UNIVERSITY OF ROME TOR VERGATA

Electronics Engineering Department

Ph.D. Thesis in “Space systems and Technologies”

Satellite and terrestrial network integration

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Contents

Chapter 1

Summary

This thesis work is the outcome of the research activities carried out during the execution of the Ph.D. in "Space systems and Technologies" at university of Rome Tor Vergata, under the supervision of Prof. Luglio and the coordination of Prof. Cardarilli. The main topic of satellite and terrestrial networks integration has been narrowed down during the studies and projects executed, resulting in a reference architecture that will be presented in the following sections, suitable for a wide range of applications and use-cases. Some choices are driven by requirements of projects run during the Ph.D., other are technology constrained, other are the results of research and proposals for real system enhancements. In the latter case, tests were performed via simulation or emulation to assess the benefit of the newer introduced modifications.

Some of the results obtained were presented in international conferences and journals, and can be grouped in papers related to architectures, application layer, transport protocol enhancements, network layer, MAC layer and physical layer. The complete list of publications derived from the Ph.D. activities are reported in the Publications section.

The rest of the thesis is organized as follows: in chapter 2 the main objectives of this work are specified, describing the goals of the integrated satellite/terrestrial network; in chapter 3 the motivations of the integrated approach are presented, with an introduction of typical services and applications of interest; section 4 contains a technology state of the art description; chapter 5 deals with the research approach followed to validate the architectural choices, followed in chapter 7 by simulation activities results, in section 8 by emulation executions results and in section 9 by tests run on network including real satellite platforms; finally chapter 10 draws conclusions related to the thesis' topics.

Chapter 2

Objectives

The thesis' objective is to outline the most suitable architectures and select the most proper technologies for an hybrid network composed of satellite segments and terrestrial segments. The design is targeted to the requirements of actual projects, typical applications and already deployed systems, aided by simulations activities, emulations and test campaigns.

The main goal of the resulting hybrid network is to be flexible enough to be re-used in different contexts and scenarios, using each time the optimal satellite technology available and exploit the resoruces available in the best possible way. For this reason a set of additional requirements must be satisfied by the hybrid network, including prioritization of flows (in order to let time-critical applications to exploit the minimum delay and jitter possible), confidentiality and integrity of the communications, per-flow call setup and dynamic quality of service, optimal dynamic bandwidth allocation and full exploitation of the channel capacity, handover tolerance of higher layer protocols, mobility, etc. Each of these briefly introduced aspects will make the satellite/terrestrial hybrid network the best solution, not to compete directly with full-terrestrial systems, but rather to offer a solid solution where the need of a satellite segment is mandatory (emergency, distance among terminals, rural areas services, etc.).

Chapter 3

Integration rationale

Through the utilization of an efficient and leading edge telecommunication infrastructure, information can be shared among a large number of users, allowing fast data distribution, interactivity, large scalability, mobility and coverage. Satellite and terrestrial systems can cooperate fruitfully to reach these goals.

Satellites can provide broadband access ubiquitously over very large areas, including remote or impervious locations where typically terrestrial telecommunication infrastructures are not present or inconvenient, and in case of service outages (natural disasters). In addition, satellite systems can offer a centralized management, ensure long range mobility and are particularly suitable to provide service during emergency when other telecommunications infrastructures can be damaged, destroyed or simply unavailable.

Terrestrial networks are suitable for high speed wired communications, using cheap devices and fixed points of access, and for medium-to-high speed wireless connectivity, resulting in a more flexible infrastructure. In both cases terrestrial networks represent a consolidated access technology, usually available in mostly all network devices (phones, PDAs, laptops, PCs, etc.) and supporting broadband applications.

The target of the thesis is the study of the integration of satellite and terrestrial wireless networks, designed to leverage on strengths of both. For instance it is possible to integrate the satellite core system with other wireless sub-systems to complement the satellite infrastructure, realizing additional terrestrial tails to ensure capillarity and improve efficiency and flexibility of the services offered. Table 3.1 makes a parallel concerning the main characteristics of the two networks, with a column dedicated to the advantage resulting in their integration.

Of course this approach introduces some drawbacks, such as multiple standard adaptations, complexity in the management or possible intra-segment incompatibilities which will be addressed in chapter 6.

Network characteristics		Integration	
Terrestrial	Satellite	Terrestrial networks can help satellites	Satellites can help Terrestrial networks
High availability in densely populated regions	Wide coverage (irrespectively of population density)	To use terrestrial COTS terminals and widespread standards	To act as backbone
Broadband Fixed connectivity, also symmetric	Asymmetric Fixed/mobile Broadband connectivity	To aggregate different sources of traffic to a satellite node	Services to high speed mobile users (train)
Fixed wired infrastructure, slow expansions	Promptness to set up (once satellite in orbit)	Performing dedicated connectivity among different satellite platforms	To interconnect several terrestrial subnets and To extend coverage
Hot-spot coverage (wireless) short to medium range	Expansive satellite bandwidth	To enhance coverage in urban areas (and indoor)	To setup connectivity in emergency
Cheap devices	Medium-cost devices	Reduce costs (aggregation)	
Not scaling well for broadcast/multicast	Suitability for multicast and broadcast	To alleviate satellite traffic if local networks are available (handover)	Deliver broadcast contents

Table 3.1: Network comparison

3.1 Services classifications

The services considered as reference for the development and dimensioning of the integrated hybrid network are representing typical services encountered and studied during the management of telecommunication projects.

3.1.1 Interactive services

One of the most widespread use of a telecommunication network consists in the transfer of data in a interactive way, in which the users request or publish some specific content with an unpredictable pattern. In other words the amount of data, the time in which it is requested or sent, the duration of a session and the endpoints are not easily represented by simple random variables. Examples of application of such services are web browsing (HTTP protocol), chat, data transfer on demand. This kind of service is usually the critical one for the dimensioning of the network, especially when peer to peer applications are involved.

3.1.2 Datacast Services

This kind of services concern the delivery of broadband contents (video, data or audio) in real time or delayed.

The use of satellite connectivity can be optimal when a large amount of data must be delivered to a large community of users. In this case the concept of broadcast and multicast are inherited directly from the the satellite characteristics (wide coverage and broadcasted signal). In most of the cases connection-less protocols are adopted (i.e., UDP [1]).

3.1.3 Emergency services

In an emergency scenario, where the existing telecommunication infrastructure is damaged or missing, emergency services are a broad class of services required for the emergency handling. Emergency handling consists of several phases: the prevention and preparation to the emergency, the detection and monitoring of the emergency, the emergency managment and remediation and finally the restore of normal conditions.

A satellite based network for emergency plays a fundamental role in the monitoring, emergency management and restoring phases, in which a strict coordination and a dedicated network, quickly deployable in replacement of damaged exsisting infrastructures, is needed. Re-locable Broadband Satellite Terminals can be deployed quickly to guarantee backbone connection to the Internet and to service centers, and can be interconnected to other wireless terrestrial technologies, to offer a more capillary service, in better conditions of mobility (smaller terminals) and even using commonly available terminals (cost effectiveness). In fact, examples of GSM pico-cells, WiFi hot-spot and several other wireless technologies, coupled to the satellite link, are part of several projects carried on during the thesis activities. For these particular services, the cost can be less relevant with respect to service reliability, coverage, time to be online, quality of service (QoS) guarantee, security, etc.

3.1.4 Telemedicine service

Telemedicine service can be exploited through a large set of devices, performed in a great number of contexts and for several reasons. One of the first targets of the telemedicine service is to reduce costs compared to ordinary medical assistance and can be applied at different stages of a medical aid: monitoring, diagnosis and therapy.

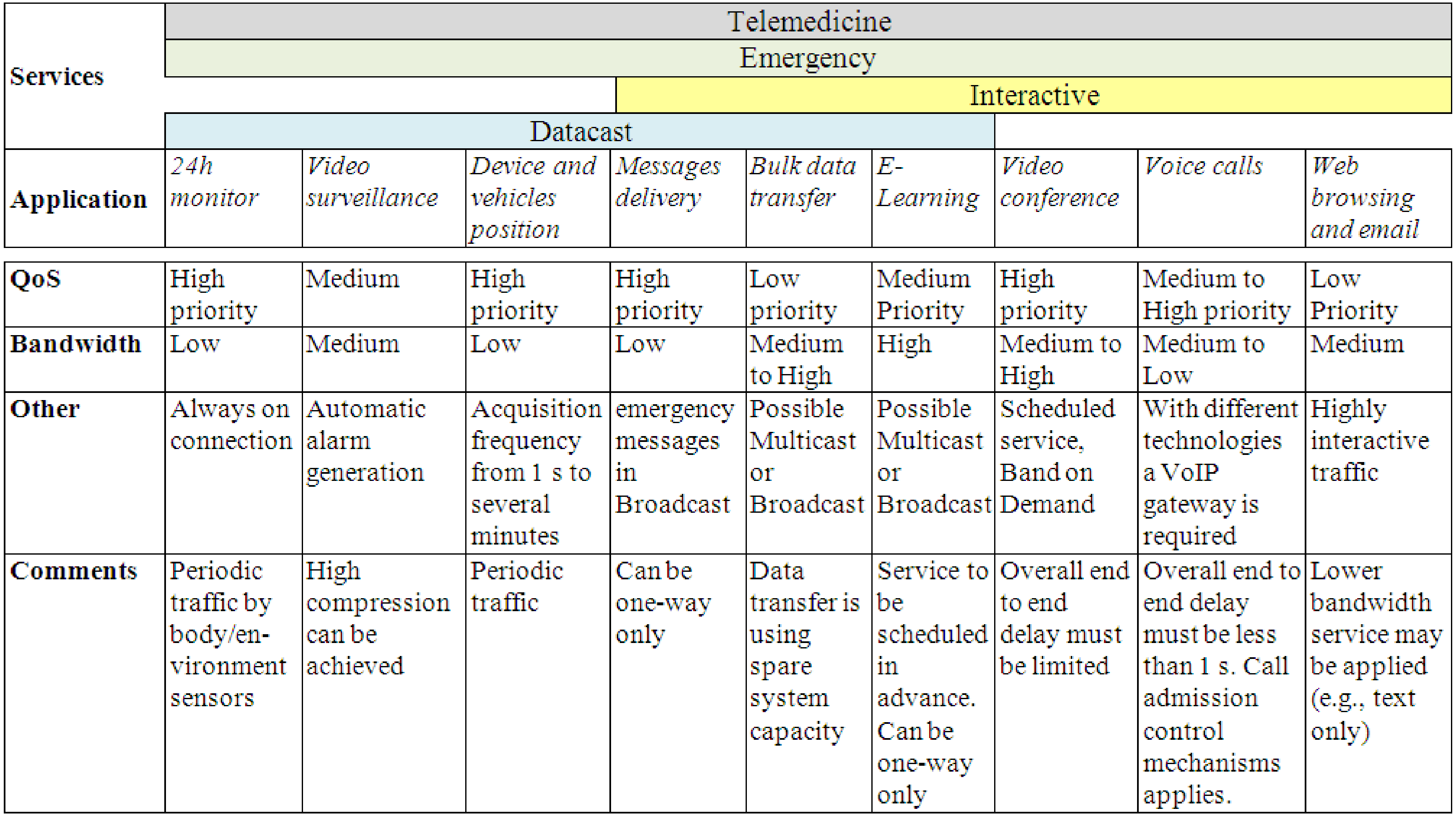
In fact in case of patients in remote areas (also on board of planes, ships and off-shore oil platforms) or needing to be continuously monitored (for example after a surgery), telemedicine can be adopted for monitoring and diagnosis of cases minimizing transfer costs of specialized personnel or patients. Moreover, the possibility to transfer preliminary information related to injured person using telemedicine is useful to decide for hospitalization and perform preliminary diagnosis, for an efficient triage execution.

Exchange of medical information and access to medical databases and patient records, together with a coverage in high density patient locations (e.g. nursing home) implies a great advantage for specialized doctors reducing the time spent for diagnosis and maximizing the number of patients visited.

In all these briefly mentioned cases, distribution of patients can be very different, from single units in remote areas, to some several in a wide disaster struck area or in a small place like a nursing house.

From the services outlined above, a set of reference applications can be proposed. Their classifications, characteristics and requirements are showed in table 3.2.

All the development and solutions identified are introduced considering all the needs of the reference applications, with the aim of making the hybrid network performing as required and introducing the proper optimizations to offer a better service and using efficiently the satellite channel.



Services	Telemedicine								
	Emergency								
				Interactive					
	Datacast								
Application	*24h monitor*	*Video surveillance*	*Device and vehicles position*	*Messages delivery*	*Bulk data transfer*	*E-Learning*	*Video conference*	*Voice calls*	*Web browsing and email*
QoS	High priority	Medium	High priority	High priority	Low priority	Medium Priority	High priority	Medium to High priority	Low Priority
Bandwidth	Low	Medium	Low	Low	Medium to High	High	Medium to High	Medium to Low	Medium
Other	Always on connection	Automatic alarm generation	Acquisition frequency from 1 s to several minutes	emergency messages in Broadcast	Possible Multicast or Broadcast	Possible Multicast or Broadcast	Scheduled service, Band on Demand	With different technologies a VoIP gateway is required	Highly interactive traffic
Comments	Periodic traffic by body/environment sensors	High compression can be achieved	Periodic traffic	Can be one-way only	Data transfer is using spare system capacity	Service to be scheduled in advance. Can be one-way only	Overall end to end delay must be limited	Overall end to end delay must be less than 1 s. Call admission control mechanisms applies.	Lower bandwidth service may be applied (e.g., text only)

Table 3.2: Applications classification

Chapter 4

State of the art

4.1 Technological aspects

Most of the standards considered for the realization of the integrated multi-segment network concerns the connectivity at lower layers using wireless and wired broadband links. As an example, the DVB-S standard for Digital Video Broadcast over Satellite only covers the first two layers of the OSI stack, leaving other aspects open to the implementer. Other standards studied include aspects which are orthogonal to lower layer specifications and are either independent higher layers architectures (e.g. Mobile IP) or part of standards extensions (e.g., 802.11i for extended security in Wi-Fi). Finally protocol specifications has also been analyzed.

The selection process was based on the most promising and widespread technologies coming from personal experience, standards analysis, execution of projects, coordination of thesis and exchange programs.

For a more clear exposition, the lower layer standards and other aspects of network integration, architectures and protocols are addressed in separate sections covering respectively security solutions, handover methods, quality of service and transport protocols highlighting commonalities and possible synergies.

4.1.1 MAC/Phy Standards

4.1.1.1 DVB-S

DVB-S (DVB over Satellite) standard [2] has been conceived for video broadcasting via satellite and is adopted for data delivery in the satellite forward link in bi-directional systems. The standard is composed of two main blocks: baseband processing and satellite medium adaptation. The base-band processing adopts MPEG-2 (Moving Picture Expert Group-2) as source coding of different input types (audio, video and data). The basic component of MPEG-2 system is known as Elementary Stream (ES). A program (i.e. TV program) is a combination of ESs generated by independent video, audio and data encoders. Each ES is an input for a MPEG-2 pro-

Modulation	FEC Coding	Efficiency (bit/Hz)	E_S/N_0(dB)
QPSK	1/2	0.99998	1
QPSK	2/3	1,322253	3,10
8PSK	3/5	1,779991	5,50
16APSK	2/3	2,637201	8,97
32APSK	5/6	4,119540	14,28

Table 4.1: Modulation and Coding examples for DVB-S2

cessor, which assembles data into a stream of Packetized Elementary Stream (PES). A PES is usually organized to contain an integer number of ES units. A number of PES are finally multiplexed into the MPEG-2 Transport Stream (TS) which is composed of packets with a fixed length of 188 bytes with a Packet Identifier (PID), which allows the receiver to reassemble the original PES packets.

The satellite channel adaptation concerns the types of modulation and coding schemes to be adopted to meet the target quality of the signal (Bit Error Rate of around 10^{-11}). The processes involved in this adaptation are transport multiplex adaptation and randomization for energy dispersal, outer coding (i.e., Reed-Solomon), convolutional interleaving, inner coding (i.e., punctured convolutional code), base-band shaping, and modulation.

In the last few years, the DVB-S standard was updated in the DVB-S2 version [3]. There are substantial changes in the satellite channel adaptation block also introducing different roll-off factors for baseband shaping, and different inner/outer coding strategies (e.g., LDPC codes, lighter FEC rates). Also baseband processing is different, intoroducing bigger frames (BBFRAMES) available in two fixed sizes, and offering improved encapsulation methods for data. Backwards compatibility is maintained through a legacy support for MPEG-TS encapsulation in the newer BBFRAMES.

In table 4.1 there is an example from the standard of the achievable spectrum efficiency using different Modulation schemes and FEC coding (MODCODS) and the respective required signal E_S/N_0 (received symbol energy over noise). It is evident that in case of better signal received it is possible to better exploit the satellite bandwidth compared to the DVB-S fixed QPSK modulation.

The new standard introduced also adaptive coding and modulation (ACM), so that the best MODCODS can be used dynamically according to the reception conditions. ACM requires the user terminals to notify the propagation channel conditions to adapt modulation and coding to comply with BER requirements.

4.1.1.2 DVB-RCS

DVB Return Channel Satellite (RCS) [4] allows bidirectional communications adopting DVB-S/S2 in the forward link and allowing up to 2 Mbit/s in the return link utilizing either MPEG-TS (as for the forward link) or ATM on top of the physi-

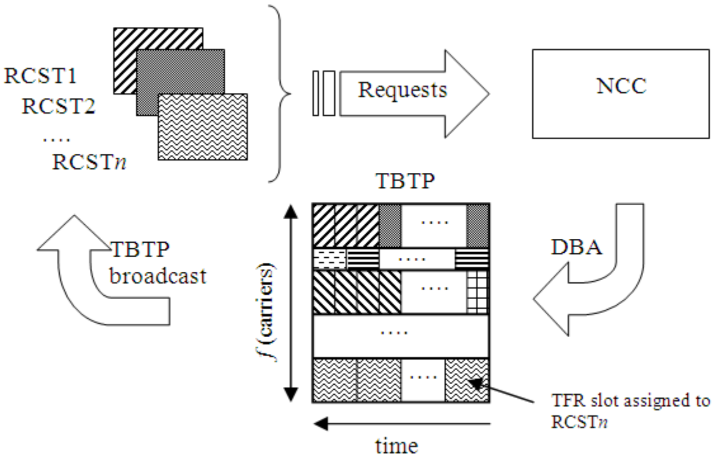


Figure 4.1: DVB-RCS messaging

cal layer. Satellite terminals (STs) share the bandwidth on a MF-TDMA discipline through a Demand Assignment Multiple Access (DAMA) scheme. NCC assigns time slots as a response to explicit requests (Bandwidth on Demand) coming from STs issuing Terminal Burst Time Plane (TBTP) in the forward link every superframe on the basis of the Service Level Agreements (SLAs) between operator and customers.

