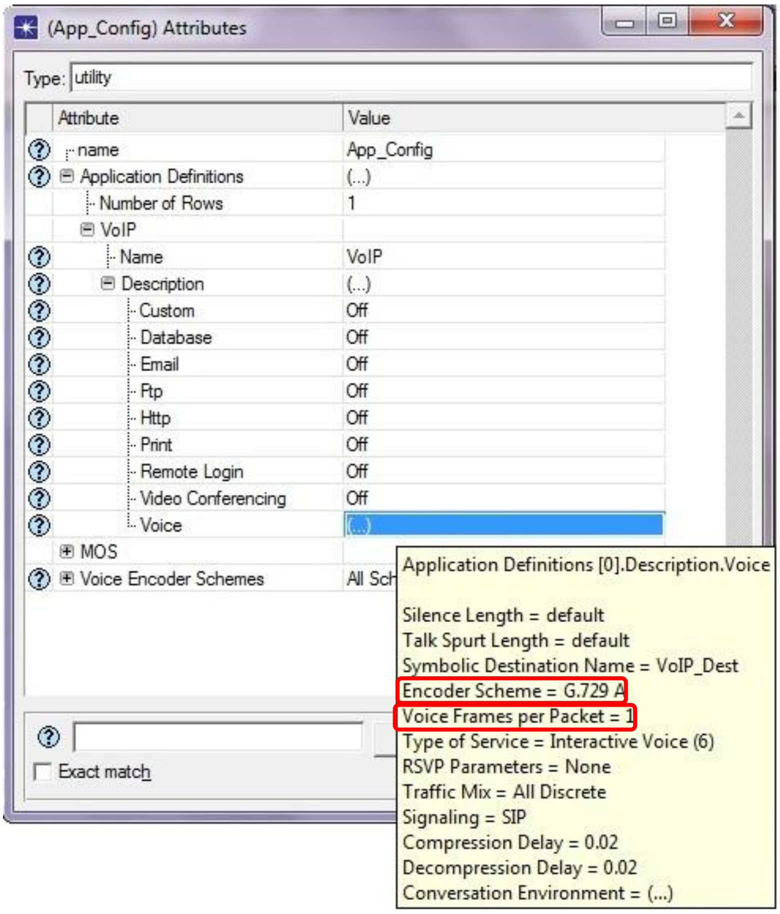



Figure 3.17: Application Configuration in OPNET.

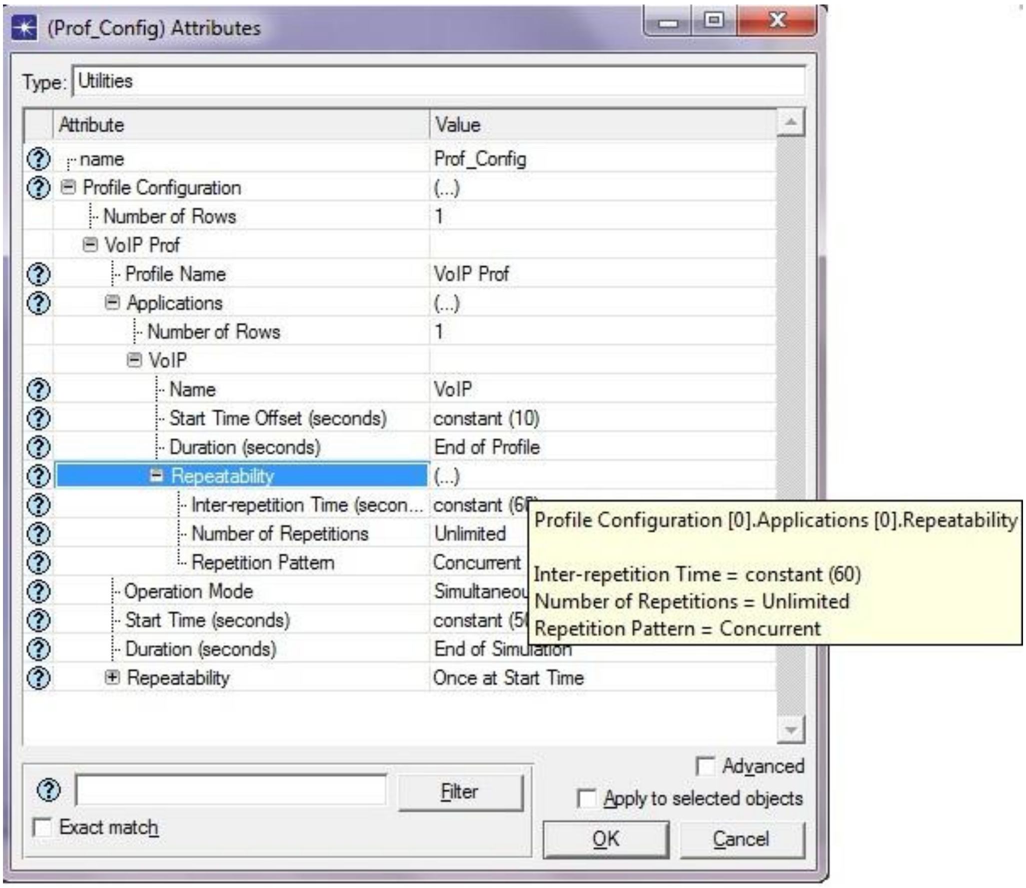



Figure 3.18: Profile configuration in OPNET.

As mentioned earlier, the objective of this part is to calculate network capacity (the maximum number of possible calls) while maintaining the quality at a good level. To achieve this objective, VoIP calls are added to a pair of sender/receiver every minute and key quality parameter is checked after each run to monitor the call's performance. This method is easier to implement instead of adding a pair of sender/receiver repeatedly, although it has a small impact on the queuing delay which is negligible [103].

Based on the result of the simulation when there is a *mismatch* between the traffic sent and received or other quality parameters such as delay is less than 150 ms or MOS decrease below 3.6, it means the number of VoIP calls are more than the network capacity. Hence, upper bound of each transmission rate is estimable based on in this method.

The simulation run time was 20 minutes for all runs according to our profile. VoIP traffic starts after 1 minute from the start time of the simulation and every 1 minute 1 VoIP call is added to the network (maximum 19 calls can be generated). To find the accurate capacity of the network in each transmission rate, as mentioned earlier some indices like the difference between traffic sent and received, delay and MOS should be considered.

In the first scenario the capacity of 11 Mbps using the G.711 codec with 5 frames per packet (fpp)[6] is evaluated. Figure 3.19 shows the mismatch of voice packets sent and received during the simulation. According to the results, after the 12th minute, the sent and received traffic do not trace each other. Since in the profile, one VoIP call was added to the network every minute and the profile started transmission after the first minute, it is concluded that after the 11th call, the capacity of this network is full.

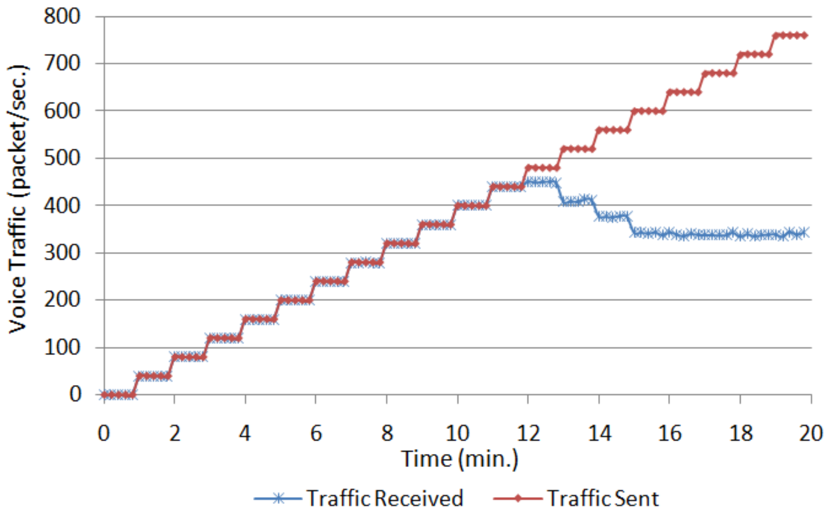



Figure 3.19: Voice traffic sends & receives using G.711codec/5fpp/11Mbps.

For further examination, MOS was also checked to indicate the voice quality of calls. Figure 3.20 shows that after the 12th minute, the quality has degraded sharply which is associated with 11 calls.

[6] Each frame is 10 millisecond.

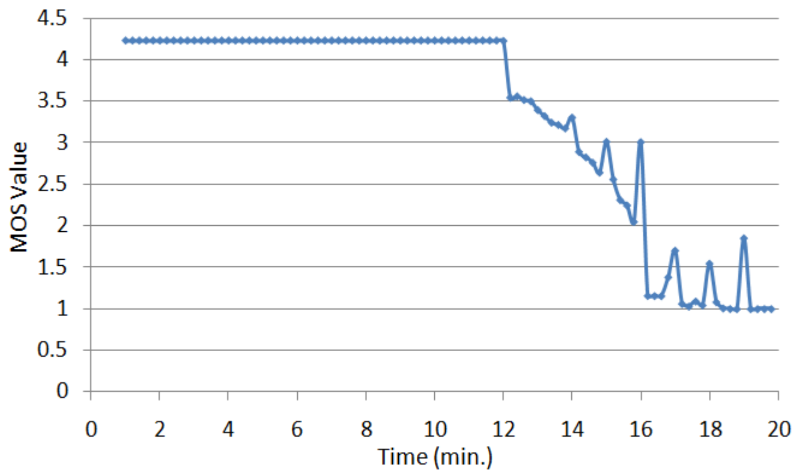



Figure 3.20: MOS level during calls using G.711codec /5fpp/11Mbps.

In addition, end-to-end delay was checked. Figure 3.21 shows when end-to-end delay is within the acceptable range (less than 150 ms) the number of calls is less than 11 (the time is less than 12 min.).

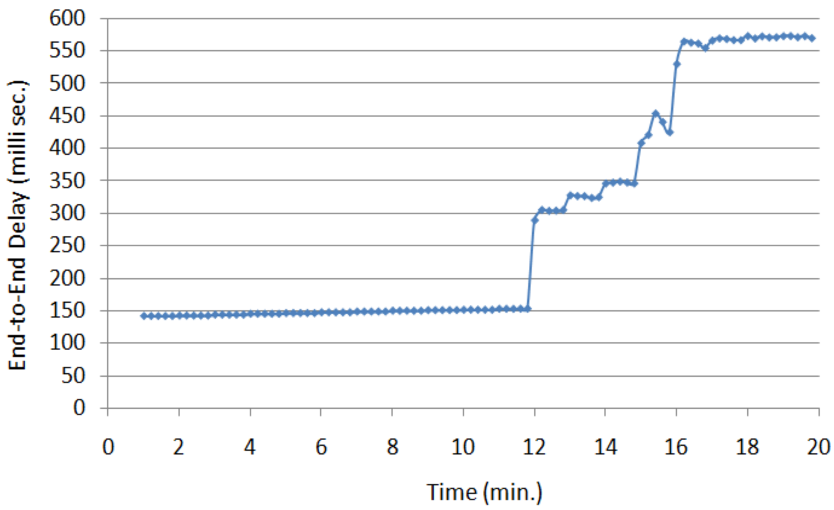



Figure 3.21: End-to-end delay for voice and delay in WLAN using G.711 codec /5fpp/11Mbps.

From last three graphs, it is concluded that the maximum number of possible calls with G.711 codec and 5 frames per packet is 11 calls when the transmission rate of both nodes is 11 Mbps.

In the previous scenario, the method has been presented to find the network capacity based on the G.711 codec with 5fpp in 11Mbps transmission rate. The method is repeated for all IEEE 802.11b transmission rates (11, 5.5, 2 and 1 Mbps) using G.711 and G.729 codec and different number of frames per packet (fpp) from 1 to 5.

Finally, the maximum number of calls where each transmission rate could support were collected and tabulated in Tables 3.6 and 3.7.

Table 3.6: The maximum number of calls for G.729.

| Transmission Rate (Mbps) \ Packet size (frame per packet) | 1 | 2 | 3 | 4 | 5 |
|---|---|---|---|---|---|
| 1 | 1 | 3 | 4 | 6 | 7 |
| 2 | 2 | 4 | 6 | 8 | 10 |
| 5.5 | 2 | 5 | 8 | 11 | 13 |
| 11 | 3 | 6 | 9 | 12 | 15 |

Table 3.7: The maximum number of calls for G.711.

| Transmission Rate (Mbps) \ Packet size (frame per packet) | 1 | 2 | 3 | 4 | 5 |
|---|---|---|---|---|---|
| 1 | 1 | 1 | 2 | 2 | 2 |
| 2 | 1 | 2 | 3 | 4 | 4 |
| 5.5 | 2 | 4 | 6 | 7 | 8 |
| 11 | 2 | 5 | 7 | 9 | 11 |

Next two Figures show the maximum possible number of calls with the acceptable quality for each transmission rate using different number of frame(s) per packet (1 to 5) and G.729 codec (Figure 3.22) and G.711 codec (Figure 3.23).

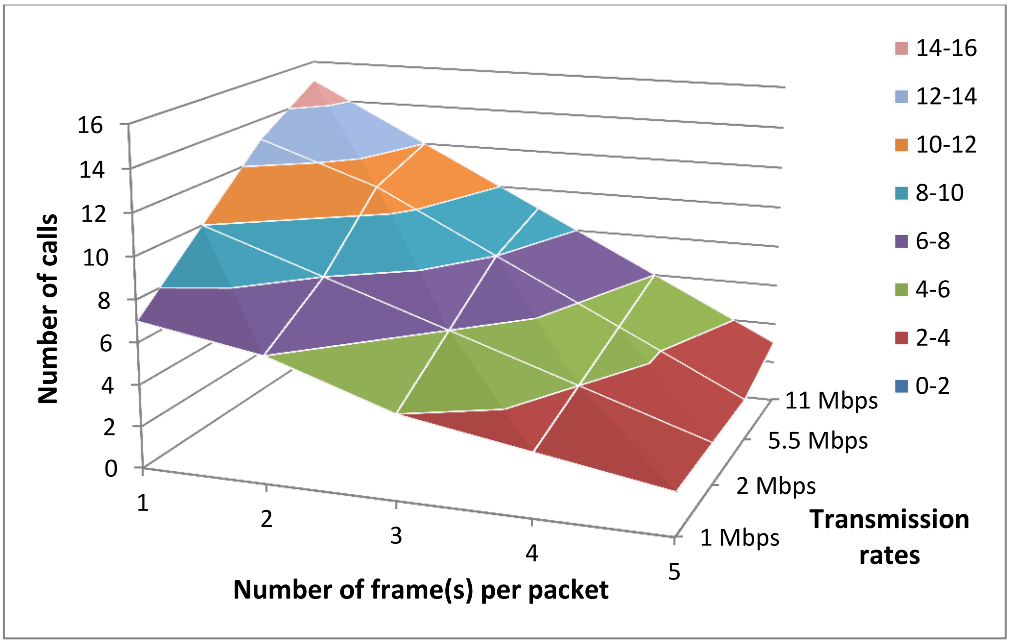



Figure 3.22: Calls capacity in different transmission rate for G.729 codec with different number of frames per packet.

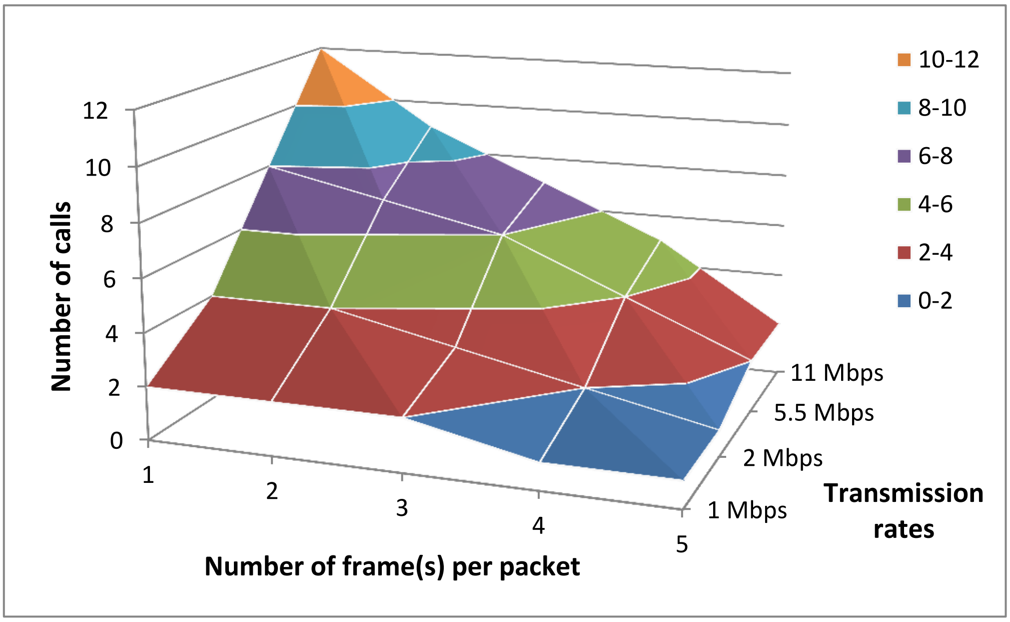


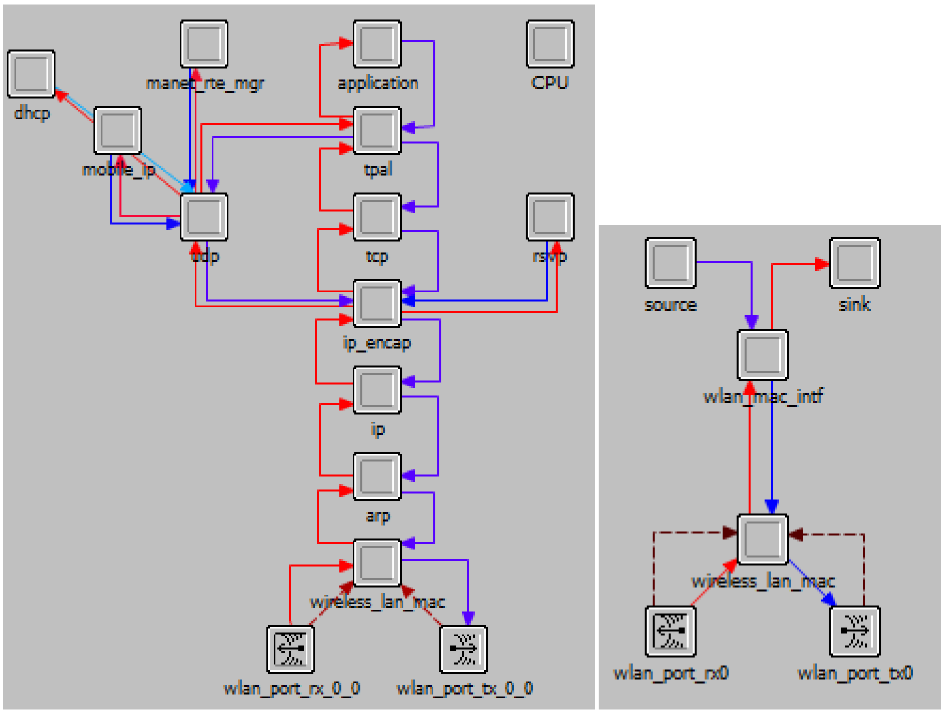

Figure 3.23: Calls capacity in different transmission rate for G.711 codec with different number of frames per packet.

According to the research methodology flowchart in Figure 3.1, after determining the capacity for each transmission rate, the next step is to model a WLAN network with the accurate number of calls to investigate the behavior of quality factors in the congested network. Apparently, in order to make a network congested, it can be through either increasing the number of calls or decreasing the transmission rate. Since in the previous section, maximum number of calls has been found and the model has been developed based on this number of possible calls then by reducing the transmission rate, congestion will be simulated and the behavior of the speech quality parameters will be investigated.

### 3.4 Simulation Model

To find an adaptive rate technique for VoIP in WLANs a stepwise the simulation approach is adopted. However, the simulation model will maintain fixed for the rest of this study. Simulation began with a development of a wireless network in the infrastructure mode, as it is more popular rather than an ad-hoc mode. Infrastructure mode has AP as a central device between senders and receivers, which in VoIP scenarios are named caller and callee. After development of network with the specific numbers of caller and callee, a call is assigned to every pair of caller-callee with the specific codec and packet size (according to the capacity estimation). Then, by including LA, the effect of transmission rate fluctuations in wireless stations in term of call quality is investigated. In the next section, the effect of codec parameter including different coding rates and different packetizations will be verified individually and in combination to find the best adoption procedure for the proposed algorithm.

Wireless nodes in the simulation are based on 802.11b standard and they are placed in the same BSS. The conversation environment is inside a building with UDP as the transport protocol. The nodes were chosen from wireless workstations since wireless station only model MAC and the physical layer but wireless workstations also have an interface which connects the MAC with the higher layers (Figure 3.24). Since wireless workstations support application layer they are used in our model.




wlan_wknstn_adv model wlan_station_adv model

Figure 3.24: Different station models.

Furthermore, SIP (Session Initiation Protocol) is used to manage the call signaling including call setting up, codec negotiation, parameter re-negotiation and call tearing down. Codecs can be established at call setup time using the signaling protocol by SIP INVITE message and it may modify dynamically during the call using SIP re-INVITE message.

Following attributes were configured to use SIP (Figure 3.25):

Application Configuration node: Voice > Signaling

Proxy Server node > SIP UAC Parameters > UAS Service Server > Server Address

Caller- Callee nodes > SIP UAC Parameters > UAC Service > Proxy Server Specification

| | |
|---|---|
| SIP | |
| SIP UAC Parameters | (...) |
| UAC Service | Enabled |
| Proxy Server Specification | (...) |
| Number of Rows | 1 |
| proxy_server | |
| Proxy Server Address | proxy_server |
| Maximum Simultaneous Calls | Unlimited |
| Proxy Server Connect Timeout (sec... | TCP Based |

Figure 3.25: SIP UAC and UAS configuration.

802.11b has been chosen due to its wide installation on network infrastructures. Furthermore, G.711 and G.729 codecs are used for adaptation between codecs in this study since G.711 provides best quality and G.729 provides lower bandwidth consumption among codecs and also due to their universal availability as they are supported by most of the voice gateways, IP phones and VoIP applications. In Figure 3.26 and 3.27 the attributes of each codec used is displayed.

| | |
|---|---|
| PCM | |
| Codec Type | PCM |
| Name | G.711 |
| Frame Size (secs) | 10 msec |
| Lookahead Size (secs) | 0 msec |
| DSP Processing Ratio | 1.0 |
| Coding Rate (bits/sec) | 64 Kbps |
| Speech Activity Detection | Disabled |
| Equipment Impairment Factor (Ie) | 0 |
| Packet Loss Robustness Factor (Bpl) | 4.3 |

Figure 3.26: G.711 codec attributes.

| | |
|---|---|
| CS-ACELP | |
| Codec Type | CS-ACELP |
| Name | G.729 A |
| Frame Size (secs) | 10 msec |
| Lookahead Size (secs) | 5 msec |
| DSP Processing Ratio | 1.0 |
| Coding Rate (bits/sec) | 8 Kbps |
| Speech Activity Detection | Disabled |
| Equipment Impairment Factor (Ie) | 10 |
| Packet Loss Robustness Factor (Bpl) | default |

Figure 3.27: G.729 codec attributes.

It is significant to mention that the reference for providing toll quality[7] encoded speech is the quality of encoded voice by G.711 coder. Therefore, as this codec provides the best perceived speech in un-congested network, the simulation starts with G.711 codec. The lowest delay in the network is associated with smaller packet size or smaller number of frames per packet so the initial number of frames per packet is one frame per packet (each frame equal to 10 ms).

According to table 3.7 with G.711 codec and 1 fpp only two calls can be established in our network model, so 2 pairs of source-destinations (4 stations) have been considered. The topology of our simulation model is shown in Figure 3.28 and two pairs of caller-callee are connected according to Figure 3.29.

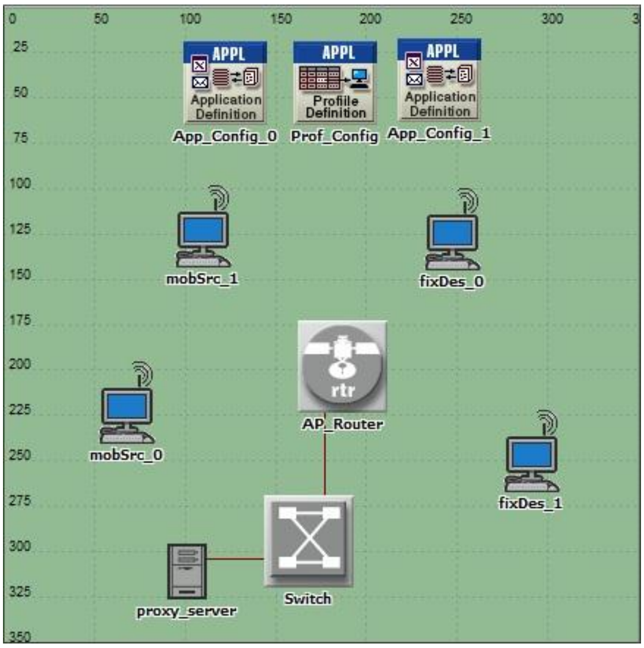



Figure 3.28: Network Topology.

[7] *Toll quality* is a satisfactory quality for end user. This level of quality normally obtain by PSTN networks and many VoIP application and VoIP equipment desire to reach to this quality level.G.711 coder is able to produce toll quality from encoded speech.

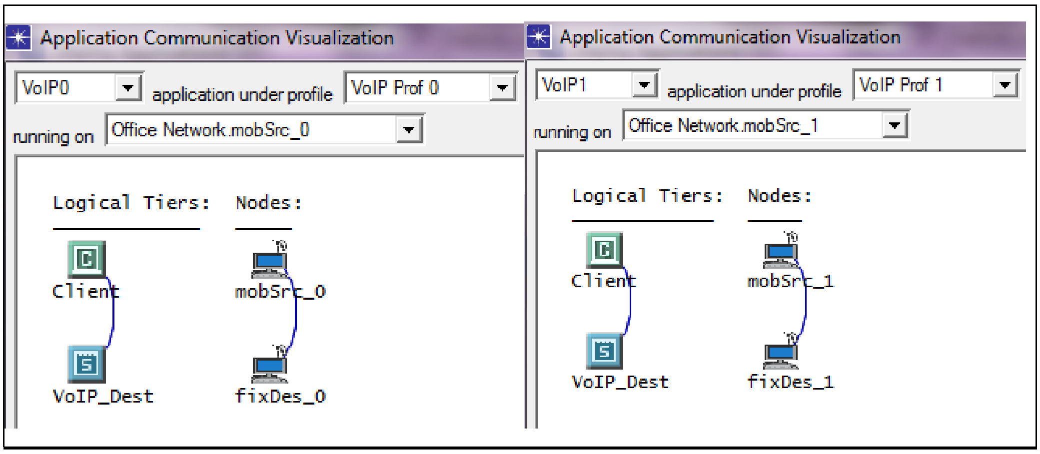



Figure 3.29: Application communication visualization.

