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Contribution to Broadband Metropolitan Networking Infrastructures Utilising Digital Terrestrial Television (DVB-T)

A Thesis Submitted in Partial Fulfilment of the Requirements for the Degree of Doctor of Philosophy (Ph.D.)

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This Ph.D. Thesis is dedicated to my parents

Η Συμβουλευτική Επιτροπή της Διδακτορικής Διατριβής του Μαστοράκη Γεωργίου:

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ΠΕΡΙΛΗΨΗ

Η Διδακτορική Διατριβή αυτή συμβάλει στις ευρυζωνικές μητροπολιτικές δικτυακές υποδομές μέσω της σχεδίασης, υλοποίησης και αξιολόγησης μίας πρωτότυπης αρχιτεκτονικής, η οποία αξιοποιεί την ψηφιακή τηλεόραση ως ένα οπισθοζευκτικό δίκτυο (backhaul/middle–mile), δίνοντας τη δυνατότητα πρόσβασης των χρηστών/πολιτών σε μονόδρομες, διαδραστικές και κατά-παραγγελία ευρυζωνικές υπηρεσίες, με εγγυημένη ποιότητα-υπηρεσίας (QoS). Βασιζόμενη σε θέματα σύγκλισης τεχνολογιών και υπηρεσιών, καθώς στην αξιοποίηση του ενδογενούς χαρακτηριστικού της επίγειας ψηφιακής τηλεόρασης να συνδυάζει ετερογενή δεδομένα στο ίδιο ρεύμα μεταφοράς (MPEG-2 TS), η Διδακτορική Διατριβή προτείνει τη χρήση του προτύπου DVB-T σε αναγεννητικές διαρθρώσεις (regenerative configuration) για την υλοποίηση ενός κοινού ευρυζωνικού δικτυακού κορμού, ο οποίος είναι διαθέσιμος σε όλη την περιοχή ευρυεκπομπής. Αυτός ο δικτυακός κορμός δρα ως οπισθοζευκτικό δίκτυο, το οποίο επεκτείνει τον κεντρικό κορμό του μητροπολιτικού δικτύου, έτσι ώστε αυτός να είναι προσπελάσιμος από κάθε χρήστη μέσα στο αποτύπωμα κάλυψης της ευρυεκπομπής (broadcasting footprint). Η πρόσβαση των χρηστών σε αυτόν τον κοινό δικτυακό κορμό επιτυγχάνεται μέσω ενδιάμεσων κόμβων (Cell Main Nodes - CMN), οι οποίοι κάνουν χρήση ενσύρματων ή ασύρματων τεχνολογιών επικοινωνίας. Για την αποδοτική λειτουργία του δικτύου καθώς και για την παροχή δεδομένων με εγγυημένη ποιότητα-υπηρεσίας, η Διατριβή αυτή παρουσιάζει τη μελέτη, σχεδίαση, υλοποίηση και ενσωμάτωση, σε μία αποκεντρωμένη δικτυακή αρχιτεκτονική, ενός μηχανισμού διαφοροποίησης των υπηρεσιών (services differentiation), με βάση τις απαιτήσεις του κάθε χρήστη και τα χαρακτηριστικά της κάθε υπηρεσίας.

Σύμφωνα με την προτεινόμενη γενική αρχιτεκτονική και το σχεδιασμό του συστήματος, η Διατριβή αναλύει και παρουσιάζει την υλοποίηση μίας πρωτότυπης μητροπολιτικής υποδομής, η οποία αποτελείται από μία πλατφόρμα DVB-T σε αναγεννητική διάρθρωση και δύο ενδιάμεσους κόμβους πρόσβασης, ο ένας εκ' των οποίων βρίσκεται (γεωγραφικά) κοντά στον κεντρικό κορμό του μητροπολιτικού δικτύου (urban-area) ενώ ο άλλος σε μία απομακρυσμένη περιοχή (rural-area). Αυτή η πρότυπη μητροπολιτική υποδομή αποτέλεσε την πειραματική διάταξη (testbed) πάνω στην οποία σχεδιάστηκαν και πραγματοποιήθηκαν μία σειρά πειραμάτων και μετρήσεων αξιολόγησης των δικτυακών επιδόσεων του συστήματος. Τα αποτελέσματα αυτών των μετρήσεων επιβεβαίωσαν αρχικά την ικανότητα της προτεινόμενης αρχιτεκτονικής στη δημιουργία μητροπολιτικών ευρυζωνικών δικτύων με δυνατότητες παροχής εγγυημένης ποιότητας-υπηρεσιας (QoS) και παράλληλα ανέδειξαν πεδία για μελλοντική έρευνα και αξιοποίηση.

ABSTRACT

This Ph.D. thesis contributes to the issue of broadband metropolitan area networking infrastructures by designing, implementing and evaluating a prototype architecture that utilises digital television as a backhaul/middle-mile network, enabling users/citizens to access linear, interactive and on-demand broadband services at guaranteed QoS. Building upon the issue of technology and services convergence, and by exploiting the intrinsic characteristic of DVB-T technology to combine heterogeneous data traffic into the same transport stream, it proposes the realisation of DVB-T in regenerative configurations for the establishment of a common IP backbone that is present and available within the entire broadcasting area. This IP-backbone acts as a backhaul/middle-mile connection, which extends the core metropolitan network to reach every user within the broadcasting footprint. Users access this IP-backbone, via intermediate communication nodes (namely Cell Main Nodes – CMN), utilising wired and/or wireless links, both for delivering and consuming broadband services. Efficient network operation and guaranteed QoS provisioning are confronted by designing, implementing and deploying, in a decentralised approach, a services-differentiation mechanism making use of QoS-aware rules according to each user's privileges and services attributes.

Following the system design of the overall network architecture, the thesis elaborates on the implementation of a prototype metropolitan network infrastructure, consisting of one regenerative DVB-T platform and two CMNs, one located within an urban area and another at a rural region. This prototype served as a testbed for carrying out experiments and network performance evaluation measurements. The experimental results verified the capacity of the proposed architecture in establishing broadband metropolitan networking infrastructures with QoS guarantees, besides drew-up fields for future research and exploitation.

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DECLARATION

Part of the contribution of this thesis has been presented in conferences and submitted and/or accepted for publication:

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INTRODUCTION

Background

A fundamental objective of the recent research activities in the fields of Internet, telecommunications and digital broadcasting, is the issue of technology and services convergence, which will allow for the realisation of a common and unified networking environment. In this direction, one of the challenges is to create broadband metropolitan infrastructures that will enable users/citizens to ubiquitously access (both receive and deliver) heterogeneous services (linear, interactive, and on-demand) via any kind of access network and at the maximum possible quality. Until nowadays, several solutions have been proposed for confronting this challenge, mainly based upon two approaches: a) the convergence of Internet and telecommunication technologies, and b) the Interactive broadcasting one.

The former, are dealing with the interconnection of the various wired/wireless access networks (e.g. xDSL, 3G, WLAN, etc.) over the core backbone (e.g. optical fibre), in order to allow text, voice and video services (usually offered via different platforms) to be accessed (received and delivered) via the same infrastructure (triple-play). At the same time they are exploiting the IP-stack as the underlying protocols for content delivery and QoS provisioning (e.g. DiffServ). Actual implementations of these solutions, especially at metropolitan area level, require the access network to be close to the core backbone, as a matter of the specific access network characteristics. Otherwise, either the core backbone (e.g. optical fibre) has to be extended, so that to reach every possible access network, or a backhaul connection (middle-mile) between the access and the core backbone has to be established. In any case, i.e. either extension of the core backbone or exploitation of backhaul connections, the decision to be made is strongly dependent on the trade-off among several issues, including economical, business and geographical ones. For example, while extension of the core backbone seems to be the best technological solution towards the efficient deployment of broadband metropolitan infrastructures, scale economies, installation costs and/or terrain variations, constitute major obstacles for fast take-up. In such cases, backhaul connections seem to be more attractive. Nowadays, backhaul networks are mainly based on wireless technologies (e.g. Microwave), providing for the fast deployment of broadband metropolitan networking infrastructures within any terrain, besides reducing the high installation costs usually required for a wired/cable backhaul connection. However, wireless backhaul installation costs still remain an issue, constraining ubiquitous access and prohibiting the deployment of such broadband metropolitan infrastructures, especially in places that are far from the core-backbone (e.g. rural and dispersed areas).

On the other hand, and in respect to the broadcasting technologies, the efforts were initiated within the "interactive broadcasting" field for the creation of a common environment, where users can access linear digital audiovisual content enhanced with interactive services. Recent actual implementations of this approach exploit the downlink channel (broadcast medium) for forward traffic, while reverse path data (uplink) is provided via wired or wireless telecommunication links (e.g. ISDN, PSTN, WLAN, UMTS, etc.). As a result, and by taking into account that in most interactive broadcasting implementations the same entity (i.e. the broadcaster) is acting both as service/content provider and network operator, users can receive broadband linear services (one-way) and request/access predefined content (interactive services provided by the broadcaster). In this respect, convergence based on this approach is partially achieved, and only at technological level, as long as telecommunications and Internet are utilised for the data requests from the user to the broadcaster. Additionally, the limited system resources (in terms of total available bandwidth) and the use of primitive QoS aware protocols/mechanisms for the delivery of interactive content, prohibit such solutions to be exploited for enabling users to provide/distribute their own content to the entire infrastructure, and for the provision of IP on-demand services with guaranteed QoS. In this respect, while current interactive broadcasting implementations can provide an alternative solution for homogeneous broadband services access/reception, they hardly contribute to the issue of convergence, for heterogeneous services access over a common metropolitan networking infrastructure.

Aim of the Research and Contribution

As the need for technology and services convergence still remains an issue, one of the main research objectives of this PhD thesis is to study, design and propose an alternative solution for the realisation of a common metropolitan networking environment, capable to provide heterogeneous services access over the same infrastructure. It anticipates that if convergence is applied among all technological fields, a unified metropolitan networking infrastructure can be realised, enabling users/citizens to both receive and distribute linear, interactive and on-demand broadband services, either from urban or rural areas, and at maximum possible QoS. In this respect, it proposes a novel network architecture that exploits the terrestrial digital video broadcasting technology (DVB-T) as a complementary wireless backhaul/middle-mile connection, capable to interconnect, anywhere within the entire broadcasting footprint, heterogeneous access IP-based networks, to each other and to the core backbone. Building upon the large coverage-area capabilities of DVB-T, and by exploiting its intrinsic characteristic to combine heterogeneous data traffic into the same transport stream (e.g. MPEG-2 and IP datagrams), it proposes the exploitation of the DVB-T stream in regenerative configurations for establishing a common IP backbone that is present and available within the entire metropolitan area. In this respect, DVB-T technology is not only used for delivering custom linear content (i.e. one-way services) but also as a medium for the provision of interactive multimedia and on-demand services, paving the way towards a fully converged environment. Users access the common IP backbone, for both distributing and consuming broadband services, via intermediate communication nodes (namely Cell Main Nodes – CMN), which may utilise wired and/or wireless links. Efficient network operation, as a matter of dynamic resource allocation and maximum possible QoS provisioning, is confronted by adopting a decentralised approach (at each CMN level). More specifically, a dynamic management system for traffic prioritisation is designed and implemented, allowing for optimised services provision, in respect to the available system resources, user's privileges and services QoS attributes. Differentiated Services (DiffServ) aware mechanisms are utilised both at the backhaul and at the core network, capable to adapt, classify and prioritise IP traffic according to specific QoS requirements.

Towards establishing the validity of the proposed architecture, an experimental networking prototype was designed and implemented according to the design specifications for conducting network performance evaluation tests under real transmission/reception conditions. The obtained experimental results verified the capacity of the proposed architecture for heterogeneous services access with QoS guarantees, establishing it as an alternative solution towards the realisation of broadband metropolitan networking infrastructures.

Structure of the Thesis

Following this introductory chapter, the rest of this thesis is structured as follows:

Chapter 1, provides a brief technology review of the existing approaches for technology/services convergence, and delves into the solutions proposed for the establishment of backhaul connections, besides analysing the interactive broadcasting ones. In this respect, it presents some representative cases for wired/wireless backhaul networking connections, while elaborating on the QoS aware mechanisms utilised for IP services distribution over the core and the backhaul networks with QoS guarantees. Following, it discusses the European standard for digital terrestrial video broadcasting (DVB-T), and presents the encapsulation techniques required for introducing interactivity (IP data into MPEG-2 transport streams). Finally, it analyses the configuration of existing interactive broadcasting installations, capable to provide linear and interactive services delivery.

Chapter 2, elaborates on the design of a networking metropolitan infrastructure that exploits DVB-T technology as an alternative solution for realising backhaul networks. In this respect, it elaborates on the exploitation of the DVB-T stream as a middle-mile network for extending the core backbone to reach every user within the broadcasting footprint (metropolitan area), and proposes the utilisation of this backhaul in regenerative configurations for the creation of a common IP-backbone. Towards enabling users to access this common IP-backbone for heterogeneous linear and on-demand services provision, it proposes a decentralised architecture consisting of a number of Cell Main Nodes, which make use of wired and/or wireless technologies in the access network. Finally, for guaranteed QoS provision, it elaborates on the design of services prioritisation mechanisms (DiffServ) and the corresponding QoS rules (according to specific users' privileges and services attributes), and describes their deployment both at the backhaul and at the CMN levels.

Chapter 3, in turn, presents the implementation and realisation of a prototype that conforms to the design specifications, and which serves as a testbed for conducting preliminary performance evaluation experiments. In this context, it presents the implementation and configuration of the regenerative DVB-T, where the common IP-backbone is created, as well as the realisation of two CMNs; one located in an urban area by utilising WLAN technology in the access network, and another CMN located in a rural metropolitan area, where no-access to the core backbone is currently established and only primitive PSTN/ISDN connections are available. Then it elaborates on the capacity of the implemented prototype in delivering on-demand (TCP) and multicast (UDP) services, via a series of tests utilising TCP data traffic and UDP streams respectively. Preliminary performance experimental results establish the validity of the proposed architecture during TCP data traffic as a matter of the configuration parameters in the prototype and the protocol attributes, besides verifying it during UDP data provision as a matter of the available network resources and the provided stream parameters.

Chapter 4, elaborates on the capacity of the proposed architecture in providing QoS guarantees according to specific users' privileges and services attributes. In this context, it describes the design and implementation of QoS aware modules that make use of DiffServ capable mechanisms and analyses the rules according to which services differentiation and QoS provisioning is obtained. Towards evaluating the overall system performance, and verifying its capacity in guaranteed QoS provisioning, it presents a number of experiments that were conducted under realistic transmission/reception conditions in respect to delay-sensitive and bandwidth-dependent services provision (i.e. multicast audio/video), comprising both row UDP data traffic and real video streams (MPEG-4 videos). Finally, it elaborates on the perceived QoS provisioning, based on a subjective picture quality assessment method, for verifying the capability of the implemented prototype in guaranteed QoS provisioning (according to users' privileges and services attributes) and for establishing the proposed architecture as an alternative solution for broadband metropolitan networks able to provide differentiated services.

The last chapter of this Ph.D. thesis, Chapter 5, concludes by summarising the scientific findings and research results, besides elaborating on fields for future exploitation.

1. TECHNOLOGY REVIEW

1.1. Introduction

The current trends in metropolitan networking infrastructures is the realisation of a common environment that converges Internet, telecommunications and broadcasting technologies, in order to enable users to ubiquitously receive and deliver heterogeneous services at maximum possible QoS. Towards these, two main approaches exist: a) the convergence of Internet and telecommunication technologies, and b) the Interactive broadcasting one.

Research results for such a convergent environment, which are stemming from the convergence between Internet and telecommunications, describe the interconnection of the various access networks over the metropolitan backbone (e.g. optical fibre), and exploit the IP-stack as the underlying protocol. In this framework, users can both receive and deliver triple-play services (text, voice and video), via the same infrastructure (e.g. xDSL), while experiencing maximum possible QoS over sophisticated IP-based mechanisms (e.g. DiffServ). Major obstacles, however, in the deployment of these solutions within the entire metropolitan area (so that to serve every place/citizen), are the technological limitations of the various access networks in achieving direct connection to the core backbone. To overcome this, two main solutions are offered; either to extend the core backbone (in order to reach every possible access network), or to establish an indirect connection from each dispersed access network to the core backbone; such an indirect connection is known as backhaul or middle-mile network.

On the other hand, research results for the realisation of a metropolitan convergent environment, stemming from the interactive broadcasting field, describe the provision of linear audiovisual content, enhanced with interactive services. In this framework, the broadcast medium is exploited for delivering the forward traffic, while services' requests are carried over wired or wireless links (e.g. ISDN, PSTN, WLAN, UMTS, etc.). Major obstacles, however, in the exploitation of these solutions as a broadband metropolitan network, are the limited system resources (in terms of total available bandwidth), and the use of primitive QoS aware protocols/mechanisms.

This chapter briefly discusses the two approaches for technology/services convergence, and delves into the solutions provided for the establishment of backhaul connections, while analysing the interactive broadcasting ones. In this respect, it presents some representative wired and wireless solutions for the backhaul networking connections, while elaborating on the QoS-aware mechanisms utilised for IP services distribution over the core and the backhaul. Thereinafter, it briefly discusses the European standard for digital terrestrial video broadcasting (DVB-T), and analyses the encapsulation techniques required for introducing interactivity (IP data into MPEG-2 streams). Finally, it presents and analyses the configuration of existing interactive broadcasting installations, capable to provide linear and interactive services.

1.2. Backhaul

Delivery of broadband services depends not only on the existence of a broadband access network, the so-called "last mile", but also on a means of connection to the mainline or backbone networks, which form part of metropolitan data transmission networks. This connection is known as backhaul and has been called the "middle mile" (see Fig. 1-1). Backhaul is a significant issue, since high-capacity backbone networks (e.g. optical fibre) are normally found in large cities (as main part of the national networks), and obtaining connection to them is a substantial factor in the provision of broadband services. Nowadays, backhaul to the nearest available main network node can be addressed by a variety of wired/cable and wireless technologies, such as optical fibres, radio and microwave links, satellite links, etc.

Fig. 1-1 Broadband access and backhaul connection

1.2.1. Microwave Links

Microwave links are utilised as a backhaul solution for transmitting signals in a point-to-point configuration, on a line of sight radio path. In microwave radio relay, radio waves are transmitted between the two locations with directional antennas, forming a fixed radio connection between the two points. The basic components required for operating a radio link are the transmitter, towers, antennas, and receiver. Transmitter functions typically include multiplexing, encoding, modulation, up-conversion from baseband or intermediate frequency (IF) to radio frequency (RF), power amplification, and filtering for spectrum control. Receiver functions include RF filtering, down-conversion from RF to IF, amplification at IF, equalisation, demodulation, decoding, and demultiplexing.

The design of microwave radio systems involves engineering of the path to evaluate the effects of propagation on performance, development of a frequency allocation plan, and proper selection of radio and link components. This design process must ensure that outage requirements are met on a per link and system basis. The frequency allocation plan is based on four elements: the local frequency regulatory authority requirements, selected radio transmitter and receiver characteristics, antenna characteristics, and potential intrasystem and intersystem RF interference.

Nevertheless, various phenomena associated with propagation, such as multipath fading and interference, affect microwave radio performance. The modes of propagation between two radio antennas may include a direct, line-of-sight (LOS) path but also a ground or surface wave that parallels the earth's surface, a sky wave from signal components reflected off the troposphere or ionosphere, a ground reflected path, and a path diffracted from an obstacle in the terrain. The presence and utility of these modes depend on the link geometry, both distance and terrain between the two antennas, and the operating frequency. For frequencies in the microwave (~2 – 30 GHz) band, the LOS propagation mode is the predominant mode available for use; the other modes may cause interference with the stronger LOS path. Line-of-sight links are limited in distance by the curvature of the earth, obstacles along the path, and free-space loss.

1.2.2. Satellite Links

Satellite systems can provide broadband connectivity realising backhaul connections to almost any place. They are, however, limited in bandwidth and experience latency problems because of the long paths that signals must traverse between the earth station, satellite and back again. A small volume of broadband provision can be accomplished with existing satellites, while widespread delivery of services would imply the launching of new transponders. Satellite technology is generally considered to be expensive, if ubiquitous.

Three types of satellite systems are currently widespread, classified upon their height of orbit. Geosynchronous earth orbit satellites (GEO), which are at 36,000 km above the earth, low earth orbit satellites (LEO), which are much nearer, 500 – 1,500 km, and medium earth orbit satellites (MEO), which circle at 7,000 – 12,000 km. However, backhaul connections and fixed broadband provision is invariably supported by GEO satellite platforms. GEO satellites have the largest terrestrial coverage. Their distance limits the bandwidth that can be sent for a given power, with the result that GEO satellites need high power transmitters and elaborate, aligned antenna systems (satellite dishes).

1.2.3. Optical Networks

Optical fibre networks can be utilised as a backhaul technology from the core backbone to the local exchange point of presence, offering a very high symmetric bandwidth with low latency and low contention. It is, however, very expensive, due to the high cost of installing new physical infrastructure and equipment. A fibre backhaul network can use different technologies providing a point-to-point connection from the local exchange to core backbone (e.g. SDH over fibre, SONET).

Synchronous optical networking (SONET) and Synchronous Digital Hierarchy (SDH), are two closely related multiplexing protocols for transferring multiple digital bit streams using lasers or light-emitting diodes (LEDs) over the same optical fibre. The method was developed to replace the Plesiochronous Digital Hierarchy (PDH) system for transporting larger amounts of data traffic over the same fibre wire without synchronisation problems.

SONET and SDH are based on circuit mode communication, meaning that each connection achieves a constant bit rate and delay. For example, SDH or SONET may be utilised to allow several Internet Service Providers to share the same optical fibre, without being affected by each others traffic load, and without being able to temporarily borrow free capacity from each other. Since SONET and SDH are characterised as pure time division multiplexing (TDM) protocols, offering permanent connections, and do not involve packet mode communication, they are considered as physical layer protocols. Both SDH and SONET are widely used today in many countries worldwide.

1.3. Quality of Service in Backhaul and Core IP-Networks

A number of several standards are utilised towards providing Quality of Service (QoS) in IP-based networks at a certain guaranteed level according to specific requirements. QoS technologies achieve this by prioritising the networking traffic and by allocating the available network resources in such way, that the desired amount of bandwidth is reserved while network performance evaluation metrics are kept in acceptable predefined levels. Internet Engineering Task Force (IETF) [1] has developed IntServ (Integrated Services) [2], DiffServ (Differentiated Services) [3] and MPLS (Multi-protocol Label Switching) [4] as the QoS technologies utilised in IP based networks. MPLS standard provides QoS by creating end-to-end circuits, with specific performance characteristics, across any type of transport medium but it requires sophisticated and expensive equipment (i.e. MPLS aware routers) to be deployed. IntServ standard provides deterministic guarantees (i.e. deterministic bandwidths and end-to-end delays) to individual flows but it is very complex and suffers from scalability issues, while DiffServ standard, compared to the other two technologies, does not require any special equipment and is very scalable.

