**DOKUZ EYLÜL UNIVERSITY**

**GRADUATE SCHOOL OF NATURAL AND APPLIED SCIENCES**

# QUALITY OF EXPERIENCE PREDICTION MODEL FOR MOBILE NETWORKS

**by**

**Zohreh SEYYEDRAHMANI**

**June, 2015**

**İZMİR**

# QUALITY OF EXPERIENCE PREDICTION MODEL FOR MOBILE NETWORKS

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## M.Sc. THESIS EXAMINATION RESULT FORM

We have read the thesis entitled **"QUALITY OF EXPERIENCE PREDICTION MODEL FOR MOBILE NETWORKS"** completed by ZOHREH **SEYYEDRAHMANI** under supervision of **ASST. PROF. DR. SEMİH UTKU** and we certify that in our opinion it is fully adequate, in scope and in quality, as a thesis for the degree of Master of Science.

Asst. Prof. Dr. SEMİH UTKU

Supervisor

Y. Doç. Dr. Reşat YILMAZ

(Jury Member)

Asst. Prof. Dr. Canan Eren ATAY

(Jury Member)

Prof. Dr. Ayşe OKUR

Director

Graduate School of Natural and Applied Sciences

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# QUALITY OF EXPERIENCE PREDICTION MODEL FOR MOBILE NETWORKS

## ABSTRACT

Rate adaptation is a general problem in streaming services where sending and receiving sides should compromise on a common rate, e.g. to minimize the re buffering failures, also called start-stop failures here. Currently, start-stop failure has the dominant effect on QoE (quality of experience) since the other types of streaming errors, such as blurring and color distortion of media frames, can be significantly avoided with the protection of the streaming channels against packet losses, e.g. using Forward Error Correction or TCP/IP protocol. In this work, the term Playing Rate Adaptation is introduced to further increase the rate adaptation capability of the streaming services by introducing a synthetic failure type, controlled fluctuations in playing rate, which in turns can reduce or eliminate the start-stop failures. Playing some video segments in negligible reduced rate is argued to provide better QoE than the start-stop failures that otherwise would be possible. As a case study, the streaming service is considered to be progressive downloading type over the 3GPP's MBMS (multimedia broadcast multicast service) as the underlying network and as worst case scenario the playing rate adaptation is applied to all frames of 2 minutes video. In order to prove the accuracy of the proposed model, objective and subjective study using DSIS (double stimulus impairment scale) and DSCS (double stimulus comparison scale) are provided. The results show that the playing rate adaptation even in worst case scenario increases the user satisfaction from the service.

**Keywords:** QoE, streaming, playing rate adaptation, MBMS, progressive download

## MOBİL NETWORKLER İÇİN DENEYİM KALİTESİ TAHMİNLEME MODELİ

## ÖZ

Hız uyarlaması video akış sürecinde ortaya çıkan verici ve alıcı olarak her iki taraf açısından da ortaklaşa uzlaşılması gereken genel bir problemdir. Tarafalar ortak bir hız uyarlaması sürecinde, örneğin yeniden bellekleme işlemini, aza indirmemesi gerekmektedir bu kavram bu çalışmada başla-dur olarak isimlendirilmektedir. Günümüzde videoların başla-dur hataları, KDK (Kullanıcı Deneyim Kalitesi)'i üzerinde diğer veri akış hataları yanında (bulanıklık, medya üzerindeki renk bozulması vb.) baskın bir etkiye sahiptir. Veri Akış hataları ileri hata düzeltme veya TCP/IP protokolü kullanılarak paket kayıplarına karşı akış kanallarının korunması ile önemli ölçüde önlenebilir. Bu çalışmada, Hız Uyarlaması Değişimleri kavramı tanıtılarak, video akış servislerin hız uyarlama yetkinliğinin artırılması hedeflenmiştir. Bu süreçte sentetik hata tipleri, hız değişimi yapılarak kontrollü dalgalanmalar sağlanmış ve başla-dur hatalarının azaltılması veya indirgenmesi hedeflenmiştir. Bazı video segmentleri üzerinde paramatrik olarak düşük oranda oynamak ile başla-dur hatalarına karşı daha iyi bir KDK sağlayacağının mümkün olacağı ileri sürülmüştür. Çalışmamızda, video yayın hizmeti ilerlemeli indirme tipi olacak şekilde 3GPP's MBMS (Çoklu Ortam Yayını Çoklu Yayın Hizmeti) ile tasarlamış olduğumuz ağ üzerinde en kötü senaryo göz önüne alınarak 2 dakikalık bir video üzerinde hız uyarlaması değiştirilerek denenmiştir. Çalışmamızın ve önerimizin doğruluğunu ispatlamak amacıyla farklı denekler üzerinde DSIS (Çift Uyarı Azalma Ölçeği) ve DSCS (Çift Uyarı Kıyaslama Ölçeği) kullanılarak nesnel ve öznel testler yapılmıştır. Elde ettiğimiz sonuçlar servislerdeki değişimlerin en kötü senaryo da bile hız uyarlamasında yapılan değişikliklerin kullanıcıları etkilemediğini göstermiştir.

**Anahtar kelimeler**: KDK, akış, hız uyarlaması, MBMS, ilerlemeli indirme

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# CHAPTER ONE
# INTRODUCTION

In this chapter, we provide a brief introduction to the concepts of Quality of Experience, Quality of Service and Rate Adaptation. Existing some researches and problems that recognized in area of Quality of Experience, lead to the motivation for performing the proposed research.

## 1.1 A General Concept of Quality of Service

Over the last, the network has been examined objectively by measuring a number of criteria to determine the network quality. This quantification is named the Quality of Service (QoS) of the network. Quality of Service (QoS) has become one of the ultimate and dominating research topics in the area of communication networks. Quality of Service for networks is an industry- comprehensive set of standards and techniques which ensures high-quality performance for network applications. QoS techniques provoke that network service providers can use available resources efficiently and ensure the required level of service without reactively widening, over-provisioning or under- provisioning their networks.

Quality of service is especially important for administrators that provide the services with high quality (such as the transportation of network traffics with high quality) and is important for users that use these provided services and are the more critical than others. Therefore, Quality of Service or QoS is a method of providing better service with high quality for selected traffic types over various types of networks. QoS is one of the most important factor for network administrators and companies that provided services.

There are some problems that many companies and administrators face them in many of times. For example, the end-users that use the provided services of the companies, say: “Our application is occasionally slow”, “Our video stream is disordered and the voice is out of sync”, “Our voice calls sound poor sometimes”, etc. For prevailing to these issues and problems, many companies try to solve these problems through the implementation of Quality of Service policies. A good QoS policy gives access preference to the applications such as voice or video. What is the good QoS policy and what is the importance of QoS is illustrated with two sample examples; in the first example, a company provides services for many of its users so that users can search in site and order their requirements from the on-line catalog. Suppose, two users use provided services of company. One of users search in sites and other user wants to download a video on an Internet movie web site, which requires considerable bandwidth resources. Without configuration QoS, a user that searched in the online catalog may encounter delays because the video download caused congestion. On the other hand, with considering QoS configuration, http packets that are sent to and from the incorporate company's web server is given preference, while other web packets could be given lower preference. Therefore QoS configuration maintains a high level of resources for user access and causes that many of users use provided services at the same time. In second example, suppose, we have two packets: one of these packets is a voice packet and the other packet is an FTP packet. Both of these packets close to router interface at the same time. Without QoS configuration, the voice packet had to wait in a queue because the FTP packet has been processed out the router interface. Therefore, this event causes that the voice path (depending on the router interface speed) encounters with an inadmissible amount of delay. The following figure describes this case:

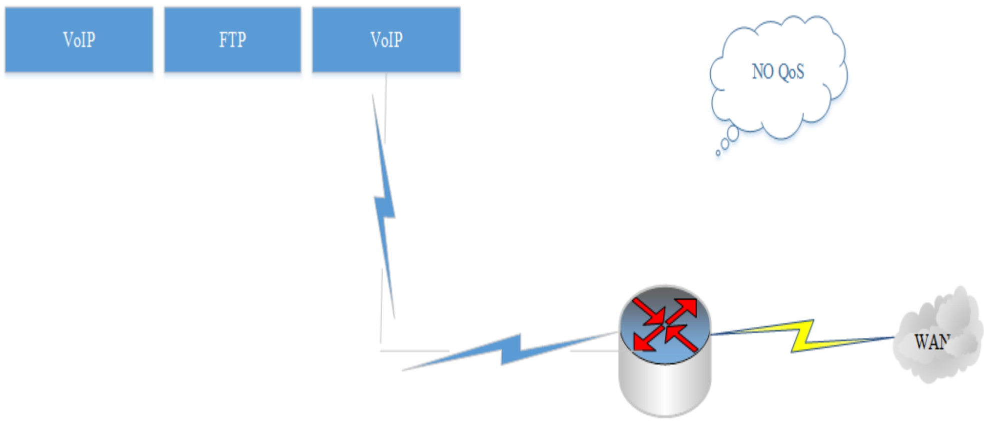



Figure 1.1 Transmission of data without QoS configuration

With considering QoS configuration, the voice packet is given priority over the FTP packet and processed first. The following figure describes this case:

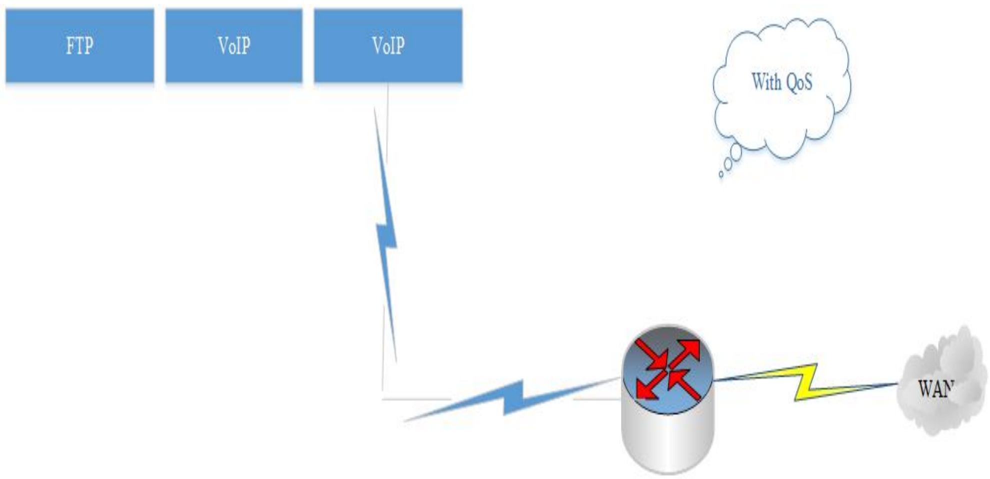



Figure 1.2 Transmission of data With QoS configuration

As a result, if the internet service providers do not consider QoS policies, in transportation of applications, each packet of application is given equal access to resources that created many problems.

- Thus we conclude that main and primary goal of QoS is to provide privileged network services for the applications by providing important parameters of QoS that include: qualified bandwidth, controlling latency and jitter, and reducing data loss. The following description illustrates these important parameters of QoS:
- Latency (Delay): The amount of time it takes a data to reach the destination after being transmitted from the source.
- Jitter (Delay variation): jitter is the variation in latency. On the other hands, the end-to-end delay difference between packets. For example, one packet is transmitted from source to the destination in 200 ms and the following packet is transmitted 300 ms then the delay variation is 100 ms.
- Loss: When the total number of packets are transmitted, maybe a number of packets were not received in destination. This event is named loss packet. If

the network is in the best situation, (highly available and non-congestion), possibility of loss existing would be zero.

- Bandwith: The rate at which traffic is carried by the network.

Also it should be added, a typical user does not involve with how a particular service is provided and how network's internal is designed, but only the most important factor that is related to users is quality of provided service. From the user's point of view, Quality of Service is represented by parameters which:

- Focus on factors which perceived by users, rather than effects within the network which are not perceivable by users.
- Network internal design is not related to user's point of view, but it is related to service providers.
- With considering user's point of view, all aspects of the service can be technically measured at the service access point.
- Are defined in network independent terms and create a common language understandable by both the user and the service provider.

QoS tries to ensure fair network resources and also manages and controls them by setting priorities for specifies type of data such as videos, voices, images ,etc. Summery, QoS is the ability of a network to provide satisfactory service for the transport of network traffic generated by applications such as video on demand, IPTV, VoIP, video streaming and also provides the following benefits:

- Control over resources hence QoS allows Internet Service Providers manage and control network resources (Optimize the use of bandwith, equipment and so on) and gives preference to resources.
- QoS provides new methods for configuring easy user interface.
- QoS improves quality of on-line applications and services thereby user experience is improved.
- Provides the best service with various methods that thereby delay and congestion event is reduced.

## 1.2 A General Concept of Quality of Experience

### *1.2.1 Concept of Quality*

From the past until now many of researchers have defined concept of quality in various article. We give the most popular definitions of quality from some literature. Guru Crosby in 1979 has stated the following: “Quality is conformance to requirements”. According to Previous definition and with considering development of the specifications and requirements, Juran, J.M in 1998 describes the following: “Quality is fitness for use” (Juran & Godfrey, 1998). The other properly accepted definition is “Quality is the degree to which performance meets expectation” (Chandrupatla, 2009). W. Edwards Deming defined quality as follows: “Good quality means a predictable degree of uniformity and dependability with a quality standard suited to the customer”. This definition describes the importance of the customer who will use the product. The definition which provides a resource to evaluate quality using a relative measure, has been adopted by the American Society for Quality (ASQ): “Quality denotes an excellence in goods and services, especially to the degree they conform to requirements and satisfy customers” (Chandrupatla, 2009). Finally, the word ‘quality’ has different meaning under different circumstances (Jain P. L., 2001). “Quality is the customers' perception of the value of the suppliers' work output. Quality is a momentary perception that occurs when something in our environment interacts with us, in the pre-intellectual awareness that comes before rational thought takes over and begins establishing order. Judgment of the resulting order is then reported as good or bad quality value” (Bin Mohd Nor, 2008). Therefore, the word "Quality" demonstrates the properties of products and/or services that are evaluated by the consumer (Porkodi, 2010). Generally, concept of Quality includes objective methods of measuring and ensuring dimensional consistency with some specific principles, for example for a product, a system or business.

### *1.2.2 Concept of Experience*

The concept of Experience is explained in the following: “experience is exposure of people to situations and the development of their skills and knowledge as a result of expose” (Watson, 1991). The general Experience concept is set of knowledge or skills of something or some event gained through involvement. Physical, mental, emotional, spiritual, religious, or social experience are types of Experience. Customer Experience is the sum of all experiences that a customer has with a supplier of goods or services over the duration of their relationship with that supplier. Alternatively, customer experience can involve subjective responses from customers to supplier via direct or undirected means ( Meyer & Schwager, 2007).

### *1.2.3 Concept of Quality of Experience*

Nowadays, the growth of internet and the telecommunication systems open excellent opportunities for multimedia services because these services are provided with the most high quality and the better than before. When the use of these services are increasing and are becoming most prevalent than the past, internet service providers and administrators encounter with the problems. Therefore, they should find efficient solution for provided services management. For example, management of Streaming multimedia services such as IP TV, Video conferencing or VoIP is very difficult because they need high resource and accurate requirements. Sufficient and efficient management of multimedia services depends on finding out what the users are satisfied from provided services, which in turn depends on the service perceived quality. The perception of multimedia services quality is related to many factors such as the multimedia fidelity and resolution, type of device, content. Traditional approaches management of network services individually focus on how transmission of media with noting to network characteristics quality such as bandwith, jitter, delay, loss rate and overlook many important factors. Thus user’s expectations from provided services were not taken into account by the network characteristic parameters. Therefore we conclude that data-centric service management cannot be as efficient with the system’s resources as user-centric management approach because it is not able to evaluate the user's perceived quality of provided services. To improve the service management, a shift from data-centric to a user-centric or user-aware multimedia service management is necessary (Agboma & Liotta, 2008).

Thus, only delivery of multimedia services to users is not sufficient, but, the most important factor for user-centric management of multimedia services is to consider the provided services quality which perceived by users. Therefore users expect multimedia services are delivered with high quality and without any difficulty and problems to them. To execute user-centric management and satisfy the user to delivered services quality, network operators need principled and accurate estimation of the perceived quality by users. For perpending of this execution and estimation, they need to take into account not only Quality of Service but also another important factor, because QoS does not consider the user's perception.

To provide better service with high quality for users and also to satisfy end-user experience, internet service providers and administrators try to improve the representation quality of multimedia content and network infrastructure respectively. Therefore should be exist an appropriate method that can evaluate the quality of provided services efficiently and reliably. By "quality" we mean Quality of Experience (QoE) (Jain, 2004), which evaluated the measure of a user's pleasure.

QoE is a comprehensive concept which applied to many topics. Therefore, should be represented a general, attractive, concise definition and appropriate for any field: "Quality of Experience is a subjective measure of customer's experiences". This short definition expresses the fact that QoE is based on measurements, in other words, many mechanisms and systems are necessary to determine what measurement methods are the most appropriate and how they can be obtained. Another important point is expressed in this definition is subject who has the most important and key role in doing the measurements. This subject is the customer/user since it is the human who is paying for a service. There is a very important point in previous expression that how difficult is the execution of a new specifications from the user's perspective has not importance not at all, but the important is whether doing the implementations and executions are useful, improved and beneficial that users can experience their perspective or not. Finally, must be attended these measurements depend on own opinion of each individual user because these measurements are subjective. Nowadays QoE has been applied in many research fields. Hence, existence of QoE in any field related to new products, technologies or processes is necessary.

Therefore, Quality of experience is a subjective measure of a user's experiences with multimedia services such as VoIP, IPTV, and Video conferencing. QoE also be named as Quality of User Experience.

Quality of Experience systems try to evaluate quality parameters which perceived by subjective users' perspective and they are different from the technical QoS parameter. For example, a person's reaction to listening to music through headphones is based not only on the frequency response of the system and the speakers, but the comfort of the unit and the individual's hearing sensitivity (Encyclopedia, n.d.).Therefore QoE plays an important role in providing multimedia services with the highest quality which caused the users be satisfied from provided services.

### 1.3 Relationship Between QoE and QoS

Most of time, concepts of QoE and QoS are confused with each other. QoS is represents of an objective system performance parameters, such as the bandwidth, delay, and loss rate of communication networks and whose main purpose is to deal with technical network aspects. However, the quality of experience (QoE), defined as "the overall acceptability of an application or service, as perceived by the end-user" by The International Telecommunication Union (ITU-T) (International telecommunication union, 2008), encompasses three important components: first, User Experience that is represents of emotions, expectations, personalization, interaction, interface, and self-efficacy. Second, Quality of Service that includes: network, jitter, packet loss, delay, and throughput. Three, context that is represents of: personal context, social context, price, content, motivation. Hence, QoE is the better method to evaluation the performance of end-to-end systems, as shown in Figure 1.3 and Figure 1.4.

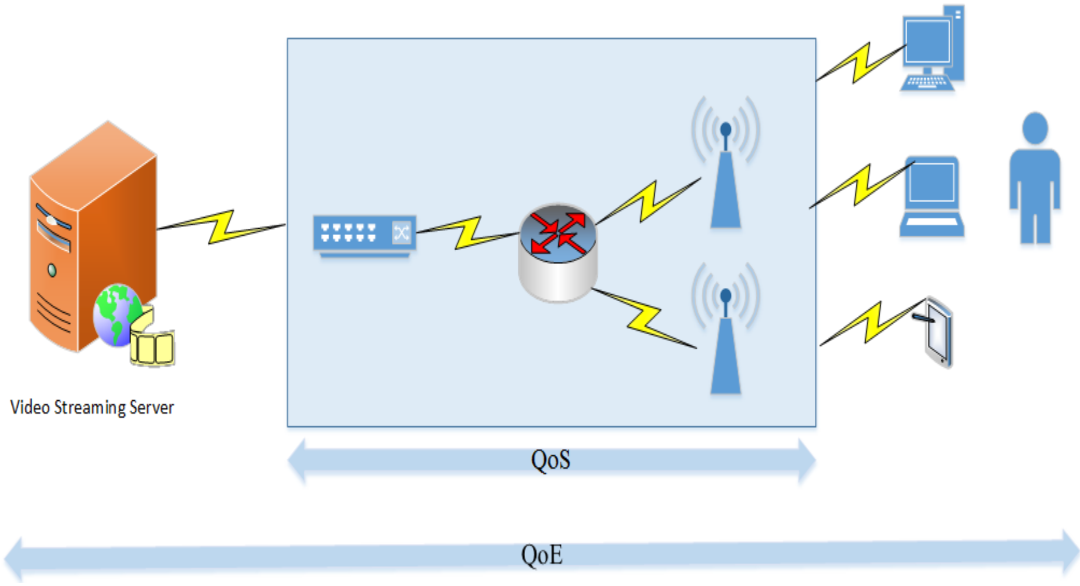



Figure 1.3 Relationship between QoE and QoS for video streaming service

QoE is an end-to-end evaluation of user satisfaction, irrespective of the network technology, while QoS measures the network performance.

