Analysis of QoS for DVB-RCS based IP network

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Keywords:

Distance education, Satellite networks, Voice and video transmission, DVB-RCS, Quality of Service

Abstract:

Purpose: Internet has been used to transport data in the form of packet. Multimedia applications including voice and video are being sent using Internet. In the past, Internet did not support any kind of sophisticated quality of service (QoS) mechanism. Although the type of service (ToS) field in the Internet protocol (IP) header has been present and allowed the differentiated treatment of packets, it was never really used on a large scale. IP applications have been mostly used on terrestrial networks so far.

Design/methodology: The use of TCP/IP has made it possible to have two way communication using open standard satellite networks based on digital video broadcast return channel via satellite (DVB-RCS). The voice and video are sensitive to delay and jitter so bandwidth must be guaranteed while transporting it. With the extensive use of IP for carrying voice and video, there is a need to add QoS functionality in satellite networks. The performance of data transfer using voice and video with reference to QoS parameters in satellite network is analyzed in this paper.

Findings: The results show that satellite networks based on DVB-RCS can carry voice and video traffic and offer good quality of service in terms of packet loss and jitter but are poor in quality in terms of packet delay.

Originality/value of paper: This paper describes the architecture of a DVB-RCS based satellite network supporting interactive and multimedia applications.

Type of paper: Technical paper

1. Introduction

The Internet protocol (IP) is the most important fabric of the Internet. IP is designed in such a way that it runs across almost every transmission media and system platform. With the proliferation of hand-held devices equipped with wireless interfaces, the users are increasingly expecting the higher end services. These services include not only voice and data but also multimedia applications which demand high bandwidth. There is also an evolution in communication networks from circuit switched to packet switched and provision of wireless access to Internet at fairly fast rate. The next generation networks are evolving towards all-IP networks.

Internet provides best effort service in which the reliable delivery of packet is important than the amount of time needed for that delivery. This service relies upon native transmission control protocol (TCP) mechanism to re-sequence packets received out of order, as well as to request retransmission of any lost or damaged packets. A number of new applications like voice over IP (VoIP), streaming video, video conferencing and e-commerce have emerged which have varying needs for delay, jitter, bandwidth and packet loss. It is therefore necessary for the network to provide quality of service (QoS) in addition to best effort service to support these applications. The present day satellite networks are supporting real-time multimedia services and integrating with the IP networks. As the IP applications are becoming more prominent, satellite networks are also being used to carry traffic to the remote areas and the areas with the lower degree of infrastructure development where users would otherwise have to wait for broadband access through terrestrial network for several years. The European Telecommunications Standards Institute (ETSI) has recently published an open standard for interactive broadband satellite services, namely DVB-RCS (ETSI, 2005). Satellite systems have been using proprietary air interfaces so far. This standard can be used to provide broadband access and broadcast services to the users who are exchanging voice, video and data based applications.

[Take in Figure 1]

Like any satellite based communication system, DVB-RCS also requires at least three stations, two on the earth and a repeater station on the satellite as shown in figure 1. In space segment, it is the satellite and the transmission paths through the atmosphere. In ground segment there are two stations. The station called "hub" communicates via the satellite and interfaces to the terrestrial networks that the satellite system serves called user terminal. A typical "user terminal" part of DVB-RCS system comprises of the outdoor unit (ODU), the indoor unit (IDU) and the peripherals such as multimedia personal computer, and so on. The DVB-RCS standard allows two-way communication between the user terminals, also called satellite interactive terminals (SIT), installed at customer's sites (Skinnemoen et al., 2004).

We focus on Edusat (Educational Satellite) which is the first exclusive satellite in geo-stationary orbit (GEO) at 74^0E to meet the growing demand for an interactive satellite based education system for India using IP based applications. A transponder on board a satellite acts as a radio relay to provide power over an RF band. Its principal parameters are the bandwidth over which it can operate and the power it can deliver. Resources on board a satellite are limited, so the capacity of a transponder is either bandwidth limited or power limited. The bandwidth of transponders which may be used to carry DVB-RCS channels is typically 36 or 72 MHz. There are six C-band transponders with their footprints covering the entire country. C-band is used particularly in the areas with heavy rainfall that degrades link budgets at higher frequencies. The Edusat is also using six Ku-band transponders for covering different regions of India. Each regional beam is using a Ku-band transponder. There are multiple regional beams covering northern, north-eastern, eastern, southern and western regions and one national beam with its footprint covering the entire Indian mainland as shown in figure 2 (ISRO, 2007).