After making a physical connection, then application and profile should be defined and assigned to the nodes. The ***application*** contains all the types of applications that can be run in the network models; for example, VoIP, video conference, Email, Http browsing, etc. Once the applications are defined, the ***profile*** should be created to define the user's behavior. Profiles describe the activity patterns of a user or group of users in terms of the applications used over a time period.

Figure 3.30 shows the flow of defining applications and profiles. Figure 3.31 illustrates the application settings that are assigned to the nodes. The profile configurations of these 2 calls are shown in Figure 3.32.

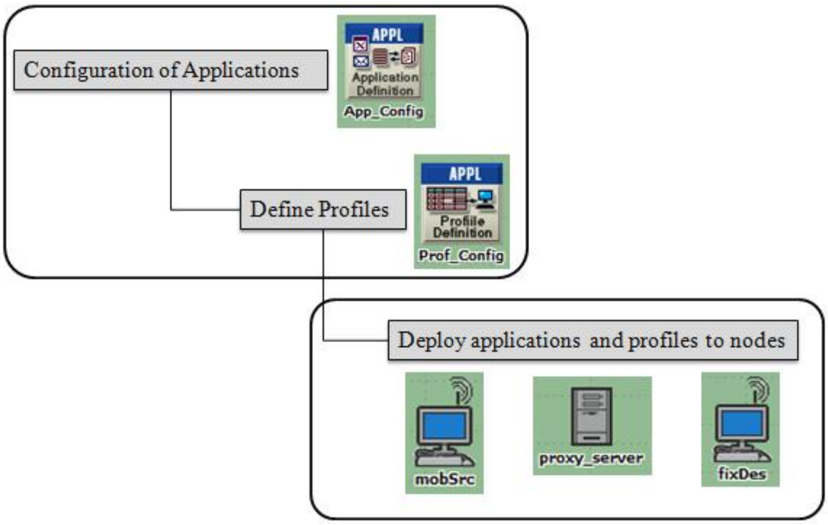



Figure 3.30: Flow of model development (assigning application and profile).

| Attribute | Value |
|---|---|
| Silence Length (seconds) | default |
| Talk Spurt Length (seconds) | default |
| Symbolic Destination Name | VoIP_Dest |
| Encoder Scheme | G.729 A |
| Voice Frames per Packet | 1 |
| Type of Service | 220 |
| RSVP Parameters | None |
| Traffic Mix (%) | All Discrete |
| Signaling | SIP |
| Compression Delay (seconds) | 0.02 |
| Decompression Delay (seconds) | 0.02 |
| Conversation Environment | (...) |

| Attribute | Value |
|---|---|
| Silence Length (seconds) | default |
| Talk Spurt Length (seconds) | default |
| Symbolic Destination Name | VoIP_Dest |
| Encoder Scheme | G.711 |
| Voice Frames per Packet | 1 |
| Type of Service | 220 |
| RSVP Parameters | None |
| Traffic Mix (%) | All Discrete |
| Signaling | SIP |
| Compression Delay (seconds) | 0.02 |
| Decompression Delay (seconds) | 0.02 |
| Conversation Environment | (...) |

Figure 3.31: VoIP application configurations for two main codecs.

| ⊟ VoIP Prof 0 | |
|---|---|
| Profile Name | VoIP Prof 0 |
| ⊞ Applications | (...) 1 |
| Operation Mode | Simultaneous |
| Start Time (seconds) | constant (50) |
| Duration (seconds) | End of Simulation |
| ⊞ Repeatability | Once at Start Time |

| ⊟ VoIP Prof 1 | |
|---|---|
| Profile Name | VoIP Prof 1 |
| ⊞ Applications | (...) 2 |
| Operation Mode | Simultaneous |
| Start Time (seconds) | constant (100) |
| Duration (seconds) | End of Simulation |
| ⊞ Repeatability | Once at Start Time |

* * *

1

| ⊟ VoIP0 | |
|---|---|
| Name | VoIP0 |
| Start Time Offset (seconds) | constant (10) |
| Duration (seconds) | constant (1080) |
| ⊞ Repeatability | Once at Start Time |

2

| ⊟ VoIP1 | |
|---|---|
| Name | VoIP1 |
| Start Time Offset (seconds) | constant (20) |
| Duration (seconds) | constant (1020) |
| ⊞ Repeatability | Once at Start Time |

Figure 3.32: VoIP profile configurations for two nodes.

The relation between simulation timing, profile timing and application timing have been set as shown in Figure 3.33 and the timing configuration of this study is demonstrated in Figure 3.34.

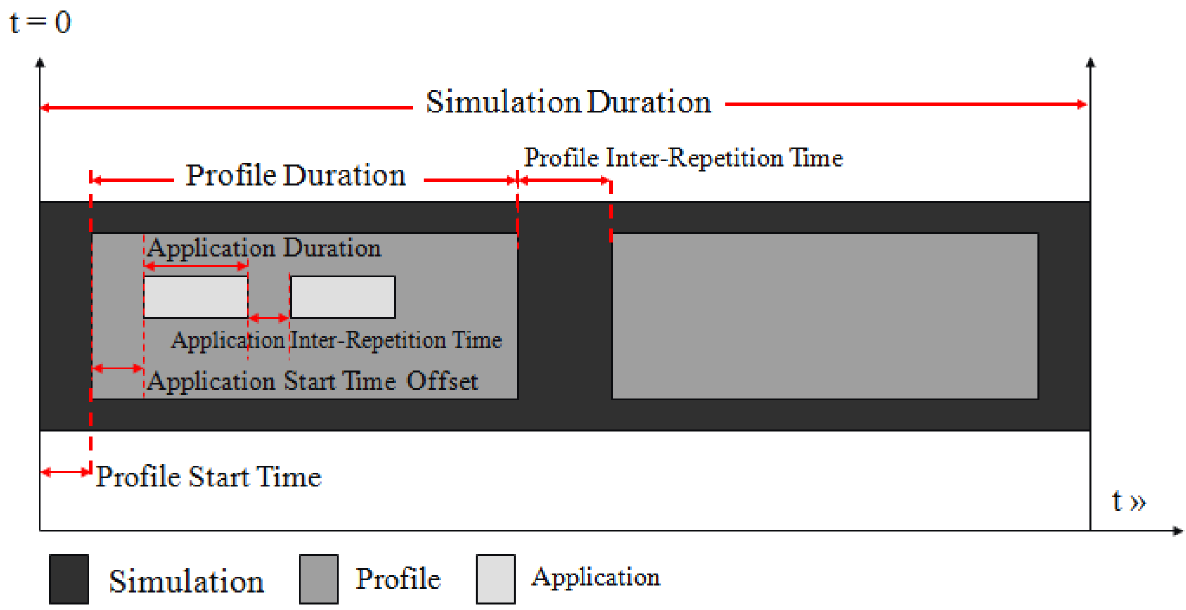



Figure 3.33: Profile timing diagram.

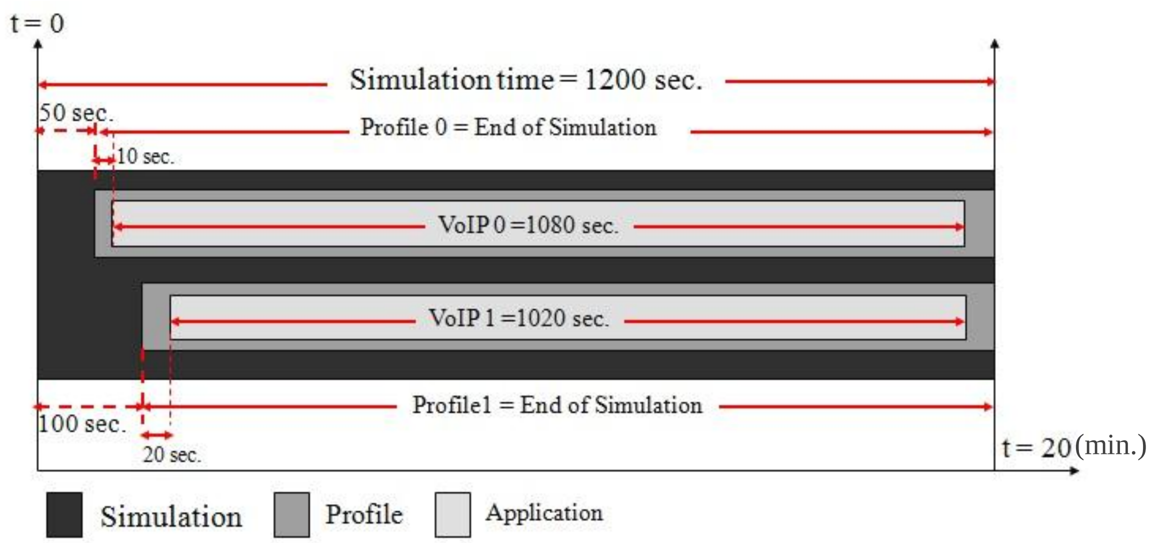



Figure 3.34: Profile timing configuration.

The simulation run time was set to 20 minutes and the profiles start in parallel, with a different offset. In the profiles, VoIP calls were made simultaneously (not serial) with dissimilar offset and calls continuously last almost until to the end of each profile (one minute before profile finish). So, after 120$^{th}$ Sec, network handles two concurrent calls which last up to the 19th minute of simulation time (Figure 3.34).

As shown in Figure 3.30, after the configuration of application and profile (application configuration and timing configuration), the caller and callee should be configured by modifying their attributes according to Figures 3.35 and 3.36

| Applications | |
|---|---|
| Application: ACE Tier Configuration | Unspecified |
| Application: Destination Preferences | (...) |
| Number of Rows | 1 |
| VoIP0 | |
| Application | VoIP0 |
| Symbolic Name | VoIP_Dest |
| Actual Name | (...) |
| Number of Rows | 1 |
| Office Network.fixDes_0 | |
| Name | Office Network.fixDes_0 |
| Selection Weight | 10 |
| Application: Multicasting Specification | None |
| Application: RSVP Parameters | None |
| Application: Segment Size | 64,000 |
| Application: Source Preferences | None |
| Application: Supported Profiles | (...) |
| Number of Rows | 1 |
| VoIP Prof 0 | |
| Profile Name | VoIP Prof 0 |
| Traffic Type | All Discrete |
| Application Delay Tracking | Disabled |
| Application: Supported Services | None |
| Application: Transport Protocol Specifi... | UDP |

Figure 3.35: Source node (caller) configuration.

| Applications | |
|---|---|
| Application: ACE Tier Configuration | Unspecified |
| Application: Destination Preferences | None |
| Application: Multicasting Specification | None |
| Application: RSVP Parameters | None |
| Application: Segment Size | 64,000 |
| Application: Source Preferences | None |
| Application: Supported Profiles | None |
| Application: Supported Services | (...) |
| Application: Transport Protocol Specifi... | UDP |

Figure 3.36: Destination node (callee) configuration.

The rest of the configurations including data rate (transmission rate) assignment, BSS identifier, access point functionality and MAC parameter settings are in the wireless section of the node's attributes. Figure 3.37 shows these setting for one of the nodes.

| | |
|---|---|
| ⊟ Wireless LAN | |
| Wireless LAN MAC Address | Auto Assigned |
| ⊟ Wireless LAN Parameters | (...) |
| BSS Identifier | 1 |
| Access Point Functionality | Disabled |
| Physical Characteristics | Direct Sequence |
| Data Rate (bps) | 11 Mbps |
| ⊞ Channel Settings | Auto Assigned |
| Transmit Power (W) | 0.005 |
| Packet Reception-Power Threshold... | -95 |
| Rts Threshold (bytes) | None |
| Fragmentation Threshold (bytes) | 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) | 256000 |
| Roaming Capability | Disabled |
| Large Packet Processing | Drop |
| ⊞ PCF Parameters | Disabled |
| ⊞ HCF Parameters | Not Supported |

Figure 3.37: Wireless LAN setting in one of the nodes.

As mentioned earlier, there are two possible ways to create congestion; (1) when the capacity is full, adding one call can cause to congestion (2) dropping transmission rate to a lower rate can also cause to congestion. If the capacity of the link is exceeded, the quality drops severely (that is why capacity was studied in section 3.3). Since the purpose of this dissertation is to study multi-rate effect of wireless links on VoIP, the second approach (reducing transmission rate) has been used to create the congestion.

### 3.5 Verifying the Best Quality Factor for Adaptation Instant

In the proposed model that wireless nodes used 802.11b standard, in order to check the effect of transmission rate reduction on the main quality factors, the transmission rate was changed from 11 Mbps to 1 Mbps gradually. Then main quality factors were checked to find the best quality factors that can determine the network congestion faster and more accurate.

In order to assess perceived speech quality, instantaneous quality and/or perceived speech quality can be used [14]. **Instantaneous quality** is measurable by jitter, delay, packet loss and some other quality impairment factors any time during the call. OPNET provides these factors in its statistics (those factors which are also available in RTCP-XR). On the other hand, **Perceived quality** is the perceived quality reported by users in a period during a call. The perceived speech quality can be expressed as E-model or MOS (MOS values may obtain by subjective method or by objective methods as mentioned in section 1.3). Since OPNET provides MOS in its statistics, so MOS is used in this study for perceived speech quality.

In the following, in three different scenarios, the transmission rate of one set of caller-callee had been reduced gradually from 11 Mbps to 1 Mbps and the results of instantaneous and perceived quality have been displayed for each rate.

In the first case that transmission rate is 11 Mbps and all the calls use the G.711 codec and 1 fpp, the general perceived speech quality for the studied model in term of MOS is around 4.3 which is good quality. That is, after the transmission rate for one of the calls drop to 5.5 Mbps, it can be seen that the overall speech quality in the network remains good (Figure 3.38).

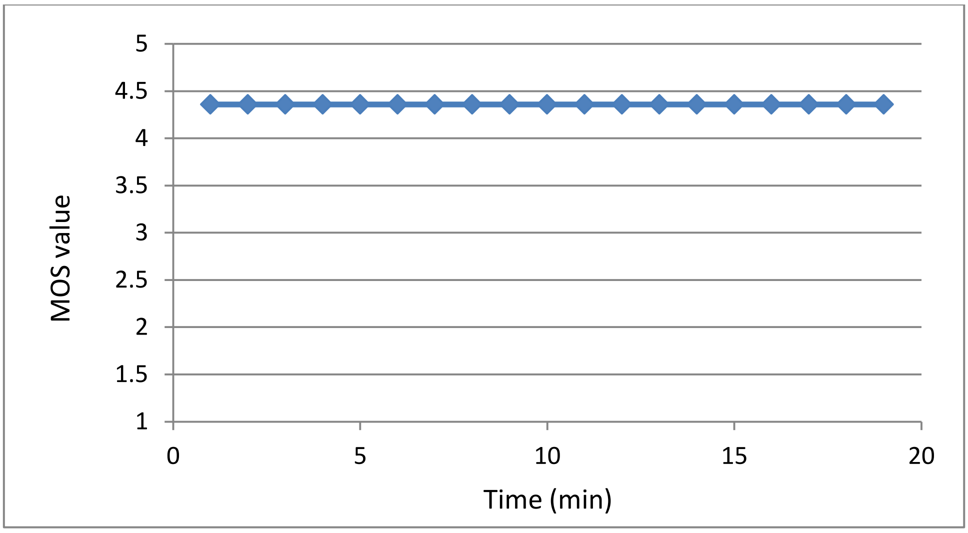



Figure 3.38: MOS result, when transmission rate of one call fall from 11 to 5.5 Mbps.

In addition, reducing the transmission rate to 2 Mbps shows the same perceived quality by MOS (Figure 3.39). So in this specific scenario that network is almost saturated with the 2 calls (G.711 codec, 1 fpp), even two step transmission rate reduction from 11 to 5.5 and then 5.5 to 2, almost does not affect MOS much. It means the good quality can be maintained even with some rate reductions.

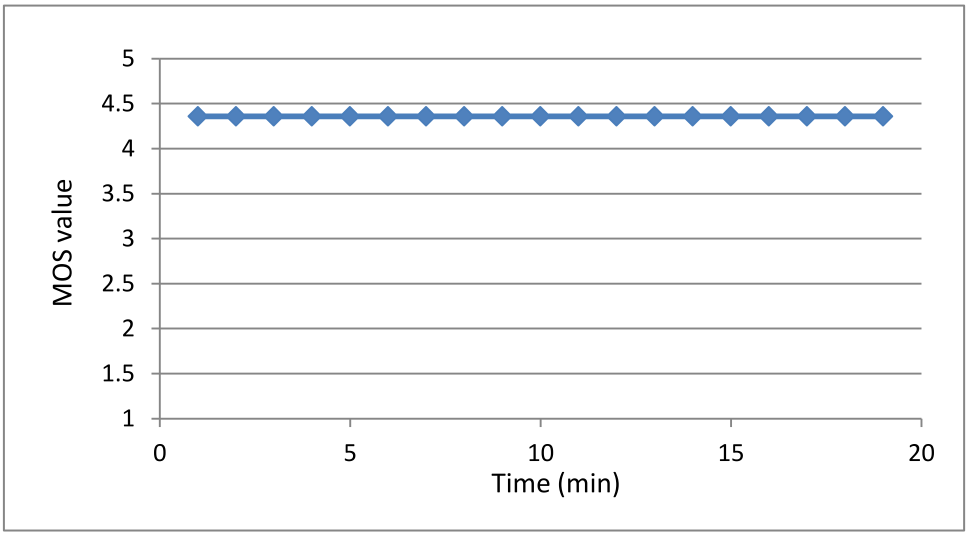



Figure 3.39: MOS result, when transmission rate of one call fall from 5.5 to 2 Mbps.

In the next Figure (Figure 3.40) the result of reducing the transmission rate to 1Mbps is shown. First with the presence of one call (according to our profile configuration), the overall perceived speech quality in term of MOS is 4.3 and then after adding the second call, MOS is sharply decreased to 1.

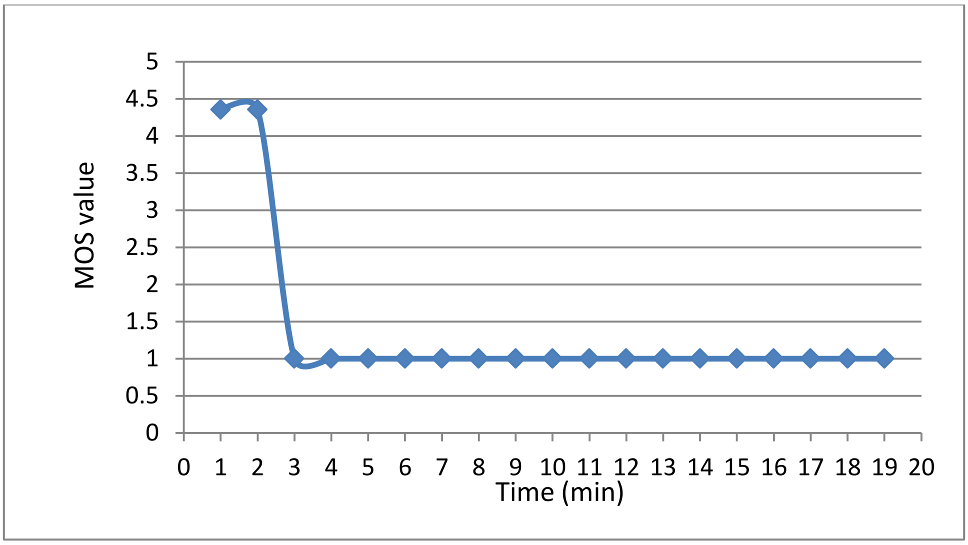



Figure 3.40: MOS result, when transmission rate of one call falls from 2 to 1 Mbps.

Obviously, from the three previous Figures it can be concluded that not all the transmission rate reduction causes severe quality degradation. However, in the critical situations (when the transmission rates of some wireless nodes are reduced remarkably), the overall quality will be affected by transmission rate reduction.

In order to have clearer observation, average MOS for different transmission rates tabulated in Table 3.8 which shows MOS does not change obviously for the three first transmission rate reduction but when the transmission rate drops to 1 Mbps the average MOS is 1.1874 which is very low in comparison with three previous rate reductions and it shows unacceptable quality.

Table 3.8: Average MOS for different transmission rate in one call.

| Transmission rate Reduction in one call | 11 Mbps | 5.5 Mbps | 2 Mbps | 1 Mbps |
|---|---|---|---|---|
| MOS | 4.3584 | 4.3582 | 4.3565 | 1.1874 |

Unlike previous works, that they changed the call parameters (codec and/or packetization) in every rate change, the results of above scenarios show that adaptation is not necessary for all transmission rate reductions. Only the transmission rate reductions that cause congestion needs adaptation and the rest of transmission rate reductions that they do not affect quality to be degraded continue without adaptation. Therefore, some other quality factors should be taken into account beside transmission rate variations to determine the time when adaptation is required.

One of the quality factors that affected quality of calls and it is available in RTCP-XR statistic is the *jitter*. Simply, variation in delay is called "jitter" [104] and it is mainly caused by network congestion. In the other words, the variation in bit arrival times against the regenerated clock at a receiver is jitter. However in IP packets there is no clock to compare the packet arrival times directly, so another way of defining jitter is finding differences in delay extracted from packet time stamps.

IETF in RFC 3393 [104] defines packet jitter as the Instantaneous Packet Delay Variation (IPDV). The IPDV is defined as the difference in *one-way delay* between successive packets. *One-way delay* defines the time duration from the start of packet transmission at the source to the end of the time when packet is received at the destination. If this process always takes equal time, clearly there is no difference in delay so this effect will be cancelled [105].

If the sequence of packets transmission are at times t(1), t(2), t(3), ... t(n) and they receive at the times t'(1), t'(2), t'(3), ... t'(n), then delays *d(i)*s are calculated using this formula:

$$d(i) = t'(i) - t(i) \qquad \text{where } d(i)>=0 \tag{3.7}$$

Consequently, the IPDV or jitter as defined by the IETF, is the sequence of d(2) - d(1), d(3) - d(2), ... d(n) - d(n-1).

In OPNET that is a discrete event simulation tool, the jitter is defined as the time difference between the instances when consecutive packets are received at the destination minus the time difference between the instances when these packets are sent from the source [105], hence the IPDV is:

$$\begin{aligned}&[t'(n) - t'(n-1)] - [t(n) - t(n-1)], \ldots [t'(3) - t'(2)] - [t(3) - t(2)], [t'(2) - t'(1)] - [t(2) - t(1)]\\ &= [t'(n) - t(n)] - [t'(n-1) - t(n-1)], \ldots [t'(3) - t(3)] - [t'(2) - t(2)], [t'(2) - t(2)] - [t'(1) - t(1)]\\ &= d(n) - d(n-1), \ldots d(3) - d(2), d(2) - d(1)\end{aligned} \tag{3.8}$$

In OPNET, the jitter is plotted as the signed maximum jitter over a particular time interval. IETF also defines Packet Delay Variation (PDV) as the difference in one way delay between successive packets. The selection criteria for PDV did not define by IETF but it could be selected packets in a sliding window or the packets which give the maximum and minimum delay in a sequence, thus:

$$PDV= \max\{d(1), d(2), d(3), ... d(n)\} - \min\{d(1), d(2), d(3), ... d(n)\} \quad (3.9)$$

that is

$$PDV= \max\{[t'(1) - t(1)], [t'(2) - t(2)], [t'(3) - t(3)],...[t'(n) - t(n)]\} - \min\{[t'(1) - t(1)], [t'(2) - t(2)], [t'(3) - t(3)], ... [t'(n) - t(n)]\} \quad (3.10)$$

Alternatively in OPNET the PDV is defined as the variance of the delay [105]. For example. Consider the sequences:

t = 1, 2, 3, 4, 5,

t' = 1.45, 2.41, 3.43, 5.44

the packet sent at t= 4 could be ignored as it missed, so:

d = t' - t = 0.45, 0.41, 0.43, 0.44

and so IPDV (jitter) = d(i) - d(i-1) = -0.04, 0.02, 0.01

and the maximum IPDV (signed) = -0.04

Using this selection criteria of packets and with the maximum and minimum delay given,

IPV = 0.45 - 0.41 = 0.04

However using OPNET's criteria of the variability of the delay is:

var (d) = sum(d(i) - u)^2/n

and u  is the mean, u= 0.43

var(d) = [(0.45- 0.43)+ (0.41-0.43)+ (0.43- 0.43) + (0.44- 0.43)] ^2)/4

hence PDV= 0.00025

Figure 3.41 shows the Packet Delay Variation (PVD) when transmission rate in one of the calls is 5.5 Mbps. When one of the call falls from 11 to 5.5 Mbps, PVD is fluctuated especially in the first few minutes (because first and second calls have begun). However, with consideration of unit of PVD on the y-axis, this fluctuation is very small and it changes around 3.45E-8.

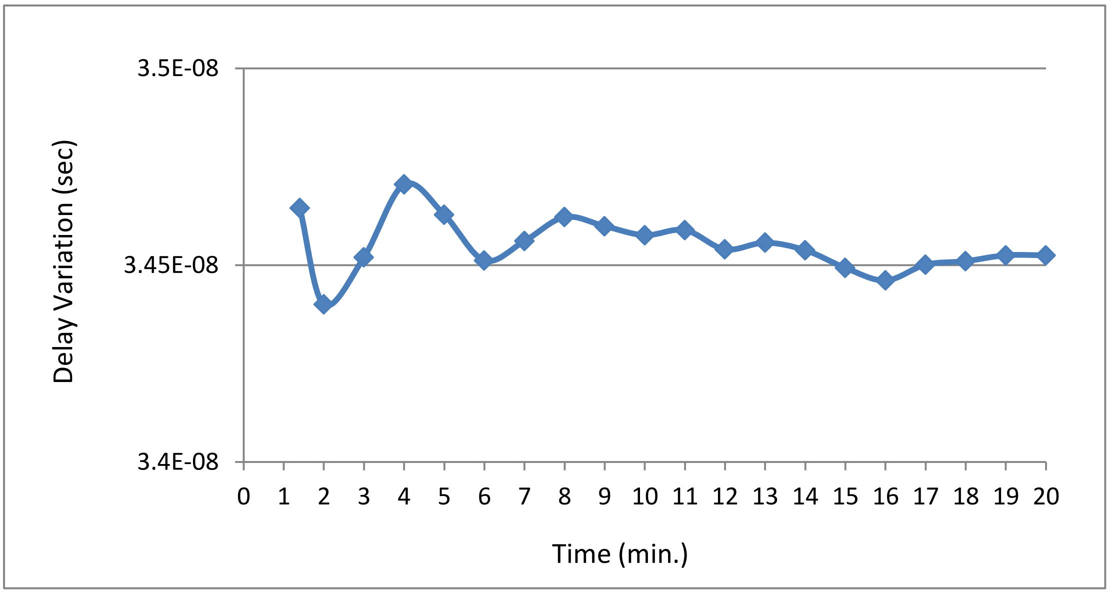



Figure 3.41: Delay variation in the call when transmission rate of caller and callee is reduced from 11Mbps to 5.5 Mbps.

