QOS For Multimedia Applications with Emphasize on Video Conferencing

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***Abstract*—Over the past few years, the fast growth of internet made it a new target for some operators which decided to use it as a new infrastructure to offer their services such as voice and video to the customers via internet. In order to fulfill this, they developed multimedia applications which changed the nature of traffic over the internet. As a result, it seemed that there is a need to a new mechanism to guarantee the quality of service. therefore, to make these applications widely used, QOS requirements must be met. In this article we are going to talk about the QOS challenges specifically for video conferencing as a multimedia application, and we try to explain how to reach the appropriate quality for this applications in a congested network.**

***Index Terms*— Quality of Service, QOS, Multimedia Application, Video Conferencing**

I. Introduction

Multimedia is a terminology that is composed of voice, video and data. These forms of traffic can be integrated into the same application called *Multimedia Applications*. There are so many kinds of multimedia applications such as: streaming media (Music and video on demand), IPTV, IP Telephony (VOIP), online games, video conferencing, industrial control systems, network operation support systems and etc.

Video conferencing is a real-time communication including voice and video between two or more people at different locations via internet. This application can be used in distinct areas like distance learning for educational purposes, remote surgery in medical institutions, business meetings and judicial systems. Some of the most popular video conferencing tools are as follows:

- Skype (One to One).
- OOVOO (One to Many).
- Mega meeting (Many to Many up to 16).

Offering appropriate quality is a dilemma for these multimedia applications which involves an undisrupted video and voice communication with satisfactory quality for users.

Quality of service is a solution to achieve this goal. QOS comprises some measurable factors like network availability, bandwidth, delay, jitter, loss, and immeasurable ones such as emission priority and discard priority. By determining the suitable criteria for these parameters we can provide selected traffic with better service over converged networks.

In this paper we try to reach a solution to gain the acceptable quality with tendency to video conferencing by describing QOS requirements sensitivities and how to implement it by considering these factors.

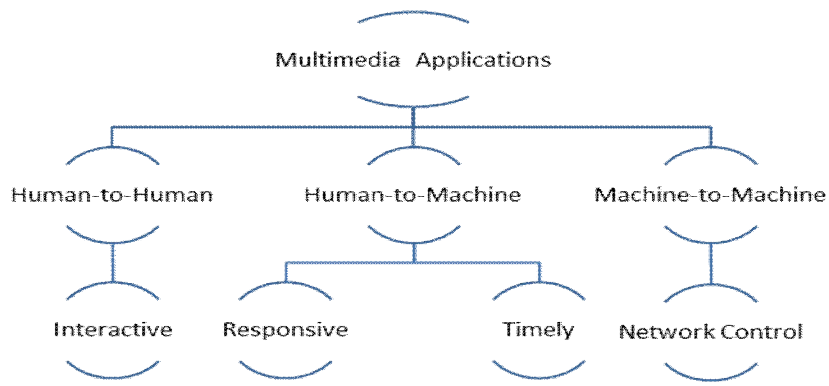
II. Background

A. Multimedia Applications

Traditionally different types of traffic required dedicated networks, for example video conferencing operated over ISDN (Integrated Service Digital Network) and voice transmitted via PSTN (Public Switched Telephone Network). With the rapid growth of internet, the idea of using of common infrastructure to handle all sorts of traffic became widespread. Some benefits lies in reducing the operational cost of using separate networks and simplification of maintenance which lead to profit improvement. Presenting text, graphic, video, animation and sound in an integrated way over IP became a new concern. To cope with that there seemed to be a necessity for an application called multimedia application. Multimedia applications can be categorized based on the kind of interactivity as shown below [9]:

- *Interactive applications*:

Applications which form human to human communications. This can be between two or more people. VoIP, interactive gaming and video conferencing are some



examples of that. It is obvious the user expects real time reaction from the applications. Real-time concept encompasses minimal delay, jitter and loss sensitivity.

- *Response applications:*

Near real-time applications between human and machine that need approximately quick response to the request. Therefore delay, jitter and loss should be minimized. Some instances are streaming audio/video, client/server transactions.

- *Timely applications:*

This is the same as previous one except for need to near real-time response but still should perform at a certain amount of time such as email. Timely applications are tolerant to jitter, loss and bounded amount of delay.