[Take in Figure 2]

Related Work

Voice communication needs priority over other services in IP environments with limited bandwidth, such as IP satellite networks. Bandwidth utilization in such networks needs to be optimized in order to reduce service costs. The use of dynamic bandwidth allocation schemes responds to the key challenges raised by the VoIP application in satellite environment (Skinnemoen et al., 2005). The use of geostationary multibeams and on-board processing provides a great opportunity for the speedy deployment of real time services such as IP telephony services over satellites. The performance of the VoIP traffic over EuroSkyWay test-bed was evaluated and it was shown that geostationary satellites can carry VoIP traffic (Cruickshank et al., 2001). IP telephony over satellite links and QoS as the enabling technology for combination of data and voice service has been implemented. The level of service quality achieved on LAN and satellite link by using QoS mechanisms available off-the-shelf router and switches have been presented in (Toegl et al., 2005). Providing high-quality multimedia-based modern distance-learning services imposes requirements in terms of cost effectiveness, bandwidth demand and full compliance with the most common networking protocols. ViaSat and Eutelsat have cooperated in the development of an IP-based satellite network to meet these requirements and supply high performance and cost-effective interactivity to end-users via a bidirectional unbalanced satellite link (Feltrin et al., 2003). A new connection admission control algorithm guarantees good QoS for real time multimedia traffic mapped over the service classes of the DVB-RCS standard (Pace et al., 2004). The performance results of laboratory experiments for VoIP over satellite under different link and traffic conditions indicate that VoIP traffic can be transported by satellites without compromising quality (Nguyen et al., 2001).

In this paper, we describe the broadband satellite IP network based on DVB-RCS technology in section 2. The QoS metrics are explained in section 3. We also present the comparison of QoS performance of multimedia traffic with voice and video in a terrestrial network and a satellite based network at 131 locations in India in section 4. Results of the tests performed on the above system are discussed in section 5. Section 6 concludes the paper.

2. IP network technology for Edusat

DVB-RCS systems are based on very small aperture terminals (VSAT). They are one of the easiest ways to provide interactive broadband over satellite, compared to existing terrestrial access networks (based on dial-up modems, leased lines and cable modems, etc.). The natural multicasting characteristics of the satellite, the wireless access and the permanent connectivity are some of the major advantages of DVB-RCS systems. They allow broadband access of the Internet and of value added services via satellite. DVB-RCS enable encapsulation of IP packets in 53 bytes ATM cells or in 180 bytes MPEG-2 transport system (MPEG-TS) at media access control (MAC) layer. To minimize total system costs, DVB-RCS networks are designed to have a single expensive hub and a large number of much smaller remote terminals. Hub is a major station in a satellite communication system that carries terrestrial network services like virtual classroom to and from terminals. The DVB-RCS systems are configured as star so their central station is called a hub. One important function of hub is to map the traffic of all remote terminals belonging to each user group in an efficient way over the satellite. The hub is usually a high performance earth station with an antenna diameter of something between 3 to 6 meters (3.6 meters in Edusat). The hub comprises of a control center for managing the network as well as microwave equipments for the transmission and reception of signals. It also consists of substantial interfacing equipments to support the wide range of terrestrial interfaces. Indian Space Research Organization (ISRO) is presently managing the hub of Edusat stationed at Space Application Centre (SAC), Ahmedabad. The hub is primarily responsible for carrying IP traffic between SITs and other external networks. It is also responsible for overall network management and SIT management.

2.1 The Hub

The DVB-RCS systems support communication on channels in two directions. Forward link (FL) from the satellite to many terminals and return link (RL) from the terminals to the satellite. The DVB-RCS systems use already existing DVB-S standard in the forward link

(FL). In DVB-RCS, forward link is a time division multiplex (TDM) carrier. For IP based connectivity, FL allows high data rate. Presently, FL is configured for 10 million bits per second considering satellite resources available and total traffic expected among all the SITs at Edusat. RL is responsible for carrying the traffic using multi-frequency time division multiple access (MF-TDMA) based on ATM or MPEG standard. The statistical multiplexing allocation mechanism allows the overbooking of return link resources so that satellite bandwidth can be optimized.

DVB-RCS specifies a MAC layer in which the hub controls sharing of the return link capacity among user terminals. The RL based on MF-TDMA, is segmented in frequency and time. The capacity is divided in super frames, made up of frames consisting of fixed time slots. The time slots are dynamically allocated to terminals based on demands (which commensurate with the transmit traffic) and resource availability. Some slots can also be statically allocated to certain terminals. The static or dynamic allocation is performed by the bandwidth scheduler, which implements the MAC protocol. The allocations are regularly broadcast over the forward channel via terminal burst time plan (TBTP) messages. The TBTP contains one or more entries for each terminal, defining the assignment of frequency and time slots. Upon receiving the TBTP messages, the terminals are responsible for calculating the capacity requests and dispatching the traffic from various queues. The bandwidth scheduler in hub assigns the capacity on return link according to the request made by the terminals. It also checks for service level agreement with the user.

