Volume 3, Issue 4, April 2013 ISSN: 2277 128X
International Journal of Advanced Research in Computer Science and Software Engineering
Research Paper
Available online at: www.ijarcsse.com

Quality of Service in Telemedicine Network

Avantika Singh*
Computer Science and Engineering
Amity University, India

Garima Vats
Computer Science and Engineering
Amity University, India

Nidhi Chandra
Computer Science and Engineering
Amity University, India

Abstract— the key focus of this paper is to analyse telemedicine technology, advantages and limitations of telemedicine as well as quality of service parameters in telemedicine network. In telemedicine IP videoconferencing is used and IP videoconferencing is one of such application that requires special attention and service from the network. In telemedicine performance is a critical factor; therefore it requires significant bandwidth with minimal delay, jitter & loss. This paper describes various methods for attaining quality of service (QoS) in a telemedicine network.

Keywords— Telemedicine technology, H.323, SIP, Call flow, jitter control

I. INTRODUCTION

Telemedicine facilitates the provision of medical aid from the distance. It is an effective solution for providing specialty healthcare in the form of improved access and reduced cost to the rural patients. Telemedicine is a "Rapid access to shared and remote medical expertise by means of telecommunications and information technologies no matter where the patient or relevant information is located."[1] According to World Health Organization, telemedicine is defined as, "The delivery of healthcare services, where distance is a critical factor, by all healthcare professionals using information and communication technologies for the exchange of valid information for diagnosis, treatment and prevention of disease and injuries, research and evaluation, and for continuing education of healthcare providers, all in the interests of advancing the health of individuals and their communities".

II. TELEMEDICINE IN INDIA

India is the seventh largest country in the World in area and has the second largest population in the world, but most of the people are living in the rural areas in India. It is the well known fact that 75% of the expertise doctors live in urban areas in India.[2] Most of the neurosurgeons in Chennai city alone exceed those in the entire North Eastern Region of India. This calls for innovative methods of utilization of science and technology for the benefit of the society. Adaptation of Telemedicine technology offers one of the best options for delivering healthcare for rural and geographically distant population spread across India. Telemedicine helps in enabling specialty healthcare to the rural and remote population of India.

III. TELEMEDICINE TECHNOLOGY

Telemedicine is not a single field it is a combination of Communication Technology, Information Technology, Biomedical Engineering and Medical Science. It requires communication and information technology for transmitting information from the patient's end to the doctor's end, similarly it requires biomedical engineering for diagnosing patient and medical science is required for giving expert advice. In actual practice telemedicine is a confluence of Communication Technology, Information Technology, Biomedical Engineering and Medical Science.

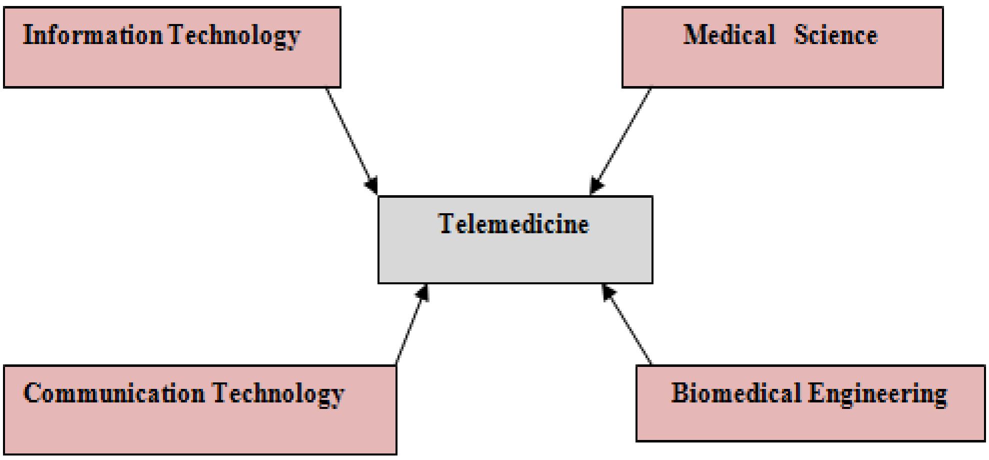

Figure 1: Telemedicine Technology

Telemedicine is the use of electronic information and communication technologies to provide and support health care when distance separates participants. Technical standards and clinical protocols must be associated with the telemedicine technology. Infrastructure requirements of telemedicine are classified into hardware standards and software standards. Hardware Standards includes standards and guidelines for basic telemedicine platform, servers, clinical devices, videoconferencing system, communication hardware and power support while software standards includes operating system, telemedicine software and server software. VSAT, PSTN, ISDN, Leased Line and Wireless LAN/WAN are the various connectivity options for telemedicine services. There are two types of telemedicine services they are:-

A. SYNCHRONOUS (REAL TIME)

In this type of service both the parties need to present at the same time, video conferencing equipments play an important role in this kind of telemedicine applications

B. ASYNCHRONOUS (STORE AND FORWARD)

This type of service allows acquiring medical data and forwarding it to doctor for medical assistance at any convenient time later on, this strategy does not require both the parties to be present at the same time.