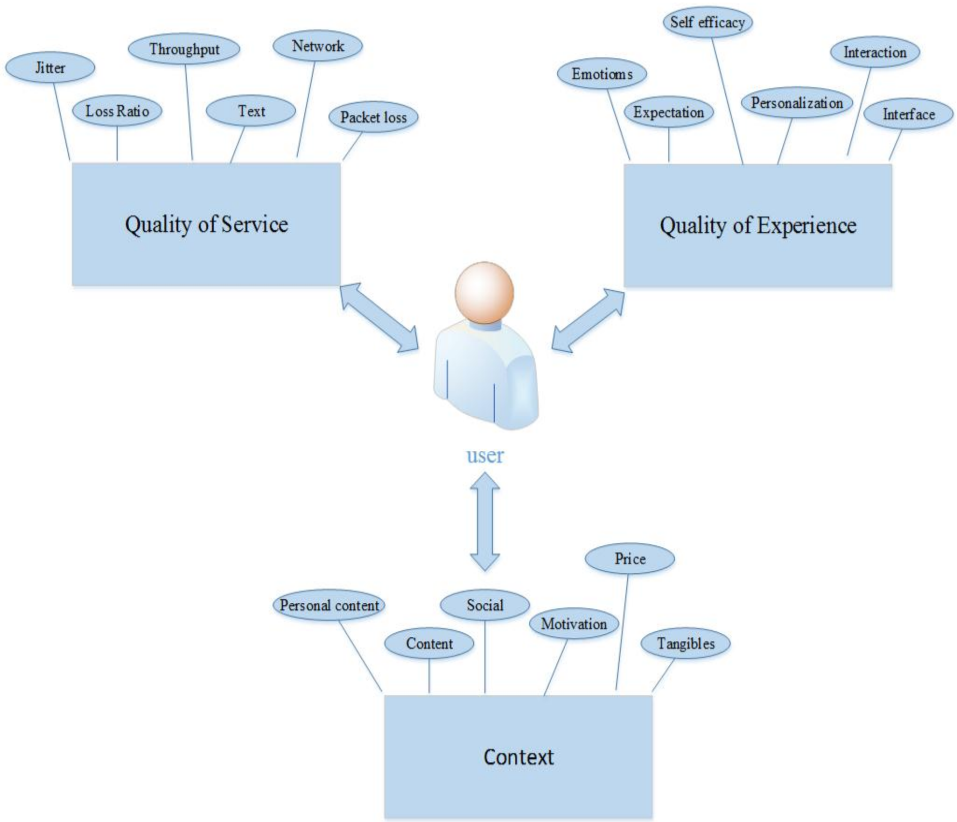



Figure 1.4 Overview of QoE

To sum up, Quality of Experience (QoE) is an important metric for business and is employed during the design and management of content delivery systems and other engineering processes. With pleasant QoE the user will adhere with service and efficiency can be earned by the service. Moreover, QoE is defined by overall experience of the users when they are accessing and using the provided services (International telecommunication union, 2008). In pleasant QoE, the user wants to stay with service and feels satisfaction in using of the service more. The ‘QoE’ is related to the perception of the user about the quality of provided services. User’s perception of a service is included with cost, reliability, availability, usability, utility and fidelity. Therefore, the users express their perception about provided services quality through the their feelings like ‘good’, ‘excellent’, ‘poor’, bad. Low QoE expresses that the user is not satisfied of the provided service quality. Waiting for user complaints to find out the level of services quality by provider services is infeasible work. Therefore many of users do not notify their dissatisfaction for the low quality service. They just leave the service and go to another one (Quality of Experience (QoE) of mobile services: Can it be measured and improved?, 2004). Therefore, QoE plays important role in the area of communication networks.

### 1.4 General Concept of Rate Adaptation

Now, should be surveyed that what has the more influence on quality of experience and finally on user’s satisfaction. At the present time, the important proposed subject is evaluation of QoE based on rate adaptation. Rate adaptation is one of the most important subject that increased QoE and improved user experience over streaming service. Playback interruptions and star-stop failure are the more current problems over video streaming that provoked by channel throughput fluctuations. The variations in rate causes video quality fluctuations and thus affects users’ Quality of Experience. Adapting video data rate during streaming can effectively reduce the consequences caused by these problems. For example, the server sends video data to the user. When this video is streamed, first, received video data is buffered at recipient side then the user receives video data. Because of the some problems that created in network such as changing the throughput of a wireless channel over time, the amount of buffered video data reduced when the channel throughput falls below the current video data rate. When user receives the all of the buffered video data, the playback process stalls that affect into users’ QoE. For solving this problem, various video rate-adaptation techniques have been proposed to reduce the risk of playback interruptions. However the variable bitrate leads to quality fluctuations, which affect users’ QoE. Therefore, Playing Rate Adaptation is a mechanisms when the available bandwith is sufficient, rate adaptation increases the sending rate quickly, and when the network becomes congested and bandwith is insufficient, decreases sending rate.

Therefore, as to improve performance of multimedia streaming transmission, we apply rate adaptation mechanism using quality of experience as indicator for rate selection.

### 1.5 Motivation

The most important goal of resources management is to find an efficient and influential solution of allocating resources justly. Internet technology is progressing day by day because to provide services to its users with better quality and more improved, contemporary and advanced features. However, because of there is the extreme competition of resources between Internet users, the dissatisfaction of Internet resources exists. The more resources are optimized, the more users are satisfied from provided resources. Therefore service providers should evaluate their services quality in an effective method and also provide an optimized solution that users can benefit from better services. As result, service providers also benefit from satisfaction of users because it causes that would increase their reputation, reliability, importance, trust and eventually their revenues.

The most important perspective for any kinds of services is Quality and service providers should evaluate their services quality expressly. Therefore, Internet Service Providers try to provide service's quality by QoS metrics. For example, with using factors such as, sufficient bandwith, minimum packet loss, minimum delay, observance in prioritize of packets, maximum throughput rate, reliable to each service can be evaluated the quality of provided services. QoS determines the service usability and utility, both of which influence the popularity of the service (Sasikaladevi & Arockiam, 2010).

Only, to be used of provided services by users is not indicative of sufficient method to conclude a fair evaluation of user's experience of the service. Therefore despite of the effects by QoS, it disregards end users' perceptions about provided services. In other words, service providers should consider the expectations of users when providing a service. Consequently, QoS is not able to provide services with a certain quality as expected by user. As a result, the most important factor to express the level of quality that users believe they have experienced is QoE. QoE is based on the user behavior while QoS is based on technical performance.

Notwithstanding difference between QoE and QoS, they are related with each other. Satisfaction of users evaluates provided services quality. Hence assessments of users about provided services quality can be evolution criterion. Thereby service providers are conducted to provide better services in order to users are satisfied.

The most important motivation of researchers is to ensure best method for evaluating QoE with the best performance and the least cost. There are three main important issues that need to be researched and answered:

1. To find a completely and comprehensive defined definition.
2. How to build a measurement model that evaluates Quality of Experience.
3. What is the best method for evaluating of QoE and user's satisfaction?

The motivation of finding the better solution for evaluating QoE causes that researchers study many methods and measurement models. The measurement model that has been attempted in the current work is related to the playing Rate Adaptation for increasing QoE over streaming service. Thus, more effort and new standards will be necessary for defining the measurement model to demonstrate the perceived quality which is experienced by end users. Rate adaptation is one of the more important subject that increased QoE and improved user experience over streaming service.

### 1.6 Problem Statement

Rate adaptation is a challenging problem in streaming services where the delivery of time critic media data over limited networks and the differences in sending and receiving device profiles, such as the processing and memory capacities, all should be considered. Particularly, for the broadcast networks or unidirectional delivery platforms the problem gets harder to be solved due to having no feedback channel to the sender side. Thus, many unwanted conditions, such as the packet losses, delays, and bandwidth limitations, may occur in these service platforms. Quality of Service (QoS) is a way of classification that manages how the unwanted conditions are controlled and mapped to the service quality. QoS could be considered as a compromise of the both sending and receiving sides on a common service quality where the rate adaptation plays an important role for regulating the fluctuations in quality. However, QoS could not reflect how the end-user experience is. At this point, the QoE describes the achieved QoS and the end-user satisfaction with the service (Yetgin & Göçer, 2015). The interaction between the QoS and QoE is shown as layers in Figure 1.5.

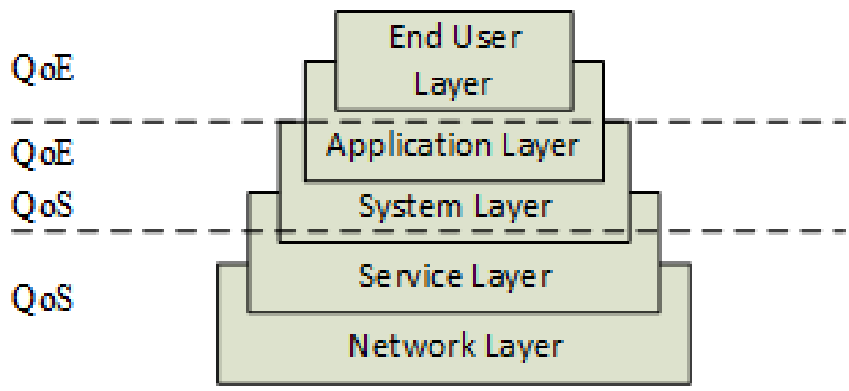



Figure 1.5 Layered approach for QoS and QoE.

At the network layer, the QoS parameters are the communication requirements, such as bandwidth, delay, jitter, loss, and reliability. System level QoS is related to operating system and processing/buffering capability of the end-user equipment. Application layer QoS are media related parameters such as media player, frame size, frame rate, media encodings as well as the player buffering method. End-user layer reflects the whole interactions between the service and end-users. At this level, QoE describes purely the degree of user satisfaction from the service as a whole. Currently, the QoE is getting an overlay over the other layers due to its increasing importance and popularity in new generation service platforms. QoE aware traffic management (Fu, Kunzmann, Wetterwald, Corujo & Costa, 2013), QoE aware service cost, QoE aware operating system (Hong, Chen, Huang, Chen & Hsu, 2014), the QoE aware error controls (Ding, Deng, Lo & Park, 2013), and QoE aware players (Yetgin & Göçer, 2015) are examples of the convergence between QoS and QoE.

### 1.7 Approach and Proposed Framework

In this research, we provide an application-layer overlay to the existing rate adaptation approaches by introducing playing rate adaptation, to the best of our knowledge, is not considered yet in the literature. The proposed adaptation provides a trade-off between the start-stop failures and the disturbance of the negligible drops in playing rate where the resulting fluctuations in playing rate can be better hided from users than that of the start-stop failures. Thus, in this work, it is claimed and proved that reducing the playing rates of some consecutive video frames (segments) in a way that the user perception is minimally affected decreases the future start-stop failures in advance and hence increases the QoE of users from the service. The proposed rate adaptation is not an alternative of the existing ones. It simply attempts to further increase the subjective quality when the start-stop failures are possible. Thus, we studied two types of quality failures, the start-stop failures versus the fluctuations in playing rate, with their effects on QoE. In the case study, the progressive download service (Yetgin & Seckin, 2008; Yetgin & Seckin, 2009; Yetgin & Çelik, 2012), which is a streaming technology using "play while download" approach, over MBMS network is considered where the channels are assumed to be packet-lossy and protected using FEC. The proposed rate adaptation is equally applicable to all types of streaming services regardless of the underlying networks. In order to prove the accuracy of the proposed model, we provide some subjective study using DSIS (double stimulus impairment scale) and DSCS (double stimulus comparison scale) as well as some objective study computing the re buffering lengths. The results show that the playing rate adaptation even in worst case scenario increases the user satisfaction from the service.

## 1.8 Thesis Outline

### Chapter One: Introduction

This chapter introduces the general concepts of Quality of Experience and Quality of Service, relationship between QoE and QoS, general concept of Rate Adaptation. And also presents motivation of our work and background for current problems involving Rate Adaptation problem in streaming service and explanation about used approach and our proposed framework in this work.

### Chapter Two: Related work

This chapter represented previous worked in evaluation of QoE field and various measurement methods of QoE.

### Chapter Three: Methods of QoE Assessment

This chapter gives an overview of existing QoE measurement studies in general. Existing QoE measurement studies are divided into subjective quality evaluation, objective quality evaluation.

### Chapter Four: Application

This chapter, represents a concise explanation about main component of proposed model. Also provides the system model and problem formulization for subjective study and objective study and demonstrates the experimental results over various MBMS link conditions.

### Chapter Five: Conclusion and Future Work

This chapter summarizes the main conclusion from the research work and proposes future work to develop the analysis of QoE over streaming service based on rate adaptation. Future work includes to discover the overall aspect of the playing rate adaptation, such as developing methods to find the suitable video segments for the playing rate adaption.

# CHAPTER TWO
# RELATED WORK

As previously mentioned, measuring Quality of Experience is one of the attractive topics in area of communication of networks and important subject in academic world. For this purpose, from past to present there are several works on the measuring QoE of multimedia streaming with different methods and researches have presented frameworks for this field.

First proposed basic framework is (Klaue, Rathke & Wolisz, 2003). EvalVid is framework and toolkit for video transmission and quality evaluation. With EvalVid not only QoS parameters such as loss rate, delay and jitter is measured but also evaluation of video quality is supported with Peak-signal-to Noise Ratio (PSNR) metric. It should be noted, EvalVid supports any kind of video codecs like MPEG, H.263, H.264 and H.26L and might be used both in real experimental set-ups and simulation experiments.

There are several works that are based on EvalVid. One of these works is utility function (Ahmed Khan & Toseef, 2011). By OPNET simulator, utility function is performed for applications such as real-time VoIP applications, non-real-time applications, FTP and video streaming. In this method, MOS value is calculated on PSNR value. Utility function assumes user satisfaction for real-time and non-real-time applications with regard to both technical and non-technical attributes. Utility function that estimates the user satisfaction for different applications, for validating proposed utility function, is compared utility-based results with the results attained from the objective measurements. Researchers have two important goals in this work: first, to develop this technique to model operator utilities and second, to find the local and global optimum solution in two levels of operator level for resource allocation and for network selection strategies (using proposed utility function) in different environments at user level.

Another work is a framework that includes a space which a 3-dimensional Eclidean space for 3-parameters (bitrate, loss and delay) used to measure quality (Venkataraman & Chatterjee, 2009). This space's name is QoE space which is used for k-dimensional for k-parameters. In steps of this measuring used PSNR, VQM and MOS metrics. In this framework, is created 18 video samples from original sample with unique combinations of the 3 parameters. First, 77 human subjects evaluate these video samples on a scale of 1 to 5 to create the QoE space. In a second set of survey, researchers choose 5 video sequence and ask 49 human subjects to rate their experience because researchers validate the accuracy of their predictions. An implementation of this framework on standard Linux PC shows, could be computed 20 MOS calculations per second with 3 parameters and 18 partitions of the QoE space. In this framework, Researches gived a lightweight, fast, scalable and efficient method for inferring the QoE of a video stream in transmission. The most important advantages of this framework is applicable for incorporating any number of parameters that can affect video quality, from network dependent to network independent.

The next framework is based on the Gradient Boosting Decision Tree (GBDT) algorithm (Chen, Yu & Xie, 2013). In this framework, is presented end-to-end QoE prediction model for video streaming service in LTE network and is developed a QoE-drive bitrate adaptation system to improve user experience. Video streaming transmitted through LTE network and simulated under NS2 platform. Video quality is measured by PSNR and MOS metrics. In this proposed QoE prediction model, with helping the gradient boosting decision tree (GBDT) algorithm, a bit rate adaptation scheme implemented at the video streaming server is proposed based on the value of estimated QoE and the feedback information of the network congestion. For example, when the load of traffic is high enough, this congestion of traffic in network causes some packets discards and transmission delay also increases. Therefore the video quality deteriorates and the QoE becomes worse. Hence, a quality adaptation mechanism plays important role to control the congestion and maintains an adequate QoE value.

EvalVid-RA (Lie & Klaue, 2008) is framework that combine with EvalVid and network simulator NS2. Evalvid-RA is for the simulation of true rate adaptive video. The solution generates real rate adaptive MPEG-4 streaming traffic, using the quantize scale for adjusting the sending rate.

The last proposed framework based on EvalVid is QoE Monitor (Saladino, Paganelli & Casoni, 2013) QoE Monitor includes network simulator NS3 that applied to QoE assessment in any simulated networks. The role of this tool is to predict the multimedia quality perceived by users through objective metrics like PSNR and SSIM. QoE Monitor is a flexible tool, usable to perform various QoE evaluation over different networks. In this framework, researches and engineers use numerical simulation tools to evaluate video and audio quality perceived by end-users and their purpose is to improve communication of network with a particular configuration and the employed codec on the quality of the received video and/or audio file.

Next framework is Open Evaluation Framework for Multimedia Over Network (Lee, Kim, Hyun & Lee, 2011). OEFMON is close to EvalVid-RA, but EvakVid-RA has some disadvantages that come from Evalvid, including limited codec support (such as VC-1 codec and ON2 codec), frame-level-only video coding that does not slice coding, etc. OEFMON combines a multimedia and network simulator QualNet. The tool applied in OEFMON for video quality measurement is MSU which asserts various video quality metrics such as MSE, PSNR and VQM. OEFMON includes several features. First, to support various types of codecs and networks, multimedia module and a network simulator are combined. Second, the multimedia module apply the network performance information which is fed back to the multimedia module in real time, into its coding algorithms. Third, to better realize the complex communication between perceived video quality and network performance transmitted video, network performance, and video quality can all be observed at the same time during simulation. Finally, because modularized structure OEFMON, it seems easily extensible.

QMON (Eckert, Knoll & Schlegel, 2013), BonaFide (Tsiaras et al., 2014) and DDIS (Bradai, Ahmed & Medjiah, 2013) are frameworks for evaluation of QoE that use MOS metric. Quality Monitoring (QMON) provides satisfactory of video quality in mobile networks. QMON be able to transparent video monitoring without the requirement to install any monitoring tools on the user's devices and on the server platform. QMON supports video codecs like H.246 (is used in mp4) and vp8 (provided by Google in WebM container).

BonaFide is an open source android application that collects essential metrics to assess QoE in mobile environments and detect traffic shaping in mobile networks. BonaFide uses client-server architecture to perform measurement test. This open source Android application, BonaFide, originally investigates traffic shaping in mobile networks, but in this work, has been used for obtaining metrics necessary to obtain service specific MOS values. Also this application support measurements across multiple servers, to prevent the affects of backbone networks becoming pronounced.

Double Stimulus Impairment Scale (DSIS) is framework for evaluating QoE under different network conditions of loss and delay. In this method videos are presented consequently in pairs. The first one is the reference, and the second one is impaired. After their playback, user with keeping in mind the first video, evaluates the second video. During this test (last up to half an hour), the user observes a series of distortion pictures or sequences randomly which original picture is included in between these pictures or sequences also to be evaluated. At the end of the series of steps, is used the opinion mean score (MOS) method for evaluating experience of users.

Profile-Based QoE Assessment Framework (PBQAF) (Serral-Graci`, Lu, Yannuzzi, Masip-Bruin & Kuipers, 2010) is framework that creates a platform for monitoring QoE multimedia service. PBQAF is deployed in Planet Lab network a set of 137 node (one server and 136 client) with sobCast P2PTV application. The applied methodology for the QoE assessment is FIXME. In this work, is presented new method to evaluate user's satisfaction of video quality. In compared to other works, this method exclusively uses the video received at destination in order to assess its quality. To perform this, researcher base their quality estimation in the analysis of the experienced frame losses and the application buffers.

In the all previous works, each framework uses different ways for evaluation QoE. For example some frameworks use subjective assessment ways or objective assessments ways and some frameworks use network simulator as well. Finally, there are some works that are relatively different with previous work, that of: use of Linux Container (LXC). This frameworks integrate LXC with network simulator NS3 for simulating several applications running in mobile network. One of this frameworks is Boxing Experience (Bustos-Jim´enez, Alonso, Fa´undez & M´eric, 2014). Boxing Experience is implemented over software NS3 and VLC and Linux Container (LXC) and surveys relation between QoS and QoE metrics for multimedia transmission. This framework describes about the distribution of video frame from a network camera to multiple client. In this proposed framework, just be attention to how Boxing Experience framework is implemented and what its performance is. Boxing Experience can easily simulate on a typical desktop computer where multiple clients are connected to one streaming server.

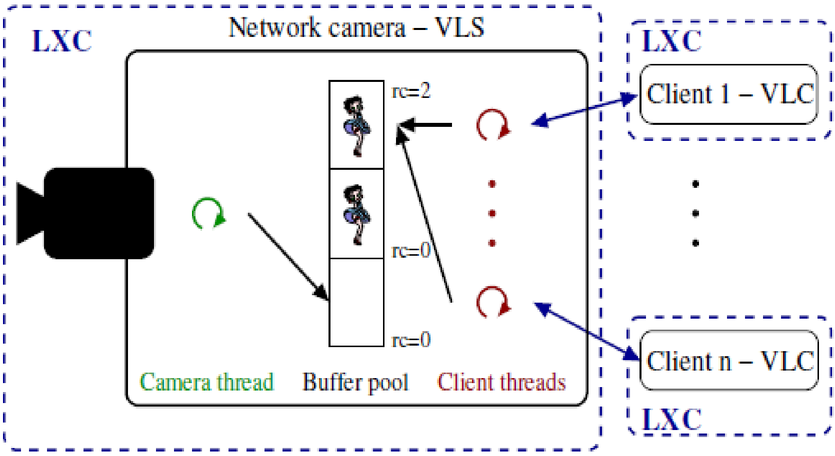



Figure 2.1 Boxing Experience in the context of video streaming from a network camera to multiple clients. The arrows between the LXCs represent the connections via NS3.