When a user terminal requests bandwidth, the scheduler checks the user profile in database. There are four types of bandwidth (capacity) which are considered: continuous rate assignment (CRA), rate based dynamic capacity (RBDC), volume based dynamic capacity (VBDC), and free capacity assignment (FCA). The CRA is used for traffic which requires a guaranteed rate with minimum delay and minimum jitter. It is assigned to a terminal which needs guaranteed bandwidth whether it has the traffic to send or not. In Edusat network, this type of time slot is assigned by hub when particular SIT logs on (switched on). CRA provides guaranteed RL bandwidth and therefore is most suitable for real time traffic like video, voice conferencing and other non-real time applications. RBDC and VBDC are allocated on request and FCA is given as a bonus. RBDC type of time slots are assigned by hub when particular SIT requests for allotment of time slots. RBDC provides RL bandwidth on demand. Hub allots the number of time slots proportional to data rate at the local area network (LAN) ports of SIT. It is suitable for real time traffic like voice conferencing and other non-real time applications like file transfer.

When terminals require additional capacity, the unused capacity is distributed among them. The traffic that supports a small jitter and does not demand guaranteed capacity uses RBDC class. The traffic that can tolerate jitter and requires best effort services is assigned VBDC class. VBDC type of time slots are assigned by hub when particular SIT demands or requests for allotment of time slots. VBDC provides RL bandwidth on demand. Hub allots the number of time slots proportional to data volume at the LAN port of SIT. It is suitable for non-real time applications. In pool of time slots with hub, some time slots are free. The free time slots are assigned by hub to SIT to fulfill the requirement of data transfer. It is like a bonus to SIT and is suitable for non-real time applications. It is important to note that DVB-RCS specification applies to the terminal (SIT) only.

2.2 The user terminals

In contrast to the hub station, the remote terminals are much simpler. The user terminals consist of an outdoor unit (ODU) and an indoor unit (IDU). The ODU consists of three main modules, the DVB-RCS transmitter (solid state power block converter), the DVB-S receiver (low noise block converter), IDU interface and the antenna generally 0.55 to 2.4 m (1.2 m in Edusat) in diameter, which can be wall, roof or ground mounted. Transmitter converts intermediate frequency (IF) (L band 950-1450 MHz) to Ku band frequency and performs high power amplification of L band signal. The receiver performs down conversion to IF (L band 950 MHz-2150 MHz). IDU interface handles all IDU and ODU communication.

Indoor unit (IDU) provides the signal processing (modulation, demodulation, multiplexing, demultiplexing, synchronization and routing) and supports the user interfaces. The IDU box is usually equivalent to the size of a domestic CD player. IDU houses network interface (Ethernet), processing unit, receiver (demodulator), transmitter (modulator), the ODU interface, clock generation unit and power supply. IP packets from MPEG stream received at the satellite interface are extracted by the processing unit and transferred to Ethernet interface for local network. IP packets to the remote LAN are received from the network interface and are embedded into asynchronous transfer mode (ATM) cells before sending them to the modulator. All signaling with the local network, satellite gateway, internal signaling with other devices in IDU and ODU are also handled by processing unit. The demodulator converts the IF signal received from ODU to baseband, performs analog to digital conversion, forwards error correction (FEC) decoding and performs other related operations. The modulator performs frequency up conversion to IF, digital to analog conversion, FEC encoding and other related operations. Processing unit provides the information of FEC encoding type and rate, symbol rate, and decides on what MF-TDMA burst to use for transmission and what IF to use from the CPU to the modulator. This information is extracted from the DVB-RCS tables available on the forward link. Through these tables, the terminal may be asked to change burst parameters (frequency, symbol rate, FEC rate etc.) on a burst-by-burst basis.

The hub provides TCP/IP stack towards the LAN and may include security and TCP enhancement options. The local IP traffic to be sent to remote LAN is first encapsulated to either ATM or MPEG format, then modulated to L-Band frequency. The available modulated signal is fed to SSPB for uplink. Similarly, modulated signal received by LNBC from hub is fed to IDU. The IDU demodulates the data and forwards it to LAN port in the form of IP packets.

3. QoS Metrics

The Internet was originally developed to provide best effort (BE) services to a wide range of data applications generating bursty traffic. Its use for real-time applications raises significant challenges over bandwidth and delay constrained satellite links. Performance requirement of current IP multimedia services differ, in terms of delay, jitter and packet loss. The voice or video streaming applications are error tolerant and these have stringent requirements on transmission delays and jitter. On the other hand, interactive applications like web browsing are very sensitive to losses but can bear considerable delays. IP services and applications have dominated terrestrial networks so far. The space segment is required to be QoS aware to seamlessly integrate with IP terrestrial networks and to efficiently use resources and serve a maximum number of connections. The QoS parameters which are applicable for applications using terrestrial links should also be applicable for satellite links. The following discussion reviews various parameters that provide the QoS performance.