DVB-RCS standard allows fixed or dynamic time slot allocations according to five different schemes:

1. Continuous Rate Assignment (CRA) which is a fixed and static allocation of resources; after an initial set-up phase with a negotiation between the RCST and the NCC a given number of time slots are periodically assigned to a RCST every superframe until RCST sends the assignment release message;

2. Rate-Based Dynamic Capacity (RBDC) which allocates capacity dynamically requested on the basis of rate measurements performed by the ST; each request is absolute and overrides all previous requests from the same ST; every request shall be subject to a maximum rate limit directly negotiated between the ST and the NCC;

3. Volume-Based Dynamic Capacity (VBDC) allocates capacity dynamically requested on the basis of data volume measurements performed by the ST; these requests are cumulative (i.e. each request shall add to all previous ones from the same RCST), indicating a total number of traffic slots that are needed to transmit all data currently present in the MAC queue; the slots can be assigned through several superframes; the minimum and the maximum VBDC capacity that can be assigned to an RCST represent the main MAC parameters; VBDC is the default mode in DVB-RCS;

4. Absolute Volume-Based Dynamic Capacity (AVBDC) is similar to VBDC, but requests are absolute (i.e. each request replaces the previous ones from the

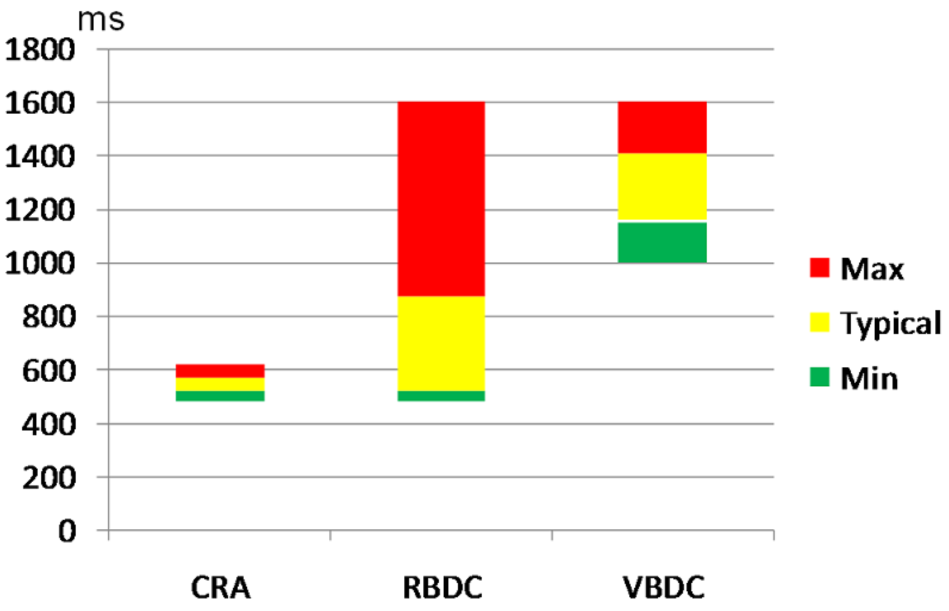


Figure 4.2: DAMA delays for TCP connections

same ST) and indicate a total number of traffic slots that are required to be assigned in several superframes; AVBDC is used for loss recovery in the VBDC mode;

5. Free Capacity Assignment (FCA) allocates capacity to STs from spare capacity that would be otherwise unused; such a resource assignment is automatic, not involving any request from the RCST.

The logical organization of a DVB-RCS system, including the messages loop and the generation of the TBTP is represented in figure 4.1.

For asymmetrical applications, DVB-RCS standard is identified as the most convenient solutions mainly for the possibility to use small low cost terminals (VSAT) and very flexible radio resource management. The star topology implies that two satellite RCS terminals must use the space segment twice (double hop) to directly exchange information: the first time to reach the HUB, and the second time to go from the HUB to the other terminal. Typical delay mainly due to the contribution of the physical propagation and the DAMA access scheme when TCP connections are established are reported in figure 4.2.

4.1.1.3 Skyplex

Skyplex [5] can be proposed as one of the candidate solution for mesh connectivity. It requires more complex terminals and bigger antennas compared to a star system but it can be convenient when the traffic load is high and symmetric and when a double hop associated latency can not be tolerated. Multiplexing of up-link data is performed on board of the satellite and broadcasted to all the terminals. It is possible to adopt on top of the Skyplex system a DVB-RCS like DAMA mechanism.

4.1.1.4 Wi-Fi (802.11)

IEEE 802.11 is a widespread wireless technology that provides qualitative connection service to mobile devices [6]. Maximal bandwidth available on 802.11 variants ranges from 11 Mbit/s (802.11b) to 54 Mbit/s (802.11a, 802.11g), which establishes a ready personal communication means to substitute non-mobile wired connections. 802.11 protocols work under two different scenarios: (i) multi-hop mobile ad-hoc networks (MANET), and (ii) last-hop wireless networks with a central star node. So far the latter case has been widely deployed in many places with the name "Hot Spot".

The basic system architecture considered is related to scenario (ii), with the use of a limited range star node called Access Point. A set of APs can serve multiple user terminal each, guaranteeing local mobility: when a terminal moves from the coverage area of an AP to another, the connection will be established through the new AP. To ensure connectivity among terminals the different APs are interconnected among one another and to allow interconnection with the global network they are connected to the backbone. Newer standard 802.11i and 802.11e are defining respectively stronger algorithms for security (WPA2) and QoS at MAC layer.

4.1.1.5 WiMAX (802.16)

WiMAX standard (Worldwide Interoperability for Microwave Access) formally referred to as IEEE 802.16, is proposed to offer broadband wireless access to medium / long range coverages [7]. The cellular like architecture allows to interconnect the different base stations both in point to point mode and in mesh mode providing up to about 70 Mbit/s capacity per station on a coverage ranging from few kilometers in NLOS and some tens of kilometers in LOS environments. WiMAX has an optional QoS enabled MAC layer and enforces strong encryption with time varying keys.

4.1.1.6 Bluetooth (802.15.1)

Ad-hoc wireless networks allow terminals to flexibly and autonomously organize themselves to communicate without a pre-existing infrastructure. To this aim an ad-hoc network connectivity is based on a peer-to-peer paradigm, and there is no functional difference among terminals. IEEE 802.15, of which Bluetooth is commercial implementation, is a transmission standard designed to support ad-hoc connectivity in a personal area [8]. When Bluetooth terminals get close enough, they can cluster into a piconet and temporarily designate one master unit to co-ordinate transmissions with up to seven slave units. Bluetooth is based on packet transmission and frequency hopping (FH) technologies to provide channelization among different piconets within the same area. Each service has a pre-defined QoS profile to announce during setup.

4.1.1.7 Other standards

Additional terrestrial broadband technologies has also been considered but not carefully studied for the aims of the thesis. These can be used to enhance and extend the satellite coverage to reach more users and at the same time aggregating the traffic to the same satellite terminal. As a brief introduction these are:

Hiperlan based on the European standard EN 300 652 V1.2.1, and similar to IEEE802.11a for what concerns modulation schemes and frequencies, is a broadband wireless technology often used as point-to-point direct bridge for line-of-sight medium distance connections. It makes use of directional antennas and fixed installations;

PLC Power Line Communications is a method of sending data using the distribution network of power lines; it is a wired technology suitable for last mile connections while avoiding the deployment of new cable/optical connections;

pico DSLAM are small DSL central nodes (managing some tens of DSL modems) that can overlay a digital signal to voice signals over simple copper wires and can be used as last mile connectivity strategy;

GPRS/EDGE/UMTS/HSDPA/LTE are data links available to cellular terminals of classic voice operators, presented in order of introduction into the market and performance; they are usually not completely manageable by the users but operated by commercial companies;

SatMODE/Astra2Connect offer a satellite narrowband return channel that can be used in alternative to DVB-RCS, where the upload capacity required is very limited, thus avoiding the use of additional wired/wireless last-mile technologies.

4.2 Security

Information Security concerns in general the concepts of cryptography, integrity, availability, authenticity, non repudiation, access control, and may have a broader meaning, applicable even to the hardware protection. For the purpose of the satellite/terrestrial integration, some security architectures and procedures of interest have been studied. Security can be implemented at different layers, with different approaches and several solutions are possible:

Layer 2 - DVB-CA [9] is an example of encryption applied at MAC layer, where single packets are encoded and decoded at the edge of the specific network. Each layer 2 technology has its own realization of security infrastructures, which is usually not interoperable.

Layer 3 - IPsec [10] is the standard currently in use to create security associations at network layer and is used as baseline for the enhanced security mechanisms embedded into IPv6 [11]. When using IP-sec the communication can be secured between the two ends of the communication, so that intermediate nodes or eavesdropper are not able to detect the content of the communication. No information are available to attackers concerning the classification of the traffic (e.g., understanding if the traffic is due to browsing, VPN, or email).

Layer 4 -Transport protocols such as TSL (or SSH) can be adopted. In this case it is possible to establish an end-to-end security association at transport layer, using SSL based solutions. This approach can be seen as a tunnel established above IP layer, where all data is encrypted and available in clear only at the end of the connection. This technology is connection oriented, and strictly based on the TCP protocol. An immediate advantage is that the intermediate nodes encountered during the routing path do not participate in the establishment of the secure communication. At the same time, protocols like SSL are widely standardized [12] and available on all platforms and operating systems (see for instance the HTTPS secure web pages).

Layer 5 - Security can be (and usually it is) applied at application layer, simply using algorithms inside client/server software to encrypt the communications. In this case unauthorized listeners are able to detect which application is in use (traffic analysis is thus possible) but still unable to decode the messages. An example is GPG [13], used to sign and encrypt emails. Security at application layer is not dependent upon the specific transport layer adopted and possible additional security mechanisms.

4.3 Handover

A wireless terminal connected to a wireless node may need to handover to another node for several reason: movement outside the coverage area, degradation of actual connection performance, possible cost saving, etc. Any wireless terminal usually support some kind of handover functionalities to another node. For instance IEEE802.11 based devices already foresee mobility support, as for instance the WDS operational mode to interconnect multiple Wi-Fi access points and create a bigger cloud of coverage allowing greater user movements.

The handover procedure is usually divided into two phases: decision and execution, either performed by the network or by the node.

Handover can involve the same physical layer at the old and new connection, resulting in a layer 2 handover (or horizontal). This kind of handover is the one usually the literature refers to: it can be soft, if the relation is established while the

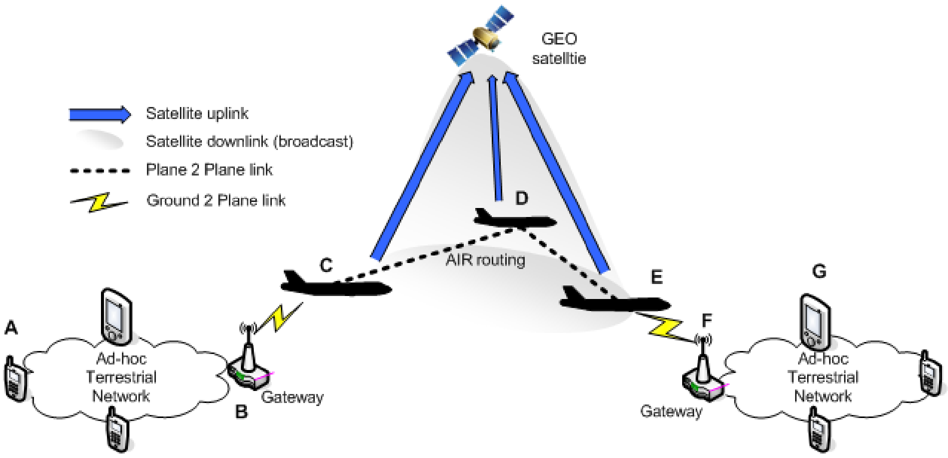


Figure 4.3: UAV to satellite handover

old connection is maintained or hard, if the old link is released before connecting to the new one.

Vertical handover implies switching between two different systems to keep the connectivity and implies the use of different hardware necessary to associate to a new channel using in general different modulation, MAC layers and frequencies, involving also higher layers (layer 3 handover). Vertical handover can be performed by dual hardware cards nodes, with both soft or hard handover execution possible, or also as a more recent application, completely in software (Cognitive Radio or Software Radio approach). In the rest of the work, this kind of handover will be deeply analyzed, since it can be used to alternate satellite connectivity to more convenient wireless connections, where available.

4.3.1 Vertical handover

The vertical handover needs a careful implementation due to technical and performance issuescoming from potential differences of the origin and destination networks in terms of latency, speed, overhead, etc.

A specific case studied is the handover performed by Airborne nodes, which are usually directly connected with a plane-to-plane link, to a higher delay geostationary connection, in case of necessity. This scenario is represented in figure 4.3. In particular TCP based end-to-end connection are established among users of different sub-nets (identified with letter A and G) which are connected by UAV (Unmanned Aerial Veichels), identified with letters C and D.

Another representative case of vertical handover analyzed is related to high speed trains performing handover from satellite broadband connections to terrestrial wireless local coverage, usually introduced as gap-filler to keep the connection into tunnels. In fact, the broadband satellite link works only in line of sight conditions, and another technology able to propagate inside the tunnel must be implemented. The terrestrial wireless connection as well can be available into train stations, where the

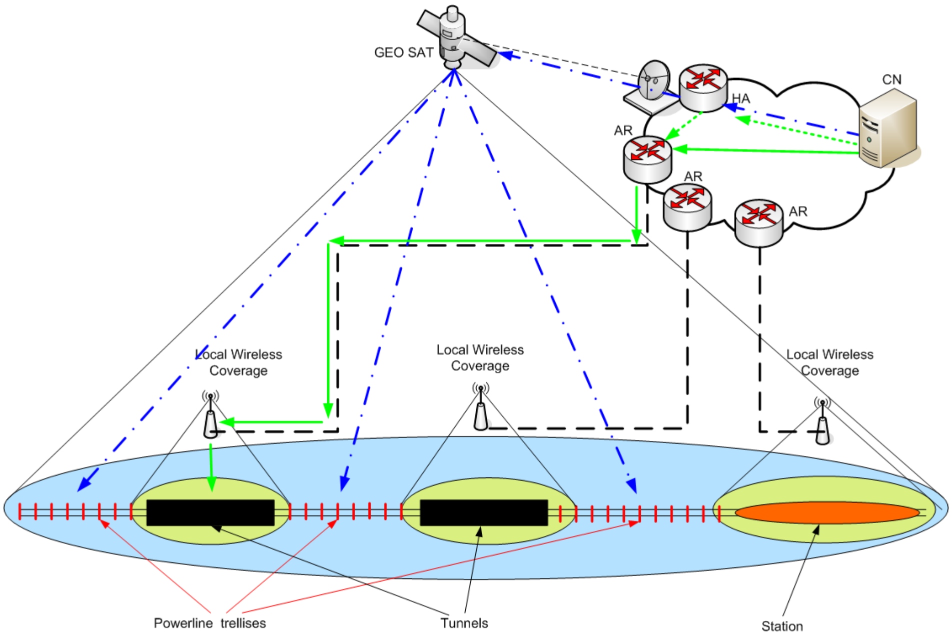


Figure 4.4: High speed train handover

train stops for a relatively long time and the satellite link can be relieved. Figure 4.4 depicts the considered scenario. Specific additional work concerned the analysis of the impact on traffic at network layer due to periodic electric trellises without introducing redundancy in hardware or specific countermeasures at lower layers.

4.3.2 MobileIP

When crossing different segments of the network which adopt different addressing, the network must support a network layer mobility mechanism (such as Mobile IP MIPv4 [14]) in order to support a terminal switching from a subnet to another, with additional network core elements (e.g. Home Agent) required. These solutions at network layer may imply a disconnection time with a consequent loss of the relations/connections already established.

If the transmission must be seamless during the switch, the network must also support seamless handover, to guarantee an uninterrupted IP data flow (fundamental for instances in TCP based applications). In order to implement efficient handover among different segments maximizing data transfer, suitable algorithms must be set up and smart solutions must be adopted to minimize inefficiencies at transport layer, together with a consistent signaling.

Mobile IP (MIP) protocol provides mobility support at network layer, allowing mobile nodes to change point of access to an IP network transparently. Mainly, MIP envisages the definition of a location management agent, called Home Agent (HA), which keeps the association between a home permanent IP address assigned to the Mobile Node (MN) in the home network and several care-of-address (CoA) assigned when visiting other networks. When MN moves to a new location, it obtains and registers a CoA with HA. CoA can be, for instance, an IP address belonging to the subnet of the new access router, called Foreign Agent (FA), to which the MN is connected to. Therefore, all the packets directed to the MN are intercepted by HA, which encapsulates and forwards them to the CoA. The FA then de-encapsulates incoming packets and delivers them to the MN. Packets generated by the MN instead have as source the permanent IP and not the CoA, so that explicit support for mobility must be implemented in the routers of the visited networks. MIP has been introduced to offer mobility to nomadic users giving a dynamic point of access from different wired networks keeping a unique IP address reference. This concept has been extended to wireless networks, which are more suitable to mobility and more demanding in terms of handovers procedures. Nevertheless, in all its realizations MIP has to deal with the so-called "Handover Latency", or out of service time, and it is not able to mask the impairments of such latency to higher layers. In fact a MIP handover results in an interruption of the communication at layer 3 with consequent packet losses. The duration of such interruption usually is composed of the time to connect MN to the new link (HO performed at layer 2) and the time needed to update routing (HO performed layer 3).

4.4 Quality of Service (QoS)

Guarantee of the QoS on a hybrid network is critical, to offer the highest degree of performance to selected applications. Several approaches are possible, either centralized or distributed, and need the cooperation of different layers of the OSI stack and of involved network nodes. TCP/IP based networks are usually best effort, relaying for instance on collision prone connections such as Ethernet links and require specific approaches to obtain QoS. Furthermore, in case of lower layer QoS connections (e.g., ATM based links), when crossing different networks adopting different technologies, which is the typical case when using IP network layer protocols, QoS solutions cannot be compatible. When a technology supports QoS at lower layer, typically it means that it supports different queues at MAC layer, and different classifications policies. An example of MAC layer QoS support is the extension of Wi-Fi standard 802.11e, covering the differentiation and handling of different types of traffic associated to connections at transport and application layer, using different hardware queues.

The specific aspects addressed during the execution of the thesis concern the implementation of DiffServ traffic prioritization in DVB-RCS network. Such aspects are detailed in the following sub sections.

4.4.1 Differentiated Services (DiffServ)

Differentiated Services (DiffServ) is an architecture to achieve statistical QoS to IP based traffic, in a scalable manner. Before entering a DiffServ domain, which is operated by the same entity (e.g., autonomous system) with a common router behavior, IP packets are inspected to determine application related information and obtain a classification of IP packets into logical flow with different priorities. For instance UDP traffic, which is associated to a real time video content, can be classified as Guaranteed Service. The classification covers all the range of applications requirements and are Expedited Forwarding (EF), related to low-loss and latency traffic, Assured Forwarding (AF) which guarantee the data delivery with good performance under normal network load, Best Effort, for all other flows. Each of the above mentioned class can be divided in subclasses, usually called AF_1, AF_2, ... AF_n.

