

Delivery of Multimedia Services using Broadband Satellite Networks

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Abstract

Satellite systems provide a solution to broadband services that is easily deployed and available across a wide geographical coverage. This can provide a significant contribution to reducing or eliminating the digital divide for users across Europe. This paper provides an overview of key technologies that will enhance the quality of satellite networks and allow closer integration of satellite networks within the next generation of multimedia-enabled Internet. The paper is based on work performed in the EC funded SatSix project.

1. Introduction

Recent years have seen an exponential growth in multimedia applications within the Internet as a whole, with a general move towards support for triple-play services that combine voice, video (e.g. TV, teleconference) and data services, all delivered over the same network. Efficient multimedia delivery requires high speed broadband networks. Unfortunately at this time, such broadband services are not available in all regions within the EU due to geography (e.g. mountains), living in small rural communities, or peripheral regions, creating a "digital divide". The SatSix project was initiated by the EC to reduce this European digital divide by enabling efficient delivery of broadband services using satellite links. The project promotes the introduction of the IPv6 protocol into satellite-based communication systems that employ the established European Digital Video Broadcast standards for satellite systems (DVB-RCS[1], and DVB-S2 [2]).

Such new satellite systems can offer an attractive solution to provide broadband networks to:

- access all types of content via the Internet and other distributed networks (e.g. Virtual Private networks).
- access by all types of users: directly, via private local networks (for Corporate and SME users), or shared local networks (WiFi or WiMax, LAN etc.)

SatSix is investigating key techniques; at the network, access and transmission level. Wireless Local Loops, WLL, with a satellite access may be deployed together with enhanced network services including mobility, multicast, security [15] and QoS [16]. Cost/performance will be optimized using enhanced adaptive physical layer methods,

a new optimized encapsulation, cross-layer optimizations and enhanced transport protocols. The research programme also considers opportunities, such as hybrid satellite payloads (combining transparent and regenerative satellites) and support for mobility.

Key issues that will impact the design of the emerging generation of satellite equipment include:

- Lowered cost of broadband access by developing new satellite access techniques (e.g. DVB-S2 to adaptively tailor performance to prevailing link conditions).
- Investigation of techniques that enable sharing access costs, such as integration with WiFi and WiMax.
- Closer integration with Next Generation Networks, piloting IPv6.
- New multimedia applications for both corporate and consumer markets (including 3-play and TV over DVB-S2).
- Investigation of new congestion control protocols such as the DCCP for multimedia traffic, so they cooperate fairly with existing Internet traffic.

Section 2 presents a brief introduction to the DVB-RCS satellite architecture that it is proposed to implement for the SatSix project. Section 3 gives an overview of the triple play services to be provided within satellite architecture. Section 4 discusses the need for congestion control protocols to carry multimedia traffic. Section 5 discusses some of the simulations carried out on the TFRC and DCCP protocol, followed by conclusions and future work.

2. DVB-RCS

The SatSix Project will use terminals built according to the Digital Video Broadcasting Return Channel System (DVB-RCS) [1] standard. This is a key European standard that specifies a two-way channel for GEO satellite based networks. This now forms the basis of a standards-based IP satellite network.

Current DVB-RCS systems utilize a forward satellite link using the ISO MPEG-2 Transport Stream (TS), in common with current-generation digital TV services. Work is in progress to extend the forward link to utilize the new DVB-S2 standard that supports advanced physical layers with the opportunity to utilize adaptive coding/modulation. Both DVB-S and DVB-S2 can support the MPEG-TS, but the

latter also provides an opportunity to define a new encapsulation method tailored to the IP-service and designed to offer improved performance when used in combination with adaptive coding/modulation providing dynamic rate adaptation.

2.1 An Example DVB RCS Satellite System

Most current RCS systems use traditional transparent satellites, efficient for a centralized star network, e.g. for access to broadband terrestrial internet services, but incurring a double-hop satellite delay, making it unattractive for multimedia communication between user terminals.

AmerHis [18,19,20,21], a joint project of the European Space Agency (ESA), CDTI and Hispasat, was designed as a response to the growing demand in multimedia broadband services and the adaptation of real time services to the satellite world. It represents the first operational regenerative, on-board processing (OBP), DVB-RCS satellite switching. OBP greatly reduces the satellite delay by reducing the double hop nature of the current RCS systems into a single hop architecture.