Next Figure (3.42) shows PVD when the transmission rate falls down from 5.5 Mbps to 2 Mbps. In the first few minutes because of two calls establishment the delay variation increases sharply but after that, the PDV remains almost constant. In comparison with the previous graph in Figure 3.41, the difference between mean delay variation values is noticeable. In Figure 3.41, the magnitude of delay variation is 5E-10 whereas in Figure 3.42 is 1E-5, means the delay variation in transmission rate of 2 Mbps is extremely higher than variation in transmission of rate 5.5 Mbps.

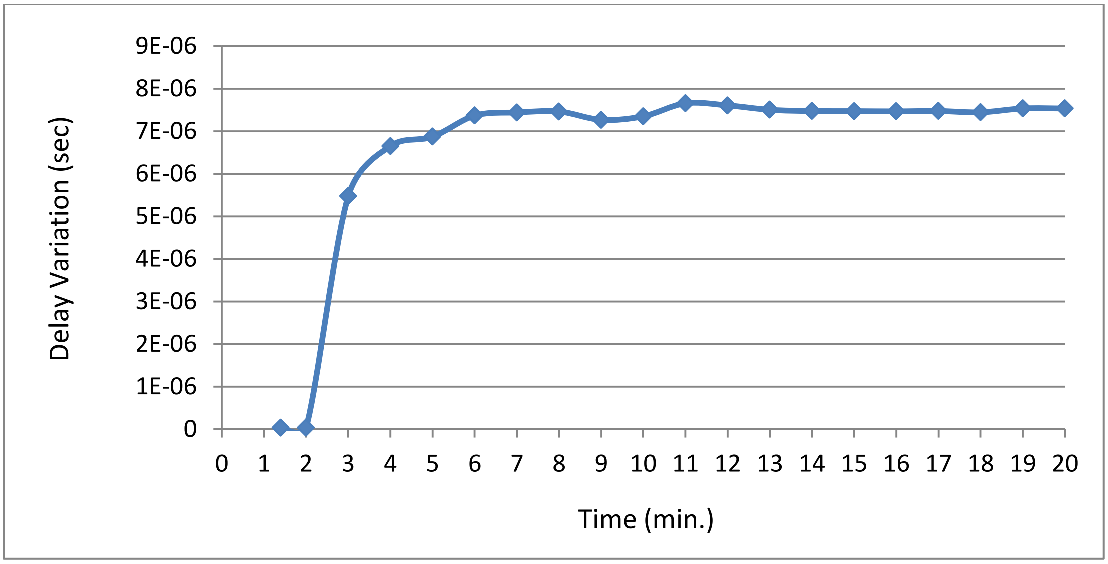



Figure 3.42: Delay variation in the call when transmission rate of caller and callee is reduced to 2 Mbps.

Accordingly, Figure 3.42 shows a huge difference in delay variation when transmission rate is decreased from 5.5 to 2 Mbps. In Figure 3.43 delay variation is demonstrated with the transmission rate reduced from 2 to 1 Mbps.

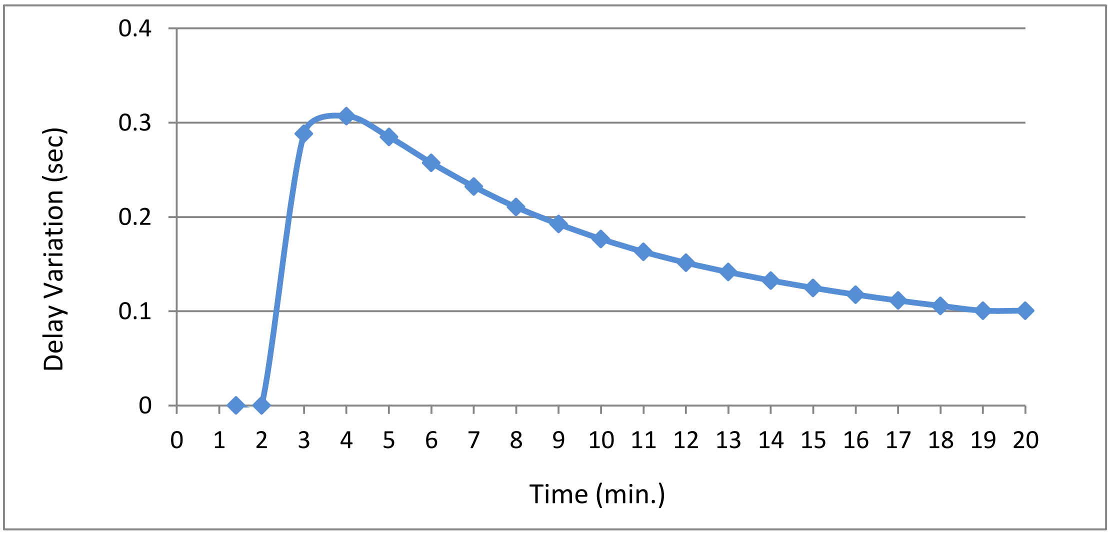



Figure 3.43: Delay variation in the call when transmission rate of caller and callee is reduced to 1 Mbps.

The graph in Figure 3.43 is sharply ascending for the first few minutes because of two calls establishment. But after a while delay variation starts to be descending which is due to small differences between delay of two successive packets in time t' and t.

However changing the graph behavior from ascending to descending does not mean the overall delay decrease in comparison with the previous transmission rate reductions (the Y axis unit is 1.0E-1, in Figure 3.43while it is 1.0 E-6 in Figure 3.42). Since the scales of graphs in last three Figures are very far from each other, illustration of them together in the form of linear graph is not possible. Therefore, in order to compare the mean value of packet delay variation in each transmission rate is tabulated in Table 3.9:

Table 3.9: Mean delay Variation results between caller and callee in one call.

| Transmission rate Reduction in one call | 11 Mbps | 5.5 Mbps | 2 Mbps | 1 Mbps |
|---|---|---|---|---|
| Mean PDV (sec) | 3.68E-08 | 3.46E-08 | 6.88E-06 | 1.71E-01 |

Whereas the overall quality indicated by MOS in Table 3.8 does not show a big difference between changing transmission rate from 5.5 to 2 Mbps, the delay variation (Table 3.9) shows a big difference between the delay variation of each transmission rate in the same scenario. So, packet delay variation would be a good index to indicate transmission rate changes.

Another quality factor gained by RTCP-XR VoIP block (available in the endpoint's report) is "End System Delay" which is the internal round trip delay. The comparison of two continuous rate changes is presented in Figures 3.44, 3.45 and 3.46.

Figure 3.44 shows the end-to-end delay of packets in source, after a round trip when calls used 11 Mbps and then 5.5 Mbps. Clearly, the end-to-end delay for rate 5.5 Mbps is higher than the rate 11 because of the lower transmission rate. However, in both graphs the reason that end-to-end delay drops in the last minute of simulation are that the calls end at the 19th minute of simulation time.

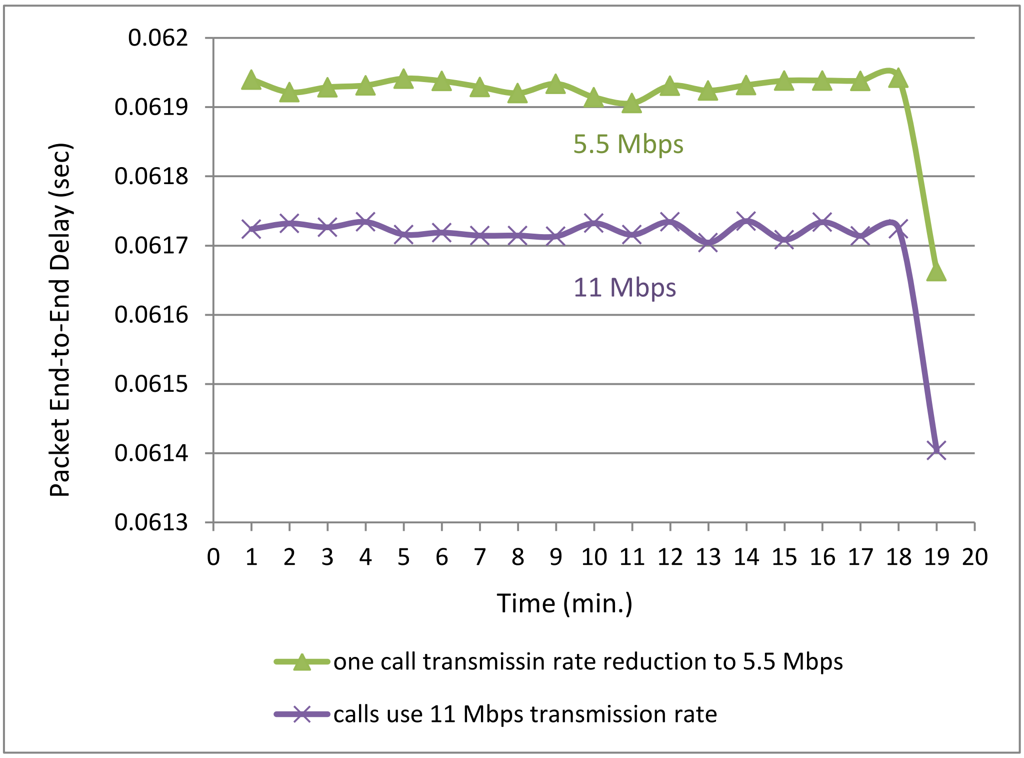



Figure 3.44: End-to-end delay for transmission rates of 11 and 5.5 Mbps in one call.

Likewise, Figure 3.45 shows the end-to-end delay when the transmission rate drops from 5.5 to 2 Mbps. Obviously the end-to-end delay in rate 2 is higher than rate 5.5 Mbps. Note, due to absence of calls in the last minute of simulation graph is descending at the end. Although the graph of end-to-end for transmission rate of 5.5Mbps looks very smooth, but as it had been shown in Figure 3.44 it is not very smooth and due to the scale limit, it looks smooth in comparison with 2 Mbps transmission rate.

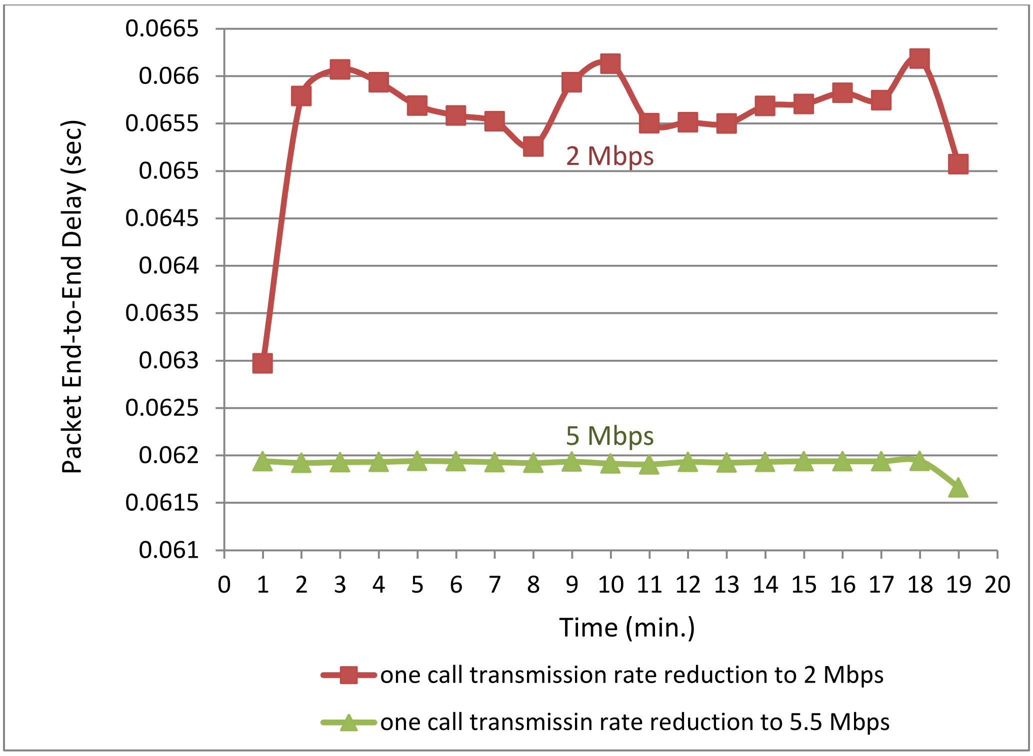


Figure 3.45: End-to-end delay for transmission rates of 5.5 and 2 Mbps in one call.

Figure 3.46 demonstrates end-to-end delay in transmission rate of 1 Mbps is higher in comparison with 2 Mbps. While the graph for 2 Mbps transmission rate also looks linear in Figure 3.46 but as it had been shown in Figure 3.45 it is not linear and it is due to the limited scale of linear presentation. However, the behavior of graph for rate 1 Mbps is sharply ascending in the call establishment time and after that it continues with such a high end-to-end delay. As mentioned in chapter one, the threshold value for end-to-end delay is 0.15 seconded (150 millisecond) but in this scenario when the transmission rate between a pair of caller-callee is dropped to 1 Mbps, the average value of end-to-end delay is 1.11 which is much more than threshold.

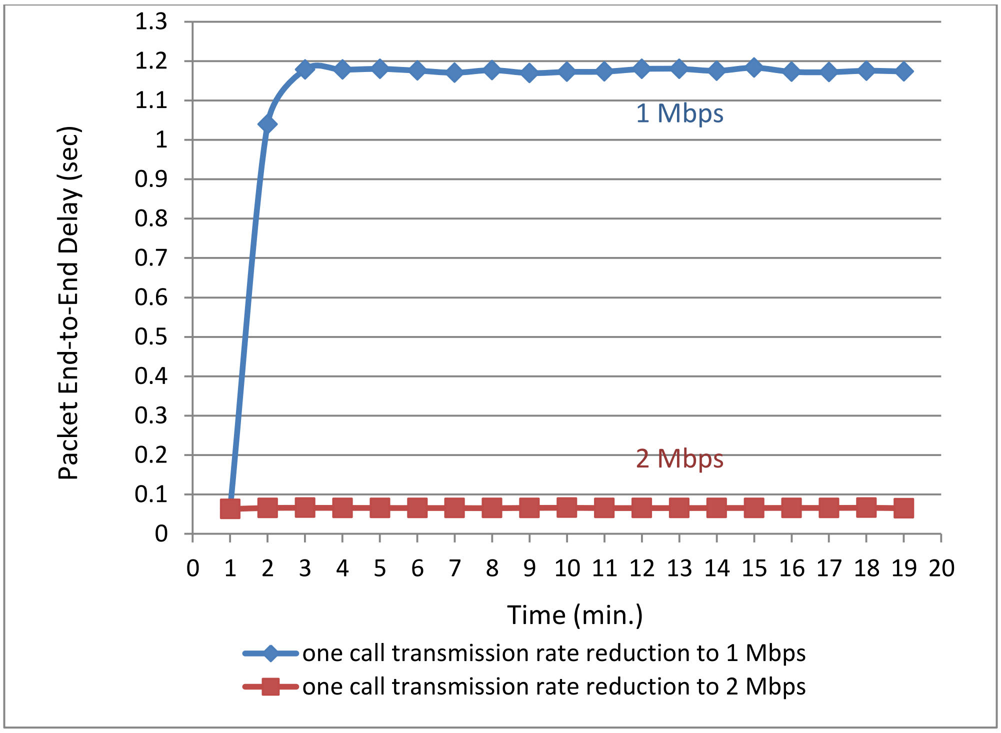



Figure 3.46: Result of end-to-end delay for transmission rates 2 and 1 Mbps in one call.

Table 3.10 tabulated the average of end-to-end delay in each transmission rate. Apparently, in the lower congestion, end-to-end delay is increased gradually, but in severe congestion, it raises very fast. A conclusion from the Table 3.10 is that delay could also be a good index to show the congestion. Note that, the disparity of PDV for different transmission rate is more obvious in comparing to delay so PDV is still the best index of adaptation time.

Table 3.10: Average End-to-End delay for different transmission rate in one call.

| Transmission rate Reduction in one call | 11 Mbps | 5.5 Mbps | 2 Mbps | 1 Mbps |
|---|---|---|---|---|
| Average end-to-end delay (sec) | 0.0617 | 0.0619 | 0.0655 | 1.1137 |

The next two Figures show the packet sent and received graphs for transmission rate of 1 and 2 Mbps. The difference between traffic sent and traffic received is the packet loss rate. Figure 3.47 shows packet loss is almost zero when the transmission rate decreases to 2 Mbps in our simulation scenario.

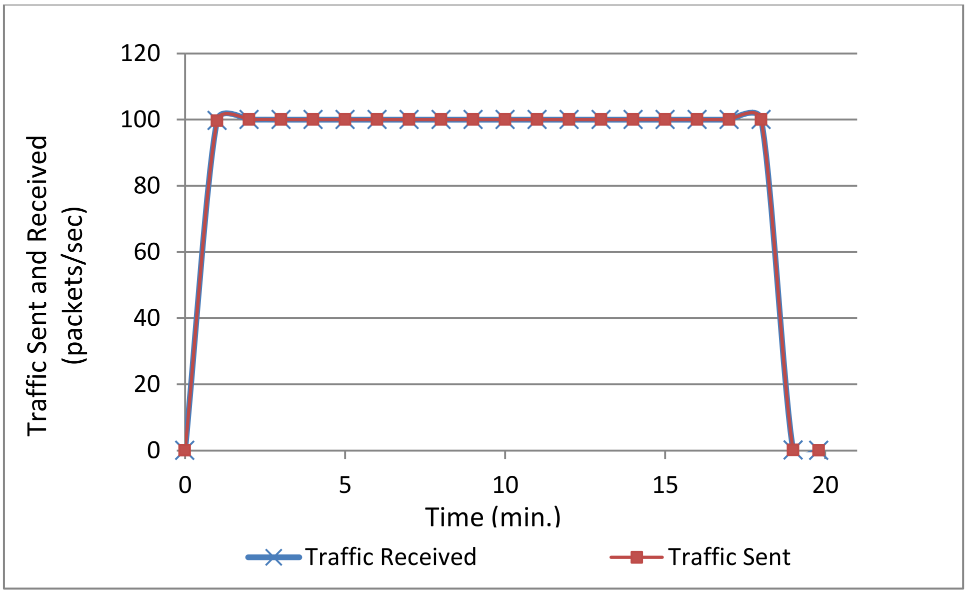



Figure 3.47: Traffic sent & traffic received in the receiver side of the call that is affected by the transmission rate reduction to 2 Mbps.

However, when transmission rate decreases to 1 Mbps (Figure 3.48) packet loss rate increases to almost 40 packets in each point. While transmission rate reduction from 11 to 5.5 and then to 2 Mbps does not cause packet loss in our simulation scenarios and only rate reduction to 1 Mbps shows packet loss. It can be concluded that packet loss is not a good index for transmission rate reduction determination. This conclusion is also according to the previous study's findings.

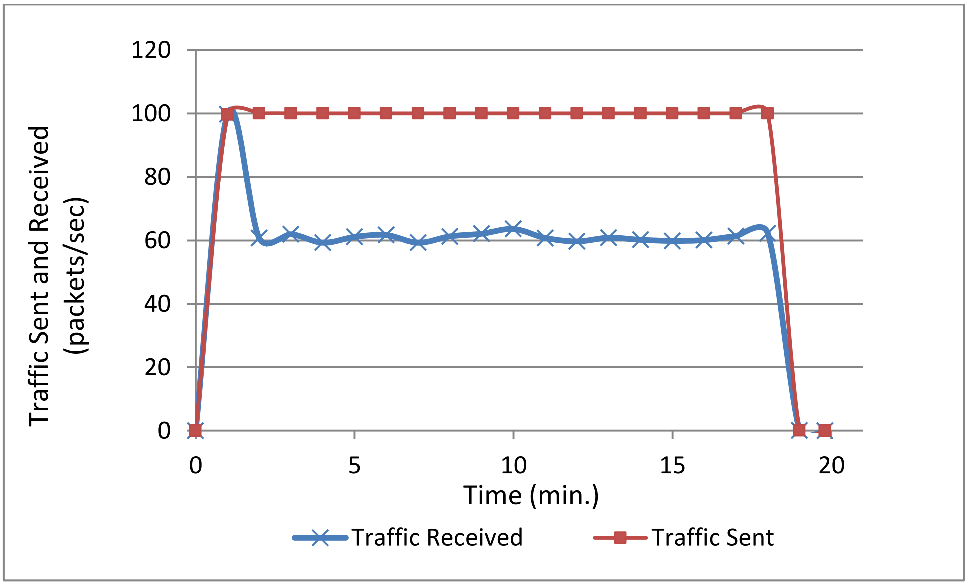



Figure 3.48: Traffic sent & traffic received in the receiver side of the call that is affected by the transmission rate reduction to 1 Mbps.

The results of successive transmission rate reduction on *instant quality factors* namely PDV (jitter), end-to-end delay, and packet loss (Table 3.9-3.10), also on the *overall quality factor* by MOS (Table 3.8) present two important concluded facts:

First, the results of MOS show, some of the rate changes do not affect quality to be degraded obviously, while most of the previous algorithms perform the adaptation process based on the result of MOS. Therefore, our technique checks other instant VoIP quality metrics beside MOS to decide for the right adaptation instance.

Second, Figures 3.41-3.48 show that among the instant quality factors, PDV (jitter) shows the different mean values for transmission rates, and it can differentiate transmission rate of 1 and 2 Mbps from 5.5 and 11 Mbps, while end-to-end delay only differentiates transmission rate of 1Mbps from other transmission rates (11, 5.5 and 2 Mbps). Thus, jitter (PDV) ) is better index to show transmission rate reductions and in practice it can be interpreted by Inter-arrival jitter and extract from RTCP-XR packets.

So, in the proposed method jitter will be monitored during the calls and when the rate drops to the lower rate fast RTCP-XR (every 2 second for minor congestion [84]) will be triggered to check other quality factors. Meanwhile, MOS and delay will check to have an accurate estimation of links status. If all these factors show rate reduction caused congestion then the adaptation phase is commenced.

### 3.6 Verifying the Best Adaptation Process

So far the best adaptation indices have been investigated to determine the right adaptation instance and according to our methodology flowchart (Figure 3.1) hereafter the best adaptation method will be investigated. In fact, adaptation of speech output rate could be done using coding rate adaptation (codec and/or payload size adaptation). As discussed in section 3.2.4.2 the effect of bigger payload size on reducing the link traffic is due to grouping of more speech frames in one packet that share a fix amount of protocol overhead among them. Therefore, the ratio of real data to overhead is higher than using a small frame size with the same overhead.

Consequently due to less protocol overhead that is carried on the network, lower traffic would be expected which will cause lower congestion.

In addition, the packet size adaptation method is preferable because it keeps the codec fixed. Consequently this method can gain the benefit of coding with higher quality (like G.711) and at the same time it can reduce the load of network traffic by switching to bigger payload size.

From another view, packet adaptation has less modification cost (since changing codec in gateways and other middle hardware also paying codec licensing fees are not required). Furthermore, when the transmission rate fluctuation has a small effect on the congestion, codec adaptation changes the traffic output volume massively, thereby in the small congestion, frame size adaptation could be more useful rather than codec adaptation.

All the above reasons lead us to use payload size adaptation first but if the congestion was high and the quality factors could not meet their acceptable range, codec adaptation would be commenced to rectify the congestion. Henceforth simulation results will be shown that in most of the cases, frame size adaptation is enough to recover the network from low to moderate congestion and they also show codec adaptation helps the network to recover from higher congestion.

The first simulation scenario was conducted with 3 stations having transmission rate fallen to 1 Mbps and 1 of the stations has 11 Mbps transmission rate. In this scenario Frame-Adaptive, Codec-Adaptive, and None-Adaptive methods were compared. The results show, where the None-Adaptive method failed to provide good quality of service, adaptive methods act much better. Furthermore, the differences between the both adaptive methods namely (Frame-Adaptive, Codec-Adaptive) are discussed. In this simulation scenario, packet size adaptation is done by switching from 1 fpp to 2 fpp and codec adaptation has been done by switching from G.711 to G.729.

As Figure 3.49 presents, the average MOS for frame-adaptive, codec-adaptive and none-adaptive graphs are compared. This Figure shows that the frame size adaptation method has the highest MOS and followed by codec adaptation. The reason is that, frame-adaptive method keeps the codec G.711 which has the highest MOS among all codecs, while Codec-adaptive uses G.729 codec which has lower MOS. Although None-Adaptive also uses G.711 but since it uses 1 frame per packet, it has higher protocol overhead (almost 2 times more redundant data in comparing to 2 frames per packet), thus it results in more traffic and more congestion causing MOS degradation.

Furthermore, codec-adaptive method which uses G.729 instead of G.711 also reduces the congestion by having lower bitrate (G.729 bit-rate is 8 times lower than the G.711) and lower load on the bandwidth which consequently reduces the congestion and causes higher MOS in comparison to none adaptive method. Nevertheless, G.729 has lower quality (MOS 4) in comparison to G.711 (MOS) 4.3 due to its lower bit-rate.