After the IP packet has been marked according to the DiffServ Domain policy (usually acting upon its TOS field), the packets are routed from one router to another following the priority criteria defined. Since this operation is performed at each router of the DiffServ Domain, it is also called Per-Hop-Behavior.

In practice DiffServ is a statistical approach, since QoS is not guaranteed to single applications' traffic, but rather to a class of traffic. In this way, the system is much scalable in presence a large amount of traffic and transport protocol sessions (stateless behavior).

Finally DiffServ classification process can be in conflict with IPSec in case the IP headers are encrypted, so that external markers should be used.

4.4.2 SatLabs specification for DVB-RCS QoS

SatLabs is a group founded by the European Space Agency (ESA) to work on extensions of DVB-RCS standards, to achieve interoperability among different equipments, also at higher layers. In a technical document on QoS support in DVB-RCS systems [15], they address the issues and provide guidelines to add QoS features to interoperable DVB-RCS networks.

The document proposes the DiffServ approach applied to the edge nodes of the DVB-RCS network, using a set of queues with different priorities together with a set of managment messages and signalling useful for a correct network setup.

From the document [15] the following parts, which are important for the system design, are recalled: *"A Request Class (RC) is a representation of a PHB at the MAC layer (in the DAMA Controller). It defines a behavior of the MAC layer for a given aggregation of traffic [...] An RC is a concept similar to the PHB, but seen from layer 2 [...]Several requests of the same category may be generated by an RCST, one per RC"*. RC are in direct relation with DAMA requests using a relation which is left open by the Satlabs documents.

4.4.3 C2P in DVB-RCS

Recently a new extension for DVB-RCS standards has been issued [16], with the introduction of a Call Control Protocol (C2P). C2P allows to adjust DiffServ queues, traffic classification and mapping to MAC frames dynamically and with extreme flexibility, in order to offer guaranteed quality services. In practice, when establishing a C2P relation involving one or more satellite terminals and the Network Control Center (NCC), the resources needed by a specific application are negotiated *a priori* and, if allowed, kept for the whole service duration. Typical C2P messages are for Service Establishment, Modify and Release, quite similar to SIP/IMS signalling [17].

Furthermore C2P channel can be used to adjust lower layer parameters, not direcly related to QoS (e.g., terminal IP address, subscription types, etc.).

4.5 Transport protocols

4.5.1 TCP basic concepts

TCP is a reliable transport protocol, based on explicit acknowledgment of received data, implementing a flow control and congestion control to transmit blocks of data at the optimal rate and avoiding congestion of the network [18]. It is a connection oriented end to end protocol, on top of connection-less IP protocol.

TCP was originally designed to work efficiently in wired and congested networks but suffers from a certain number of factors when running in hybrid networks including satellite networks. Specifically, high latency, large bandwidth-delay product (BDP), link asymmetry, sudden BDP changes due to handover and channel errors can negatively impact TCP performance, as illustrated for instance in [19]. In literature a large number of solutions have been proposed to improve TCP performance in such cases. A possible classification of proposed solutions is the following:

- Non-TCP enhancements – provision of enhanced lower layers; in other words, TCP can benefit passively from mechanisms running at different protocol layers;

- TCP standard enhancements – extensions compliant to standard TCP protocol specification;

- TCP variants – modifications of the standard flow, congestion and error recovery schemes;

- Performance Enhancing Proxies (PEPs) – modifications of the TCP end-to-end semantic (including architectural modifications).

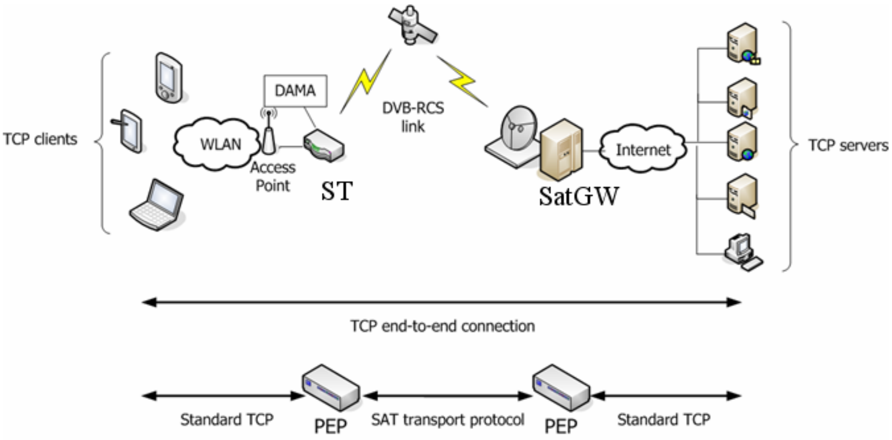


Figure 4.5: PEP: Split Architecture

Several solutions has been studied, triggering the definition of original ones and new architectures, which are introduced and discussed in the following chapters.

4.5.2 UDP-based protocols

Where possible, UDP based protocols can be used. UDP is connection-less protocol which extends IP packets with ports to reach a specific application (socket). All features which can be required, such as reliability, must be implemented at application layer. This protocol is especially adopted for real time (unreliable) transmission and multimedia streaming, and also as inner transport protocol when adopting splitting based solutions ([20]).

A UDP-based protocol studied for the thesis includes higher layer FEC coding (of the class called digital fountain and specifically LT coding [21]) to achieve reliability and maintain efficiency in the transmission especially for noisy channels and in multicast configuration.

4.5.3 Split architecture for satellite

Where referring to PEP techniques, different solutions are usually contemplated ([22]). The split architecture has been studied and implemented, to enhanche the performance of standard TCP end-to-end protocol when crossing the satellite. At the ingress of the satellite segment, TCP flows are intercepted and translated into a different transport protocol, specific for the satellite segment. At the satellite segment egress, the connection is restored to standard TCP. The principle is depicted in figure 4.5 in relation to a DVB-RCS typical scenario.

To conclude the section, an overview of the protocol stacks and the logical blocks involved in the satellite segment for a DVB-RCS link are reported in figure 4.6, including elements used for QoS enforcment and details on the specific protocols and standards used.

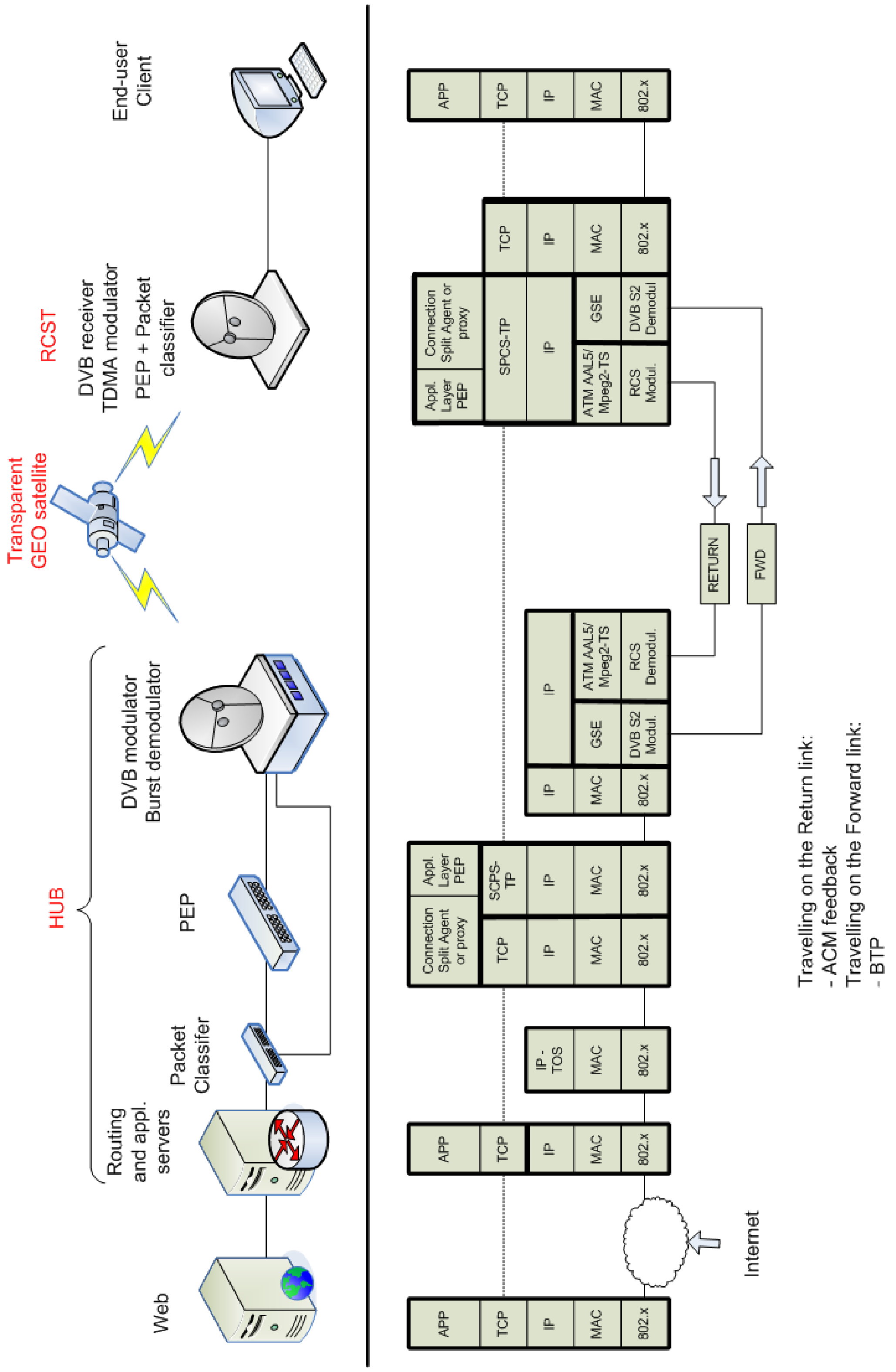


Figure 4.6: Protocol stack for a complete DVB-RCS system

Chapter 5

Proposed Approach

5.1 Analysis

The first step for the identification of the final integrated architecture, the most suitable protocols and promising solutions, has been in first instance a critical analysis of the requirements of the final network, coming from experience developed during execution of several projects making use of satellite networks, with collaboration with end user and institutional partners. A very high level list of requirements is reported below:

R1. The final telecommunication network must support segments using different technologies;

R2. The services of interest must be available in a broad set of scenarios (e.g., search and rescue in disaster struck areas), without substantial modifications to the network architecture;

R3. The network must be easily deployable, with a coverage adequate to the scenario and available to satellite terminals but also other COTS wireless devices by means of an adaptation (gateway);

R4. The services must be offered by means of one or a combination of several applications, with different requirements (e.g., real-time, timeliness);

R5. The network must offer to the applications the right degree of security;

R6. The network must include functions to differentiate the data flows and enforce QoS;

R7. The bandwidth of the satellite segment must be wisely used (due to its high cost);

R8. The network must be usable in emergency situations, with an high degree of reliability.

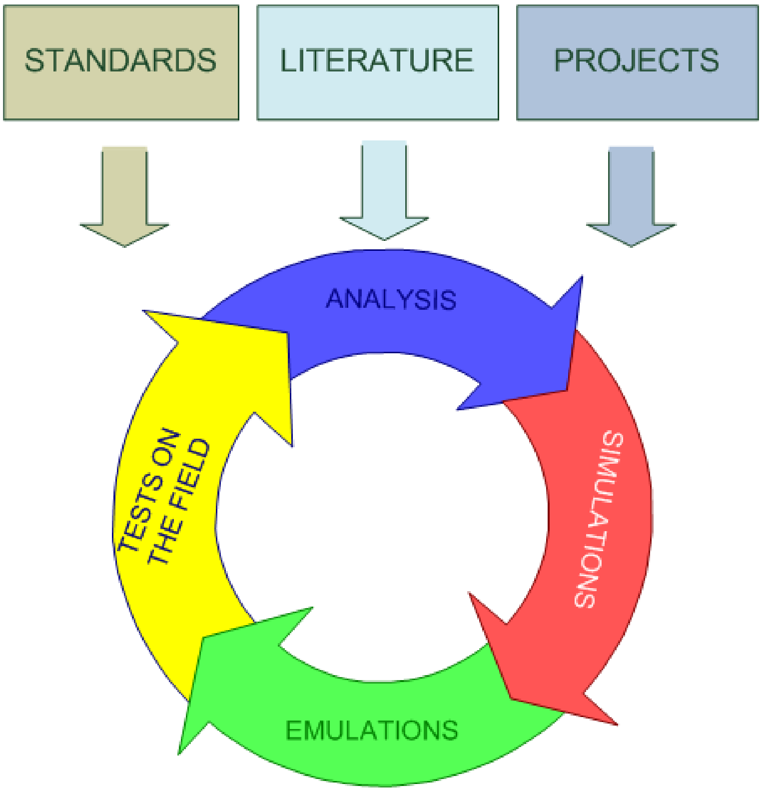


Figure 5.1: Followed approach

To meet the requirements adopting a common and flexible architecture, a careful study by means of analysis of standards has been performed, with feedback from real installations and existing solutions. After a harmonization process, the analysis activity continued with the evaluation of possible technological limitations, best choices for what concerns the network, the protocols, the applications, integration issues at different levels (physical, logical, security, QoS support, etc.).

Then, a series of proposal and enhancements are introduced reinforced by a process of simulation, emulation and real tests, following the study logic represented in figure 5.1. Specific details on solutions, tests execution and outcomes are included in the following sections.

5.2 Simulation

Most of the activities concerning the simulations for the scenario under investigation were performed using Network Simulator. The Network Simulator 2 (NS2) package has been developed in the frame of VINT Project [23] to analyze protocol's behavior in the wired network. Subsequently, several software modules have been added to extend the analysis also to the wireless and the satellite networks. In particular, NS-2 is an object oriented simulator, written in C++, with an OTCL (Object Tool Command Language) interpreter as a front-end. The simulator supports a class hierarchy in C++, and a similar class hierarchy within the OTCL interpreter. Users can implement new simulator objects through the interpreter; these objects are instantiated within the interpreter, and are closely mirrored by a corresponding object in the compiled hierarchy. Communication architectures and simulation parameters are defined through an input file (script), written in OTCL language. As results, the simulator provides either output files or graphs. In the frame of the proposal's objectives, NS2 already includes the implementation of many TCP enhancements (i.e. TCP Westwood, TCP Vegas, TCP Hybla, etc.) and easily allows to handle a large set of TCP parameters (i.e. socket buffer size, initial slow start threshold, initial congestion window, MSS, etc.). Furthermore, thanks to a modular structure and a particular care of the transport layer issues, NS2 makes the implementation of new transport protocol relatively easy.

NS2 has also been extended with custom modules, which are described in the the following chapters.

Some simulation activities have been complemented by Matlab pre-processing of data (for instance to obtain specific channel models) or ad-hoc simulations written in C++.

5.3 Emulation

In addition to simulations activities, emulations based tests has been performed.

A Satellite emulation platform allows to test both real applications and TCP/IP protocols in a communication scenario similar to the real one, but greatly increasing flexibility and avoiding costs of expensive satellite capacity and terminals/hardware leases. It is however necessary that the emulated environment matches the real system as much as possible. The emulation tool considered in the rest of the thesis is called Satellite Network Emulation Platform (SNEP), validated to match the behavior of a real DVB-RCS system. It is realized with a network of five computers, each one performing the functionalists of one or more elements of a real DVB-RCS network. The emulation platform is based on work performed by previous students [24] and enhanced during the execution of the Ph.D. with additional modules and features, which are all detailed in the following chapters.

5.4 Tests on Real Platform

Some of the proposed solution have been tested using real satellite terminals, also taking advantage of the availability of ESA/ESRIN laboratories during the execution of the TOP project [25]. The satellite platform considered is the DVB-RCS compliant Alcatel 9720 Mini HUB with EMS RCSTs. In chapter 9 the main outcomes are illustrated.

Chapter 6

Analysis

6.1 System Integration

The integration of different links using different standards is not straightforward. For instance differences in framing sizes, in procedures and protocols may lead to inefficiencies or inability to inter-work at all. The solution, widely adopted nowadays in several contexts, is the convergence towards IP based communication. In this way, the network layer is responsible to perform addressing functions, adaptation to lower layers, enforcing flows prioritization, security, mobility as identified in the previous section. Of course this approach has some drawbacks, such as increased complexity of each node (which is required to support the TCP/IP stack), complexity in the management of the network, increased communication overhead and computational power. Sometimes it is also necessary to force the splitting of connections on different segments, change protocols and break end-to-end communications at transport layer.

Nevertheless this approach has been considered the best tradeoff, and proved a perfect suitability for the hybrid network subject of this thesis. Nowadays TCP/IP protocol is included in almost any device connected to a network, but particular attention is needed to use such protocol over satellite segments, where bandwidth is usually expensive and delay is huge.

A practical integration of segments using different technologies has been performed by means of an emulation platform using different wireless access points and wired connections, testing end-to-end Quality of Service, data encryption at layer 3 and other aspects, detailed in chapter 8. Also testing with real IP based devices (medical equipment) has been performed and validated.

6.1.1 Resource management

The resources available to the system should be carefully assigned with a certain degree of efficiency, especially when considering the satellite bandwidth. In the specific case, resource management is considered only on the satellite segment, assuming that terrestrial links do not need to grant resources on request but can work on overprovisioning.

In DVB-RCS systems a Demand Assignment Multiple Allocation mechanism is adopted (DAMA). All requests form satellite terminals are collected by the NCC on superframe basis (typically each 26.5 ms or multiples) on a dedicated signaling channel. The NCC than computes the resource allocation using a Dynamic Bandwidth Allocation algorithm (DBA), whose response is broadcasted back to all terminals to schedule their transmission. The response assignment to each Satellite Terminals (ST) is a subset of slots of the MF-TDMA, which are either ATM cells or (optionally) MPEG-TS frames.

The DVB-RCS standard does not define any specific DBA algorithm: DBA can be a key element for an efficient and fair distribution of resources among all RCSTs. Some DBA algorithms has been proposed and studied:

1. Round Robin - grant slots one by one in turn to each terminal until resources are over or all terminals requests are satisfied;

2. Proportional - assign a number of resources proportional to the absolute value of the request, normalized to the total amount of resources available;

3. Proportional with minimum guaranteed - as proportional but avoiding that a ST with a smaller request than the average could starve;

4. First Come First Served - first ST logged into the system are served in order until resources are over;

5. Priority - as in the proportional but with a weighting factor which can be assigned by the network manager to give different priorities to terminals; this specific DBA algorithm pursue the maximization of an utility function [26] and is computationally more complex than the Proportional one.

Also for what concerns the request elaboration by the STs, the standard leaves freedom to the implementer to use its own algorithms. For instance the way rate is estimated for Rate Based Capacity Requests (RBDC) can be performed in several ways, even with predictive filters [27]. In first instance first order filters has been studied [28] and implemented.