DiffServ standard aims to provide Quality of Service to traffic aggregates rather than to individual flows. A traffic aggregate can consist of many independent to each other flows that share the same QoS requirements. These requirements are defined in a pre-established agreement between the service provider and the end user that is called Service Level Agreement (SLA). From the SLA derives the Traffic Condition Agreement (TCA), which specifies when the incoming traffic is "in profile" or "out of profile" and the actions to be taken in both cases. These actions consist on the classification and conditioning of the incoming traffic by the DiffServ aware nodes in such way that the agreed QoS level of the provided services is guaranteed.

The overall DiffServ architecture comprises two subsystems. The first subsystem consists of a number of DiffServ Edge Routers (DERs), which are located at the edge of a network and according to the subscribed SLAs and TCAs their main operation is to perform traffic conditioning and classification. This can be achieved by marking accordingly the DS (Differentiated Service) field of every IP packet (Type of Service field in IPv4, Traffic Class field in IPv6) [5], setting a specific DSCP (Differentiated Service Code Point) value. IP packets having the same DSCP value belong to the same Behaviour Aggregate (BA) class. DERs also perform metering, policing/dropping or/and shaping of the incoming traffic in such way that will guarantee the QoS requirements of the aggregated flows. The second subsystem consists of a number of DiffServ Core Routers (DCRs), which are located inside the core network and according to the subscribed SLAs and TCAs, their main operation is to forward the traffic belonging to every BA to its destination according to a certain Per Hop Behaviour (PHB) [6]. A PHB refers to the kind of forwarding treatment that BAs will have from the DCRs in order to meet the desired QoS level and is achieved by prioritising accordingly every BA class with the use of a range of queue service and/or queue management disciplines.

According to the DiffServ concept every BA (traffic aggregate) must be mapped to a PHB group. The Expedited Forwarding (EF) PHB [7] is used to provide QoS in Premium Services by offering them low packet losses, low one way delay, low jitter and assured bandwidth. The Assured Forwarding (AF) PHB [8] is used to provide different levels of forwarding assurances for incoming traffic. It succeeds that by defining four AF classes of different priority. Every class supports additionally three different drop levels. The incoming traffic is classified to a certain AF class accordingly to its priority. AF can be used to implement the "Olympic Service" model, where three different service classes exist (i.e. Gold, Silver and Bronze class). Traffic assigned to gold class has higher priority towards the other two classes so in case of congestion it will receive a more beneficial treatment from the DiffServ routers. The Best Effort (BE) PHB is also used for services with less QoS requirements.

1.4. Digital Video Broadcasting (DVB)

1.4.1. DVB Evolution

The technological sector of digital television [9], [10] in Europe was established by the Digital Video Broadcasting (DVB) project [11], [12] as an industry-led consortium of over 270 broadcasters, manufacturers, network operators, software developers and regulatory bodies. DVB defines the systems [13] for the delivery of MPEG-2 transport/programme streams [14], [15], [16] over satellite, cable and wireless terrestrial links. More specifically, DVB-S standard [17], [18], [19] defines the modulation and channel coding utilised for digital television services transmission over satellite links. Distribution of digital television services over cable systems is achieved by utilising the DVB-C standard [20]. DVB-C signals are received by subscribers terminals (Set-Top-Box) enabling the support of interactive multimedia and networking services. Finally, digital television broadcast services can also be provided over terrestrial wireless links in the VHF/UHF frequency bands, by exploiting the DVB-T standard [21], [22]. In order to meet, however the increasing requirements for mobile reception of DVB-T signals via handhelds devices, and alleviate the need for battery-saving on them, DVB-H technology [23], [24], [25], [26], [27] was introduced as an extension/addition to the DVB-T one.

As DVB-T is nowadays the dominant standard for digital TV programmes provision over terrestrial channels [28], the following subsections briefly describe the principles of its operation and the basic building blocks.

1.4.2. DVB-T Standard

DVB-T standard [21], [22] defines the functional block of equipment performing the adaptation of the baseband television signals from the output of an MPEG-2 transport stream multiplexer, to the terrestrial channel characteristics. The following processes are applied to the input data stream (i.e. MPEG-2 TS) in a DVB-T modulator:

- Transport multiplex adaptation and randomisation for energy dispersal;
- Outer coding (i.e. Reed-Solomon code);
- Outer interleaving (i.e. convolutional interleaving);
- Inner coding (i.e. punctured convolutional code);
- Inner interleaving (i.e. either native or in-depth);
- Mapping and modulation;
- Orthogonal Frequency Division Multiplexing (OFDM) transmission

DVB-T standard is directly compatible with MPEG-2 coded television signals [16]. It was designed enabling for the provision of digital terrestrial broadcasting services [29], [30], [31], [32] in the existing VHF and UHF spectrum allocated for transmission of analogue television programs. DVB-T standard provides sufficient protection against high level of co-channel and adjacent channel interference emanating from existing PAL/SECAM/NTSC services. It also allows for maximum spectrum efficiency when utilised in the VHF and UHF bands in Single Frequency Network (SFN) operation [33].

DVB-T specifications define different levels of QAM modulation and different inner code rates utilised in order to trade off between bit rate and signal ruggedness. Two level hierarchical channel coding and modulation is also supported including uniform and multi-resolution constellation. The block diagram of a DVB-T modulator is depicted in Fig. 1-2.

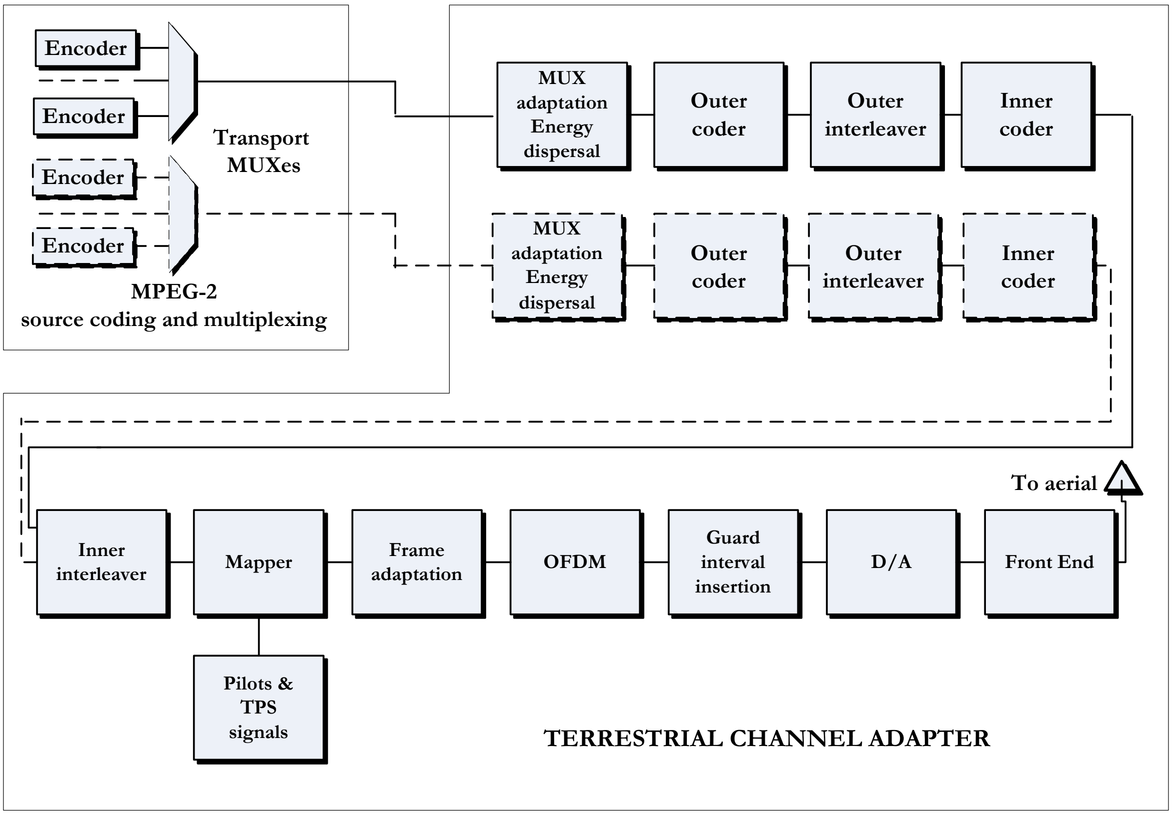


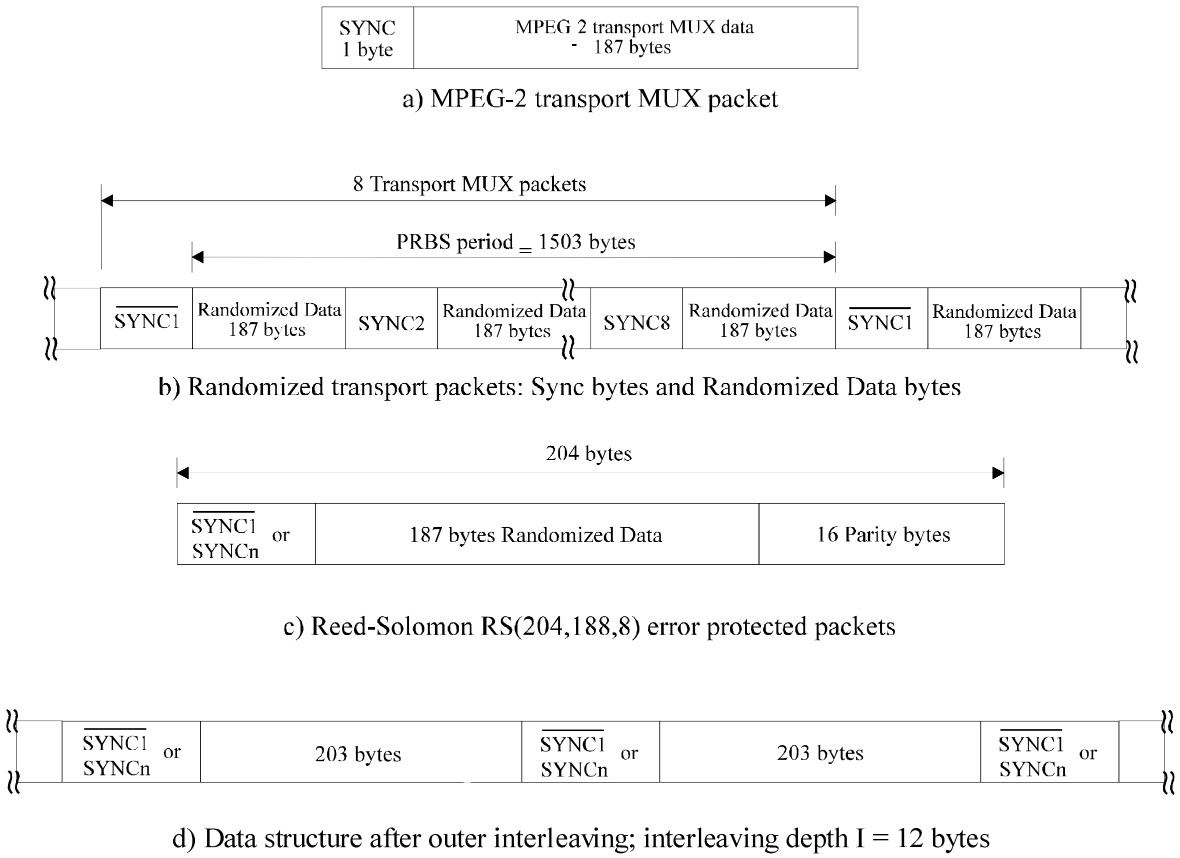
Fig. 1-2 Functional block diagram of DVB-T modulator

1.4.2.1. Transport Multiplex Adaptation and Randomisation for Energy Dispersal

The output data stream from an MPEG-2 transport multiplexer is organised in fixed length packets (188bytes), each one including 1 sync-word byte. The processing order at the transmitting side always starts from MSB of the sync-word byte. In order to ensure adequate binary transitions, the data of this MPEG-2 multiplex output is randomised. This process assures that data is randomly spread in the available bandwidth of the DVB-T signal. This is achieved inserting MPEG-2 transport stream in a scrambler, which utilises a Pseudo Random Binary Sequence (PRBS) generator. The PRBS generator has a period of 1503bytes and handles 8 data packets each time. In order to provide an initialisation signal for the descrambler, the MPEG-2 sync-byte of the first transport packet in a group of eight packets is bit-wise inverted. This process is referred to as transport multiplex adaptation.

1.4.2.2. Outer Coding and Outer Interleaving

The outer coding and interleaving is performed on the input packet structure as depicted in Fig. 1-3a. Reed-Solomon RS (204,188, t=8) shortened code derived from the original systematic RS (255,239, t=8) code is applied to each randomised transport packet of Fig. 1-3b in order to generate an error protected packet (see Fig. 1-3c). Reed-Solomon coding is also applied to the packet sync-byte, either non-inverted or inverted. The Reed-Solomon code has length 204bytes, dimension 188bytes and allows correcting up to 8 random erroneous bytes in a received word of 204bytes. The shortened Reed-Solomon code may be implemented by adding 51bytes, all set to zero, before the information bytes at the input of an RS (255,239, t=8) encoder. After the RS coding procedure these null bytes are discarded, leading to an RS code word of N=204bytes. After the process of outer coding a convolutional byte-wise interleaving with depth I=12 is applied to the error protected packets (see Fig. 1-3d). The convolutional interleaving process is based on the Forney approach which is compatible with Ramsey type III approach, with I=12. The interleaved data bytes are composed of error protected packets and are delimited by inverted or non-inverted MPEG-2 sync bytes (preserving the periodicity of 204bytes).



a) MPEG-2 transport MUX packet

b) Randomized transport packets: Sync bytes and Randomized Data bytes

c) Reed-Solomon RS(204,188,8) error protected packets

d) Data structure after outer interleaving; interleaving depth I = 12 bytes

Fig. 1-3 Steps in the process of adaptation, energy dispersal, outer coding and interleaving

1.4.2.3. Inner Coding and Inner Interleaving

DVB-T standard allows for a range of punctured convolutional codes, based on a mother convolutional code of rate ½ with 63 states. This allows the selection of the most appropriate level of error correction for a given service or data rate in either non-hierarchical or hierarchical transmission mode. In addition to the mother code rate ½, the standard allows punctured rates of 2/3, ¾, 5/6 and 7/8. DVB-T standard also specifies the native inner interleaving processes utilised for 2K and 8K transmission modes. After inner interleaving process a sequence of bits is generated, which is organised in symbols. The available modulation schemes, that can be utilised for signal constellation, are QPSK, 16QAM, 64QAM (2bits/symbol, 4bits/symbol and 6bits/symbol in each carrier respectively). Data is then transmitted to the terrestrial environment according to the Coded Orthogonal Frequency Division Multiplexing (COFDM) [34], [35].

1.4.2.4. Coded Orthogonal Frequency Division Multiplexing

A COFDM signal is comprised of an ensemble of closely spaced carriers generated by an Inverse Fast Fourier Transform (IFFT) technique within the modulator (see Fig. 1-4). The transmission signal is organised in frames (4 frames constitute one super-frame) each one consisted of 68 subsequent COFDM symbols. Every COFDM symbol is constituted by a set of 6817 (8K mode) or 1705 (2K mode) carriers and transmitted with a duration T_s. Two subsequent carriers are separated by Δf=1116Hz and Δf=4464Hz for 8K and 2K operation mode respectively and each symbol is composed of two parts: a useful part with duration T_u=896μs (8K mode) or T_u=224μs (2K mode) and a guard interval with a duration Δ (four values of guard interval may be used; i.e. $T_u/4$, $T_u/8$, $T_u/16$, $T_u/32$. The guard interval consists in a cyclic continuation of the useful part of the COFDM symbol and is inserted before it. The guard interval is the period of time within each total symbol period when no new data is modulated onto the carriers. It is used to allow reception in a multipath environment, where time delayed signals are added to form a composite receive signal. During this guard interval any time delayed signal that is received will add constructively to the main signal without causing interference. The guard interval utilisation in a symbol increases its total duration reducing the capacity of the symbol. As depicted in Fig. 1-5, the guard interval duration for a fixed modulation state and inner code rate, affects the DVB-T channel capacity. However, the utilisation of increased guard interval durations in a DVB-T system, optimises the multipath performance of the system and the SFN transmitter separation distance.

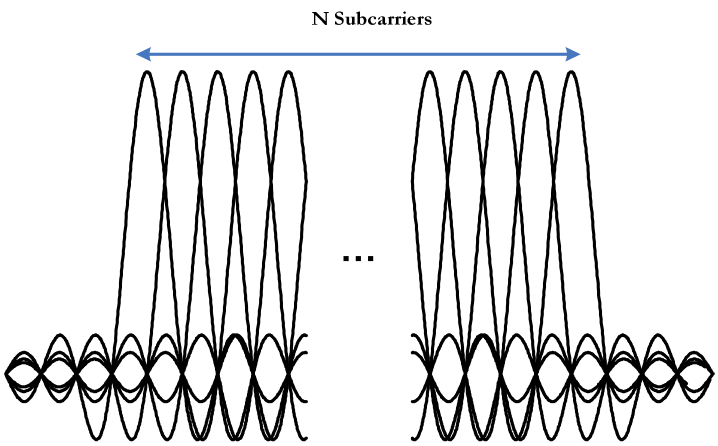


Fig. 1-4 COFDM subcarriers

The orthogonality condition in the COFDM modulation technique implies a fixed relationship between all the carriers in the ensemble (i.e. subsequent carriers are always separated by $\Delta f=1/T_u$). The side lobes of each carrier extend in both frequency directions and tend to zero as they move away from the main lobe (see Fig. 1-4).

Since the COFDM signal comprises many separately-modulated carriers, each symbol can in turn be considered to be divided into cells, each corresponding to the modulation carried on one carrier during one symbol. In addition to the transmitted data a COFDM frame contains scattered pilot cells, continual pilot carriers and Transmission Parameter Signalling (TPS) carriers. The pilots can be used for frame synchronisation, frequency synchronisation, time synchronisation, channel estimation, transmission mode identification and can also be used to follow the phase noise. The TPS carriers are used for the purpose of signalling parameters related to the transmission scheme; i.e. channel coding and modulation.

The choice for 2K or 8K broadcasting mode in the DVB-T modulator does not affect the total available bit rate during digital terrestrial television transmission. Useful bit rate is affected by modulation scheme, code rate and guard interval configured in a DVB-T modulator (see Fig. 1-5).

Modulation	**Code rate**	**Guard interval**			
		1/4	**1/8**	**1/16**	**1/32**
QPSK	1/2	4.98	5.53	5.85	6.03
	2/3	6.64	7.37	7.81	8.04
	3/4	7.46	8.29	8.78	9.05
	5/6	8.29	9.22	9.76	10.05
	7/8	8.71	9.68	10.25	10.56
16-QAM	1/2	9.95	11.06	11.71	12.06
	2/3	13.27	14.75	15.61	16.09
	3/4	14.93	16.59	17.56	18.10
	5/6	16.59	18.43	19.52	20.11
	7/8	17.42	19.35	20.49	21.11
64-QAM	1/2	14.93	16.59	17.56	18.10
	2/3	19.91	22.12	23.42	24.13
	3/4	22.39	24.88	26.35	27.14
	5/6	24.88	27.65	29.27	30.16
	7/8	26.13	29.03	30.74	31.67

Fig. 1-5 Useful bitrate (Mb/s) for all combinations of guard interval, constellation and code rate for non-hierarchical systems (8MHz channels)

The COFDM signal bandwidth can be tailored for different channel bandwidths. The typical channel bandwidth utilised in the most DVB-T systems is 8MHz (6MHz and 7MHz channel bandwidths are also available). The COFDM signal is normally centred co-incidentally with the centre of the assigned Radio Frequency channel. This leaves approximately 0.2MHz on either side of the COFDM signal as a guard band. The theoretical DVB-T transmission signal spectrum for guard interval $\Delta=T_u/4$ in an 8MHz channel is depicted in Fig. 1-6.

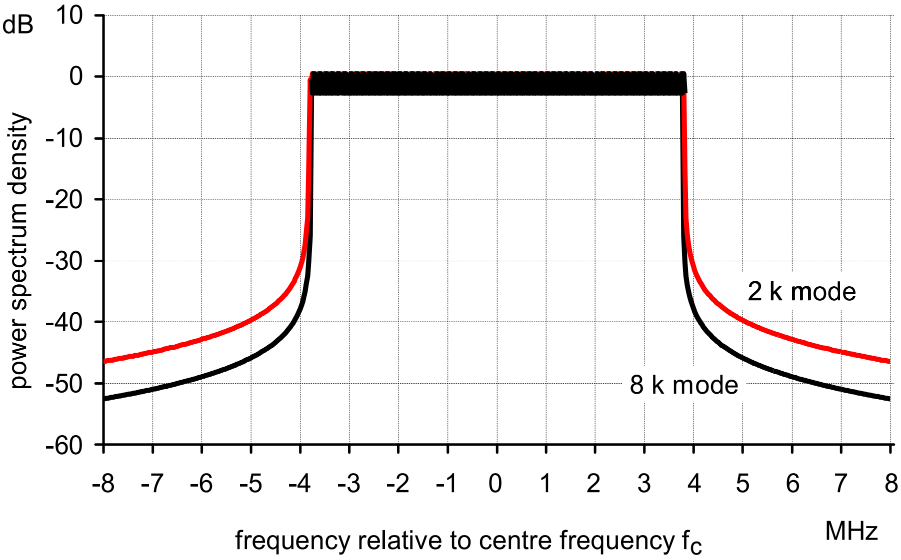


Fig. 1-6 Theoretical DVB-T transmission signal spectrum for guard interval $\Delta = T_u/4$ (for 8MHz channels)

1.5. IP Encapsulation in DVB Systems

1.5.1. Overview

Towards enabling for the provision of IP based traffic through DVB channels several IP encapsulation techniques have been defined and standardised (ETSI EN 301 192 [36]), including the Data Piping, Data Streaming, Data Carousels, and the Multi Protocol Encapsulation technique, which supports data broadcast services that require the transmission of communication protocols datagrams via DVB compliant broadcast networks. Moreover, recently a new encapsulation method has been proposed and standardised, namely Unidirectional Lightweight Encapsulation (ULE) protocol [37], offering optimised performance when compared to the MPE one [38].

1.5.2. Multi Protocol Encapsulation (MPE)

The Multiprotocol Encapsulation (MPE) supports data broadcast services that require the transmission of communication protocols datagrams via DVB compliant broadcast networks. The transmission of datagrams according to the multiprotocol encapsulation specification is done by encapsulating them in DSM-CC (Digital Storage Media Command and Control) sections [39] (see Fig. 1-7), which are compliant with the MPEG-2 private section format [16]. This process includes the addition of a header in the beginning of each section before the IP datagram and a 4-byte CRC or checksum at the end. Among others the header contains fields declaring the length of the section, the scrambling status and the MAC address of the receiver. Once formed the MPE section is split into several fragments and transferred to the payload MPEG-2 TS packets. If the length of the MPE section is not an integer multiple of the MPEG-2 TS packet payload (184bytes), the encapsulator module has the option of either padding the rest of the last packet with stuffing bytes (padding) or beginning a new MPE section in the remaining area (packing). The main drawback of MPE is the inclusion of several

MPEG-specific fields in the section header, which in fact can be omitted. Moreover, the declaration of the receiver MAC address is mandatory in MPE, adding an overhead of 6 more bytes. Another issue is the absence of the declaration of type of data contained in the MPE section. MPE offers the option of either having a pure IP payload or carrying the data with an LLC/SNAP (Logical Link Control/Sub-Network Access Point) header. So, there is no uniform representation of the type of the encapsulated data as it exists e.g. in Ethernet framing with the Type field.