The AmerHis system includes the following elements in its architecture (refer Figure 1):

- OBP (On-Board Processor)
- MS (Management Station).
- RCST (Return Channel Satellite Terminal)
- RSGW (Regenerative Satellite Gateway)
- VSP (Video Service Provider)
- STB (Set-Top Box)

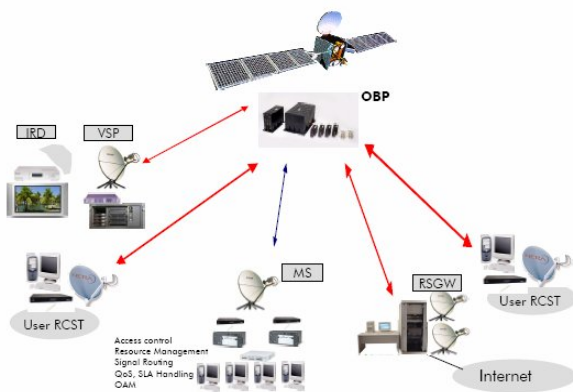


Figure 1: AmerHis: System Architecture

The MS provides some functions of the Network Control Centre (NCC) in a traditional DVB-RCS system, for instance it determine how satellite uplink capacity is allocated to individual RCSTs within the network.

SatSix partners including AAS-E, AAS-F, and Hispasat are fully committed to the standardization process of DVB-RCS. The latest version of the standard now includes DVB-S2 on the forward link and the capability to support Dynamic Rate Adaptation (DRA) and Adaptive Coding and Modulation (ACM) both on the forward and the return link.

SatSix will consider AMERHIS regenerative architecture and also the traditional transparent architecture. Future hybrid systems would provide the benefit of both regenerative (low delay) and transparent (efficient broadband) architectures, plus the support of DVB-S2.

3. Satellite Services Description

3.1 Triple Play

The availability of additional capacity presents new opportunities for convergence of TV, multimedia and data (Internet) services into an integrated service, in a broadband bundling now often known as “triple play”. This service is recognized by incumbent operators as a potential “killer application” to counteract the actions of competitors and to allow the creation of broadband services that can increase the Average Revenue Per user (ARPU).

3.1.1 Data Services

IP transmission is a key service and is supported not just for TCP-based data services (internet access, file transfer, etc) but as the primary service for voice and video (as described in the following two sections). IP flows can be assigned a different QoS depending on the user requests.

Delivery of broadband services over the satellite network, demands that the satellite network should become an integral part that may be easily integrated with the fixed and mobile Internet. This requires the networking protocols used by the multimedia services should cooperate in harmony with the existing protocols that are used for data services. This is a challenging requirement both in terms of the need to offer a quality of service that meets the needs of multimedia applications (in terms of required bandwidth, minimal jitter, maximum delay, etc), and the consumption of only a fair share of the available capacity that is available over the end-to-end Internet Path.

3.1.2 Video Services

Most current TV services use the MPEG-2 for TV distribution, e.g. based on the standards produced by the DVB Forum. DVB-S use the MPEG-2 TS as a link technology that allows video programmes to be separately sent (MPEG-2 Single Program Transport Stream, SPTS) or more commonly combined into a single transmission (Multiple Program Transport Stream, MPTS). The MPEG-2 video codec is still widely used, although new compression algorithms are being considered such as H.264/AVC (standard MPEG-4 part 10, Advanced Video Coding) or the Windows Media 9.

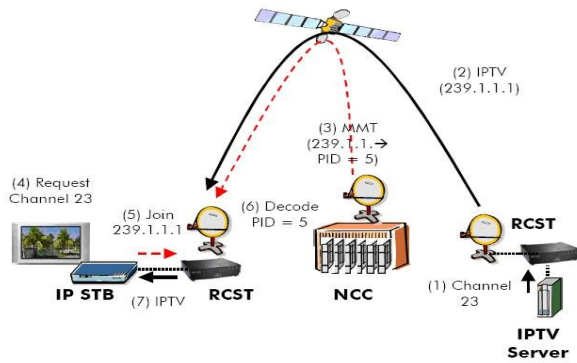


Figure 2: Multicast IPTV scenario

TVoIP (IPTV) may also be sent over a MPEG-2 TS as a data service. This requires that the video is encapsulated independently and therefore the MPEG2-TS contain only one program (SPTS). A unicast transport may also be used between the server and the setup box (STB). The files are stored in the server and the SPTS is currently encapsulated in UDP/IP to the unicast address of the STB. In the case of Video on Demand (VoD), the video delivery normally follows a user request to join a video stream (Figure 2). To exploit the multicast capabilities of satellite a user can be logically attached to a particular TVoIP stream, using IP multicast to transport the UDP/IP data.

