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Delivery of triple-play services over broadband satellite networks

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Abstract: Digital video broadcast return channel via satellite system (RCS) is a key European standard that specifies a broadband satellite system using dynamic capacity assignment and multi-frequency time-division multiple access. This study introduces new architectural components for an RCS system to support advanced Internet protocol (IP)-oriented multimedia. This will allow closer integration of satellite networks with the next generation of multimedia-enabled networks enabling operators to offer triple-play services that simultaneously support voice over IP, television-over IP and other services. These types of services are particularly suited to next generation satellite systems that combine transparent and multi-spot beam satellites with on-board regenerative signal processing, and can evolve to use adaptive coding and modulation.

1 Introduction

The digital video broadcast (DVB) return channel via satellite system (DVB-RCS) is a key European standard published by the European Telecommunications Standards Institute (ETSI) [1, 2]. The specification leverages the highly successful DVB standards for television (TV) distribution [1] to provide a bidirectional network using geostationary satellites. This has been adopted by many companies in the satellite industry [3]. Deployment, interoperability and management aspects of DVB-RCS are addressed by the SatLabs forum [3], whereas internetworking and the systems architecture are being standardised by the broadband satellite multimedia (BSM) working group of ETSI [4]. An Internet engineering task force (IETF) working group Internet protocol over DVB (IPDVB) assisted in protocol issues relating to IP, including IP version 6.

Current DVB-RCS systems can offer broadband access to Internet services and employ traditional transparent satellites, forming a centralised star network where all communication passes via a gateway terminal (Fig. 1). A common DVB-RCS system profile [4] utilises a fixed-rate time-division multiplexed forward satellite link using DVB-S [1].

As in many broadband technologies, capacity in the return link, from an RCS terminal (RCST) to the gateway, is not dedicated, but is shared using radio resource management (RRM). The RRM subsystem is controlled by a network control centre (NCC), usually co-located with the gateway. Individual RCSTs transmit using multi-frequency time-division multiple access (MF-TDMA), in timeslots allocated by the RRM subsystem using a terminal burst timeplan (TBTP). Allocations may employ a static pre-assigned transmission rate (based on a service-level agreement (SLA)) or may be allocated in response to either explicit capacity requests (sent by RCSTs) or dynamically predicted traffic characteristics. Usually a combination of the three methods is used to seek a compromise between offered quality of service (QoS) and efficient use of the satellite capacity. In a satellite system, these tradeoffs are under direct control of the NCC. The DVB-RCS standard was amended in 2004 to allow use of the more advanced DVB-S2 [5] physical layer as an alternative forward link, and work is presently underway to define a next generation RCS system (RCS NG) with improved efficiency that will target a range of applications.

This paper extends the current RCS architecture by proposing a QoS subsystem to enable classification and

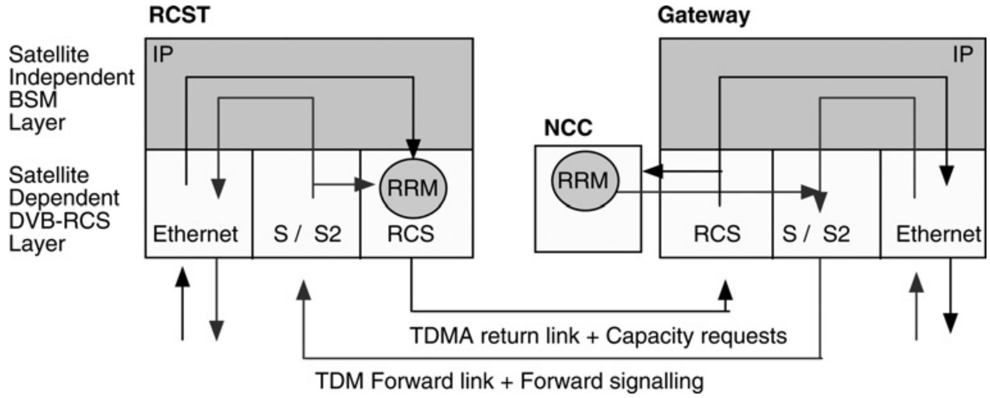

Figure 1 *DVB-RCS terminal (RCST) to gateway communication*

appropriate scheduling of multimedia traffic. It analyses the simulated performance of congestion-controlled multimedia and demonstrates that with appropriate support a range of triple-play services (including congestion-controlled IP multimedia) could be successfully transmitted with acceptable performance over a DVB-RCS system.

The remainder of the paper is organised as follows: Section 2 describes triple-play services over DVB-RCS; Section 3 describes the proposed QoS subsystem; Section 4 discusses the need for congestion-controlled multimedia; Section 5 analyses performance for congestion-controlled multimedia traffic; Section 6 gives a general discussion on the evolution of the physical layer; followed by the conclusion in Section 7.

2 Triple-play services over broadband satellite networks

There has been an exponential growth in multimedia applications within the Internet as a whole, with an increasing desire to combine voice, video and data services over a common IP network, known as 'triple-play bundling'. This is seen by many operators as a potential 'killer application'. Satellite-delivery can be an enabling technology for deploying this broadband service, particularly to small rural communities or peripheral regions, where it provides an alternative to wired connections. In supporting 'triple play', satellite systems need to be integrated with the Internet to provide appropriate performance across a range of applications (required capacity, maximum jitter, maximum delay, etc.). This presents a challenge that requires careful consideration of the RRM and QoS functions in the satellite terminal and coordination with IP-based network and transport protocols that control sharing of capacity along the end-to-end Internet path.