So far, all of applied methods are the same approximately. In the other words, each framework uses objective or subjective method or LXC for evaluating QoE. But these evaluation methods are not complete frameworks and each of them has advantages or disadvantages. Sometimes, these methods do not evaluate QoE precise and properly. Therefore must be exist another solutions for evaluating of QoE. As noted above, the important subject that proposed is evaluating QoE based on rate adaptation. Rate adaptation approaches for increased QoE is to enhance the user experience over streaming service. Now we ask, are there rate adaptation solutions for assessment of QoE. There are many solutions for rate adaptation problem depending on the target platform considered. One major classification could be based on whether the approaches are QoE aware (Thakolsri, Kellerer & Steinbach, 2010; Chen, Zhu & de Veciana, 2015; Piamrat, Ksentini, Bonnin & Viho, 2009), QoS aware (Kuschnig, Kofler & Hellwagner, 2010; Yu & Chao, 2007) or Non-Quality aware (Li et al., 2014; Tian & Liu, 2012).

Work (Thakolsri, Kellerer & Steinbach, 2010), proposes a Quality of Experience based rate adaptation scheme selection approach for multi-user wireless video delivery. Two used instances of rate adaptation schemes are transcoding and packet dropping, and researchers investigate impact of these instances on user perceived video quality. Generally, the aim of suitable rate adaptation scheme for each video stream is that the overall quality degradation is minimized. With allocating QoE-based resource and constrained transmission resources the proposed scheme selection approach is integrated. Simulator which used in proposed scheme is High Speed Downlink Packet Access (HSDPA). With HSDPA simulator, the QoE-based approach leads to user perceived quality much better than other approaches including a non-optimized HSDPA systems and user's perception of services quality is improved considerably. A network operator with using algorithm, can arrange and control multiple video streams with various contents and also to each video stream select an appropriate rate adaptation scheme dynamically. To achieve an optimal resource allocation that increases the user satisfaction for and network resources, this proposed algorithm is integrated with the QoE-driven network resource allocation optimization. Simulation results show that the QoE-based approach achieves a better user perceived quality.

Another work (Chen, Zhu & de Veciana, 2015) is correlated to show how rate adaptation and admission improves the QoE of video users. In this work, researchers try to find that users' QoE was strongly correlated with the empirical Cumulative Distribution Function (eCDF) of the predicted video quality. Based on this observation, for applying QoE restrictions on empirical Cumulative quality Distribution per user, a rate-adaptation algorithm is proposed. Then, users whose empirical cumulative quality distribution is not likely to satisfy their QoE constraint, is blocked by a proposed threshold-based admission control policy. Therefore, researchers make an online adaptation algorithm to automatically optimize the threshold. Simulation results display that the proposed scheme can decrease network resource deterioration by maximized rate-adaptation algorithms.

In work (Piamrat, Ksentini, Bonnin & Viho, 2009), is suggest a new Dynamic Rate-Adaptation Mechanism based on Quality of Experience, namely Q-DRAM. According to the user's feedback on QoE, researchers reduce the multicast transmission rate when users had bad QoE and they increase the multicast transmission rate when users had good QoE. Compared to existing solutions according to the IEEE 802.11 standard, simulation results display that Q-DRAM increases the wireless channel utilization and maximizes users' QoE.

In second major classification, QoS aware, work ( Yu & Chao, 2007) proposed an efficient link rate adaptation algorithm, named QoS-aware Link Rate Adaptation (Q-LRA). To specify the best mode which affects the system performance significantly, Q-LRA takes both QoS demand and channel condition into consideration. Improving system throughput and supporting QoS guarantee are the most important goals of proposed Q-LRA algorithm.

In Non-Quality aware classification, work (Li et al., 2014) shows the technology for video streaming over the Internet is based on HTTP-based adaptive streaming

(HAS). By using HTTP/TCP, HAS provides network-friendly TCP to achieve both firewall/NAT traversal and bandwidth sharing. In this work is shown that when multiple HAS clients use the provided services at a network, a limitation exist. Because of diversity in the video bitrates results, it is difficult for a client to perceive its fair-share bandwidth correctly. This video bitrate fluctuation caused undesirable behaviors that negatively impact the video viewing experience. For overcoming this limitation, researchers aim to design at the application layer using a “probe and adapt” principle for video bitrate adaptation (where “probe” mention to trial increment of the data rate, instead of sending secondary piggybacking traffic). For achievement of “probe and adapt” principle, researchers exhibit a client-side rate adaptation algorithm for HAS. This algorithm is PANDA. PANDA decrease the video bitrate fluctuations by over 75% without increasing the risk of buffer underrun.

Work (Klaue, Rathke & Wolisz, 2003) is based on Dynamic Adaptive Streaming over HTTP. DASH is widely applied for live and on-demand video streaming services. Video adaptation algorithms in existing DASH systems contain disadvantage. They are too slow to respond to changes of congestion level or too sensitive to variations of short-term network bandwidth which degrade user video experience. In this work, researchers surveys DASH through analysis and experiments for solving this problem. They display that client-side buffered video time is a good feedback signal to guide video adaptation. Another goal of researchers in this work is to provide new video rate control algorithms that balance the needs for high bandwidth utilization and adjustment of video rate. For representing that DASH designs are highly efficient in realistic network environment, with applying of comprehensive experiments on a network test bed and the Internet, researchers expand a fully-functional DASH system and measure its performance.

Majority of the related works in literature could be considered as Non-Quality aware in that they just aim to manage the rate-control without considering any QoS issue. Usually these works describes best-effort services that attempt to maximize the average quality. The QoS aware approaches take one or more of the QoS parameters into account in decision process of the rate-control algorithms. Thus, they attempt to preserve the QoS level despite of the fluctuations in network conditions. Recently, QoE aware rate adaption approaches are considered for various service platforms (Vergados et al., 2013; Hu et al., 2012; Jammeh et al., 2012). For example the used platform in work (Vergados et al., 2013), exists a hopeful solution to mobile connectivity namely the Long-Term Evolution (LTE) standard. To provide high data rates at a relatively low cost is most important feature of LTE platform. In this work researchers survey the concept of Quality of Experience for video traffic in LTE systems. Because unpredictable disruption degrades video quality and it affect on satisfaction of video user, to 1improve QoE in LTE networks, researchers proposed an adaptive video coding scheme. In the proposed model the users who perceive quality the same, are sent to groups. These groups are categorized into a number of service levels. The task of service levels is to provide different QoE satisfaction thresholds to their members. After adapting the rate of the transmitted video by the QoE driven adaptation scheme adapts, users who remain in satisfactory levels, experienced the QoE. For transmission rate adaptation, two different policies for transmission rate adaptation are surveyed, namely the adaptive and the coordinated approach. The linear slow start and/or the exponential increase adjust the level of the transmission rate for both policies. The adjusting the video resolution lead to proposed algorithm for the both transmission rate adaptation policies minimize packet loss and delay in the video transmission.

In work (Hu et al., 2012), for adapting the scalable video streams at the edge of a wireless network researchers present a proxy-based solution, which can respond quickly to highly dynamic wireless links. The technique which used for lightweight rate adaptation at the edge is scalable video coding (SVC). For providing the maximum subjective quality under appropriate rate, researchers present QoE model namely rate-quality tradeoff model. When congestion over the wireless link happens, proxy allows that rate adaptation controls the buffer level. If the proxy finds out variations in bottleneck buffer level, will reflect this difference in the throughput and delay of wireless links for all users. Technique which applied in this work is TCP-friendly rate control (TFRC). TFRC is suitable for adaptive SVC streaming and is used as a comparison rate control mechanism, as it is targeting for media streaming. When channel condition changes, the sending rate is adapted by proxy-based adaptation quickly and the resulting video playback quality is significantly improved over the TFRC scheme.

Work (Jammeh et al., 2012), studied about Network quality of service (NQoS) of IP networks and its disadvantages (such as be unpredictable and to impact the quality of networked multimedia services) and tried to find solutions for these problems. The considerable issue in voice over IP (VoIP) services is Adaptive voice and video scheme which affect to quality of experience. Traditional adaptation schemes based on NQoS do not evaluate perceived quality by users. Moreover, not only the design of adaptation schemes will be difficult because of the uncertainties inherent in NQoS parameter measurements, but the performance of adaptation schemes will not be optimal as well. To solve the optimization problem, this work presents a QoE-driven adaptation scheme for voice and video over IP which provide optimal QoE for networked voice and video applications. For implementation and testing of the adaptive VoIP, this work used NS2 and Open IMS Core network as an extensive simulation and test-bed evaluation. Results of simulation display that the scheme caused that network bandwidth be available and congestion is controlled optimally for both voice and video and delivered QoE for different network conditions is optimized. Therefore all of these consequences lead to satisfaction of users from provided services.

These approaches attempt to preserve the QoE level rather than the QoS level. Thus, they usually need a QoE measurement method by which the rate adaption might be triggered. The QoE aware rate adaption was first introduced for a traffic optimization, in which a utility function capturing the user satisfaction as a function of data rate is applied (Kelly, 1997). Later approaches further improved the utility function to measure the QoE for various service platforms (Thakolsri, Kellerer & Steinbach, 2010; Chen, Zhu & de Veciana, 2015). For example, (Thakolsri, Kellerer & Steinbach, 2010) supplies the impact of different rate adaptation techniques on the user perceived video quality where the selection of the rate adaptation scheme is based on the QoE based utility function.

Another classification of the rate adaptation approaches in literature could be based on whether the adaptation is triggered by sender-side (Kuschnig, Kofler & Hellwagner, 2010; Argyriou, 2007) receiver-side (Li et al., 2014; Tian & Liu, 2012) or network/content-centric approaches (Fu et al., 2012; Li, Chuah & Yoo, 2004). Sender-driven approaches control the transmission rate depending on the feedback from receiver or feedback from network nodes that keep track of the available bandwidth in receiver-side. For example, in work (Argyriou, 2007) is presented a mechanism for video streaming with the transmission control protocol (TCP) that uses a new rate–distortion metric for optimizing real-time Internet video streaming with the transmission control protocol. In this work, is proposed an algorithm for rate–distortion optimized mode selection (RDOMS-TCP). This algorithm specifies the encoding of each macroblocks. Researchers in this work try to display that TCP presents a viable solution for the transport of real-time encoded video bit streams.

Receiver-driven approaches usually no need a reverse channel to the sender instead they drop/increase the rate in some way with the cost of losing/gaining some kind of quality. For example, in (McCanne, Jacobson & Vetterli, 1996) the video streaming is deployed over a few channels where the primary channel provides an average quality and the others provide incremental improvement in video quality. Depending on the network conditions the receivers can choose to join or leave the channels. So the common property of the rate adaptation algorithms is their adaptation into some quality where the quality is any matter of quality degrading failures, simply called quality failures, in network, service, system, application, or end-user layers (see Figure 1.5). At this point, what makes the work different from those in literature is the introduction of a new quality metric that is degraded by the synthetic fluctuations in playing rate. Synthetic means the fluctuation is created by the algorithm itself, e.g. according to the stream-time media content, in order to reduce the future the start-stop failures. Actually, the playing rate adaptation requires finding suitable video segments in streaming where applying the playing rate adaptation has minimal effect on the perceived quality. For example, some segments may contain exciting scenes with fast actions while some others may contain stable scenes with slow actions and the user perceived quality depend on the stream-time content. Finding the suitable segments for playing rate adaptation is out of scope of the study. However, as a worst-case study, playing rate adaptation is considered to span all frames of the 2 minutes video in that worst-case test allow us to make some generalization.

# CHAPTER THREE
# METHODS OF QoE ASSESSMENT

As mentioned, Quality of Experience is related to how users perceive the quality of an application and also QoE is overall acceptability of applications or services that user perceives and experiences them. For measuring of applications quality such as video, audio, image there are subjective assessment methods and objective assessment methods.

## 3.1 Objective assessment

Objective assessment is algorithmic and mathematical method that assess the video quality without having the user test. There are some popular objective metrics to determine QoE such as: Peak-Signal-to Noise Ratio (PSNR) that is ratio between signals' maximum power and the power of the signals' noise, Mean square Error (MSE) PSNR is usually derived via MSE, Perceptual Speech Quality (PSQM) is used for voice quality measurement, Structual SIMilarity (SSIM) is a method for measuring the similarity between two images, etc.

### *3.1.1 Signal-to Noise Ratio (PSNR)*

#### *3.1.1.1 Mean square Error (MSE)*

The objective assessment method that evaluates average squared difference between an original image and a distorted image is Mean Squared Error. According to shown formula, for computing MSE, is assessed pixel-by-pixel by adding up the squared differences of all the pixels and dividing by the total pixel count.

For images A = {a1 .. aM} and B = {b1 .. bM}, where M is the number of pixels:

$$\text{MSE (A, B)} = 1/M + \sum_{i=1}^{M}(a_i - b_i)^2 \quad (3.1)$$

Peak Signal-to-Noise Ratio (PSNR) is method that evaluate the ratio between the original signal and the distortion signal in an image, with decibels measurement unit.

It is proved that if the evaluated PSNR is high, the impaired image is closer to the original image. Results of the tests and researches prove that a higher PSNR value is not always indicative a higher quality image. The advantages of PSNR is easy and fast calculation. Because of giving fair results PSNR is a popular quality metric in between objective assessment methods.

For images A = {a1 .. aM}, B = {b1 .. bM}, and MAX equal to the maximum possible pixel value (2^8 - 1 = 255 for 8-bit images):

$$\text{PSNR (A, B)} = 10\log10_{10}\left(\frac{\text{MAX}^2}{\text{MSE (A.B)}}\right) \tag{3.2}$$

The algorithm of PSNR is shown in below:

```
function PSNR = PeakSignaltoNoiseRatio(origImg, distImg)

origImg = double (origImg);
distImg = double (distImg);

[M N] = size (origImg);
error = origImg - distImg;
MSE = sum (sum (error .* error)) / (M * N);

If (MSE > 0)
   PSNR = 10*log (255*255/MSE) / log(10);
else
   PSNR = 99;
End
```

### *3.1.2 Perceptual Speech Quality (PSQM)*

Perceptual Speech Quality (PSQM) method is described in ITU-T P.861 standard. PSQM is the most appropriate objective method to evaluate the perceived quality of real-time voice transmission which compressed by compression codecs. In this method, is compared a distorted voice sample to an original voice. In this comparison, for evaluating the difference between two sample, is used a complex analytical process and viewpoint of users. With using of mean opinion score (MOS) presented by users, the conclusive distortion score is obtained.

### *3.1.3 Structural Similarity (SSIM)*

The Structural SIMilarity (SSIM) is an objective method which measuring the similarity between two images. The algorithm of SSIM evaluates difference between distorted image and original image. At evaluating subjective image quality, SSIM is a much better method than MSE or PSNR.

These objective assessment methods ignores an important factor: that of human subjectivity. Therefore objective methods are unsuitable to predict multimedia quality in term of human perception and satisfactory of users cannot be predict by traditional network metric. Hence need to be another method for evaluating QoE: that of subjective method.

## 3.2 Subjective Assessment

The subjective assessment method is evaluation of users' experience that they perceive the provided services quality. This method is applied with perception of users and is different from user to user. A common subjective assessment ways to evaluate QoE are Mean Opinion Score (MOS), DSIS (Double Stimulus Impairment Scale), DSCS (stimulus comparison scale), DSCQS (Double Stimulus Continuous Quality Scale), SCACJ (Stimulus Comparison Adjectival Categorical Judgment), SAMVIQ (Subjective Assessment Method for Video Quality evaluation).

### *3.2.1 Mean Opinion Score (MOS)*

The most popular, foundation and important of the Subjective Quality Assessment method is known as Mean Opinion Score (MOS). The MOS includes responses and answers of users who are asked for evaluating of provided services quality.

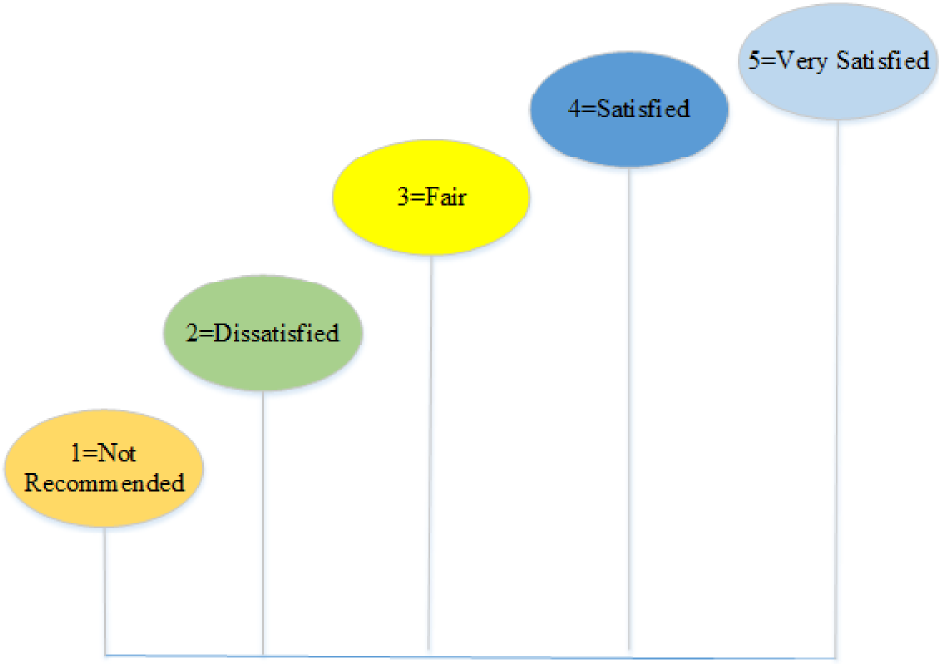



Figure 3.1 Five level scale Mean Opinion Score (MOS)

The MOS provides a numerical indication of the perceived quality of received media after compression and/or transmission. As Figure 3.1 shows, MOS is based on five point subjective scale of 1=Not Recommended, 2= Dissatisfied, 3=Fair, 4= Satisfied, 5= Very Satisfied. Result of any subjective assessment is related to MOS and a number of different scales based on MOS scores. Also subjective assessment applies two popular scales include: the five grade absolute quality rate scale and a five grade impairment scale. In five grade absolute quality: 5= Excellent, 4=Good, 3=Fair, 2=Poor, 1=Bad and in a five grade impairment scale: 5=Imperceptible, 4=Perceptible but not annoying, 3=slightly annoying, 2=annoying, 1=very annoying. Figure 3.2 shows this issue:

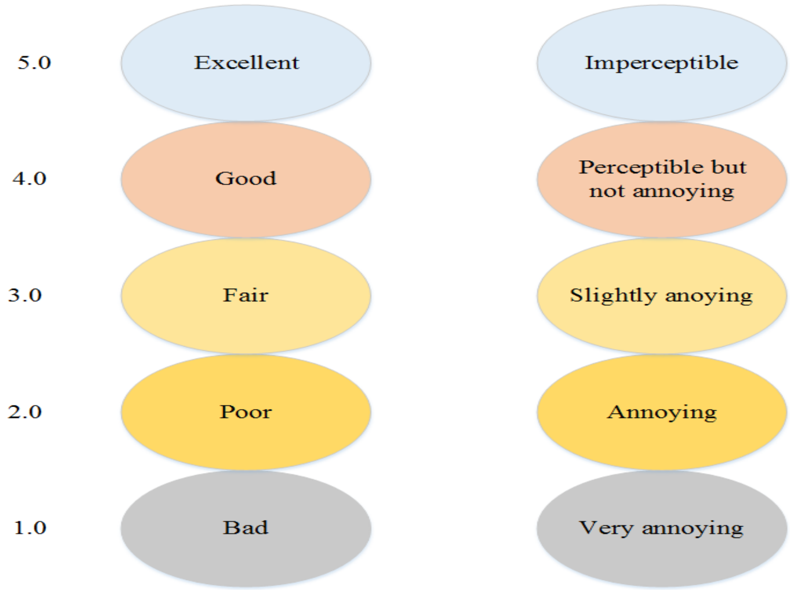



Figure 3.2 Five Grade Absolute Quality Rate Scale (Right) and Five Grade Impairment Scale (Left)

### *3.2.2 Double Stimulus Impairment Scale (DSIS)*

DSIS (Double Stimulus Impairment Scale) is described in ITU-R BT.500-11. DSIS method evaluates distortion images quality which have been transported through transportation channel. In this method, first, the users observes an unimpaired original image, then is presented the same image impaired. In second step, user evaluates the second image and keeping in mind the first. In this step (last up 30 minutes) the user observes a series of distortion pictures or sequences randomly which original picture is included in between these pictures or sequences also to be assessed. At the end of the series of steps, the users state their opinion with MOS method.