TABLE 1. ADVANTAGES AND LIMITATIONS of TELEMEDICINE

Advantages	Limitations
Eliminate distance barriers and improve access to quality health services	The main limitation of telemedicine is lack of telecommunication technologies to provide appropriate quality of service to the telemedicine network
In emergency and critical care situations where moving a patient may be undesirable and/or not feasible	Setting up a telemedicine infrastructure is very costly
Facilitate patients and rural practitioners' access to specialist health services and support	Till now there is no strong provision for telemedicine reimbursement legal laws must be made in this regard
Lessen the inconvenience and/or cost of patient transfers	
Reduce unnecessary travel time for health professionals	
Reduce isolation of rural practice by upgrading their knowledge through tele-education	
Developing virtual communities that interacts and shares knowledge	

IV. **Videoconferencing In Telemedicine**

For supporting synchronous telemedicine applications videoconferencing is required. The two main protocols that are used are H.323 and SIP. Both of these are signalling protocols that are required for establishing connections between the communicating parties[6]. These are standard protocols for real time transfer of audio and video over the network.

TABLE 2. COMPARATIVE ANALYSIS

Distinctive Factors	**H.323**	**SIP**
Compression/decompression	More	Less
Complexity	high	Less
CPU	More Processing	Less
Supported Services	More Explicitly Defined	Less defined
Delay Time	Same	Same
Admission Control	Provided	Not Provided

Maintaining the quality of a videoconferencing call is must in telemedicine. In telemedicine network quality of service is a very important factor; if the quality of network suffers medical advice given by a doctor may lead to disasters in some cases. The networks with guaranteed quality of service (QoS) are highly desirable. Applications like emails, FTP are not sensitive to delay and jitter but this is not the case with the telemedicine applications as they are highly susceptible to delay, loss and jitter. QoS mechanisms provide a set of tools that enables the network to recognize traffic belonging to certain users and applications such that preferential services may be provided to them. QoS does not create an additional capacity on the network. It just helps in maintaining the available resources according to the policies. QoS in telemedicine network guarantee the end-user application a certain level of performance.

V. QOS SOLUTIONS FOR TELEMEDICINE

The goal of IP videoconferencing in a telemedicine network is to provide a quality audio and video experience to the end user. One of the challenges to achieving this goal is getting acceptable service from the network. H.323 gatekeeper is the one component for providing some QoS in the case of H.323 based telemedicine network [3]. An end to end QoS solution for telemedicine network needs endpoints to support resource reservation protocol (RSVP). It is a signaling protocol that is used to request resources from the network. RSVP enables applications to communicate their traffic profile and to request a specific quality of service from the network based on the applications bandwidth and delay requirements. It is not enough for the communicating endpoints to just generate RSVP signaling. It is also necessary to coordinate the functioning of the application with the RSVP signaling.

VI. Call Flow

When placing a call from the patient's end to the doctor's end, the patient's end point communicates its ability to reserve resources to the gatekeeper. The gatekeeper can respond with whether or not it would like the endpoint to attempt resource reservation. In the H. 245 phase the endpoints communicate whether they are capable of resource reservation. Using this knowledge they can make decision as to whether to proceed with the call or not. Once the logical channels have been opened RSVP reservation messages can be sent. Major steps involved are:-

1. At first patient end at the remote location sends the admission request message to the gatekeeper.
2. Then the gatekeeper either sends GCF (gatekeeper admission confirm) or GRJ (gatekeeper admission reject) message back to the patient's end.
3. Then a set-up is established between the patient's end point and the gatekeeper.
4. A set-up is also established between the gatekeeper and the other communicating end i.e. with the doctor's end.
5. Then the doctor's end sends the admission request message to the gatekeeper.
6. In return gatekeeper sends either GCF or GRJ message back to the doctor's end.
7. Then the doctor's end is finally connected to the gatekeeper.
8. Now both the parties that is patient's endpoint and the doctor's endpoint can communicate.

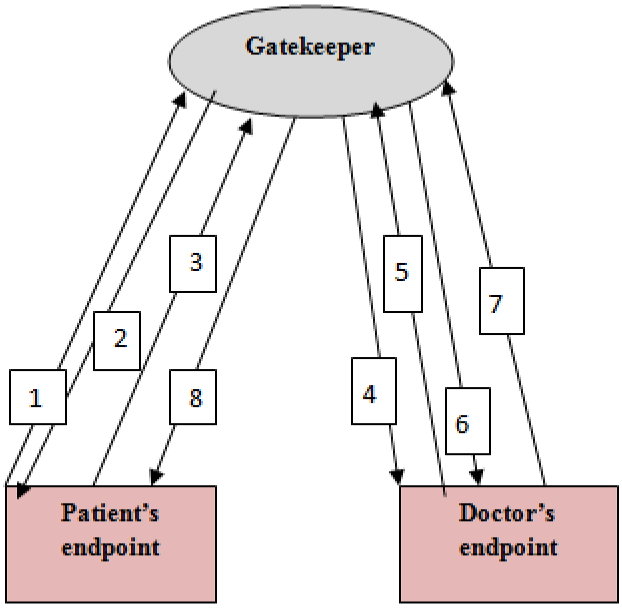

Figure 2: Connection Establishment

This is how H.323 gatekeeper helps in providing QoS solution for telemedicine networks.