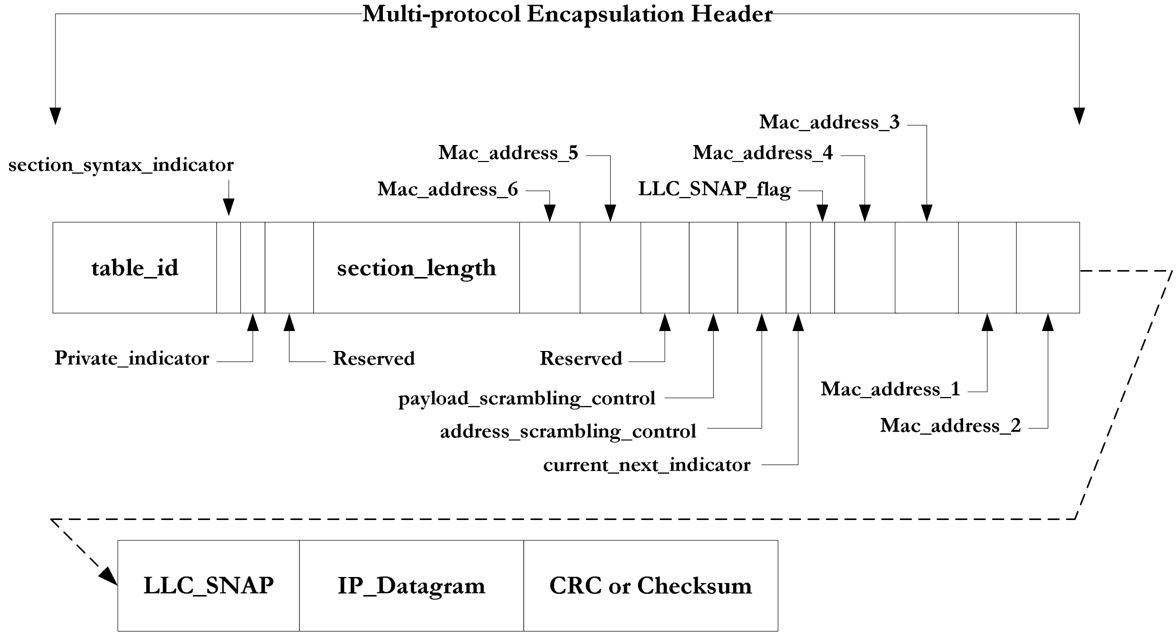


Fig. 1-7 MPE section format

MPE has been the most suitable method for conveying IP datagrams and is adopted by all IP/DVB gateways in the last few years [40]. Additionally, a new lightweight encapsulation method (Unidirectional Lightweight Encapsulation - ULE) has been defined supporting optimum efficiency in comparison to MPE encapsulation method, which adds extra information due to framing and signalling [41].

1.5.3. Unidirectional Lightweight Encapsulation (ULE)

ULE was designed with the aim of making the IP datagrams encapsulation process as lightweight as possible [42], [43]. It follows the approach of data piping method (i.e. directly mapping the datagrams into the MPEG-2 transport stream payload), adding only a small header. ULE header contains just a length field which declares the length of the ULE section and a Type field which has the same functionality with that of Ethernet (i.e. it declares the type of the payload) [44]. ULE provides native support for state of the art network protocols such as IPv6 and MPLS [45], [46]. Depending on the value of this field the Protocol Data Unit (PDU) can be an IPv4 datagram, an IPv6 datagram, an MPLS or even a whole bridged MAC-frame. The ULE header can also include a 6-byte destination address corresponding to the receiver's Network Point of Attachment (NPA). The NPA address is used to uniquely identify a receiver in the MPEG-2 transmission network and is mandatory only in the case that the PDU is to be processed by a receiver-router. If this is not the case and the data is directly received by the destination terminal, this field can be omitted and filtering can be performed at IP level. Additionally, a CRC-32 tail is appended to ensure proper reception and synchronisation. Fig. 1-8 depicts the structure of the ULE section. The framing is as lightweight as possible retaining only the necessary fields for proper de-encapsulation and forwarding of the IP datagram.

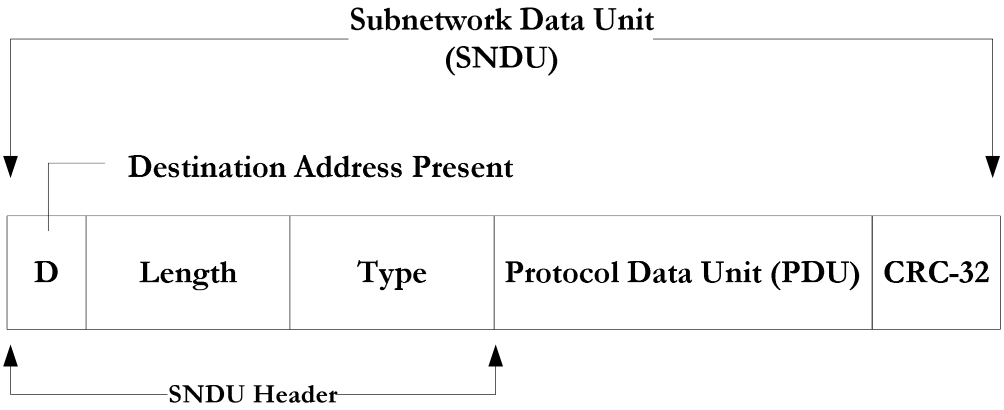


Fig. 1-8 ULE section structure

After framing, the ULE section is mapped to the payload of the MPEG-2 transport stream packets. In the case that the ULE section length is not an integer multiple of the packet payload, one of the aforementioned techniques of Padding or Packing can be employed. A simple observation of Fig. 1-7 and Fig. 1-8 is sufficient to demonstrate the simplicity introduced by ULE. By reducing the framing fields only to necessary ones, ULE saves bandwidth and processing time at the encapsulator (IP/DVB gateway). Moreover, ULE restricts the filtering at the receiver to the network layer only and eliminates the need for an IP-to-MAC association table in the encapsulator, which would need additional resources in order to be created and maintained. In the case of MPE, the encapsulation of each datagram requires a lookup in this table in order to specify the MAC address corresponding to each IP destination.

1.6. Interactive Broadcasting

1.6.1. Overview

DVB standards define the specifications enabling for the provision of custom linear services (i.e. one way) to users located inside large coverage areas. Towards supporting offline interactivity the first interactive broadcasting systems [47], [48], [49], [50] enabled for the provision of IP datacasting, where a large volume of data was pushed with constant bit rate to the end users' terminals through the DVB channel with a repeated process (data carousel). Data was then received offline by the end user via a graphic interface. There was no real interactivity in such systems since the end user was not able to request for real time services (e.g. Internet) and provide his own services due to lack of an interaction channel. If the forward DVB channel is combined with an interaction channel using any of the existing wired or wireless access technologies then an interactive network is realised able to provide non-linear multimedia and networking services. Fig. 1-9 depicts the generic model for DVB interactive systems [51].

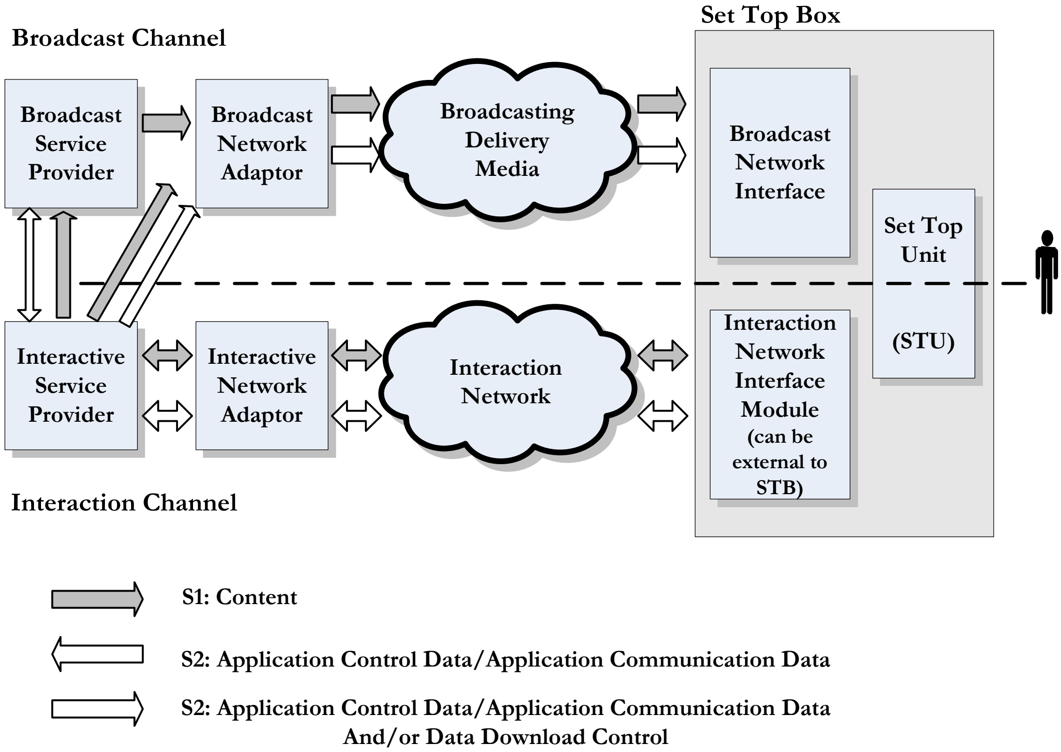


Fig. 1-9 Generic model for DVB interactive systems

In this configuration, the service provider's side may incorporate both an interactive service provider and a broadcaster. The former may enable access to interactive multimedia services such as video and audio on demand, or may provide Internet facilities such as WWW access, e-mail services, etc. The latter is responsible for the distribution of broadcast services, such as TV programmes, that utilise digital transmission formats (i.e. MPEG-2 TS). Both interactive and broadcast services are multiplexed into one transmission stream by the broadcast network adapter and distributed to the users via the broadcasting delivery media according to a DVB standard. Each user receives the multiplexed broadcasts via a broadcast interface module (e.g. antenna, front-end amplifier, etc.) that passes the appropriate data to an end-user module (TV receiver, PC screen, PC station, etc.) via a set top unit (de-multiplexer, decoder, etc.). The users' requests for interactive services are forwarded by the set-top unit to an interaction interface module that may utilise wired access technology (PSTN, ISDN, xDSL, etc.) [52], [53] or wireless (LMDS, DECT, DVB-RCT, DVB-RCS, GSM, GPRS, UMTS etc.) [54], [55], [56], [57], [58], [59], [60], [61] depending on the interaction media's specifications. The interaction media, in turn, passes the users' requests to the interactive service provider via the interactive network adapter.

The generic model for DVB interactivity has been utilised realising interactive broadcasting systems [62], [63], [64], [65], [66], [67], [68], [69], [70] enabling for the provision of non-linear services through satellite, cable and wireless terrestrial links.

1.6.2. Interactive DVB-T System Configuration

Based on the generic model of interactivity an interactive DVB-T system may utilise wired or wireless access technologies towards realising the interaction channel and providing non-linear services. The configuration of an interactive DVB-T system is depicted in Fig. 1-10. In this configuration the service provider is the television broadcaster that delivers TV programmes via the UHF band. The linear multimedia television programmes are multiplexed together utilising an MPEG-2 TS multiplexer forming in this way a final DVB-T stream. The interactive services data traffic is then encapsulated into the DVB-T stream that is 64QAM/OFDM modulated, up-converted to the UHF frequency band, amplified and finally broadcasted to the terrestrial coverage area. The users receive the DVB-T signals via a common UHF antenna, that passes the signal to an interface module (signal splitter). The output of this device is fed into a DVB-T compatible receiver/demodulator and to a DVB-T PC card. The former takes the responsibility to demultiplex/demodulate the received UHF information data and provide the appropriate video-audio signals to a TV receiver. The latter (PC DVB-T card) demultiplexes/demodulates the received UHF signal and feeds the PC with the appropriate multimedia information data. This data may be any multicasted/broadcasted multimedia information, or the service provider's (broadcaster's) reply information upon the user's request for interactive services.

In such an interactive DVB-T system whenever the user requests interactive services, the appropriate data (generated by the PC station) is passed to an interaction channel interface module, which takes the responsibility to deliver the information to the interaction channel adapter. The latter passes the data to the interactive service provider that processes the user's request, forwards them to the appropriate networking services server producing the appropriate reply signals. This forward data traffic is then carried to the television broadcaster via the IP/DVB gateway. Finally, the broadcaster distributes the reply signals back to the appropriate user via the UHF downlink.

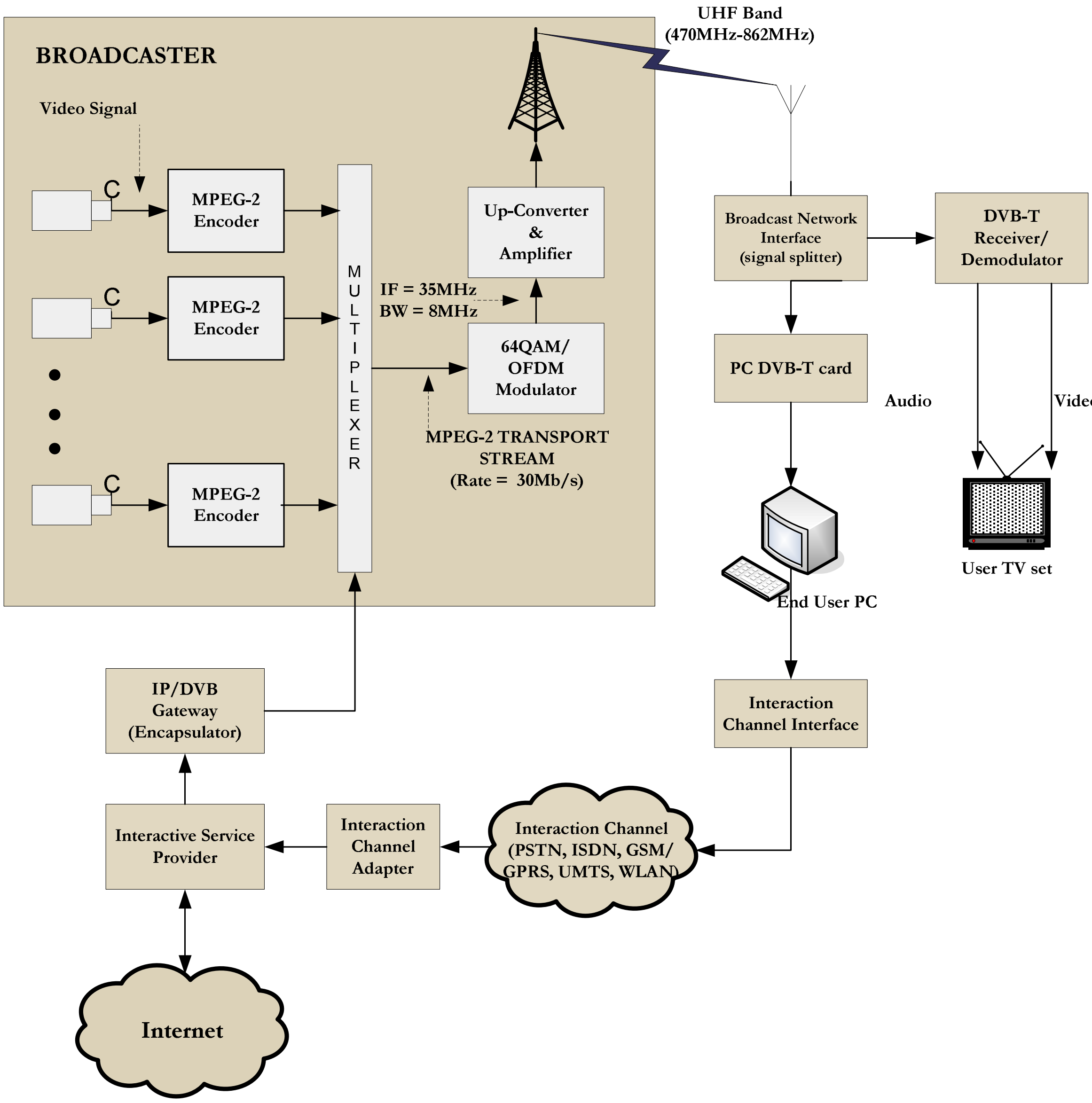


Fig. 1-10 Interactive DVB-T system

Such interactive DVB-T platforms have been extensively studied under the framework of a number of research and development efforts since 1999 [71], [72], [73], [74], [75]. The support of a large number of users in such systems requires, however, complex equipment from the service provider side and multiple interfaces modules for different interaction channel technologies. This considerably limits the end users' number and system capacity preventing the utilisation of such architectures for the realisation of broadband metropolitan networking infrastructures. Additionally, the low bandwidth interaction channel of these systems limits end users capability becoming active ones manipulating and distributing their own services to the entire coverage area (i.e. the target group of these systems includes passive citizens accessing pre-defined content). Also the provision of QoS aware services utilising such systems, requires the use of complex centralised mechanisms in service provider's side in order to control the available network resources and DVB-T available bandwidth.

1.7. Summary

This chapter has briefly discussed the current two approaches for technology/services convergence, which are currently exploited for the realisation of metropolitan infrastructures, capable to provide heterogeneous services via any kind of access network, and at maximum possible QoS. Technology review that was carried out concerning the solutions provided by the Internet and telecommunications convergence, revealed the exploitation of wireless backhaul connections (communication between the access and the core backbone) as a very promising solution for allowing triple-play services access and for creating such convergent environments within the entire metropolitan area. However, installation costs still remain an issue, prohibiting their fast take-up and constraining, therefore, their deployment especially in places that are far from the core-backbone. On the other hand, technology review concerning the efforts for convergence from the interactive broadcasting sector, indicated that major obstacles towards a common metropolitan networking environment are the limited system resources (in terms of total available bandwidth) and the lack of sophisticated QoS aware protocols/mechanisms. Therefore, these solutions are not suitable for the delivery of on-demand IP services with guaranteed QoS services, while prohibiting users to provide/distribute their own content to the entire metropolitan infrastructure.

2. SYSTEM DESIGN

2.1. Introduction

Following the technology review carried out in chapter 1, this chapter elaborates on the design of a prototype architecture that arises as a complementary solution to the issues of broadband metropolitan networking infrastructures. It anticipates that if convergence is applied to all technological sectors (i.e. Internet, telecommunications and broadcasting), an alternative approach can be realised, allowing for the realisation of a common environment capable to provide access (within the entire metropolitan area) to broadband linear, interactive and on-demand services at guaranteed QoS.

Building upon the issue of wireless backhaul connections and by taking into account the principles of interactive DVB-T technology, it presents the exploitation of the DVB-T stream in regenerative configurations for the creation of a common IP-backbone that is present and available within the entire broadcasting footprint. This IP-backbone acts both as a core network for interconnecting various access networks within the metropolitan area, and as backhaul for providing the connection to the core-backbone (e.g. fibre optic). Users access it, for both delivering and consuming broadband multimedia and networking services, via intermediate communication nodes (namely Cell Main Nodes – CMN) utilising wired and/or wireless links.

For guaranteed QoS provision, according to users' privileges and services attributes, the chapter elaborates on the design of a dynamic management system for traffic prioritisation. In this respect, it presents the architecture and network configuration that exploits Differentiated Services (DiffServ) aware mechanisms, both at the backhaul and at the core network, capable to adapt, classify and prioritise IP traffic according to specific QoS requirements.

2.2. Overall Architecture

As already shown in chapter 1, Internet and telecommunications convergence can pave the way towards the deployment of broadband metropolitan infrastructures, by providing access to triple-play services at guaranteed QoS. Their partial inability, however, to serve every possible place and user within the entire metropolitan area, is overcome by exploiting backhaul connections, as a matter, however, of economical and/or geographical issues. On the other hand, convergence based on the interactive broadcasting solutions can enable users to ubiquitously access, within in any place of the metropolitan area, broadband linear content of high quality, enhanced with a short of interactivity. Nevertheless, current interactive broadcasting solutions cannot contribute to the issue of broadband metropolitan networking infrastructures, due to their inability in providing on-demand IP-services at guaranteed QoS. Therefore, a novel approach is required.

In this section it is anticipated that such an approach can be revealed if convergence is applied among Internet, telecommunications and broadcasting sectors, providing a complementary solution for the realisation of a common environment capable to provide access (within the entire metropolitan area) to broadband linear, interactive and on-demand services at guaranteed QoS. Taking into account the large coverage-area capabilities and by exploiting the intrinsic characteristic of the European terrestrial digital video broadcasting standard (DVB-T) to combine heterogeneous data traffic into the same transport stream (e.g. MPEG-2 and IP datagrams), this section proposes the exploitation of DVB-T stream as a backhaul connection for interconnecting various access IP-based networks to each other and to the core-backbone. In this respect, it proposes the design of DVB-T stream in regenerative configurations for establishing a common IP backbone that extends the core backbone to reach every place within the broadcasting footprint (including rural and urban areas), contributing, therefore, to the fast and efficient deployment of metropolitan infrastructures. Users access this backbone, for both delivering and consuming broadband multimedia and networking services, via intermediate communication nodes (namely Cell Main Nodes – CMN) utilising wired and/or wireless links. In this respect, a decentralised architecture is anticipated, capable to support broadband services access via a single and common infrastructure, contributing therefore, to the issue of broadband metropolitan infrastructures with triple-play capabilities. Finally, for the efficient network operation, traffic prioritisation techniques are anticipated, for guaranteed QoS provisioning according to each user's privileges and services' attributes. In this respect, the exploitation of DiffServ QoS-aware mechanisms in a decentralised approach both at the backhaul and at each CMN level, contributes to the issue of broadband metropolitan infrastructures with QoS guarantees.

The overall network architecture of such a decentralised infrastructure is depicted in Fig. 2-1. It consists of two core subsystems: a central broadcasting point (regenerative DVB-T) and secondly, a number of distributed Cell Main Nodes (CMNs) located within the terrestrial broadcasting area. Each CMN enables a number of users (geographically adjacent to the specific CMN) to access IP unicast services that are hosted by the entire infrastructure (e.g. by the ISP and multimedia provider as depicted in Fig. 2-1). The communication between the users and the corresponding CMN (access network) is achieved via broadband point-to-multipoint links (e.g. WLAN, xDSL). The CMN gathers all IP traffic stemming from its own users and forwards it to the central broadcasting point (UHF transmission point visible by all CMNs) via dedicated point-to-point uplinks. IP traffic stemming from all CMNs is received by the broadcasting point, where a process unit filters, regenerates, and multiplexes it into a single transport stream (IP-multiplex) with the digital TV program(s) stemming from the TV broadcaster(s) (TV studio), towards forming the final DVB-T bouquet.