Delay is the time taken by packets to reach from the source to destination. It is typically 270 ms for geostationary orbit (GEO) satellites. The round-trip time (RTT) is defined as the time from the transmission of a packet until a response is received, and is at least twice the forward delay. There will be double hop, when source and destination are connected through satellite.

Jitter is the variations in inter packet arrival time due to variable transmission delay over the network. It is generated in satellite system and IP networks due to their multiple access schemes and variations in router queue loading in time. Jitter must be absorbed at the listener side to play out the voice sample with right timing; otherwise it degrades the voice or video quality. For this reason, dynamic jitter buffering at the listener side is required. Such buffering requires collecting packets and holding them long enough to allow the slowest packet to arrive in time to be played in the correct sequence. This causes additional delays. The jitter buffers add delay, which is used to remove the packet delay variation that each packet is subjected to as it transits through the packet network.

Packet loss is the result of propagation delays and channel impairments (noise, interference). IP networks cannot provide a guarantee that there will be no packet loss and the packet will certainly be delivered in the desired order. Packets may be dropped under peak loads and during periods of congestion caused by link failures or inadequate capacity. Due to the time sensitivity of voice and video transmission, the normal TCP based re-transmission schemes are not suitable. Packet losses greater than 10 percent are generally not acceptable. Receiver-side algorithms may help to conceal individual packet losses but they usually cannot handle bursty losses, more common across wireless links than in wired networks.

In case of above three QoS parameters, the challenge in satellite network is to jointly optimize them based on trade-offs. For instance, larger buffers would help reduce the jitter but add more delay. There are two main QoS models, which are defined for IP based networks: Integrated Services (IntServ) (Wroclawski, 1997) and Differentiated Services (DiffServ) (Blake et al, 1998). In case of IntServ model, the resources are explicitly identified and reserved to guarantee QoS. The applications state their resource requirements ahead of time, enabling the network to anticipate demands and deny new requests if the required resources are not available. This concept implies the use of resource reservation protocol. An application requiring QoS must first setup and reserve resources before the packet is transmitted. Resource Reservation Protocol (RSVP) is a signaling protocol used by Intserv model for per-flow based traffic. In addition to best effort service, Intserv offers two service classes: guaranteed bandwidth and delay service, and controlled load service. Controlled load service approximates the behavior with same QoS that an uncongested packet network would deliver. It is similar to receiving the best effort service under low traffic conditions.

IntServ provides end to end service guarantee, but it is not scalable as it requires the per-flow states to be maintained in all the network nodes along the end to end path. The scalability problem is resolved by replacing per-flow service by per hop behavior (PHB) in DiffServ. Concatenation of multiple PHBs provides end to end performance. There are five PHB classes that have been defined in DiffServ. These are expedited forwarding (EF) (Jacobson et al., 1999) and four classes of assured forwarding (AF) (Heinanen et al., 1999). When packets reach different network nodes in DiffServ domain, they are treated according to their class, which is defined by type of service (ToS) bits in packet header. ToS byte of IP header is redefined as differentiated service code point (DSCP) and it has 6 bits. DiffServ model requires the devices that are able to classify, mark, shape and drop packet which enter and leave DiffServ domain.

If a data and a voice packet have the same priority, the voice packet’s timing is most likely to be disturbed due to the various sizes of data packets and their bursty traffic generation pattern. It is therefore necessary to assign a higher priority to voice packets over data packets to reduce jitter for satellite and IP networks. This is effectively done using DiffServ model. The voice and video packets are required to be mapped into PHB class like EF that provides rate guarantee. This is necessary in order to get priority over the data associated with lower priority service classes. Table 1 shows the suggested mapping for voice and video applications (Cisco Systems, 2004).