- *Network control applications:*

It involves interaction between machines on the both sides for network controlling and administration of data transactions. Such applications include critical alarms, routing protocols, billing applications. They are not jitter sensitive. They can accept minimal amount of delay and loss.

B. Video Conferencing

Video conferencing is a multimedia application that provide clients with the possibility of participating in an real-time audiovisual session while they may be in distinct geographical locations. There are two types of video conferencing: point to point and multi point.

- *Point to point:*

When two individuals are involved in a two way communication. No extra equipment is needed. Participants just dial up either IP address or domain name to access other side.

- *Multi point:*

It is a session among several users. Each connects to a device called MCU (multi point control unit) through gatekeeper which will be introduced later.

Video conferencing benefits and barriers

Video conferencing has facilitated communication in diverse ways such as:

- *Travel saving:*

Attending a meeting necessitates participants to spend a lot of time and money for traveling. Furthermore you will be ignorant of current affairs happening in your workplace. So using video conferencing is an alternative to overcome these problems.

- *Security:*

When privacy has to be concerned video conferencing is a good choice to provide appropriate safety for us.

- *Productivity increment:*

Using video conferencing will enables participants to be in a permanent contact with each other. Reducing the meeting arrangement and attendance time will lead to speed up products and service enhancement.

- *Extension of communications beyond international boundaries :*

Video conferencing let us to aggregate different ideas from various sites of the world whereby face to face contact will transfer the latest information to all at once.

Video conferencing is a very useful and effective communication tool in many cases, but it is not widely used, since it encounters some problems:

- *Primitive deployment:*

At the beginning it was very costly to use video conferencing because of its expensive equipment. By evolution of technology they became cheaper. Even home users could buy it easily with reasonable cost. Despite lower cost, industries prefer to test different types of instruments to choose the appropriate instruments.

- *Availability of high speed medium:*

Some small businesses doesn't have suitable infrastructure for high speed transmission. Therefore they must add up the cost of promoting their platform to expenses of video conferencing investment.

- *Quality:*

In the past the tendency to use video conferencing wasn't high due to the low video quality caused by limitations of network components. But nowadays as technology developed and device capabilities enhanced, video conferencing got a chance to be taken into consideration as a new tool for communication. For example frame rate is one of the quality dependent factors that has experienced a tremendous jump in value during this time. Another one is multicast supporting networks growth.

- *Interoperability:*

The main issue proposed here is intercommunicating between devices of different technologies and protocols. Now there is an attempt to reach an agreement about a universal standard which cover this subject entirely.

- *Lack of ubiquity:*

Deploying video conferencing whenever and wherever is not feasible. Because it requires some facilities if not available.

- *User acceptance:*

The acceptance of video conferencing depends on personality of company members whether they are prepared to accept changes or resist in front of technology.

- *Comfort with hardware:*

As much as the user feels comfortable with the hardware the adaption process will be faster. Training has a great influence on this.

- *User preparation:*

The persons in charge of video conferencing establishment should be trained to apply user expectations properly. For example a user who doesn't need to see opposite ones in multipoint conference.

C. *Mega conference [6]*

It is one of the most popular and largest video conferencing summit annually held across the world which use the standard protocol (H.323) engaging hundreds of people of different layer such as vendors, researchers, ordinary people and etc.

III. Multimedia Application Solutions

Typically traditional networks were designed to service specific kind of traffic, like telecommunication networks which carry voice or computer networks which transmit data and cable networks for video transfer. But today multiservice networks has come into existence that can handle different types of flows simultaneously, but some barriers are still in the path. Such as congestion capacity constrain and interaction diversity. Ways to dominate these problems are:

- Over provisioning
- Separate networks
- QOS

Over provisioning : It means increasing bandwidth which assures offering various traffic types without any delay, conflict at once. It is a good solution for small networks, but when expansion is subject to consider there might be need to multiply network resource capacities. Because of high resource consumption and costly implementation, it isn't always a desirable solution.

Separate networks: This is another guideline that isolates network components for each kind of traffic which solve the interference among voice, video and data. But the tackle with bandwidth and delay still remains. Furthermore cost is another concern in this solution. There is a possibility to solve the bandwidth problem by merging over provisioning for each separate network, but the cost will burst.

QOS: QOS is an ability to have various behaviors toward different types of traffic to meet their requisites. Implementing QOS needs a predefined level of network capacity. Having supplied that would manage the traffic in the best way without additional resources.