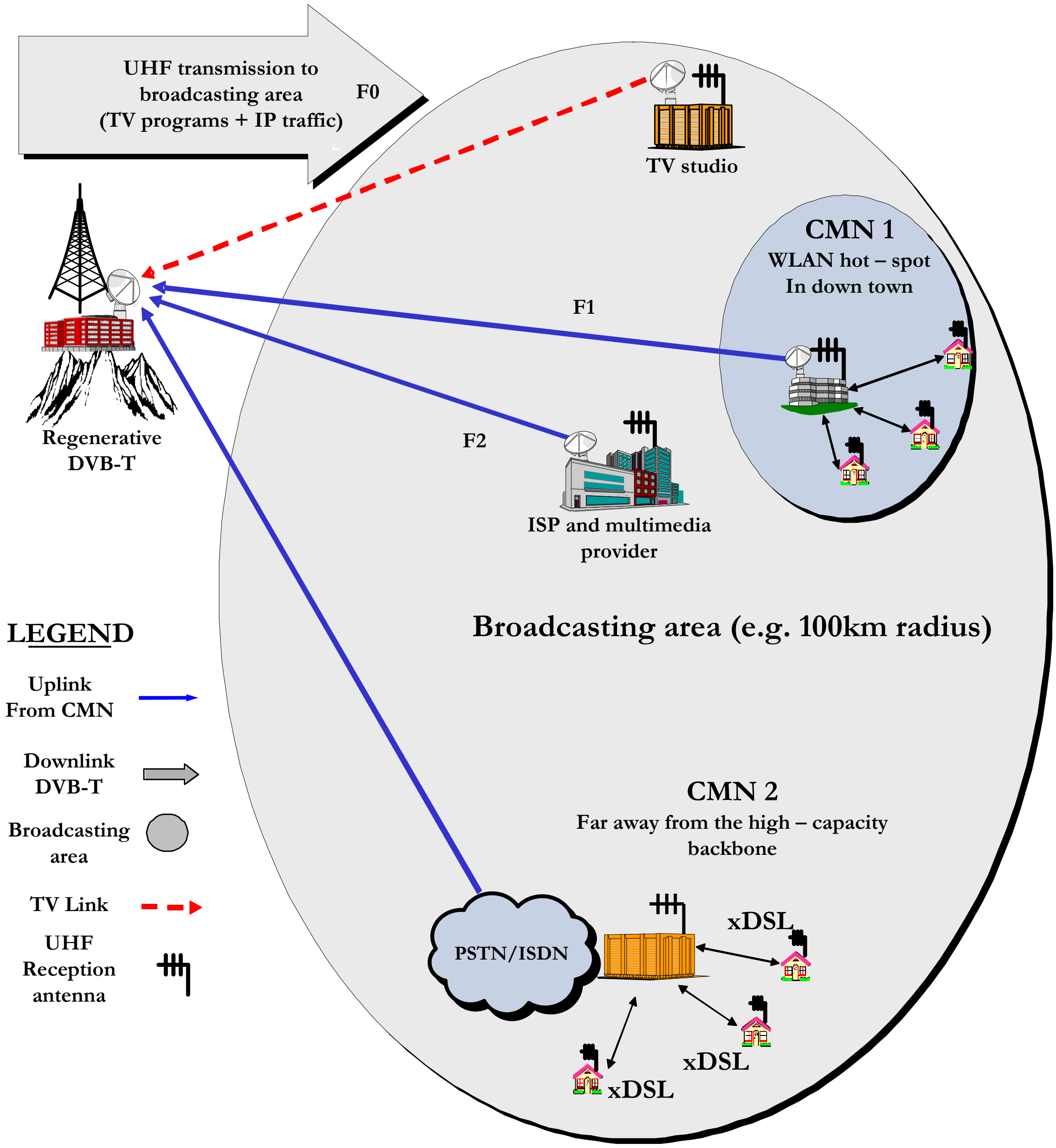


Fig. 2-1 Overall networking architecture

Each user receives the appropriate IP reply signals indirectly via the corresponding CMN, while receiving a custom digital TV program (e.g. MPEG-2) and IP multicast data (e.g. IPTV) directly via the common DVB-T stream (by utilising a simple/custom UHF antenna). In such an architecture, both reverse and forward IP data traffic is encapsulated into the common DVB-T stream, thus improving the flexibility and performance of the networking infrastructure. Furthermore, the cellular conception that is adopted utilises the DVB-T stream in a backbone topology, which interconnects all cells that are located within the broadcasting area. Thus, a unique, virtual common IP backbone is created that is present at every cell via its CMN. The DVB-T bit stream supplies the IP traffic of this IP backbone, while users access the network via the appropriate CMN.

2.2.1. Regenerative DVB-T Design

The architecture of regenerative DVB-T (depicted in Fig. 2-2) is capable of:

- Receiving the users IP traffic over the uplinks (via the appropriate CMN — see PSTN/ISDN uplink and F1 in Fig. 2-1).
- Receiving any local digital TV program(s) with IP multicast and Internet services, stemming from the TV studio broadcaster and the ISP/multimedia provider respectively (see TV Link and F2 in Fig. 2-1).
- Creating and broadcasting a common UHF downlink that carries the digital TV program(s) and the IP data.

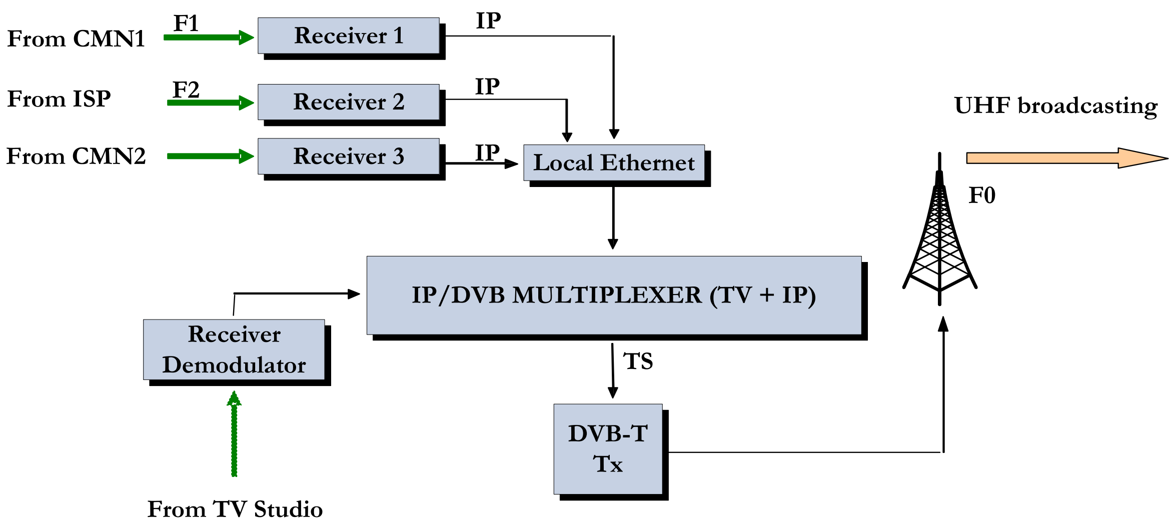


Fig. 2-2 Architecture of regenerative DVB-T

Following the architecture depicted in Fig. 2-2, the multiplexing device receives any type of data (IP and/or digital TV programs), adapts it into a DVB-T transport stream (IP to MPEG-2 encapsulation), and finally broadcasts this DVB-T stream to the entire broadcasting area following the DVB-T standard (COFDM scheme in the UHF band). In this respect, all users and providers contribute to the creation of a backbone (DVB-T stream) that is present and available within the entire broadcasting area.

An actual exploitation of this architecture can be realised for the deployment of xDSL infrastructures that enable not only triple-play services and access, but also and most predominantly, always-on connectivity. The deployment of xDSL access networks require the head-end unit (i.e. DSLAM) to be placed close to a high-capacity core backbone (e.g. fiber), while the end-user equipment must be located no further than a few kilometres from the point-of-presence (e.g. 5Km from the DSLAM). As a result, in areas that are far away from the core backbone, xDSL deployment cannot be realised unless extension of the high-capacity backbone is achieved to reach these areas.

However, taking into account that DVB-T transmissions utilise coverage areas of many kilometres (e.g. 100Km), the proposed architecture (Fig. 2-2) can be exploited for realising the common DVB-T stream in middle-mile/backhaul configurations, extending the core backbone within the entire broadcasting area and making it available or present at any CMN within the coverage footprint. This type of backhaul solution, which conforms to the proposed architecture, is presented next, with a description of an urban-based CMN that hosts an active user (IP multicaster) and a rural-based CMN that is placed away from the core backbone and the service provider (ISP and multimedia provider in Fig. 2-1), providing users with always-on connectivity and access to triple-play services.

2.2.2. Cell Main Nodes Design

The following two subsections discuss in detail design issues regarding urban and rural-based CMNs of the proposed metropolitan networking infrastructure.

2.2.2.1. Urban-based CMN

The urban-based CMNs of the networking infrastructure contribute to the active participation of users (in the context of not only consuming custom and predefined content) but also in the capability of creating, manipulating, and distributing their own content and services over a commonly exploited metropolitan infrastructure. The active participation of these potential content and application providers (stemming from traditional, passive users) is the key to generating revenues, creating rich activity in the market chain, and spearheading new progress in the broadcasting, Internet, and telecommunication sectors. The realisation of such a converged networking environment decouples the service provision function from the network operators and offers this privilege to all potential interested players, changing the traditional passive users to active ones.

In this context, urban users who wish to become active participants in the information society, can access the entire infrastructure for distributing their own services via a CMN (urban-based CMN), the architecture of which is depicted in Fig. 2-3. This architecture utilises a broadband uplink for realising the communication between CMN and regenerative DVB-T (e.g. microwave RF link) and WLAN technology in the access network, consisting of an access point (AP) at the CMN site, which maintains a full duplex communication with the station adapters (SAs) at the users' sites. The output from each SA is in IP form, which can be transparently processed by the upper layers of the software at the users' terminals.

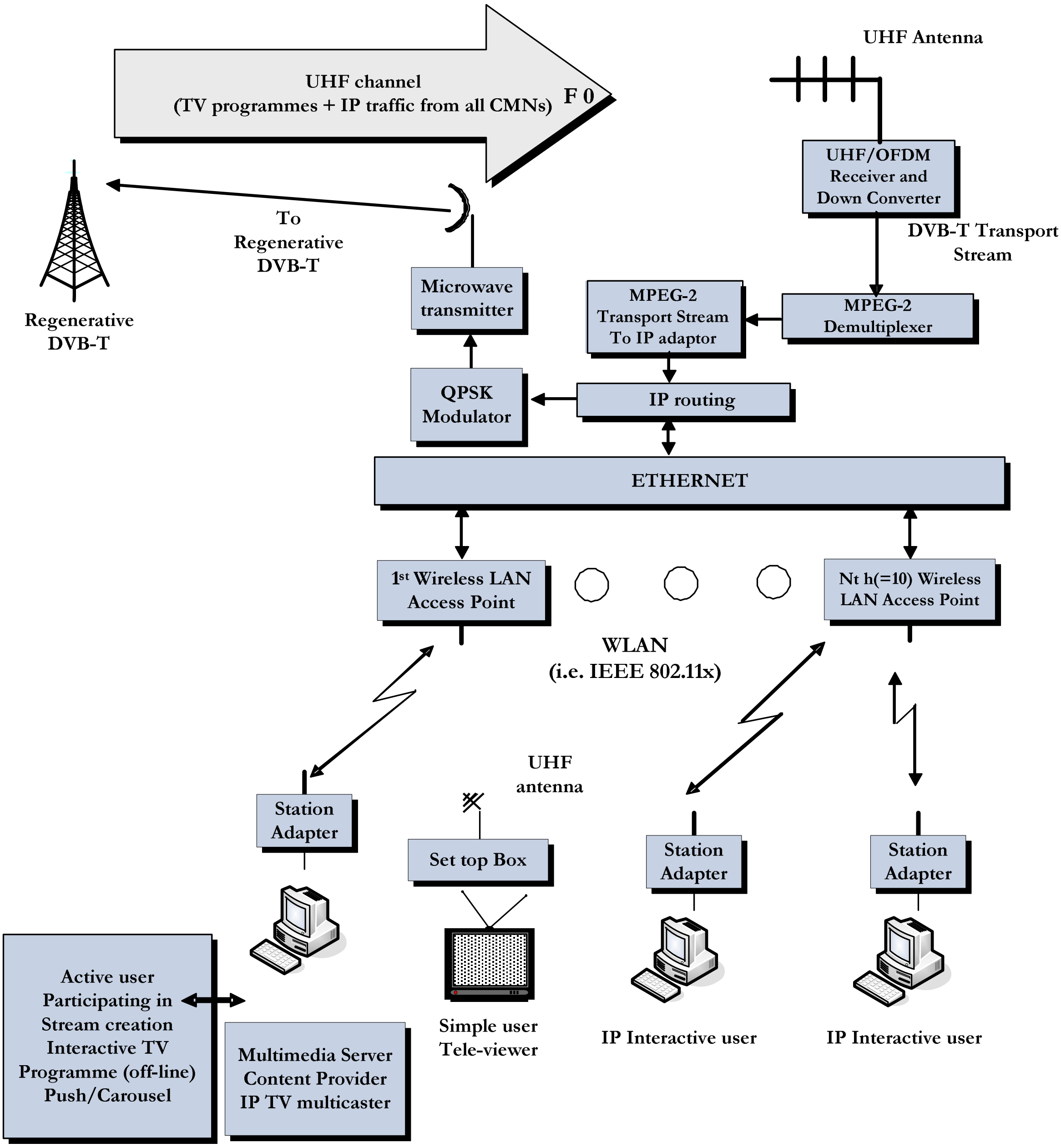


Fig. 2-3 Overall architecture of an urban-based CMN with WLAN technology in the access network

The CMN gathers all IP traffic stemming from its users (custom and active users, potential content providers, etc.), and forwards it to the central broadcasting point (regenerative DVB-T). This IP traffic is processed, regenerated, and multiplexed with all other IP traffic (stemming from other CMNs) into a single transport stream (IP-plex), with the digital TV program(s) stemming from the TV broadcaster, participating in this way in the creation of the final DVB-T bouquet. In this respect, an active user exploits the common DVB-T stream for maintaining his own e-business, which is virtually present at any place within the entire broadcasting area (via the common DVB-T stream). Such an e-business may be the realisation of an IPTV multicaster, capable of targeting customers in a radius of 100Km, located both in urban and rural areas.

Upon a user's request for inter-cell communication (i.e. between a specific user and any other content provider/user/server of the entire infrastructure), its CMN provides the required IP data (i.e. data requests or data acknowledgments) to the entire infrastructure via the common

DVB-T stream (created at the regenerative DVB-T; see Fig. 2-2). The corresponding reply signals (that stem from the content provider/user/server located within another CMN) are received by a DVB-T compliant downlink adapter via the common DVB-T stream (in the UHF band) and are forwarded to the user (who initiated the IP data request) via its SA (Fig. 2-3). In this respect, both reverse and forward IP data traffic is encapsulated into the common DVB-T stream, realising a common IP backbone (present and available within the entire broadcasting area) that is shared and commonly exploited by all users (active or passive ones).

2.2.2.2. Rural-based CMN

In areas that are far away from a core backbone network (e.g. fibre) and where only custom PSTN/ISDN is currently available, access to triple-play services and always-on connectivity cannot be realised, unless extension of the core backbone to these regions is accomplished. As already mentioned, cost constitutes the major obstacle for such extension of the core backbone in these areas.

For these regions or cases, the exploitation of the proposed networking infrastructure is a very promising solution, enabling not only provision of triple-play services but also of always-on connectivity via cost-effective (marginal cost) extension of the core backbone. Primarily, the common DVB-T stream enables broadcast and multicast data (such as digital TV MPEG-2, and IPTV services) to be present and available within the entire coverage area. Users can easily and cost-effectively access services by utilising a simple/custom UHF reception antenna and the corresponding DVB-T reception equipment on their premises. Secondly, the exploitation of the common DVB-T stream in backhaul/middle-mile configurations, enables the fast/immediate interconnection between the core backbone and any CMN within the entire broadcasting footprint. With such an approach, in a way, the core backbone is transferred to rural-based CMNs, enabling for the provision of triple play services and always-on connectivity (e.g. ADSL).

The overall architecture of such a rural-based CMN is depicted in Fig. 2-4. It comprises DVB-T compliant equipment for receiving the common DVB-T stream (UHF channel), a reverse path channel (uplink communication from this CMN to the regenerative DVB-T), making use of the PSTN/ISDN technology (already available in this area), and an ADSL-based access network. At this point it should be noted that the deployment of the ADSL technology in the access network would not have been feasible due to high costs for the backhaul connection (physical connection between this rural-based CMN and the nearest optical backbone network). However, the proposed architecture enables the low cost and fast deployment of an ADSL network, by exploiting the common DVB-T stream as a backhaul and middle-mile infrastructure, capable of interconnecting the core backbone (present in urban areas) with the rural-based CMN. Such an approach enables rural users to maintain always-on connectivity (over the ADSL network) and triple-play services access over the common UHF beam. Upon a user's request for personalised/unicast IP services that are hosted by the core high-capacity network (e.g. Internet), the corresponding IP data requests are forwarded by the ADSL modem to the IP-DSLAM module, which takes the responsibility of forwarding them to the local Ethernet of this CMN (rural-based CMN). The local Ethernet, in turn, forwards them to the uplink interface module, which passes them to the central broadcasting point (regenerative DVB-T) via the uplink chain (PSTN/ISDN network). For maintaining one-way communication between the CMN and the regenerative DVB-T, the central broadcasting point comprises a routing mechanism that blocks downlink data traffic from the regenerative DVB-T to the CMN via the PSTN/ISDN. Finally, the data requests reach the ISP and multimedia provider via the common DVB-T UHF channel. The corresponding reply signals are provided by ISP and multimedia provider to the regenerative DVB-T over the uplink (F1 in Fig. 2-1) and reach the Ethernet of the rural-based CMN via the DVB-T UHF channel. Users receive the corresponding reply data via their CMN Ethernet over the IP-DSLAM module. It should be noted that in such a configuration (regenerative), both the reverse and forward IP data traffic is encapsulated into the common DVB-T stream (i.e. all IP data are in the same stream along with the digital TV program/bouquet). Multicast IP services (e.g. IPTV, IP-Radio stemming from the ISP and multimedia provider), are received by all users (rural and urban ones) directly via the common DVB-T stream, through a custom UHF antenna.

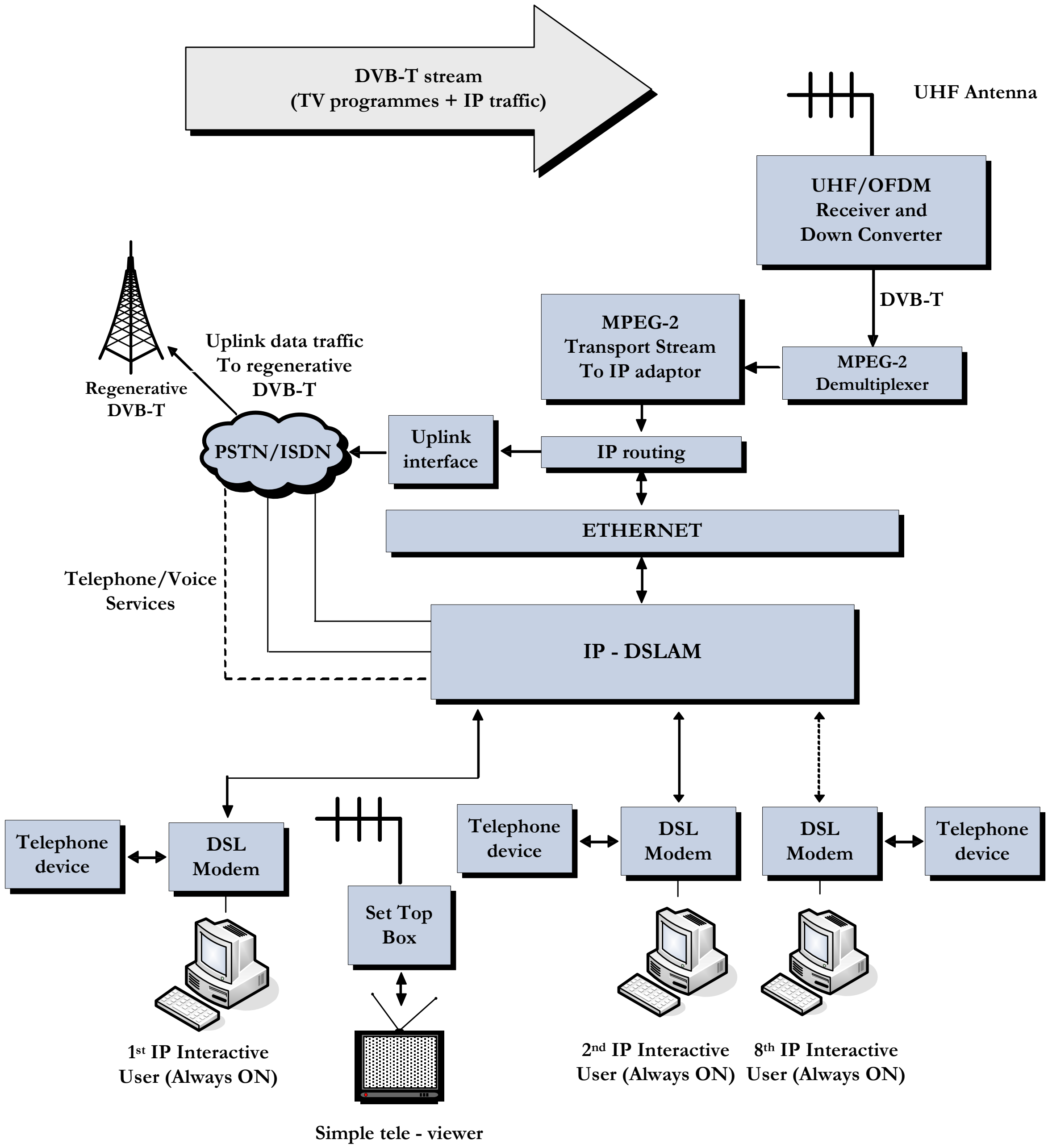


Fig. 2-4 Overall architecture of a rural-based CMN with ADSL technology in the access network

2.3. QoS aware Mechanisms

Towards enabling for the provision of IP services at a guaranteed quality, the proposed metropolitan infrastructure was enhanced with QoS aware modules, making use of traffic prioritisation techniques. More specifically, the networking architecture was enhanced with DiffServ mechanisms [3], [76] for the provision of end-to-end IP traffic differentiation, both across the backhaul and the backbone infrastructure. Fig. 2-5 depicts the overall architecture of the QoS aware networking infrastructure, featuring DiffServ-enabled modules.