VII. Telemedicine Jitter Control

Jitter measures the variability of delay of packets in the given stream, which is an important property for telemedicine networks. In ideal conditions packets should be delivered in a perfectly periodic fashion, but if source generates packets in an uneven manner unavoidable jitter is introduced by the network due to the variable queuing and propagation delays. This jitter introduced in the telemedicine network can drastically reduce the service quality provided by it. Jitter is measured in two ways. One measure called delay jitter which measures the maximum difference in the total delay of different packets. The second measure is called rate jitter which measures the difference in packet delivery rates at various times. In this paper we are addressing only the delay jitter in telemedicine network. Jitter control implementation is usually modelled as follows. The incoming packets are input into the jitter regulator, which reshapes these packets by holding these packets in the internal buffer. Jitter regulator passes these packets to the link-scheduler, which schedules the packet transmission on the output link. There are two types of jitter control algorithms one is on-line algorithms and the other are off-line algorithms. An algorithm is called on-line if its action at time t is a function of the packet arrival times and release times which occur before or at t. Likewise, an algorithm is called off-line if its action can depend on future events also. We are assuming that all the packets in telemedicine application are of equal size. Each packet k

arrives at a fixed time and whenever a packet arrives it is stored in the buffer upon arrival and it is released some time after its arrival. Packets are released in a FIFO order and the time of packet released is governed by the jitter control algorithm. Let s_a (k) is the time at which packet *k* is released by the algorithm A and this algorithm uses a buffer of size B. Then the release time sequence generated by algorithm A using a buffer of size B must satisfy the following equation for all

$0 \leq k \leq n$:

$a(k) \leq s_a(k) \leq a(k+B)$

where *n* is the total number of packets and $a(k) = \infty$ for $k > n$. Here a(k) is the lower bound which specifies that the packet cannot be send before it arrives and a(k+ B) is the upper bound which specifies that when the packet (k+ B) arrives packet *k* must be released due to the FIFOness of the algorithm.

Let X_a is the time difference in perfectly ideal conditions between the arrivals of the different packets. Then time sequence σ is given by:

$\sigma = (t_i)^n_{i=0}$

The delay jitter measures how far off the difference of delivery times of different packets from the ideal time difference in a perfectly periodic sequence, where packets are exactly X_a time units apart. We define delay jitter as[4]:

$J_\sigma = \max_{0 \leq i, k \leq n} (| t_i - t_k - (i - k) X_a |)$

Consider a buffer space 2B. First the buffer is loaded with B packets and when the (B + 1) packet comes the algorithm releases the first buffered packet. From this time the algorithm releases packet *k* after time kX_a, for all $0 \leq k \leq n$

$S_a(k) = a(B) + kX_a$

VIII. Conclusion

Telemedicine has enormous benefits in the sector of human care. Here in this paper we presented various methods by which we can improve the quality of service in the telemedicine network. Network QoS is an important component for successful telemedicine application. With QoS the network administrator is able to control the resource consumption by the application. It is important that any QoS solution for telemedicine application should support all the elements of network quality of service including classification, shaping, and scheduling and admission control. Here we discuss jitter control mechanism in the network and the use of RSVP (Resource Reservation Protocol) and H.323 enabled gatekeepers for enhancing the overall quality of service in the telemedicine network.

XI. Recommendations For Future Scope

Here, our emphasis is mainly on delay jitter control for enhancing the quality of service in telemedicine network, we can also study rate jitter control, latency control methods for further improvement of quality of service in telemedicine network.

References

[1] Ankita S Dwivedi, Dr. Devarshi S Mehta, Dr. Samir Mehta, Dr. Surendra Dwivedi “Telemedicine in India- A case study on TeleElectrocardiograph “,CSI Communications, June 2012

[2] “ISRO- Telemedicine document – healing touch through space”, Published by Publication and Public Relation Unit, ISRO Headquaters, Banglore, February, 2005

[3] Mark Handley , “ Signalling Protocol- Overview”

[4] Jagadish, S.; Manivasakan, R ,“Analysis of jitter control algorithms in QoS networks”, Internet (AH-ICI) Second Asian Himalayas International Conference on 4-6 Nov, 2011

[5] Ismail Dalgic, Hanlin Fang ,“Comparison of H.323 and SIP for IP Telephony Signaling”, Multimedia Systems and Applications System , November 1999.

[6] Subha Dhesikan , “Quality of Service for IP Videoconferencing”, WhitePaper ,Cisco Systems, August 15, 2002

[7] Pavlos Papageorgiou ,“A Comparison of H.323 vs SIP”, University of Maryland at College Park. June 4, 2001

[8] Jill Gemmill, “Network basis for Telemedicine”, Journal of Telemedicine and Telecare Volume 11 Number 2, 2005