For example, the reference picture or sequence and the test picture or sequence are presented:

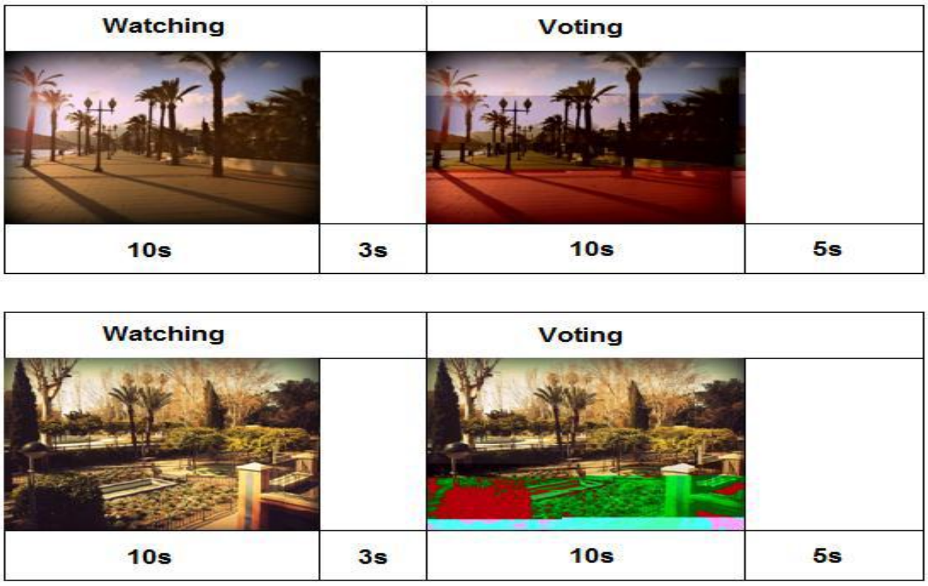



Figure 3.3 The five-grade impairment scale

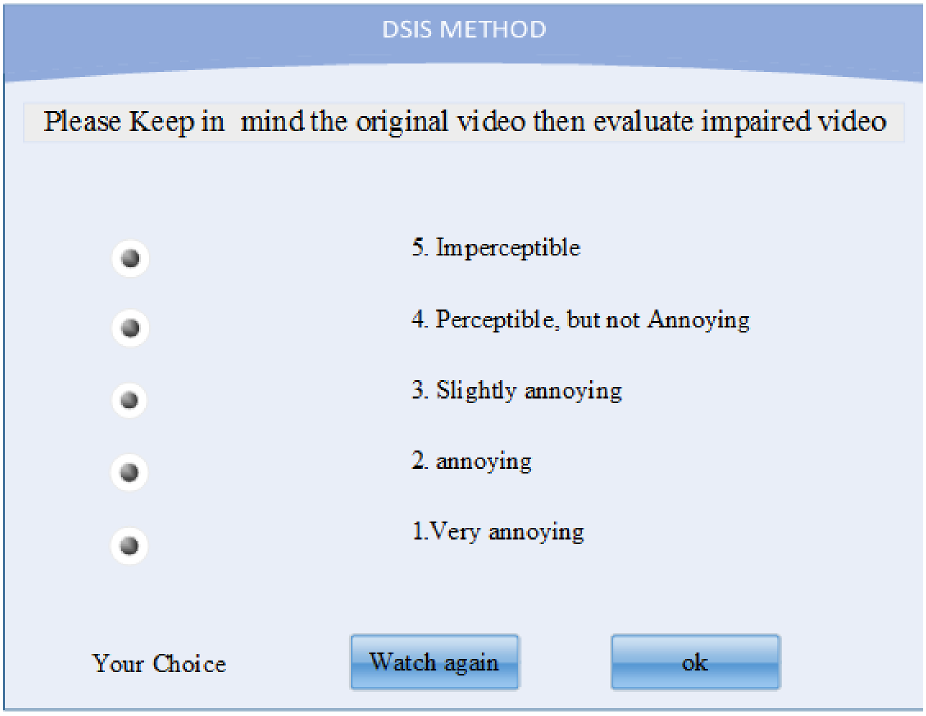



Figure 3.4 Double Stimulus Impairment Scale

### *3.2.3 DSCS (Double Stimulus Comparison Scale)*

As mentioned before, DSIS evaluates the impairment of the test sequence with respect to the original sequence which uses the MOS with 5-point scale from “very annoying” to “imperceptible”. If the test sequence is higher perceived quality than the original sequence, DSCS will be a more suitable method. Therefore it uses a comparative scale ranging from “much worse” to “much better”.

### *3.2.4 DSCQS (Double Stimulus Continuous Quality Scale)*

One of the most popular method which used for the quality evaluation of systems and transmission channel used for television broadcasts is DSCQS (Double Stimulus Continuous Quality Scale) method. DSCQS is described in ITU-R BT.500-11. This method can assess simultaneously difference in quality between an original video/image and an assessment video/image. The users are not aware of which one is the either the original or the test sequence in test duration (8-10 s). The ranges of rating score is from “bad” to “excellent”, what is equivalent to a 0 to 100 scale as it can be seen in Figure 3.5. The most important advantage of this method is effective in cases where it is not possible to present the full range of quality conditions.


Figure 3.5 Video quality assessment scale used in subjective MOS tests

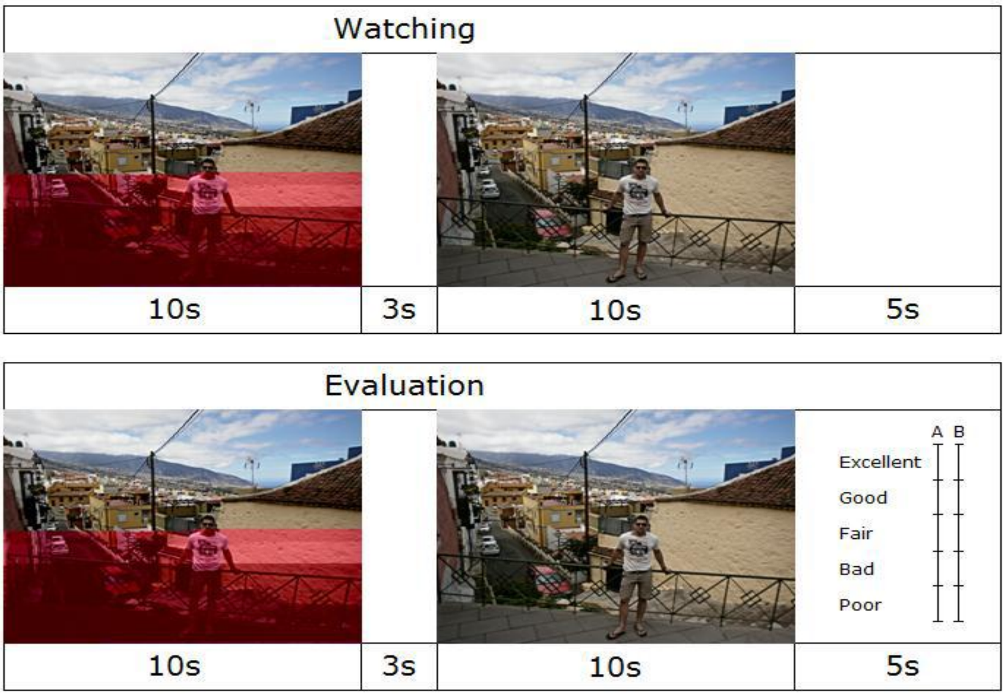



Figure 3.6 Evaluation between an original video/image and an assessment video/image

### *3.2.5 SCACJ (Stimulus Comparison Adjectival Categorical Judgement)*

The ITU-R BT.500-11 standard has described SCACJ method. In this method original image and impaired image are showed simultaneously. Then service providers ask users to give their opinion using following comparison scale:

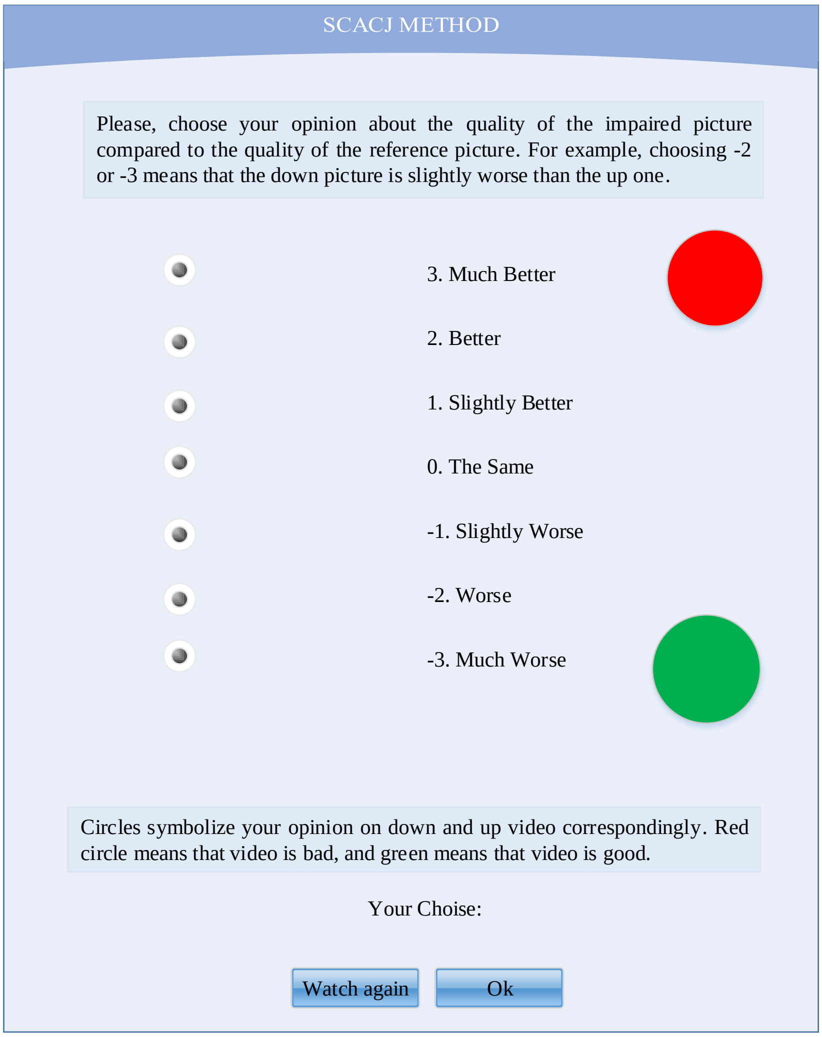



Figure 3.7 Stimulus comparison adjectival categorical judgement

### *3.2.6 SAMVIQ (Subjective Assessment Method for Video Quality evaluation)*

SAMVIQ was created by European Broadcasting Union (EBU) and was recently sent for standardization. In this method, exist two videos. One of this video is original and the other is impaired. Service providers show these videos to users and then ask users to give their opinion. The users after watching videos give mark to impaired video. Mark is in the range from 0 to 100. Figure 3.8 shows this method:


Figure 3.8 SAMVIQ (subjective assessment method for video quality evaluation)

Subjective methods are time consuming and costly but they present more accurate results, while objective methods are not time consuming relatively but their accuracy is depend on the prediction method and they focus on QoS data only. Therefore, in this research, is applied subjective method for evaluating QoE.

# CHAPTER FOUR
# APPLICATION

In this chapter, we present the main components of our model and show how they are interconnected and we show how our model works. Our proposed model relies on NS2 software, VLC media player and Linux containers. First of all, we describe about NS2 and LXC in summary.

## 4.1 Network Simulator 2

Network Simulator 2 (NS2) is an open source network simulator for Internet systems.

Network researchers use NS2 in:

1. To simulate various types of wired/wireless local and wide area networks
2. To implement network protocols such as TCP and UPD
3. Traffic source behavior such as FTP, Telnet, Web, CBR and VBR
4. Router queue management mechanism such as Drop Tail, RED and CBQ
5. Routing algorithms such as Dijkstra

NS2 is object oriented simulator that written C++ with an OTCL interpreter, which OTCL is to configure and set up a network (i.e., user fronted) and C++ is to run simulation (i.e., internal mechanism). Figure 4.1 shows NS2 Directory structure:

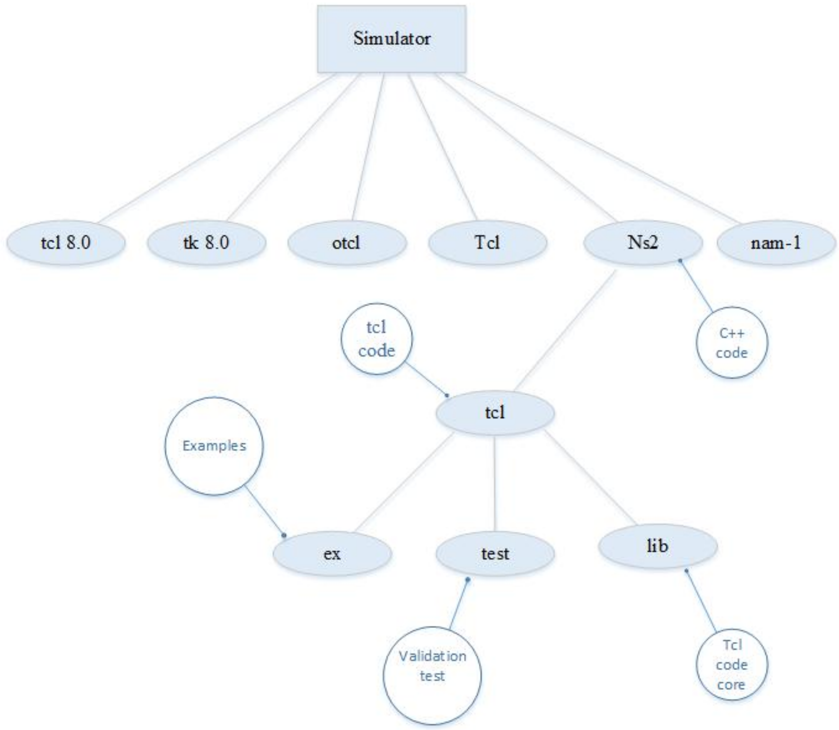



Figure 4.1 NS2 Directory structure

### 4.2 Linux Container

Linux Container (LXC) is a user interface which helped millions of users connect to the operating system of one server. For example there is one server and millions of user want to use this server. For being implemented to this work, is installed LXC in every clients which each client can use mentioned server. Therefore, LXC is a user interface that created environment as close as a standard Linux installation to use users of one operating system and one service. LXC are lightweight virtualization mechanism that does not require you set up a virtual machine on an emulation of physical hardware and provide a free software virtualization system for computers running Linux, this is accomplish through kernel level isolation allows one to run multiple virtual unit (containers) simultaneously on the same host. Figure 4.2 shows the above statement:

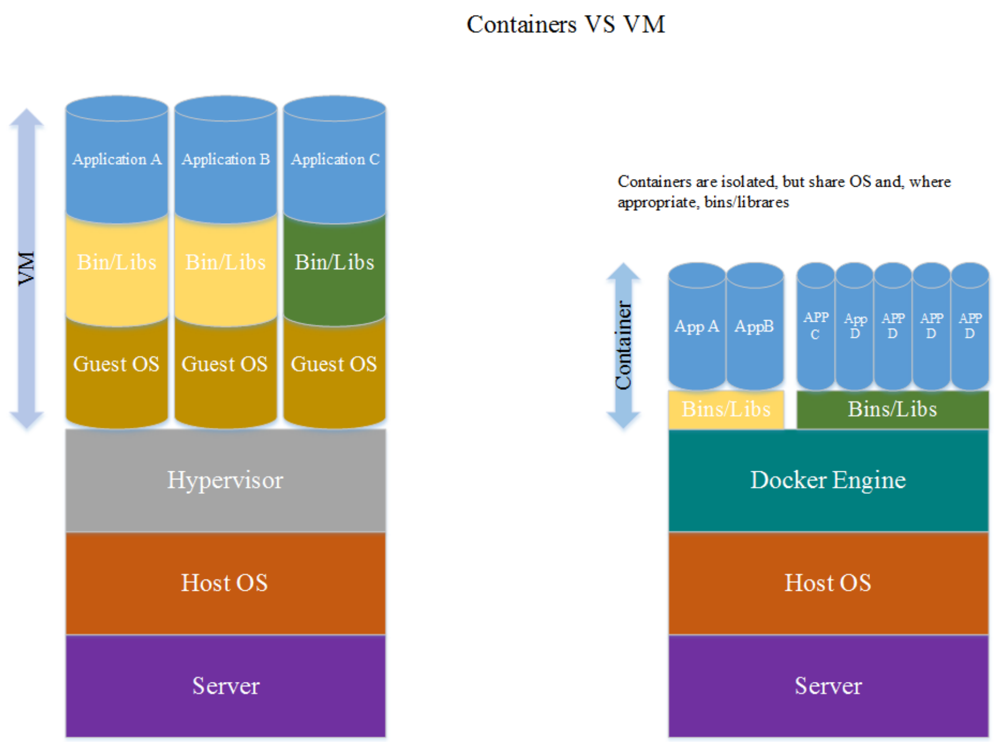



Figure 4.2 Difference between virtual machine and containers

The VLC media player is used as a server and as a client. We used LXC because: First, VLC media player cannot be directly linked to NS2. Second, the application running inside the containers can be easily changed. Third, LXC enable to monitor the resource usage (such as CPU or memory) independently for each container.

In our model, there is one LXC for client and one for server. The LXCs are connected through NS2. The client runs VLC while the server runs VLC.

### 4.3 System Model

The progressive download client is based on the model introduced in (Yetgin & Göçer, 2015), which is shown in Figure 4.3.

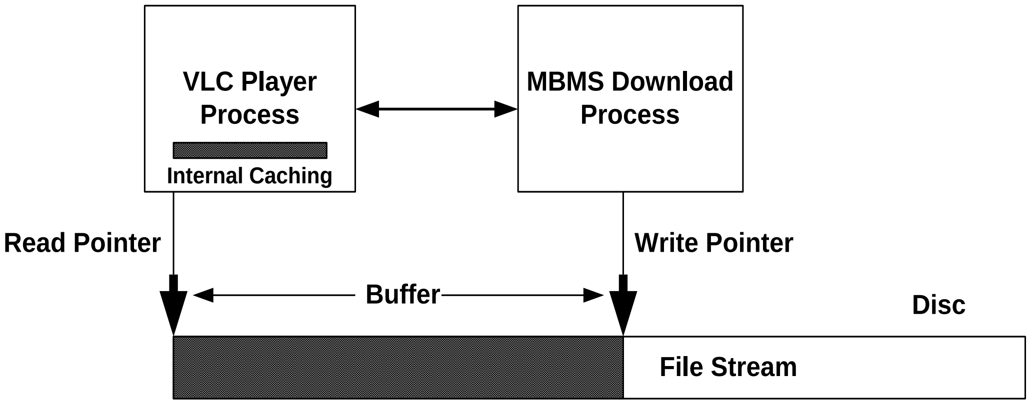



Figure 4.3 MBMS progressive download client

In the model, MBMS download client and MBMS download server introduced in (Yetgin & Seckin, 2009) are used. The progressive download client contains both the downloading process and media player process. Thus, to emulate progressive download clients, the prototype in (Yetgin & Seckin, 2009) is integrated with the VLC player's open source codes (VideoLAN, n.d.). The download services are fully protected against loss errors using FEC overheads, which are given in the experimental results section.

### *4.3.1 Buffering Model*

A QoE aware player introduced in (Yetgin & Göçer, 2015) is considered in the model where the maximum initial delay (MID) that the end-user can tolerate and a minimal blocking length (MBL) that the users prefer are subjective parameters of the buffering model. They in some sense personalize the buffering behaviors and create a value for the user expectation. The MID provides the user tolerance against the initial delay. This parameter also shows the user's tradeoff between the initial disturbance and the intermediate disturbances. For example, users who prefer higher initial delay know in advance or expect that they will see smaller number of intermediate rebuffering events.

A receiver doing a reliable progressive download is shown in Figure 4.4a and 4.4b for buffering and no-buffering cases respectively. In the figures, the receiver is assumed to have a constant media play rate, shown in thick lines, just for the visualization purpose, and the receiving rate, shown in thin lines, is assumed to be less than the media play rate in order to formulize the problem. The player starts playing media only after the time $T_d$, which is the initial delay for the progressive download. Formally, buffering occurs at time $T_i$ when the downloaded media size $F_i$ is consumed by the player, or similarly the read pointer is reached to the write pointer in Figure 4.3. The bufferings cause the end-user experience to be divided into phases, shown in Figure 4.5.