Data services vary widely in their expectations of capacity, and ability to handle congestion. Some are delay sensitive (e.g. synchronous E-services, interactive gaming) and some are delay tolerant (e.g. asynchronous E-services, alert messages, media download). The services using the transmission control protocol (TCP) can be particularly impacted by the variable delay/capacity resulting from RRM. Most satellite networks therefore tune TCP performance (especially for web traffic) using a performance enhancing proxy (PEP) [3], integrated in the RCST and gateway. PEPs can not only modify transport protocol behaviour, they can also improve the performance of higher layer protocols (HTTP acceleration, adaptive compression, etc.). PEPs also often provide a (cross-layer) interface to inform RRM mechanisms.

DVB-RCS systems can deliver TV using either MPEG-2 (as in traditional DVB TV distribution networks) or using IP-based TV, in which content may be streamed as either a unicast or multicast IP flow [6, 7] (e.g. video broadcast, video on demand and audiovisual conferencing). Many DVB-RCS systems also support voice over IP (VoIP). These multimedia applications typically use user datagram protocol (UDP) transport. Successful operational systems have tuned the RRM algorithms for VoIP (e.g. controlling jitter by allocating transmission opportunities equally distributed in time) [8]. The delay goal for good perceived VoIP quality may be achieved with tuning, despite the transmission delay of 300–400 ms for a transparent system when either the caller or callee communicates via the gateway.

2.1 Transparent and regenerative satellites

In a satellite system, the design of the satellite payload constrains the terminal design. The power-budget of a traditional transparent satellite payload therefore dominates the size of the RCST antenna and the peak transmit rate. This necessitates communication between a caller and callee attached to different RCSTs to pass via the gateway, which incurs a double satellite delay (i.e. at least 550 ms in each direction). Although acceptable for some applications (including VoIP where the caller and callee are familiar with the delay) it degrades the perceived quality below the threshold for acceptance for traditional telephony (Section 5.2).

In contrast, a regenerative satellite payload that employs on-board processing (OBP) [9] can demodulate the uplink signal, select an appropriate downlink and re-modulate the signal using a (potentially different) downlink waveform. This provides a more favourable power-budget allowing smaller terminals at comparable or higher data rates. The ability to switch data between several spot-beams (rather than the continental beams typically employed in transparent satellites) also allows more efficient use of the frequency/power of the satellite.

AmerHis [10] is the first operational OBP system that supports DVB-RCS satellite switching, developed as a joint project of the European Space Agency, CDTI, and Hispasat. The AmerHis uplink signal uses DVB-RCS, which the satellite regenerates and transmits as a downlink DVB-S signal. Amheris also supports on-board multicast replication towards a set of spot-beams, resulting in saving of bandwidth when a subset of RCSTs requires the same data.

Fig. 2 shows a network with connectivity to external networks provided via a larger terminal, called a regenerative satellite gateway (RSGW). The RSGW also manages the subscriber SLA functions, similar to a gateway in a transparent architecture. The configuration and operation of RCSTs and RSGWs is controlled by a network control centre/network management centre (NCC/NMC). This provides session control, routing and radio resource allocation to the subscriber RCSTs, and manages the OBP configuration by generating the required satellite control tables. This is normally replicated for redundancy.

RCSTs using AmerHis can support an enhancement to the DVB-RCS standard that adds a signalling protocol, known as the connection-control protocol (C2P) [9]. This allows an RCST to request the NCC to allocate capacity for bidirectional communication with another RCST. The protocol specifies the destination identifiers for traffic flows and their associated QoS, and defines mechanisms for acceptance, establishment, modification and release of connections. C2P allows the regenerative payload to provide a mesh connection. Once radio resources have been allocated, data may be directly sent between RCSTs using a single satellite hop, effectively halving the delay.

2.2 *Transparent and regenerative services*

To optimise operational cost/performance, a system architecture may combine a satellite (or satellites) that support both transparent and regenerative payloads. A service can be provided using either payload. The regenerative payload offers an advantage in delay or the support for multicast spot-beam switching. In other cases, the transparent payload can offer acceptable performance at lower overall cost. Each service is therefore associated with a preferred payload (Table 1) with actual allocations according to the requirements and available resources.

3 QoS architecture for DVB-RCS multimedia services

To meet the QoS requirement for an IP flow, an RCST must implement buffer and resource management functions that control queuing delays and assure fair sharing of the capacity allocated by the RRM among the arriving IP flows. The RRM subsystem of DVB-RCS provides a variety of capacity request mechanisms [2, 3] that can, in principle, be used to support such QoS. However, there is no currently defined architecture for the QoS subsystem that specifies how to manage queuing or scheduling, or how this should be integrated to support IP IntServ or Diffserv at the network layer. To meet this need, the paper derives a QoS architecture for DVB-RCS based on the initial work of the ETSI BSM working group [11].