6.1.2 Security

IPSec applicability to the DVB-RCS environment has been studied, including the impact of additional headers and its compatibility with PEPs. IPSec introduces a measured overhead of about 4% and in tunnel mode is completely hiding the content of the packet, avoiding the possibility of using split proxies.

DiffServ PHB by SatLabs	DVB-RCS RC	DAMA requests
EF	Real time (RT)	CRA_{EF} (+ RBDC_{EF})
AF_1	Critical Data (CD)	CRA_{AF} + RBDC_{AF}
AF_2	Critical Data (CD)	RBDC_{AF} or (CRA_{AF} + VBDC_{AF})
AF_3	Critical Data (CD)	VBDC_{AF}
BE	Best Effort (BE)	VBDC_{BE} + FCA

Table 6.1: PHB to RC to MAC mapping (SatLabs)

Secondly higher layer security has been considered (TLS), so that content of TCP and IP header is completely visible and the split architecture approach is feasible. Nevertheless, an attacker in the middle of the connection, even in the impossibility of obtaining the content of the communication, is able to understand that a security association is established, with which protocol versions and between which end users (IP address). With this information some statistical information can be obtained, such as traffic profile and application classification, which is not beneficial for higher security users.

Starting from this preliminary analysis some tests has been performed with both simulations and emulation of real IP traffic.

6.1.3 QoS

The guidelines of SatLabs [15] should be followed to implement QoS support in a satellite system of an hybrid network. The architecture suggested in the document, which can be considered as a DVB-RCS DiffServ Domain, has been analyzed. The table 6.1 has been proposed after an accurate analysis to map Per-hop-Behaviors with mandatory Request Classes and then MAC requests, in compliance with the SatLabs guidelines.

The proposal of mapping is listed in column 3, with in subscript a further distinction in priority for DAMA requests (e.g., VBDC_{BE} is satisfied only after VBDC_{AF} has been assigned) and will be in particular applied to the emulation environment.

The QoS can be enforced by means of a table with static entries (protocol type, source address, destination, address, source port, etc) to map connections to a PHB in each of the Satellite Terminal of the network.

During the execution of the thesis a preliminary implementation of C2P signaling for dynamic QoS achievement has been addressed and shown in the emulation platform chapter. For instance the sequence diagram shown in figure 6.1 has been derived from the standards and its compatibility with static QoS handling has been studied and adjusted. In this case a new connection triggers a request for a given type of service (RT – Real Time, CD – Critical Delivery, BE – Best Effort), a Sustainable Data Rate (SDR) and a Peak Data Rate (PDR). If the PDR/SDR request is compatible with the terminal profile and the resources available, the nodes involved

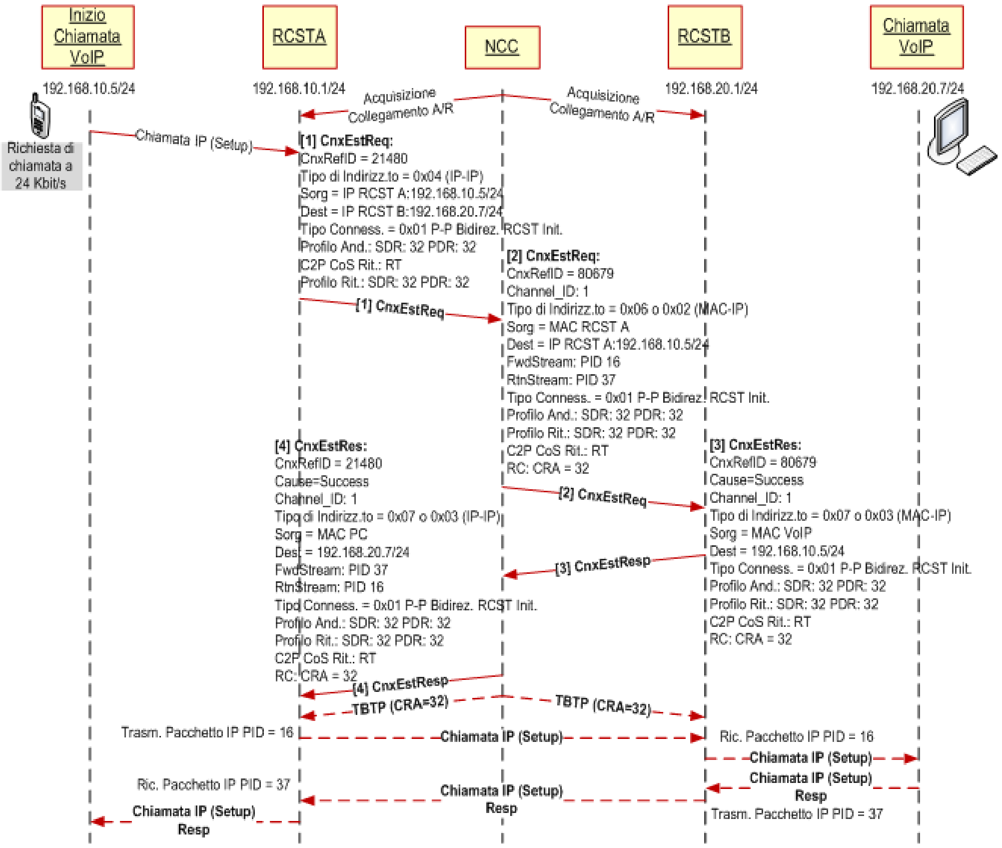


Figure 6.1: C2P messages for VoIP connection establishment

in the connection receive a dedicated channel for data transfer (modified TBTP). Such channel will have a priority according to the traffic already prioritized for static DiffServ QoS, in practice doubling the number of queues at MAC layer.

With the support of C2P the NCC must perform operation of Call admission Control (CAC) with a local state associated to each connection in order to evaluate if resources are available and take into account the connections released.

6.2 Handover impacts on TCP

Theoretical studies on handover effects on classic transport protocol has been performed. In fact although a full MIPv6 infrastructure can be set up to support seamless HO, several problems affect TCP performance. In fact, TCP protocols are not designed to optimally handle mobility, especially when HO involves links with different delay and bandwidth-delay product, as in the hybrid satellite terrestrial network considered. The following considerations are made for the specific case of connectivity onboard of a high speed train performing handover from a satellite connection to a wireless terrestrial gap-filler inside tunnels, but are applicable in general to other scenarios (vehicles, planes, etc).

Reordering and burst generation The packet reordering may occur when packets of a TCP flow take different network paths characterized by quite different propagation delay. This problem is extremely important in the considered scenario, where a soft-HO is performed at both the beginning and the end of each tunnel. There is a time period after the connection establishment with the new network, during which the Mobile Node exchanges data over both satellite and terrestrial link. While the old link continues to deliver packets in-order, the new interface may already deliver packets with a much higher sequence number. This problem arises particularly when the propagation delay of the new link is lower than propagation delay of the old one. For this reason, at the receiver side it is necessary reordering packets. In addition, arrival of out-of-sequence packets causes generations of duplicated ACKs (with or without SACK option enabled) of the last packet received in sequence. This behavior may trigger premature retransmissions.

On the other hand, TCP sender may also generate large burst of packets. Specifically, ACK S_i for the first packet sent over the new terrestrial wireless link could anticipate a quite large number (N) of ACKs $[S_{i-1}, S_{i-2}, ..., S_{i-N}]$ still in transit over the satellite link. In case a cumulative ACK scheme (i.e. TCP Reno), a burst of N packets is injected over the terrestrial link (if we are in Congestion Avoidance phase otherwise $2 \cdot N$). TCP SACK behaves in the same way since receiver is assumed to generate ACK in the correct order.

Time-out expiration The retransmission timeout (RTO) is the default mechanism to detect permanent losses and perform retransmissions: if a packet is not acknowledged before the RTO expiration, TCP sender performs a retransmission. Simultaneously, TCP sender dynamically updates RTO value on the basis of RTT measurements. RTO value is dynamically updated on the basis of RTT measurements performed by the TCP sender and it usually has the same order of magnitude of RTT. In addition, “exponential back-off” algorithm doubles RTO upon an expiration. During HO from terrestrial wireless and satellite, RTO is set according to the RTT experienced in the terrestrial wireless, while packets exchanged over satellite link experience a quite larger RTT. This means that RTO most likely expires one or more times before receiving ACKs of packets correctly received. Accordingly, several unnecessary retransmissions are performed. This problem affects all the considered TCP versions. Both TCP Reno and TCP SACK restart with a *cwnd* equal to 1 and a *ssthresh* equal to half of the *cwnd* value before RTO expiration.

Bandwidth-Delay product change Both TCP *cwnd* and *ssthresh* are dynamically changed on the basis of the sender estimate of the BDP (through RTT measurements). Since BDP of the satellite link (BDP_{sat}) is generally much larger than BDP of the wireless terrestrial link (BDP_{tl}), two different problems can be experienced during handovers:

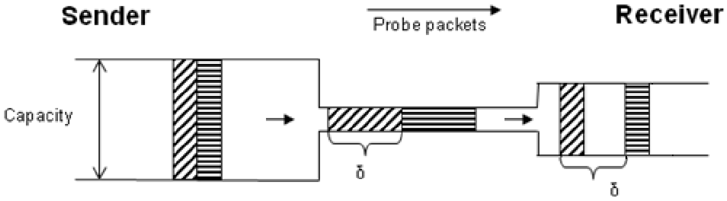


Figure 6.2: Measure the bottleneck link capacity

1. Moving from satellite to terrestrial wireless network, *cwnd* and *ssthresh* are usually set to values higher than BDP_{tl}; then, the number of packets injected over the new network exceeds link capacity potentially leading to buffer overflows, with the possibility to drop a large amount of packets triggering TCP timeouts.

2. Moving from terrestrial wireless to satellite network, both *cwnd* and *ssthresh* are set to values much lower than BDP_{sat} (even considering a possible timeout expiration as explained before); in these conditions, TCP sender must perform a large increase of *cwnd* before matching the optimal value equal to BDP_{sat}; since most likely TCP performs Congestion Avoidance algorithm (linear *cwnd* increase on a RTT basis) the increase of the *cwnd* up to the optimum value requires long time, resulting in a significant underutilization of the satellite resources.

6.3 Path characteristic discovery

It is possible to perform an analysis of channel characteristics (e.g., available and maximum bandwidth) based on the observation of the statistical properties of received packets [29, 30, 31].

A better knowledge of the lower layers for the transport protocols is beneficial to try to exploit better all the available resources, avoiding congestion, improving fairness and better reacting to sudden channel variations (for instance due to a handover).

As an example, the concept of dispersion to measure the bottleneck link capacity is presented in figure 6.2. Three links with different capacity (proportional to the height of the pipe) are interconnected, and IP packets are sent from a sender to a receiver. If two packets of size P_{size} are sent back to back, the second bottleneck link will space them according to its capacity, so that when packets are going through the third link, the same spacing δ is maintained. The bottleneck link capacity can be estimated as $\tilde{C} = P_{size}/\delta$.

Considering other metrics, such as variation and jitter in the measures of δ, variation in the absolute value of packets delivery (RTT), packets losses, with opportune filtering strategies it is possible to detect some conditions in the network that can help the transport protocol to behave better, such as detection of handover, or determine the presence of a DAMA scheme. The study in this context has been continued through simulations in NS2.

At the same time these observations, which are usually performed without introducing additional traffic but monitoring real traffic packets (usually ACKs of the TCP protocol) have also been considered for the design of a new transport protocol for DVB-RCS links.

6.4 Enhanced TCP for networks with DVB-RCS links

End to end protocols crossing a satellite segment usually underperform, mainly because of the high BDP of the link . For this reason an activity about the design of a specific TCP for the satellite environment, compatible with I-PEPs defined by ESA [32], was carried out. The main goal of this new protocol was to optimize the transfer of small objects, because other TCP versions usually end the transmission before reaching the maximum channel capacity. At the same time a good channel utilization must be reached, to avoid resource wasting.

The analysis process lead to a variant of TCP called TCP-Noordwijk in which the transmission of packets and the reception of acknowledgments (ACKs) are parallel processes. The main idea proposed for the protocol is that the reception of ACKs is used to generate detailed statistics of the underlying DVB link, such as channel capacity (which is varying over time), channel usage and rate measurements. Such statistics are used to decide the optimal transmission rate for new packets. The transmission is performed in packet bursts, which allows to reach the maximum efficiency and let to estimate correctly all the parameters required. The rationale of this approach is to exploit at best the satellite resources, pushing data into the ST buffer and at the same time to enforce congestion control to decrease the rate upon signal of presence of other traffic. This protocol is applicable in a controlled environment, where all the size of buffers and main characteristic of the channel are known in advance, not limited to a DVB-RCS link. TCP-Noordwijk must also guarantee fairness in case of multiple flows and friendliness against flows running over other transport protocols.

TCP-Noordwijk detailed functional blocks are represented in figure 6.3. In the dashed boxes there are the functions in charge of evaluating the channel conditions and the need of retransmissions. According to these statistics, the transmission rate can be reduced (“Rate Adjustment”) or increased (“Rate Tracking”), up to the actual estimate of the bottleneck channel capacity. The “Burst transmission scheduling” is a parallel process which effectively sends bursts of packets correctly spaced and

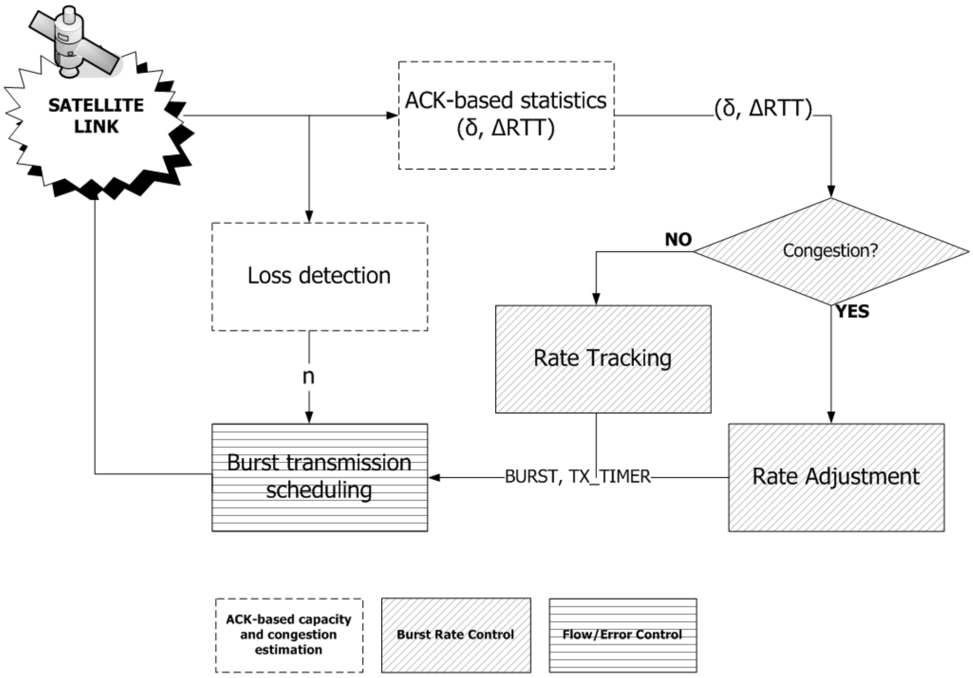


Figure 6.3: TCP-Noordwijk logical blocks

sized according to the previous algorithms, taking into account the necessary retransmission (the SNACK recovery algorithms is used [32]). It is a parallel process because it can run independently from the reception of ACKs, which are only used to update statistics. In this way it is possible to avoid the direct connection between packets sent and acknowledgments received typical of most TCP versions, which in case of high propagation delay can lead to lower transmission rates. At the same time, proper timeouts must be enforced to avoid the unnecessary overflooding of the channel in case of problems.

The algorithms detailed implementation is illustrated in [47].

6.5 Cross layer

A meaningful approach to keep good performance on hybrid satellite terrestrial wireless networks is based on the application of the cross-layer approach. When applicable, the traditional layered structure can be modified allowing the different layers to exchange information, in order to make each layer aware on events handled at other layers and then to adapt its behavior according to the global status of the protocol stack.

6.5.1 Cross layer architecture for handover

A specific cross-layer mechanism to increase performance for vertical handover enabled devices and help TCP improve performance during handover events has been proposed. Three cross-layer classes of messages are introduced:

1. From Link to Transport Layer (s_1), on the basis of L2 status (e.g., measured BER) a) if the Mobile Node is connected to the terrestrial wireless network, as soon as the signal-noise ratio over the terrestrial wireless link degrades under a pre-selected threshold b) if MN is connected to the satellite link, if a new terrestrial wireless network is in the MN's range; these messages can be used to "prepare" TCP to the forthcoming HO event (e.g., freeze the transmission to avoid losses during the transition);

2. From Link to Network layer (s_2), when the new link is available to be accessed, a L2 generated message advertise the L2 handover to the network layer; this message indicates that handover procedures can be started.

3. From Network to Transport layer (s_3), when the handover procedure is executed the network layer sends a disconnection signal to transport layer in order to announce the disconnection; in addition, when handover is completed and the new link connection is established, it can send a connection signal; both connection and disconnection signals are then used to make proper adjustments to the TCP status variables.

These messages can improve data transfer performance and are all local messages, assuming that the node that performs the handover is the same endpoint of the TCP connection, or that a split architecture is defined. They can be easily implemented in reference TCP/IP stacks (in NS2 or over Linux platforms) introducing a mechanism of message exchange and a central dispatcher.

6.5.2 Cross-layer in DVB-RCS systems: TCP layer

Standard TCP connections performance over DVB-RCS suffers of the effects of DAMA dynamic allocation for the data transfer from RCSTs to NCC. For this reasons TCP can be designed to better cope with any underlying variations that might occur using information derived from MAC layer queues. This information is in direct relation with the amount of resources granted by the DBA algorithm execution in the NCC and is locally available to the RCST. The scope of the cross layer enhancements for TCP that are proposed is to introduce countermeasures to limit some negative reactions of TCP connections running from the RCST to the NCC. In fact TCP, which is experiencing a very high RTT, needs a very large buffer at the bottlenecks of the network to reach the optimal throughput. TCP Reno, which is the standard version adopted in common operating system (with Selective Acknowledgment support [42]), is increasing its transmission window up to this optimal value, keeping on pushing packets in the buffer until a loss occurs. Usually the buffer is large enough so that when the loss occurs, the RTT is several seconds. This has as consequence a very high degree of losses, since the transmission window is oversized, and also decrease the fairness against new connections, which start with a very high RTT and need a large amount of time to converge to a fair use of the channel. The negative loss events due to the big window and the fast increase in slow start are presented by Hoe in [34].

The proposed Transport-MAC cross-layer interaction is applicable locally to each RCST or PEP agent of the satellite network, mainly based on modifying TCP sending window $wnd[n]$ when a new connection enters and/or sudden increase/decrease of the sender queue size is experienced (n is the discrete value representing the $n-th$ packet to send). In particular, quick variations of the sender queue can be assumed linked to variation of the available capacity at MAC layer. The adjustments will be designed in order to maintain a limited use of the link buffers and avoid the above mentioned impairments for each TCP connection related to a given RCST.