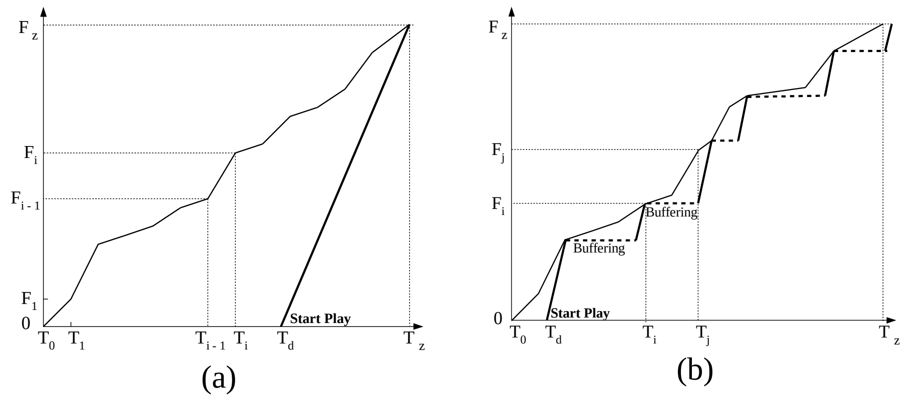



Figure 4.4 Examples of progressive download with a) no-buffering case, b) buffering cases.

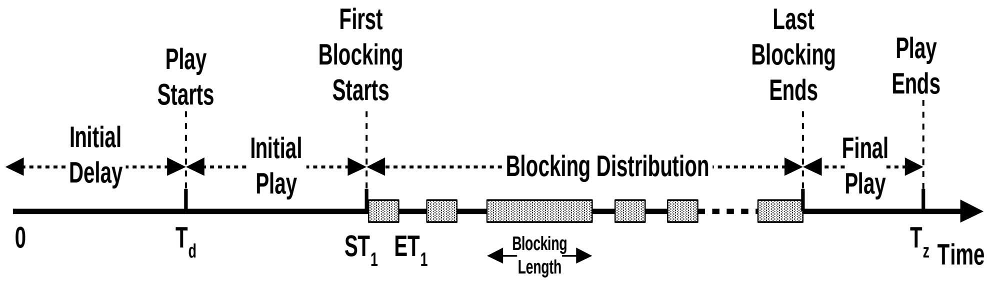



Figure 4.5 Possible phases during streaming over the progressive download

The expected initial delay predicted at Equation (4.1) is the required waiting time to have no intermediate disturbance. However, in any case the initial delay will not take longer than the MID value. Initially the player starts with the buffering state for the initial delay computed at Equation (4.2). The player decides its state later on using Equation (4.4).

$$ExpectedInitialDelay_k \approx MediaSize * \left( \frac{1}{ExpectedDownloadingRate_k} - \frac{1}{ExpectedPlayingRate_k} \right) \quad (4.1)$$

$$InitialDelay_k = \min(MID, ExpectedInitialDelay_k) \quad (4.2)$$

$$Diff_k = DownloadSize_k - (PlayedSize_k + CacheSize) \quad (4.3)$$

$State_k$:

$$\begin{aligned}
&\text{If } T_k < InitialDelay_k \\
&\quad then \quad State_k = BufferingTill(InitialDelay_k) \\
&ElseIf \text{ Player is at EOF} \\
&\quad then \quad State_k = BufferingFor(MBL) \\
&ElseIf \ (State_{k-1} = Buffering) and \ (Diff_k > MBL * MediaSize \ / \ MediaLength) \\
&\quad then \ State_k = Playing \\
&ElseIf \ (State_{k-1} = \text{Buffering}) \\
&\quad \text{then } State_k = BufferingFor(MBL)
\end{aligned} \quad (4.4)$$

Once the player switches to the playing state, it will not compute the initial delay any more. The player switches to the buffering state when it reaches to the eof of available data where the difference in downloaded data and played data becomes zero, meaning Diffk=0 (see Eq. (4.3)). The further details of the player can be found in (Yetgin & Göçer, 2015).

### *4.3.2 Problem Formulation for Objective Study*

The aim of the objective test is to compute the reduction in the total delay due to the playing rate adaptation. The total delay involves the initial or later delays (critical delay) in the critical region. The critical region is defined as the time interval between the first and last blocking shown as blocking distribution in Figure 4.5. Let's consider two identical streaming service, meaning the same network, link and buffering conditions, identified by the same configuration parameters in Table 4.1. One of the service has the playing rate adaptation enabled and the other is not. Let $CriticalDelay^{PRA}$ and $InitialDelay^{PRA}$ indicate the critical delay and the initial delay for the playing rate adaptation enabled service respectively. Similarly, CriticalDelay and InitialDelay is for the playing rate adaptation disabled service. One objective way to find how much blocking-time is prevented by the playing rate adaptation enabled service with regards to the adaptation disabled service for a particular configuration is given at Eq(4.5-4.6).

$$TimeGain(\%) = \frac{InitialDelay + CriticalDelay - (InitialDelay^{PRA} + CriticalDelay^{PRA})}{InitialDelay + CriticalDelay} \tag{4.1}$$

$$TimeGain(Sec) = InitialDelay + CriticalDelay - (InitialDelay^{PRA} + CriticalDelay^{PRA}) \tag{4.2}$$

### *4.3.3 Problem Formulation for Subjective Study*

The aim of the subjective test is to experience the quality difference from the user perspective between the two identical streaming services, identified by the same configuration parameters in Table 4.1, where one stream is the playing rate adaptation enabled and the other is not.

Two subjective tests, namely DSCS and DSIS, are considered over 20 subjects. With DSCS, subjects are presented with a pair of video streams. The order within the pair is randomized. Subjects directly rate the quality difference of the second stream from the first one on a seven point scale, -3(much worse), -2(worse), -1 (slightly worse), 0 (same), 1 (slightly better), 2 (better), 3 (much better). With DSIS, subjects are presented with a pair of video streams where the first one is the reference, and the subjects are informed about it, second one is impaired. After their playback, subjects are asked to give their opinion using five impairment scales, 5 (imperceptible), 4 (perceptible, but not annoying), 3 (slightly annoying), 2 (annoying), 1 (very annoying).

Let Comparison $Score_i$ is the score of the *i.* subject among S=20 subjects for the pair of streaming service over the same network, link and buffering conditions. The comparison scores of the subjects for a particular configuration parameter is averaged.

## 4.4 Experimental Results

The parameters used in the experiments are given in Table 4.1. Experiments are emulated on the single computer where MBMS download server and MBMS progressive download clients running together constitute the system. Each configuration involves buffering configuration, such as MBL and MID, link and bandwidth configuration considered for MBMS. The optimum values of the FEC parameters to overcome the packet losses are given at the first row of each table for the link layer losses indicated. For playing rate adaptation, the drop in playing rate is considered to be 5%, which is empirically found. For objective tests, each streaming test for a particular configuration is repeated 12 times and the results are averaged.

Table 4.1 Configuration parameters of the experiments.

| **Parameter** | **Experiment Set** |
|---|---|
| Media Source | 138 sec. of the Ice Age 3 trailer (9.46 MB) |
| Frame Resolution | 480 x 254 |
| Media Encoding | AAC+ Stereo 44100 Hz / H.264 AVC codec, 25 fps |
| FEC Codec | Reed S. Encoding-ID="129" , Instance-ID="0" |
| SDU Block Size | {800,1000} Byte |
| Symbol Length | {SDU –48}Byte |
| SB Size | {200, 214}Symbol |
| IP Packet Size | {SDU} Byte |
| PDU Block Size (RLC Block Size) | 1280 Byte |
| PDU (RLC Link Layer) Loss Rate | {0,1,5} % |
| Transmission Rate | 612 Kbps |
| VLC Player Caching Time | 0.750 sec. |
| Minimum Blocking Length | {2,4,6} sec. |
| Minimum Initial Delay | {5, 10} sec. |
| Drop in Playing Rate for playing rate adaptation | 5 % |
| Critical Delay, Blocking Frequency, Time Gain, Comparison Score | Target |

The results of the objective tests for 1% and 5% link layer losses are given in Table 4.2-4.4. For subjective test, 20 users are considered. Each user is allowed to experience the same streaming service for a particular configuration with the playing rate adaption disabled versus enabled. The comparison scores of the subjects are then averaged. The results of the subjective tests for 1% and 5% link layer losses are given in Figure 4.3, 4.4 where DSCS method is used to assess the users' QoE for comparison of the two services. The subjective test also includes DSIS study for no-loss case where no blockings occurs at all and users only assess the impairment imposed by the playing rate adaptation itself. The results for DSIS study is given in Figure 4.8.

The objective results show that the start-stop failures can be avoided significantly with the cost of negligible drop rate in playing rate where the fluctuations in the playing rate is almost hided from the users' perception(see Figure 4.6 and 4.7). The playing rate adaptation for low loss rate (*1%*) produce better objective results where 10 sec. blocking-time is prevented on the average for the initial delay of 5 sec, than that of the higher loss rate (*5%*). The main reason is the users are able to distinguish the two services having small number of start-stop failures. When the number of blockings or start-stop failures increases the ability of the users to differentiate the service quality decreases. That is, the higher packet losses put additional fluctuations and dominate the effect of the playing rate fluctuations. For small initial delays (*MID =5*), a better time gain is achieved than that of the higher initial delay (*MID=10*).

Table 4.2 Objective results with the 1% pdu losses.

| SDU Size 1000 B, SB Size 200 Symbols, FEC 8% | | | | | |
|---|---|---|---|---|---|
| Playing Rate Adaption Enabled/Disabled | MID (sec) | MBL (sec) | Average Download Rate (Kbps) | Critical Delay (sec) | Blocking Freq. |
| Disabled | 5 | 2 | 516 | *10* | *4* |
| Disabled | 5 | 4 | 516 | *8* | *2* |
| Disabled | 5 | 6 | 516 | *12* | *2* |
| Disabled | 10 | 2 | 516 | *6* | *2* |
| Disabled | 10 | 4 | 516 | *4* | *2* |
| Disabled | 10 | 6 | 516 | *6* | *1* |
| Enabled | for all | for all | 516 | *0* | *0* |

Table 4.3 Objective results with the 5% pdu losses

| SDU Size 800 B, SB Size 214 Symbols, FEC 19% | | | | | |
|---|---|---|---|---|---|
| Playing Rate Adaption Enabled/Disabled | MID (sec) | MBL (sec) | Average Download Rate (Kbps) | Critical Delay (sec) | Blocking Freq. |
| Disabled | 5 | 2 | 448 | *28* | *10* |
| Disabled | 5 | 4 | 448 | *32* | *8* |
| Disabled | 5 | 6 | 448 | *30* | *5* |
| Disabled | 10 | 2 | 448 | *24* | *8* |
| Disabled | 10 | 4 | 448 | *24* | *6* |
| Disabled | 10 | 6 | 448 | *24* | *4* |
| Enabled | 5 | 2 | 448 | *22* | *8* |
| Enabled | 5 | 4 | 448 | *24* | *6* |
| Enabled | 5 | 6 | 448 | *24* | *4* |
| Enabled | 10 | 2 | 448 | *18* | *7* |
| Enabled | 10 | 4 | 448 | *20* | *5* |
| Enabled | 10 | 6 | 448 | *18* | *3* |

Table 4.4 Time gain from playing rate adaptation

| MID | MBL | %1 PDU Loss | | %5 PDU Loss | |
|---|---|---|---|---|---|
| | | Time Gain (Sec) | Time Gain (%) | Time Gain (Sec) | Time Gain (%) |
| 5 | 2 | *10* | *67* | *6* | *18* |
| 5 | 4 | *8* | *62* | *8* | *22* |
| 5 | 6 | *12* | *71* | *6* | *17* |
| 10 | 2 | *6* | *38* | *6* | *18* |
| 10 | 4 | *4* | *29* | *4* | *12* |
| 10 | 6 | *6* | *38* | *6* | *18* |

The playing rate adaptation can be considered as a time-saving method, with the cost of negligible drop in temporal quality of the streaming, and the saved time can be used in later time, e.g. to reduce the start-stop failures. So small initial delay means that the playing rate adaptation, or the time-savings, is started early during streaming, which gains more time than that of the higher initial delay. *MID* and *MBL* pair is just a parameter of the QoE aware player. As seen in tables, their various combinations create different characteristics in fluctuations of the start-stop failures. From the experiments, one can easily discover that higher MBL reduces the number of the start-stop failures with the cost of increased blocking lengths. However, its effect on users' perception can vary since the users' experienced quality depends on highly their internal states, such as expectations and psyche. This study does not aim to discover the overall effect of the MBL and MID pair on subjective quality. So a single MBL and MID pair (MBL=2, MID=5) is chosen for the subjective tests given in Figure 5 and 6 in order to show the effect of the playing rate adaptation on subjective quality. The results are important in that we can make some generalization about the effect of the playing rate adaptation on perceived quality.

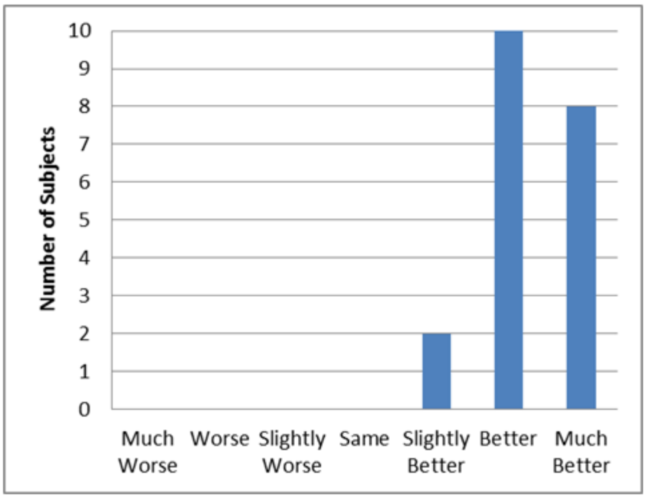



Figure 4.6 Comparison scores according to DSCS for 1% pdu losses

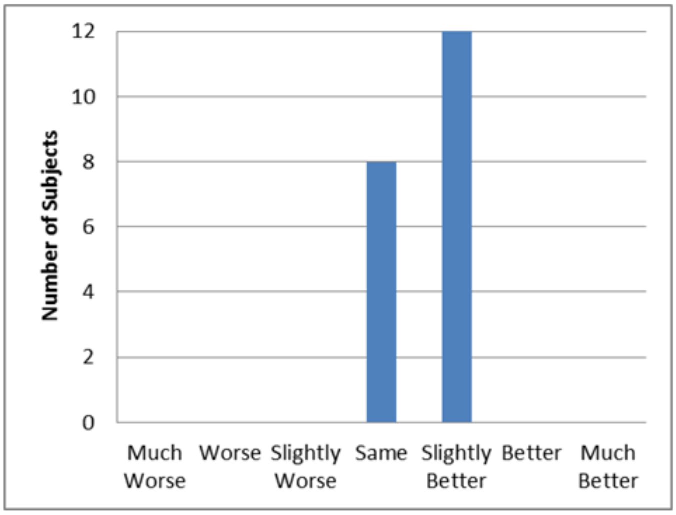



Figure 4.7 Comparison scores according to DSCS for 5% pdu losses

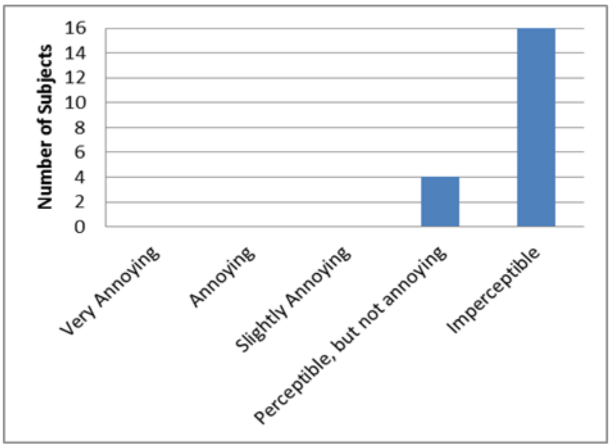



Figure 4.8 Comparison scores according to DSIS for 0% pdu losses

Generally, the playing rate adaptation can be completely hided from the users' perception when suitable adaptation rate is given. Also, the objective results conform to the subjective results where a better QoS is achieved with the low loss rate (*1%*). Figure 4.6 shows that all subjects perceived a better quality, (*Comparison Score = Better*), with the service having the playing rate adaptation. For higher loss rates, the higher number of blockings dominates and reduces the overall quality perceived by the users. However, as seen in Figure 4.7, even in higher loss rate (*5%*), users are still able to perceive slightly better QoS, (*Comparison Score = Slightly Better*), with the service having the playing rate adaptation. In order to prove that the playing rate adaptation is completely hided from the users' perception, an additional comparison test according to the DSIS method is done. The aim of the test is to remove the dominant factor caused by the losses and discover the standalone effect of the adaptation.

The results in Figure 4.8 shows that majority of the subjects are unaware of the playing rate adaptation with the *Comparison Scores = Imperceptible*. With the results, the streaming service can be better perceived by the users by enabling the playing rate where suitable in the stream-time content. Although this study only considers a worst-case test, meaning adaptation applied to all video frames of 2 minutes video, the results provide us to make following generalization; applying the playing rate adaptation to some selected segments will surely provide better achievement.

# CHAPTER FIVE
# CONCLUSION AND FUTURE WORK

As mentioned, networks transport a multitude of applications, such as real-time voice, high-quality video and delay sensitive data. Network administrators should provide satisfied, presumable, measurable, and reliability services for customers by managing bandwidth, delay, jitter and loss parameters on a network. Collection of technologies and techniques manage network resources and have the most important role for network convergence is QoS. The primary goal of QoS technologies is to transport data application such as voice, video properly and efficiently. Transporting Voice, video, and critical data applications should not be degrade in during of transportation and must be guaranteed which these data is received without any problem. Therefore, QoS is a crucial, essential and main element for successful network convergence.

QoS not only protects pleasant network traffic, but provides compatible services to undesirable traffic. Without QoS capabilities, it is impossible for a service provider to offer service to customers, who expect to be provided network services without any problem and they use these service efficiently. In communication networks, the most important subject is to be satisfied user from services; for achieving to this destination, network services providers need to take into account not only Quality of Service but also QoE should be considered because QoS does not consider the user's perception. Quality of Experience is a subjective measure of user's experiences. QoE refers to parameters from the subjective users' perspective and differ from the technical QoS parameter. We said that QoE refers the perception of the user about the quality of a specific service. This can be stated by the user's feelings such as 'good', 'excellent', 'poor', etc. for any kind of service, QoE is very important factor. User's satisfaction of a provided services is consisted: price, reliability, accessibility, fidelity, usefulness and usability. Low QoE is indicative of dissatisfaction of users from provided services. Therefore, QoE plays important role in the area of communication networks. Then we expressed relationship between QoE and QoS which QoE covers QoS. We argue about method of measuring QoE which includes subjective assessment and objective assessment. In our thesis, we use DSIS (double stimulus impairment scale), DSCS (double stimulus comparison scale) and MOS (mean opinion score) subjective assessment method for evaluating QoE. Finally, in our work, playing rate adaption is introduced as proposed method to increase the QoE of the streaming services. The proposed adaptation is an application layer overlay of the existing rate adaptation methods. So, it can be applied to any type of streaming services regardless of the underlying network. The playing rate adaption introduces new quality failure, namely synthetic fluctuations in playing rate, which degrades the temporal characteristics of the streaming video. Thus, the proposed method aims to convert one failure type to another one where the effect on QoE is better with a tradeoff between the two failure types. The objective and subjective results show that playing rate adaptation provides better QoE achievements even in worst-case tests. The work also provides the case study that shows the objective/subjective analysis of the model for various MBMS link conditions. The work opens new research door in the area of QoE over streaming. Further study is needed to discover the overall aspect of the playing rate adaptation, such as developing methods to find the suitable video segments for the playing rate adaption.

## REFERENCES


Agboma, F. & Liotta, A. (2008). QoE-aware QoS management. *MoMM '08 Proceedings of the 6th International Conference on Advances in Mobile Computing and Multimedia*, 111-116.

Ahmed Khan, M. & Toseef, U. (2011). User utility function as quality of experience(QoE). *The Tenth International Conference on Networks*, 99-104.

Argyriou, A. (2007). Real-time and rate–distortion optimized video streaming with TCP. *Signal Processing: Image Communication, 22*(4), 374–388.

Bin Mohd Nor, K. A. (2008). *Quality improvement using classical method concept in injection molding manufacturing*. Master's Thesis, Universiti Teknikal Malaysia Melaka.

Bradai, A., Ahmed, T. & Medjiah, S. (2013). QoE assessment for SVC live distribution and adaptation: ENVISION case study. *Electronics, Circuits, and Systems (ICECS)*, 333 - 336.

Bustos-Jim´enez, J., Alonso, R., Fa´undez, C. & M´eric, H. (2014). Boxing Experience: Measuring qos of multimedia streaming using NS3, LXC and VLC. *8th Workshop on Network Measurements*, 659-662. Edmonton: WNM.

Chandrupatla, T. R. (2009). *Quality and reliability in engineering.* New York: Cambridge University Press.

Chen, C., Zhu, X. & de Veciana, G. (2015). Rate adaptation and admission control for video transmission with subjective quality constraints. *IEEE Journal of Selected Topics in Signal Processing, 9*(1), 22-36.