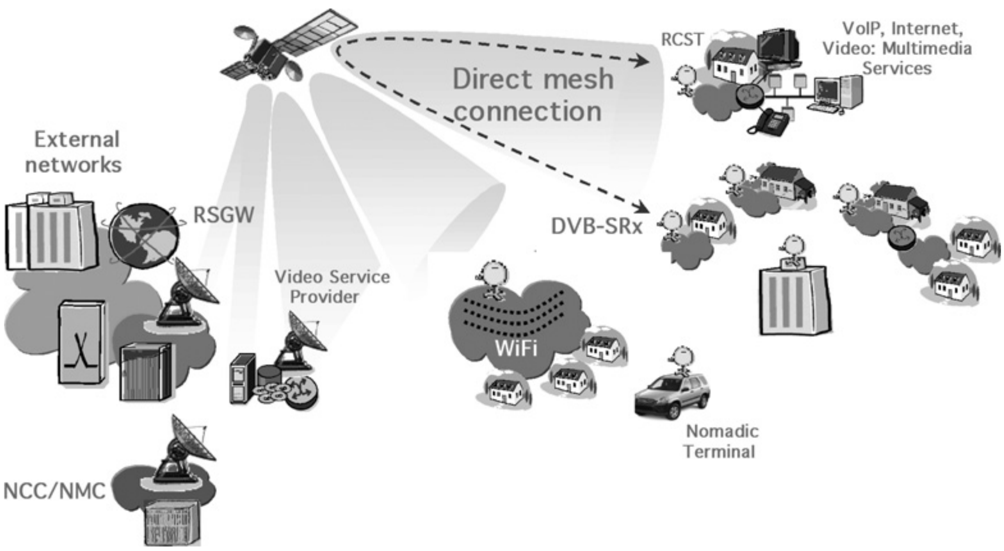

Figure 2 *DVB-RCS network using the AmerHis regenerative satellite system*

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Table 1 Service class by satellite payload architecture

Service	Transparent	Regenerative
internet access	X	
intranet virtual private network		X
IP multicast data		X
LAN interconnection		X
audio/video (AV) conference		X
asynchronous E-services	X	
synchronous E-services		X
video on-demand (IPTV)	X	
video broadcast (IPTV)	X (one/all destinations)	X (several destinations)
software download	X	
alert messages	X	
media content download	X	
interactive gaming		X
bulk peer to peer (file download/upload)	X	
interactive peer to peer (video, audio)		X

3.1 Classification of traffic

The architecture supports three models that associate an IP traffic flow with an appropriate set of QoS attributes:

Best-effort (BE) model: The BE model uses default QoS attributes. The treatment of the traffic depends on the policy implemented at the medium access control (MAC) QoS module and the NCC. This is the default for data services.

IS: The integrated services (IS) model is call-based, as in the IETF IntServ framework (RFC 1663). A QoS server associates explicit QoS attributes with an individual traffic flow [12]. This can use a proxy in the RCST that intercepts or participates in a specific signalling protocol. Since this occurs in the control plane, it is transparent to user applications. The QoS server determines the committed and available resources, and new sessions are only admitted when both resources are available, and permitted by the customer SLA. Admission may be associated with a billing model for resource usage. Examples are

- A proxy module intercepts session information protocol (SIP) messages between the caller and callee to extract media attributes that are translated to QoS attributes in an XML record and passed to the QoS server.

- A proxy for other resource reservation protocols (RSVPs), for example, RSVP may be directly interfaced to the QoS server.

- A PEP module may anticipate TCP QoS requirements for data sessions and assign QoS attributes for the QoS server.

- Where QoS information is not available at the IP layer or cannot be derived from IP signalling, a user-oriented QoS agent may explicitly communicate with the QoS server to assign QoS attributes. A bandwidth broker in the access network (e.g. at a gateway or NCC) could be directly interfaced using this method.

DS: The differentiated services (DS) model matches each IP flow with a flow descriptor (IP addresses, ports, differentiated service code point) and assigns it to one of the set of traffic aggregation classes [13]. Each class is associated with pre-assigned QoS attributes, following the IETF DiffServ framework (RFC 2475). The model may be used for any service that can share network capacity. It may meter traffic against a policy [13], where non-compliant traffic may be shaped or discarded. The framework provides one means to configure appropriate policies. The DS policy may be set locally in a system or coordinated between different networks.

3.2 Resource request by the RCST

The QoS subsystem queues the traffic and associates this with a capacity request in the RRM subsystem. IP queues are managed using class-based queuing (CBQ). Each queue is mapped to one of the small number of per-hop behaviours (PHBs) [3, 13] corresponding to each link-layer destination (i.e. return link to the gateway or a mesh connection to another RCST). In this proposal, we introduce three user traffic PHBs:

- best effort (BE);

- expedited forwarding (EF);

- assured forwarding (AF).

Each PHB is mapped to an RCST link service that determines the policy applied to a MAC queue at the transmission interface. It also determines the combination of RRM methods to be used to request capacity [2, 3, 13]. The BE PHB employs a mixture of rate-based dynamic capacity (RBDC) and volume-based dynamic capacity requests. The EF PHB relies on constant rate allocation

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(CRA) (e.g. reserving using the C2P [9] or by intercepting network signalling using a proxy), whereas the AF PHB uses a mixture of CRA and RBDC. RRM design is particularly challenging when using an adaptive physical layer, such as DVB-S2, where the total capacity of the system varies as a function of the required QoS, and propagation conditions for the destination terminals. The variable traffic resulting from variable bit rate (VBR) services needs to be either statically provisioned or require the RRM system to dynamically predict a suitable allocation rate. Over allocation can guarantee QoS requirements, but may reduce system utilisation (unless other traffic arrives that can take advantage of the free capacity). This resembles contention for wired access links, but in the case of DVB-RCS this is not dictated by physical constraints and becomes a policy decision under the control of the NCC and provisioned SLA.