6.5.3 Cross-layer in DVB-RCS systems: DAMA mechanisms

When using Adaptive Coding on the return channel of a DVB-RCS link, the coding information K_i of the terminal $i-th$ coming from lower layers, representing the number of ATM cells into a time-slot, can be part of the DBA algorithm itself in order to guarantee even more fairness in the allocation decision. According to the FEC codes actually in use, there is a difference in the amount of cells per slots.

To this aim, a cross-layer mechanism must be defined, so that the coding rate information detected at layer 1 is fed as input to the DBA algorithm. The K_i information is available at both the RCST and the NCC at MAC layer because it is known at the moment of burst transmission and reception. The mechanism for cross-layer messages exchange involving the upper layer (DBA) in this case has no impact in the satellite communication, since it can be performed out of the satellite channel bandwidth using the operating system inter-process communication (IPC) primitives. Furthermore, additional cross-layer information can be proposed to be part of the DBA algorithm, as discussed in [46], to increase DAMA algorithm flexibility. At present an additional static priority parameter p_i can be associated with the terminal $i-th$, to reflect a sort of pre-determined priority element for terminal traffic (different Service Level Agreement - SLA). This priority value p_i could also be adjusted during the session following several criteria, for instance proportional to K_i, or taking into account traffic classification (e.g. DiffServ classification at the edge routers of the satellite link) or DAMA slots request classes (higher priority for RBDC requests).

The proposed equations where a conditional maximum is searched are the following, with N the number of competing satellite terminals, x_i the amount of slots assigned, p_i the fixed priority factor, K_i the cross layer information of cells per slot, c_i' the amount of ATM cells requested, m_i the number of ATM cells already granted (CRA) assignment, M_i the pre-configured maximum capacity request in ATM cells allowed to the terminal based on its traffic contract/profile and S' is the difference in terms of timeslots (not ATM cells) between the number of slots available S considering the CRA assignment.

$$\underset{\{x_i\}}{maximize} \prod_{i=0}^{N} x_i^{p_i}$$

$$subject\,to: \begin{cases} \sum_{i=0}^{N} \left\lceil \frac{x_i}{K_i} \right\rceil \leq S' \\ 0 \leq x_i \leq c_i' \leq M_i \qquad \forall\, 0 < i < N \end{cases}$$

$$with: \begin{cases} S' = S - \sum_{i=0}^{N} \left\lceil \frac{m_i}{K_i} \right\rceil \\ c_i' = max\{(c_i - m_i), 0\} \end{cases}$$

This DBA, when the cross-layer extensions are disabled ($K_i = p_i = 1$), corresponds to the last one proposed in section 6.1.1.

Chapter 7

Performance evaluation through Simulation

An extensive simulation activity was carried out to evaluate performance improvements and the feasibility of the new approaches proposed. In addition feedback from the simulations performed helped to tune and refine the mechanisms identified during the analysis phase. Almost all simulations done was performed on the Network Simulator 2 (NS2) framework, which is a discrete-time (event based) simulator oriented to the analysis of traffic on packets networks.

The topology of the network is represented by a set of nodes and links, with different parameters characterizing each. Then, after defining the routing directives, the source and destination of packets can be selected. Simple applications are available, making use of either TCP or UDP packets delivery. If using TCP, several alternatives for congestion/flow control are available, such as Reno or Vegas. In the latest versions, by means of a special TCP container called Linux, all the congestion control algorithms available into the Linux kernel are also included: Cubic, Westwood+, Highspeed, etc. ([35]).

All the setup (topology, links characteristics, application/transport protocol details and data sources and sinks) is defined in a script file written in TCL language, than interpreted by NS2 which generates at the end of the execution the outputs of the simulation (usually in terms of text file or graphs). A graphical view of the simulations performed is also available by means of the tool NAM [36].

Examples of NS2 basic scripts, and methods to obtain and trace variables of interest from simulations are available in [37].

In the rest of this chapter, specific modifications and setup of NS2 are presented, followed by relevant results obtained.

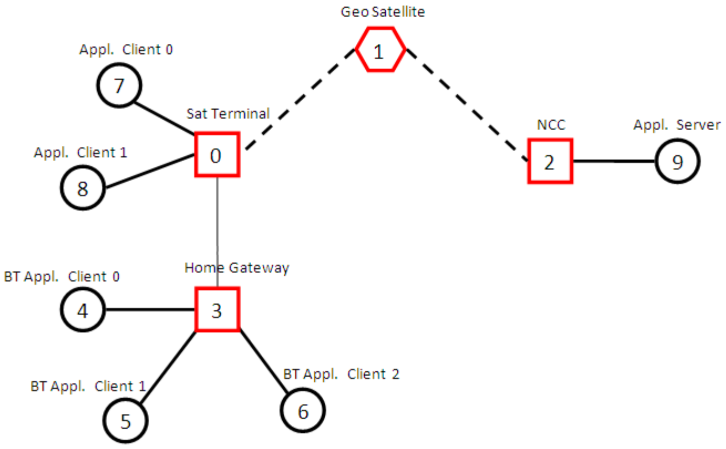


Figure 7.1: NAM network for telemedicine

7.1 Integrated network

An activity performed concerned the feasibility evaluation of running telemedicine applications in an hybrid network consisting of Bluetooth sensors connected to a wireless access point, connected in turn to a satellite terminal. A NS2 simulation script was used to define a scenario with multiple nodes of different kinds, each using the physical characteristics of the lower layers (e.g., Bluetooth) and using IP protocol for routing of packets. The output obtained with the NAM visualizer for NS2 is reported in figure 7.1.

The application control center is located at the NCC side of the network, while wireless clients (7 and 8) and Bluetooth clients (4, 5 and 6) are connected via a home gateway to the satellite terminal.

Tests performed confirmed the possibility of running typical applications (voice, data, video) using simultaneously different source and destination nodes, and the activity lead to a preliminary assessment of the dimensioning of the satellite bandwidth required for a specific telemedicine traffic aggregate.

7.2 DAMA extensions

In order to evaluate the differences of the proposed DBA algorithms and request strategies proposed, a DVB-RCS simulation including DAMA messages required by the standard has been implemented.

Figure 7.2 shows the overall simulator architecture highlighting the equivalent OSI stack of the simulated nodes (STs and gateway/NCC). Most of the DAMA functions have been included into a new “DamaMac” class (Mac/Sat/Dama), which both manages allocation signaling among STs and NCC (dotted arrows) and performs DAMA algorithms concerning the request computation (ST side requests) and the allocation decision (NCC side DBA). The different DBA algorithms proposed

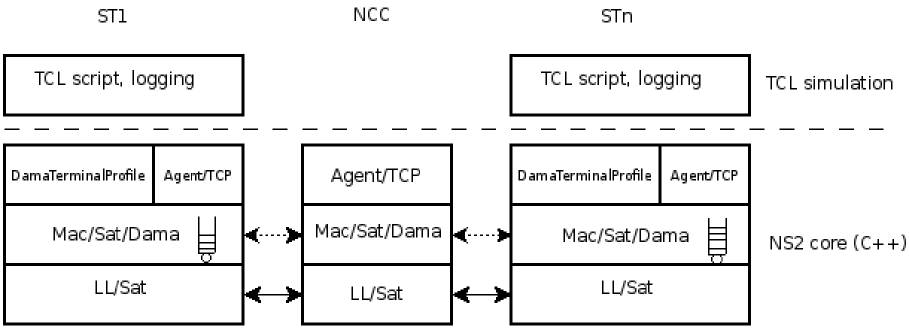


Figure 7.2: DVB-RCS DAMA NS2 based simulator architecture

have been coded in C++ and included in the NCC module.

The "DamaTerminalProfile" class manages the DAMA profile of each ST through a simplified login procedure. Such a class is also in charge to monitor the MAC queue status, periodically delivering allocation signaling and adjusting the ST sending rate accordingly. On top of DAMA, TCP agents are configured to operate as either connection source or sink. All the packets coming from STs are delivered to the DAMA agent that, on the basis of the assigned resources, schedules their transmission towards the GW.

A template TCL script has also been created to help set up simulations for DVB-RCS and DAMA support including:

- network configuration functions (i.e. number of STs, node latitude/longitude);
- definition of the DAMA profile parameters for each ST (i.e. CRA granted slots, maximum number of VBDC requests);
- functions for the generation of the traffic patterns required (connection directions, start and stop, data size);
- configuration of probes to log the state variables in order to generate post-simulation plots.

The DAMA message exchange managed by DAMA module is represented in figure 7.3. The time elapsing between sending a capacity request and the corresponding "capacity response" included in the TBTP constitutes the access delay and mainly includes the propagation round-trip delay and a processing delay at the NCC side. This process requires the management of several pending requests at all times through a First In First Out (FIFO) queue. Queue processing time is equal to the superframe duration, so that the "access delay" can be expressed in terms of number of superframes.

The difference in the use of the proposed DBA algorithms for the allocation of slots is represented in table 7.1, for two terminals competing for a maximum amount of 32 time slots per frame. CRA assignment is static and pre allocated to terminals without contention, so not considered for the DBA. Requests can be of two different

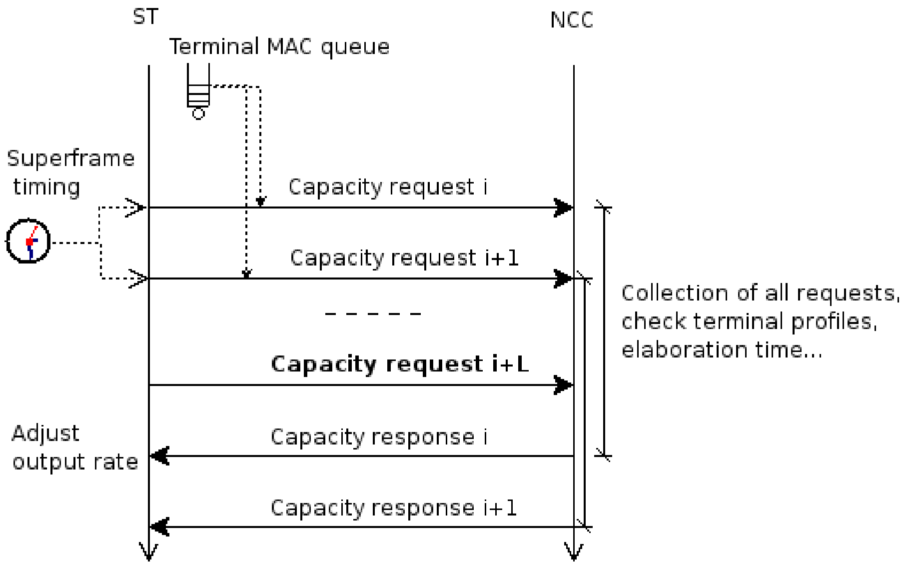


Figure 7.3: DAMA messages

Request RCST1	Request RCST2	DBA	Priority	Response to RCST1	Response to RCST2
RBDC = 32	VBDC = 32	Any	-	32	0
RBDC = 20	RBDC = 20	Proportional	-	16	16
RBDC = 20	RBDC = 30	Proportional	-	13	19
RBDC = 20	RBDC = 30	FCFS	-	20	12
RBDC = 20	RBDC = 30	Round Robin	-	16	16
RBDC = 20	RBDC = 30	Priority	p_1= 2.5 p_2= 1.0	18	14

Table 7.1: DAMA slots assignment for a total of 32 slots available

kinds: RBDC, based on rate necessary to terminals for transmission and (A)VBDC, based on the absolute instantaneous size of data to transmit. FCA is not considered because it is an allocation of spare slots even in case a terminal do not require it. RBDC has been considered of higher priority than VBDC requests.

The different effects of the different DBA algorithms can be used according to the setup required and scenario needs. In the case of proportional DBA, the following figures have been obtained for data transfer on the return link using different versions of TCP, showing the typical reactions to a DVB-RCS DAMA enabled system.

In figure 7.4 the round trip time is shown; in figure 7.5 the instantaneous throughput is shown, showing lower performances of TCP Vegas [43] fooled by the DAMA RTT induced variations; figure 7.6 shows the efficiency in terms of rate reached compared to the maximum available; finally in figure 7.7 several flows were running and

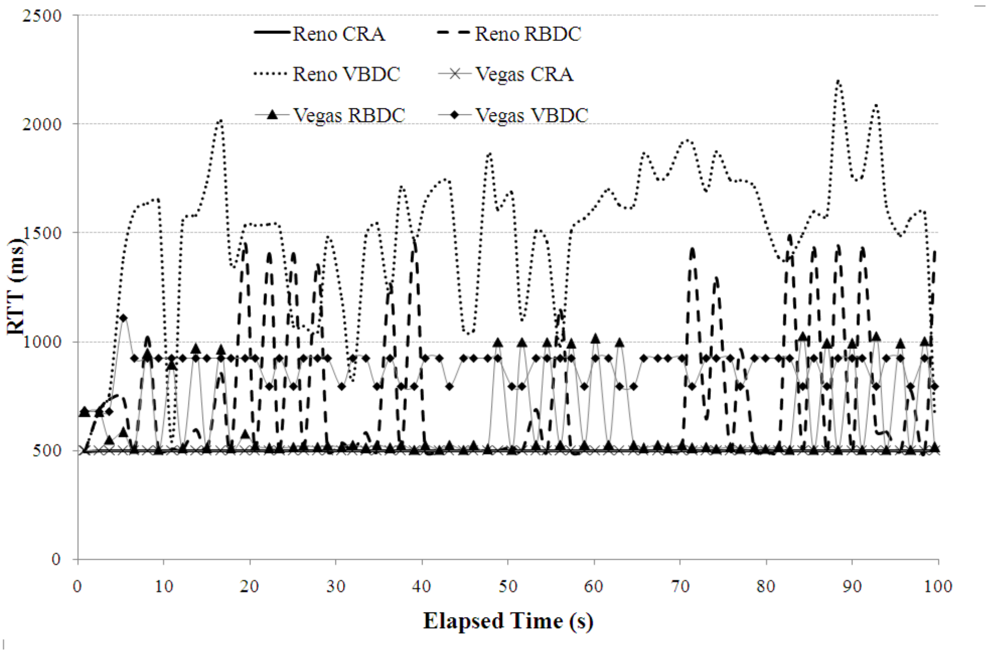


Figure 7.4: DVB-RCS DAMA: Different TCP round trip time

the Jain fairness index [45] is calculated to highlight the ability of different protocols using different DAMA schemes to share the bandwidth equally.

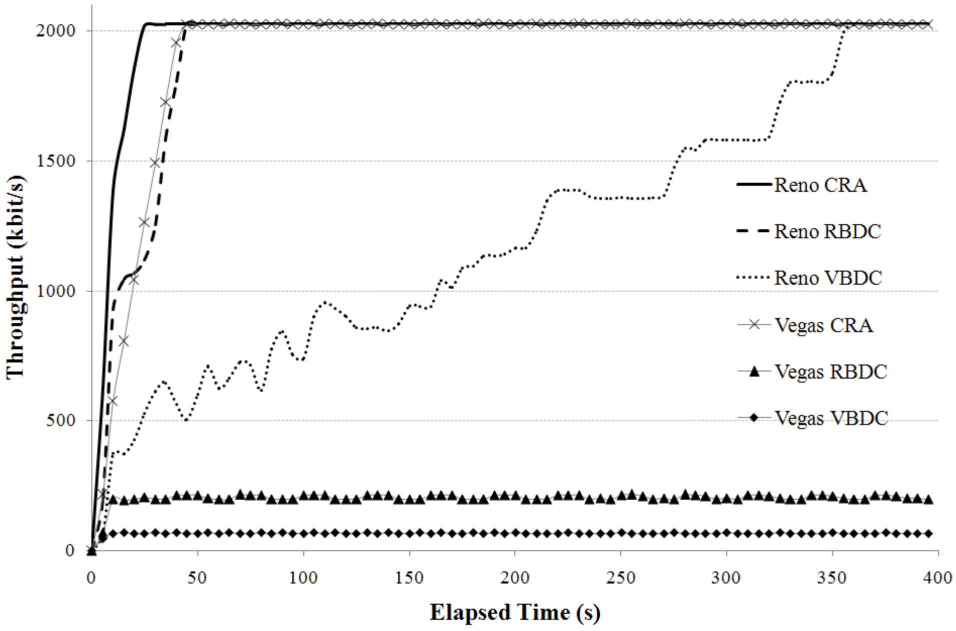


Figure 7.5: DVB-RCS DAMA: Throughput

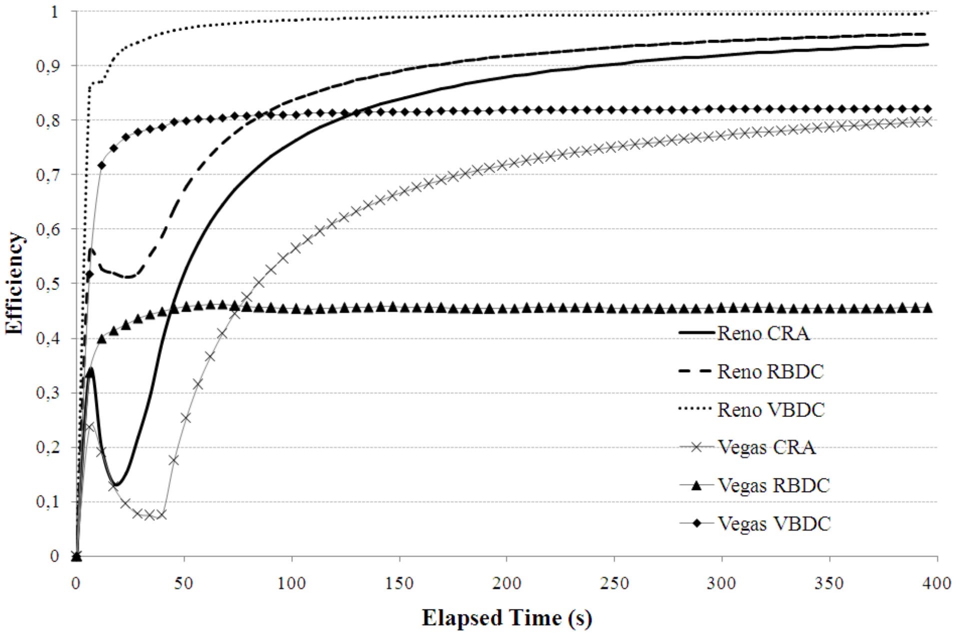


Figure 7.6: DVB-RCS DAMA: Channel utilization

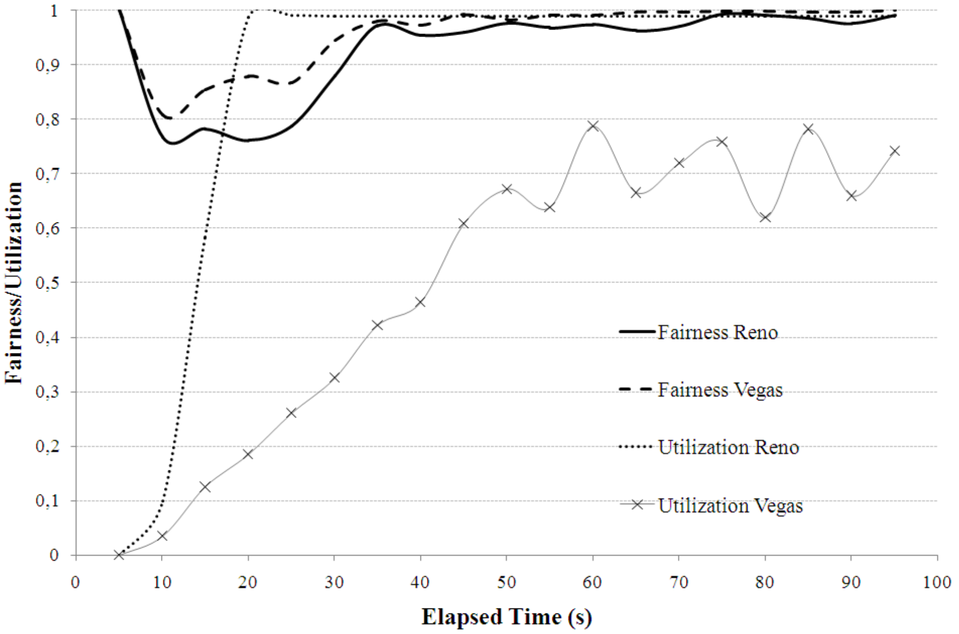


Figure 7.7: DVB-RCS DAMA: Fairness index (5 connections)

7.3 Handover implementation

A new feature was added to NS2, to include a double link between two nodes. The C++ core of NS2 is composed of several classes, of which "Connector" is the main one, in charge of defining the characteristics of a connection between two items. Several other classes inherited from Connector are needed to create a physical link between two nodes, represented in figure 7.8 as different boxes with the relative name.