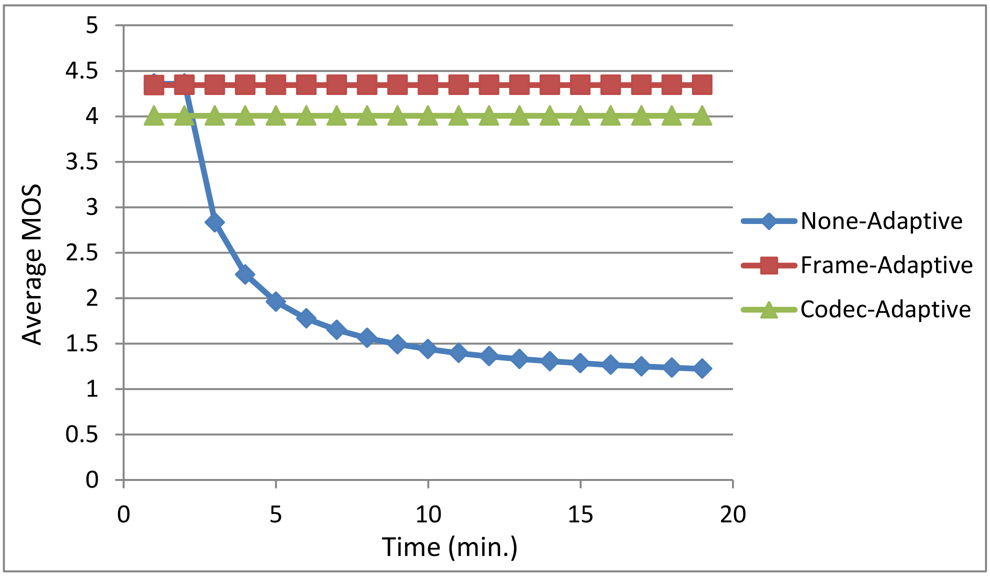



Figure 3.49: MOS of different adaptation methods when 3 stations have fallen to 1 Mbps transmission rate and 1 of the stations has 11 Mbps transmission rate.

The load on the WLAN is shown in Figure 3.50. As mentioned earlier, in this scenario, frame-adaptive approach uses 2 frames per packet therefore overhead is shared for higher amount of data in comparison to the none adaptive approach which uses 1 frame per packet. That is why frame-adaptive has lower load in comparison to the none adaptive method.

Furthermore, codec-adaptive approach uses G.729 that has transmission rate of 8Kbps and here none adaptive approach and frame-adaptive approach uses the default codec (which is G.711 that has transmission rate of 64 Kbps). Since the coding rate of G.729 is 8 times lower than G.711, the load for codec-adaptive approach is massively lower than none adaptive and it is also lower than the frame-adaptive method. This comparison shows codec-adaptive methods are more effective to reduce the traffic load, although they are more costly compared to frame-adaptive method.

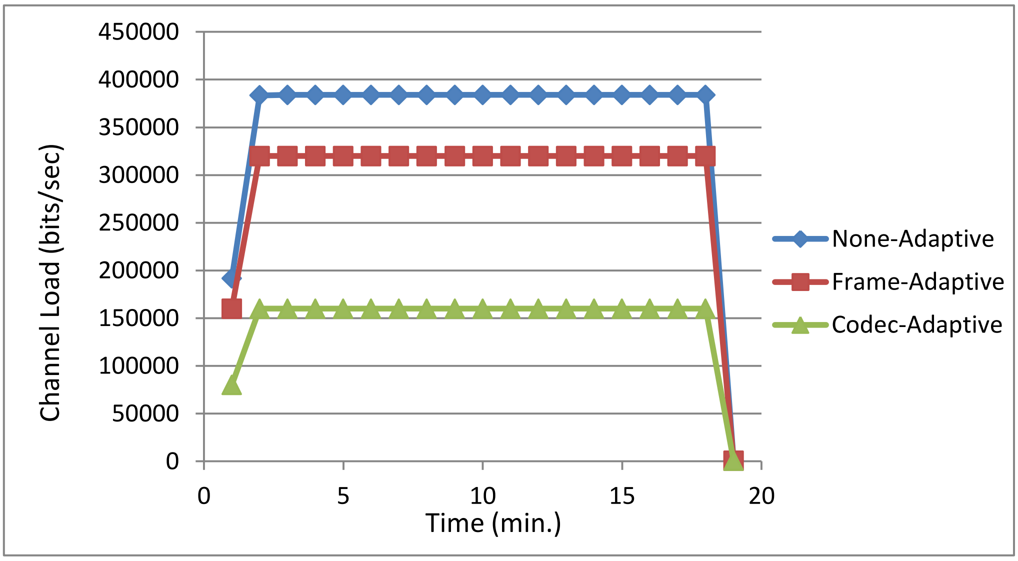



Figure 3.50: Channel load of different adaptation methods when 3 stations have fallen to transmission rate of 1 Mbps and 1 of the stations has 11 Mbps transmission rate.

Another comparison between the three graphs for the same simulation scenario is shown in Figure 3.51 in terms of End-to-End delay. As the acceptable range of delay is less than 150ms, this Figure demonstrates that both frame-adaptive and codec-adaptive methods afford the acceptable delay, while delay for frame-adaptive graph is a little bit higher than codec-adaptive. As codec-adaptive method uses only 1 frame per packet and frame-adaptive uses 2 frames per packet which needs more packetization time, thus results in more End-to-End delay. In addition, since the bit-rate of G.711 is more than G.729, so it has more traffic loads causing more transmission delay which effects on higher end-to-end delay. It should be mentioned that none adaptive graph in Figure 3.51 starts from almost 0.1 which is due to having only one call at that moment of simulation time.

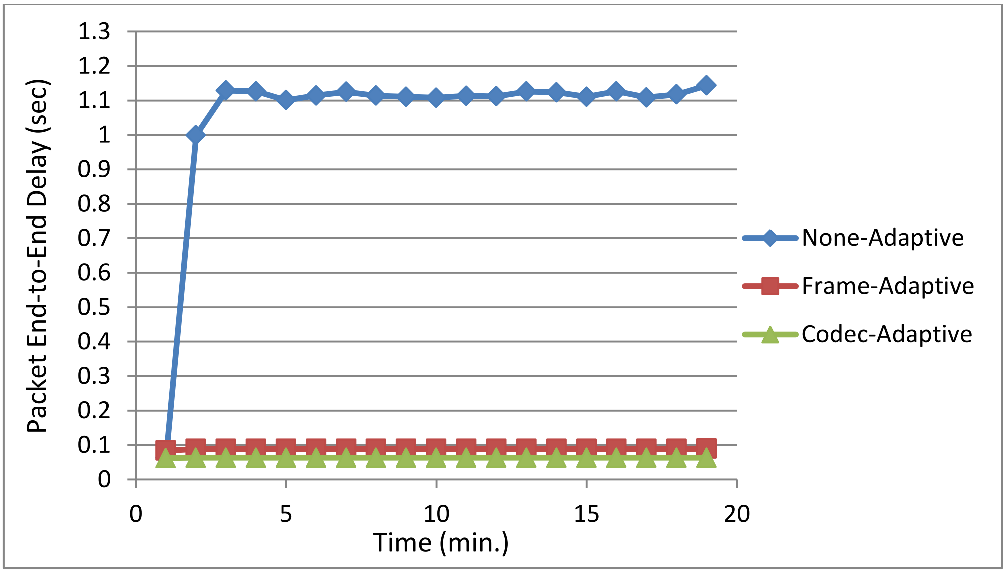



Figure 3.51: Delay of different adaptation methods when 3 stations have fallen to 1 Mbps transmission rate and 1 of the stations has 11 Mbps transmission rate.

Figure 3.52 shows very tiny data loss for both adaptive methods against non-adaptive method. Here also having data loss almost zero at the start and end of none adaptive graph are because of having only one call at the start and having no call at the last minute of simulation time.

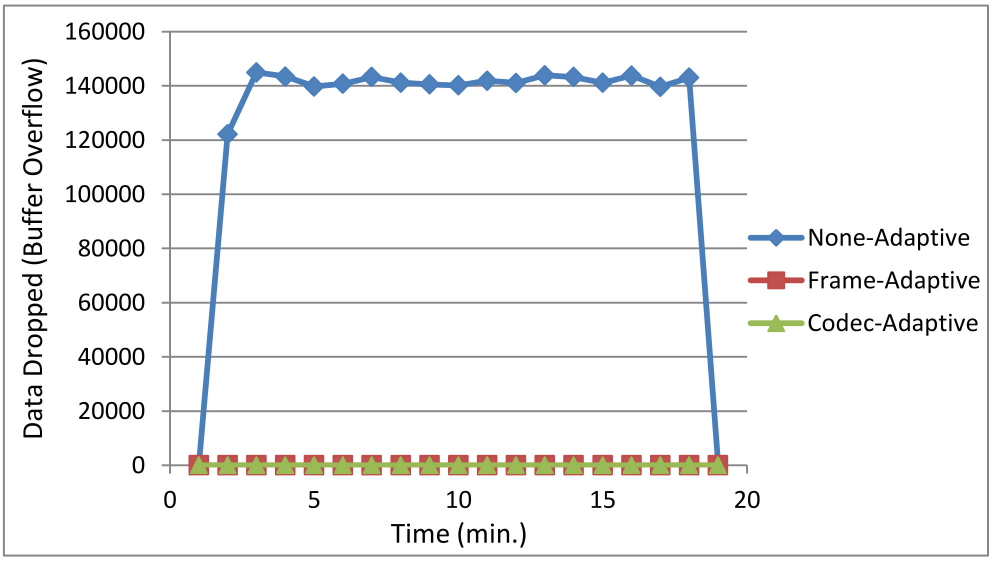



Figure 3.52: Data drop due to buffer over-flow for different adaptation methods when 3 stations have fallen to transmission rate of 1 Mbps and 1 of the stations has 11 Mbps transmission rate.

From the results obtained, it is concluded that frame-adaptive and codec-adaptive systems act better than none-adaptive systems. Meanwhile, it is concluded that frame-adaptive method can rectify low congestions but as codec-adaptive method has lower load compared to frame-adaptive method it would be better to use this method for higher congestion.

Now in order to clarify how many frame adaptation steps are needed before the codec adaptation, *previous scenario has been repeated*, except that none adaptive graph has been removed, because the main concern is to compare different frame adaptation steps. The outcomes are shown in Figures 3.53 to 3.56.

In this scenario first level of packet size adaptation is done by switching from 1 fpp to 2 fpp and then the second level of packet size adaptation has been done by switching from 2 fpp to 3 fpp and the third level is by switching from 3 to 4 fpp. Furthermore, a comparison is done between these different levels of packet size adaptation and codec adaptation (switching from G.711 to G.729).

As Figure 3.53 illustrates, there is a difference in quality level of frame-adaptive methods and codec adaptive method. It is because all frame adaptation levels use G.711 in our scenario, so their speech quality is approximately MOS 4.3. While the codec-adaptive method used G.729 in the same scenario the MOS is almost 4. Since both methods produce good quality of service the MOS are illustrated from 3.8 to 4.4 for clearer observation.

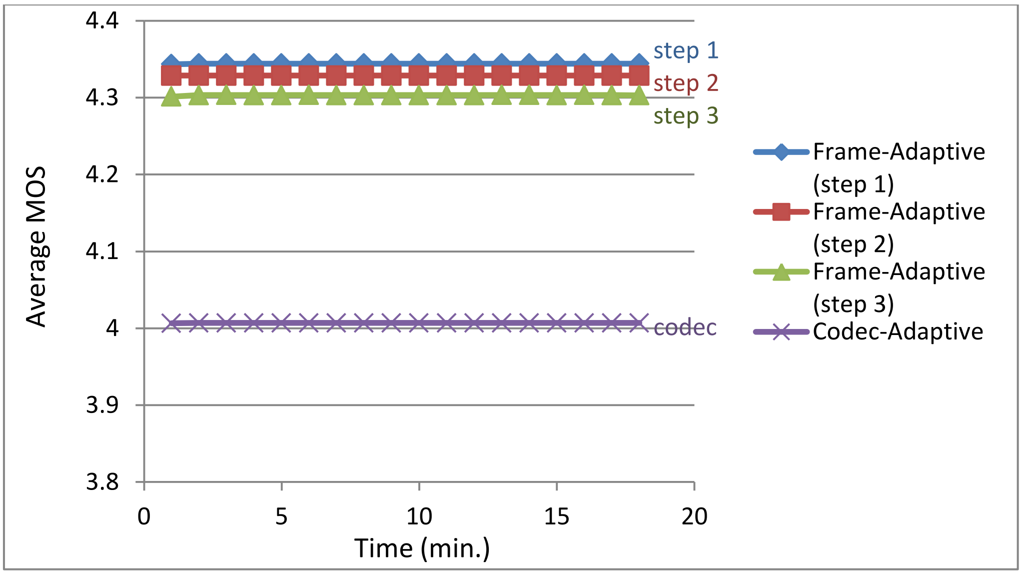



Figure 3.53: Comparison of different level of frame size adaptation methods and codec adaptation method in term of MOS.

As Figure 3.54 shows, due to having less overhead for the packets with 4 frames compare to 3 the channel load of 4fpp is less than 3fpp and with the same concept 3fpp has lower load than 2fpp. On the other hand, G.729 with having 8 Kbps bit-rate has less load than frame size approaches that used G.711 with 64 Kbps.

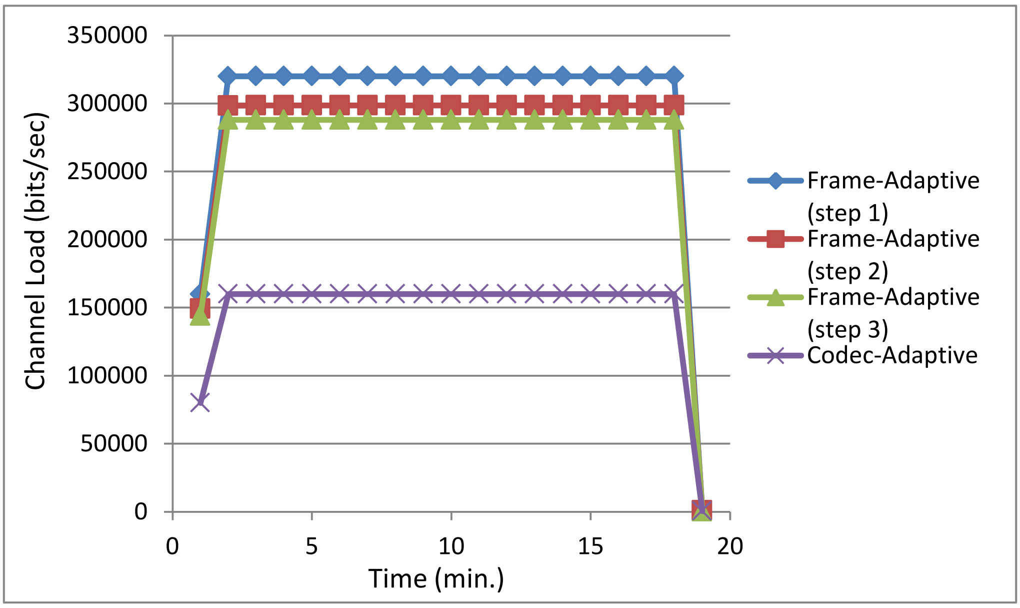



Figure 3.54: Comparison of different level of frame size adaptation methods and codec adaptation method in term of Channel load.

According to Figure 3.14, increasing number of frames per packet decreases the amount of overhead to data ratio, so it should cause to lower traffic in the network. Consequently, due to lower traffic in the network, packets are supposed to be delivered earlier at the receiver side. Therefore, in Figure 3.55, the step 2 frame-adaptive graph which used 2fpp supposed to have lower end-to-end delay comparing to step 1 which used 1 fpp. However, since the congestion in this scenario is not high the effect of overhead sharing is less obvious against packetization delay. In this scenario packetization delay (the time taken by the packetizer to pack the frames as a packet) has a more obvious effect on end-to-end delay.

As Figure 3.55 illustrates, among three levels of frame adaptation, third level (3 fpp to 4) has the highest end-to-end delay that is due to longer time which is taken by the packetizer. The same reason goes to the difference in end-to-end delay value of other levels of adaptation.

From the other side, codec adaptation uses G.729 codec instead of G.711 but with only 1 frame per packet which has the lowest packetization delay. Furthermore, G.729 codec has 8 times less output traffic comparing to G.711 which noticeably effect on the lower bandwidth consumption and the lower congestion and lower end-to-end delay.

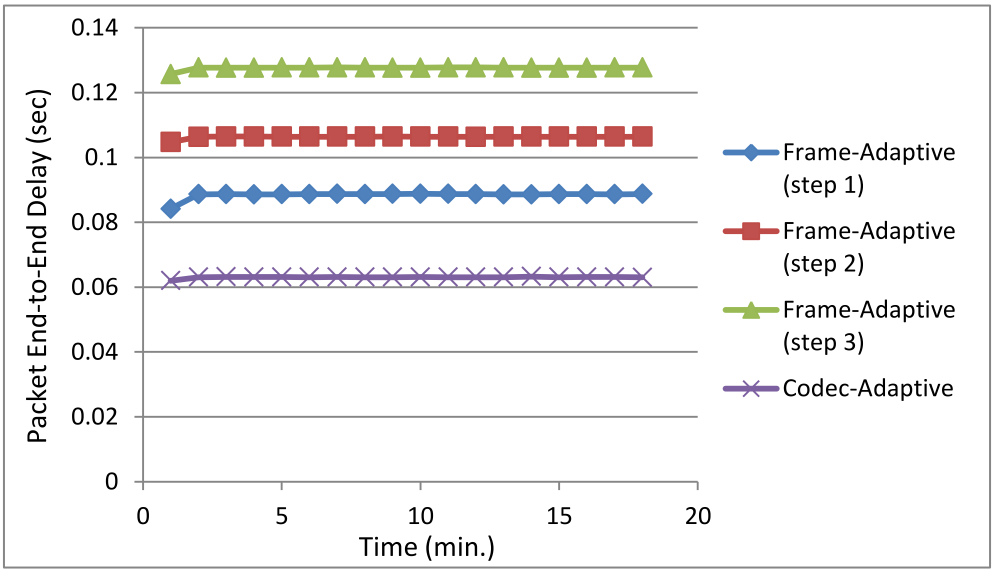



Figure 3.55: Comparison of different level of frame size adaptation methods and codec adaptation method in term of delay.

It should be mentioned here that, step 4 of frame-adaptive method has not come into further consideration which is due to the high end-to-end delay (very close to the upper limit of acceptable range) and it would create a critical situation for the adaptation algorithm later (Figure 3.56).

Furthermore, the 4th step of frame size adaptation switches to 5 fpp and each frame is 10 ms which means the packet size is 50ms. However, most of the commercial implementation does not support it. Therefore, only 3 steps of frame size adaptation are considered for the proposed algorithm. Figure 3.57 shows, all these adaptive mechanism can eliminate data drop in the AP.

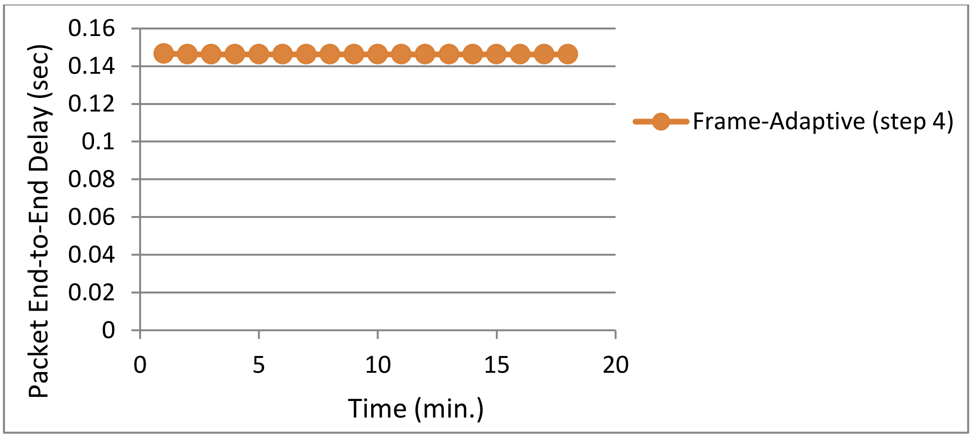



Figure 3.56: End-to-End delay of step 4 adaptation (4fpp).

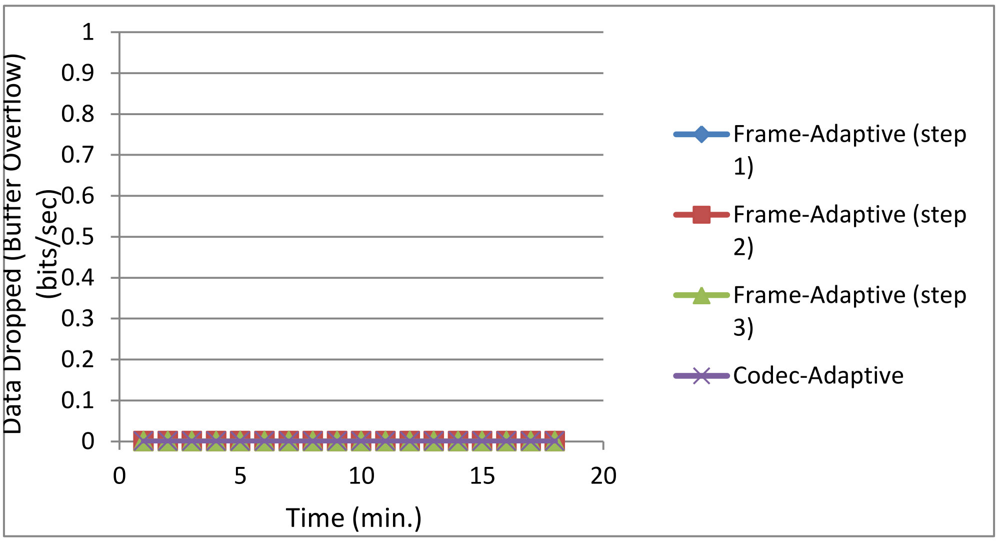



Figure 3.57: Comparison of different level of frame size adaptation methods and codec adaptation method in term of data drop.

The results of this scenario show where non-adaptive methods fail to provide good quality of service codec and frame adaptive methods are the best solution to rectify the failure of none adaptive methods. In addition, from the results obtained, the proposed mechanism can increase the number of frames up to three steps and check the quality factors after each of them, if the quality turns into the acceptable range. This means the system does not need to undergo the codec adaptation but if frame size adaptation does not rectify the congestion then codec adaptation would be commenced.

The next scenario was conducted with *all 4 stations fall to 1 Mbps* transmission rate (consequently the congestion is high). In different steps frame adaptation, codec adaptation also both approaches together have been simulated to compare their results with the none adaptation method.

In this scenario, in the none-adaptive method, calls used G.711 codec with 1 frame per packet (fpp), in frame-adaptive method calls use G.711 codec and number of frames increased to 2 fpp, in codec-adaptive method, codec is changed to G.729 with 1 fpp, and in the frame & codec adaptation method codec is changed to G.729 and number of frames is increased to 2 fpp.

The result of average MOS is shown in Figure 3.58. Where none adaptive method fails to provide good MOS (after adding the second call), frame-adaptive and codec-adaptive also failed to provide acceptable voice quality, frame and codec-adaptive methods can maintain on a good quality.

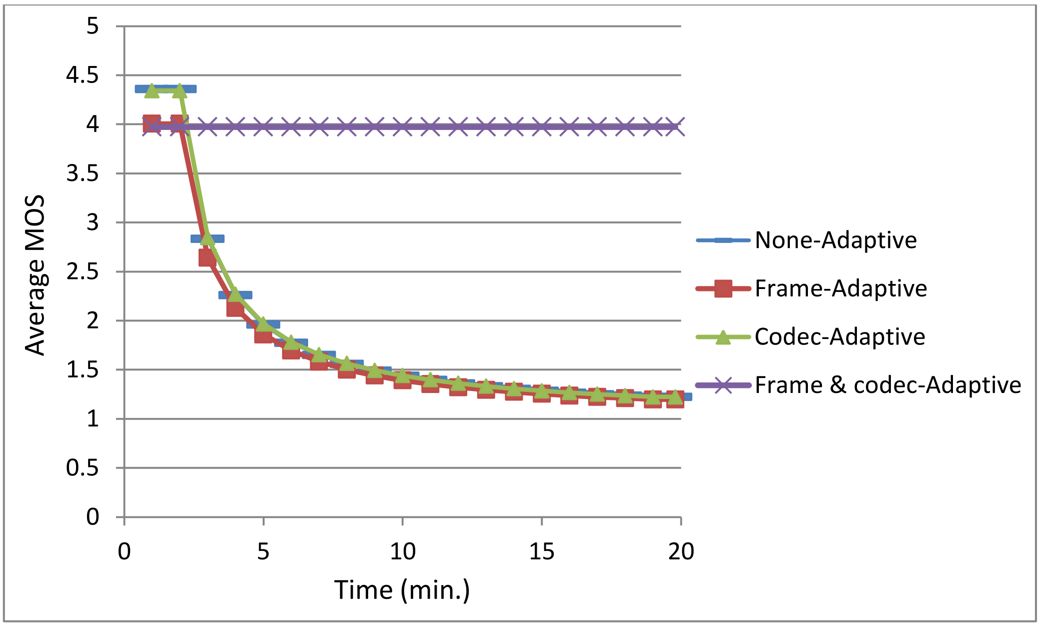



Figure 3.58: MOS of different adaptation methods when all 4 stations have fallen to 1 Mbps transmission rate.

Codec-adaptive approach uses G.729 that has lower output rate compared to frame-adaptive approach that use codec G.711. In comparison between codec-adaptive method and 'frame & codec' adaptive method; while both methods use of them G.729 codec, the second method uses larger packet size (2 frame per packet) which that has lower overhead and consequently it has lower load (Figure 3.59).

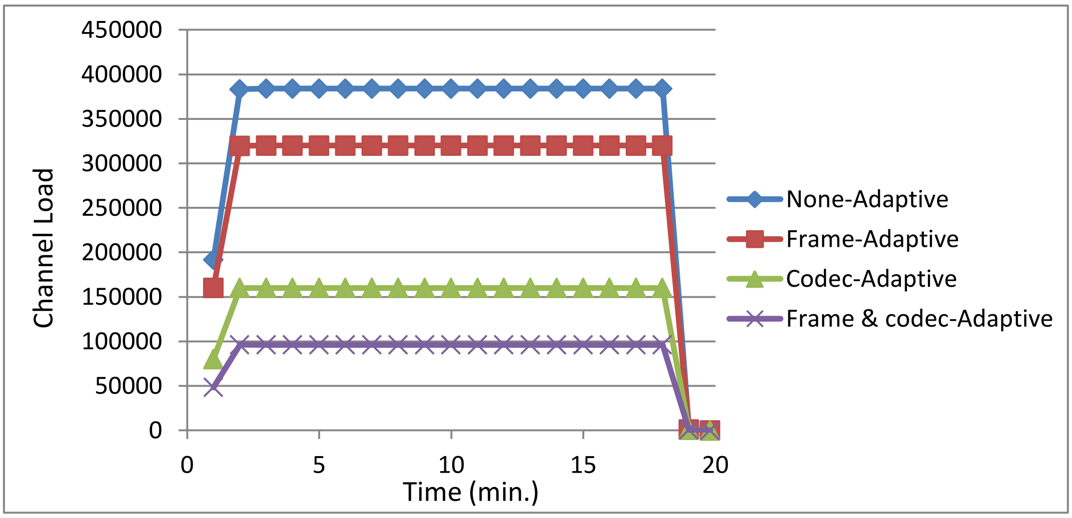



Figure 3.59: Channel load of different adaptation methods when all 4 stations have fallen to 1 Mbps transmission rate.