Fig. 3 shows the QoS architecture (in this case for mesh communication), identifying the following components:

• IP QoS: This module classifies traffic flows, queues IP traffic for transmission using CBQ and performs any required marking/shaping/dropping to enforce the QoS.

• MAC QoS: This module schedules queued traffic into allocated transmission timeslots.

• RRC agent: This RCST module is responsible for generating capacity requests and controlling transmission using timeslots allocated in the TBTP.

• RRC server: This NCC module collects capacity requests from RCSTs and combines these with local information to construct a TBTP for all RCSTs.

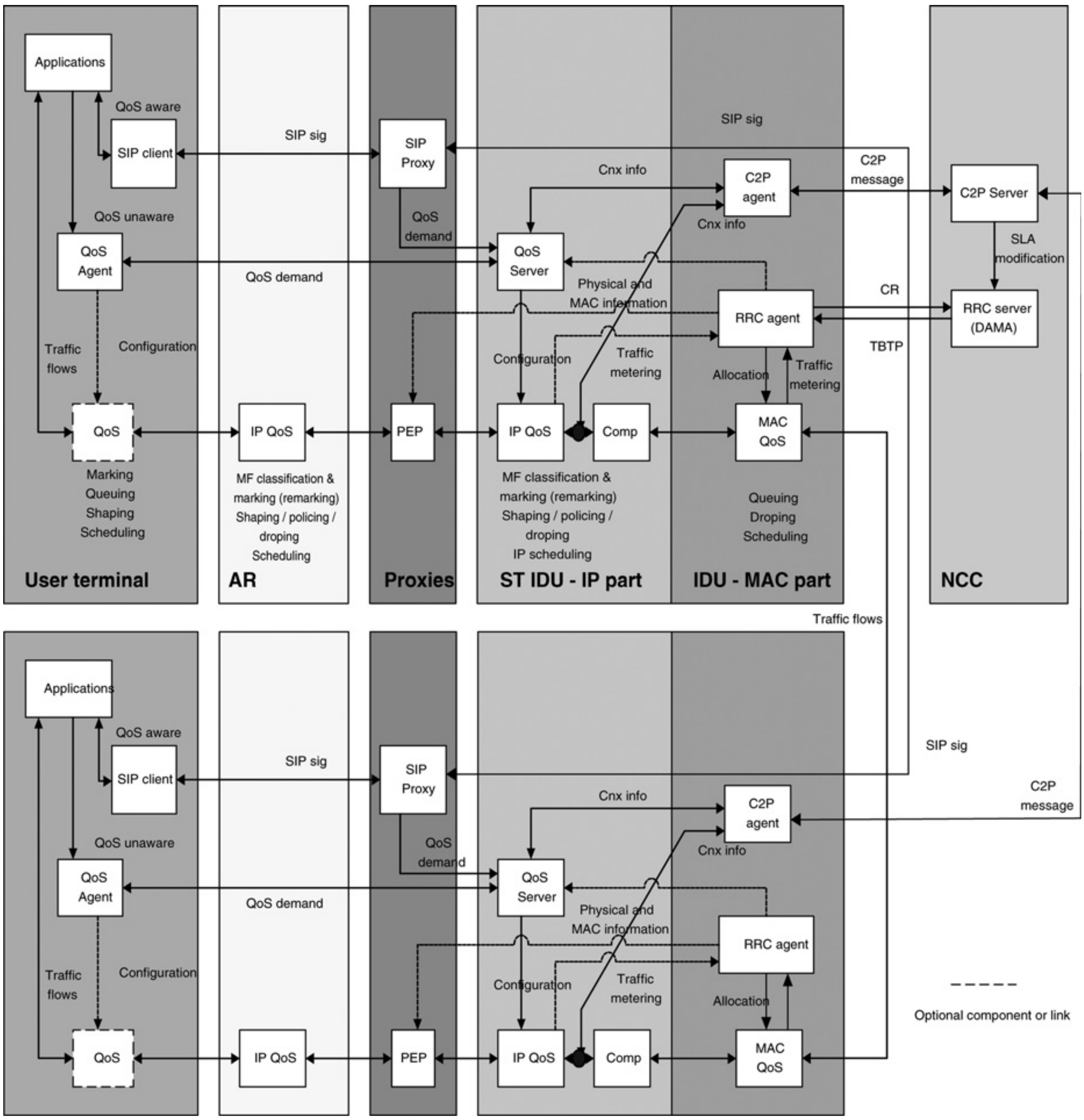

Figure 3 *Mesh QoS functional architecture*

- C2P agent: This RCST agent establishes, modifies and releases a satellite RRM connection at the C2P server. Once established, traffic is appropriately classified and associated with the connection, whereas the RRC server allocates the corresponding capacity using CRA.

- C2P server: The server provides satellite QoS information to the RRC server. It can also utilise signalling to configure IP QoS at the gateway.

- QoS server: Each RCST implements a QoS server that maps new traffic flows to QoS attributes in the IP QoS module. Traffic flows may be admission controlled and/or associated with a policy, as required to support an SLA. Each class is statically or dynamically mapped to a particular QoS treatment and a PHB and capacity request method(s).

- SIP proxy: A proxy supports QoS-aware applications by intercepting SIP signalling informing the QoS server of new SIP sessions and key SIP parameters (traffic mask, codec, etc.).