Chen, H., Yu, X. & Xie, L. (2013). End-to-end quality adaptation scheme based on qoe prediction for video streaming service in LTE networks. *First International Workshop on Wodelling, Measurements and Optimization over Wireless Networks*, 627-633.

Ding, J.-W., Deng, D.-J., Lo, Y.-K. & Park, J. H. (2013). Perceptual quality based error control for scalable on-demand streaming in next-generation wireless networks. *Telecommun Systems, 52*, 445–459.

Eckert, M., Knoll, T. M. & Schlegel, F. (2013). Advanced MOS calculation for network based QoE estimation of tcp streamed video services. *Signal Processing and Communication Systems*, 1 - 9.

*Encyclopedia*, (n.d.). Retrieved May 29, 2015, from pcmag: http://www.pcmag.com/encyclopedia/term/57607/qoe#

Fu, B., Kunzmann, G., Wetterwald, M., Corujo, D. & Costa, R. (2013). QoE-aware traffic management for mobile video delivery. *IEEE International Conference on Communication*, 652-656.

Fu, T., Li, Y., Lin, T., Tan, H., Tang, H. & Ci, S. (2012). An effective congestion control scheme in content-centric networking. *Parallel and Distributed Computing, Applications and Technologies*, 245 - 248.

Hong, H.-J., Chen, D.-Y., Huang, C.-Y., Chen, K.-T. & Hsu, C.-H. (2014). Placing virtual machines to optimize cloud gaming experience. *Proceedings of Annual Workshop on Network and Systems Support for Games. 3*, 42-53.

Hu, H., Zhu, X., Wang, Y., Pan, R., Zhu, J. & Bonomi, F. (2012). QoE-based multi-stream scalable video adaptation over wireless networks with proxy. *Communications*, 7088 - 7092.

*International telecommunication union*. (2008). Retrieved May 29, 2015, from QoE challenge tackled in new ITU-T REC: http://www.itu.int/ITU-T/newslog/QoE+Challenge+Tackled+In+New+ITUT+Rec.aspx

Jain, P. L. (2001). *Quality control and total quality management.* New Delhi: Tata McGraw.

Jain, R. (2004). Quality of experience. *IEEE Multimedia*, *11*(1), 95-96.

Jammeh, E., Mkwawa, I., Khan, A., Goudarzi, M., Sun, L. & Ifeachor, E. (2012). Quality of experience (QoE) driven adaptation scheme for voice/video over IP. *Telecommunication Systems, 49*(1), 99-111.

Juran, J. M. & Godfrey, B. (1998). *Juran's quality handbook* (5th ed.). New York: McGraw-Hill Professional.

Kelly, F. (1997). *European Transactions on Telecommunications, 8*(1), 33-37.

Klaue, J., Rathke, B. & Wolisz, A. (2003). EvalVid - A framework for video transmission and quality evaluation. *13th International Conference on Modelling Techniques and Tools for Computers Performance*, 255-272. Illinois.

Kuschnig, R., Kofler, I. & Hellwagner, H. (2010). An evaluation of tcp-based rate-control algorithms for adaptive internet streaming of H.264/SVC. *MMSys '10 Proceedings of the First Annual ACM SIGMM Conference on Multimedia Systems*, 157-168.

Lal, C., Laxmi, V. & Gaur, M. S. (2012). A rate adaptive and multipath routing protocol to support video streaming in MANETs. *ICACCI '12 Proceedings of the International Conference on Advances in Computing, Communications and Informatics*, 262-268.

Lee, C., Kim, M., Hyun, S. J. & Lee, S. (2011). OEFMON: An open evaluation framework for multimedia over networks. *Communications Magazine, 49*(9), 153 - 161.

Li, D., Chuah, C.-N. & Yoo, S. .. (2004). Proxy-driven rate-distortion optimized video streaming wireless network using asynchronous clocks. *IEEE Packet Video Workshop*.

Li, Z., Zhu, X., Gahm, J., Pan, R., Hu, H., Begen, A. C. & Oran, D. (2014). Probe and adapt: rate adaptation for http video streaming at scale. *IEEE Journal On Selected Areas In Communications, 32*(4), 719-733.

Lie, A. & Klaue, J. (2008). Evalvid-RA: Trace driven simulation of rate adaptive mpeg-4 vbr video. *Multimedia Systems, 14*(1), 33-50.

McCanne, S., Jacobson, V. & Vetterli, M. (1996). Receiver-driven layered multicast. *SIGCOMM '96 Conference proceedings on Applications, Technologies, Architectures, and Protocols for Computer Communications*, 117-130.

Meyer, C. & Schwager, A. (2007). Understanding customer experience. *Harvard Business Review, 85*(2), 116-120.

Piamrat, K., Ksentini, A., Bonnin, J.-M. & Viho, C. (2009). Q-DRAM: QoE-based dynamic rate adaptation mechanism for multicast in wireless networks. *Global Telecommunications Conference*, 1 - 6.

Porkodi, S. (2010). *Patient Care Services and Hospitals.* New Delhi: Excel Books.

(2004). *Quality of experience (QoE) of mobile services: Can it be measured and improved?* White paper.

Saladino, D., Paganelli, A. & Casoni, M. (2013). A tool for multimedia quality assessment in NS3: QoE monitor. *Simulation Modelling Practice and Theory, 32*, 30-41.

Sasikaladevi, N. & Arockiam, L. (2010, April). Relaibility evaluation model for composite web services. *International Journal of Web. & Semantic Technologi, 1*(2), 16-22.

Serral-Graci`, R., Lu, Y., Yannuzzi, M., Masip-Bruin, X. & Kuipers, F. A. (2010). Packet Loss based quality of experience of multimedia video flows. In E. Osipov, A. Kassler. & T. M. Bohnert, *Wired/Wireless Internet Communications: 8th International Conference, WWIC*, 180-191.

Thakolsri, S., Kellerer, W. & Steinbach, E. (2010). QoE-based rate adaptation scheme selection for resource-constrained wireless video transmission. *Proceedings of the 18th International Conference on Multimedea*, 783-786.

Tian, G. & Liu, Y. (2012). Towards agile and smooth video adaptation in dynamic HTTP streaming. *CoNEXT '12 Proceedings of the 8th International Conference on Emerging Networking Experiments and Technologies*, 109-120.

Tsiaras, C., Sehgal, A., Seeber, S., Dönni, D., Stiller, B., Schönwälder, J. & Rodosek, G. D. (2014). Towards evaluating type of service related Quality-of-Experience on mobile networks. *Wireless and Mobile Networking Conference (WMNC)*, 1-8.

Venkataraman, M. & Chatterjee, M. (2009). Evaluating quality of experience for streaming video in real time. *Global Telecommunications Conference*, 1 - 6.

Vergados, D. J., Sgora, A., Michalas, A., Vergados, D. D., Laulajainen, J.-P. & Jiang, Y. (2013). A qoe-driven adaptation scheme for video content delivery in lte networks. *The Tenth International Symposium on Wireless Communication Systems*, 773-777. Berlin: Vde verlag gmbh.

*VideoLAN*. (n.d.). Retrieved May 29, 2015, from VLC media player source code: http://www.videolan.org/vlc/download-sources.html

Watson, S. J. (1991). An analysis of the concept of experience. *Journal of Advanced Nursing, 16*(9), 1117-1121.

Yetgin, Z. & Çelik, T. (2012). Efficient progressive downloading over multimedia broadcast multicast service, Computer Networks,. *Computer Networks: The International Journal of Computer and Telecommunications Networking, 56*(2), 533-547.

Yetgin, Z. & Göçer, Z. (2015). Quality of experience prediction model for progressive downloading over mobile broadcast networks. *Telecommun Systems, 58*, 55-66.

Yetgin, Z. & Seckin, G. (2008). Progressive download for 3G Wireless Multicasting. *International Journal of Hybrid Information Technology, 1*(2), 67-82.

Yetgin, Z. & Seckin, G. (2009). Progressive download for multimedia broadcast multicast service. *IEEE MultiMedia, 16*(2), 76-85.

Yu, Y.-T. & Chao, H.-L. (2007). QoS-aware link rate adaptation in IEEE 802.16 broadband wireless access systems. *The 18th Annual IEEE International Symposium on Personal, Indoor and Mobile Radio Communications,* 1 - 5.

## APPENDICES

### 1.2 Installation of NS2

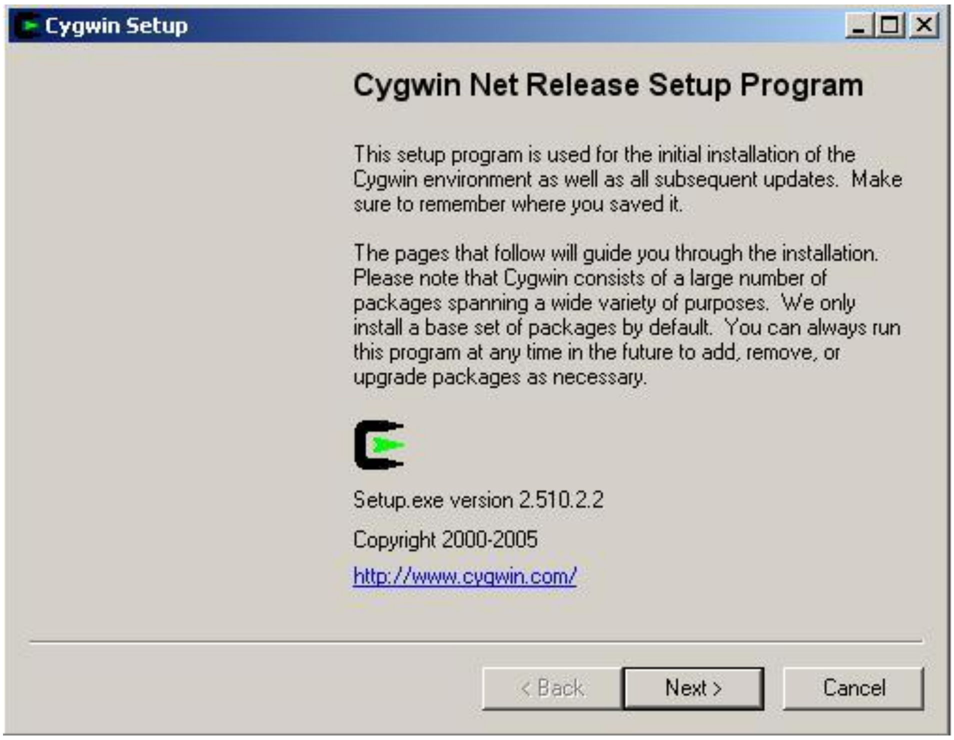
There are two commonly operating systems to create simulation environments namely: Microsoft windows-xp and other is Linux. First, we try installing NS2 in window-xp environment then we try installing in Ubuntu which is based on Linux environment.

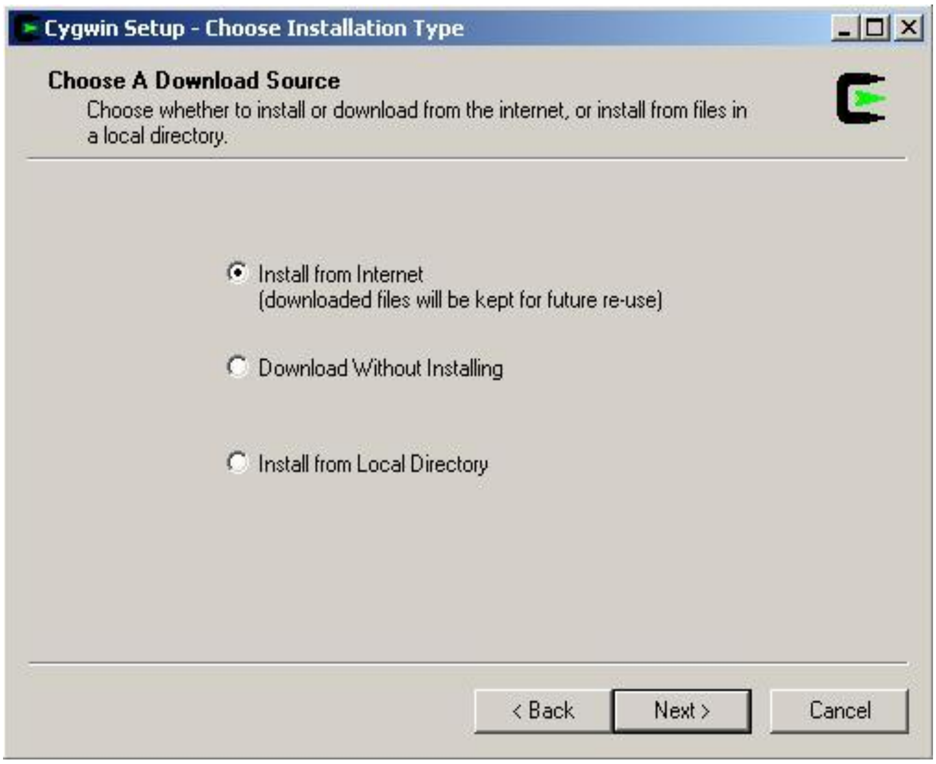
#### 1.2.1 Installation in Windows-xp operating system

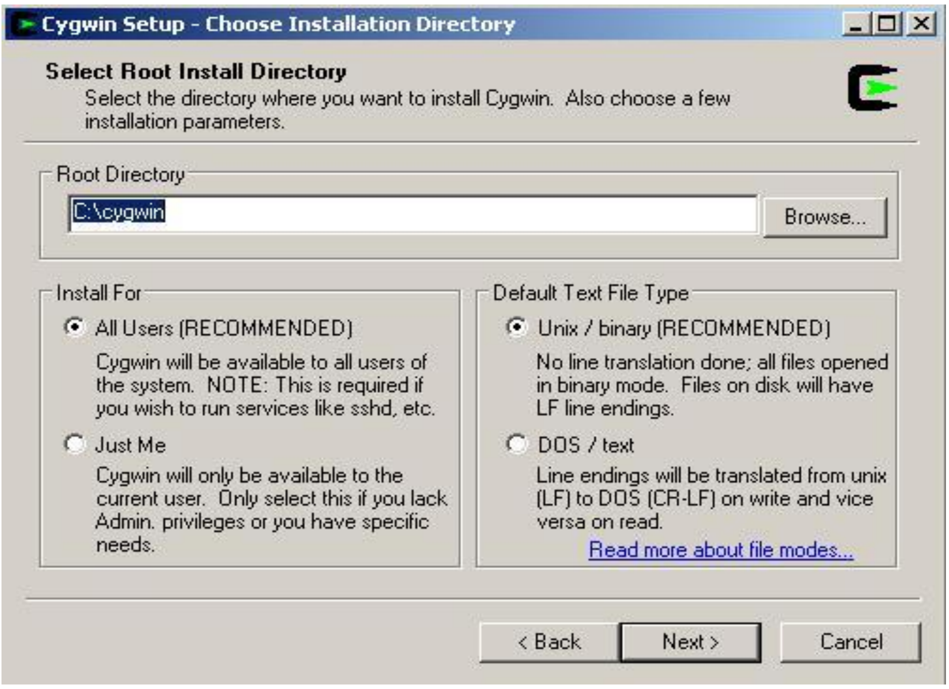
#### 1.2.1.1 Installation of Cygwin

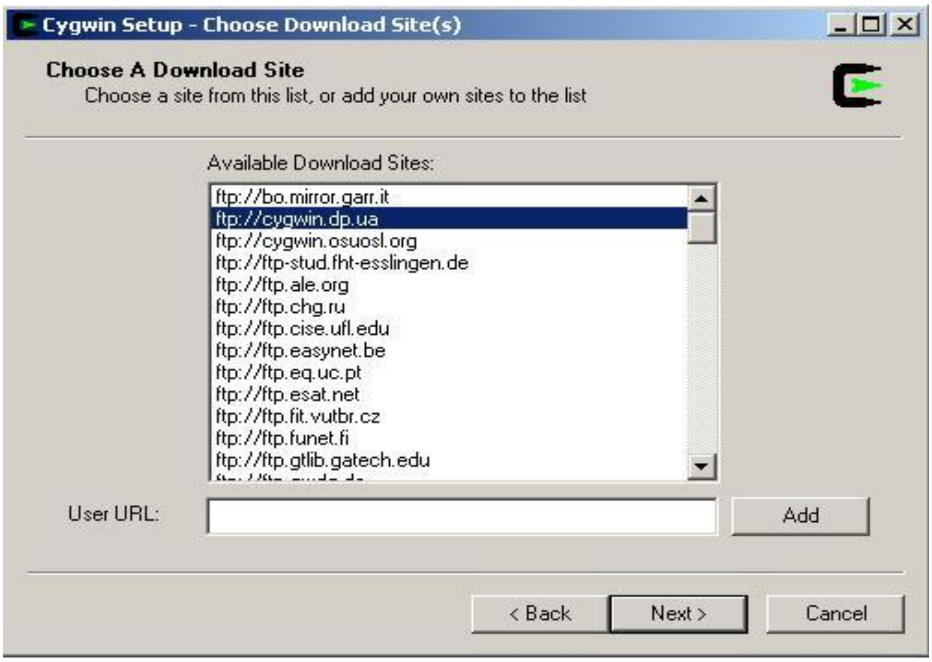
Before of installing NS2, we should install Cygwin.

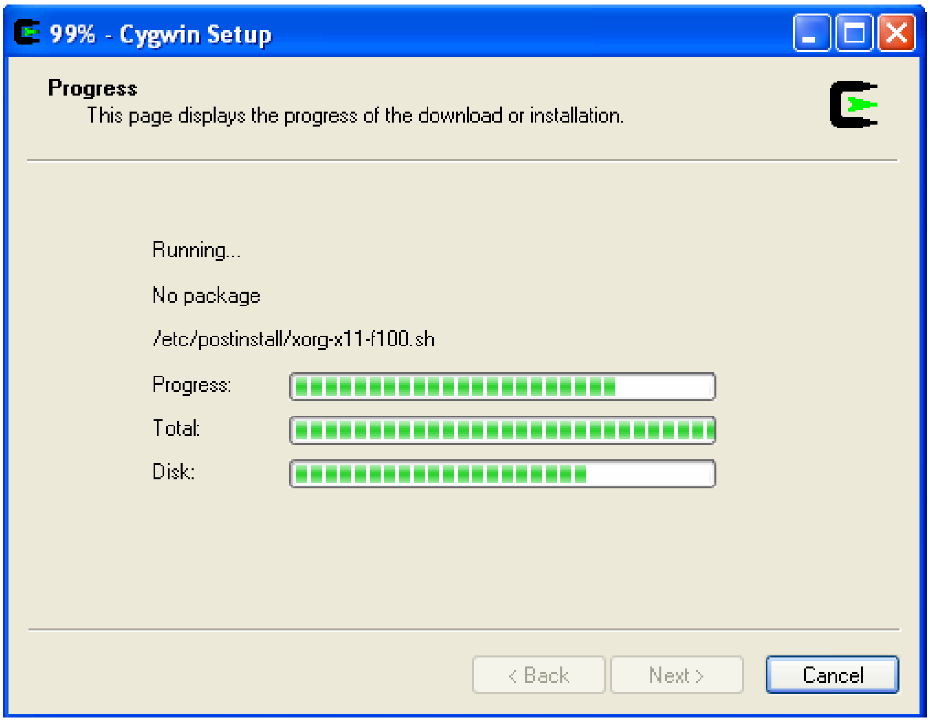
- Download cygwin.exe from http://www.cygwin.com/
- Click the "cygwin.exe"
- Cygwin Installation

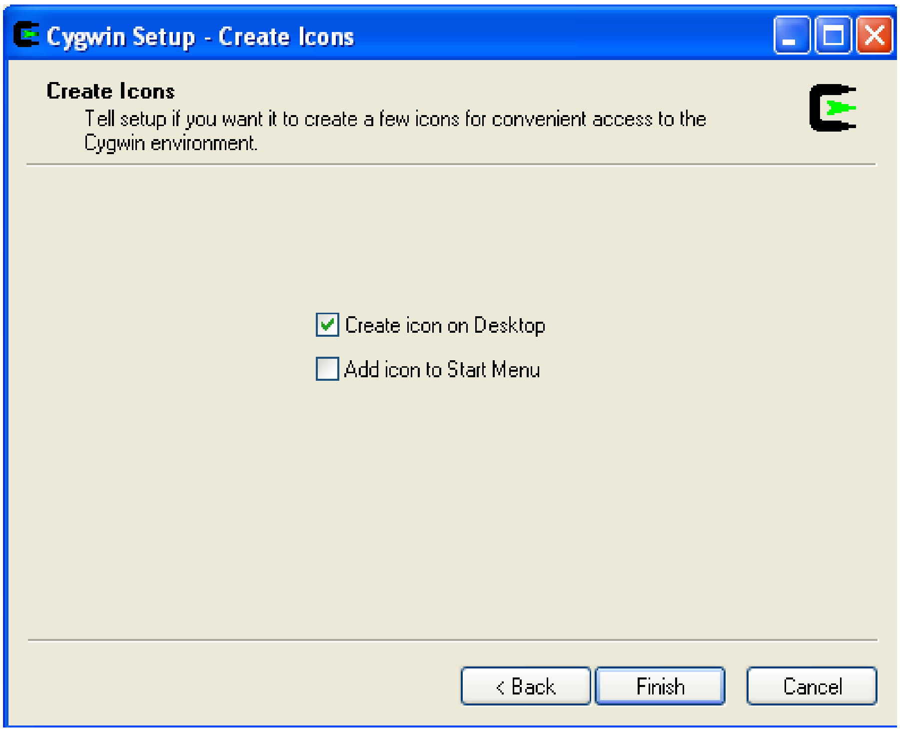



Cygwin Setup - Choose Installation Type
Choose A Download Source
Choose whether to install or download from the internet, or install from files in a local directory.
Install from Internet
(downloaded files will be kept for future re-use)
Download Without Installing
Install from Local Directory
< Back
Next >
Cancel