In the frame of the SATSIX Project, AAS-E and AAS-F will work on the definition of a new generation satellite network that will include a hybrid OBP that will combine transparent and regenerative payloads. The downlink will use DVB-S2, with an increase in bandwidth of up to 30%. The use of advanced physical layer methods (DRA and ACM) allows the introduction of lower data rates protecting the link in the case of heavy rain, reducing the cost of the terminal. The introduction of dynamic multicast will also provide important resource savings in terms of bandwidth, since the multicast flow is sent only once and replicated on board towards each beams with a terminal that requires to receive a particular multicast flow.

This combination of techniques will allow SatSix to realize new services, including IPTV, VoD , Near VoD and interactive TV and enable Triple Play service.

3.1.2 Voice Services

Traditional transparent satellite systems require all communication to pass via the hub/gateway. This introduces a double satellite delay when terminals wish to communicate directly with one another (i.e. at least, 550 ms of delay each direction). This double-hop delay is not attractive when offering a telephony service and can not reach a quality perception that is near to that of the terrestrial PSTN/ISDN.

OBP technology enables an alternative network design, which eliminates the need data to pass via a hub station. A mesh communication (two users connected via different

satellite terminals) therefore has a transmission delay of around 270 ms, which is within the goal for good quality perception (400 ms) for VoIP communication. This allows the introduction of all types of voice communications without a loss in quality.

The bandwidth –sharing algorithms used for the satellite up-link can introduce considerable jitter, unless carefully controlled. Jitter is therefore also a concern for VoIP services and can be reduced by cross-layer methods that allow an RCST to detect the signaling messages associated with a UDP flow that carries VoIP. The RCST may inform the NCC to adjust the capacity allocation plan to cater for the QoS requirements of the flow, by allocating capacity so that the bursts are equally distributed during the in the transmission of the RCST.

The following connections between endpoints is possible:

- In the satellite network using one-hop mesh connectivity and VoIP (H.323 and SIP compatible).
- With PSTN/ISDN networks using the interconnection capabilities of the RSGW.
- With Internet users with VoIP and data connectivity.

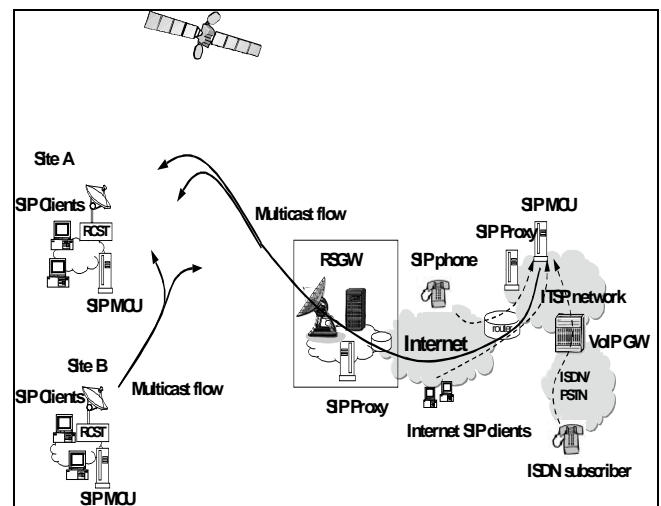


Figure 3: Star MCU scenario with external ITSP

The audio (and video) multiconference services require the use of multicast MCUs (Multipoint Control Unit), but also take advantage of the satellite multicast platform to save bandwidth resources. Terminals may be SIP phones or standard SIP clients registering at a RSGW SIP proxy.

4. QoS for Multimedia

TCP can introduce an arbitrary delay because of its reliability and in-order delivery requirements, making it unsuitable for real-time media. Most current multimedia applications use UDP as the underlying transport protocol to deliver its packets, chosen in preference to TCP. UDP however offers no performance guarantees.

Performance may be controlled by invoking network-layer QoS [16] (e.g. using the IETF Integrated Services or Differentiated Services model). The Integrated Services

approach requires support in the layer 2 network (ie. at the DVB-RCS RCST) to enable satellite capacity to be reserved for individual packet flows. Changes to the RCS specification and the network-layer interface of equipment will allow services to emerge for corporate applications [16]. This approach has however not gained widespread use within the general Internet, due to the need to establish policing policies between networks and the need to deploy complex control protocols in routers.