- QoS agent: A user terminal may use a QoS agent to inform the QoS server of the required QoS treatment for a specific traffic flow.

- PEP: This module can perform satellite-specific TCP and HTTP acceleration to mitigate the delay for specific applications. A PEP modifies the transmitted traffic pattern and may interact with the RRC agent to associate a flow with a particular QoS treatment.

- Compression: This module may compress protocol header information to decrease transmission overhead, and hence modifies the transmitted traffic pattern.

This QoS architecture is independent of the type of satellite payload. A mesh satellite system requires C2P [9] to communicate QoS attributes between terminals. To avoid delay in the transparent case, connections between the RCST and the gateway are established when the terminal logs onto the network. An RCST-initiated unicast connection would require several RTTs to establish a connection to a peer RCST via the NCC. Once established, QoS parameters may be dynamically modified by the QoS server using C2P (e.g. removing the need for a SIP proxy at the gateway).

3.3 RCST transmission scheduling

Traffic awaiting transmission (i.e. allocation of radio resource) is queued by the MAC QoS module. Our implementation uses a scheduler implementing a derivation of CBQ with a token-bucket algorithm called hierarchical dual leaky bucket [14]. This method introduces a scheduler inside each hierarchical class to improve performance for bursty traffic.

The QoS server updates the corresponding queuing parameters in the MAC QoS module to reflect changes in IP QoS and network capacity. The efficient scheduling implies close coordination between these two modules. Lack of coordination has been shown to lead to decisions that are neither relevant nor optimal [8]. Several options have been investigated, including feedback information aggregating physical information such as queue sizes, capacity requests and allocations. The transmission rate is determined by the air interface waveform required to reach each specific destination RCST (mesh or star), and can vary with time. It is therefore difficult to optimise the control loop, especially with variable rate traffic, and has drawbacks including the difficulty of avoiding starvation at the MAC air interface. A simpler solution could rely on back-pressure feedback of the air interface queue size.

4 Multimedia transport

Internet multimedia can be transported over IP networks using a range of transport protocols. Many current applications transmit at a constant rate using UDP. Such applications have fixed QoS requirements for delay, jitter and guaranteed throughput. In this case, the IS model allows a network operator to provide service guarantees to a session when the operator controls the transmission path. For example, this approach is suitable for the current use of multicast, and hence is favoured in many current satellite IPTV deployments where content servers can be directly provisioned with appropriate network resource.

Fixed QoS assignments are however difficult to manage when applications do not utilise QoS signalling, or for traffic with unknown or highly variable characteristics (e.g. the use of VPN or aggregated traffic). Such applications can be supported using DS. In this model, QoS is provisioned to a service as a whole (rather than individual sessions) and applications need to share the network capacity, particularly sessions that operate over network paths with dynamically changing capacity (including satellite broadband services). The applications that do not require any QoS guarantees, such as bulk traffic file transfers, can be supported using the BE service.

4.1 Congestion-controlled multimedia

Traffic flows using the DS and BE models need to receive a fair distribution of available capacity. This is often achieved using a protocol method known as congestion control, a standard feature of TCP. Congestion control is also vital for multimedia applications to ensure the overall health of the Internet [15], although it is not generally provided by applications that use UDP, or is provided in some non-standard way at the application layer.

Work in the IETF has resulted in the datagram congestion-control protocol (DCCP) [15]. This new transport protocol is designed to support unicast

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multimedia flows in next generation networks. DCCP utilises transport-layer feedback to determine an appropriate transmit rate.

DCCP provides a standard way to provide congestion control using one of the three congestion-control profiles. In this paper, we consider only congestion-control identifier 3 (CCID-3) [16] and CCID-4 [17]. The former is based on TCP friendly rate control (TFRC) [18], and is appropriate for flows that prefer to minimise abrupt changes in sending rate, including conversational streaming, whereas the latter is based on an experimental small packet variant of TFRC that is more suited to voice and narrowband video.

5 Performance analysis of the DVB-RCS QoS architecture

To investigate the impact of RRM on congestion-controlled multimedia, a DVB-RCS satellite system implementing appropriate RRM methods was simulated using the ns-2 network simulator [19]. Previous work has shown that the adaptive link technology employed by next generation RCS systems can result in abrupt changes in delay and/or capacity, with implications on the performance of VoIP over DCCP [20]. The performance for various network traffic patterns will therefore be compared using the simulator.

5.1 Simulation of DVB-RCS

The characterisation of a complete RRM system requires the definition of many parameters. These include the duration of an allocation frame, the set of allowed modulation and coding schemes, the choice of framing and number of timeslots per frame, the encapsulation and fragmentation method, the number of cells per timeslot, predictive and free-capacity assignment strategies, and so on [21]. Since the joint optimisation of all these parameters is generally impractical, some parameters have to be fixed for a particular simulation. DVB-RCS specifies RCST transmission using MF-TDMA with either asynchronous transfer mode (ATM) and/or MPEG encapsulation. These simulations considered an ATM-based DVB-RCS satellite network topology, as this is the most commonly used mode. The topology connected the satellite network with the Internet via a small delay of 29 ms (representative of intra-Europe delay). QoS provisioning in RCSTs allowed appropriate classification of traffic by mapping them into supported PHB. Table 2 presents the simulation parameters.