Cygwin Setup - Choose Installation Directory
Select Root Install Directory
Select the directory where you want to install Cygwin. Also choose a few installation parameters.
Root Directory
C:\cygwin
Browse...
Install For
All Users (RECOMMENDED)
Cygwin will be available to all users of the system. NOTE: This is required if you wish to run services like sshd, etc.
Just Me
Cygwin will only be available to the current user. Only select this if you lack Admin. privileges or you have specific needs.
Default Text File Type
Unix / binary (RECOMMENDED)
No line translation done; all files opened in binary mode. Files on disk will have LF line endings.
DOS / text
Line endings will be translated from unix (LF) to DOS (CR-LF) on write and vice versa on read.
Read more about file modes...
< Back
Next >
Cancel


Cygwin Setup - Choose Download Site(s)
Choose A Download Site
Choose a site from this list, or add your own sites to the list
Available Download Sites:
ftp://bo.mirror.garr.it
ftp://cygwin.dp.ua
ftp://cygwin.osuosl.org
ftp://ftp-stud.fht-esslingen.de
ftp://ftp.ale.org
ftp://ftp.chg.ru
ftp://ftp.cise.ufl.edu
ftp://ftp.easynet.be
ftp://ftp.eq.uc.pt
ftp://ftp.esat.net
ftp://ftp.fit.vutbr.cz
ftp://ftp.funet.fi
ftp://ftp.gtlib.gatech.edu
User URL:
Add
< Back
Next >
Cancel

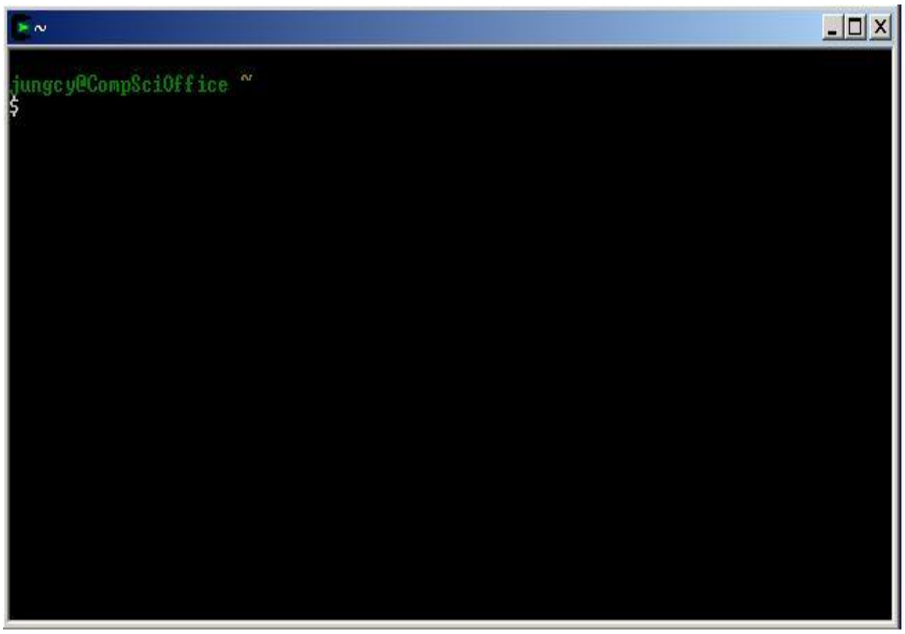

99% - Cygwin Setup
Progress
This page displays the progress of the download or installation.
Running...
No package
/etc/postinstall/xorg-x11-f100.sh
Progress:
Total:
Disk:
< Back
Next >
Cancel



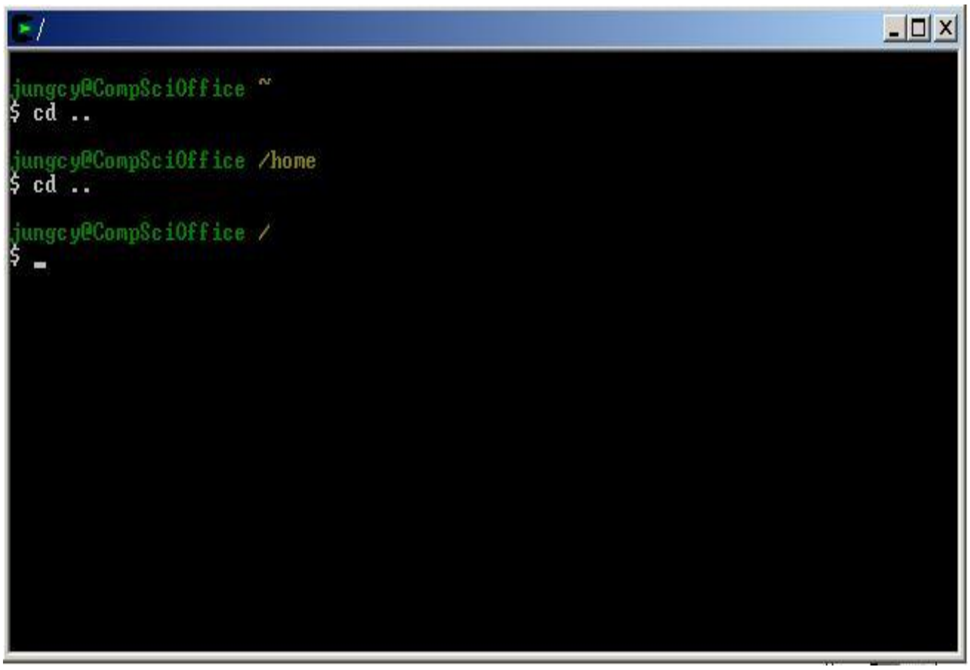

Cygwin is installed. Then we install NS2.

### 1.2.1.2 Installation of NS2

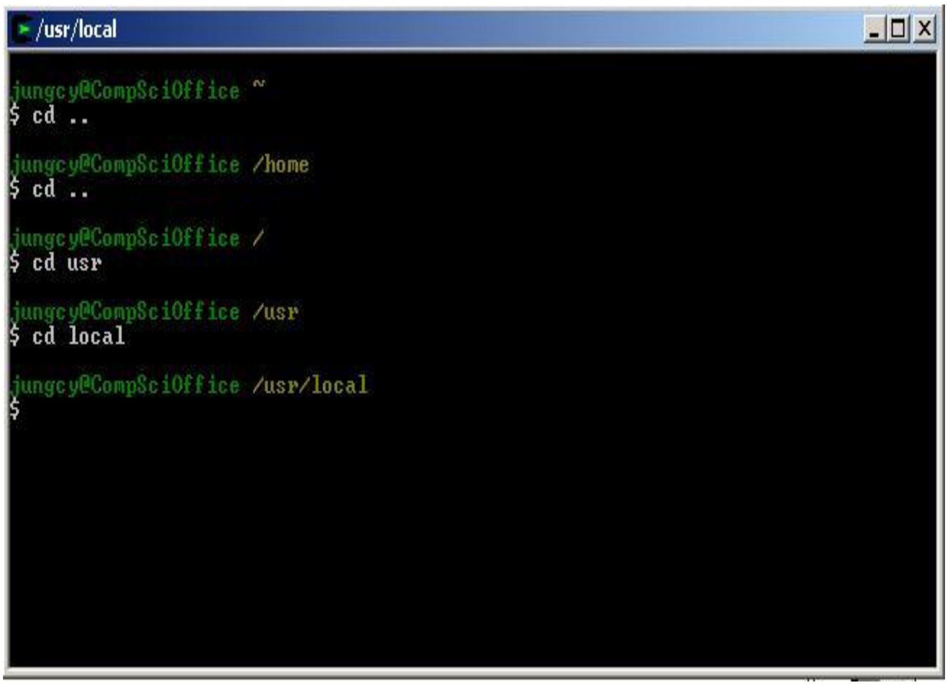
- Download ns-allinone-2.29.2.tar.gz from Website http://sourceforge.net/projects/nsnam/files/ns-2/ns-2.29/ and Save it to the c:/Cygwin /usr/local
- Extract it: tar xvfz ns-allinone-2.29.2.tar.gz
- Click on desktop icon "Cygwin"



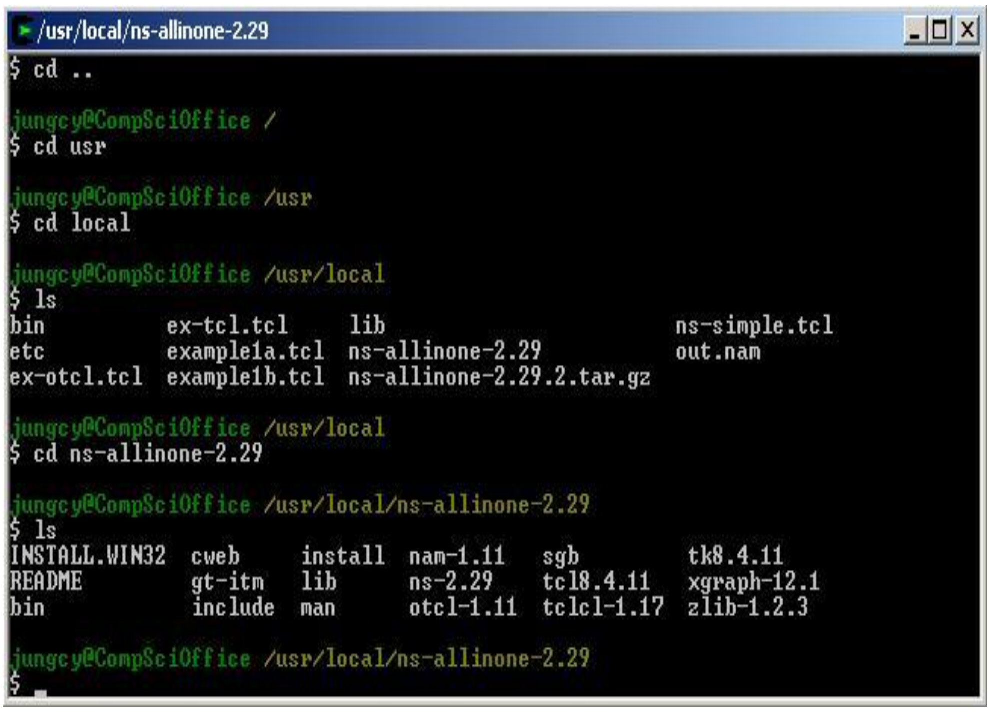

- Type “cd ..” to go to the upper folder(“cd” must be low case. And there is one space between “d” and “.”)
- Type “cd ..” again.



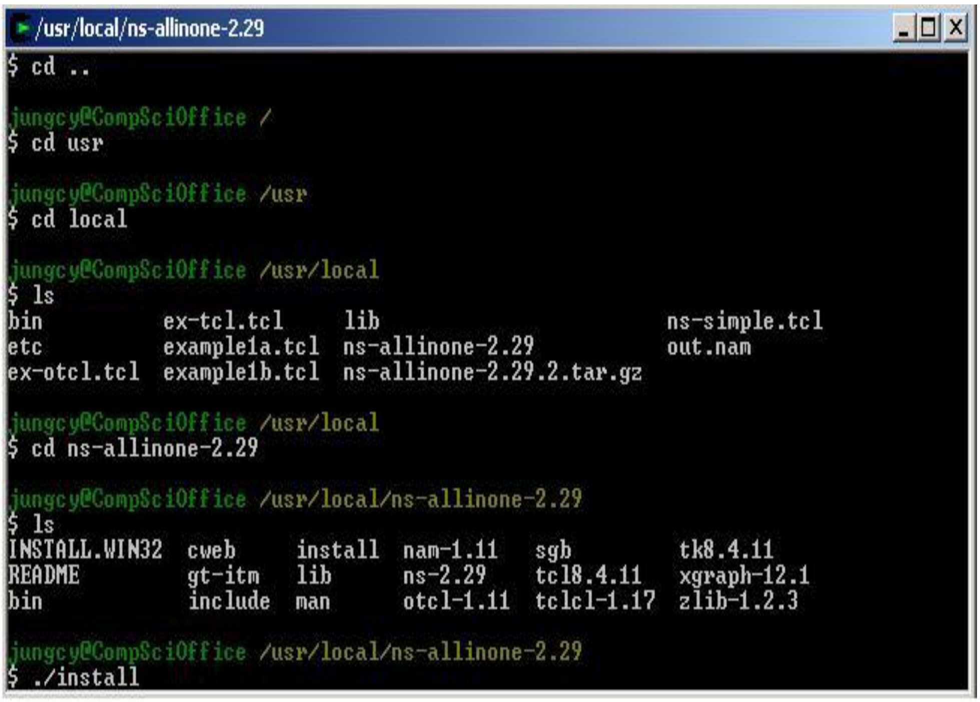

- “cd usr”, go to folder “usr”
- “cd local”, go to folder “local”



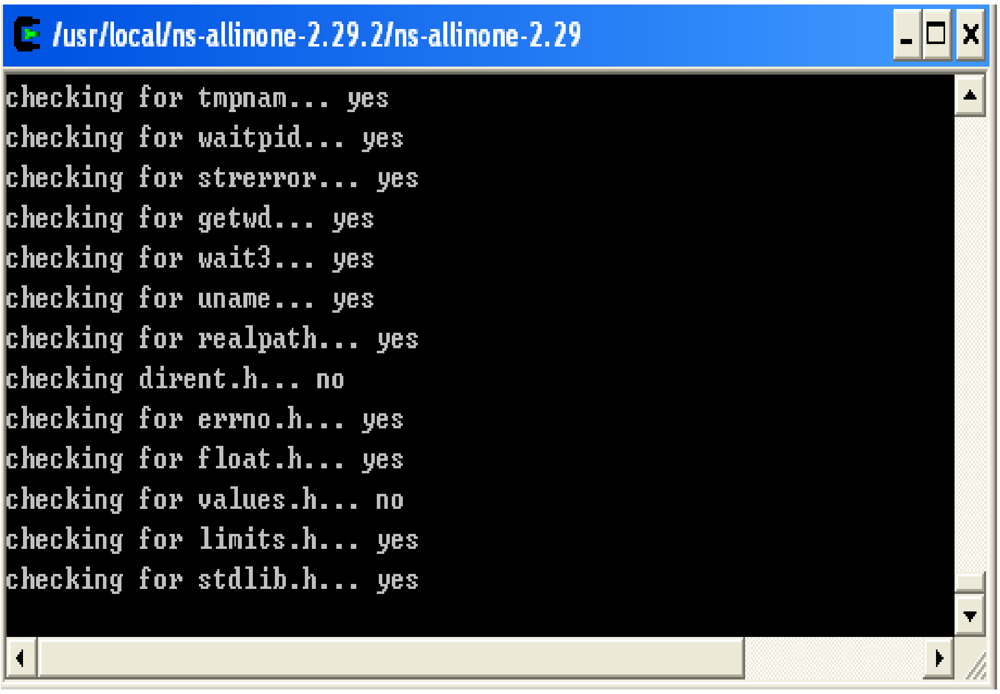

- Find "install.exe"

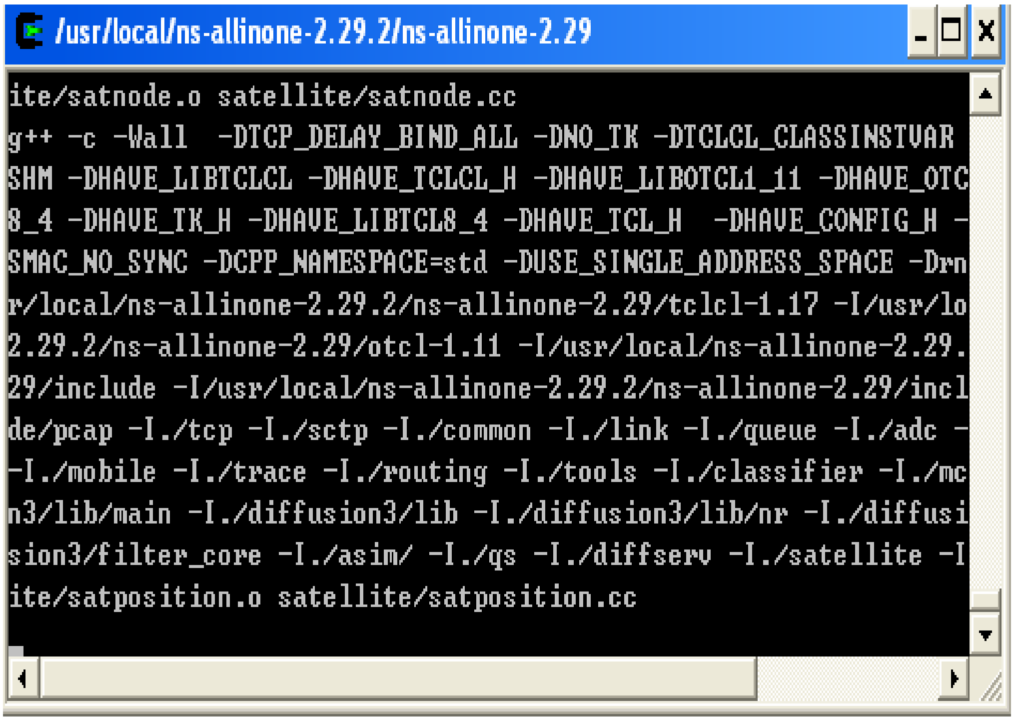



- Start to run the installation "./install"


/usr/local/ns-allinone-2.29
$ cd ..
jungcy@CompSciOffice /
$ cd usr
jungcy@CompSciOffice /usr
$ cd local
jungcy@CompSciOffice /usr/local
$ ls
bin ex-tcl.tcl lib ns-simple.tcl
etc example1a.tcl ns-allinone-2.29 out.nam
ex-otcl.tcl example1b.tcl ns-allinone-2.29.2.tar.gz
jungcy@CompSciOffice /usr/local
$ cd ns-allinone-2.29
jungcy@CompSciOffice /usr/local/ns-allinone-2.29
$ ls
INSTALL.WIN32 cweb install nam-1.11 sgb tk8.4.11
README gt-itm lib ns-2.29 tcl8.4.11 xgraph-12.1
bin include man otcl-1.11 tclcl-1.17 zlib-1.2.3
jungcy@CompSciOffice /usr/local/ns-allinone-2.29
$ ./install

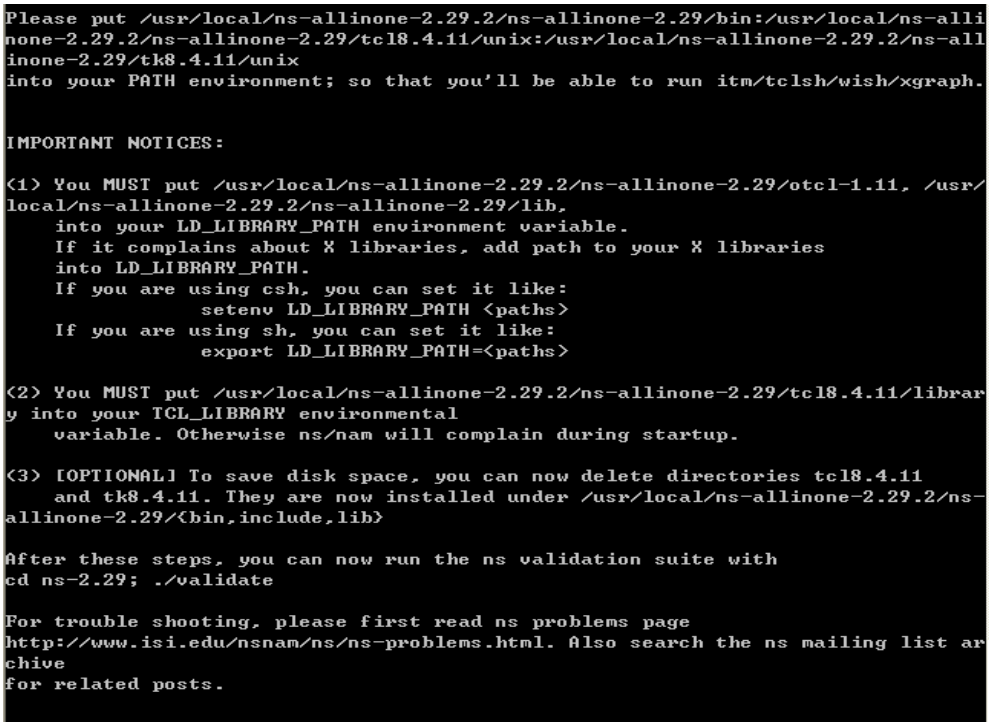
- Installing


/usr/local/ns-allinone-2.29.2/ns-allinone-2.29
checking for tmpnam... yes
checking for waitpid... yes
checking for strerror... yes
checking for getwd... yes
checking for wait3... yes
checking for uname... yes
checking for realpath... yes
checking dirent.h... no
checking for errno.h... yes
checking for float.h... yes
checking for values.h... no
checking for limits.h... yes
checking for stdlib.h... yes