[Take in Table 1]

4. Network Architecture

An independent educational WAN on Edusat using SITs at 131 locations all across India (Saxena & Jasola, 2006) has been used to evaluate the QoS performance of voice, video and non-real time IP traffic . This network architecture is shown in figure 3. In this network the teaching end (TE) location generates and transmits multimedia and other IP based traffic related to teaching. The receiving end is defined as classroom end (CE). There is one teaching end and 130 classroom ends in this network. In case of teaching end, the SIT is connected to several personal computers, IP camera and IP telephone, via a LAN. The main characteristic of these satellite SITs is the low cost and small dish size. IDU of SIT is a single channel L-band modem with inbuilt router. Single SIT allows data rate up to 2 Mbps. Each SIT is configured in hub for maximum 624 or 512 Kbps bandwidth. In MF-TDMA each carrier is divided in logical time slots. The hub assigns particular time slots to specific SIT for traffic to be sent from SIT to hub. The SIT transmits either ATM or MPEG cell in time slot in burst form. RL for TE is assigned as 624 kbps in MPEG format. For CE, RL is assigned as 384 kbps in ATM format. Each MPEG carrier is divided in 11 numbers of traffic time slots, where single time slot supports 56.75 kbps data rate. Each TE is configured permanently for CRA in hub with these time slots, and it supports up to 624 kbps RL traffic. Each ATM carrier is divided into 32 numbers of traffic time slots and single time slot supports 16 kbps data rate. Each CE is configured in hub with request based 24 time slots, i.e. 4 RDBC and 20 VBDC, and it supports up to 384 kbps RL traffic.

The use of TCP/IP provides the network with the facility to use file transfer, online and offline multimedia transfer, application sharing, taking remote control of user's computer and access to Internet. Different types of applications have been used to analyze the QoS over Edusat based satellite network and terrestrial network. As the focus is on education, real time multimedia content is used. The applications like video conferencing, voice over IP (VoIP), video on demand, web pages, slides, e-mail and chat are used. Our tests included the H.326 tools for point to point video conferencing and VoIP using different codec standards. H.261 and H.263 have been used for video and G.711 and G.729 for audio. G.711 and G.729 are widely used International Telecommunication Union (ITU) recommendations for audio codec standards. G.711 defines the 64 kb/s pulse code modulation (PCM) format, which provides no algorithmic gain, only some logarithmic compression in amplitude. In contrast, G.729 compresses the voice down to 8 kb/s with close to toll quality. When SIT is used, audio and video interaction using video conferencing can be achieved. Video conferencing can be initiated from TE for direct interactions with the other end. The CE is used to receive live lectures and interact with TE through live video conference session. VLC (video LAN client) media player and NetMeeting have been used. The tests are based on 600 samples each of file transfer, video conferencing and VoIP sessions between the users connected via two different SITs, one at teaching end and other at classroom end.

[Take in Figure 3]

Offline access of content can be done through online repositories. In case of TE, multicast facility is enabled and CRA with 624 kbps RL bandwidth is allotted for multicast and other traffic. However, for CE, multicast facility is disabled. In case of video streaming, high quality is associated with rich content distribution. When this rich content is distributed in multicast mode and low rate return channel is used for interactivity, the complexity is increased. The satellite networks are best suitable for multicast, so the focus is on multicast delivery. There are several possibilities for content contribution and delivery based on Edusat network. The distribution server is connected to the hub through a terrestrial link. The contribution is transmitted in unicast mode using leased link and transmission of multimedia content through forward link in multicast mode. In other case, distribution server is kept at the teaching end and contribution can be collected from other classroom ends. The distribution of multimedia content is done in the same way as in the earlier case. Low cost devices like PCI cards or small LAN routers are used to receive the same content contribution and delivery by the network. Besides multimedia real time applications, other non-real time applications for data transfer, such as FTP, are used. Large files of 2 Mbps have been sent and received from TE to CE and vice-versa.

5. Results analysis

Series of data transfers have been performed for measuring the delay, jitter and packet loss in various real time multimedia and non-real time applications mentioned in previous section. The results are shown in figures 4 and 5. Each point in the graphs in the figures represents the average of at least 100 data transfers. The results have been compared for previously mentioned applications for satellite link as well as for terrestrial links. There are several locations which are also connected through Internet via terrestrial links of 512 kbps each. The applications have been run on these terrestrial links and satellite network.

The delay across terrestrial links and satellite links for three different applications is shown in figures 4 and 5 respectively. In case of satellite links, the delay imposed is very high as shown in figure 5 in comparison to terrestrial link which is shown in figure 4.

[Take in Figure 4]

[Take in Figure 5]

TCP protocol is severely affected by the significant propagation delay in geostationary satellite based networks. Packets are stored in a buffer for future retransmission as and when they are received. When packets are acknowledged, they are removed from the buffer. The acknowledgement time is increased because of the long delay inherent in satellite links. In case of Edusat, double satellite hops between a teaching and classroom end increase the average RTT as shown in figure 5. The UDP is also used to transport data in multicast mode to allow one to many data transfers. When UDP is used, a higher throughput is achieved, but it is necessary to increase the reliability. In the satellite links, the delay is compensated by the bandwidth. In specific case of VoIP, more voice frames are packetized in each IP packet and it affects the MPEG encapsulation efficiency. For all these applications, the physical delay produced by satellite link did not have any noticeable effect. In practical terms, this significant delay has some impact on bidirectional and interactive connections even with reliable protocols such as TCP. As most of the applications involve personal computers (PCs), the choice of application highly influences the end to end delay. For example, Microsoft NetMeeting uses de-filtering buffers.