5.2 Simulation of VoIP traffic over DCCP

This section analyses the performance of VoIP over DCCP using DVB-RCS for both regenerative and transparent payload scenarios. The quality was assessed using the E-model [22] to determine a quality factor 'R', using sub-factors that consider the impact of loss and delay for each voice call. For VoIP traffic, the R-score is calculated based on sub-factors such as impact of loss (I_e) and impact of delay (I_d) using the formula

Table 2 Simulation parameters

Configuration	Value
frame (duration)	53 ms
superframe (frames/superframe)	1
ATM packets per timeslot	6
granularity	48 Kbps
uplink capacity	12.48 Mbps
downlink capacity	33.18 Mbps

$$R = 94.2 - I_e - I_d \tag{1}$$

where

$$I_e = \lambda_1 + \lambda_2 \ln(1 + \lambda_3 e) \tag{2}$$

and

$$I_d = 0.024\,d + 0.11(d - 177.3)I(d - 177.3) \tag{3}$$

$I(x)$ is a unity function. The factor 'R' was translated to determine a mean opinion score (MOS) using the formula

$$\begin{aligned} \text{MOS} = 1 + 0.035R + 7 \times 10^{-6} R \times (R - 60) \\ \times (100 - R) \end{aligned} \tag{4}$$

A call set-up feature was simulated (corresponding to proxying the SIP exchange prior to a VoIP session). Following call set-up, a sender started transmitting media packets only when the sending rate reached the desired encoding rate for the codec (perceived by the user as a small delay in addition to the calling time). Two classes of VoIP traffic were considered:

- *Class I:* constant bit rate VoIP simulated by a model for a G.711 voice codec (media rate 64 kbps, 50 pps).

- *Class II:* conversational VoIP with silence suppression used a G.711 codec with an ON/OFF model to represent variable periods of fixed rate voice separated by periods of silence (mean burst idle periods of 0.352 and 0.65 s). This model is consistent with the third-generation partnership program assessment of VoIP codecs [23], and thereby results in VBR traffic with a peak rate identical to the Class I traffic. The results are shown for both UDP and DCCP CCID-4 transport of the VoIP flow. Class I and Class II traffic over UDP may be expected to offer equivalent performance (only Class II is shown in the following results).

The transport-layer transmit buffer was five packets (i.e. no more than five media samples were queued at the sender).

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The varying levels of TCP web BE traffic shared the capacity (each web client randomly started 10 web sessions, with each web session requesting 10 web pages, each page having 10 objects. The size of the objects followed a Pareto II distribution (mean = 10, shape = 1.05)). Three service models were simulated:

- BE (one service class).
- IS-CBQ (QoS assigned by the control plane using C2P with CBQ).
- DS-CBQ (QoS provisioned by the operator with CBQ).

Table 3 presents the MOS values for the VoIP call for RCST-to-RCST communication using a traditional transparent payload and using the user model described in (1)–(4). Acceptable VoIP quality was observed with a transparent payload providing that one of the communicating peers was behind the gateway. A VoIP traffic from the gateway to an RCST uses the forward link. This resulted in a MOS of 3.9, irrespective of other traffic. A VoIP call to the gateway from an RCST was impacted by the delay from the RRM and resulted in a similar MOS when using either the IS or DS model. However, for the case of the caller and callee attached to different RCSTs, the additional delay from retransmission via the gateway using a transparent system resulted in low quality.

Table 4 shows the MOS achieved by a VoIP call, when sharing with a varying number of web clients using a regenerative satellite payload. A medium quality was achieved by the VoIP flow. BE resource allocation did not differentiate the traffic. The quality of a voice call therefore decreased, as the level of web traffic increased, leading to competition between traffic flows using the allocated resources, and resulting in additional queuing delay with impairment of voice quality from medium to poor quality (irrespective of the transport).

Using the IS model, we assume that an RCST associates QoS attributes with a flow (e.g. proxying SIP messages), allowing it to request capacity based on future expected need. This allows an RCST to reserve the appropriate number of timeslots for the packets of a VoIP call without incurring the variable delay in signalling the request to the RRM subsystem. With constant rate VoIP traffic (Class I), this seems to offer an acceptable performance, but led to an unused capacity when used with bursty VoIP traffic (Class II traffic), since this did not use allocated timeslots during silence suppression. When DCCP was used with Class II traffic, the resulting quality was similar to that for Class I traffic and Class II traffic with UDP.

Using the DS model specific capacity is not reserved for a flow, but real-time packets may be prioritised (e.g. transmitting these ahead of queued web traffic). In contrast to the BE model, an increase in the volume of web traffic resulted in higher allocation of slots to the MAC QoS module providing more transmission opportunities for VoIP traffic, which consequentially improved the VoIP quality.

These simulation results show that acceptable quality can be achieved using DCCP for VoIP flows over networks with the expected delay offered by a DVB-RCS network. The results also show that with appropriate QoS provisioning, DCCP could be used to efficiently carry next generation VoIP traffic with silence suppression achieving a quality similar to that of UDP. The use of DCCP also provides an incentive to VoIP application designers since it provides a way to adapt the VoIP traffic to the presence of network congestion, ensuring applications fairly share the network capacity with other Internet traffic.