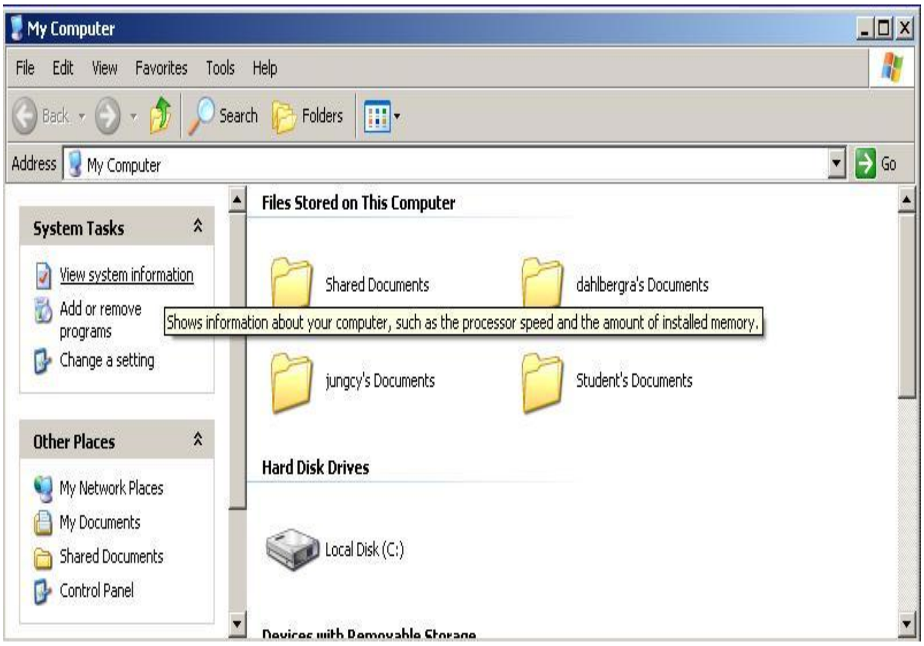
```
/usr/local/ns-allinone-2.29.2/ns-allinone-2.29
ite/satnode.o satellite/satnode.cc
g++ -c -Wall  -DTCP_DELAY_BIND_ALL -DNO_TK -DTCLCL_CLASSINSTVAR
SHM -DHAVE_LIBTCLCL -DHAVE_TCLCL_H -DHAVE_LIBOTCL1_11 -DHAVE_OTC
8_4 -DHAVE_TK_H -DHAVE_LIBTCL8_4 -DHAVE_TCL_H  -DHAVE_CONFIG_H -
SMAC_NO_SYNC -DCPP_NAMESPACE=std -DUSE_SINGLE_ADDRESS_SPACE -Drn
r/local/ns-allinone-2.29.2/ns-allinone-2.29/tclcl-1.17 -I/usr/lo
2.29.2/ns-allinone-2.29/otcl-1.11 -I/usr/local/ns-allinone-2.29.
29/include -I/usr/local/ns-allinone-2.29.2/ns-allinone-2.29/incl
de/pcap -I./tcp -I./sctp -I./common -I./link -I./queue -I./adc -
-I./mobile -I./trace -I./routing -I./tools -I./classifier -I./mc
n3/lib/main -I./diffusion3/lib -I./diffusion3/lib/nr -I./diffusi
sion3/filter_core -I./asim/ -I./qs -I./diffserv -I./satellite -I
ite/satposition.o satellite/satposition.cc
```

- Configure system variables and library paths
- After finishing installing, following window appears

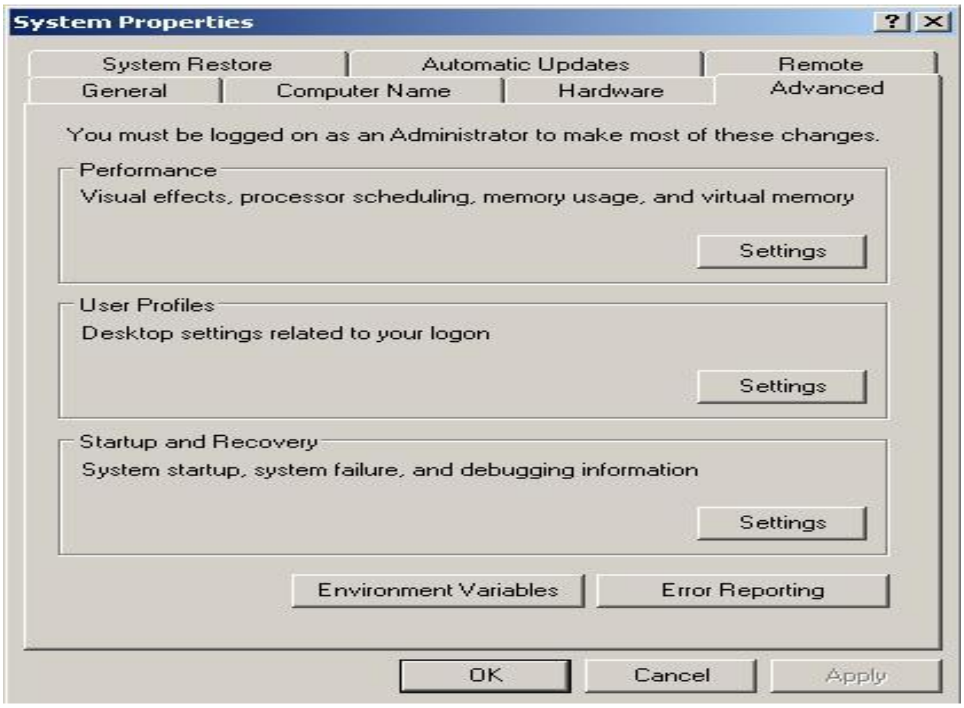


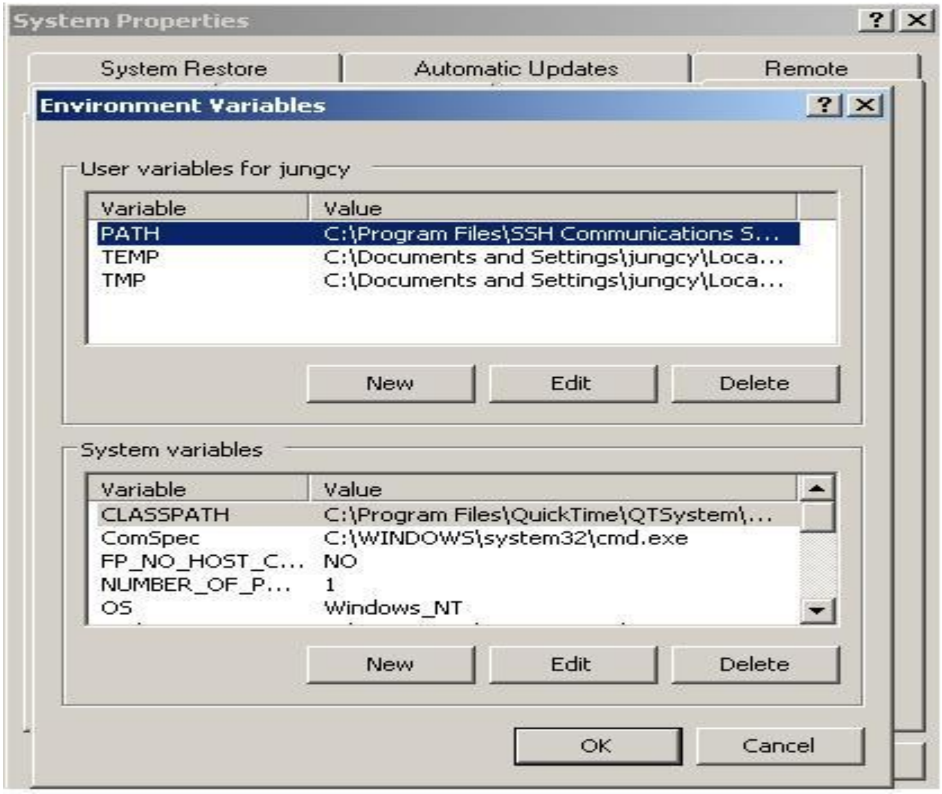
```
Please put /usr/local/ns-allinone-2.29.2/ns-allinone-2.29/bin:/usr/local/ns-alli
none-2.29.2/ns-allinone-2.29/tcl8.4.11/unix:/usr/local/ns-allinone-2.29.2/ns-all
inone-2.29/tk8.4.11/unix
into your PATH environment; so that you'll be able to run itm/tclsh/wish/xgraph.


IMPORTANT NOTICES:

(1) You MUST put /usr/local/ns-allinone-2.29.2/ns-allinone-2.29/otcl-1.11, /usr/
local/ns-allinone-2.29.2/ns-allinone-2.29/lib,
    into your LD_LIBRARY_PATH environment variable.
    If it complains about X libraries, add path to your X libraries
    into LD_LIBRARY_PATH.
    If you are using csh, you can set it like:
                setenv LD_LIBRARY_PATH <paths>
    If you are using sh, you can set it like:
                export LD_LIBRARY_PATH=<paths>

(2) You MUST put /usr/local/ns-allinone-2.29.2/ns-allinone-2.29/tcl8.4.11/librar
y into your TCL_LIBRARY environmental
    variable. Otherwise ns/nam will complain during startup.

(3) [OPTIONAL] To save disk space, you can now delete directories tcl8.4.11
    and tk8.4.11. They are now installed under /usr/local/ns-allinone-2.29.2/ns-
allinone-2.29/{bin,include,lib}

After these steps, you can now run the ns validation suite with
cd ns-2.29; ./validate

For trouble shooting, please first read ns problems page
http://www.isi.edu/nsnam/ns/ns-problems.html. Also search the ns mailing list ar
chive
for related posts.
```

- Configure system variables
- Go to “My computer” and click “view system information”

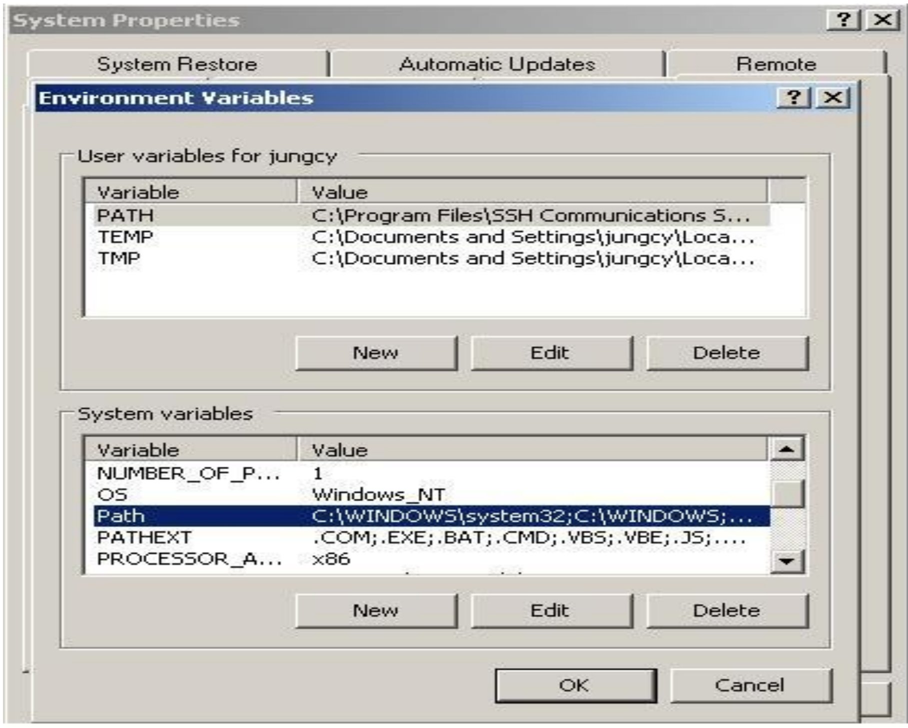


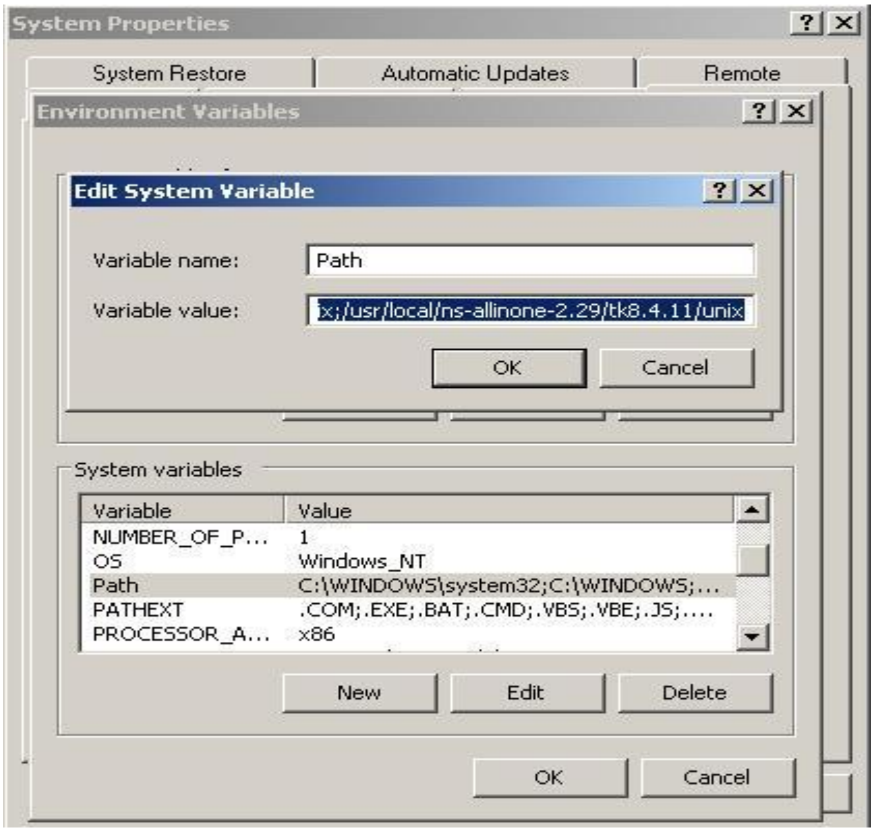

- Go to “advanced” Tab and Click “Environmental variables”



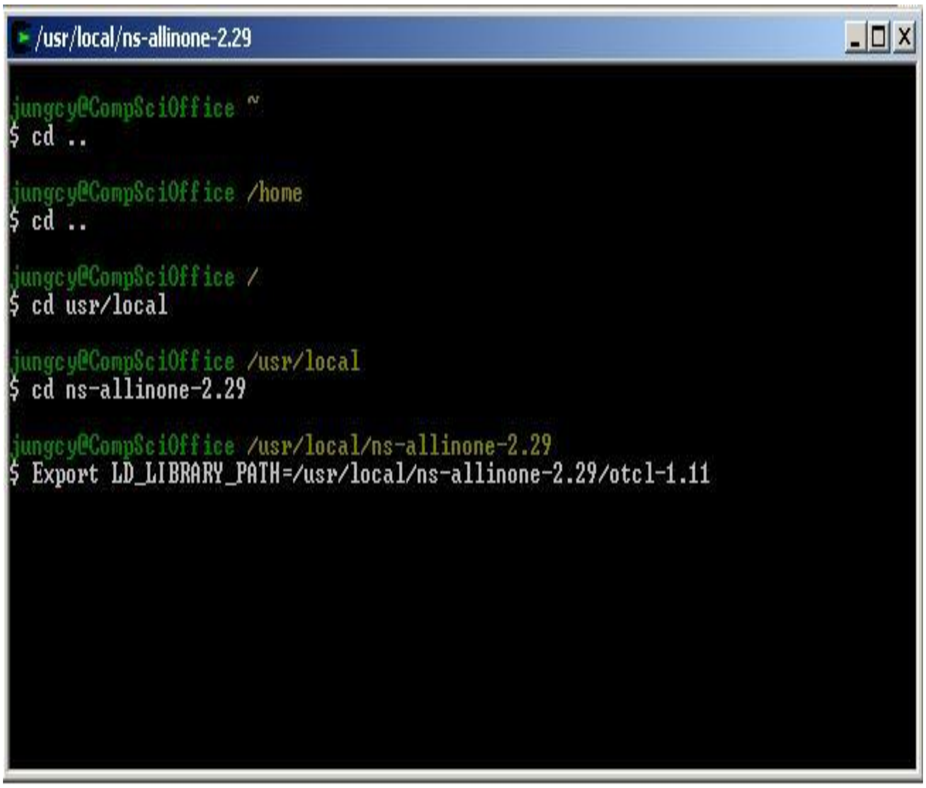



- Highlight "path" which is "system variable" box and press the "Edit" button.






- Add the following path separated with ";" make sure to not change the existing path.
  /usr/local/ns-allinone-2.29.2/bin
  /usr/local/ns-allinone-2.29.2/tcl8.4.11/unix
  /usr/local/ns-allinone-2.29.2/tk8.4.11/unix
- Go to cygwin and type the following content
  Export LD_LIBRARY_PATH=/usr/local/ns-allinone-2.29/otcl-1.11
  Export LD_LIBRARY_PATH=/usr/local/ns-allinone-2.29/lib
  Export TCL_LIBRARY_PATH=/usr/local/ns-allinone-2.29/tcl8.4.11/library




NS2 is installed successfully. But the Cygwin has various limitations. Cygwin works only in Microsoft Window-xp properly and correctly. The new version of Cygwin software does not work in Network Simulation. It would not contain all facility which is provided by Linux. Even though this operating system itself also freeware. So it is better to install NS2 for research in Ubuntu because Ubuntu support full Linux environment.

**1.2.2 Installation NS2 in Ubuntu operating system**

PROCEDURE 1:

1) First, we Update our Ubuntu installation

$ sudo apt-get update

2) Install ns2.29, nam and xgraph

$ sudo apt-get install ns nam xgraph

PROCEDURE 2:

Step1: Download the ns-allinone-2.29 from this site: http://sourceforge.net/projects/nsnam/files/ns-2/ns-2.29/

Step2: Place the ns-allinone-2.29.tar.gz package in our home folder (/home/zohre). Right click the package and extract the contents in the same folder.

Step3: Next, open the Terminal (Applications-->Accessories-->Terminal)

Step4: Change to ns-allinone2.29 directory

$ cd /home/ zohre /ns-allinone-2.29

Step5: First install all the dependencies

$ sudo apt-get install build-essential autoconf automake libxmu-dev gcc-4.3

Note that we are downgrading the gcc version, as ns2.29 works well with gcc4.3

Edit Makefile.in found at this location ns-allinone-2.29/otcl-1.13/Makefile.in as follows:

Find the line that says: CC= @CC@

And change it to: CC= gcc-4.3

Step 6: Begin ns2.29 installation

$ sudo su

#./install

Step 7: Once the installation is successful i.e without any errors, we need to add the path information to the file ~/.bashrc

#gedit /home/zohre/.bashrc

Step8: Append the following lines to the file ~/.bashrc

# LD_LIBRARY_PATH

OTCL_LIB=/home/ zohre /ns-allinone-2.29/otcl-1.13

NS2_LIB=/home/ zohre /ns-allinone-2.29/lib

X11_LIB=/usr/X11R6/lib

USR_LOCAL_LIB=/usr/local/lib

exportLD_LIBRARY_PATH=$LD_LIBRARY_PATH:$OTCL_LIB:$NS2_LIB:$X11_LIB:$USR_LCAL_LIB

# TCL_LIBRARY

TCL_LIB=/home/ zohre /ns-allinone-2.29/tcl8.4.18/library

USR_LIB=/usr/lib

export TCL_LIBRARY=$TCL_LIB:$USR_LIB

# PATH

XGRAPH=/home/ zohre /ns-allinone-2.29/bin:/home/zohre/ns-allinone-2.29/tcl8.4.18/unix:/home/ zohre /ns-allinone-2.29/tk8.4.18/unix

#the above two lines beginning from xgraph and ending with unix should come on the same line

NS=/home/ zohre /ns-allinone-2.29/ns-2.29/

NAM=/home/ zohre /ns-allinone-2.29/nam-1.14/

PATH=$PATH:$XGRAPH:$NS:$NAM

Step 9: For the changes to take effect immediately, do the following:

#source ~/.bashrc

After this, type ns to see %and type nam to show the nam startup window. This proves that our installation has been successful.