The end-to-end delay for different methods in Figure 3.60 shows that 'frame & codec' adaptive method results in the lowest delay among all methods. In this scenario codec-adaptive method that switches from G.711 to G.729 effects on consuming lower bandwidth and consequently lower delay in compare to frame-adaptive method which uses G.711. Also, frame-adaptive method that uses 2 fpp needs more packetization time causing to more delay and it is not in the acceptable range. Even though the end-to-end delay of *codec-adaptive method* is less than 0.1 second which is in the acceptable range, but since *its* MOS (Figure 3.58) is very low, that means this method is not useful for very high congestion.

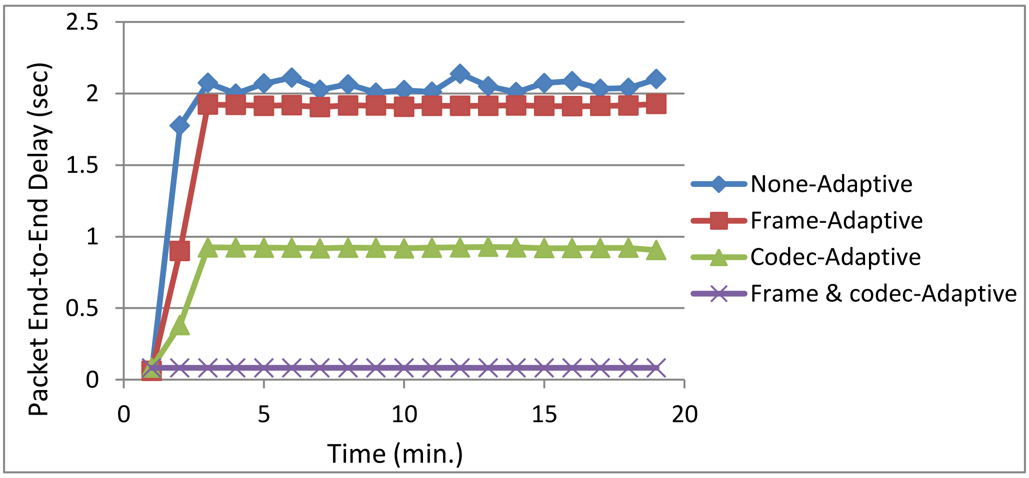



Figure 3.60: Delay of different adaptation methods when all 4 stations have fallen to 1 Mbps transmission rate.

For data loss due to buffer overflow, as Figure 3.61 illustrates, frame & codec-adaptive method has the best result among 4 tested methods. Codec-adaptive method that switches from G.711 to G.729 causes lower bandwidth consumption and consequently lower congestion which causes less accumulating of data in buffer and less data loss as compared to the frame-adaptive method.

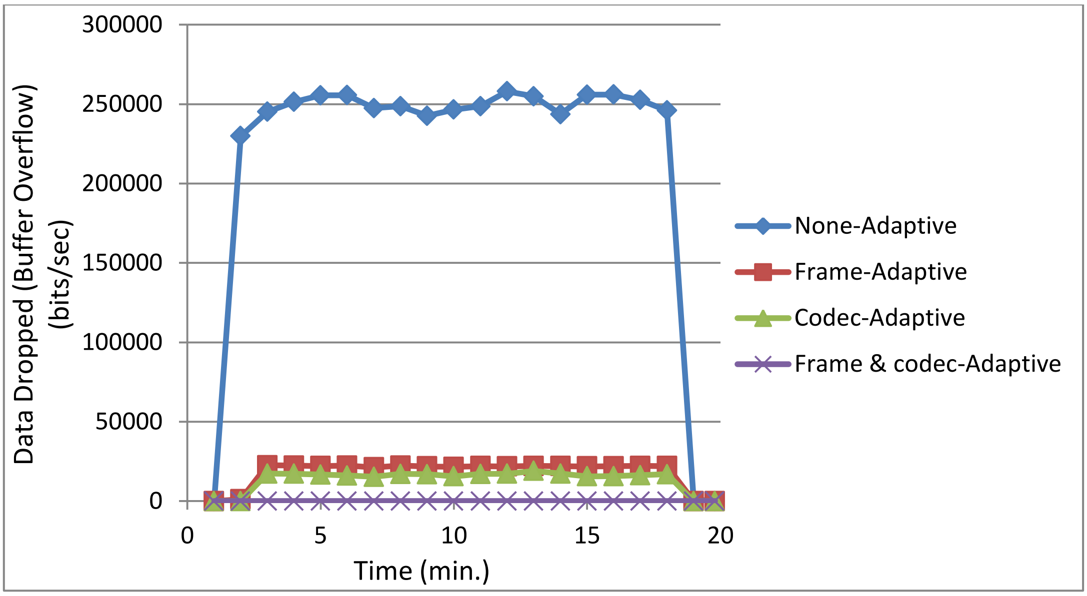



Figure 3.61: Data drop due to buffer over-flow of different adaptation methods when all 4 stations have fallen to 1 Mbps transmission rate.

Up to now, the comparison between ‘codec-adaptive’, ‘frame-adaptive’, ‘frame & codec-adaptive’ and ‘none adaptive’ methods have been analyzed and results prove that in most of the cases when the congestion is low or moderate, frame adaptation improves the perceived quality and results on lower load and lower end-to end delay and lower packet loss compared to none adaptive methods. However, in terms of severe congestion, codec adaptation acts better than frame adaptation since it has a biger effect on the speech output rate.

## 3.7 The Algorithm Flowchart and Pseudo Code

In the following, the flowchart of the proposed adaptation algorithm will be presented (Figure 3.62). The proposed algorithm is constructed based on the findings in sections 3.5 and 3.6 (finding the adaptation instance, and finding the proper adaptation process).

The algorithm initiates with the assumption of non-congested channel, so to have the best speech output quality, the algorithm selects G.711 codec and 1 frame per packet for coding. Afterwards, along with monitoring the changes in VoIP media stream, RTCP-XR is also monitored by the algorithm, if the transmission rate is reduced and/or or inter-arrival jitter is increased (compare to the former RTCP-XR) means that that current rate change requires for commencing the adaptation process. To make sure that adaptation will be done accurately, delay and MOS are also checked in this step.

Adaptation starts with packet size adaptation first, followed by codec adaptation (if it is necessary). Packet size adaptation will be performed (up to 3 frames per packet) and a counter determines these steps. After each step, quality factors will be checked to verify if packet size adaptation is successful. In the case that packet size adaptation does not rectify the congestion, the algorithm switches the current codec with the lower bit-rate codec and again quality factors will checked by sending fast RTCP-XR after the adaptation. However, when the codec is changed to the lower one, and the congestion still exist in the network, frame size adaptation should be applied on the recent codec to rectify the congestion.

When the congestion was able to be solved by any of above adaptation steps, the system will continue with the current coding parameters. The call goes on the monitoring parts again until the call being ended.

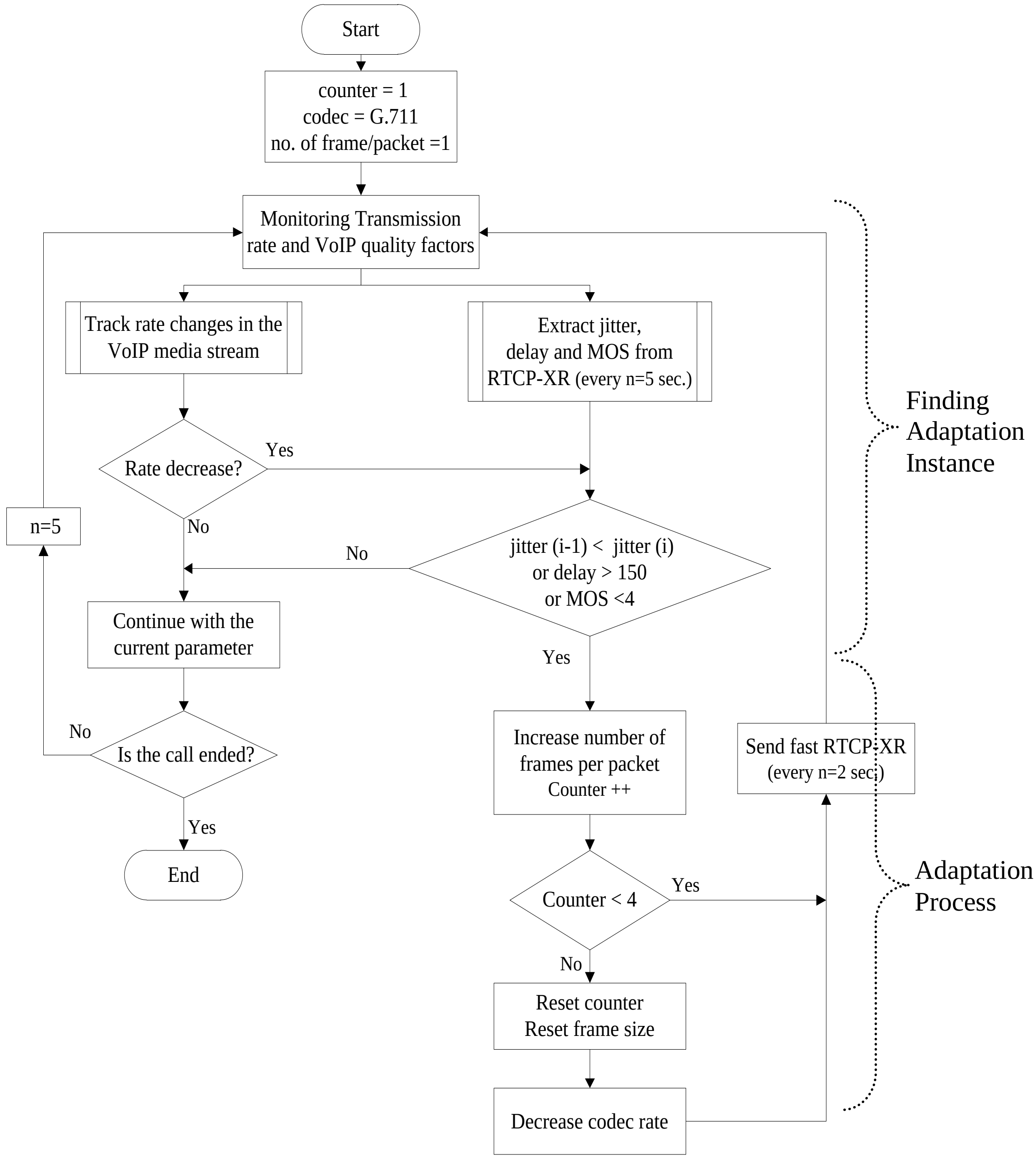



Figure 3.62: Proposed adaptation algorithm flowchart.

The pseudo-code of the proposed algorithm is as follow:

**Main Procedure Adaptive_VoIP**

Begin

1. Initialization of parameters
   - 1a. Set Counter=1
   - 1b. Set codec=G.711
   - 1c. Set number of frames per packet = 1
   - 1d. Set frequency of updates (n) = 5
2. *Call Monitoring_Proc*
3. *Call Track-rate_Proc*

End

***Sub Monitoring_Proc***

Begin

1. Monitor RTCP-XR in every n seconds and extract jitter, delay and MOS
2. If jitter(i-1)<jitter(i) or delay >150 or MOS<4 then
   - 2a. Increase the number of frames per packet
   - 2b. Counter++
   - 2c. If counter<4 then
     - 2c-i) *Call Monitoring-Frequency-Proc*
   - 2d. Else
     - 2d-i) Reset counter=1
     - 2d-ii) Reset frame-size=1
     - 2d-iii) Decrease codec rate
     - 2d-iv) *Call Monitoring-Frequency-Proc*
3. Else continue operation with current coding parameters

End sub

***Sub Track-rate_Proc***

Begin

1. Track rate changes in the VOIP media stream
2. If the rate track is stable and not decreasing then
   - 2a. Continue operation with current coding parameters
   - 2b. If the call ended then stop
   - 2c. else goto step-1
3. Else
   - 3a. *Call Monitoring-Frequency-Proc*

End sub

***Sub Monitoring-Frequency-Proc***

Begin

1. update n=2
2. send fast RTCP-XR in every n seconds
3. *call Monitoring_Proc*

End sub

### 3.8 Chapter Summary

This chapter discussed the main components of an adaptive rate control algorithm for VoIP. Then the capacity of a 802.11 b wireless network was estimated and based on this capacity estimation, a network model (Figure 3.28) was designed for further investigation.

Since this model has the maximum possible number of calls (based on capacity estimation in 3.3) the effect of reducing the transmission rate of call was investigated. *The results of transmission rate reduction on instant quality factors* namely jitter, PDV, end-to-end delay and packet loss were demonstrated in Figures 3.38- 3.40 and 3.47 -3.50, also *the results of using different adaptation method for transmission rate reductions* (shown in Figures 3.52 to 3.64), present two main conclusions (that have been used in the proposed algorithm):

First, the results of MOS show, unlike previous works, some of the transmission rate changes do not affect speech quality critically and call continues with almost the same level of MOS for some rate changes. Therefore, MOS is not a good index for defining transmission rate changes. Therefore, the proposed technique in this study checks other instant VoIP quality metrics beside MOS to decide for the right adaptation time. As it was shown in Figures 3.49 and 3.50, between the instant quality factors packet delay variation (jitter) is the best index to show transmission rate reductions.

Second, the evaluation of frame adaptation, codec adaptation and combination of both approaches showed that frame adaptation can rectify low to moderate congestion. Since this approach has less impact on the system in term of simplicity and lower coding cost, so in this study the algorithm will be started by frame adaptation in the case of low congestion. Results also showed if the system turns to higher congestion where frame adaptation is not enough, the algorithm will commence codec adaptation. Furthermore, in very high congestion, if codec adaptation could not rectify the congestion, both techniques (codec and frame size adaptation) will work together.

# CHAPTER 4

# RESULTS AND DISCUSSION

## 4.1 Chapter Overview

This chapter presents the most significant results gained by our simulation in order to validate our findings and compare our approach to the known previous approaches. The first part of this chapter will show that adaptation is not required in every transmission rate change which is one of the contributions of this study. In the second part, the comparison was made between the result of “Adaptive” and “None-Adaptive” methods. Delay, data loss, MOS[8] and channel load are used to evaluate the proposed mechanism. In addition, the comparison is done between the adaptive mechanism in this study and the previous approaches.

It should be mentioned that in all the simulation, the traffic is purely VoIP and network does not include any other traffics. The VoIP calls are between points in the office environment where clear line of sight is available between the sender to AP and AP to the receiver. In this environment when both parties use 11 Mbps transmission rate, two calls could be available with good quality, which are simulated trough two pairs of caller-callee (four stations). The calls have been assigned to each pair of caller-callee according to section 3.4.

In addition, in this chapter, the wireless stations with the transmission rate of 1 Mbps has been named low-rate stations. The default frame size is 10 milliseconds. The results were collected from Voice and Wireless LAN global statistics of OPNET.

---

[8] Objective MOS in the results of this chapter is one of the outputs of OPNET modeler.

## 4.2 Evaluating the Algorithm on Adaptation Instance

The graph in Figure 4.1 shows the fact that in IEEE 802.11 WLANs, congestion does not happen gradually rather it happens instantly. It means sometimes network can sustain the extra traffic but sometimes by adding of a tiny extra traffic network tends to switch from an un-congested condition with good perceived speech quality to a congested condition with bad perceived speech quality [51].

In order to demonstrate that every transmission rate reduction does not need adaptation, some scenarios are conducted with different transmission rate and in all of them the calls use G.711 codec with one frame per packet and the transmission rate in one of the calls (a pair of source and destination) changes in descending form (from 11 to 5.5 then to 2 and end up with 1 Mbps).

As Figure 4.1 shows the first three rate reductions in one of the calls do not need adaptation mechanism, since the average MOS is maintained high (almost 4.3). But if this call drops to transmission rate of 1 Mbps, the MOS decreases to 1 which is unacceptable quality, so the adaptation approach is needed in this instance (not for all previous transmission rate reductions). Consequently, the system can withstand some transmission rate reductions until the network gets congested.

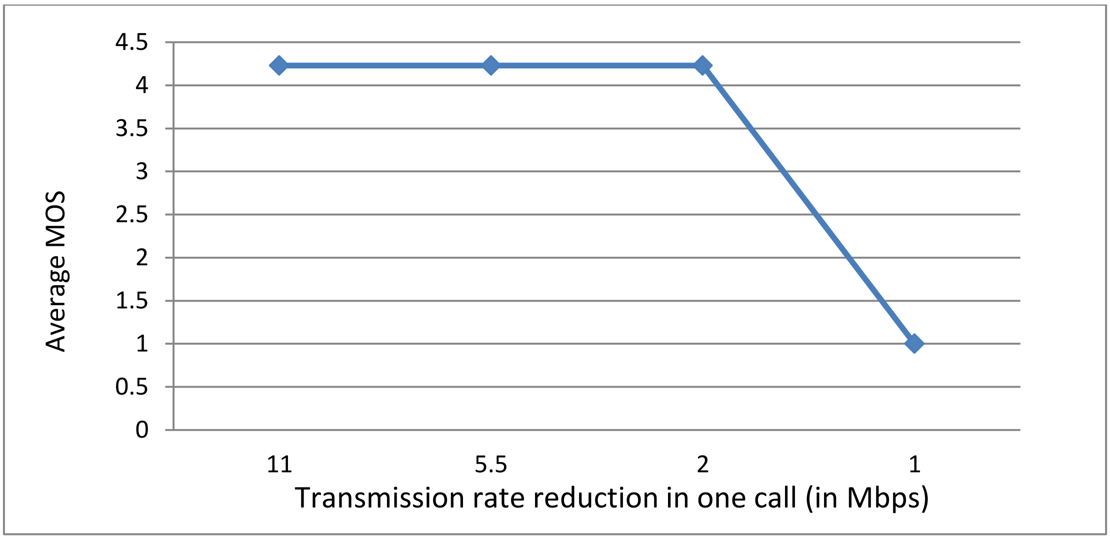



Figure 4.1: MOS results when the transmission rate of one call drops gradually.

Figure 4.1 also shows only the transmission rate changing cannot be an adequate indication for triggering the adaptation mechanism. Unlike the previous works that they changed the coding rate with all variations in transmission rate (MAC alarm), the proposed method in this study only changes the coding rate when is needed.

The adaptation instance in the proposed algorithm is determined by monitoring the quality factors such as jitter and delay besides monitoring the transmission rate variations (as discussed in the methodology chapter). That is, when jitter and delay are going to increase sharply and beside that, MOS is going to decrease to the unacceptable range, adaptation should be commenced. Consequently, unlike some previous studies, the proposed algorithm by this study determines the adaptation instance not only by MAC alarm rather, beside MAC, quality factors will be checked to perform the adaptation timely and accurately.

in Figure 4.2, *Codec-Adaptive* graph demonstrates the codec adaptation method by Anna's work [27], for the scenario in that one of the call's transmission rate is changed to the lower transmission rate gradually. Besides, the non-adaptive graph also presented to compare these two methods in term of perceived quality.

Obviously, in the sufficient bandwidth, G.711 codec provides the best quality among the codecs (MOS 4.3) and other codecs (like G.729) provide lower quality (MOS 4) [106]. In regard to the first three points of both graphs in Figure 4.2, when the bandwidth is still enough and network is not congested yet, the system can still continue with higher bitrate codecs and adaptation is not needed. In this situation, switching the codec from G.711 to G.729 causes quality degradation from 4.3 to 4 (first three points of Codec Adaptive graph). As previous codec adaptation approaches like Anna's work [27] change the codec for all transmission rate, they scarify the good quality of G.711, even in *non-congested situation*. But the proposed algorithm in this study trys to keep the G.711 till the real need of adaptation time (which is determined by quality factors beside transmission rate changes).

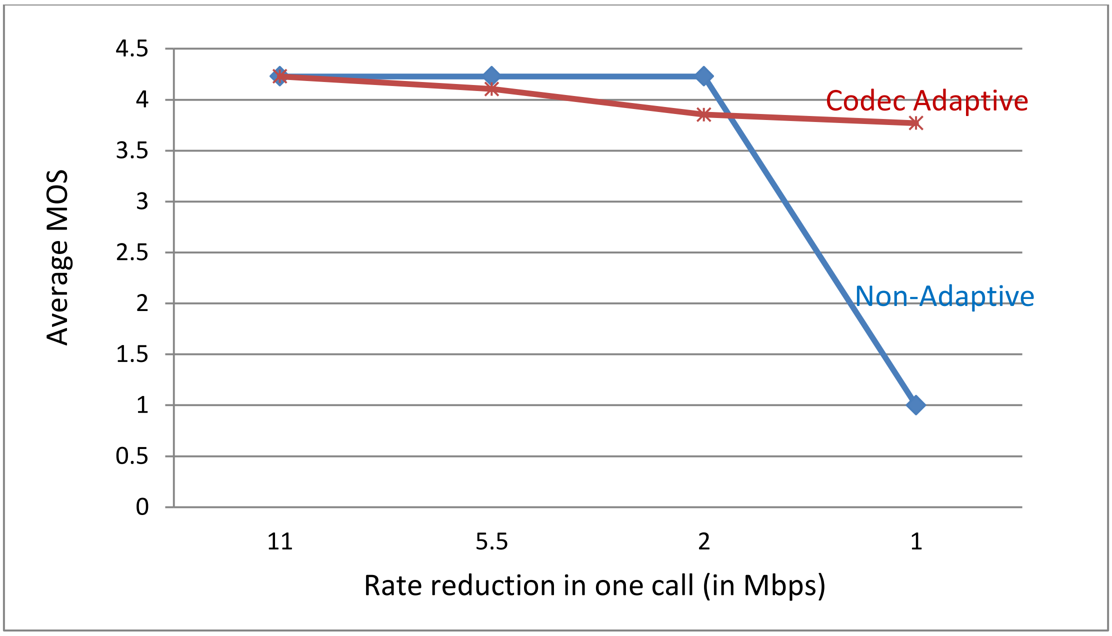



Figure 4.2: Comparison between MOS results of Anna's work [27] & non-adaptive method when the transmission rate in one of the calls is reduced gradually.

Evidently, as Figure 4.2 shows, for the last transmission rate reduction to 1Mbps, codec adaptation acts better than no adaptation. However, for the first three transmissions rate changes from 11 to 5.5 and then to 2 Mbps "No Adaptation" has the top quality, even better than codec-adaptive method, because it uses G.711 with higher output rate.

Accordingly, with the mechanism used in this study, the system does not enter to the adaptation phase till it would be required. The main advantage of this approach is cost reduction in term of changing the call's coding parameters frequently during the call.

So far, the right instance to commence adaptation is evaluated and hereafter the adaptation method will be evaluated. In fact, adaptation phase could be accomplished using codec rate adaptation and/or using frame size adaptation. The results in chapter 3 showed that in most of the cases frame size adaptation would be enough for the system to recover from congestion. It is due to grouping of more frames in one packet which cause less overhead to carry in the network so lower congestion would be expected. Also frame adaptation is more flexible and has less cost in terms of modification impact and it is free of paying license fees. Moreover, when the transmission rate is changed slightly, the frame size adaptation can control the rate of traffic carried by network but codec adaptation changes the traffic volume hugely. So in the small congestion, frame size adaptation could be used instead of codec adaptation. That is why the proposed mechanism increases the number of frames up to three steps and checks the quality factors after each of them. If the speech quality turns into the acceptable range, the system does not undergo the codec adaptation but if frame size adaptation does not rectify the congestion then codec adaptation would be commenced. The rest of this chapter is to evaluate the proposed algorithm.

## 4.3 Comparison between Adaptive and None-Adaptive Approaches

Henceforth sections (4.4.1, 4.4.2, and 4.4.3) demonstrate the comparison between non-adaptive and the proposed adaptive method for different congestion level, namely (low, moderate, and high congestion).

### 4.3.1 First Simulation Scenario, Low Congestion

In order to compare proposed adaptive method and none adaptive method on the low congestion level, the simulation scenario is conducted with 4 stations in the network and transmission rate of one of the calls (between two stations) is 11 Mbps and the other call is 1 Mbps. Figure 4.3 illustrates the average MOS using an adaptation method and without adaptation method.

In non-adaptive method, both calls used G.711 codec and 1 frame per packet (fpp) regardless of network condition. As non-adaptive graph shows that adding, the second call (in the $2^{nd}$ minute) will cause more congestion and it affects the MOS to reach to almost 1 which is a very bad quality. On the other hand, when the call is dropped to 1Mbps transmission rate, performing the adaptation and increasing the number of frames from 1 to 2 frames per packet maintains the quality at the high level, even when the second call is added.

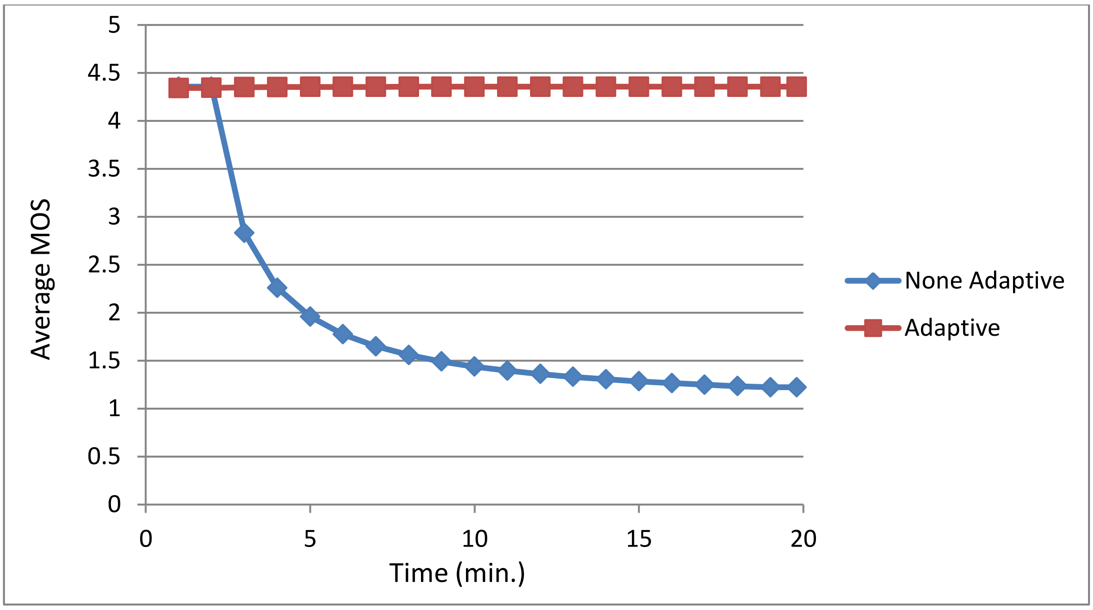



Figure 4.3: Comparison between the MOS result of non-adaptive and adaptive methods when 2 stations use 11 Mbps and 2 others use 1 Mbps transmission rate.