Table 3 Transparent payload: MOS for VoIP sharing a link with a set of web clients

Number of web clients	BE			IS-CBQ			DS-CBQ		
	Class I with DCCP	Class II with UDP	Class II with DCCP	Class I with DCCP	Class II With UDP	Class II with DCCP	Class I with DCCP	Class II with UDP	Class II with DCCP
0	1.9	2.1	1.9	2.1	2.1	1.9	1.9	2.1	1.9
5	1.9	2.1	1.8	2.1	2.1	1.9	1.9	2.1	1.9
10	1.9	2.1	1.8	2.1	2.1	2.0	2.0	2.1	2.0
20	1.7	1.7	1.4	2.1	2.1	2.1	2.1	2.1	2.0
30	1.0	1.5	1.0	2.1	2.1	2.1	2.1	2.1	2.1
40	1.0	1.1	1.0	2.1	2.1	2.1	2.1	2.1	2.1
50	1.0	1.0	1.0	2.1	2.1	2.1	2.1	2.1	2.1

Table 4 Regenerative payload: MOS for VoIP sharing a link with a set of web clients

Number of web clients	BE			IS-CBQ			DS-CBQ		
	Class I with DCCP	Class II with UDP	Class II with DCCP	Class I with DCCP	Class II With UDP	Class II with DCCP	Class I with DCCP	Class II with UDP	Class II with DCCP
0	3.7	3.9	3.6	3.9	3.9	3.8	3.7	3.9	3.7
5	3.7	3.9	3.6	3.9	3.9	3.8	3.8	3.9	3.8
10	3.8	3.8	3.7	3.9	3.9	3.8	3.8	3.9	3.8
20	3.2	3.7	3.3	3.9	3.9	3.9	3.9	3.9	3.8
30	1.5	3.4	1.9	3.9	3.9	3.9	3.9	3.9	3.9
40	1.1	3.2	1.3	3.9	3.9	3.9	3.9	3.9	3.9
50	1.0	1.3	1.0	3.9	3.9	3.9	3.9	3.9	3.9

5.3 Simulation of video transmission

This section simulates conversational and streaming video using DCCP with the network presented in the previous section. Conversational video requires a small play-out time at the receiver to satisfy its real-time constraints. In contrast, streaming video (e.g. standard definition TV) is able to tolerate a larger play-out time, since this does not have real-time constraints.

Although there are readily available models for voice traffic and methods for evaluation, it is more difficult to model video traffic and assess performance in a way that is appropriate to a range of different content. A trace-driven approach was therefore adopted. A set of video clips (and associated packet traces) were encoded and stored in a raw YUV format for input to the simulations. These could be used to perform reproducible experiments over a range of RRM scenarios. The following steps detail the methodology used:

1. Raw YUV video was compressed to MPEG-4 format, and streaming hints inserted.

2. Tools from the Evalvid toolkit [24] were used to create an input trace for a simulation.

3. The DVB-RCS system was simulated, creating two-packet capture files (a sender packet dump and a receiver packet dump).

4. From these dump files and the input trace, the video was reconstructed.

5. The reconstructed video could be assessed subjectively or objectively.

The video content encoded in step 1 could also be streamed over an actual network for comparison with the simulated video. CCID-3 was used for DCCP, since this was appropriate to both conversational and streamed video [25].

5.3.1 Quality estimation: When analysing multimedia it is important to understand the impact of network performance on the actual user-perceived quality, such details may be easily overlooked when using simulation, but failure to consider the actual effects of loss/jitter can lead to inappropriate optimisations. The subjective quality of MPEG4-encoded video (PAL format with 25 fps, media rate 1 Mbps) was therefore assessed by combining the models developed for DCCP and UDP in ns-2 with the viewer tool in Evalvid. Two receiver scenarios were considered:

1. Video conferencing using a play-out buffer of 400 ms, typical of conference-style receivers.

2. Streaming video with a large receiver play-out buffer of 20 s (also suitable for IP multicast when using UDP).

The subjective video quality estimation for the conferencing service using UDP observed pictures of good quality with no noticeable artefacts resulted from the RRM interaction with timing. The transmission of the same video using CCID-3 (Fig. 4) resulted in noticeable distortion during the initial set of frames (Fig. 4 shows some of the distorted initial frames). This was a side-effect of the slow-start of the congestion-control algorithm, which delayed transmission of initial packets, and resulted in high levels of packet loss for the first few (e.g. 3–5) seconds, after which quality became again comparable to that of UDP (in the absence of congestion). There are several possible mitigations to this effect: adaptive video codecs may vary the encoding rate based on the sending rate indicated by the underlying congestion-control protocol; stream-thinning could be used to remove specific frames when the media rate exceeded that was allowed by

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Figure 4 *Subjective quality of conversational video during DCCP slow start over DVB-RCS*

the transport protocol; and advanced coding/concealment techniques can hide the effects of loss. These methods on their own or in combination could allow acceptable delivery of video during periods of congestion.

When the receiver play-out buffer was increased to 20 s, a value more suited to streaming video than interactive use [26], no frames were discarded due to late arrival, resulting in excellent quality.

5.4 Triple-play services

This section analyses the overall performance of triple-play services (simultaneous video, audio and data) over a simulated DVB-RCS regenerative system. The quality of the same MPEG4-encoded video using DCCP was assessed by measuring the average peak signal-to-noise ratio (PSNR) of each video frame at the receiver. VoIP calls from Class II traffic using DCCP were assessed using the E-model, resulting in an average MOS for each call. The varying levels of HTTP data traffic shared the satellite capacity (each web client randomly started 10 sessions from the RCST, with each session requesting 10 web pages of 10 objects. The size of the objects followed a Pareto II distribution (mean = 10, shape = 1.05)) and the average latency for these HTTP flows was determined.