Most current Internet traffic is carried using either the best effort service or Differentiated Services. In this model many traffic flows share the same network capacity, and a method known as congestion control is required to fairly distribute the available capacity between the various flows. This is a standard feature of TCP [13], but is not provided by many current applications that use UDP for multimedia. Current UDP-based applications can (and often do) transmit at a constant rate, irrespective of the available capacity. Growth of such long-lived non-congestion-controlled traffic posed a real threat to the overall health of the Internet [7], and in particular form an obstacle to the deployment of triple-play services over bandwidth-limited technologies such as satellite systems.

5. Congestion Control

5.1 TCP Friendly Rate Control (TFRC):

TFRC [6] was designed to allow multimedia applications to provide congestion control when using UDP with RTP [4]. In TFRC, the receiver periodically sends a feedback report informing the recent loss event rate for the connection, from which the sender calculates the allowed sending rate for the next RTT period by using an equation that models the equivalent throughput that would have been obtained by a TCP flow. TFRC is suitable for multimedia applications such as TVoIP and VoIP since it allows the sending rate to vary more smoothly by decreasing and increasing the sending rate gradually. It can be used for both unicast and multicast traffic.

TFRC was designed to be reasonably fair when competing for capacity with TCP connections that use a fixed packet size. This has consequences when used for voice services that utilize small IP packets: a low rate TFRC flow sharing a network capacity (e.g. the satellite link) with high-bandwidth TCP flows (large-packets) may need to slow down, even though the nominal rate of the TFRC flow is less than the rate achieved by the TCP flows.

To enable TFRC to be used for Voice traffic, TFRC-SP [12], a Small-Packet (SP) variant of TFRC was designed. TFRC-SP achieves the same capacity as a TCP flow using packets of up to 1500 bytes. It also enforces a minimum interval of 10 ms between packets, to prevent a single flow sending packets too fast. TFRC and TFRC-SP are sufficient to provide congestion control to UDP multimedia, however the application designer needs to implement these protocols correctly. They also seek to provide only congestion control, and not other important services such as feature-negotiation,

firewall traversal and mobility support.

5.2 Datagram Congestion Control Protocol (DCCP):

To provide a standard way to introduce congestion control into multimedia applications, the IETF defined the DCCP transport protocol [8] as an alternative to UDP for unicast multimedia applications that prefer timeliness of data to reliability. This new transport protocol is designed for deployment as a standard feature in end hosts (PCs, VoIP codecs, and other internet-enabled multimedia appliances).

DCCP is intended for applications that require the flow-based semantics of TCP, but which do not want TCP's in-order delivery and reliability (e.g. TVoIP, VoIP, VoD). It provides a comprehensive set of features for the multimedia enabled Internet:

- DCCP provides a standard way to implement congestion control and congestion control negotiation for multimedia applications, according to three profiles:
 - CCID 2 [9] - TCP SACK-like Congestion Control, appropriate for flows that like to receive as much as possible over the long term (e.g. download of streaming content).
 - CCID 3 [10] - TFRC Congestion Control, appropriate for flows that prefer to minimize abrupt changes in the sending rate, including conversational streaming media applications.
 - CCID 4 [12] - TFRC Congestion Control for Small Packets, appropriate for flows that prefer to minimize abrupt changes in the sending rate, but that generate small packets at a low rate, including VoIP.
- DCCP provides an interface to other standards-based network techniques such as Explicit Congestion Notification, ECN, and Path MTU Discovery.
- DCCP implements reliable connection setup and teardown enabling firewall traversal.

DCCP also provides facilitates to negotiate connection properties between end nodes and copious space for options, allowing new features to be added as the needs of applications becomes better understood.

The SatSix project will provide implementation experience of DCCP with IPv6 and will study the implications of using DCCP in an Internet environment that includes next generation satellite links. This work will assess how the satellite link should be adapted to optimise end-to-end performance, and the impact this has on key applications.

6. Simulation

This section presents results to illustrate the congestion control behavior of both TFRC-SP and DCCP-CCID 4. The network simulator ns-2 was used [17] to simulate a scenario that includes a large delay satellite link connected to the Internet, using the topology shown in Figure 4. The bottleneck (between R1 and R2) represents the DVB-RCS satellite link with variable capacity (up to 2 Mbps) and delay

(at least 300ms delay, the approximate forward link delay in a DVB-RCS system). We modeled our application traffic using constant bit rate (CBR) traffic. Results are presented for both single and multiple IP flows.