In the proposed simulation model when the transmission rate of two stations drop from 11 to 1 Mbps and the call uses G.711 codec with 1 frame per packet, the MOS is very low and the call quality is not acceptable. The results obtained from [26] also shows very low quality in term of MOS (MOS=1) in saturation time. This MOS is calculated in real-time using the E-model. However, after adaptation and by changing the packet size to 2 frames per packet the MOS turns good.

The same simulation scenario is used in evaluation of end-to-end delay. Figure 4.4 shows the delay is acceptable (less than 0.15 Sec or 150 millisecond) when the adaptation method is applied to the network and in none adaptive graph the delay is very high.

According to result analysis of [26], also when 2 stations change their transmission rate from 11 to 1 Mbps, due to saturation of queue in AP, packet delay reaches approximately 1 sec. which is matched with the proposed result in this study.

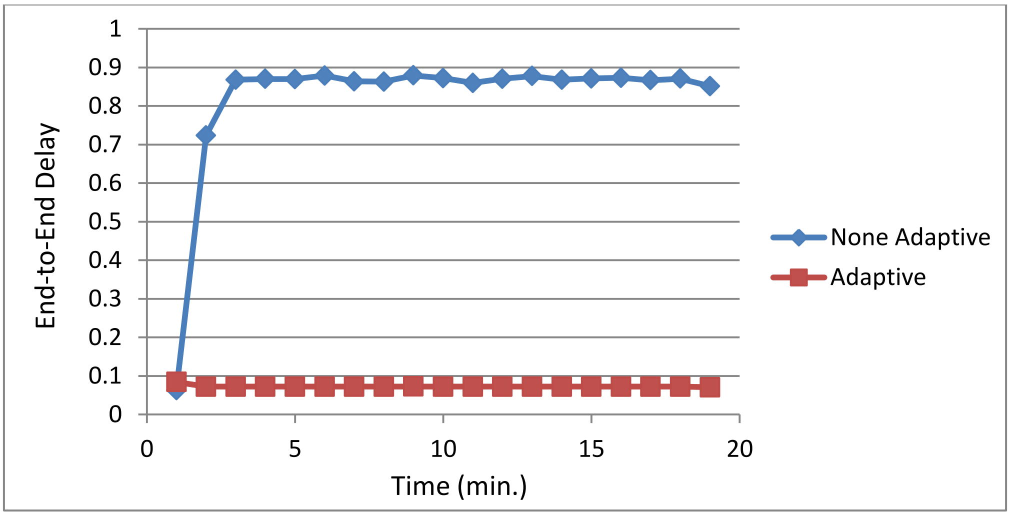



Figure 4.4: Comparison between the delay result of non-adaptive and adaptive methods when 2 stations use 11 Mbps and 2 others use 1 Mbps transmission rate.

Data loss due to buffer overflow for the same simulation scenario is shown in Figure 4.5 for both methods (using the adaptive method and without using it). As result of Figure 4.5 shows data loss due to buffer overflow is almost zero for adaptive method. It is because of less overhead due to having larger packet size. As it is shown in Figure 4.5, for none adaptive system in the first minute and at the last minute the data loss is almost zero which is according to the setting of calls in the profile definition (Figure 3.34) in that no call exists in the first and last minute of our simulation time.

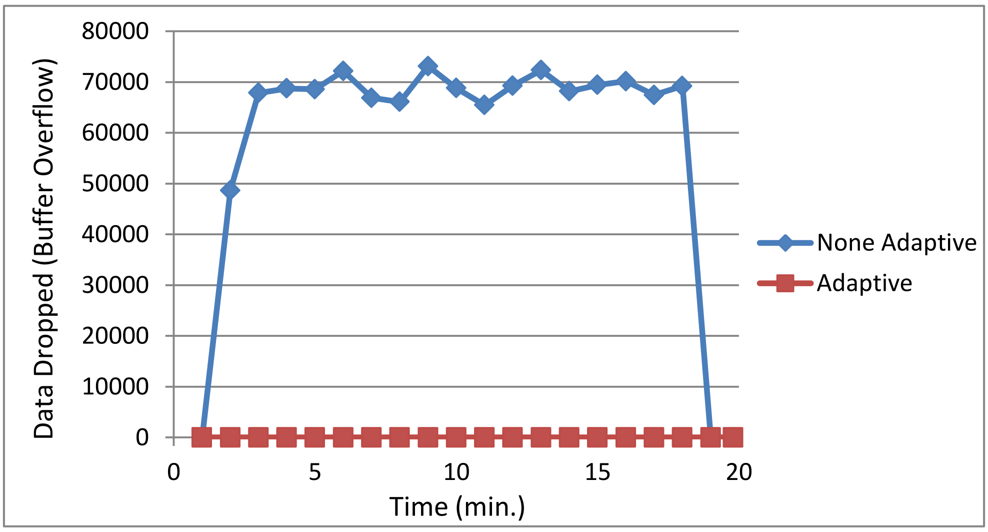



Figure 4.5: Comparison between the data drop rate of non-adaptive and adaptive methods when 2 stations use 11 Mbps and 2 others use 1 Mbps transmission rate.

The channel load in wireless LAN model is shown in Figure 4.6, that illustrates adaptive methods has lower load compared to the none adaptive method. Although the difference is not huge but as the previous Figures (4.3-4.5) presented, even this small change has a large effect on the quality metric improvements.

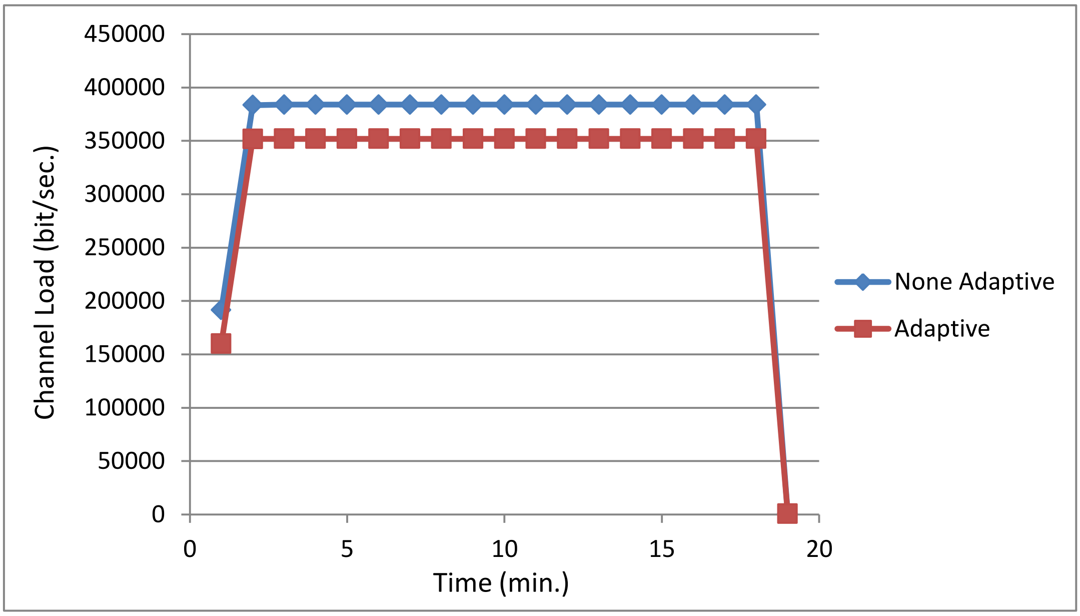



Figure 4.6: Comparison between the channel load of non-adaptive and adaptive methods when 2 stations use 11 Mbps and 2 others use 1 Mbps transmission rate.

### 4.3.2 Second Simulation Scenario, Moderate Congestion

Next four Figures show the effect of using adaptive method and none adaptive method when number of low-rate stations increase to 3 and only one station uses 11Mbps transmission rate. It means comparing to the previous scenario that had 2 low-rate stations here 3 low-rate stations are simulated. In the non-adaptive systems, the calls start and maintain using G.711 codec and 1 frame per packet but in adaptive method the packets increase to 2 frames per packet.

Figure 4.7 illustrates that MOS for none adaptive graph descend very sharp on the second minute. That is, when the 2nd call with the low transmission rate is added into the network and causes congestion which results on poor perceived quality with the very low MOS. However, adaptive graph shows (in the same scenario) when both calls increase their number of frames from 1 to 2 frames per packet the calls maintain on a good quality.

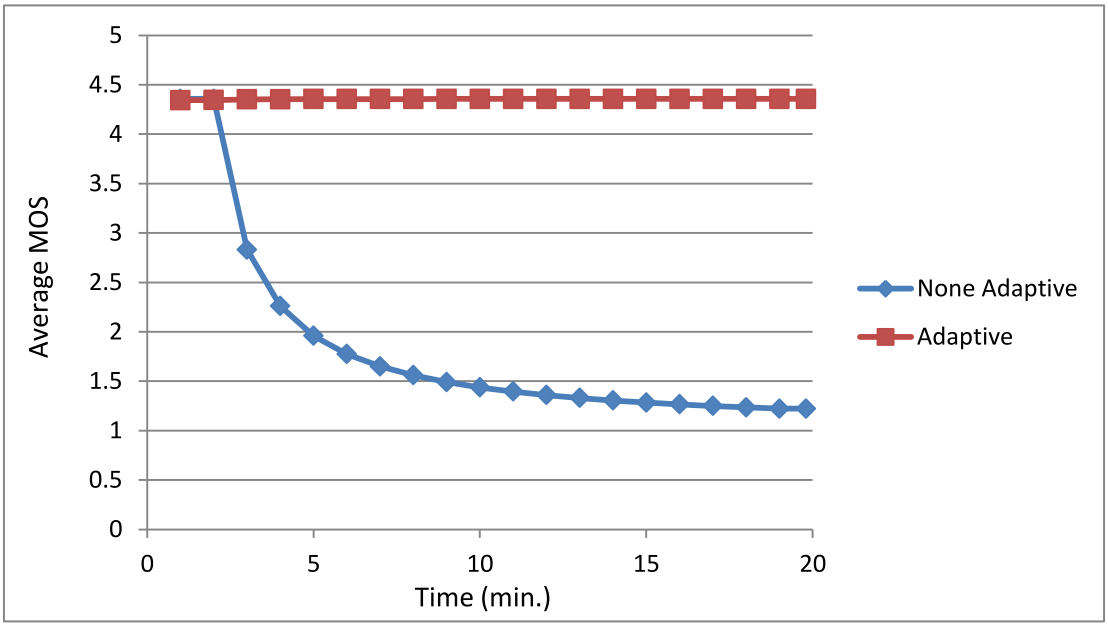



Figure 4.7: Comparison between the MOS of non-adaptive and adaptive methods when 1 station uses 11 Mbps and 3 others use 1 Mbps transmission rate.

In Figure 4.8, End-to-End delay is demonstrated for adaptive and non-adaptive systems. For non-adaptive method, on the second minute by adding the second low-rate call into the network the delay increase sharply to the unacceptable range and it maintains on that range. On the other hand, the adaptive method the graph shows that end-to-end delay is less than 0.1 second (100 millisecond) in the first minute although it riches to 0.1 seconds after adding the second slow calls. However, adaptive method using one-step bigger frame size (2 frames per packet) keeps the delay in the acceptable range.

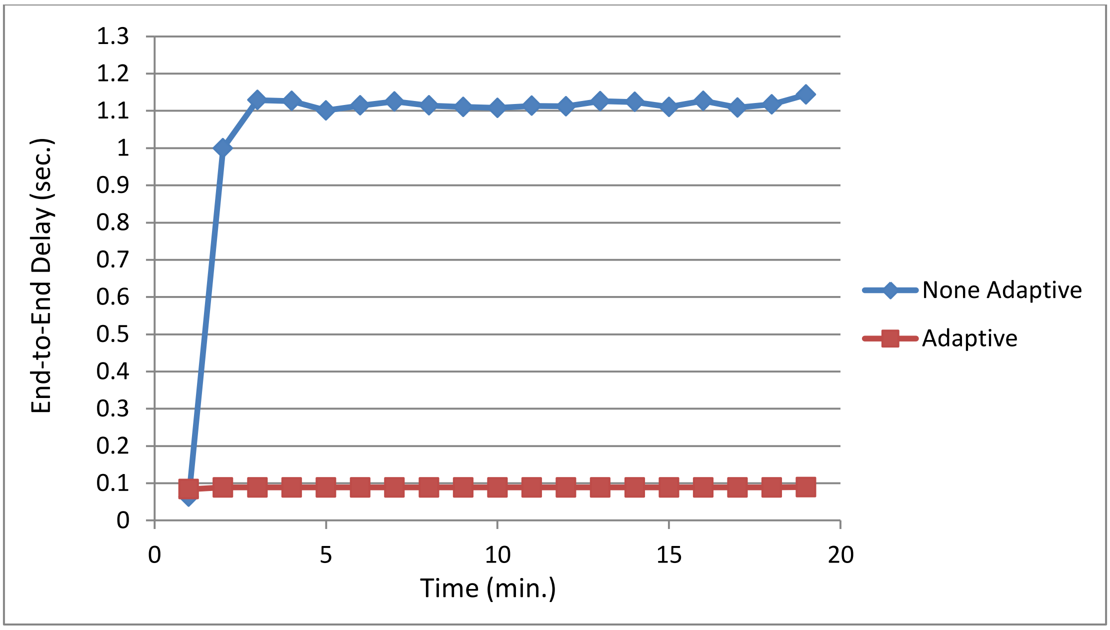



Figure 4.8: Comparison between the delay in non-adaptive and adaptive methods when 1 station uses 11 Mbps and 3 others use 1 Mbps transmission rate.

Data loss due to buffer overflow is shown in Figure 4.9 which demonstrates adaptive method has almost no packet loss while data loss is noticeably higher for none adaptive method. The big difference between the mean values of these two graphs is related to the amount of overhead. While the none adaptive method uses one frame per packet the overhead is added to the packet is for only one frame but for adaptive method which uses two frames per packet overhead that is added to the packet is shared for two frame pieces and it means less amount of overhead is dedicated to the adaptive method. Take a note that, having no call at the first and last minute of simulation caused no data loss for none adaptive method at these times.

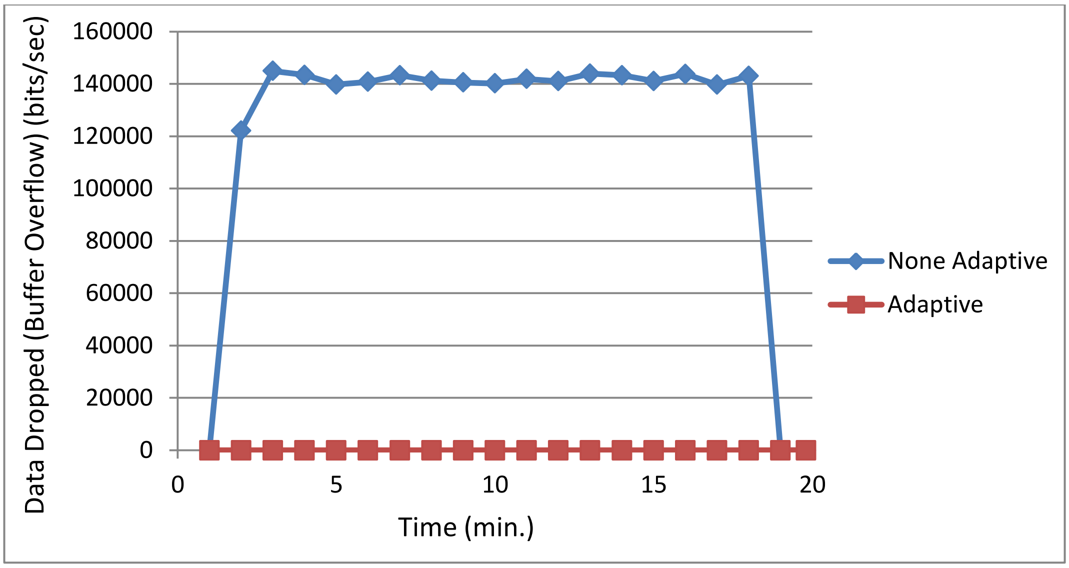



Figure 4.9: Comparison between the data loss rate of non-adaptive and adaptive methods when 1 station uses 11 Mbps and 3 others use 1 Mbps transmission rate.

Load of wireless channel is presented (for both methods) in Figure 4.10 graphs show low load up to the second minute of calls because of having only one call, and no load at the first and last minute of simulation time because of having no call at that moment. The total load of adaptive method is lower than none-adaptive method due to having less overhead.

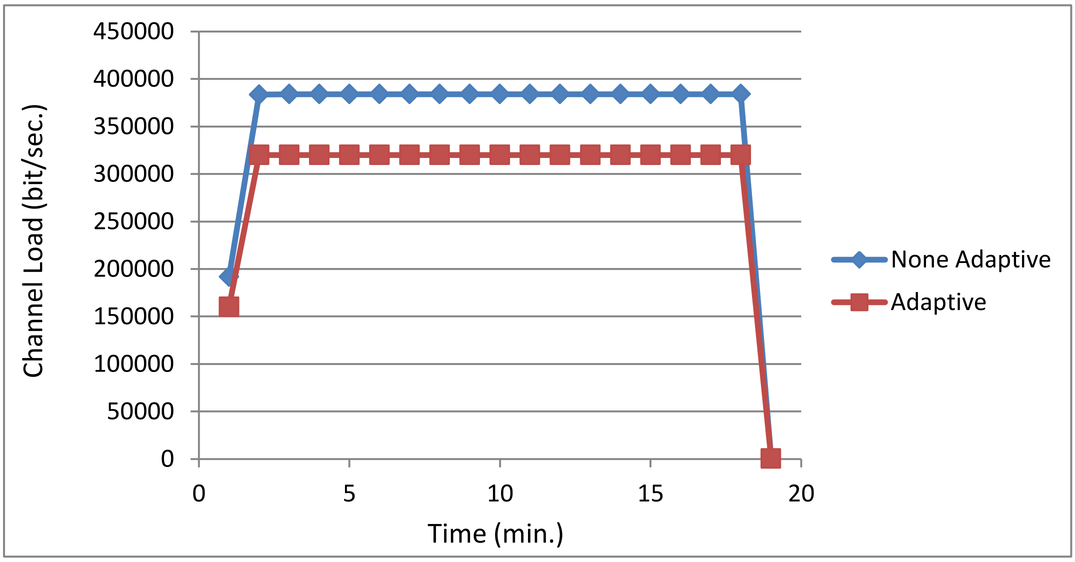



Figure 4.10: Comparison between the channel load of non-adaptive and adaptive methods when 1 station uses 11 Mbps and 3 others use 1 Mbps transmission rate.

### 4.3.3 Third Simulation Scenario, High Congestion

In order to evaluate the proposed algorithm for higher congestion, the simulation scenario has been set with 4 stations in the network that all of them falls into 1 Mbps transmission rate. In this scenario since all stations fall to 1 Mbps transmission rate and the AP also has the transmission rate of 1 Mbps, congestion is high. In this situation, codec adaptation (switching from G.711 codec to G.729) alone or packet size adaptation (switching from 1 frame to 2 frames per packet) alone would not be enough to recover the network. Consequently, combination of both approaches taken to achieve the lower load in congestion situation.

The result of none adaptive method and adaptive method are shown in in Figures 4.11 to 4.14. Non-adaptive graph is the result of using G.711 codec with 1 frame per packet, and adaptive graph is the result of using G.729 codec beside using 2 frames per packet.

The average MOS is shown in Figure 4.11. None adaptive method fails to provide acceptable quality after adding the second call. However, frame and codec adaptive method can maintain at a good quality level.

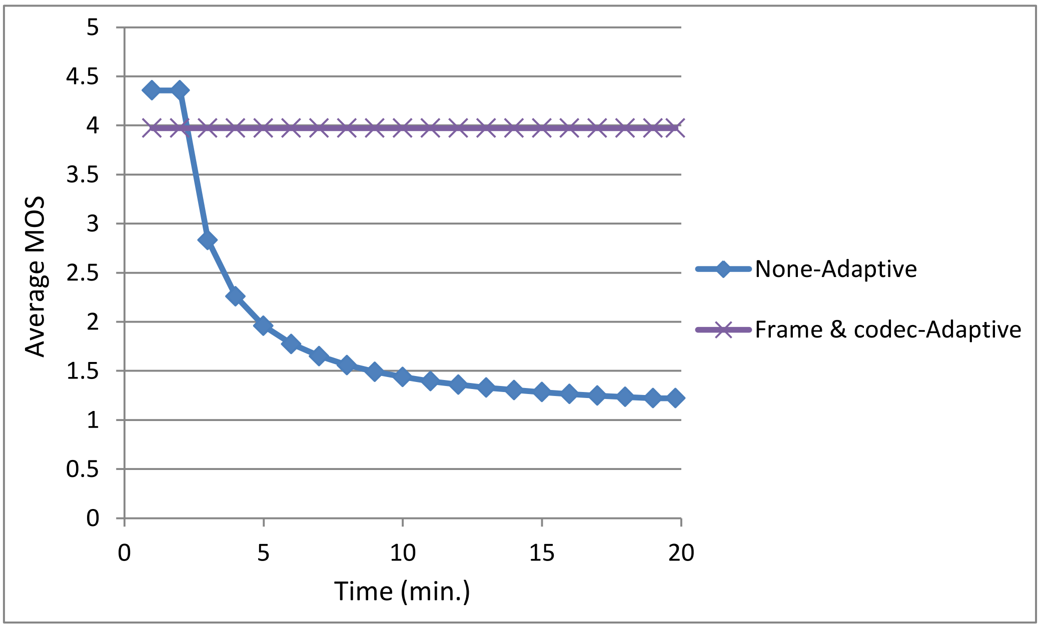



Figure 4.11: Comparison between the MOS of non-adaptive and adaptive methods when all 4 stations fall to 1 Mbps transmission rate.

The result of end-to-end delay in Figure 4.12 shows that 'frame & codec' adaptive method results in the lower delay comparing with non-adaptive methods. The reason is using more frame in one packet reduce the overhead amount in the data transmission and it effect on lower traffic load, so the packets reach to the destination with the lower delay.

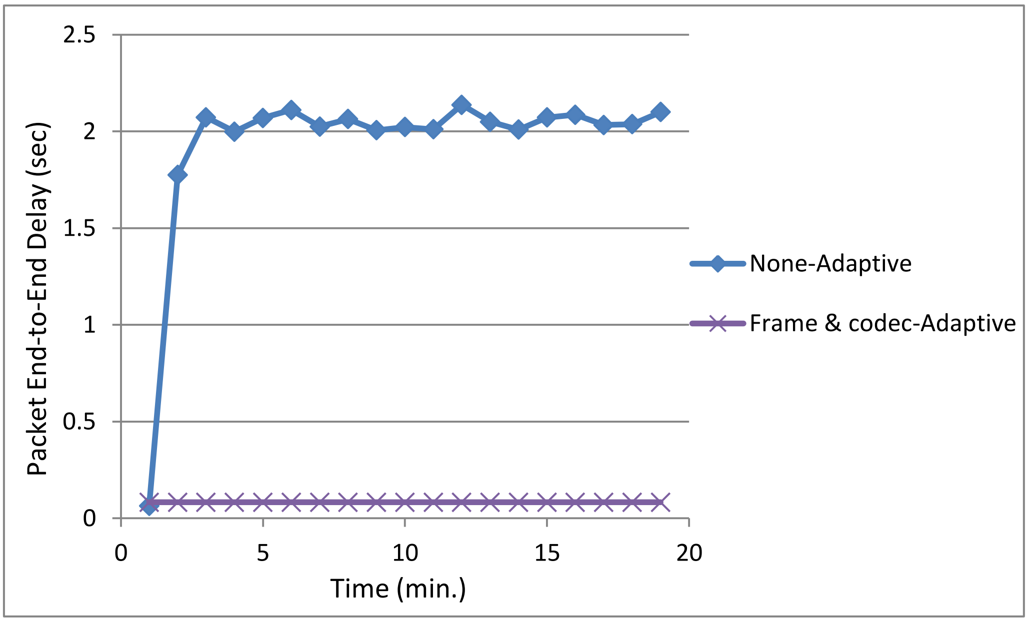



Figure 4.12: Comparison between delay of non-adaptive and adaptive methods when all 4 fall to 1 Mbps transmission rate.

For data loss due to buffer overflow as illustrates in Figure 4.13, the proposed adaptive method has very low data loss. The proposed adaptive method that switches from G.711 to G.729 and from 1 fpp to 2 fpp, leads to lower bandwidth consumption and consequently lower congestion which cause less accumulation of data in the reciver buffer and thus less data loss (in comparison to non-adaptive method).

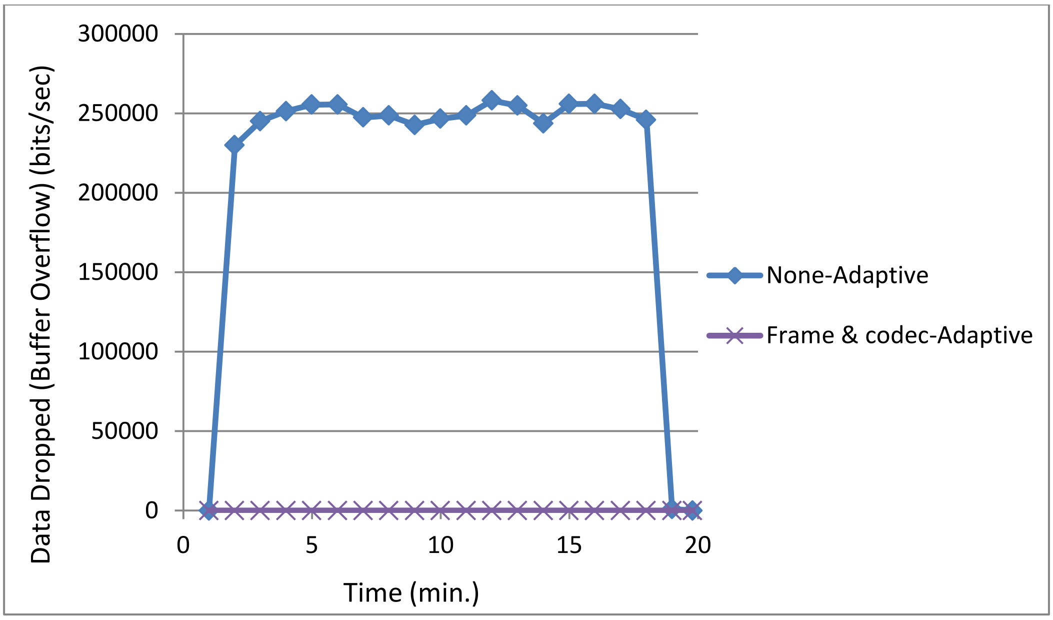



Figure 4.13: Comparison between the data drop rate of non-adaptive and adaptive methods when all 4 stations fall to 1 Mbps transmission rate.