Table 5 summarises the overall performance. The IS and DS models allowed voice traffic to improve with an increase in the terminal aggregate traffic load. This corroborates with the previous results for VoIP traffic, where an increased level of aggregated traffic presented more transmission opportunities, resulting in better performance. While IP broadcast video and video on demand (IPTV) could tolerate additional delay (accommodating this in the appreciable multimedia play-out buffer of 20 s [26]), appropriate QoS scheduling was required to support video conferencing in a heavily loaded network. The quality of the conference call was observed to decrease with increasing load, due to increased jitter resulting in arrival after the play-out deadline (set at 400 ms). The results improved for both IS and DS, with similar performances for each model. This means that DS could be used instead of IS, since this does not require

Table 5 Performance of video, audio and data over a regenerative satellite link

Type of flow	BE	DS	IS
1 IPTV flow	PSNR = 58	PSNR = 58	PSNR = 58
1 IPTV flow	PSNR = 58	PSNR = 58	PSNR = 58
5 VoIP flows	average MOS = 2.7	average *R*-score = 3.7	average *R*-score = 3.7
10 web sessions	average latency = 2 s	average latency = 3 s	average latency = 3 s
1 IPTV flow	PSNR = 58	PSNR = 58	PSNR = 58
10 VoIP flows	average MOS = 2.6	average *R*-score = 3.8	average *R*-score = 3.8
20 web sessions	average latency = 5 s	average latency = 5 s	average latency = 5 s
1 AV flow	PSNR = 58	PSNR = 58	PSNR = 58
1 AV flow	PSNR = 55	PSNR = 55	PSNR = 55
5 VoIP flows	average MOS = 2.7	average *R*-score = 3.7	average *R*-score = 3.7
10 web sessions	average latency = 2 s	average latency = 3 s	average latency = 3 s
1 AV flow	PSNR = 43	PSNR = 47	PSNR = 57
10 VoIP flows	average *R*-score = 2.6	average *R*-score = 3.8	average *R*-score = 3.8
20 web sessions	average latency = 5 s	average latency = 5 s	average latency = 5 s

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signalling with prior-knowledge of QoS attributes and facilitates better link sharing without requiring RCSTs to implement sophisticated call admission control functions. The performance of web traffic was acceptable in all cases.

Table 5 shows results for the three QoS models. The QoS mechanisms at the link or network layers benefited when traffic flows may be differentiated. The results demonstrate that congestion-controlled triple-play services can be successfully transmitted using a DS QoS model with acceptable performance over a DVB-RCS satellite deploying a regenerative payload.

6 Evolution of DVB-RCS

The DVB-RCS physical layer continues to evolve, and a new version of the specification is expected in 2010. Future systems will be benefitted from the flexible use of DVB-S2 [5] on the forward link, which provides a family of physical layer waveforms that increase the spectral efficiency from 1 bit/s/Hz using quadrature phase shift keying (QPSK) with rate 1/2 forward error correction (FEC) to 1.98 bit/s/Hz for eight phase shift keying (PSK) with 2/3 FEC; and 2.97 bit/s/Hz for 16 amplitude and PSK with 3/4 FEC.

The regenerative approach offers improved efficiency for the system and enhanced user performance. The principle drawbacks are an increase in the cost of the satellite payload and the need to freeze parts of the on-board configuration at the time of launch (e.g. the payload of AmerHis [10] predates standardisation of DVB-S2, although there are plans for future spacecraft that can combine DVB-RCS and DVB-S2 OBP).

The current DVB-RCS standard specifies a fixed FEC code with QPSK modulation for the return link. Research is also exploring an extension to allow RCSTs to support other modulation (e.g. 8-PSK) and coding schemes that are expected to form a part of the new RCS-NG standard. An adaptive transmission system is required in both the forward and return link to fully take advantage of the operational cost savings offered by the new physical layer waveforms. The resulting increase in spectral efficiency reduces operational cost, crucial to successful competition with other satellite and terrestrial Internet services. However, this also results in a dynamically varying transmission rate (as the waveform tracks the prevailing conditions). Future work therefore needs to explore the impact of adaptive coding and modulation and dynamic rate adaptation on the performance of congestion-controlled multimedia in the presence of rain fades, although the authors note that rain fade events may often vary on much longer time scales than transport protocols, and therefore the congestion-control function of DCCP may be able to adjust the transmission rate of applications under such conditions. This analysis is beyond the scope of the current paper.

It is expected that the RCS-NG standard will also define RCST higher layer functions, such as IP QoS. A DS model may be suitable for consumer networks, but the ability to support IS may also be attractive for professional networks. The architecture proposed in this paper could be used as the basis for adding QoS support.

7 Conclusion

The satellite systems designed according to the DVB-RCS specification can provide broadband services. Advances in system design and a hybrid transparent/regenerative satellite payload will continue to reduce the operational cost of DVB-RCS systems, enabling operators to offer triple-play services. The support for multiple services demands new IP-orientated architectural components to be introduced providing QoS for VoIP, television over IP (TVoIP) and other services. This paper presents a candidate architecture that supports the three QoS models. The performance for each case was analysed through simulation for both traditional constant rate multimedia and congestion-controlled next generation multimedia transport protocols. Congestion-controlled techniques when used with a differentiated service model were seen to offer acceptable performance across a range of scenarios without the need for explicit reservations of call admission control functions.

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