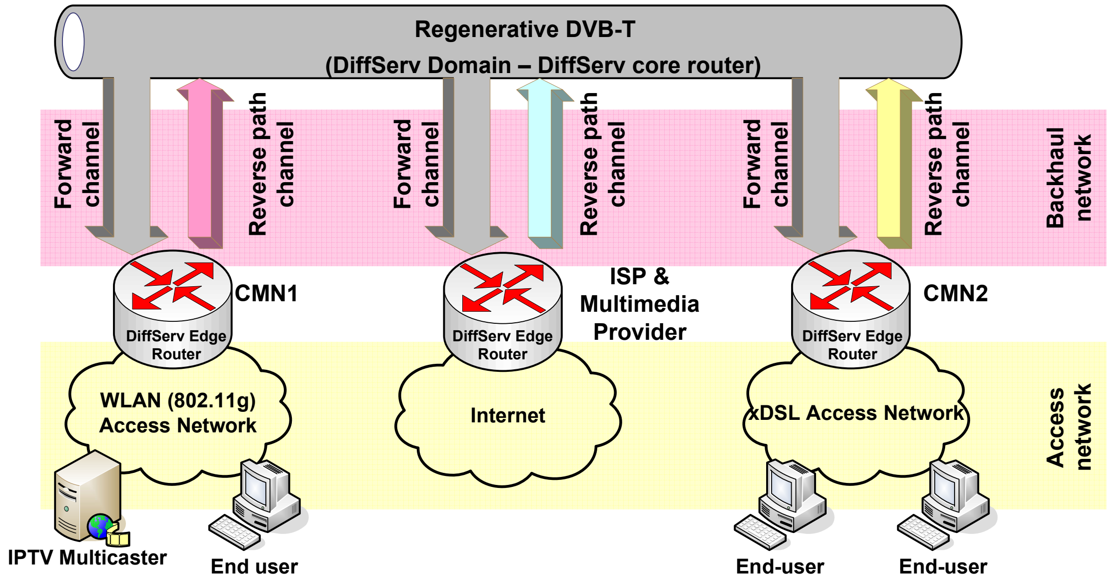


Fig. 2-5 QoS aware networking architecture

In this architecture, every CMN is considered as an entry-point to the DiffServ domain, classifying incoming IP traffic into a specific quality level, by marking accordingly the Differentiated Service Field (DS Field) of every IP packet with a certain Differentiated Service Code Point value (DSCP) [5], associating every data flow to a certain QoS class. Each class was created according to specific QoS requirements (i.e. bandwidth, one way delay, packet-losses), following a certain Service Level Agreement (SLA), signed between the Service Provider (e.g. IPTV multicaster) and the users.

All IP traffic stemming from the QoS aware CMNs, passes through a DiffServ core router located in regenerative DVB-T (see Fig. 2-6). This DiffServ core router analyses the DS field of all IP packets and forwards them according to a specific Per-Hop-Behaviour (PHB) [6] to all CMNs, i.e. to the entire broadcasting area.

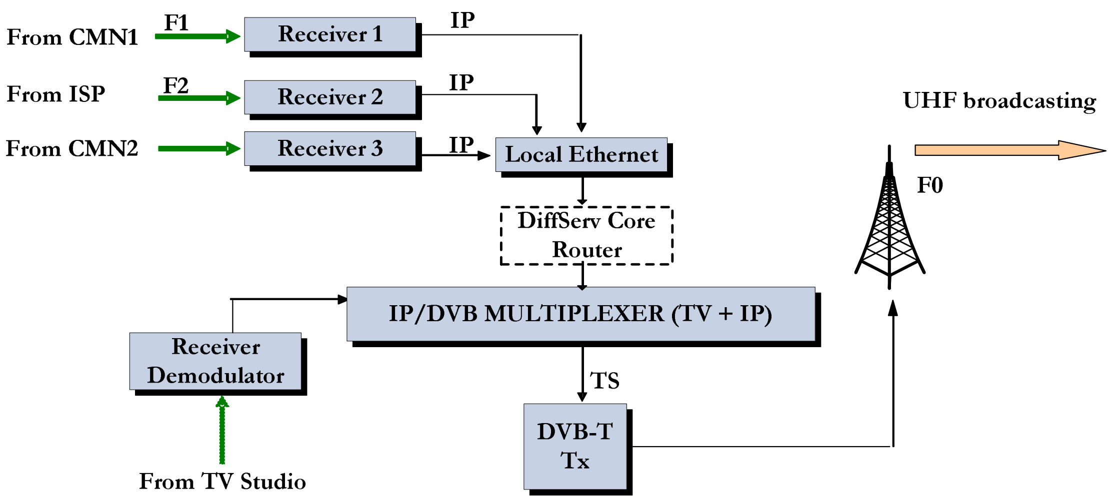


Fig. 2-6 Architecture of regenerative DiffServ aware DVB-T

Fig. 2-6 depicts the architecture of the regenerative DVB-T utilising a DiffServ core router for IP traffic prioritisation, ensuring that services are delivered according to specific QoS requirements. Fig. 2-7, on the other hand, presents the overall architecture of a DSL-based CMN (i.e. CMN with DSL technology in the access network – namely CMN2 in Fig. 2-5), utilising a DiffServ edge router for classifying the IP traffic according to specific services and users' priorities. Following this network architecture, traffic originated from a user located within a certain CMN (i.e. service provider such as the IPTV Multicaster in CMN1) and destined to another user placed in a different CMN (e.g. end-user in CMN2), is classified and forwarded according to specific QoS requirements.

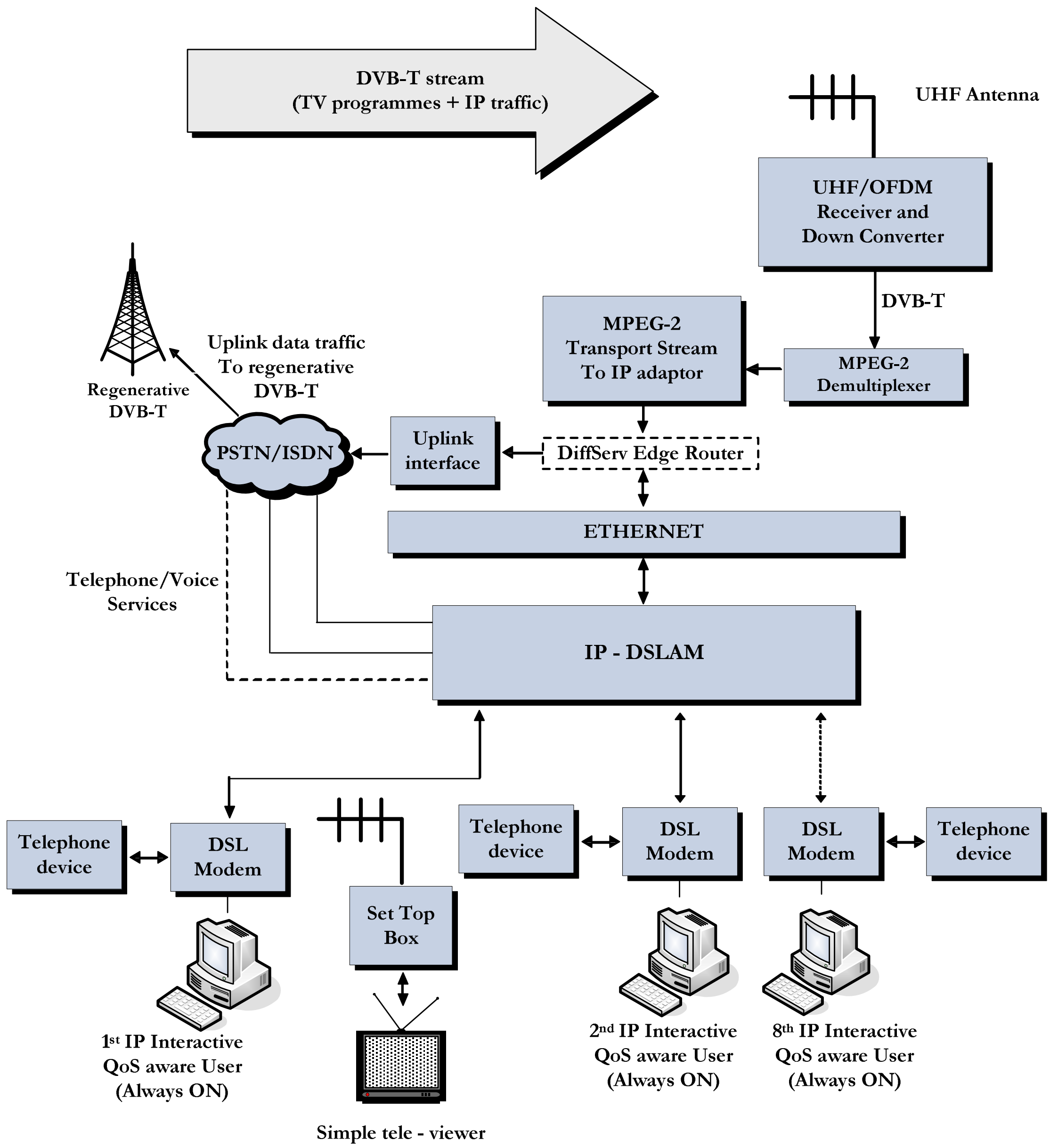


Fig. 2-7 Overall architecture of a DiffServ enabled CMN

2.4. Summary

This chapter presented the design of a prototype architecture that contributes to the issue of broadband metropolitan networking infrastructures by exploiting the convergence among Internet, telecommunications and broadcasting sectors. In this respect, it elaborated on the use of the DVB-T stream in backhaul configuration, for extending the core-backbone to reach every user of the metropolitan area, and proposed its utilisation in regenerative configurations for the creation of a common IP-backbone, carrying heterogeneous services within the entire broadcasting footprint. Towards enabling users to access this common IP-backbone, it proposed a decentralised architecture consisting of a number of Cell Main Nodes, making use of wired and/or wireless technologies in the access network. Finally, for guaranteed QoS provision, according to users' privileges and services attributes, it presented the design and overall network architecture that exploits Differentiated Services (DiffServ) aware mechanisms, both at the backhaul and at the CMN levels.

The work carried out and described in this chapter, forms the basis towards the implementation of a prototype testbed that conforms to the overall design and architectural specifications. Such an implementation is presented in chapter 3, where preliminary experimental tests for verifying the validity of the proposed architecture are also carried-out, while the overall system performance in respect to QoS provisioning is described and analysed in chapter 4.

The research work presented in this chapter was published in [77], [78], [79], [80], [81], [82], [83], [84].

3. PROTOTYPE SET-UP AND PRELIMINARY PERFORMANCE EVALUATION

3.1. Introduction

Following the system architecture, presented in chapter 2, this chapter elaborates on the implementation of a prototype that conforms to the design specifications, which will serve as a testbed for verifying the validity of the proposed architecture, via a series of preliminary performance experiments. In this context, it presents the implementation of the regenerative DVB-T, where the common IP-backbone is created, as well as the realisation of two CMNs; one located in an urban area by utilising WLAN technology in the access network, and another CMN located in a rural metropolitan area (where only primitive PSTN/ISDN lines exist and no access/connection to the core-backbone is available), by exploiting xDSL technology in the access network.

Towards verifying the validity of the proposed architecture, a number of preliminary experiments were designed and conducted under realistic transmission/reception conditions, elaborating on the overall system performance regarding TCP and UDP data traffic. More specifically, and in the case of TCP data traffic, the experimental tests were conducted in respect to useful throughput, round trip time (RTT) and retransmitted packets. On the other hand, experimental tests concerning UDP data traffic were conducted in respect to inter-arrival jitter, one way delay and total packet losses.

Analysis of the experimental results, verified the capacity of the proposed architecture in converging Internet, telecommunications and broadcasting technologies, establishing it as an alternative/complementary solution for the realisation of broadband metropolitan networking infrastructures.

3.2. Overall Configuration of the Prototype

Based on the architectural design specifications, derived in chapter 2, this section elaborates on the system configuration of a real/actual prototype, consisting of the regenerative DVB-T platform, one CMN located within an urban area and one CMN located within a rural metropolitan area. The overall architecture of this prototype is depicted in Fig. 3-1, while the configuration of each one of the three building blocks is detailed and analysed in sub-sections 3.2.1, 3.2.2 and 3.2.3, respectively. Furthermore, the prototype also comprises a local TV broadcaster and an ISP/IP multicaster. The former (i.e. TV studio in Fig. 3-1), exploits the proposed metropolitan networking infrastructure for transmitting its own analogue TV content in digital mode (simulcast), by maintaining all the required equipment for content digitisation and a microwave uplink for delivering the digital TV content to the regenerative DVB-T platform. The latter, i.e. the Internet Services Provider (ISP), provides the prototype testbed with a direct connection to the core-backbone, as well as multicast IPTV/Radio services (IP multicaster). The communication between the ISP and the regenerative DVB-T is achieved by utilising a free space optical (FSO) link, enabling for broadband communication and services delivery.

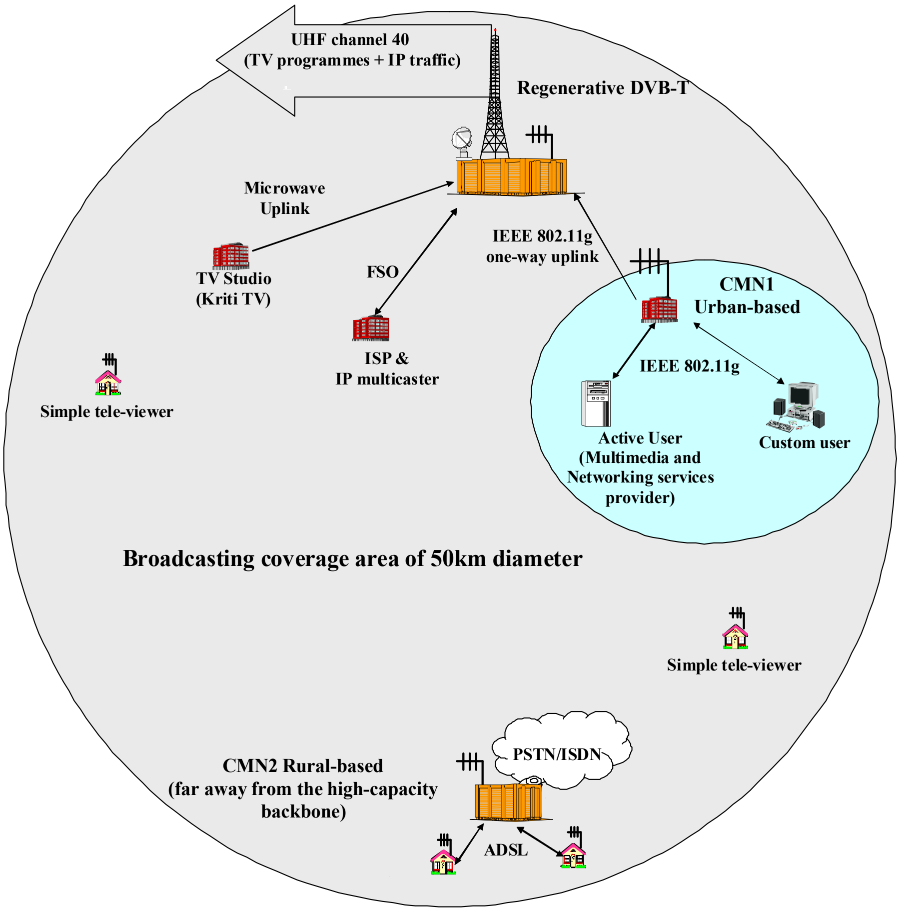


Fig. 3-1 Network architecture of the prototype

3.2.1. Configuration of the Regenerative DVB-T

As already mentioned, the regenerative DVB-T is the place where the common stream (IP-backbone) is created. It utilises a UHF transmission in channel 40 (622-630MHz), by exploiting 8K operation mode with 16QAM modulation scheme, 7/8 code rate, and 1/32 guard interval. These configuration settings/parameters provide for a total available bandwidth of 21.11Mb/s, in the final transmitted MPEG-2 transport stream, according to the DVB-T standard. Furthermore, encapsulation of the IP datagrams into the MPEG-2 transport stream is based on the multi-protocol encapsulation mechanism (MPE). Following Fig. 3-2, below, the regenerative DVB-T is in charge of:

- Receiving the local digital TV programmes from the TV studio.
- Receiving, via satellite, live digital TV broadcasts and re-multiplexing them with the local digital TV programme into the same MPEG-2 transport stream (TS).

- Receiving and processing the ISP's/IP-multicaster's content and IP-streams over the free space optical link (FSO).
- Receiving and processing the users/citizens IP request/acknowledgements over the wired terrestrial uplink (provided via CMN2).
- Receiving and processing the active users' data (VoD/AoD services) over the wireless terrestrial uplink (provided via CMN1).
- Encapsulating all IP data (received over the uplinks and the wireless optical) into the same MPEG-2 transport stream (TS).
- Remultiplexing the encapsulated IP data with the MPEG-2 TV programmes (terrestrial and satellite) into a common MPEG-2 transport stream (TS).
- Broadcasting this common MPEG-2 TS via the common UHF downlink to the entire broadcasting area (metropolitan area).

Furthermore, the regenerative DVB-T can also host and provide non-real time MPEG-2 TV programmes (via the MPEG-2 TS Server).

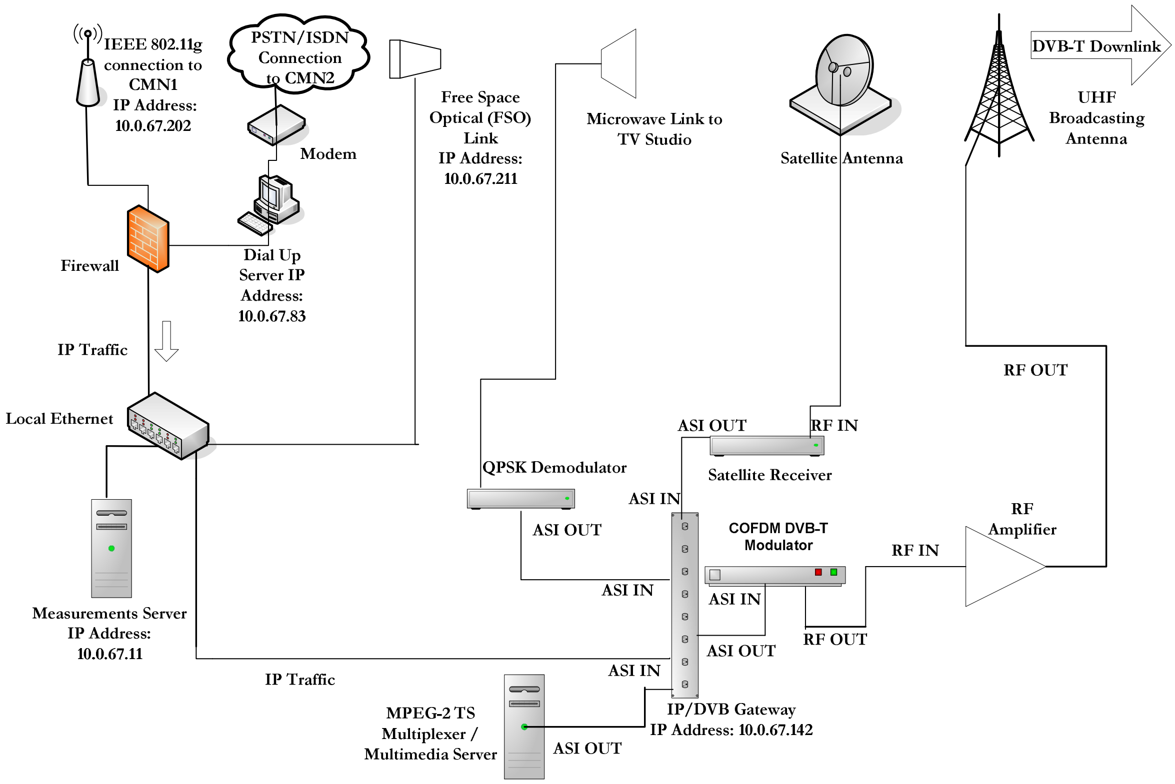


Fig. 3-2 Regenerative DVB-T configuration

3.2.2. Configuration of the Urban CMN

The implemented prototype also consists of one urban-based CMN (namely CMN1 in Fig. 3-1), located near the core/optical broadband backbone, but with no direct connection to it. This CMN accommodates not only custom users, but also active ones, who can create, manipulate, host, and provide access to their own multimedia and networking services (e.g. AoD/VoD service providers). For this reason, a broadband wireless communication uplink is utilised, to carry their content/services to the regenerative DVB-T. This link was established by exploiting IEEE 802.11g technology in a point-to-point configuration. It should be noted that as this technology principally provides for full-duplex communication, a routing mechanism was integrated at the regenerative DVB-T side, to maintain one-way point-to-point transmission (from CMN1 to the regenerative DVB-T). This routing mechanism is mainly operating at the IP level and not at the RF part (Firewall in Fig. 3-2). The overall configuration of the urban based CMN is depicted in Fig. 3-3.

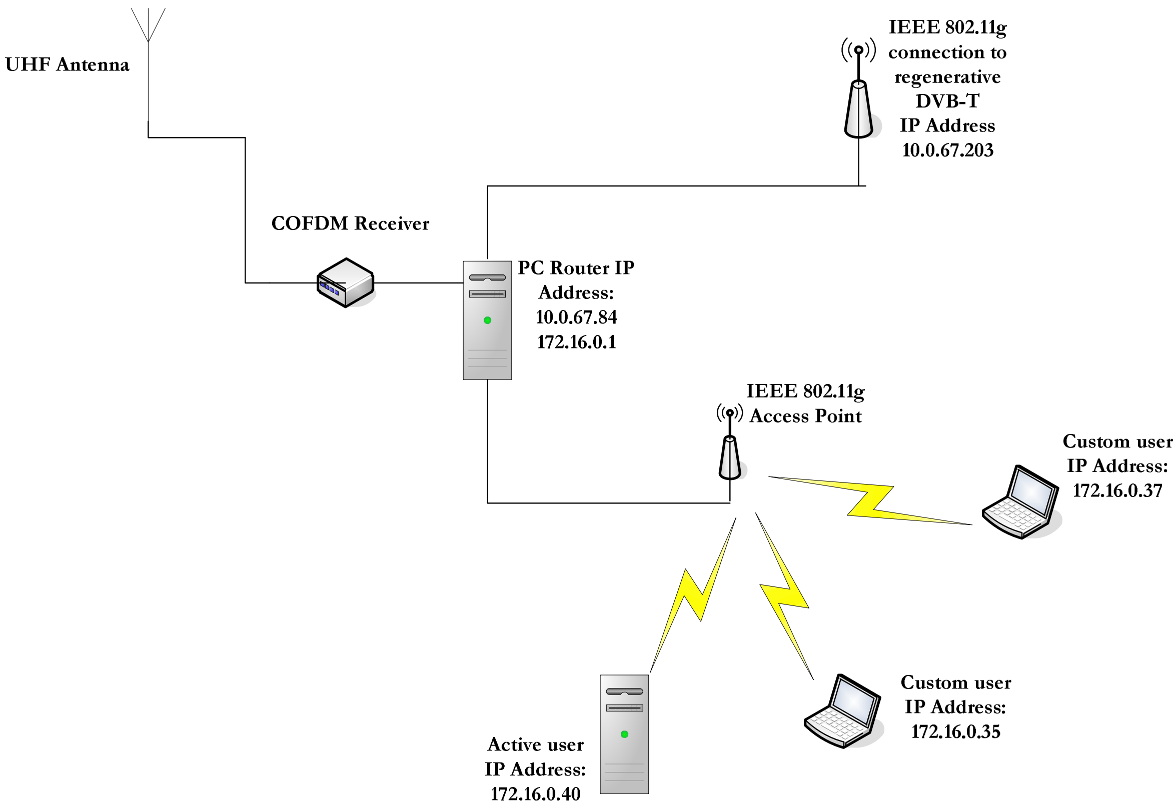


Fig. 3-3 Urban based CMN configuration

According to this configuration, IP data requests for AoD/VoD services provision, stemming by any custom user/citizen within CMN2, are received by CMN1 via the common UHF beam (UHF antenna), demultiplexed/demodulated, and forwarded to the VoD/AoD server via a point-to-multipoint IEEE 802.11g communication link ("Access Point" in Fig. 3-3). The corresponding data replies are forwarded by the access network to the regenerative DVB-T, over the one-way IEEE 802.11g link (uplink making use of the routing mechanism see Fig. 3-3 and Fig. 3-2), where they participate to the creation of the common downlink (DVB-T stream). Finally, the IP data replies (requested AoD and VoD services) are distributed to the entire infrastructure over the common IP backbone (UHF channel), and reach the appropriate custom users/citizens through the corresponding Cell Main Node (i.e. CMN2). Towards these, the urban based CMN comprises of a central processing node (Linux based PC Router), which utilises three interfaces:

- A DVB-T compliant module (COFDM Receiver) for gaining access to the common UHF channel (DVB-T downlink stream). This interface de-multiplexes and demodulates the IP data from the entire MPEG-2 transport stream.