The jitter values for all the three applications are shown in figures 6 and 7. Although voice packets are given higher priority over data packets, a large data packet may disrupt the timing of voice packets and create jitter. It can be seen that the jitter values are very low in case of satellite links, about 15 ms, as shown in figure 7.

[Take in Figure 6]

[Take in Figure 7]

The packet losses in terrestrial links and satellite links are shown in figures 8 and figure 9 respectively. In case of satellite link, loss is less in comparison to terrestrial link. IP packet loss is due to header corruption. In case of voice and video, this loss is higher than IP packet loss. This is because of damaged payload, so if an IP packet is received because its header was intact, its damaged payload may cause loss of voice or video frames. One more reason for packets loss in video and voice traffic is the late arrival of these packets at destination rendering these packets unusable by the voice and video decoders.

[Take in Figure 8]

[Take in Figure 9]

6. Conclusion

The architecture of a DVB-RCS based satellite network supporting interactive and multimedia applications is described in this paper. The star topology of the Edusat makes it possible to concentrate the complexity in the hub and reduce the complexity and cost in user terminals. The QoS performance of the real time and non-real time multimedia applications, such as video conferencing, voice over IP, video on demand and data transfer on Edusat network have been shown. Test performed with these applications have indicated acceptable jitter and packet loss but a high round trip time due to double satellite hop as the major limitation of such network. This delay impairs the quality of highly interactive multimedia applications.

The future research in this area can include the mobility aspect of the users who are connected to the user terminals. Performance of these new scenarios for QoS issues needs to be evaluated. The future area of research can also be related to the TCP parameters like congestion avoidance and Queue management. The user terminals are also being installed in the remote areas where other communication facilities are limited. The social and cultural impact of the satellite based education in these areas should also be explored.

References

Blake, S., Blake, D, Carlson, M., Davies, E, Wang, Z., & Weiss, W., (1998), "An Architecture for Differentiated Services", *IETF*, RFC 2475, Retrieved October 29, 2007, from http://www.ietf.org/rfc/rfc2475.txt.

Cisco Systems, (2004), “Cisco IP Telephony Solution Reference Network Design”, Cisco Systems, Inc, San Jose, USA, Retrieved October 29, 2007, from http://www.cisco.com/application/pdf/en/us/guest/netsol/ns268/c649/ccmigration_09186a0080479538.pdf.

Cruickshank, H., Sun, Z., Sanchez, A., & Carducci, F., (2001), “Analysis of IP voice conferencing over EuroSkyWay Satellite system”, *IEE Proceeding-Communications*, Vol. 148, No. 4, pp. 202-206.

ETSI, (2005), “Digital Video Broadcasting (DVB); Interactive Channel for Satellite Distribution Systems”, Report ETSI EN 301 790, V l. 4.1, Retrieved October 29, 2007, from http://pda.etsi.org/pda/home.asp?wki_id=G6RH4V2UWXFHGJMOLi_CR.

Feltrin, E., Guinamand, Y., Weller, E., & Oweis, L., (2003), “An IP based satellite network for distance learning”, in Ansari, N., Deek, F., Lin, C.-Y., & Yu, H. (Eds.), *Proceedings of the International Conference on Information Technology: Research and Education*, IEEE, Piscataway, NJ, USA, pp. 416-420.

Heinanen, J., Baker, F., Weiss, W., & Wroclawski, J., (1999), “Assured Forwarding PHB Group”, *IETF*, RFC 2597, Retrieved October 29, 2007, from http://www.ietf.org/rfc/rfc2597.txt.

ISRO (2007). “EDUSAT CSLV-F01”, Retrieved October 29, 2007, from, http://www.isro.org/edusat.

Jacobson, V., Nichols, K., & Poduri, K., (1999), “An Expedited Forwarding PHB”, *IETF*, RFC 2598, Retrieved October 29, 2007, from http://www.ietf.org/rfc/rfc2598.txt.

Nguyen, T., Yegenoglu, F., & Sciuto, A., (2001), “Voice over IP Service and Performance in Satellite Networks”, *IEEE Communications Magazine*, Vol. 39, No. 3, pp. 164-171.

Pace, P., Aloi, G., & Marano, S., (2004), “Efficient Real-Time Multimedia Connections Handling over DVB-RCS Satellite”, *Proceedings of Global Telecommunications Conference 2004,* IEEE Communications Society, Piscataway, NJ, USA, pp. 2722-2727.