IV. QOS

A. *QOS requirements*

End to end QOS of multimedia applications can be considered from two aspects: network and end points. Network and Different end points have distinct QOS parameters. Considering these facts QOS can be classified as follows [4]:

- *Application level QOS:*

User related metrics like throughput, latency, availability and continuity of service (Frame size, Frame rate, Image and audio clarity).

- *System level QOS:*

End point system requirements such as CPU and OS requirements.

- *Network level QOS:*

The most significant parameters of QOS which are communication related such as bandwidth, jitter, delay, loss and reliability (network availability).

QOS requirements are presented as follows:

Bandwidth and throughput: Bandwidth is the available capacity of connection between two terminals as the most popular term for that is (bps). Throughput slightly differs from bandwidth as it stands for effective bandwidth that is provided by network.

Delay or latency: It specifies the time it takes for a packet to leave source until reaching the destination. Applications and network devices can cause delay.

Jitter (delay variation): Jitter is an interval between subsequent packets. It is occurred by network congestion, route alternation and etc.

Loss: It is amount of packets out of all that are not received at destination. The success of QOS depends on this factor.

Reliability: Some applications are sensitive to packet loss such as real-time applications. Thus there must be some mechanism either in application or network to minimize the packet loss, such as forward error correction (FEC).

Table 1: Applications QOS Metrics Sensitivity [9]

Application	Bandwidth	Sensitivity to:		
		Delay	Jitter	Loss
VOIP	Low	High	High	Med
Video Conferencing	High	High	High	Med
Streaming Video	High	Med	Med	Med
Streaming Audio	Low	Med	Med	Med
Client/Server Transactions	Med	Med	Low	High
Email	Low	Low	Low	High
File Transfer	Med	Low	Low	High

Availability and continuity of service: This parameter talks about the user satisfaction level of service. Some of the most important ones are:

Frame size: It is the size of image on the screen. The bigger the frame size the more bandwidth it requires. QOS can define the frame size.

Frame rate: It refers to frames per unit of time which sent to network higher frame rate needs more bandwidth.

Image clarity: It is user perception of quality of received image.

Audio clarity: It specifies the rate of audio recording or emitting per unit of time. Higher audio quality demands higher rate.

Video conferencing QOS requirements

Video conferencing is an interactive video application with following recommended QOS factors [1]:

- For best quality it should be marked to DSCP AF41, but less qualities correspond to AF42 or AF43.
- Loss shouldn’t exceed 1%.
- End to end delay should be below 150 ms.
- Jitter should be under 30 ms.
- 28 Kbps-1.5 Gbps bandwidth is required depending on designated video format).

Table 2: Typical bandwicth for some video formats [6]

Video Format	Typical Bandwidth Requirement
Uncompressed HDTV	1.5 Gbits/sec
HDTV, Interim Format	360 Mbits/sec
Standard Definition TV (SDTV), SMPTE	270 Mbits/sec
Compressed MPEG-2 4:2:2	25-60 Mbits/sec
Broadcast Quality HDTV (MPEG-2)	19.4 Mbits/sec
MPEG-2 Standard Definition TV (SDTV)	6 Mbits/sec
MPEG-1	1.5 Mbits/sec
MPEG-4	5 Kbits/sec-4 Mbits/sec
H.323 (H.263)	28 Kbits/sec-1 Mabits/sec

B. QOS Solutions For Multimedia Applications

There are some mechanisms to handle traffic most efficiently. Two most important are Differentiated Service (DiffServ) and Integrated Service (IntServ).

IntServ

This solution of QOS was developed by demand of interactive applications. It performs QOS with the use of resource reservation and admission control mechanisms. IntServ relies on Resource Reservation Protocol (RSVP) to request expected QOS requirement from network and reserve bandwidth. If reservation attempt succeeds application can begin the communication, if not application may reduce its essentials to meet the agreement with network. IntServ assures network QOS metrics such as bandwidth, loss and delay so it can be named Hard QOS.

Strengths and shortcomings of IntServ:

- *Explicit admission control:* It guarantees that actual request resources will be dedicatedly delivered to the applicant.
- *Application coordination:* This mechanism ease the communication by using dynamic port numbers to answer the application request in case of request denial application can lower its expectations and resend the request.
- *Non-scalable architecture:* Because of increasing overhead of continuous signaling and controlling flows due to RSVP stateful architecture, IntServ is not suitable for enterprise networks.
- All the devices along the path between end points must be RSVP enabled to satisfy required QOS.