Figure 4: Simulation Topology

The first sets of tests considered both TFRC-SP and CCID 4 carrying streaming audio. The audio codec was assumed to send 50 packets per second each of 160 bytes (64kpbs).

Figure 5 presents the results of TFRC-SP when there are no packet losses and also in the presence of packet loss. 1% of packets were dropped. In the absence of packet loss, TFRC steadily increases the rate up to the media encoding rate and maintains this rate. With packet loss, TFRC steadily decreases the rate appropriately at the same time increasing the rate gradually in the absence of packet loss. Thus it is clear that the TFRC protocol is reactive to a reduction of the capacity available on the satellite link and alters its sending rate accordingly.

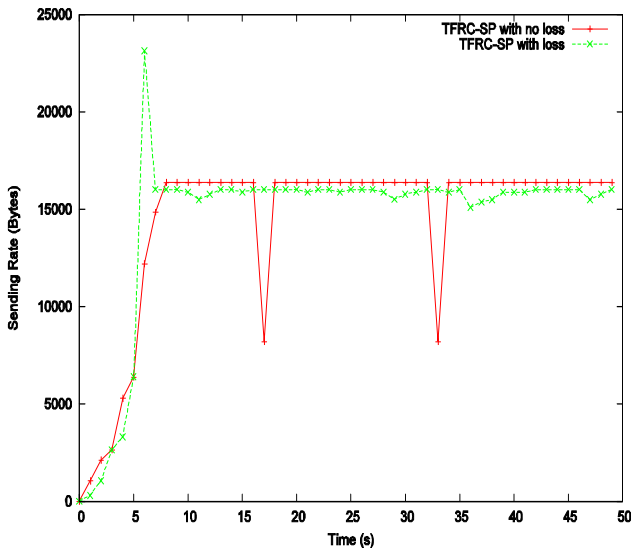


Figure 5: TFRC-SP Sending Rate Vs Time. With both packet loss and no loss. Packet loss rate = 1%.

Figure 6 presents the result for DCCP CCID 4 using small packets during both presence and absence of loss of packets. These results show that CCID 4 performs similarly to the TFRC protocol described above.

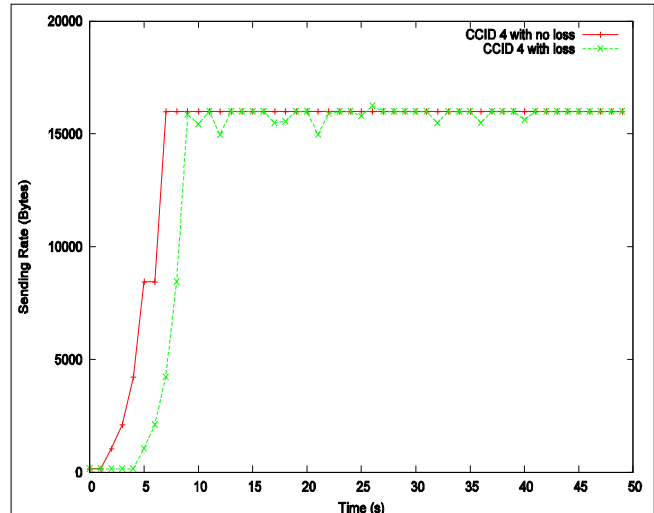


Figure 6: DCCP CCID 4 Sending Rate Vs Time. With both packet loss and no loss. Packet loss rate = 1%.

The next set of results consider the fairness of DCCP CCID 4 when the satellite capacity used for multimedia streaming is shared with other TCP flows. To verify the fairness, we first simulate a scenario where multiple UDP flows carrying streaming video cooperated along with a single TCP flow. The video application sent 50 packets per second each packet having a fixed 1000 bytes of data (400 kbps - a low-rate video codec). The single TCP flow carried FTP traffic having 1460 bytes packets. The maximum congestion window (cwnd) was set 40 packets per RTT. Initially the test used one single UDP flow and one single TCP flow. Tests were then repeated by gradually increasing the UDP flow up to 8 flows.

Figure 7 shows that as the number of UDP flows is increased, the throughput of the single TCP flow gradually reduces. When the number of UDP flows is 8, the single TCP is deprived of its fair share of the bandwidth and is unable to increase its cwnd appropriately. When such multimedia applications use UDP they would not fairly cooperate with existing TCP flows unless they use appropriate congestion control mechanism, such as TFRC.

The tests were repeated by replacing the UDP flows by DCCP CCID 4 flows. Figure 8 shows that in the presence of CCID 4 flows, the single TCP flow is still able to recover from its losses even though it encounters packet losses, and gradually increase the cwnd appropriately.