A modification of the code, shown in figure 7.9 allowed to define a couple of paths for the packets, selectable by means of a switch introduced in the dynamic class (also inherited from Connector). In this way the two links defined can have different queues, separate log of tracing events (classes ending by "...T_") different drop models in order to resemble the two links involved in the handover, either soft or hard. In fact, if the command of switching sent to the Dynamic connector is soft, it just changes its target pointer to the new link, else if the command is hard, at the same time the path is changed, the old link target defaults to drop. In case of hard handover, it is also possible to define a certain time interval of complete black out for transmission (hard handover execution time).

A test of handover has been performed with this conditions:

- at time 0 s the first link is used, with a a throughput of 300 kbit/s and a propagation delay of 20 ms;

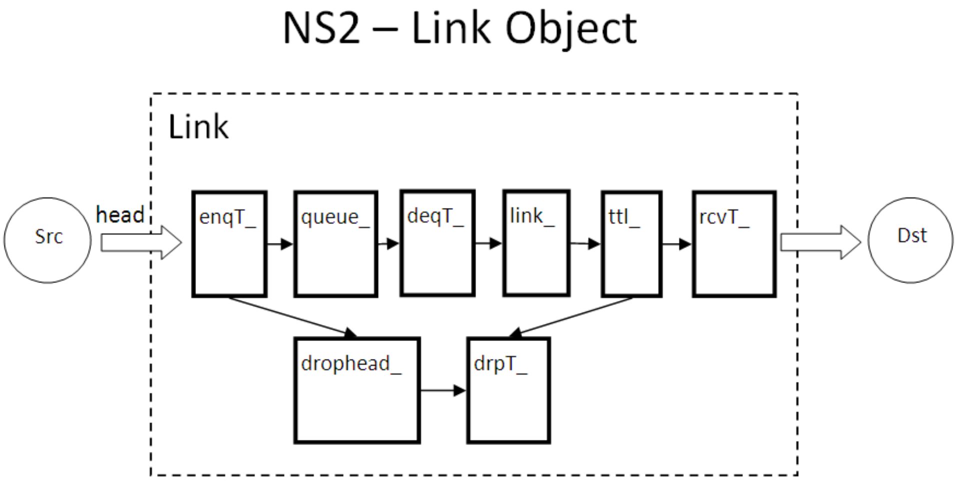


Figure 7.8: NS2 link structure

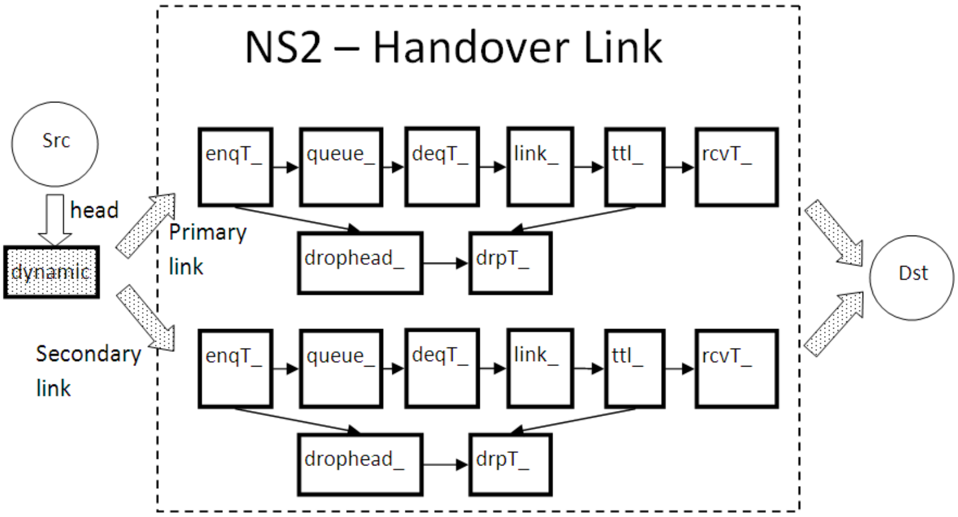


Figure 7.9: NS2 new handover link structure

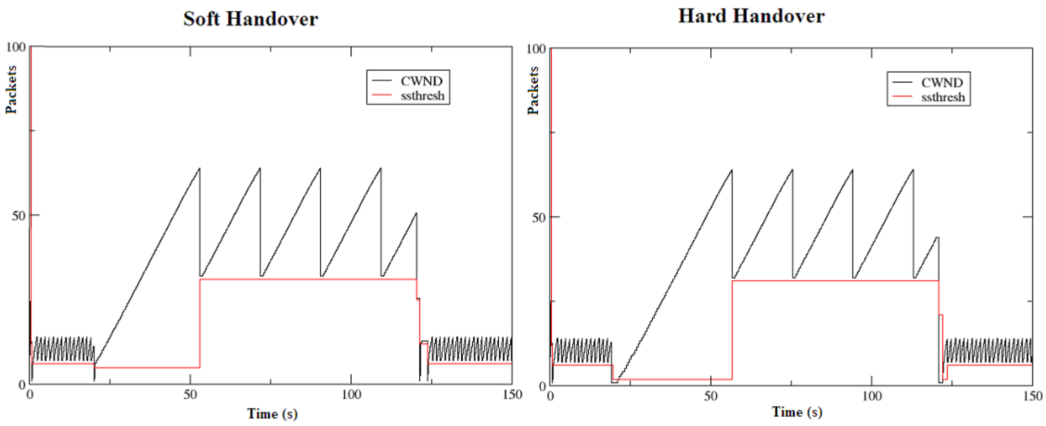


Figure 7.10: NS2 simulation of Soft versus Hard handover

- at time 20 s the handover towards the second link happens, with a a new throughput of 1 Mbit/s and a propagation delay of 520 ms;
- at time 120 s there is the handover to the first link;
- at time 150 s simulation ends.

Figure 7.10 represents the simulation results. Although standard TCP Reno has in general troubles during the transition, as illustrated in a previous chapter, soft handover has less impact on the connection at time 20 s making the connection to react to handover better (e.g., higher *ssthresh*, so less losses detected) while at time 120 s soft handover behaves worst, because the buffer is big (the second link has a high BDP) and reordering (with discarded packets) happen.

7.4 TCP for satellite environment

A dedicated activity of simulation has been performed for the validation of the TCP-Noordwijk congestion, flow control and error recovery algorithms for the use on DVB-RCS DAMA links. At the same time thanks to the simulation campaign, the algorithms parameters were adjusted to obtain a good balance of performance increase in terms of throughput, fairness of multiple flows and friendliness with coexistent standard TCP flows.

In figure 7.11 the throughput of a single connection over a 2 Mbit/s DVB-RCS return channel with resources assigned with DAMA RBDC assignment is shown and the great improvement on the startup phase when using TCP-N is evident.

In figure 7.12 the friendliness of TCP-N can be appreciated, since the overall capacity is equally distributed among all the connections active in the same time range.

Finally also friendliness against other protocols has been evaluated. In fact, figure 7.13 shows that a background TCP-Reno connection is only marginally affected by a competing TCP-N one. The Reno reference connections measures the

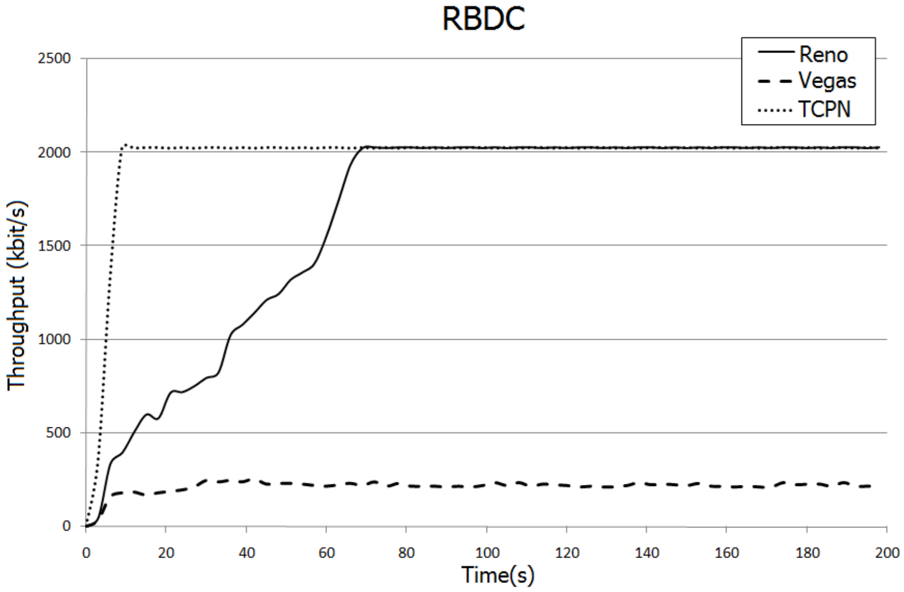


Figure 7.11: TCP-Noordwijk transfer rate on RBDC assignment

throughput achieved by a TCP connection alone using VBDC access, while in the second simulation between 150 and 350 s a TCP-N connection is started, minimally affecting the existing TCP- Reno.

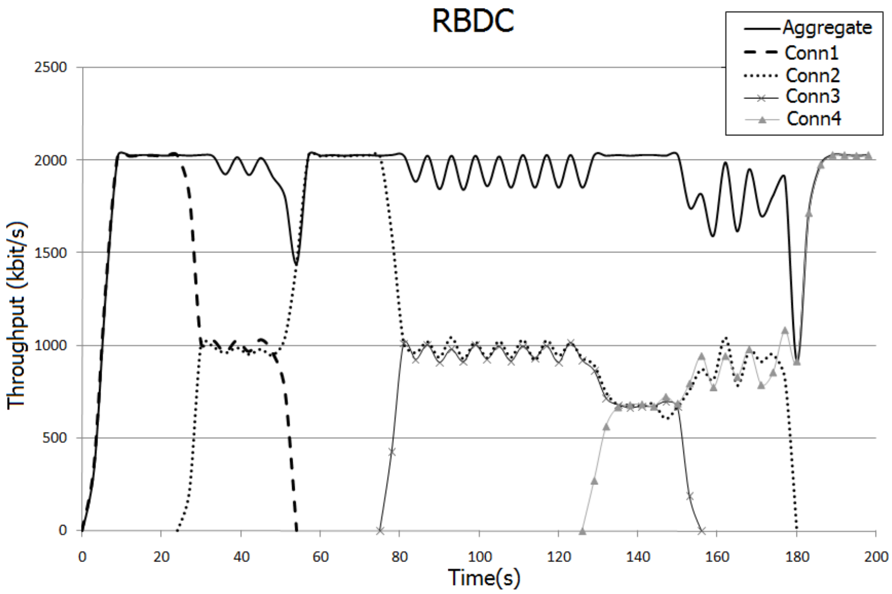


Figure 7.12: TCP-Noordwijk fairness on RBDC assignment

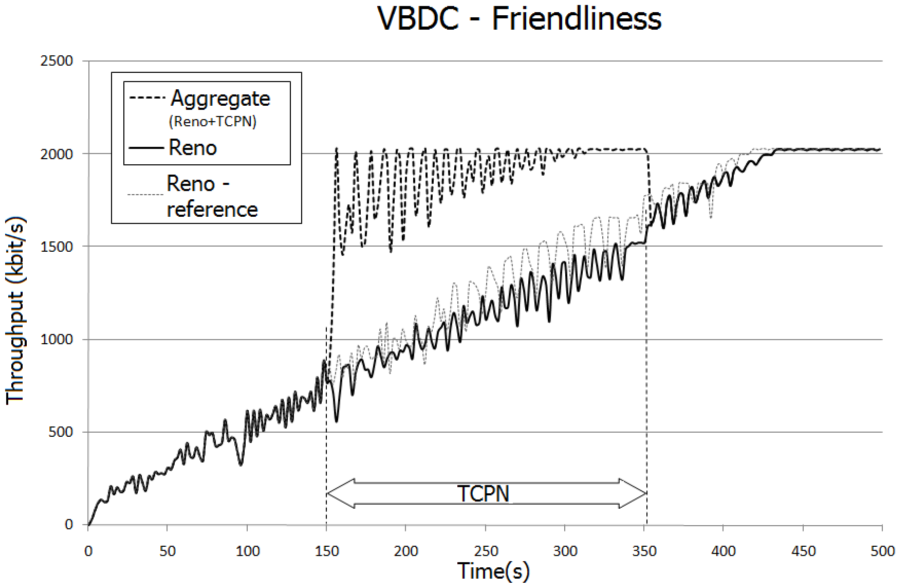


Figure 7.13: TCP-Noordwijk transfer rate on RBDC assignment

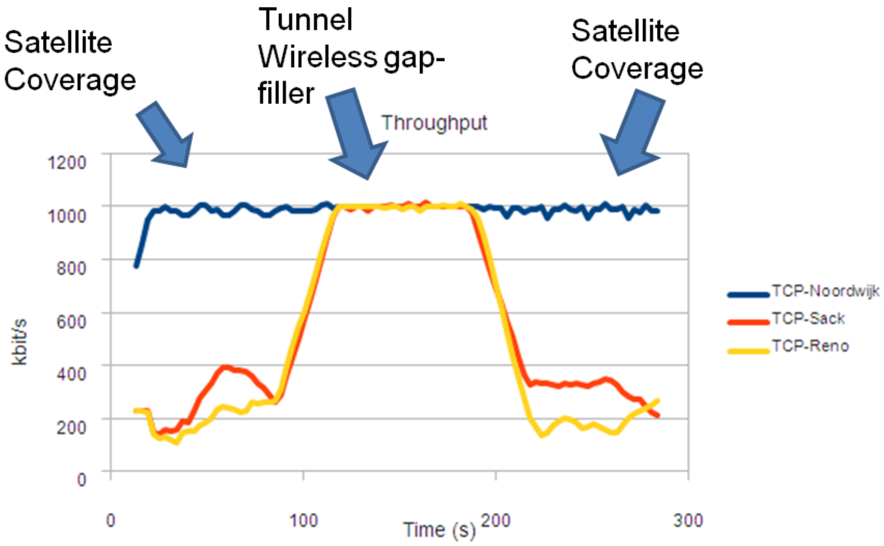


Figure 7.14: Use of TCP-N on high speed trains

7.4.1 TCP Noordwijk use on high speed trains

TCP-Noordwijk has been applied successfully in a mobile environment, where a satellite connection is established on board of an high-speed train by means of a DVB-RCS satellite connection. In particular a model for trellises induced link deterioration has been studied and modeled in NS2, as a semi-periodical loss of frames at layer 2. At the same time a hard handover mechanism has been hypothesized whenever the train enters into a tunnel (gap filler connectivity) or approaches a train station where a wireless connectivity coverage is available. Preliminary tests show that TCP-Noordwijk performs better than other standard end-to-end protocols, which heavily suffer of the periodic losses and only achieve a good throughput when using the terrestrial wireless link (e.g., inside tunnel). In figure 7.14 the results are shown, considering a handover inside the tunnel at 100 s, and a backoff to the satellite connectivity at 200 s. TCP-N demonstrated to be also smartly adaptable to very high change in BDP (from several tens of packets to few) and avoided buffer overflows.

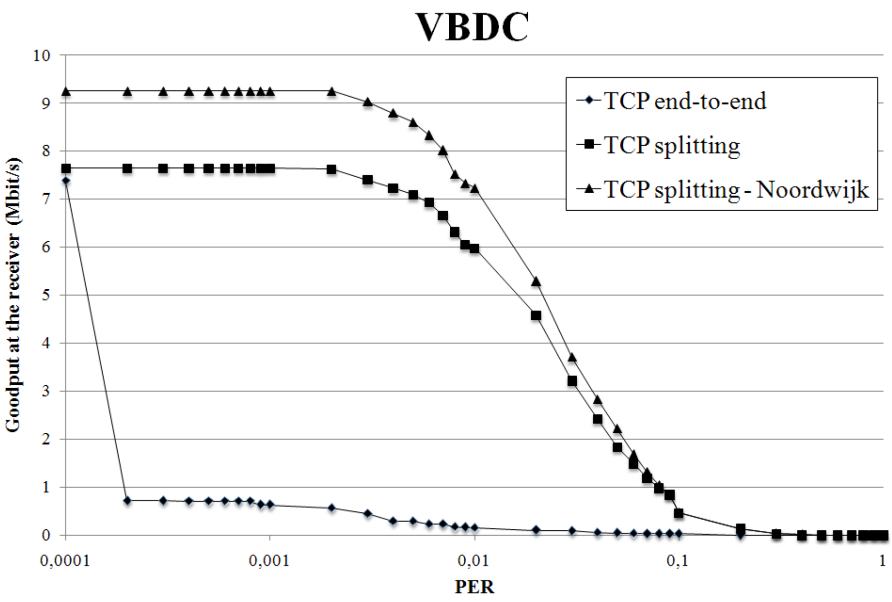


Figure 7.15: Split architecture for wireless access network.