The proposed adaptive method switches from high bitrate codec (G.711) to the lower bit-rate codec (G.729). Therefore, this method reduces the load of traffic greatly. In addition, the adaptive method uses larger packet size, which has lower overhead to data ratio, consequently it contributes to have a lower load (Figure 4.14).

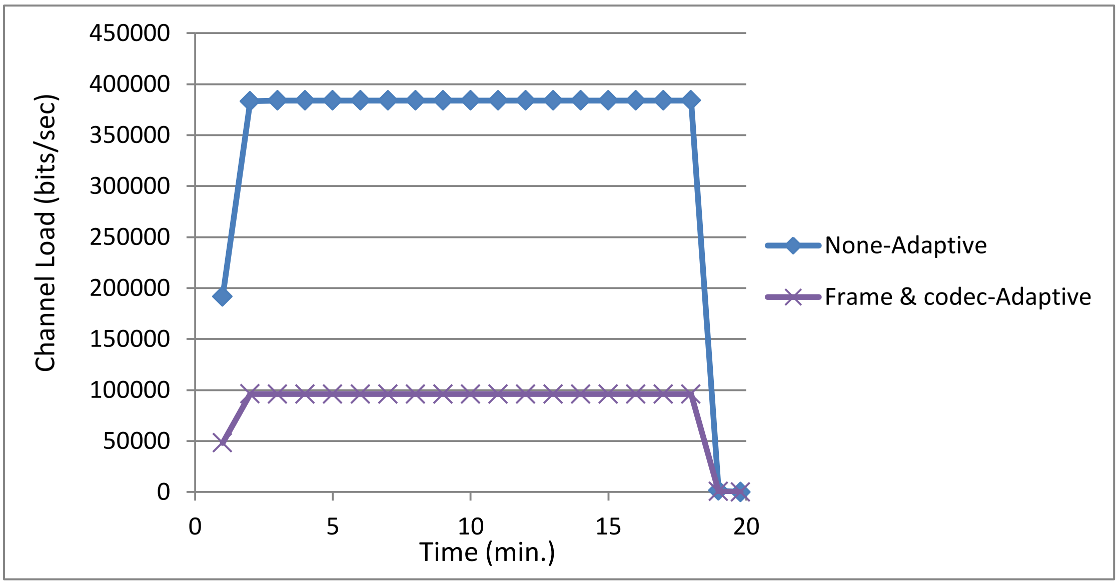



Figure 4.14: Comparison between the channel load of non-adaptive and adaptive methods when all 4 stations fall to 1 Mbps transmission rate.

Until now, the comparison between the proposed adaptive method and the none-adaptive method were done and the results proved that during the congestion, adaptive approach improves the perceived quality with providing the lower load, and lower end-to end delay, and lower packet loss.

## 4.4 Algorithm Validation

To compare this research work with other closely related work, it is necessary to flashback to the literature review chapter. According to the comparative study of codec solution for VoIP over WLANs by Anna et al. [59], every codec adaptation approach has a *decision policy* which determines the mechanism of codec adaptation. They categorized these policies into three groups; *None-Adaptive policies*, *Single-Adaptive policies* and *Multi-Adaptive policies*.

*None-Adaptive policy:* In the first category, a fix codec is used and all calls maintain the codec for the whole call duration (no codec adaptation is performed). So, in the congestion, the calls suffer from a very bad quality will be simply dropped.

*Single-Adaptive policies:* In this category, calls that cause capacity reduction (for example by reducing their transmission rate) are those that perform adaptation.

*Multi-Adaptive policies*: In the third category, to remedy the congestion, any number of calls can be chosen to change their codec to the lower one. In order to attain the fairness, those calls that changed their rates are the first options for adaptation since they caused the congestion. If the congestion is not solved yet, other calls also need to change their codec, so further policies should be considered which make this solution more complex.

Table 4.1: Policy-based categories of previous adaptive rate VoIP algorithm.

| Single-Adaptive | Multi-Adaptive |
|---|---|
| MC Govern [51] | Tüysüz [35], [75], [76] |
| | Sfairopoulou<br>[67], [68], [26], [107], [59] |
| | Tebbani [81] |

This research work is a kind of Multi-Adaptive policies because the first phase of the proposed algorithm (finding the right adaptation instance) will be executed by all the stations, so even if a node does not face with transmission rate reduction but suffers from a bad quality due to network congestion, may also enter the adaptation process.

Among the Multi-Adaptive policies only Tüysüz and Sfairopoulou have used RTCP packet as feedback. RTCP has accurate information about speech quality factors of the current session. Since the proposed algorithm also uses a new version of RTCP, Tüysüz et al. work [76], and Sfairopoulou works [26] are considered as most related studies to this research.

Tüysüz et al. [76] proposed an algorithm that is implemented on AP, while Sfairopoulou considered her algorithm in the distributed way (on stations) too. However, according to [67] implementation of the algorithm on the AP give more processing effort to the AP but the distributed method (on the stations) gives almost equally satisfactory results. Since the implementation of distributed mode in practice is easier and it does not need programmable APs or specific device [51] we considered the distributed method for the proposed approach. Thus findings are compared to the method in [26].

Nonetheless, as compared to the study by Tüysüz et al. [76], it is suggested to use RTCP-XR packet instead of RTCP. In view of the fact that RTCP-XR provides more quality factors and it omits the quality measurement part of their algorithm, so it can improve their algorithm with faster reaction.

The rest of this chapter compares the proposed adaptive method and other previous adaptive method presented by [26] in two main scenarios; *first, when the rate decreases in one call, and second, when the transmission number of slow stations are increased*.

The abbreviations given below are used in our graphs; CA is Codec Adaptation method that is presented by [26], FCA is the Frame size–Codec Adaptation method presented by this study and NA is when No Adaptation method is applied.

With the same simulation model, CA, FCA and NA are evaluated *when the rate decreases in one call* and the results are shown in Figures 4.15 and 4.16.

Figure 4.15 shows that when a call drop to 1 Mbps transmission rate, both CA and FCA approaches have very good quality in term of MOS while NA has an unacceptable quality (MOS 1). As been demonstrated in Figure 4.15, FCA has the higher value of MOS for all transmission rate reduction compare to CA.

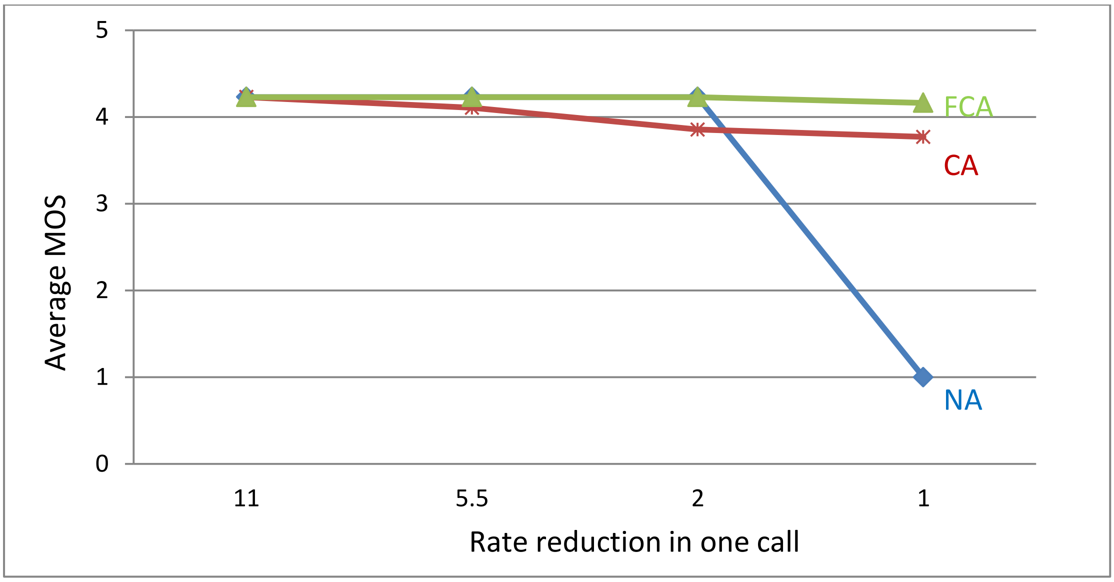



Figure 4.15: The comparison between FCA, CA and NA in term of MOS when transmission rate of one call is being reduced gradually.

In our adaptation method, when the transmission rate of station is not changed, and quality factors are in the acceptable range, the algorithm does not enter to the adaptation part. While in [26], after every transmission rate reduction in the wireless link codec in-use should be switched to the lower bitrate codecs (according to the table 4.1 order).

Table 4.2: Different codec used in the codec adaptation of [26].

| No | Codec | Data bitrate (kbps) |
|---|---|---|
| 1 | G.711 | 64 kbps |
| 2 | G.726 | 16, 24, or 32 kbps (here 32 has been used) |
| 3 | G.723.1 | 5.3 or 6.3 kbps |
| 4 | G.729A | 8 kbps |

As the NA graph in Figure 4.15 shows, when transmission rate reduction does not cause congestion, the non-adaptive method can also maintain a good quality and adaptation is not required (first three points on the NA graph). Consequently, in such a situation, switching from higher bitrate codec to lower bitrate codec causes degradation in quality and further it gives a heavy burden to the system to send SIP re-invite message and prepare the call with the new parameter while it is not needed. Moreover, it needs several codecs, which imposes more cost. These are the disadvantage as compared to the proposed method of this study.

As the FCA graph in Figure 4.15 shows, with the first three transmission rate reductions in one call, the network does not affect by congestion and so calls continue with the previous coding parameters. However, when in the last transmission rate reduction (to 1 Mbps), the frame size adaptation has been used (switching to the bigger frame size).

Figure 4.16 demonstrates the comparison between CA and FCA and NA from End-to-End delay point of view. It shows both adaptive methods act better with the lower delay in comparison with to the none adaptive system and the delay is maintained in the acceptable range.

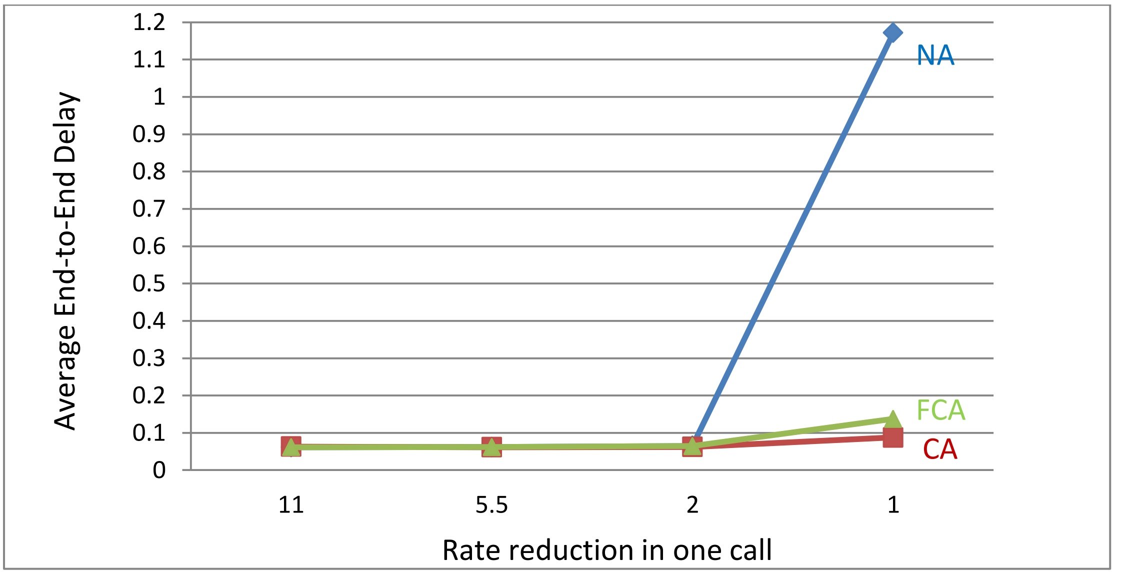



Figure 4.16: Comparison between end-to-end delay of FCA, CA and NA when transmission rate of one call is being reduced gradually.

As Figure 4.16 illustrates, the FCA has a little higher end-to-end delay when transmission rate of one call drops to 1 Mbps. This is related to the larger packet size (2 frame per packet) which the proposed method been used for adaptation in the last transmission rate reduction. While, CA uses lower bit-rate codec and so the load is lower, consequently packets are delivered to the destination with lower delay. However both graphs show the acceptable range of delay for calls.

For more investigation, a new scenario was conducted in that *number of low-rate stations increases in the network* (in the same simulation model). Again, CA, FCA and NA are compared and the results are presented in Figures 4.17 and 4.19.

Figure 4.17 shows that using adaptation methods (CA and FCA), the MOS is high and in the acceptable range, while NA graph shows an instantly sharp drop when two stations fall into the low transmission rate. Take note that, the first point of the NA graph (in 1 low-rate stn) supports our previous finding that all rate reduction does not need adaptation process. Here, when only one station's transmission rate drops into the lower rate, the slight traffic load does not affect MOS too much. The strongest reason why NA graph has dropped sharply is that, congestion in the wireless 802.11 link is not gradual and even tiny extra traffic will cause quality degradation [51].

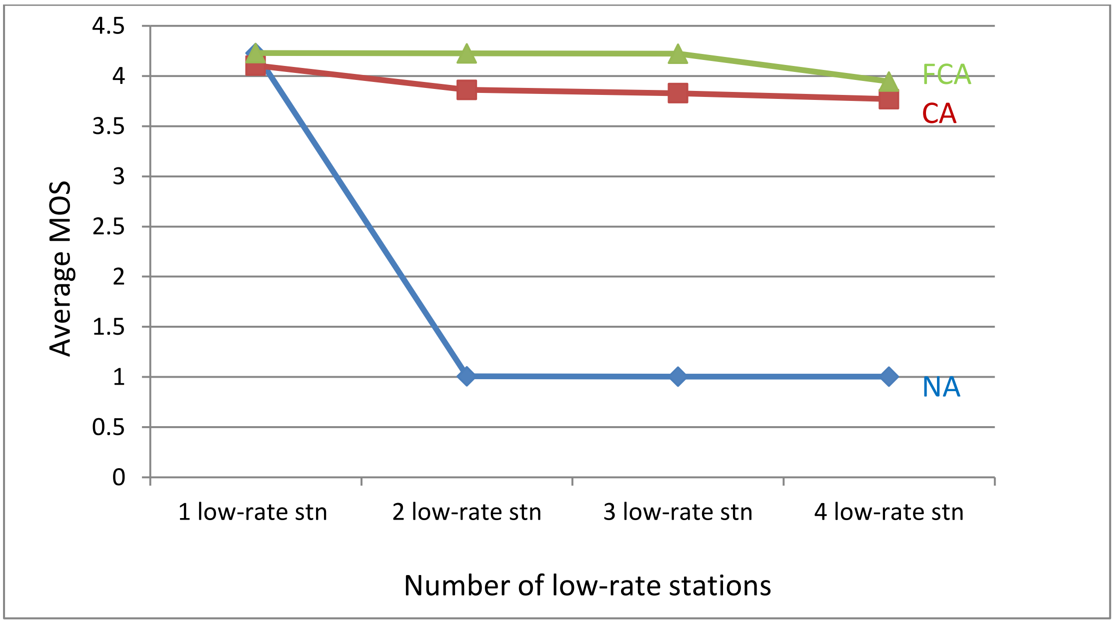



Figure 4.17: The comparison between FCA, CA and NA in term of MOS when number of low rate stations are increased.

Comparison between Anna's work [26] and this study (CA and FCA graphs) in term of MOS shows that, while the CA method of [26] has used different codec in each step (according to Table 4.1), the proposed method uses G.711 codec for the first three points of FCA. In the last point of FCA graph (4-low rate stn), it used packet size adaptation but since the congestion is high switching to the bigger packet size is not enough and codec switches to G.729 codec. This shows that the proposed method needs less codec adaptation that results in lower cost and better MOS in overall.

Figure 4.18 presents the relation of end-to-end delay with increasing number of low-rate stations. While NA fails to provide acceptable delay for more than 1 low-rate station, apparently CA and FCA have a huge effect on delay reduction and they act excellent to diminish end-to-end delay and keep it in the acceptable range (less than 0.15 Sec).

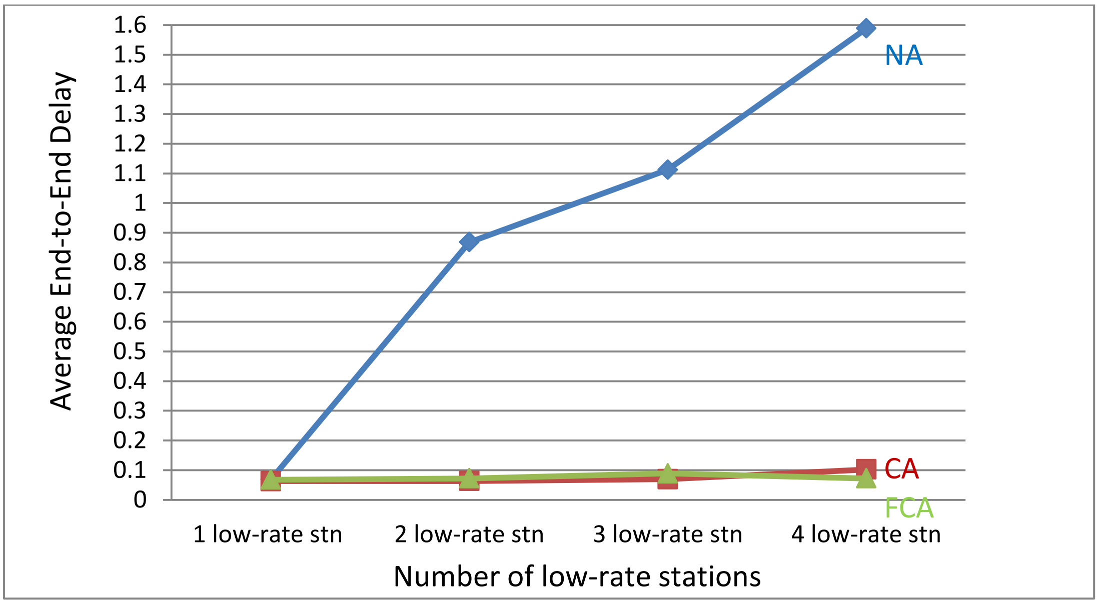



Figure 4.18: Comparison between end-to-end delay of FCA, CA and NA when number of low transmission rate stations are increased.

Figure 4.19 presents the larger scale of Figure 4.18, in order to compare CA and FCA. Since FCA uses the bigger frame size for adaptation as a preliminary choice, the first three points of FCA graph have insignificant higher end-to-end-delay compare to CA graph. This is due to longer packetization time that is taken by the packetizer.

On the other hand, the most surprising result is in the last point, when number of low rate stations reaches to 4 when the network is heavily loaded. In this point, FCA has had a lower delay compared to CA since it uses frame adaptation and codec adaptation together, while CA only uses codec adaptation. Consequently, in the higher level of congestion FCA act better than CA.

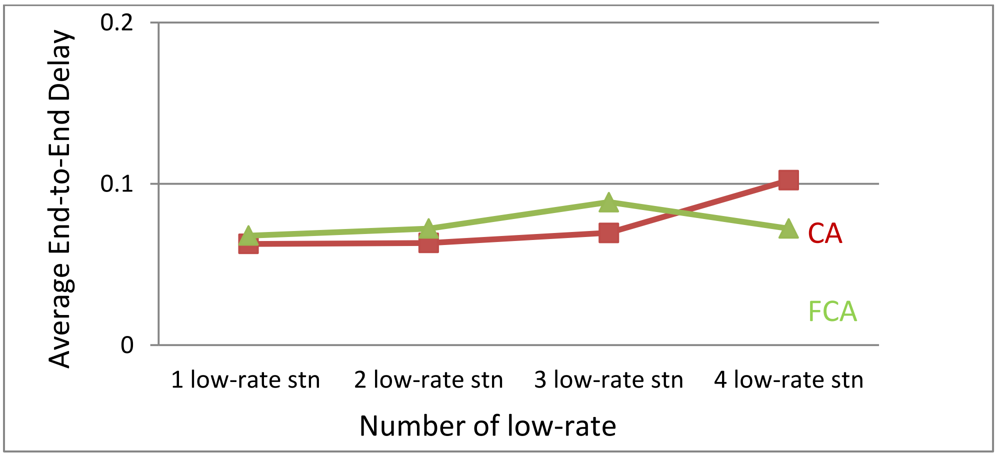



Figure 4.19: Comparison between end-to-end delay of FCA, CA in the bigger scale.

## 4.5 Discussion

Prior studies noted the importance of employing adaptation methods for VoIP over wireless networks [67], [51], [50], [108] , [81]. In reviewing the literature, a strong relationship has been reported between input rate parameters namely ‘codec’ and/or ‘packet size’ adaptation and the speech output rate. It has been found that in order to decrease the load of network in the congestion condition, switching from a higher bit-rate codec to the lower bit-rate codec and/or switching from smaller packet size to the larger packet size in the sender’s side can reduce the speech output rate.

As mentioned in the literature review, most of these studies only considered one of the mentioned parameters: only codec adaptation [43], [26], [69], [33], or only frame size adaptation [62], [49], [63]. However, there are few researchers who have studied both mentioned input rate parameters for their adaptation method [50], [20], [75]. Along with this group, this research work also considers both parameters. The most interesting finding of this study was to determine a sequence of codec and packet size adaptation. Packet size adaptation was chosen first as a more local and lower cost solution for minor congestion which was followed by codec adaptation as more expensive but effective solution.

Another important finding from studying different transmission rate reduction was that adaptation process is not necessary for every transmission rate change (Figure 4.1). This finding certified the idea of [51] that found all link adaptations do not cause to congestion. Consequently, changes in the VoIP media stream is not a precise and an adequate index to perform adaptation phase.

Hence, in order to find the right adaptation instance and to answer our first research question, 'When to make the decision of adaptation?' Some simulation scenarios were discussed in chapter 3, which showed delay variation (jitter) is the best options of congestion determination and adaptation instance, beside the track of transmission rate changes in the VoIP media stream. This finding is consistent with those of other studies that used MAC alarm ([75], [26]) and also with those only considered the quality feedback ([44] and [45], [33], [46], [47], [49], [50], [51] , [52]).

In compare to Tuysuz et al. [76], employing RTCP-XR instead of RTCP can provide faster execution time for the algorithm, because it provides more statistics about speech quality in the session and also calculated R-factor.

There are similarities between the behavior of FCA in this study and CA studied by Anna et al. ([26]) in terms of MOS. The observed difference between the behavior of CA and FCA in term of end-to-end delay in Figure 4.16 and 4.18 show that FCA method had a slight higher delay in some points, which is due to using a larger packet size and having more packetization delay. However, CA uses codec adaptation but changing codec needs to be supported by gateways and other middle hardware and in the called-party. In addition, some codecs need to be paid for their license fee. So, the benefit of using larger packet size including cost reduction and modification impact encourage us to use packet size adaptation first and if it was not enough codec adaptation can be performed next.

In purely VoIP scenario, distributed implementation of the proposed adaptation algorithm gives satisfactory performance, in addition it does not require modifications of network components such as AP and router. The performance evaluation of the proposed algorithm compare to the none-adaptive system was performed in three different congestion levels (low, moderate, and high). The results show good speech quality for adaptive system in comparison to none adaptive systems.

Also, a further comparison between the proposed algorithm and the algorithm in [26] was conducted in two main scenarios. First is a simulation scenario, in that transmission rate of one call is fallen gradually to the lower transmission rate. The second is a simulation scenario in that numbers of low-rate stations are increased in the network. In both scenarios, the proposed method was promising as it is able to improve the call quality by offering minimum packet loss, minimum end-to-end delay, and higher average MOS. These findings are significant to improve the current adaptive rate control algorithm for VoIP application.

# CHAPTER 5

# CONCLUSION

This dissertation studies the effect of transmission rate reduction of wireless channels on VoIP calls. A consequence of transmission rate reduction is lack of bandwidth for speech traffic, thereby congestion is created. The results of this study showed that congestion in the lower level leads to increasing jitter and end-to-end delay, and in the higher level it results in packet loss.

Previous related studies (which were discussed earlier in Chapter 2) revealed that the effect of transmission rate reduction in WLANs channels can be rectified using an adaptive methods such as varying coding rate, different packet size and dynamic play-out buffer. Technically, adaptive VoIP systems provide better-perceived speech quality over multi-rate wireless LANs compared to non-adaptive systems.

This research focused on Codec and Packet-size adaptation on the sender side and tried to improve previous algorithm for finding the right adaptation moment and getting faster speech quality feedback. Unlike previous studies, this research results showed that not all the transmission rate reduction require adaptation process. So those that require adaptation processes are determined by checking speech quality factors that include jitter and delay. The proposed method can also address the Ping-Pong effect, which is caused by immediate tendency of change between two subsequence transmission rates. A node that is on the border of two different transmission rates normally creates the Ping-Pong effect and its rate frequently is subjected to change . Hence, it needs adaptation process regularly. Since monitoring PDV (jitter) can indicate whether the transmission rate reduction leads to congestion or not, adaptation does not commence unless it is needed. Consequently, the Ping-Pong effect also less likely will affect the voice quality.

In another perspective, most of the previous adaptive algorithms used RTCP packet to provide feedback about the quality of call sessions, thus they need to calculate R in their algorithms. On the other hand, the RTCP-XR that was used in the proposed method provides the calculated R and MOS by the “VoIP metrics report block”, thus omitting the R calculation part.

Another major effort of this study was to compare different adaptation methods, which include codec adaptation, packet size adaptation, and combination of these two methods. The results of this study showed that packet size adaptation can only rectify normal to medium congestion, while codec adaptation can rectify medium to high congestion. A combination of these two methods can rectify very high congestion level. Taken together, the proposed method brought along the advantage of using both codec and packet size adaptation.

Eventually, to provide a clear observation of call quality perceived by each user, as well as diagnosing and isolating the problems quickly, an important practical implementation option is to use this algorithm in the VoIP software, in the network hardware chipsets of the stations or in performance analysis software. The finding of this study is strongly recommended to be considered in the industry approach of VoIP over multi-rate WLANs.