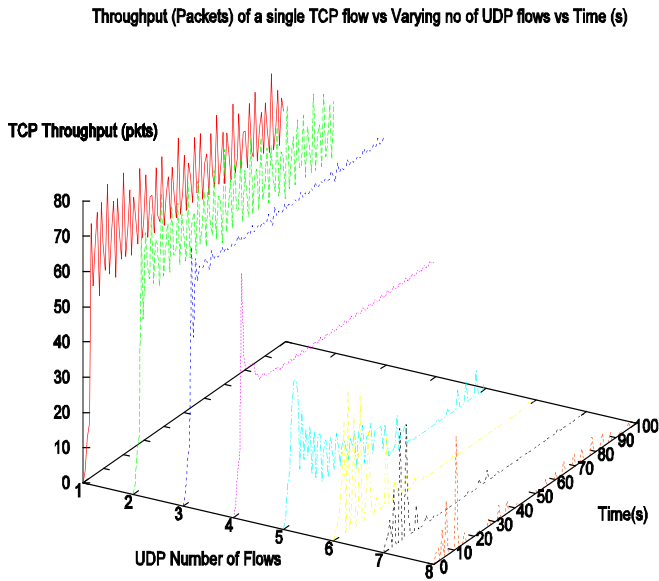


Figure 7: Throughput (packets) of a single TCP flow (vs) Varying no of UDP flows (vs) Time (s). The x-axis gives the performance of the TCP flow when the number of UDP flows increase from 1 to 8 flows.

DCCP CCID 4 flows gradually decrease their sending rate on experiencing congestion, based on their reported loss event rates. This enables cooperating TCP flow to gradually achieve a fair share (in contrast to normal UDP).

These simulations demonstrate that DCCP enables multimedia flows to coexist with TCP traffic within the same QoS class without causing unfair sharing of network capacity. This fairness is important when deploying a mixture of voice, video and data services, and particularly important for links with limited and varying capacity, such as those offered by broadband satellite systems. As such, the deployment of DCCP can act as an enabling technology to the growth of widespread support for triple-play services.

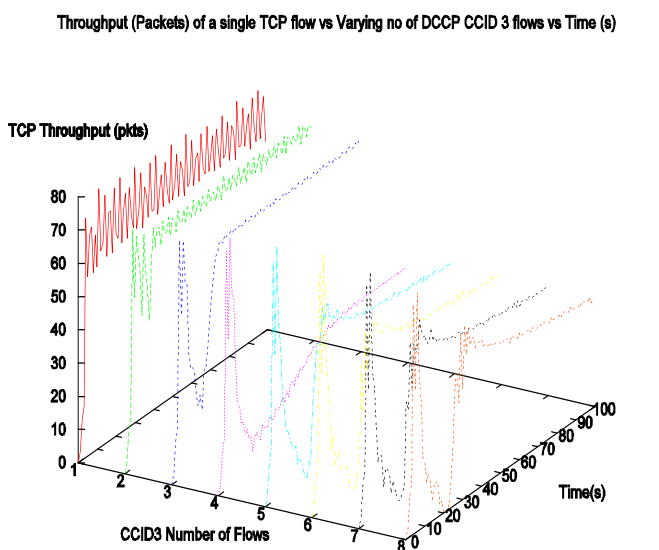


Figure 8: Throughput (packets) of a single TCP flow (vs) Varying no of CCID 3 flows (vs) Time (s). The x-axis gives the performance of the TCP flow when the CCID 3 flows increase from 1 to 8 flows.

Conclusions and Future Work

This paper provides an overview of some key work within the SatSix project to enhance delivery of broadband services for the next generation of IPv6-enabled satellite systems. It presents a brief overview of the system architecture (using DVB-RCS), focussing on the regenerative satellite system. It also describes the way in which triple play services will be offered, highlighting the use of multimedia transport in a satellite environment, and in particular describing the use of the DCCP transport protocol for multimedia.

Some initial results are presented to illustrate issues that need to be considered. These results will be used to drive our choice of scenarios for the emulation test bed. These issues continued to be studied together with other systems issues (including adaptive physical layer designs, radio resource allocation methods, new link technologies, network quality of service, security, mobility, and the architecture of the satellite itself) within the SatSix Project. The Project seeks to relate this work to the requirements of service providers. Results from both the simulation and emulation will therefore be used during the live trials involving a set of Eastern European partners.

Acknowledgments

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