7.5 Split architecture

In some cases it is useful to separate end-to-end connections in several segments, in relation to the variations of characteristics of the links used by means of proxies (PEPs). In the specific case of wireless access of users attached to a common satellite terminal, than traveling over a DVB-RCS link towards a gateway to a terrestrial network, the impact of a split architecture has been analyzed.

With the assumption that errors on transmission are due to degradation of BER on the wireless access network, while the satellite link is working in optimal *Quasi Error Free* conditions (as claimed in the DVB standard for data transmission over satellite), the following simulations have proved that using the split architecture is a big advantage in terms of throughput. In fact, the losses are recovered locally in the wireless access network, before that data is sent over satellite.

In particular, the use of an optimal TCP for DVB-RCS link, namely TCP-Noordwijk, boost even more the performance making transmission closer to the theoretical maximum reachable of 10 Mbit/s on the forward link (using DVB-S technology). In figure 7.15 the throughput of several TCP connections using VBDC resource allocation, which is the most critical due to its higher access delay, is shown in function of increasing Packet Error Rate.

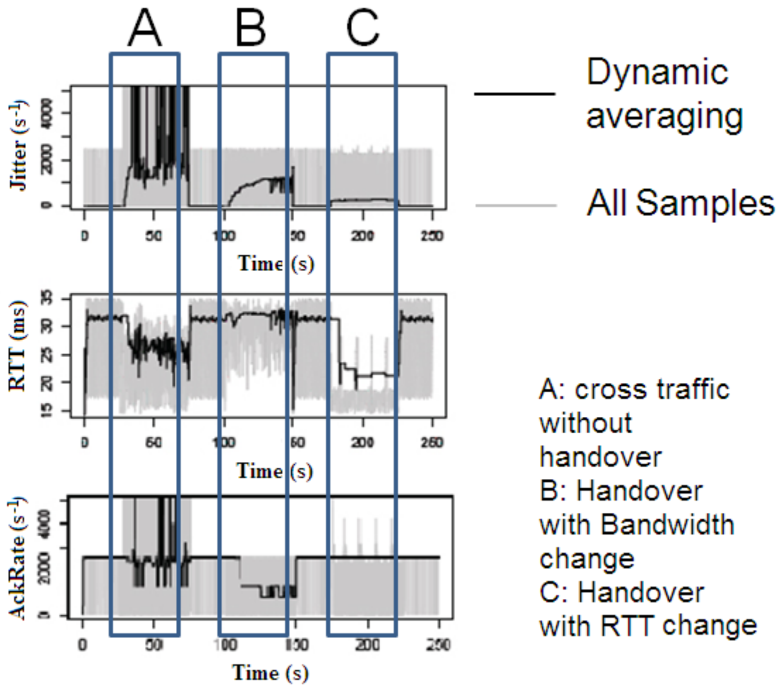


Figure 7.16: TCP ACKs analysis

7.6 TCP for capacity estimation

The detection of underlying network characteristics variations can be important to design more intelligent TCPs. Starting from TCP Probe concept [38], which is a TCP used to transfer data and simultaneously detect the bottleneck link capacity as described in section 6.3, some enhancements have been proposed and tested. In particular TCP Probe considers only samples when the perceived RTT is the minimum, hoping to avoid bad samples and producing wrong estimates. This approach introduces a limitation to the correct capacity estimation when the network is loaded, or the minimum RTT is variable over time, such as in a DVB-RCS system or during an handover event.

For this reason other values have been considered, such as jitter in packets arrival, variation in average RTT, etc. With an improved 3 dimensional model, by the observation of the flow of TCP acknowledgments received opportunely processed with adaptive filters, it was possible to detect the channel capacity estimate more accurately. In particular it was possible to discriminate events of handover, RTT changes and network congestion, better than TCP Probe. In figure 7.16 it is shown a first analysis obtained in NS2; further details are provided in [48].

Measures of the actual state of the channel capacity are also performed continuously by TCP-Noordwijk to determine the optimal transmission parameters, as illustrated in section 7.4.

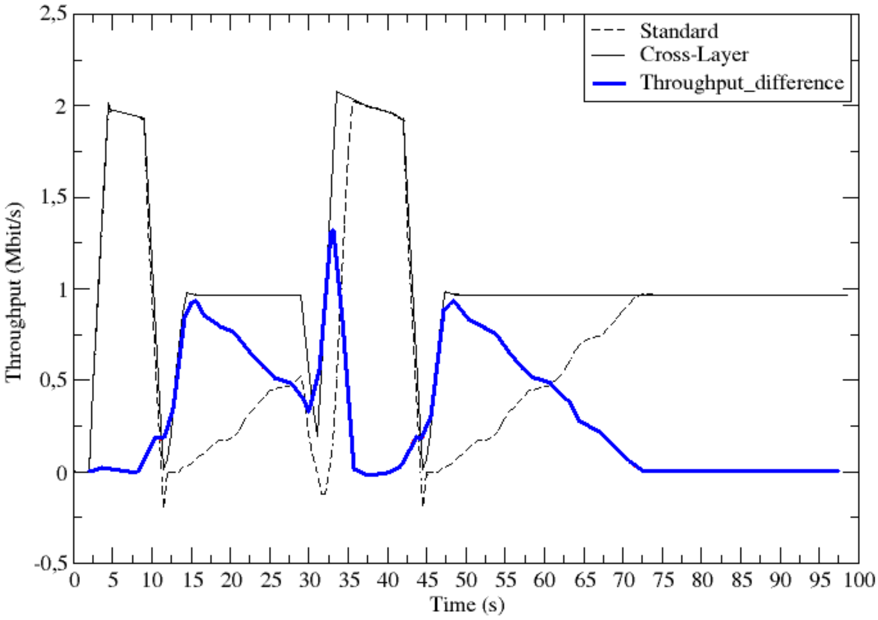


Figure 7.17: Handover TCP comparison

7.7 Cross-layer Handover management

The cross-layer framework proposed in section 6.5.1 introduces three cross-layer messages identified as (s_1), (s_2) and (s_3) for an efficient management of handover, with the goal of optimizing standard TCP connections transfer rate. The feasibility of the proposed approach was evaluated with several NS2 based simulations. Not only the TCP transfer obtained better results, but also the use of the queues and losses was reduced using this cross layer mechanism.

In particular figure 7.17 shows the throughput increment in function of time, when several handover events happens to and from terrestrial and satellite wireless networks, in comparison with a reference standard TCP Reno. The wireless terrestrial network has a maximum goodput of 2 Mbit/s, the satellite of 1 Mbit/s and handover events happen at time 10 s 30 s and 45 s.

A complete dissemination on this topic is available in [49].

		No cross layer		Cross layer	
Connection	Start time	Timeout at Syn/ACK	Packets TX	Timeout at Syn/ACK	Packets TX
1	0 s	0	1103	0	1054
2	5 s	1	331	0	931
3	10 s	0	847	0	878
4	10 s	0	707	0	881
5	20 s	0	707	0	815
6	30 s	0	707	0	764
7	40 s	0	507	0	733
TOTAL		**1**	**4909**	**0**	**6056 (+23.4%)**

Table 7.2: Loss events without and with cross layer modifications

7.8 Cross-layer TCP optimization in DVB-RCS environment

Cross-layer messages introduced into a dynamic bandwidth environment, typical of a DVB-RCS system with DAMA, can lead to performance increase in terms of increased fairness of the existing flows and reduction of queues usage. The results obtained by means of NS2 simulation, accurately tuned according to the analysis performed and proper cross layer messages can be summarized in table 7.2 and in figure 7.18. In the table the number of timeouts and the number of packets transmitted until the end of simulation at 300 s of several connections, which in sequence are started over the return link, is presented, with and without cross-layer modifications to TCP. The number of packets transmitted is better distributed (better fairness index) and the number of losses at the startup (dangerous for the triggering of long startup timeouts) are limited. Also the number of packets transmitted is better, because no congestion at the bottleneck queue of the link happens. In fact in the figure, the queue utilization (and consequently the delay perceived by a new connection established which can affect the overall system fairness) is shown, with a clear lowering of its maximum size.

7.9 Cross-layer adaptations for DVB-RCS DAMA

To validate the cross layer extensions for a DBA algorithm proposed in 6.5.3, a simulation activity was performed in NS2 with realistic traffic and compared with previous results obtained with the Matlab based scripts described in [50]. In the simulations the value K_i represents the number of ATM cells that are currently

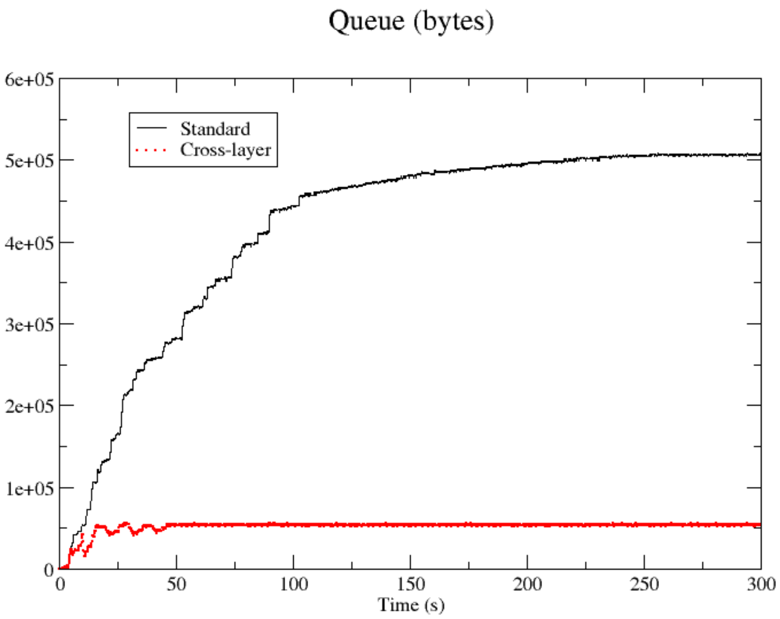


Figure 7.18: Cross Layer TCP queue usage

fitting in one time slot of the MF-TDMA structure of the DVB-RCS system and depends on the required FEC rate which can be adjusted during the simulation according to the link BER. The FEC used are the ones commonly available for DVB-S transmission. The K_i information is cross-layered to the DBA algorithm in order to determine the optimal allocation of resources, not penalizing too much the terminals with higher FEC rates.

The main advantage of the simulation is that the series of requests is reflecting real needs of simulated connections rather than fixed arbitrary numbers. In this way also the system stability and interaction among DBA and TCP congestion control mechanisms can be evaluated.

The results are compatible with the specification and tests in realistic simulated environments with multiple RCSTs and different priorities/channel conditions are undergoing.

The C++ code implementing this DBA algorithms, together with the other discussed in 6.1.1, is available online as patch to NS2 at

`http://www.tlcsat.uniroma2.it/DAMA/`.

Chapter 8

Performance evaluation through Emulation

An emulation platform for satellite systems allows to test both real applications and TCP/IP protocols in a communication scenario similar to the real one, avoiding costs of expensive satellite capacity and terminals/hardware leases and introducing invaluable flexibility. It is however necessary that the emulated environment matches the real system characteristics as close as possible. The emulation tool developed and enhanced during the execution of the thesis works is called Satellite Network Emulation Platform (SNEP), specifically targeted to the emulation of DVB-RCS networks.

The emulation strength resides mainly in the capability of using real applications and assessing performance, from objective and subjective point of view.

The hardware of the SNEP emulator is composed of Linux machines with ordinary hardware using specific configuration and additional software modules.

8.1 Architecture of the emulator: SNEP

The SNEP is composed of a network of five computers, each one performing the functionality of one or more elements of a real DVB-RCS network. Figure 8.1 shows the relation between each emulator machine and the elements of a real system.

At first there is an application Server providing commonly required Internet services (e.g., FTP, HTTP, etc.). Both satellite gateway and the Network Control Center (NCC) functionalities are reproduced in the same machine. Another machine introduces satellite physical constraints in terms of both physical delay and bandwidth. Furthermore, different virtual machines over the same physical machine (ratio $1 : n$) play the role of multiple RCSTs or UTs. Host virtualization offers flexibility and scalability at the cost to have highly performing hardware (CPUs, RAM, etc.) and managing processes related to emulation operations with a high scheduling priority. The routing tables of each machine of the system (both real and

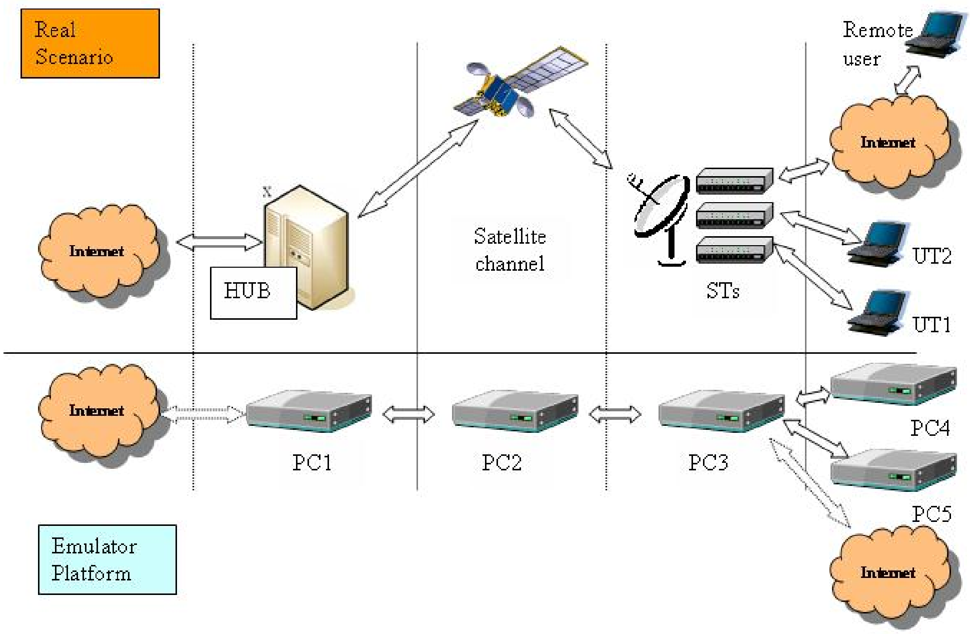


Figure 8.1: SNEP matching with a real system

virtual machines) has been done so that each RCST is reachable from the Server and vice-versa, with RCST that must cross the satellite segments twice (double hop) in case of direct communication among them.

The main contributions to the platform development were a set of scripts for the setup of satellite emulation PC (enforcement of delay, error, etc.), the introduction of a complete support to DAMA by means of a centralized NCC software and multiple client software on each terminal in charge of issuing requests and regulating the output on a virtual TDMA frame according to the TBTP generated by NCC. The algorithms used are carefully ported from NS2, with a direct translation of the algorithms from C++ language of NS2 to ANSI C. Furthermore, a set of automated scripts for execution of applications (iperf, ftp and ping) at different points of the emulator, with an automated generation of results in terms of performances graphs, packets loss, etc. has been designed.

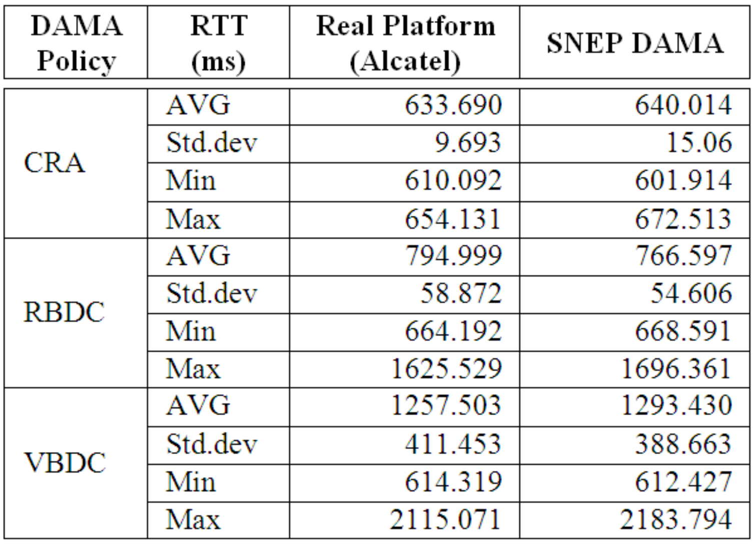
8.2 DAMA extension

The complete signaling for the DAMA allocation mechanisms has been designed, using dedicated channels, with a multi-client server application. The following procedures listed in table 8.1 have been reproduced by specific UDP messages, to resemble real messages defined by the standard with some simplification.

The system was tested and produced the expected results, compared to the NS2 simulations and real systems. For instance the results from a single RCST performing several pings at 1 Hz with different DAMA profiles is shown in table 8.2 in comparison with a real system.

Name	Description
Login Request	A terminal wish to enter the system, communicates its unique identifier, its traffic profile (MaxRBDC, MaxVBDC, CRA), its geographical position
Login Response	The NCC verify that the terminals eligible of the profile announced, reserve its CRA slots and send the login success/failure state
Course Synchronization Request	A logged terminal send a reference burst to ask for a grain tuned synchronization
Course Synchronization Response	The NCC reports its absolute internal clock for the superframe reference
Fine Synchronization Request	A logged terminal sends a message using the TDMA reference time to determine its offset in the time slot structure
Fine Synchronization Response	The NCC reports the offset of the received message in comparison with the TDMA frame and the other terminal logged into the system. In this way it is possible to synchronize terminals with small differences in RTT
DAMA capacity Request	Issued periodically (at maximum each superframe) to request the resources needed to transmit (RBDC or (A)VBDC)
DAMA capacity Response (TBTP)	The response to the requests performing the DBA algorithm
Logout Request	The terminal leaves the network
Logout Response	NCC frees resources assigned the leaving RCST and sends an OK message

Table 8.1: SNEP Emulator DAMA messages



DAMA Policy	**RTT (ms)**	**Real Platform (Alcatel)**	**SNEP DAMA**
CRA	AVG	633.690	640.014
	Std.dev	9.693	15.06
	Min	610.092	601.914
	Max	654.131	672.513
RBDC	AVG	794.999	766.597
	Std.dev	58.872	54.606
	Min	664.192	668.591
	Max	1625.529	1696.361
VBDC	AVG	1257.503	1293.430
	Std.dev	411.453	388.663
	Min	614.319	612.427
	Max	2115.071	2183.794

Table 8.2: Ping statistics on SNEP

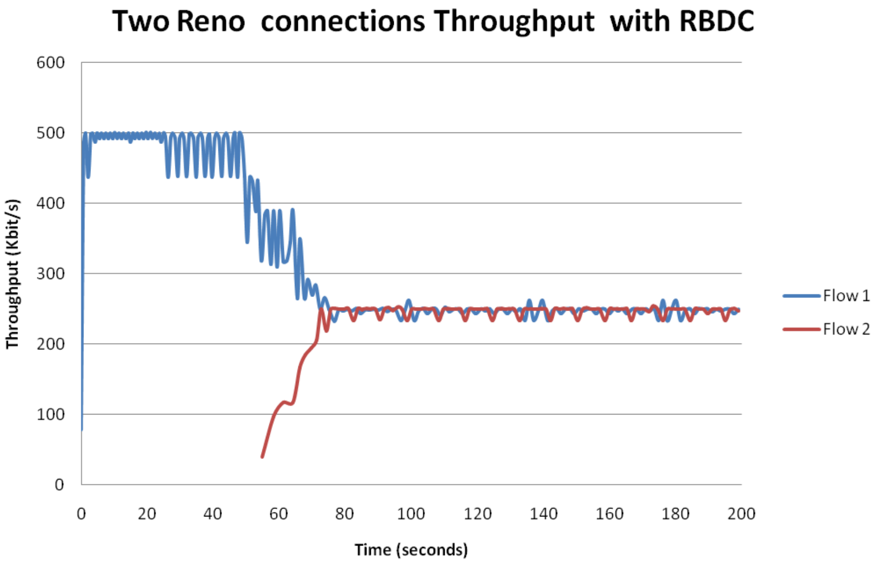


Figure 8.2: Competing TCP Reno flows over RBDC: throughput over time

Finally an example of multi terminal resource contention is presented in figure 8.2, with two RCSTs competing for the resources available on the return link with the same MaxRBDC = 32, with a TCP connection starting at 0 s from RCST1 and other at about 4 s (manual launching a FTP upload) from RCST2. It is visible how, after a convergence time interval, both terminals have the same bandwidth assigned. The DBA used is the number 3 described in section 6.1.1 (proportional).