## 5.1 Future Works

This research makes several noteworthy contributions to the adaptive methods for VoIP over WLANs. Yet, a number of important limitation need to be considered as future works.

First, the present study only considered *transmission rate reduction* in wireless links (based on 802.11 standards) which cause the lack of bandwidth and create congestion. However, it is worth to investigate the effect of using adaptive methods on *transmission rate enhancement*, to use the higher bit-rate codecs and/or smaller packet size, in order to provide better speech quality and to use the bandwidth more efficiently.

Second, because VoIP is real-time and the short delay is desired to delivering speech traffic, it would be interesting to compare the recent adaptive algorithms from the time consumption point of view. It is suggested that future researches study the quality factors to find the most accurate adaptation indices and their optimum threshold.

## REFERENCES

[1] R. Beuran, "VoIP over Wireless LAN Survey," Internet Research Center Japan Advanced Institute of Science and Technology (JAIST), Research Report, 2006.

[2] P. V. Mockapetris. "Telephony's Next Act.", 2006. Available: http://spectrum.ieee.org/computing/networks/telephonys-next-act

[3] J. F. Ransome and J. W. Rittinghouse, *Voice over Internet Protocol (VoIP) Security*, Elsevier, 2005.

[4] A. D. Keromytis, "Voice over IP: Risks, Threats and Vulnerabilities," in Proceedings of the *Cyber Infrastructure Protection (CIP) Conference*, (June 2009).

[5] H. Schulzrinne, S. Casner, R. Frederick, and V. Jacobson, "RTP: A Transport Protocol for Real-Time Applications," IETF, RFC 3550, July 2003[Online]. http://www.rfc-editor.org/rfc/rfc3550.txt

[6] J. Light and A. Bhuvaneshwari, "Performance analysis of audio codecs over real-time transmission protocol (RTP) for voice services over Internet protocol," in *Proceedings of Second Annual Conference on Communication Networks and Services Research*,2004, pp. 351-356.

[7] J. Postel, "Transmission Control Protocol," IETF, RFC 0793, Sept. 1981. [Online]. Available: http://www.rfc-editor.org/rfc/rfc793.txt

[8] V. Paxson, M. Allman, J. Chu, and M. Sargent, "Computing TCP's Retransmission Timer," IETF, RFC 6298, June 2011. http://www.rfc-editor.org/rfc/rfc6298.txt

[9] S. Shankar, J. del Prado Pavon, and P. Wienert, "Optimal packing of VoIP calls in an IEEE 802.11 a/e WLAN in the presence of QoS constraints and

channel errors," in *IEEE Global Telecommunications Conference, 2004. GLOBECOM '04.*, Vol.5, pp. 2974-2980.

[10] Recommendation ITU-T G.1020 (2006), *Performance parameter definitions for quality of speech and other voiceband applications utilizing IP networks.*

[11] L. Sun, "Speech Quality Prediction For Voice Over Internet Protocol Networks," PhD thesis, School of Computing, Communications and Electronics, University of Plymouth, 2004.

[12] C. Jianguo and M. Gregory, "Performance Evaluation of VoIP Services using Different CODECs over a UMTS Network," in *Telecommunication, Networks and Applications Conference, 2008. ATNAC 2008. Australasian*, 2008, pp. 67-71.

[13] Recommendation ITU-T P.800.1 (2003), *Mean Opinion Score (MOS) terminology.*

[14] L. Carvalho, E. Mota, R. Aguiar, A. F. Lima, and J. N. de Souza, "An E-model implementation for speech quality evaluation in VoIP systems," in *10th IEEE Symposium on Computers and Communications, 2005. ISCC 2005*, pp. 933-938.

[15] S. Karapantazis and F. N. Pavlidou, "VoIP: A comprehensive survey on a promising technology," *Computer Networks*, vol. 53, pp. 2050-2090, 2009.

[16] R. J. B. Reynolds and A. W. Rix, "Quality VoIP - An Engineering Challenge," *BT Technology Journal*, vol. 19, pp. 23-32, 2001.

[17] F. D. Rango, M. Tropea, P. Fazio, and S. Marano, "Overview on VoIP: Subjective and Objective Measurement Methods," *IJCSNS International Journal of Computer Science and Network Security*, vol. 6, p. 14, 2006.

[18] Recommendation ITU-T G.107 (2009), *The E-model: a computational model for use in transmission planning.*

[19] L. Sun and E. C. Ifeachor, "Voice quality prediction models and their application in VoIP networks," *IEEE Transactions on Multimedia*, vol. 8, pp. 809-820, 2006.

[20] E. S. Myakotnykh and R. A. Thompson, "Adaptive Speech Quality Management in Voice-over-IP Communications," *Fifth Advanced International Conference on Telecommunications, AICT '09* pp.64-71, 24-28 May 2009.

[21] K. Wallace and T. Sauer. *CCNP Voice CVoice 642-437 Quick Reference (3rd Edition)*. Cisco Press, February 2011. Print.

[22] H. Y. Zhang, J. P. Homer, G. Einicke, and K. K. T. Kubik, "Performance comparison and analysis of voice communication over ad hoc network," presented at the 1st IEEE International Conference on Wireless Broadband and Ultra Wideband Communications (AusWireless'06), Sydney, Australia, March 2006, pp.1-6.

[23] S. Sahabudin and M. Y. Alias, "End-to end delay performance analysis of various codecs on VoIP Quality of Service," in *IEEE 9th Malaysia International Conference on Communications (MICC'9)*, 2009, pp. 607-612.

[24] R. S. Ramakrishnan and P. V. Kumar, "Performance Analysis of Different Codecs in VoIP Using SIP" *Proceedings of the Conference on Mobile and Pervasive Computing* (CoMPC 2008), August 07–08, 2008

[25] Recommendation ITU-T G.711 (1988), *Pulse code modulation (PCM) of voice frequencies.*

[26] A. Sfairopoulou, "A cross-layer mechanism for QoS improvements in VoIP over multi-rate WLAN networks," PhD thesis, School of Information and Communication Technologies, Universitat Pompeu Fabra, Barcelona, 2008.

[27] Recommendation ITU-T G.729 (1996), *Coding of Speech at 8 kbit/s using Conjugate Structure Algebraic Code Excited Linear Prediction (CS-ACELP).*

[28] B. Lipiczky, "Voice over WLAN," in *Information security management handbook*, ed: Auerbach Publications, 2007, pp. 145-153.

[29] Recommendation ITU-T G.114 (2003), *One-way transmission time.*

[30] K. Nisar, " A Voice Priority Queue (VPQ) Scheduler and Algorithm for VoIP over WLAN Network," PhD thesis, Computer and Information Scince Department, Universiti Technologi Petronas, Perak, Malaysia, 2011.

[31] J. S. Liu and C. H. R. Lin, "A Relay-Based MAC Protocol for Multi-Rate and Multi-Range Infrastructure Wireless LANs," *Wireless Personal Communications*, vol. 34, pp. 7-28, 2005.

[32] Jeff. (2008). *Introduction to Wi-Fi (802.11 or WiFi).* Available: http://en.kioskea.net/contents/wifi/wifiintro.php3

[33] A. Barberis, C. Casetti, J. C. De Martin, and M. Meo, "A simulation study of adaptive voice communications on IP networks," *Computer Communications*, Volume 24, Issue 9, May 2001, Pages 757-767.

[34] E. Myakotnykh, "Adaptive speech quality in Voice- over-IP communications," PhD thesis, Information Sciences and Telecommunications department, University of Pittsburgh, USA, 2008.

[35] M. F. Tüysüz and H. A. Mantar, "A cross layer QoS algorithm to improve wireless link throughput and voice quality over multi-rate WLANs," *Proceedings of the 6th International Conference on Wireless Communications and Mobile Computing*, Caen, France,2010, pp. 209-213.

[36] A. MOHD and O. L. LOON, "Performance of Voice over IP (VoIP) over A Wireless LAN (WLAN) for Different Audio/Voice Codecs," *Jurnal Teknologi © Universiti Teknologi Malaysia*, pp. 39–60, 2007.

[37] R. M. Hirannaiah, A. Jasti, and R. Pendse, "Influence of Codecs on Adaptive Jitter Buffer Algorithm," in *IEEE 66th Vehicular Technology Conference, 2007. VTC-2007 Fall. 2007*, 2007, pp. 2015-2019.

[38] S. Madhani, S. Shah, and A. Gutierrez, "Optimized Adaptive Jitter Buffer Design for Wireless Internet Telephony," in *IEEE Global Telecommunications Conference, 2007. GLOBECOM '07.*, 2007, pp. 5248-5253.

[39] L. Jing and N. Zhisheng, "An adaptive receiver buffer adjust algorithm for VoIP applications considering voice characters," in *the 2004 Joint Conference of the 10th Asia-Pacific Conference on Communications and the 5th International Symposium on Multi-Dimensional Mobile Communications Proceeding Communications*, 2004, vol.2, pp. 597-601.

[40] M. Baratvand, M. Tabandeh, A. Behboodi, and A. F. Ahmadi, "Jitter-Buffer Management for VoIP over Wireless LAN in a Limited Resource Device," in *Fourth International Conference on Networking and Services, 2008. ICNS 2008.*, 2008, pp. 90-95.

[41] T. Bially, B. Gold, and S. Seneff, "A Technique for Adaptive Voice Flow Control in Integrated Packet Networks,", *IEEE Transactions on Communications*, vol. 28, pp. 325-333, 1980.

[42] N. Yin and M. G. Hluchyj, "A Dynamic Rate Control Mechanism for Source Coded Traffic in A Fast Packet Network," *IEEE Journal on Selected Areas in Communications*, vol. 9, pp. 1003-1012, 1991.

[43] A. Trad, Q. Ni, and H. Afifi, "Adaptive VoIP Transmission over Heterogeneous WiredWireless Networks," in *Second International Workshop*

*on Multimedia Interactive Protocols and Systems (MIPS)* Grenoble, France, 2004, pp. 25-36.

[44] J. C. Bolot and A. Vega-Garcia, "Control Mechanisms for Packet Audio in the Internet," in *Proceeding of Fifteenth Annual Joint on the IEEE Computer and communication Societies. Networking the Next Generation. INFOCOM '96.*, 1996, vol.1, pp. 232-239.

[45] S. Mohamed, F. Cervantes-Perez, and H. Afifi, "Integrating Networks Measurements and Speech Quality Subjective Scores for Control Purposes," in *Proceedings of Twentieth IEEE International Conference on Computer Communications, Annual Joint Conference of the IEEE Computer and Communications Societies. (INFOCOM 2001)*, 2001, vol.2, pp. 641-649.

[46] I. Busse, B. Deffner, and H. Schulzrinne, "Dynamic QoS Control of Multimedia Applications Based on RTP," *Computer Communications,* vol. 19, pp. 49-58, 1996.

[47] T. Chunxia, L. Xiaojun, and W. Zhe, "A Research on Adaptive QoS Technology," in *International Conference on Future Computer and Communication, 2009. FCC '09.*, 2009, pp. 164-166.

[48] A. Sfairopoulou, C. Macian, and B. Bellalta, "QoS Adaptation in SIP-based VoIP calls in Multi-rate IEEE 802.11 Environments," in *International Symposium on Wireless Communication Systems, 2006. ISWCS '06. 3rd*, 2006, pp. 475-479.

[49] B. Ngamwongwattana, "Sync & Sense Enabled Adaptive Packetization VoIP," Phd thesis, University of Pittsburgh, USA, 2007.

[50] T. Kawata and H. Yamada, "Adaptive Multi-Rate VoIP for IEEE 802.11 Wireless Networks with Link Adaptation Function," in *IEEE Global Telecommunications Conference, 2006. GLOBECOM '06.*, 2006, pp. 1-5.

[51] P. McGovern, S. Murphy, and L. Murphy, "Addressing the Link Adaptation Problem for VoWLAN using Codec Adaptation," in *IEEE Global Telecommunications Conference, 2006. GLOBECOM '06.*, 2006, pp. 1-6.

[52] K. J. Koo and D. Y. Kim, "A scalable speech/audio coder control algorithm to improve the Qos of VoIP calls over WLANs," in *14th European Wireless Conference, 2008. EW 2008.*, 2008, pp. 1-6.

[53] J. Matta, C. Pepin, K. Lashkari, and R. Jain, "A source and channel rate adaptation algorithm for AMR in VoIP using the Emodel," *Proceedings of the 13th International Workshop on Network and Operating Systems Support for Digital Audio and Video,* Monterey, CA, USA, 2003, pp. 92-99.

[54] Y. Huang, J. Korhonen, and Y. Wang, "Optimization of source and channel coding for voice over IP," in *Multimedia and Expo, 2005. ICME 2005. IEEE International Conference on Computing & Processing (Hardware/Software),* 6-8 July 2005.

[55] 3GPP (3rd Generation Partnership Project) TS 26.090 (Technical Secification) *"Mandatory Speech Codec speech processing functions; Adaptive Multi-Rate (AMR) speech codec; Transcoding functions"*. Release 6, 2004.

[56] Speex codec, A Free Codec for Free Speech, Available: http://www.speex.org/

[57] Z. Bojovic, Z. Peric, V. Delic, E. Secerov, M. Secujski, and V. Senk, "Comparative Analysis of the Performance of Different Codecs in a Live VoIP Network using SIP Protocol," Journal of Electronics And Electrical Engineering (eejournal),vol.117, pp. 37-42, 2012.

[58] M. Werner and P. Vary, "Speech Quality Improvment in UMTS by AMR Mode Switching," in *ITG Conference on Source and Channel Coding (ITGSCC),* Berlin, Germany, 2002, pp. 281-284.

[59] A. Sfairopoulou, B. Bellalta, C. Macián, and M. Oliver, "A Comparative Survey of Adaptive Codec Solutions for VoIP over Multirate WLANs: A

Capacity versus Quality Performance Trade-Off," *EURASIP Journal on Wireless Communications and Networking*, 2011, pp.1-13.

[60] S. Khan, S. A. Mahmud, K. K. Loo, and H. S. Al-Raweshidy, "A cross layer rate adaptation solution for IEEE 802.11 networks," *Computer Communications*, vol. 31, iss. 8, pp. 1638-1652, 2008.

[61] S. Garg and M. Kappes, "Can I add a VoIP call?," in *IEEE International Conference on Communications, 2003. ICC '03.*, 2003, pp. 779-783 vol.2.

[62] H. Oouchi, T. Takenaga, H. Sugawara, and M. Masugi, "Study on appropriate voice data length of IP packets for VoIP network adjustment," in *Global Telecommunications Conference, 2002. GLOBECOM '02. IEEE*, 2002, vol.2, pp. 1618-1622.

[63] C. Mahlo, C. Hoene, A. Rostami, and A. Wolisz, "Adaptive Coding and Packet Rates for TCP-Friendly VoIP Flows," *Proceedings of 3rd International Symposium on Telecommunications IST2005*, Shiraz, 2005.

[64] L. Wang, M. Wu, L. Wei, and M. Li, "An adaptive network congestion control method of mobile VoIP," in *International Conference on Signal Processing Systems (ICSPS), 2010 2nd*, 2010, pp. V1-567-V1-570.

[65] A. Mukhopadhyay, T. Chakraborty, S. Bhunia, I. S. Misra, and S. K. Sanyal, "Study of enhanced VoIP performance under congested wireless network scenarios," in *Third International Conference on Communication Systems and Networks (COMSNETS), 2011*, 2011, pp. 1-7.

[66] S. Floyd, M. Handley, J. Padhye, and J. Widmer, "Equation-based congestion control for unicast applications," *SIGCOMM Computer Communication Review*, vol. 30, pp. 43-56, 2000.

[67] A. Sfairopoulou, C. Macian, and B. Bellalta, "VoIP Codec Adaptation Algorithm in Multirate 802.11 WLANs: Distributed vs. Centralized

Performance Comparison," in *Dependable and Adaptable Networks and Services*. vol. 4606, Aiko Pras and Marten van Sinderen, Eds., ed: Springer Berlin / Heidelberg, 2007, pp. 52-61.

[68] A. Sfairopoulou, C. Macian, and B. Bellalta. (2007), Dynamic measurement-based codec selection for VoIP in multirate IEEE 802.11 WLANs. *The 8th COST 290 Management Committee Meeting Universidad de M´alaga*. Available: http://www.cost290.org/td2007/tds/td07018.pdf

[69] N. Costa and M. S. Nunes, "Adaptive Quality of Service in Voice over IP Communications," in *Fifth IEEE International Conference on Networking and Services, 2009. ICNS'* 09, pp. 19-24, 2009.

[70] P. Papadimitriou and V. Tsaoussidis, "Evaluation of Transport Services for VoIP," in *IEEE International Conference on Communications, 2006. ICC '06.*, 2006, pp. 891-896.

[71] E. A. Yavuz and V. C. M. Leung, "Modeling Channel Occupancy Times for Voice Traffic in Cellular Networks," in *IEEE International Conference on Communications, 2007. ICC '07.*, 2007, pp. 332-337.

[72] P. McGovern, P. Perry, S. Murphy, and L. Murphy, "Endpoint-Based Call Admission Control and Resource Management for VoWLAN," *IEEE Transactions on Mobile Computing,* vol. 10, pp. 684-699, 2011.

[73] Recommendation ITU-T G- series (2006), *G.729-based embedded variable bit-rate coder: An 8-32 kbit/s scalable wideband coder bitstream interoperable with G.729*.

[74] S. A. A. Alshakhsi and H. Hasbullah, "Improving QoS of VoWLAN via Cross-Layer-Based Adaptive Approach," in *International Conference on Information Science and Applications (ICISA) 2011,* 2011, pp. 1-8.

[75] M. F. Tüysüz and H. A. Mantar, "Minimizing the Effects of Multi-rate WLANs by Adapting Link Adaptation and Call Admission Interfaces," Proc. *Networked Digital Technologies (NDT)*, 2010, pp. 262-273.

[76] M. F. Tüysüz and H. A. Mantar, "Evaluation of cross layer QoS aproachs for improving voice quality over multi-rate WLANs," in *International Conference on Computer Engineering and Systems (ICCES 2010)*, 2010, pp. 73-78.

[77] W. Chiapin and L. Tsungnan, "A Cross-Layer Link Adaptation Algorithm for IEEE 802.11 WLAN with Multiple Nodes," in *IEEE Asia-Pacific Services Computing Conference, 2008. APSCC '08*, 2008, pp. 1162-1167.

[78] F. Cuervo, N. Greene, A. Rayhan, C. Huitema, B. Rosen, and J. Segers, "Megaco Protocol Version 1.0," IETF, RFC 3015, Nov. 2000 [Online]. http://www.rfc-editor.org/rfc/rfc3015.txt

[79] Z. Qiao, L. Sun, N. Heilemann, and E. Ifeachor, "A new method for VoIP quality of service control use combined adaptive sender rate and priority marking," in *Communications, 2004 IEEE International Conference on*, 2004, pp. 1473-1477 Vol.3.

[80] L. Sun and E. C. Ifeachor, "Prediction of perceived conversational speech quality and effects of playout buffer algorithms," in *Communications, 2003. ICC '03. IEEE International Conference on*, 2003, pp. 1-6 vol.1.

[81] B. Tebbani and K. Haddadou, "Codec-based adaptive QoS control for VoWLAN with differentiated services," in *Wireless Days, 2008. WD '08. 1st IFIP*, 2008, pp. 1-5.

[82] W. Mazurczyk and Z. Kotulski, "Adaptive VoIP with Audio Watermarking for Improved Call quality and security," *Journal of Information Assurance and Security*, vol. 2, pp. 226-234, 2007.

[83] H. J. Kim, Y. H. Choi, J. Seok, and J. Hong, "Audio Watermarking Techniques," in *Intelligent watermarking techniques*, Lakmi C. Jain, Jeng Shyang Pan, and Hsiang Cheh Huang, Eds., ed: World Scientific, 2004.

[84] X. Liu and C. Tu, "An VoIP Application Design with Dynamic QoS Control " in *Theoretical and Mathematical Foundations of Computer Science*. vol. 164, Qihai Zhou, Ed., ed: Springer Berlin Heidelberg, 2011, pp. 605-612.

[85] E. S. Myakotnykh and R. A. Thompson, "Effect of packet size and compression variation on quality of VoIP communications," in *Proceedings of 1st International Workshop on the Security and Communication Networks (IWSCN2009)*, 2009, pp. 1-6.

[86] H. Schulzrinne, S. Casner, R. Frederick, and V. Jacobson, "RTP: A Transport Protocol for Real-Time Applications," IETF, RFC 3550, July 2003 [Online]. http://www.rfc-editor.org/rfc/rfc3550.txt

[87] T. Koistinen, "Protocol overview: RTP and RTCP," *Nokia Telecommunications* (2000).Available: https://www.netlab.tkk.fi/opetus/s38130/k99/presentations/4.pdf

[88] M. Masuda, K. Yamamoto, S. Majima, T. Hayashi, and K. Kawashima, "Performance Evaluation of VoIP QoE Monitoring Using RTCP XR," in *Management Enabling the Future Internet for Changing Business and New Computing Services*. vol. 5787, Choong Hong, Toshio Tonouchi, Yan Ma, and Chi-Shih Chao, Eds., ed: Springer, 2009, pp. 435-439.

[89] T. Friedman, R. Caceres, and A. Clark, "RTP Control Protocol Extended Reports (RTCP XR)," IETF, RFC 3611, Nov. 2003 [Online]. http://www.rfc-editor.org/rfc/rfc3611.txt

[90] M. Siddique and J. Kamruzzaman, "VoIP Call Capacity over Wireless Mesh Networks," in *IEEE Global Telecommunications Conference, 2008. IEEE GLOBECOM 2008*, 2008, pp. 1-6.

[91] R. G. Cole and J. H. Rosenbluth, "Voice over IP performance monitoring," *SIGCOMM Comput. Commun. Rev.*, vol. 31, pp. 9-24, 2001.

[92] Cisco, "Voice Over IP - Per Call Bandwidth Consumption", *Document ID: 7934*, (2006). Available: http://www.cisco.com/application/pdf/paws/7934/bwidth_consume.pdf

[93] O. Hersent, J. P. Petit, and D. Gurle, *Beyond VoIP Protocols: Understanding Voice Technology and Networking Techniques for IP Telephony*: Wiley, 2005.

[94] S. Korchi, M. Boulmalf, and A. Lakas, "Study and Design of QoS-Based VoIP over Wireless LAN," in *International Wireless Communications and Mobile Computing Conference, 2008. IWCMC '08*, 2008, pp. 547-551.

[95] Cisco, "*Cisco Codec Support FAQ*" (2005), Available: http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_qanda_item09186a00801b34cc.shtml

[96] J. V. Meggelen, L. Madsen, and J. Smith, "*Asterisk™: The Future of Telephony*," (2007) Available: http://astbook.asteriskdocs.org/en/2nd_Edition/asterisk-book-html-chunk/asterisk-CHP-8-SECT-3.html

[97] L. Wei, M. Wu, and D. Wu, "Wireless VoIP Adaptive Source Rate Control Algorithm," *5th International Conference on Wireless Communications, Networking and Mobile Computing, 2009. WiCom '09.*, 2009, pp. 1-4.

[98] Recommendation ITU-T H.323 (2006) *Packet-based Multimedia Communications Systems.*

[99] J. Rosenberg, H. Schulzrinne, G. Camarillo, C. Johnson, J. Peterson, R. Sparks, M. Handley, and E. Schooler, "SIP: Session Initiation Protocol," IETF, RFC 3261, June 2002 [Online]. http://www.rfc-editor.org/rfc/rfc3261.txt

[100] H. Schulzrinne and J. Rosenberg, "A Comparison of SIP and H.323 for Internet Telephony," in *proceedings of the International Workshop on Network and Operating System Support for Digital Audio and Video (NOSSDAV '98)*, Cambridge, England, 1998, pp. 83-86.

[101] B. S. De, P. P. Joshi, V. Sahdev, and D. Callahan, "End-to-end voice over IP testing and the effect of QoS on signaling," in *System Theory, 2003. Proceedings of the 35th Southeastern Symposium on*, 2003, pp. 142-147.

[102] J. Glasmann, W. Kellerer, and H. Moller, "Service architectures in H.323 and SIP: A comparison," *Communications Surveys & Tutorials, IEEE,* vol. 5, pp. 32-47, 2003.

[103] K. Salah and A. Alkhoraidly, "An OPNET-based simulation approach for deploying VoIP," *International Journal of Network Management*, vol. 16, pp. 159-183, 2006.

[104] C. Demichelis and P. Chimento, "IP Packet Delay Variation Metric for IP Performance Metrics," IETF, RFC 3393, Nov. 2002.[Online]. Available: www.rfc-editor.org/rfc/rfc3393.txt

[105] A. Simmonds,"*Jitter and Packet Delay Variation in the Internet - definitions,"* (2007), Available: http://www-staff.it.uts.edu.au/~simmonds/

[106] bandwidth.com. "*MOS score*", (2010), Available: http://bandwidth.com/wiki/article/MOS_score

[107] A. Sfairopoulou, B. Bellalta, and C. Macian, "How to tune VoIP codec selection in WLANs?," *Communications Letters, IEEE,* vol. 12, pp. 551-553, 2008.

[108] H. Chao, Y. Chu, and T. Tsuei, "Codec Schemes Selection for Wireless Voice over IP (VoIP)," in *Advances in Multimedia Information Processing*. vol.

2195, Heung-Yeung Shum, Mark Liao, and Shih-Fu Chang, Eds., ed: Springer Berlin / Heidelberg, 2001, pp. 622-629.

APPENDIX A

LIST OF PUBLICATIONS

| Research Title | Author's Name | Recognition |
|---|---|---|
| A Survey on Voice over IP over Wireless LANs | HaniyehKazemitabar, Sameha Ahmed, KashifNisar, Abas B Said, Halabi B Hasbullah | Conference paper in World Academy of Science, Engineering and Technology, 71, 2010 |
| Capacity Analysis of G.711 and G.729 Codec for VoIP over 802.11b WLANs | HaniyehKazemitabar, Abas Md. Said | Informatics engineering and information science book Chapter<br>From series of Communications in Computer and Information Science, 2011, Volume 253, Part 7, 519-529<br>by Springer Berlin Heidelberg |
| Performance Analysis of VoIP over Multi-Rate WLAN | HaniyehKazemitabar, Abas Md. Said | International Journal of Machine Learning and Computing (IJMLC) |
| An adaptive rate control algorithm for VoIP over multi-rate WLANs | HaniyehKazemitabar, Abas Md. Said | Journal of Procedia Information Technology and Computer Science (P-ITCS) |
| Adaptive QoS Control<br>Approaches for VoIP Over<br>Multi-Rate WLANs | HaniyehKazemitabar, Abas Md. Said | ICCIS 2012conference proceedings indexed by IEEE Explore and SCOPUS |