- One IEEE 802.11g WLAN (namely "Access Point" in Fig. 3-3) accommodating custom users/citizens and one active user/citizen who enables access to AoD and VoD services. This interface receives the IP requests/acknowledgments for AoD/VoD services' provision (carried by the common IP backbone in UHF to the PC router), and forwards them to the Active User. The corresponding reply signals (AoD and VoD content provided by the Active User) are forwarded through the uplink interface to the regenerative DVB-T, and from there back to the appropriate custom users/citizens over the common UHF channel.
- An IEEE 802.11g based interface for carrying all IP data stemming from the users of this CMN to the regenerative DVB-T.

3.2.3. Configuration of the Rural CMN

Finally, the implemented prototype also includes one rural-based CMN (namely CMN2 in Fig. 3-1), located about 10 kilometres from the regenerative DVB-T platform (under LOS conditions with it) and in an area, where only PSTN/ISDN is currently available. Users at this area cannot (primarily and in principle) be always-online and exploit triple-play broadband services as well as on-demand content, unless connection to the core-backbone is provided. Towards this, and by exploiting the common DVB-T stream in backhaul/middle-mile configurations, the core-backbone is extended to reach this area, via the corresponding cell main node (i.e. CMN2).

The configuration of this CMN (depicted in Fig. 3-4), utilises it as part of the local PSTN exchanger (placed within the same building), and exploits ADSL technology in the access network, which provides for always-online connectivity, as well as custom PSTN/ISDN channel(s) for the communication with the regenerative DVB-T (uplink). In this respect, and given that establishment with the core-backbone has been achieved via the DVB-T stream, this CMN enables for triple-play broadband services access as well as for on-demand content provision. The low bandwidth, however, of the uplink (PSTN/ISDN link) prohibits the users of this area to become active ones, in order to deliver their own broadband content to the entire infrastructure.

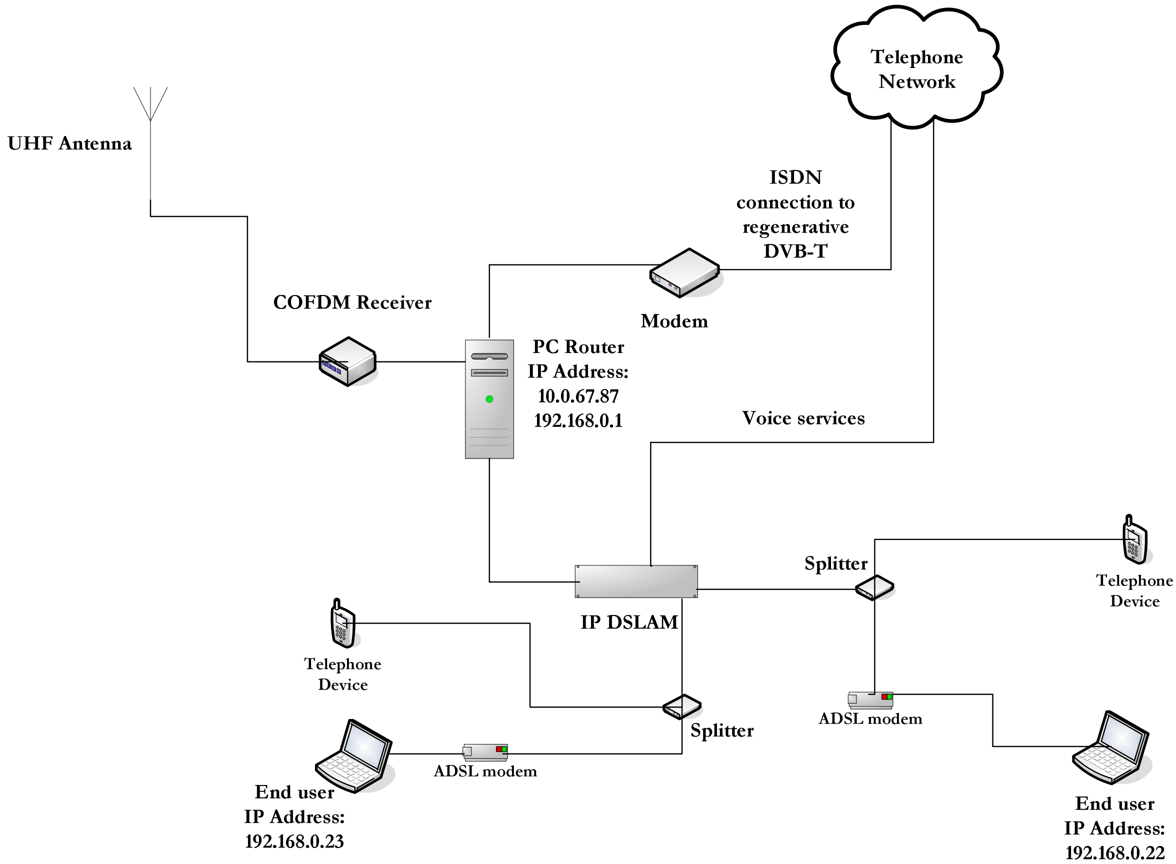


Fig. 3-4 Rural based CMN configuration

3.3. Preliminary Performance Evaluation

Towards evaluating the performance of the implemented prototype, and verifying the capacity of the proposed architecture in establishing broadband infrastructures within the entire metropolitan area, two classes of preliminary experiments were designed and conducted under realistic transmission/reception conditions in terms of multiple network performance evaluation metrics [85]:

- The first class of experiments was designed in respect to on-demand IP-services provision (TCP based data traffic) taking into account the useful throughput, the round trip time (RTT) and the retransmitted packets.
- The second class of experiments was designed in respect to multicast IP-services provision (UDP based data traffic), taking into account the inter-arrival jitter, one way delay and total packet losses.
- It should be noted that during these preliminary tests, the total available bandwidth of the common DVB-T stream (i.e. 21.11Mb/s) was exploited as follows:
- 11Mb/s were allocated for three digital television programs (i.e. MPEG-2 live and non-live broadcasts),
- 8Mb/s were allocated to TCP traffic (i.e. for on-demand services provision), and

- 2Mb/s were allocated for UDP traffic (i.e. IP-multicast streams stemming from the active users).

3.3.1. Network Performance Evaluation based on TCP Traffic

In order to evaluate the network performance during on-demand IP-services provision, a hierarchical and structured framework was adopted, including the evaluation of each individual networking component, and the evaluation of the entire infrastructure/testbed. In this respect, this section presents the evaluation of a) the access networks, b) the uplink-downlink chains, c) the communication of a user and the regenerative DVB-T, and d) an end-to-end communication between a user located in CMN2 and the active user located in CMN1.

During these experiments the Iperf traffic generator [86] was used for establishing TCP connections and providing TCP bulk data traffic between the modules-under-test (e.g. end user located in CMN2 and the active user located in CMN1). Furthermore, the Tcpdump application [87] was utilised for capturing the transmitted packets, extracting information about their delivery over the network-under-test (e.g. from packets headers), and storing this information in certain "dump" files. Finally, the information stored in these "dump" files, was processed by utilising the TCPTrace protocol analyzer [88], providing results regarding the instantaneous useful throughput [89], round trip time (RTT) [90], and retransmitted packets.

It should be noted that during these experiments the Best-effort scheme was adopted, the TCP BIC congestion control algorithm was utilised (as the default mechanism in Linux operating systems), and the TCP auto-tuning option was also selected for an automatic adjustment of the advertised TCP window size according to the bandwidth-delay product [91].

3.3.1.1. Access Network Performance Evaluation

The first set of experiments was designed in order to evaluate the performance of the two access networks, i.e. the communication between a user and CMN1 over IEEE 802.11g (providing a nominal bit-rate of 27Mb/s both in the downlink and in the uplink communication), and the communication between a user and CMN2 over ADSL (utilising a nominal bit-rate of 8Mb/s in the downlink path and 512Kb/s in the uplink).

IEEE 802.11g Access Network

Towards evaluating the performance of the IEEE 802.11g access network, the Iperf traffic generator was used for establishing a TCP connection between the active user (IP address 172.16.40 – see Fig. 3-3), which was placed at 100m away from CMN1 under line-of-sight conditions, and the PC Router (IP address 172.16.0.1) located within CMN1. Upon establishment of this connection the Iperf tool was generating TCP traffic between the two communicating entities. At the same time, the Tcpdump application (running both at the user and at the PC router), was capturing the headers of the transmitted/received data packets, besides storing them as "dump" files. These tests were lasting for a period of 180s, which was sufficient for the network-under-test to reach a stable state. Upon completion of the TCP connection, the data stored within the "dump" files were collected and analysed by the TCPTrace tool, providing results regarding the instantaneous useful throughput, round trip time (RTT), and retransmitted packets. As already mentioned, TCP BIC congestion control algorithm was utilised, and TCP auto-tuning was selected. The experimental results obtained for this access network performance, indicated an average instantaneous useful throughput of

22.93Mb/s, an average RTT of 32.29ms, while no retransmissions occurred (as a matter of un-congested network conditions). Graphical representation of these experimental results versus time is depicted in Fig. 3-5 and Fig. 3-6 respectively, during the time period of 180s.

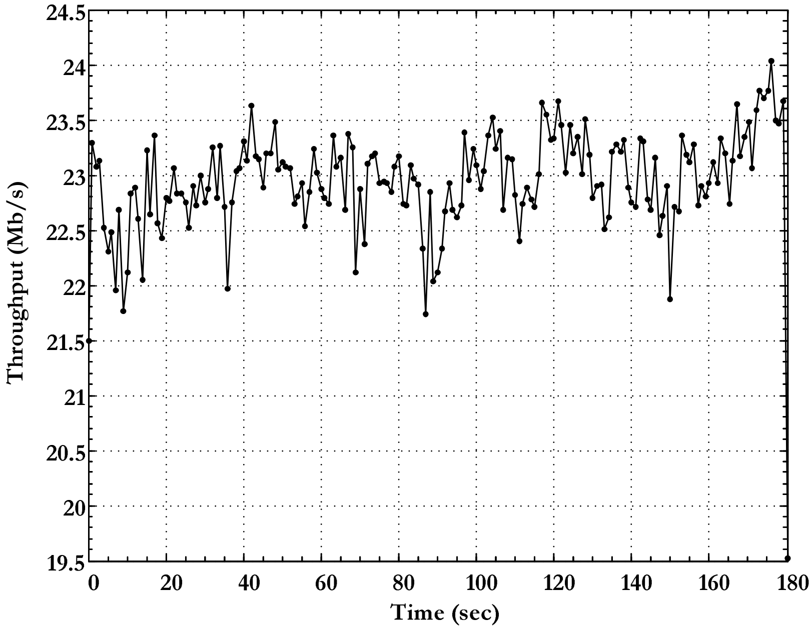


Fig. 3-5 Useful throughput of IEEE 802.11g based access network

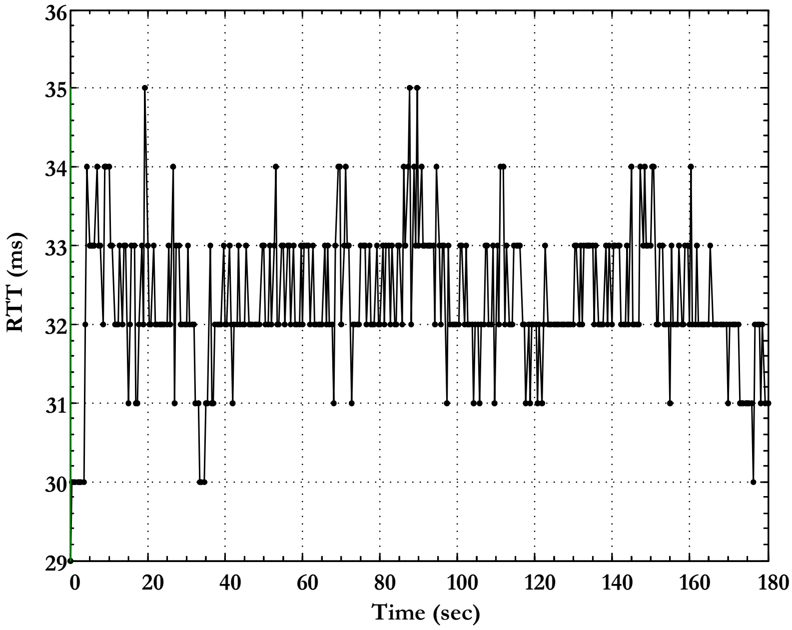


Fig. 3-6 Round trip time of IEEE 802.11g based access network

<u>**ADSL Access Network**</u>

Similarly as for the IEEE 802.11g access network, the evaluation process of the ADSL network utilised the Iperf traffic generator for establishing the TCP connection between an end user (IP address 192.168.0.22 – see Fig. 3-4), which was placed at about 800m away from

CMN2 (local PSTN exchanger), and the PC Router (IP address 192.168.0.1) located within CMN2. Upon establishment of this connection the Iperf tool was generating TCP traffic between the two communicating entities, while at the same time, the Tcpdump application was capturing the headers of the transmitted/received data packets. These were stored in "dump" files both at the user and at the PC router. The experimental results provided by the TCPTrace tool, indicated an average useful throughput of 6.98Mb/s and an average RTT of 297.2ms. No retransmissions were also observed, as a matter of the un-congested network conditions. Graphical representation of these experimental results versus time is depicted in Fig. 3-7 and Fig. 3-8 respectively, during the time period of 180s, which was sufficient for the network-under-test to reach a stable state.

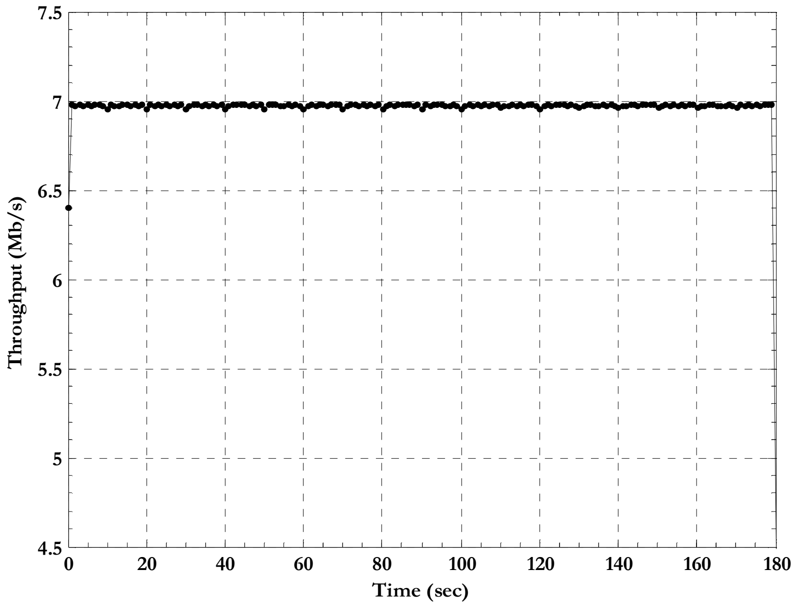


Fig. 3-7 Useful throughput of ADSL based access network

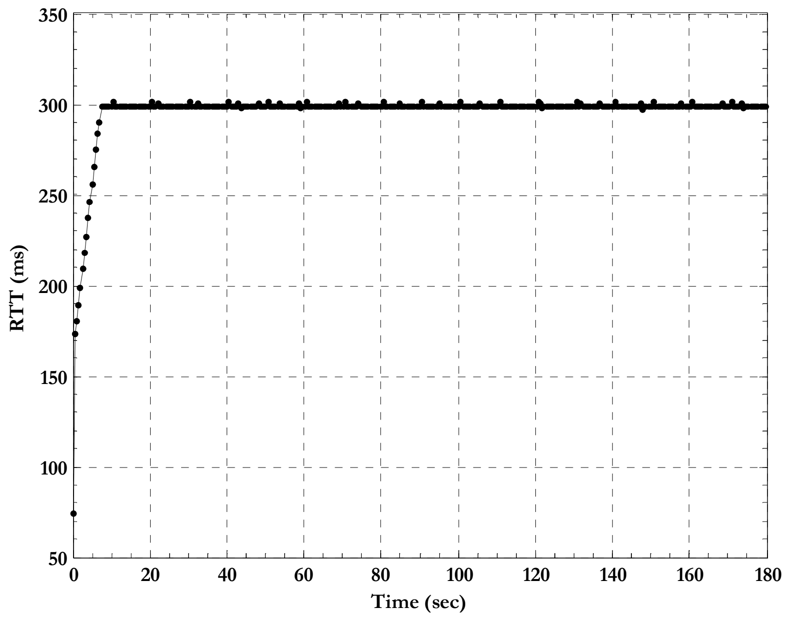


Fig. 3-8 Round trip time of ADSL based access network

3.3.1.2. Performance Evaluation of the Uplink-Downlink Chain

The next sets of experiments were designed in order to evaluate the performance of the uplink-downlink communication chains, i.e. between each CMN and the regenerative DVB-T. Towards these, similar methodology as the one used for the evaluation of the access network, was followed. It should be also noted that during these preliminary experiments the IP channel of 8Mb/s (devoted to TCP data traffic in the DVB-T stream), was un-congested, i.e. no other traffic was present in the TCP channel than the one generated during these tests. The overall testbed architecture and setup during this set of experiments is depicted in Fig. 3-9, exploiting either ISDN technology (in the case of CMN2) or IEEE 802.11g technology (in the case of CMN1) for the uplink.

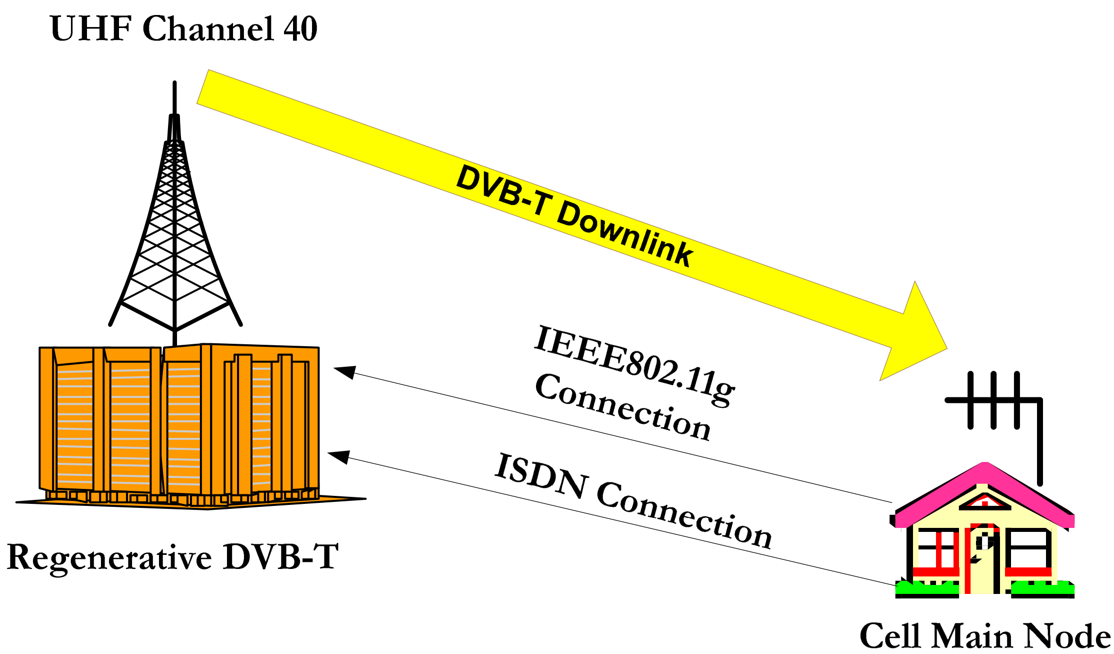


Fig. 3-9 Networking architecture of the uplink-downlink chain (ISDN and IEEE 802.11)

<u>ISDN Uplink – DVB-T Downlink</u>

The first set of tests was carried out between the PC Router at CMN2 (IP address 10.0.67.87 – see Fig. 3-4) and the Measurements Server (IP address 10.0.67.11 – see Fig. 3-2) at the regenerative DVB-T platform. TCP reverse data traffic (i.e. requests/acknowledgments from CMN2 to the regenerative DVB-T platform) was being carried over the ISDN link, while the TCP forward data traffic was provided through the UHF transmission channel. The obtained results (provided by the TCPTrace tool) indicated an average useful throughput of 4.54Mb/s and an average RTT of 124.48ms, as a matter of the high asymmetry of the uplink-downlink chain. Graphical representations of the average useful throughput and the average RTT are depicted in Fig. 3-10 and Fig. 3-11 respectively, for a time period of 180s, which was sufficient for the network-under-test to reach a stable state. No packet retransmissions occurred, as a result of the un-congested DVB/IP channel.