8.3 TCP for satellite environment

A reference implementation of the TCP-Noordwijk for satellite environment has also been ported to Linux from NS2, as additional protocol inside an application for PEPs by MITRE. In this frame the validation of the results on the SNEP platform compared to the results obtained in NS2 was performed. Figure 8.3 shows the comparison in terms of throughput reached by a single TCP connection when using RBDC access strategy (1 CRA and 31 RBDC slots out of 32) in both simulated scenario and on the SNEP emulation platform (Linux), with a very close trend in both plots.

8.4 Telemedicine service testing

Telediagnosis in the field of neurophysiology has been studied, as illustrated in figure 8.4, with the traffic related to the telemedicine service flowing through the SNEP emulation platform. The satellite terminal was installed by the patient premises, with a technician executing an ElecroEncephaloGraphy (EEG) with an TCP/IP device by EBNeuro, supported by voice and video transfer. At the other edge, a

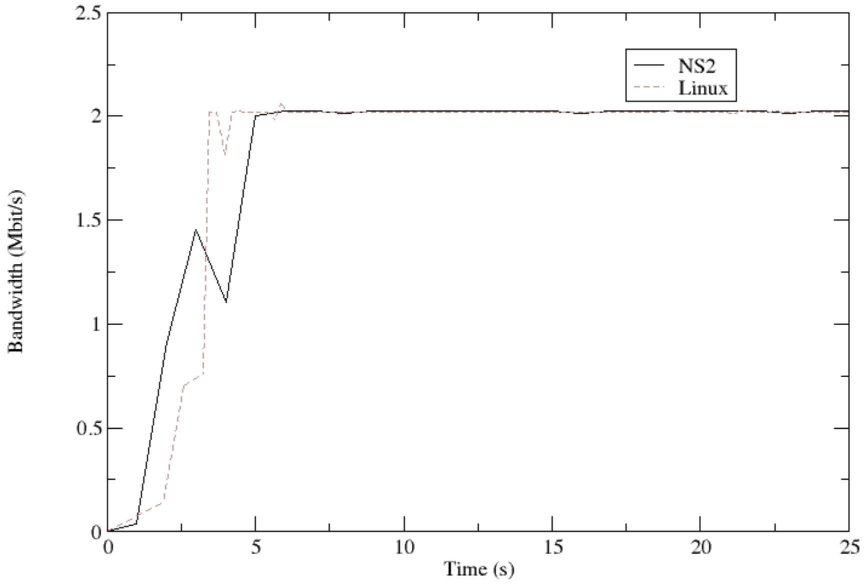


Figure 8.3: SNEP and NS2 comparison

specialized doctor was in charge of directing the technician in the acquisition of data, and than performed the diagnosis.

To correctly handle the traffic aggregation of the telemedicine services in relation to the limited resources available, the SNEP platform was enhanced with a support for application prioritization at IP layer. In this way for instance the EEG data is transferred in realtime with higher priority compared to other traffic.

The exam was performed with the assistance of a doctor from the Neuroscience department of "Politecnico Tor Vergata", as visible in fig 8.5, to evaluate the quality of the exam perceived by the doctor and to verify the capability of the applications to bear the satellite latency, respecting bandwidth constraint of a typical DVB-RCS system.

The test was successful[1] and also let the telemedicine application and devices to be adjusted for the satellite communication (for instance adjusting the timeouts), so that their use on a satellite system can be possible. The integration at IP level of custom medical devices has also been validated.

The overall average bitrate observed during a session of 40 minutes was about 190 kbit/s, compatible with the capacity of a DVB-RCS service return link and including real time delivery of EEG data, bi-directional voice via a custom Asterisk PBX setup and video via the commercial application Skype.

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[1] Special thanks to Prof. L. Bianchi, to the technician M. Abbafati and the "volunteer" for scalp electrodes placement F. Belli

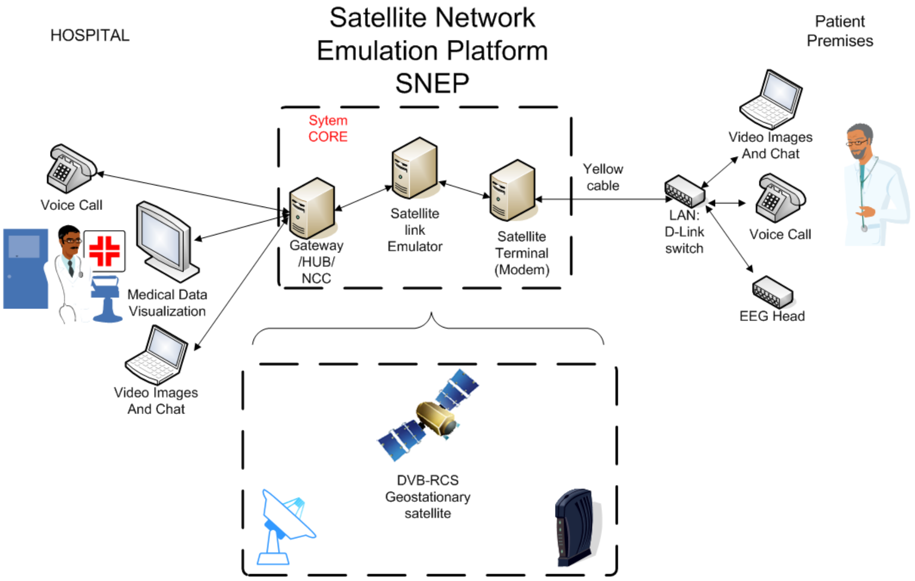

Figure 8.4: Telemedicine setup

Figure 8.5: Telemedicine real test with medical equipment

	No traffic		Cross traffic		Cross Traffic + QoS	
Parameter	Return link	Forward link	Return link	Forward link	Return link	Forward link
Min. (ms)	0	0	0.1	0	0.6	0
Average (ms)	30	0.02	32	0.3	31	0.02
Max. (ms)	74	5	239	201	98	28
Std. Dev.	21	0.4	28	1.5	21	1.8
Quality index	4		2		4	

Table 8.3: VoIP calls evaluation

8.5 VoIP Application performance evaluation

The Asterisk PBX software suite [39] has been installed in a dedicated server, connected in a LAN with the NCC/Gateway of the SNEP. In this way it was possible to establish voice calls using the SIP standard [40] among satellite terminals and between a satellite terminal and another user connected to the gateway with a terrestrial network. Several tests with different configurations of DAMA resource allocations and codecs for the real-time transfer of voice packets have been performed. Specifically a command-line software phone was installed at the NCC and on the RCST called SJphone, able to reproduce a reference PCM *.wav* audio file at one side of the call and record (together with play) the audio file received at the other side.

Several calls were executed, analyzing the statistics measurements of packets sent and received, together with an evaluation of the overall voice quality, with a scale from 1 to 5 (1 being the lowest quality). It was noticed that as soon as interfering traffic was running together with the voice call, serious quality degradation of the voice happened.

For this reason an additional full support for QoS has been implemented at each of the RCST client component of DAMA. It is a preliminary version following the SatLabs specifications, with also a support for dynamic QoS by means of C2P signaling. The work is still open and part of the ongoing EMERSAT project [41].

The first results obtained are summarized in table 8.3 showing the minimum, average, maximum and standard deviation of the measurements of inter-arrival time of voice packets, with an indication of the subjective call quality indication.

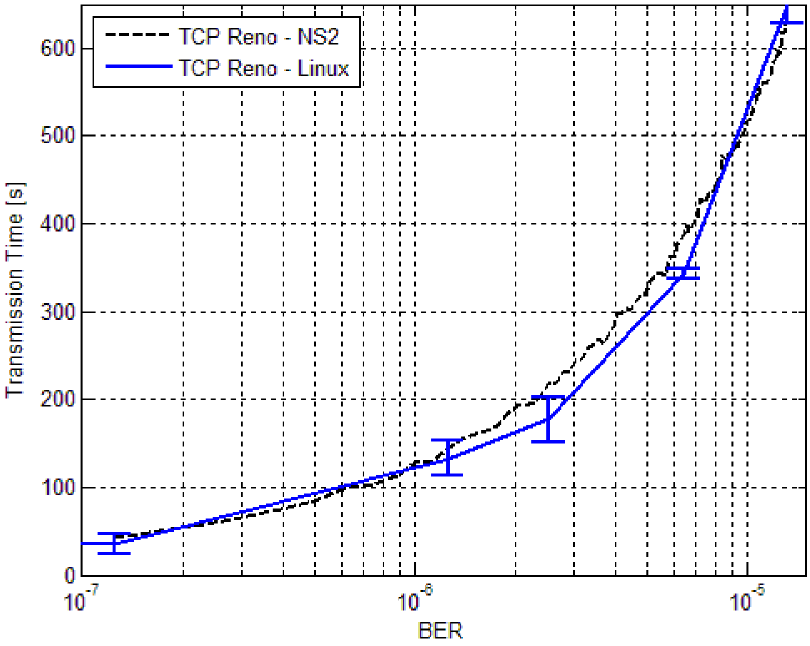

Figure 8.6: TCP Reno transfer time and BER

8.6 FTP performance evaluation

Performance of TCP based data transfer over a satellite segment has been studied on the SNEP platform. In particular, a focus on impact of different BER on the satellite segment of an hybrid network for an FTP data transfer was considered. The time necessary to transfer a file and the efficiency of the transfer has been analyzed, including in the evaluation also a comparison with results obtained using a similar setup in NS2. The satellite segment consisted in a 2 Mbit/s DVB-S forward link and a 384 kbit/s RCS return link (for the ACK traffic), with a FEC of 1/2 and IP packet size of 1460 bytes.

The result was that TCP is heavily affected by high loss rate, making the transmission inefficient and time consuming even at few packets lost per second. In figure 8.6 the time necessary for the transfer of a 4.8 Mbytes file is represented as a function of time: the Linux line represents the values obtained running the client and server applications at the edges of the SNEP several times, providing the average value obtained and the confidence interval; the NS2 line represents an average of several hundreds tests for each BER value, each time varying the seed of the random error generator. In figure 8.7 the efficiency of transfer for different sized files is represented (only results form Linux SNEP platform), as the ratio of bytes to transfer divided by bytes actually transferred. The ideal case represents the use of an ideal protocol without headers, acknowledgments and with perfect retransmissions.

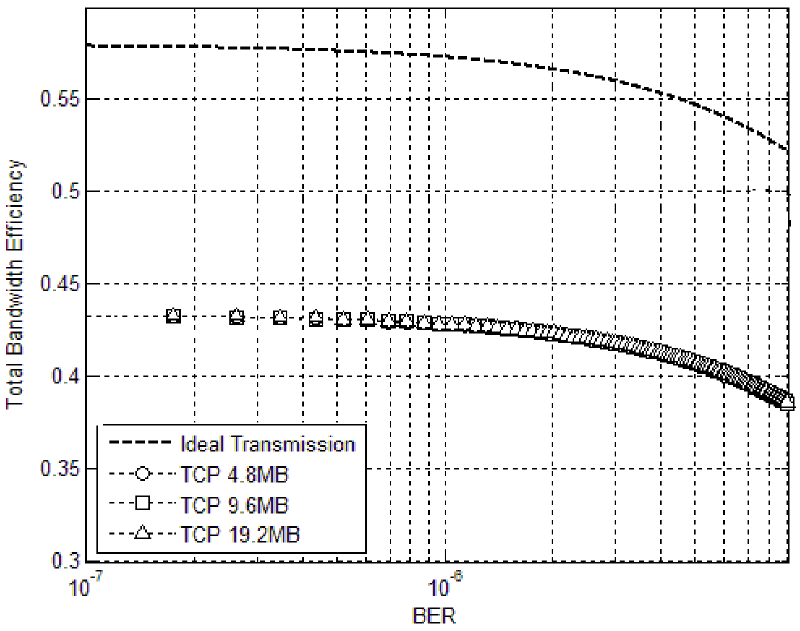

Figure 8.7: TCP Reno transfer time and BER

In this context, a proposal for UDP based transfer using the Fountain Coding [21] was tested, running a Linux implementation of encoder and decoder, obtaining a marginal improvement in system efficiency and a reduction in the transfer times, as accurately described in [51].

Chapter 9

Test on Real Platform

The test campaign for the ESA TOP project was in part executed at ESRIN premises, using the Alcatel 9720 Mini HUB V1.0 with a EMS RCST. The setup of tests included the MITRE PEPs extended with TCP-Noordwijk. In this frame the test execution and post processing of data was set up. Among the several results obtained, the throughput comparison using split architecture and default TCP (Reno), TCP Vegas and TCP-Noordwijk respectively is presented in figure 9.1. The time needed to reach the maximum available rate is less than half for TCP-N, although competing with an high aggressiveness of the other TCP versions due to the initial setup of *ssthresh* in Linux to infinite [44].

In particular, it is confirmed that TCP Vegas in presence of high and variable delays typical of the VBDC case does not reach the maximum throughput. Nevertheless the throughput measured on the real platform in this case is higher than the one measured by means of NS2 simulations in the same conditions, so that further investigations are ongoing.

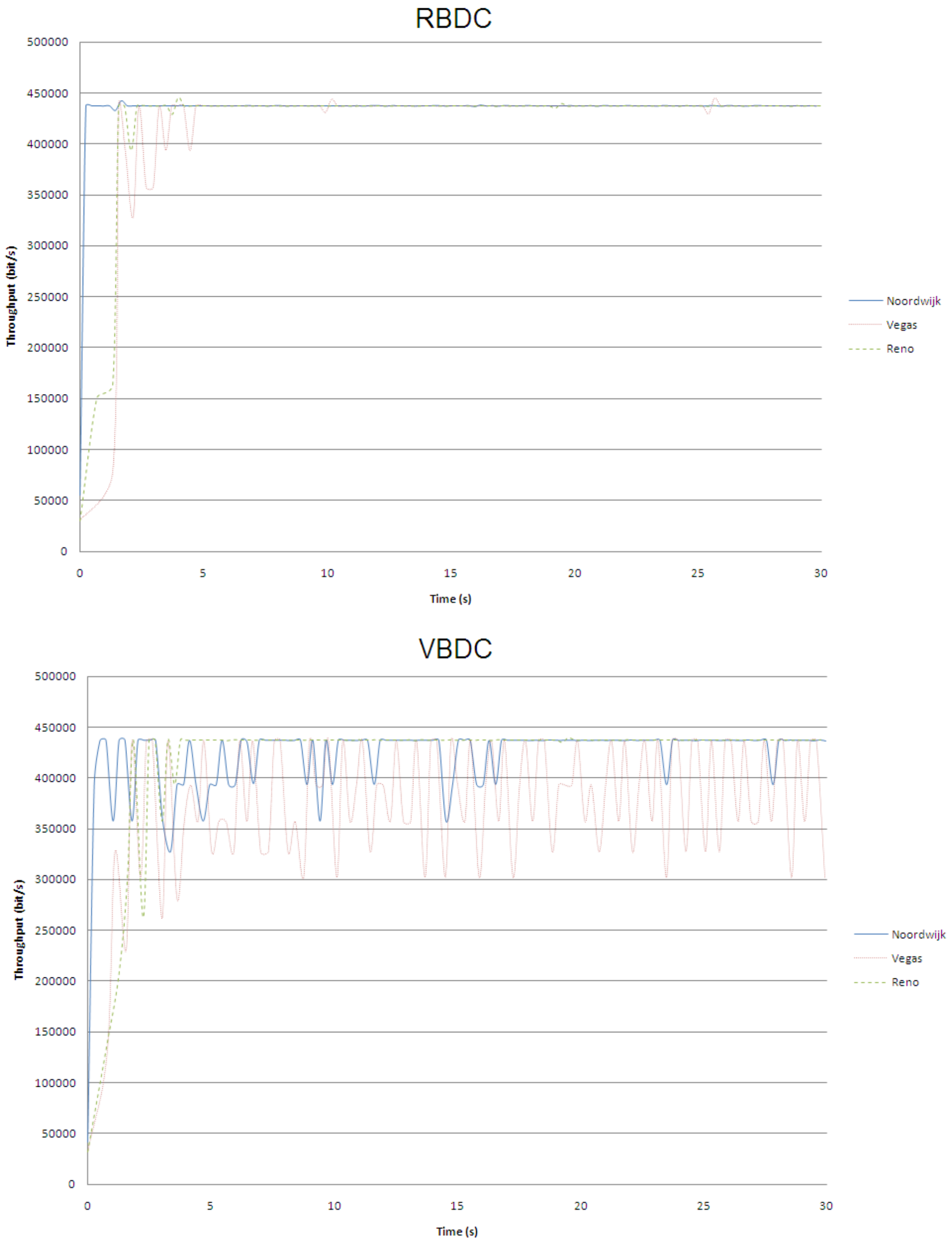

Figure 9.1: TCP comparison of real DVB-RCS system

Chapter 10

Conclusion

The activities performed during the three years of Ph.D. are related to heterogeneous networks composed of satellite and terrestrial segments. The critical aspects of integration have been analyzed through a phase of study and a phase of analysis of requirements, leading to the proposal for new solutions which were tuned and validated performing many simulation, emulation and real platforms testing.

Special attention has been put on DVB-RCS systems with particular regard to resource allocation, end-to-end transport protocol optimizations, cross-layer mechanisms, handover and QoS. In the thesis several outcomes are presented, showing the feasibility and performance of the hybrid network studied for different applications and scenarios of interest, and offering in many cases improvements compared to present technologies usually inherited by terrestrial networks (and thus not optimized for hybrid networks).

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