Saxena, P. C., & Jasola, S., (2006), “Edusat: Taking Quality Education to unreached”, *Proceedings of 12th International Conference on Terrestrial and Satellite broadcast*, Broadcast Engineering Society, New Delhi, India, pp. 127-130.

Skinnemoen, H., Leirvik, R., Hetland, J., Fanebust, H., & Paxal, V., (2004), “Interactive IP network via satellite DVB-RCS”, *IEEE Journal on Selected Areas in Communications,* Vol. 22, pp. 508-517.

Skinnemoen, H., Vermesan, A., Luoras, A., Adams, G., & Lobao, X., (2005), “VoIP over DVB-RCS with QoS and bandwidth on demand”, *IEEE Wireless Communications*, Vol. 12, pp. 46-53.

Toegl, R., Birnbacher, U., & Koudelka, O., (2005), “Deploying IP telephony over Satellite Links”, *IEEE Proc of 2nd International Symposium on Wireless Communications Systems* (ISWCS'2005), Siena, Italy, pp. 624-628.

Wroclawski, J., (1997), “The Use of RSVP with IETF Integrated Services”, *IETF*, RFC 2210, Retrieved October 29, 2007, from http://www.ietf.org/rfc/rfc2210.txt.

Traffic Type	IP precedence	DSCP	PHB
Voice RTP	5	46	EF
Voice control	3	24	CS3
Video conferencing	4	34	41
Data	0, 1, 2	10 to 22	BE to AF 23