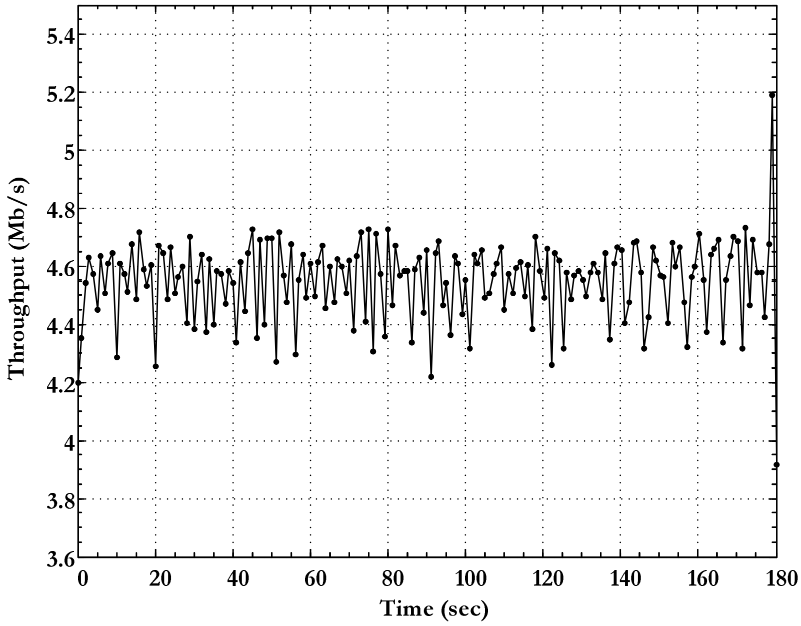


Fig. 3-10 Useful throughput of the uplink-downlink chain (ISDN case)

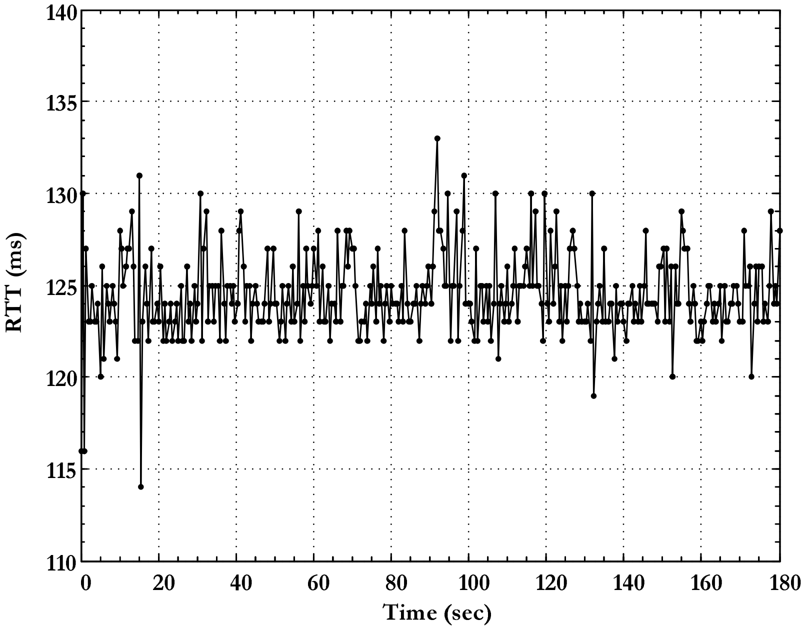


Fig. 3-11 Round trip time of the uplink-downlink chain (ISDN case)

<u>IEEE 802.11g Uplink – DVB-T Downlink</u>

The second set in these experiments was carried out for evaluating the performance of the uplink-downlink chain between the PC router at CMN1 (IP address 10.0.67.84 – see Fig. 3-3) and the Measurements Server at the regenerative DVB-T platform (IP address 10.0.67.11), exploiting the IEEE 802.11g technology only for TCP reverse data traffic and the DVB-T stream for the downlink. The results provided by the TCPTrace tool indicated an average useful throughput of 7.42Mb/s (see Fig. 3-12), and an average RTT of 78.36ms (see Fig. 3-13), while no retransmissions occurred, as a result of the un-congested DVB/IP channel.

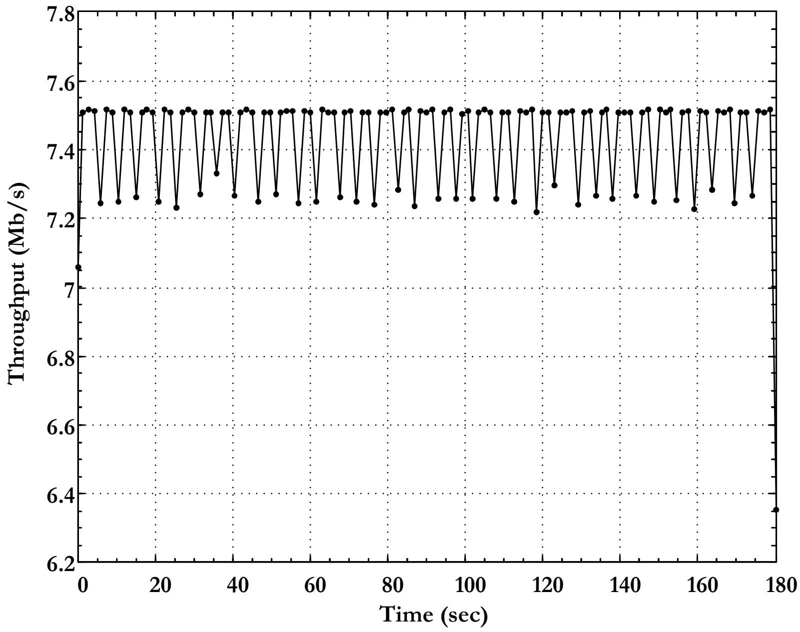


Fig. 3-12 Useful throughput of the uplink-downlink chain (IEEE 802.11g case)

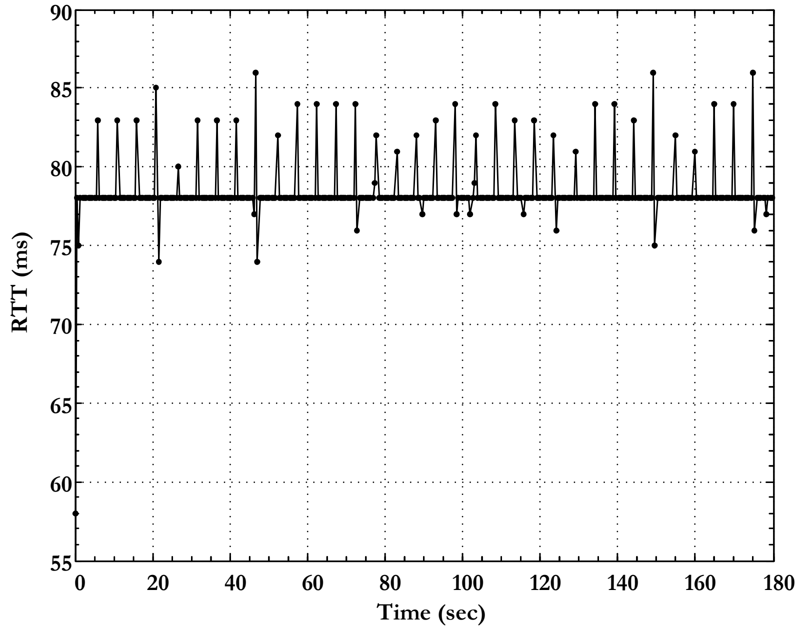


Fig. 3-13 Round trip time of the uplink-downlink chain (IEEE 802.11g case)

3.3.1.3. Performance Evaluation of the Communication between a User and the Regenerative DVB-T Platform

The next sets of experiments were conducted towards evaluating the performance of the communication network between a user and the regenerative DVB-T platform. The overall testbed architecture and setup during this set of experiments is depicted in Fig. 3-14, where IEEE 802.11g technology is exploited both in the access and the uplink networks when the user is located at CMN1, while ADSL and ISDN technologies are exploited, in the access and the uplink respectively, when the user is located CMN2. As in the previous sub-section, the total available bandwidth of the common DVB-T stream (i.e. 21.11Mb/s) was similarly exploited, providing for 8Mb/s for TCP traffic. However, this IP channel was un-congested during these tests, i.e. no other traffic was present in the TCP channel than the one generated during these tests.

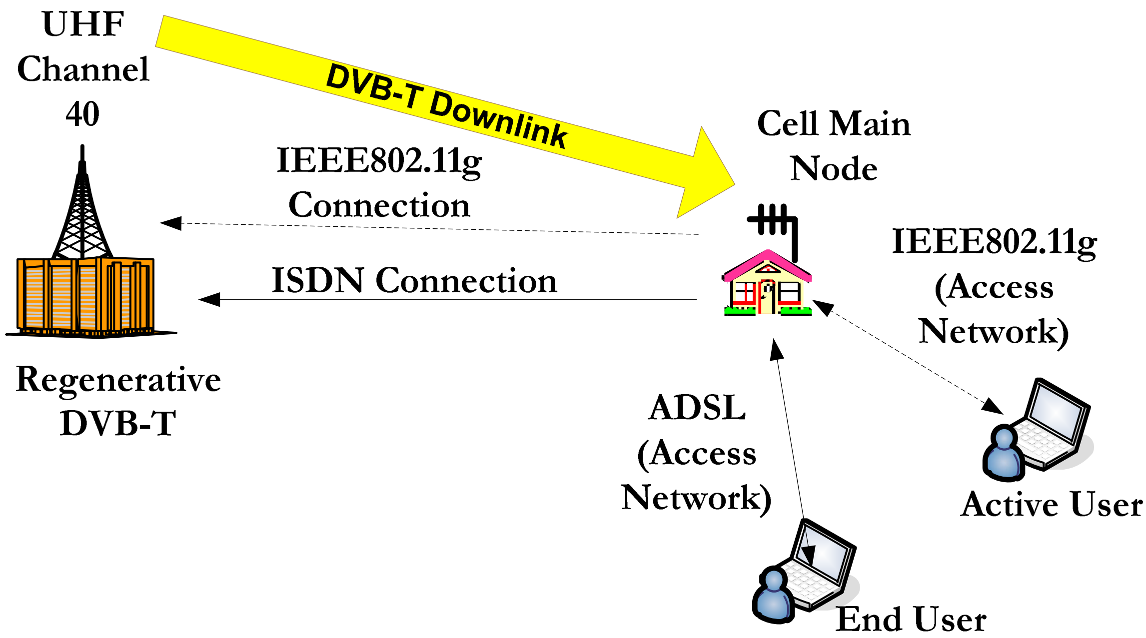


Fig. 3-14 Networking architecture of the communication between a user and the regenerative DVB-T platform

User at CMN1

The performance evaluation experiments for CMN1 exploited the Iperf traffic generator for establishing a TCP connection between the active user (IP address 172.16.0.40) and the Measurements Server (IP address 10.0.67.11) at the regenerative DVB-T platform. Upon establishment of this connection the Iperf tool was generating TCP traffic between the two communicating entities. At the same time, the Tcpdump application (running both at the active user's terminal and at the Measurements Server), was capturing the headers of the transmitted/received data packets, besides storing them as "dump" files. Upon completion of the TCP connection, the data stored within the "dump files" were collected and analysed by the TCPTrace tool, providing results regarding the instantaneous useful throughput, round trip time (RTT), and retransmitted packets. The experimental results, provided by the TCPTrace tool, indicated an average instantaneous useful throughput of 7.42Mb/s (see Fig. 3-15), an average RTT of 78.27ms (see Fig. 3-16), while no retransmissions occurred, as a result of the un-congested DVB/IP channel.

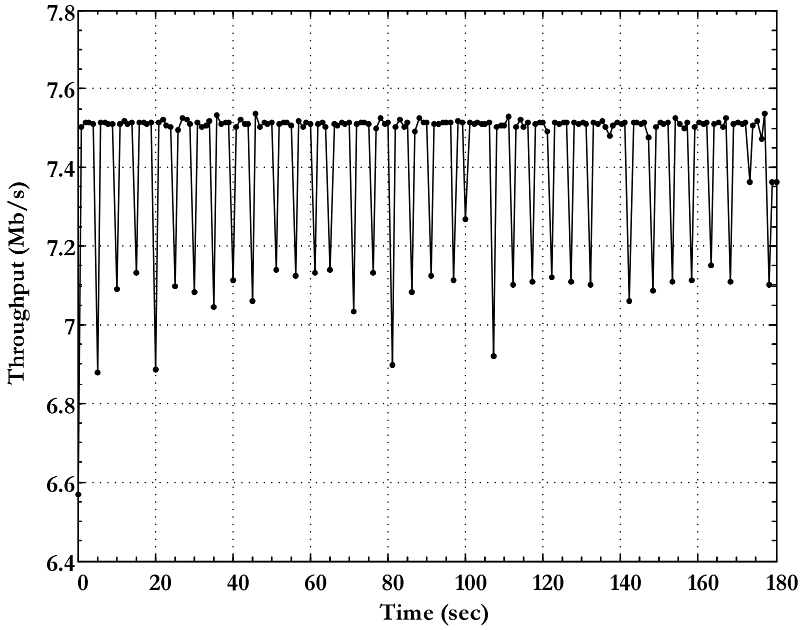


Fig. 3-15 Useful throughput of the network connection between the active user and the regenerative DVB-T platform

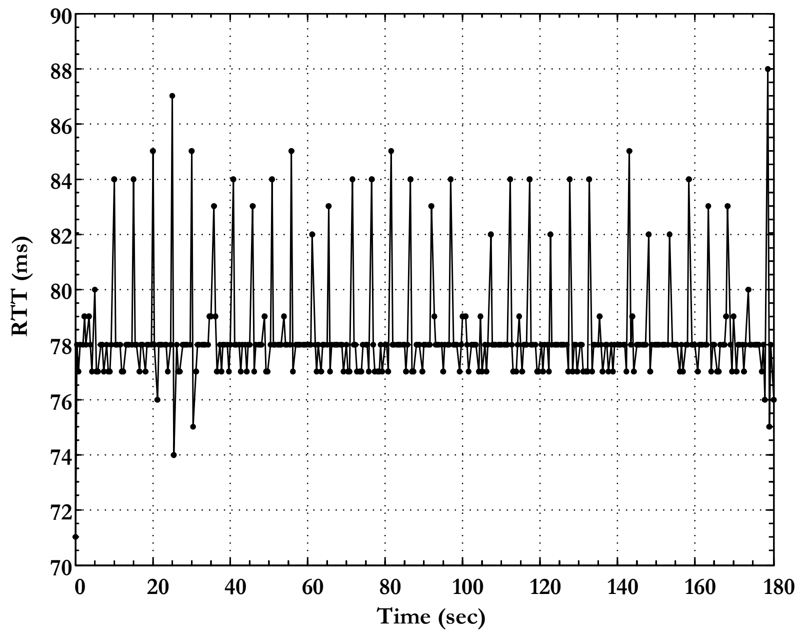


Fig. 3-16 Round trip time of the network connection between the active user and the regenerative DVB-T platform

User at CMN2

Similar tests were also conducted for the communication path between an end user located at CMN2 (IP address 192.168.0.22) and the Measurements Server (IP address 10.0.67.11) at the regenerative DVB-T platform. It should be mentioned that in this communication, the TCP reverse data traffic (i.e. requests/acknowledgments), originated by the end user's terminal, are forwarded to CMN2 through the ADSL based access network and routed to the regenerative DVB-T platform through the one way ISDN based uplink. The TCP reply data traffic was provided over the DVB-T downlink to CMN2, where it was distributed back to the end user's terminal through the ADSL based access network. The obtained results indicated an average instantaneous useful throughput of 4.53Mb/s (see Fig. 3-17) and an average RTT of 128.83ms (see Fig. 3-18), while no retransmissions occurred, as a result of the un-congested DVB/IP channel.

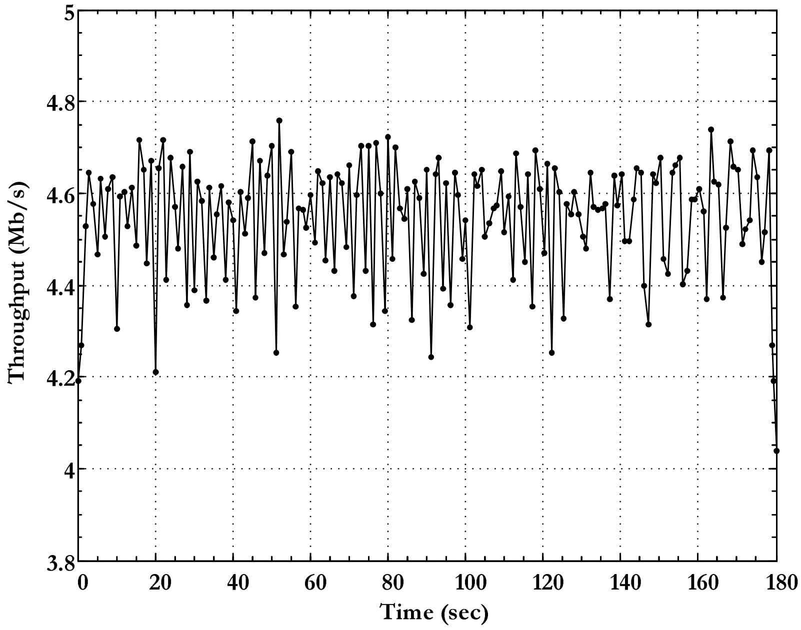


Fig. 3-17 Useful throughput of the network connection between the end user and the regenerative DVB-T platform

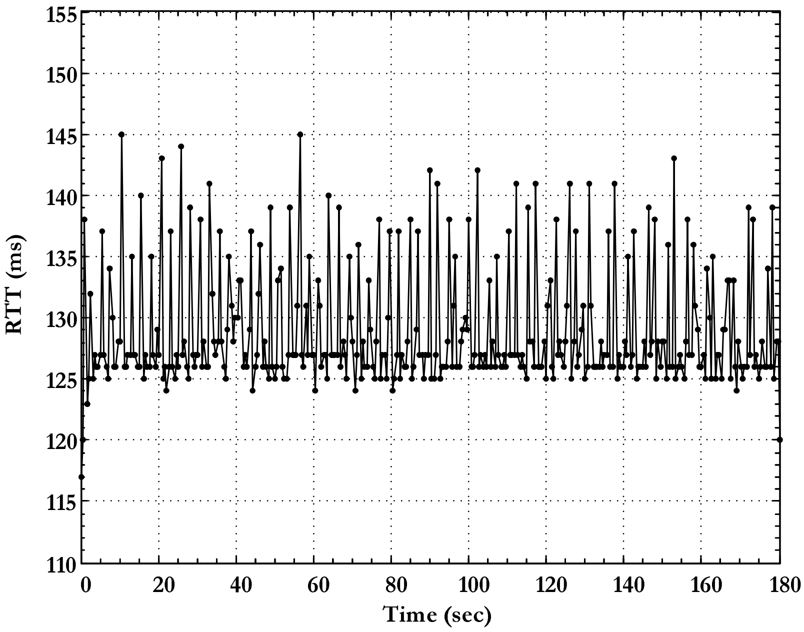


Fig. 3-18 Round trip time of the network connection between the end user and the regenerative DVB-T platform

3.3.1.4. End-to-end Network Performance Evaluation based on TCP Traffic

Performance during Single Access

The next set of experimental tests was conducted in order to evaluate the network performance based on TCP traffic for an end-to-end communication, i.e. when a user located in CMN2 is accessing content hosted by the active user located in CMN1. The overall system architecture regarding these experimental tests is depicted in Fig. 3-19. As in the previous sub-sections, the total available bandwidth of the common DVB-T stream (i.e. 21.11Mb/s) was similarly exploited, providing for 8Mb/s for TCP traffic. It should be noted that during these experiments, this IP channel was un-congested, i.e. no other traffic was present in the TCP channel of the DVB-T stream than the one generated during these tests.

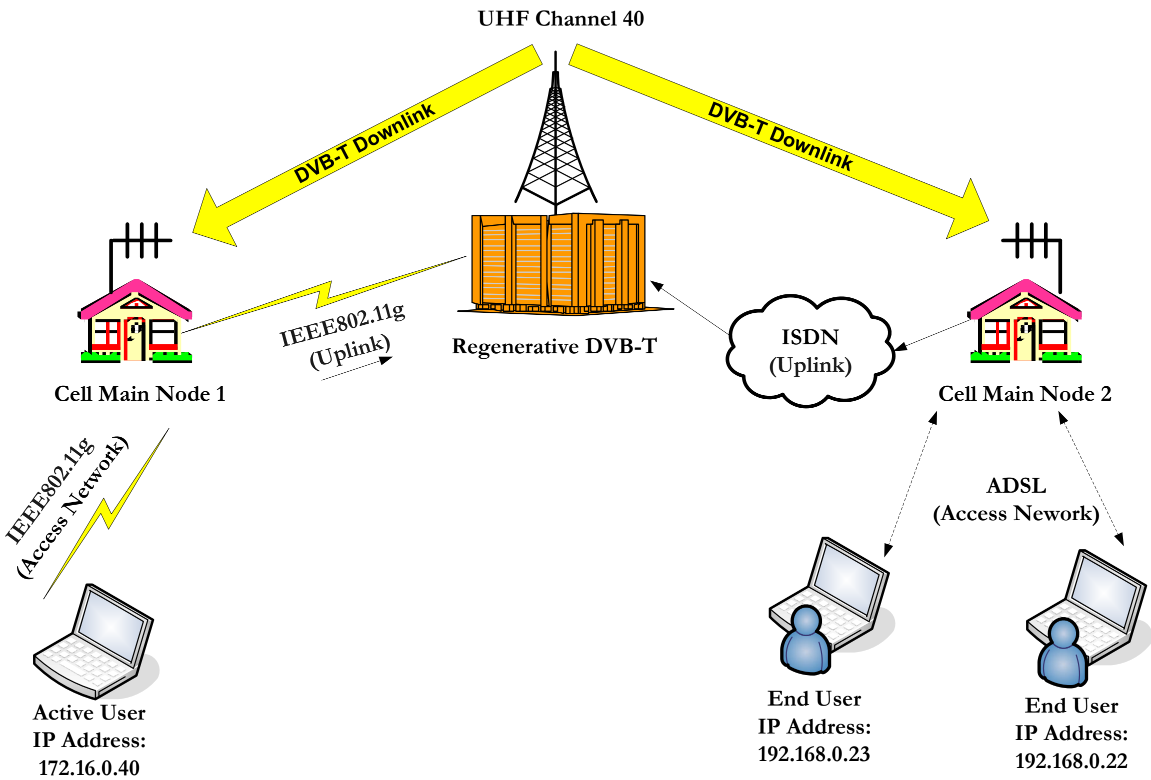


Fig. 3-19 Networking architecture for end-to-end communication

Similarly as in the previous tests, the Iperf traffic generator was used for establishing the TCP connection between an end user's terminal (IP address 192.168.0.22) at CMN2, and the active user's terminal (IP address 172.16.0.40) located within CMN1. Upon establishment of this connection, the Iperf tool was generating TCP traffic (i.e. requests/acknowledgements), which was received by the IP-DSLAM and forwarded to the regenerative DVB-T platform over the ISDN uplink. At the regenerative DVB-T platform, it was encapsulated in the common DVB-T stream by the IP/DVB gateway, prior to be broadcasted to the entire DVB-T coverage area together with other information (e.g. digital television programs). Finally, this traffic was received by CMN1 via a COFDM receiver, which routed it to the active user's terminal through a point-to-multipoint wireless link (i.e. IEEE 802.11g connection). In turn, the TCP reply data traffic that was originated by the active user's terminal, was forwarded to CMN1 through the wireless access network, where CMN1 was then in charge of forwarding it to the regenerative DVB-T platform utilising the one-way point-to-point connection (i.e. IEEE

802.11g connection). At this point, the TCP reply data traffic was encapsulated in the IP/DVB gateway and was broadcasted to the entire coverage area according to the DVB-T standard. Finally, the reply data traffic was received by a COFDM (at CMN2) and forwarded to the end user's terminal through the ADSL based access network.