DiffServ

This model works based on classifying different classes of traffic. DiffServ uses a field in IP packet header which called DiffServ Code Point (DSCP) to mark services it would get from network.

DSCP has two popular values:

- Expedited Forwarding (EF)
- Assured Forwarding (AF)

EF: It provides the packet with low latency, loss and jitter to achieve the highest possible priority from the network. So it fits for VoIP. Since the priority for voice and video will be marked the same and in regard to larger size of video packets delay will increase for voice packet while waiting for video packets to be processed. In addition, small size of EF queues lead to video packet loss growth. Furthermore since the video packets same priority would precede the video ones therefore sync problem arises.

AF: It guarantees the delivery of packets as long as the path is not oversubscribed. If congestion happens it will drop the packets according to a twelve DSCP value pattern. Some reasonable amount of jitter, delay and loss are tolerated. So it is a good idea for video conferencing.

Table 3: DiffServ AF DSCP values [7]

Drop Precedence	Class 1	Class 2	Class 3	Class 4
Low	AF11	AF21	AF31	AF41
Medium	AF12	AF22	AF32	AF42
High	AF13	AF23	AF33	AF43

Followings are the strengths of DiffServ:

- *Scalability:* Since its flow based independent and traffic is embedded in DSCP. There is no overhead, thus it supports large networks.
- *Simplicity:* It is flexible to cover different kinds of applications.

Shortcomings of DiffServ are:

- *Shortage of admission control:* Because of defect of supplying admission control it cannot guarantee the services.

Bundling IntServ and DiffServ

Based upon former pros and cons of solutions it can be drawn that they complement each other by resolving their limitations. IntServ can accommodate as the edge boundary where admission control is an issue and DiffServ on an areas such as high speed back bones where scaling and traffic aggregation matter. Hence, it can be concluded that integrating IntServ and DiffServ would be a comprehensive solution for all networks.

V. Related Prospect

Gatekeeper

Gatekeeper is an optional tool of video conferencing networks working based on H.323 format. It can provide some compulsory services for terminal gateways and MCUs such as [3]:

- *Address translation:* Translating caller id to IP and vice versa for end points.
- *Bandwidth control:* Deciding about connection bandwidth based on applied configuration .
- *Admission control:* Applying permit/deny access policies defined by administrator to the H.323 network.
- *Zone management:* Traffic management among gatekeepers.

Gatekeeper performs in two modes: directly between two end points or connecting them via itself. Since amount of bandwidth is statically configured on gatekeeper, it doesn't know the capacity of each network device. It is not aware of how the request would be served. On the other hand gatekeeper is only operable on H.323 networks while it is ignorant of available bandwidth. Because of directly connected endpoints it is useless to have decision control mechanism for bandwidth. Due to reasons above, gatekeeper cannot be used as an ideal solution for networks. By using a policy server with the knowledge of network topology, it is possible to provide the gatekeeper by the bandwidth information of network. This will solve the problem of QOS for small H.323 based networks. Nevertheless there are some limitations using gatekeeper. It is becoming more common due to its features. In order to overcome its problems, it can be put together with DiffServ and IntServ as a joint solution [8].

VI. Conclusion

This repot talked about the multimedia application categorization based on kind of interactions, one of them is interactive application which is a kind of human to human communication such as video conferencing.

Video conferencing was introduced as useful tool of communication operating in different model with features like expense saving and *Productivity increment*. Despite the benefits, it is not commonly used because of some lacks, such as costly initial deployment, quality and user acceptance. To resolve the quality issue there are several solutions as it mentioned the most practical and utilized of them is QOS.

QOS as mentioned is a general solution for multimedia application that can be discussed from different viewpoints of different layers (application, system, network). This paper focused on parameters of network level, such as bandwidth, jitter, and delay. While figuring the needs of multimedia applications to these parameters. IntServ and DiffServ were presented as two main ways for QOS implementation, as far as they are not perfect merely, and can resolve each other limitations, their integration seemed to be a favorable solution.

Gatekeeper is an evolving solution offering good facilities, like address translation and admission control. Due to some drawbacks such as bandwidth ignorance, it should be joint to previous solution to perform perfectly.

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