Table 1 :IP Traffic mapping guidelines

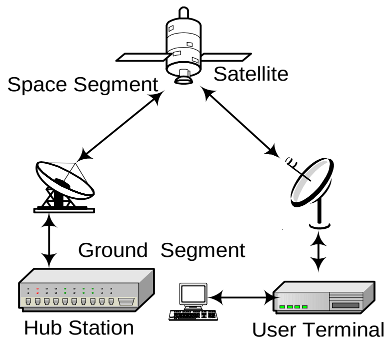

Figure 1: DVB-RCS system

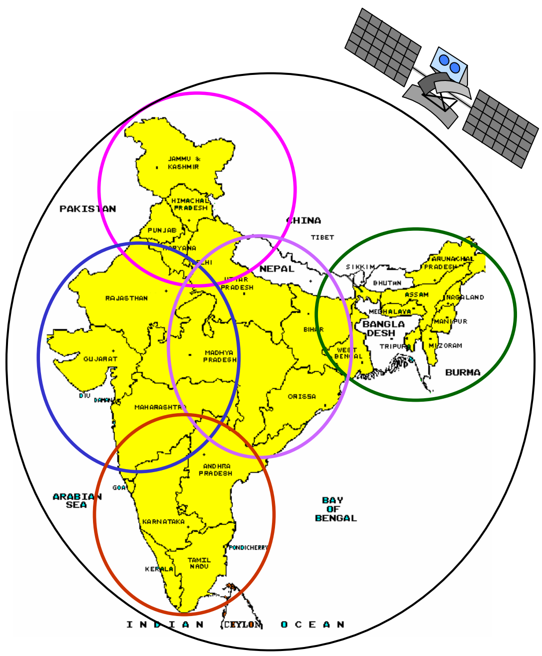

Figure 2 Edusat Coverage

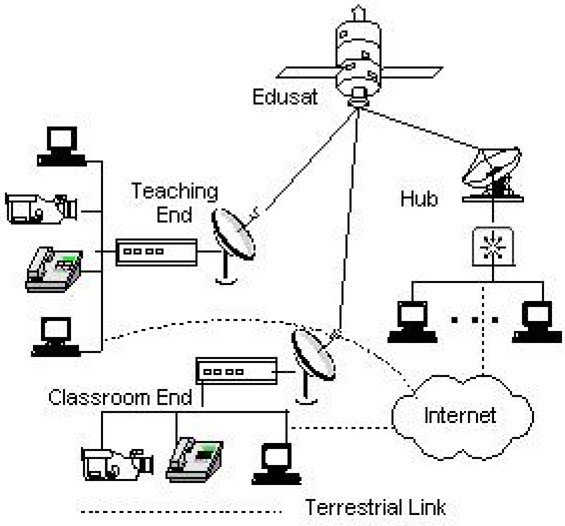

Figure 3 Network architecture

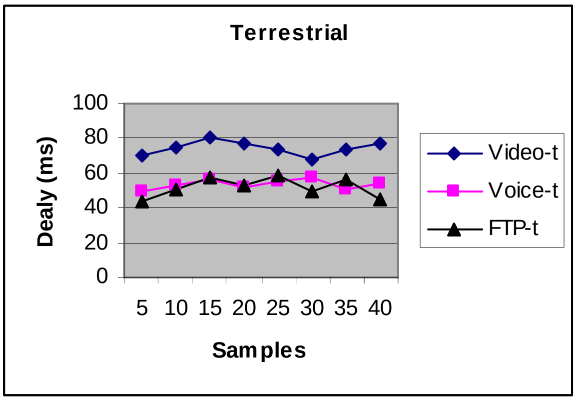

Figure 4 Delay across terrestrial links

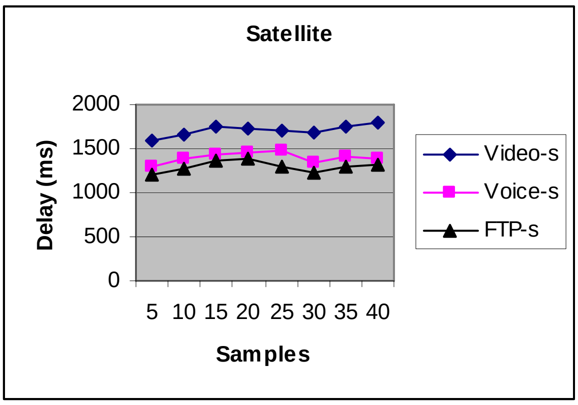

Figure 5 Delay across satellite links

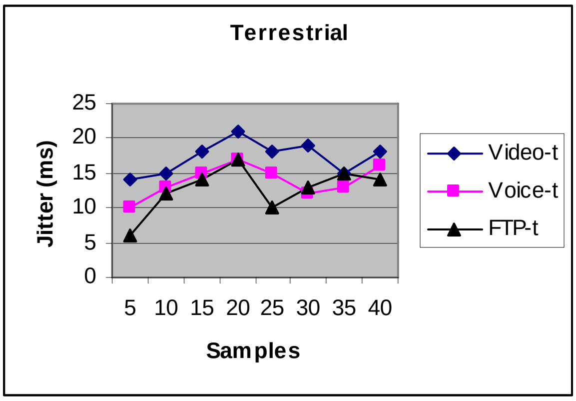

Figure 6 Jitter experienced on terrestrial links

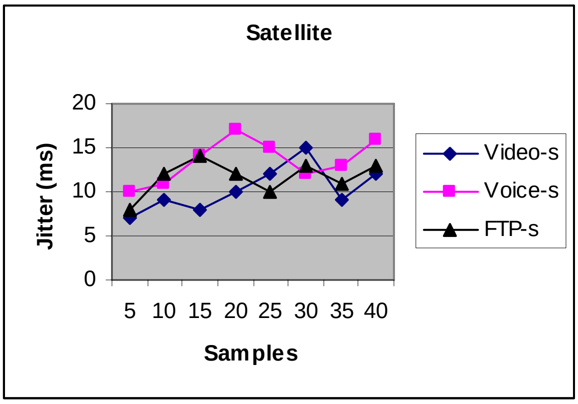

Figure 7 Jitter experienced on satellite links

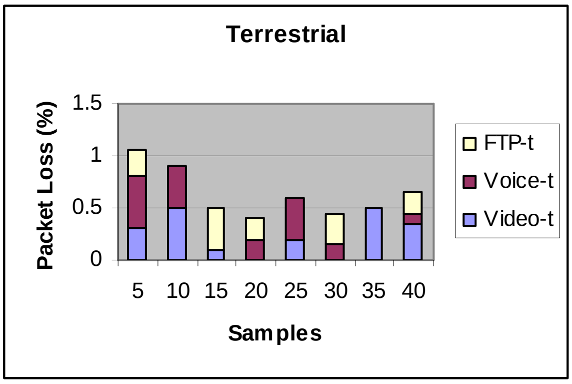

Figure 8 Packet losses in terrestrial links

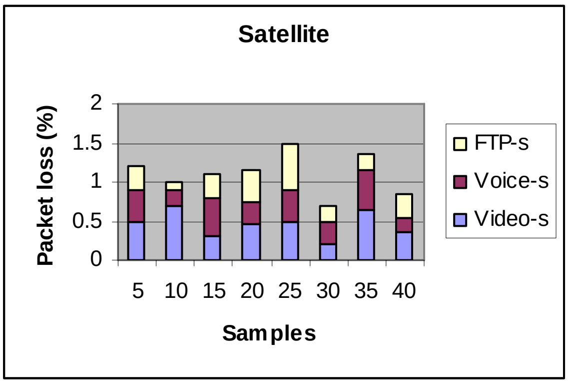

Figure 9 Packet losses in satellite links