During this process, which was lasting 180s during each experiment (sufficient for the network-under-test to reach a stable state), the Tcpdump application was also active (running both at the end user's and active user's terminals), for capturing the headers of the transmitted/received data packets, besides storing them as "dump" files. Upon completion of the TCP connection, the data stored within the "dump" files were collected and analysed by the TCPTrace tool, providing results regarding the instantaneous useful throughput, round trip time (RTT), and retransmitted packets. The experimental results indicated an average useful throughput of 4.4Mb/s and an average RTT of 273.69ms, during this end-to-end TCP communication. Moreover, no retransmissions were observed as a result of the un-congested DVB/IP channel. Fig. 3-20 and Fig. 3-21 depict the graphical representations of the useful throughput and the RTT versus time, respectively.

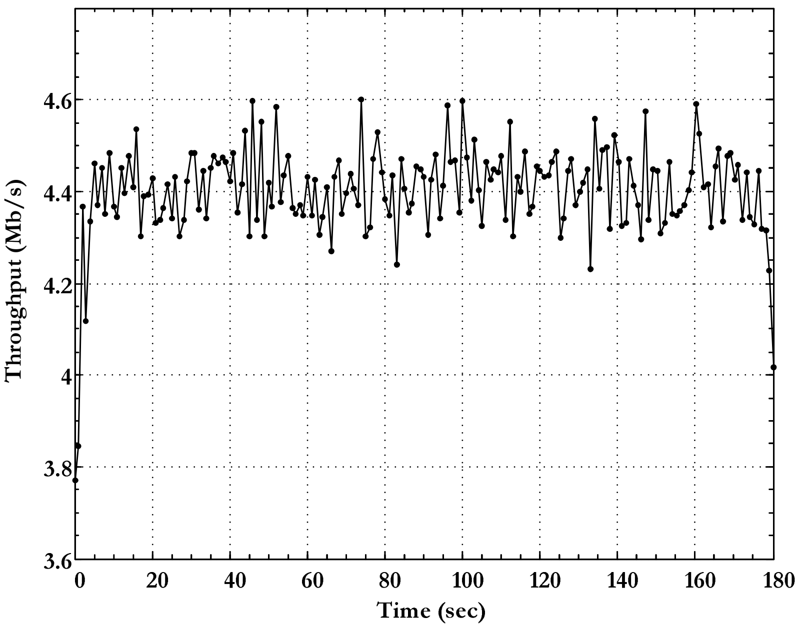


Fig. 3-20 Useful throughput for end-to-end communication

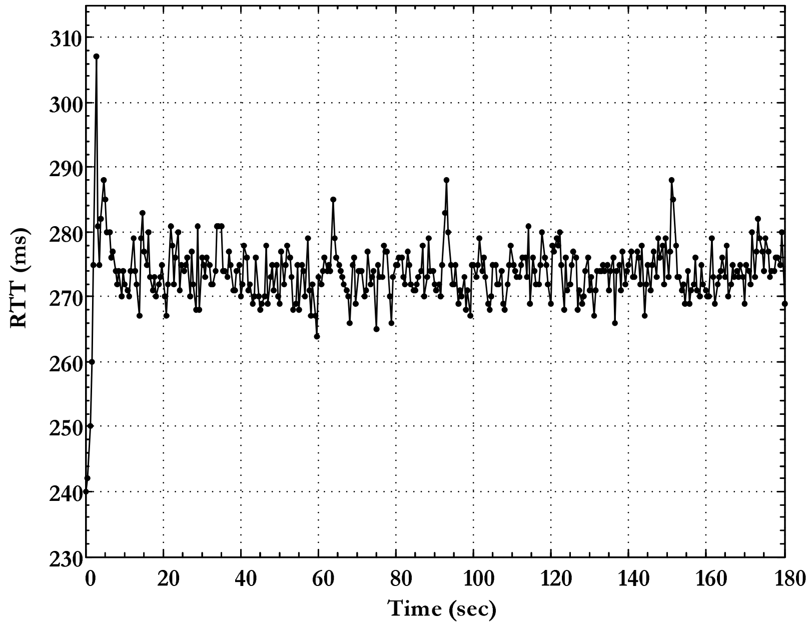


Fig. 3-21 Round trip time for end-to-end communication

Performance during Concurrent Access

Similar experiments concerning the end-to-end network performance evaluation were also conducted in the presence of two end-users (located at CMN2), i.e. when they concurrently communicate with the active user at CMN1 (see Fig. 3-19). In these experiments, two Iperf traffic generators were used: a) the first one for establishing the TCP connection between the end user's terminal with IP address 192.168.0.22 and the active user's terminal (IP address 172.16.0.40), and b) the second one for establishing the TCP connection between the end user's terminal with IP address 192.168.0.23 and the active user's terminal at CMN1. In this respect, and for the purposes of the experimental process, the two Iperf generators were adjusted so that the first one start generating TCP traffic for 120s, while the second starting 60s after the first one. The experimental results indicated that the useful throughput of 4.4Mb/s is evenly shared among the two end-users when they are concurrently communicating with the active one, exploiting the total available bandwidth. More specifically, the experimental results during concurrent access indicated an average useful throughput of 2.17Mb/s and 2.15Mb/s and an average RTT of 399.48ms and 400.43ms, for 1st for 2nd TCP connection respectively. A graphical representation of the network throughput when two ADSL users access the active user's terminal is depicted in Fig. 3-22, while Fig. 3-23 and Fig. 3-24 depict the corresponding RTT for 1st and 2nd TCP connections respectively.

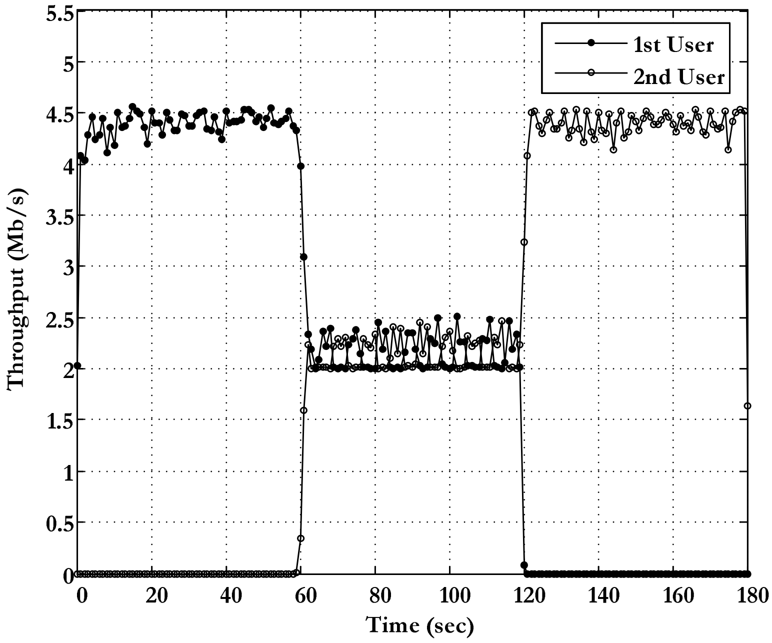


Fig. 3-22 Useful throughput for two TCP connections

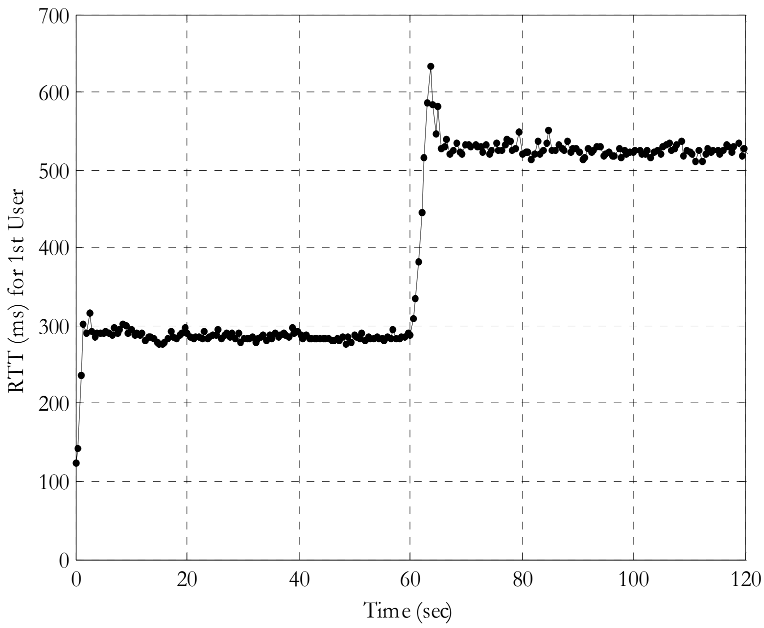


Fig. 3-23 RTT for 1st TCP connection

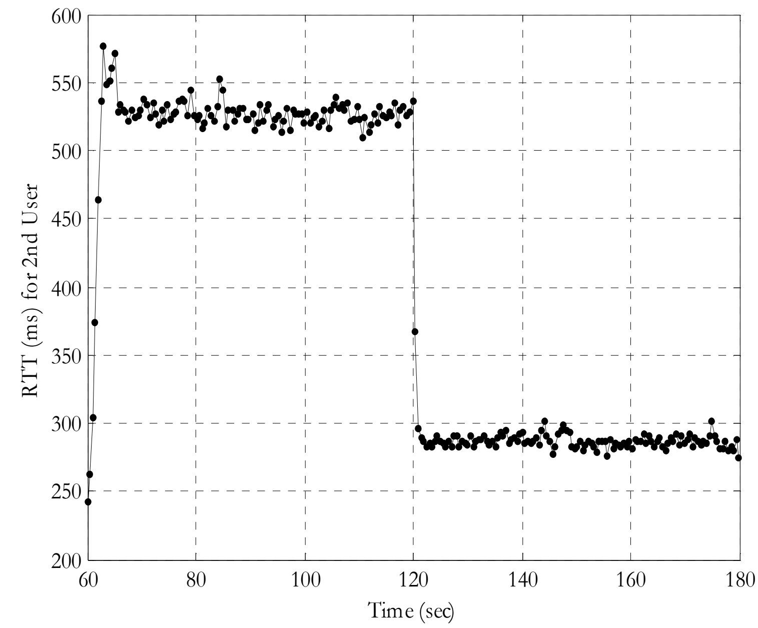


Fig. 3-24 RTT for 2nd TCP connection

Performance under Congested Network Conditions

As already stated, the experimental tests presented until now, utilised an un-congested IP channel in the common DVB-T stream devoted for TCP traffic provision (total available bit-rate of 8Mb/s), which is not, however, the case in real networking infrastructures. In this context, another set of experiments was designed in order to evaluate the end-to-end network performance under network congestion conditions. Towards this, it was initially determined the type and the characteristics of the data traffic that would congest the IP channel in the DVB-T stream. For preliminary experimental purposes, it was chosen the Burst-traffic model for delivering data packets of 1024bytes size, at given time intervals and at given bit-rates. In this respect, the MGEN tool [92] was utilised, running at the Measurements Server in the regenerative DVB-T platform, configured so that constantly providing a UDP traffic of 1Mb/s, during the entire duration of each experiment, while increasing its bit-rate to 7Mb/s every 30 seconds and for a time-interval of 5 seconds, commencing from the 10th second of the overall evaluation time. During these experiments, several TCP congestion avoidance algorithms were exploited, including the Reno, Veno, H-TCP, Vegas, Cubic, Hybla, etc, elaborating on the end-to-end network performance, in respect to the average instantaneous useful throughput, round trip time (RTT), and packet losses (as a matter of total retransmitted packets and triple duplicate acknowledgments). The experimental tests indicated variations in the average values of the above performance evaluation metrics, establishing therefore, the close relation between the overall network performance and the chosen TCP congestion avoidance algorithm. Fig. 3-25 depicts the experimental results in respect to the average instantaneous useful throughput for the each one of the utilised TCP congestion avoidance algorithms, where an optimised performance is observed when TCP Hybla is utilised. Fig. 3-26 depicts the average RTT for the various tested TCP congestion avoidance algorithms, indicating that the exploitation of TCP Veno results in an optimised end-to-end performance, in terms of round trip time. Finally, Fig. 3-27 depicts the total retransmitted packets and triple duplicate acknowledgments for the different TCP congestion avoidance algorithms, indicating that when TCP Veno algorithm is utilised, the TCP connection experiences less total retransmitted packets.

The experimental measurements in terms of useful throughput, RTT and loss events (i.e. retransmitted packets and triple duplicate acknowledgements) for this set of tests are presented in Appendix (A.I).

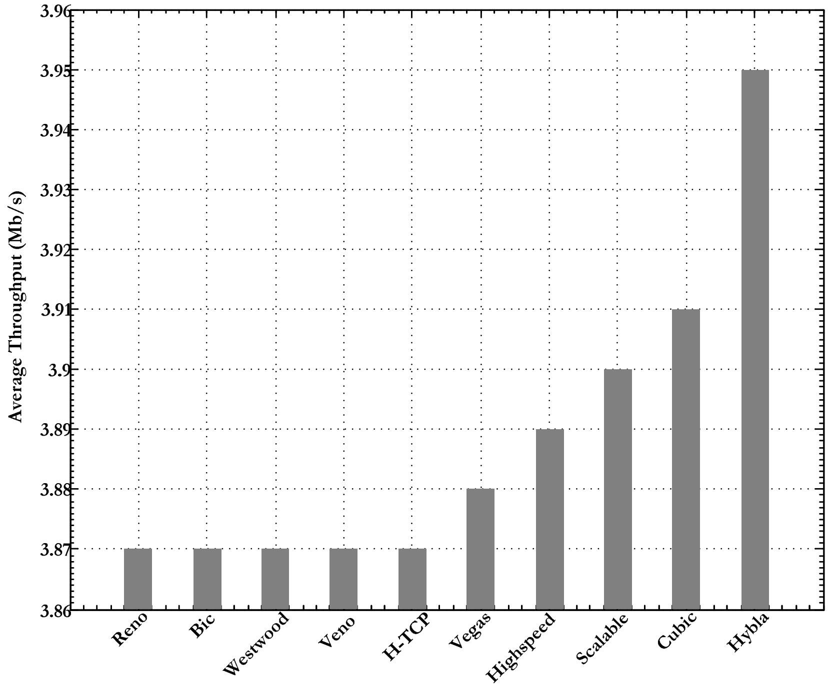


Fig. 3-25 Average useful throughput for different TCP congestion avoidance algorithms

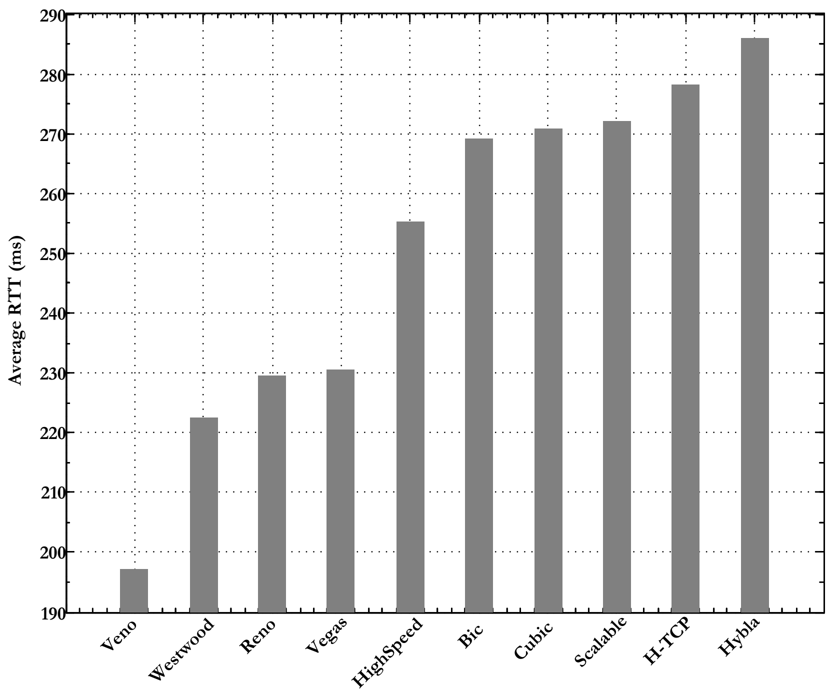


Fig. 3-26 Average round trip time for different TCP congestion avoidance algorithms

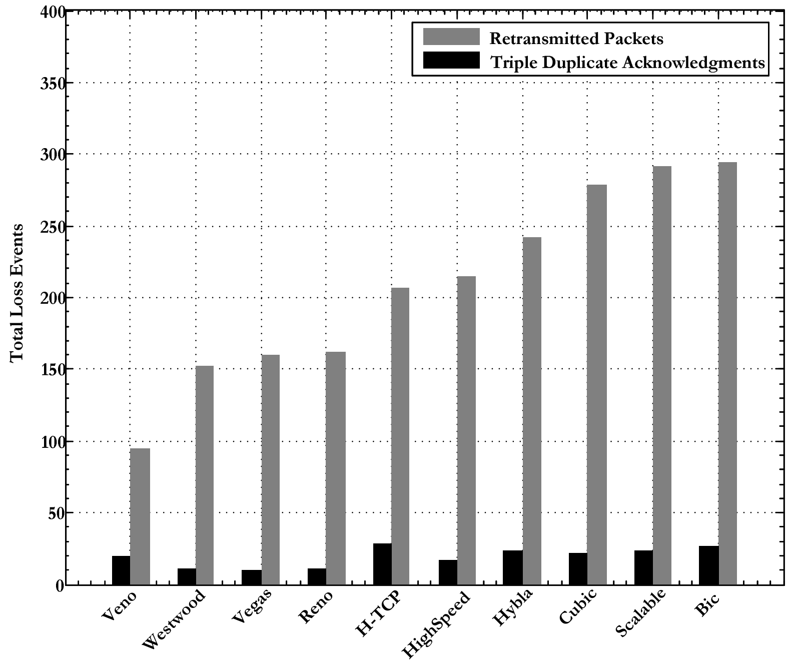


Fig. 3-27 Total retransmitted packets and triple duplicate acknowledgments for different TCP congestion avoidance algorithms

3.3.1.5. End-to-end Network Performance Evaluation for Different IP/DVB Gateway Bandwidth Allocation for TCP Traffic

Another critical factor that strongly affects the overall network performance during an end-to-end communication is the IP/DVB encapsulation process, during which the IP datagrams are fed-onto the MPEG-2 transport stream packets (MPE). In other words, the time needs for encapsulating TCP packets in MPEG-2 ones is directly related to the bandwidth allocated each time for IP services in the common DVB-T stream. Until now, the IP-channel within the common DVB-T downlink was being allocated a total bandwidth of 8Mb/s for TCP traffic, while the remaining 13Mb/s were distributed between digital TV programmes (MPEG-2) and UDP traffic. This set of experiments elaborates on the overall end-to-end system performance when more bandwidth is allocated to MPEG-2 services and less to TCP traffic, as a matter of current services priority in DVB-T systems.

In this respect, a set of experimental tests was designed in order to evaluate the end-to-end network performance when different bandwidth is allocated for TCP traffic, starting from 1Mb/s up-to 8Mb/s in steps of 1Mb/s, in terms of useful throughput and round trip time. Fig. 3-28 presents the experimental results obtained for the average useful throughput versus the bandwidth allocated in the IP/DVB gateway for TCP traffic, and indicates that for rates above 6Mb/s the network performance does not change significantly (useful throughput about 4.4Mb/s). On the other hand, Fig. 3-29 presents the average round trip time versus the bandwidth allocated in the IP/DVB gateway for TCP traffic. The obtained experimental results indicated an RTT of about 271.3ms when 8Mb/s are allocated, up-to 1.37s when 1Mb/s are allocated for TCP traffic.

The experimental measurements in terms of useful throughput and RTT for this set of tests are presented in Appendix (A.II).

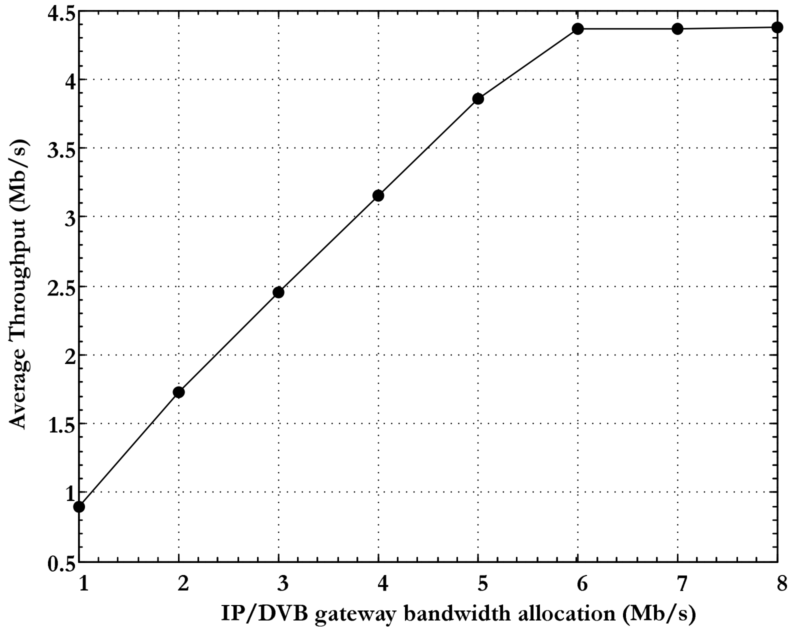


Fig. 3-28 Average useful throughput for different IP/DVB gateway bandwidth allocation for TCP traffic

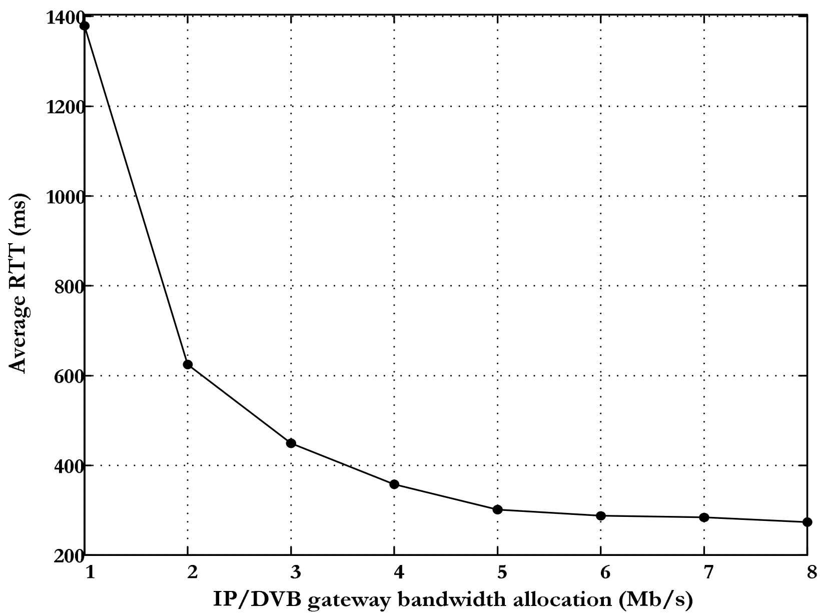


Fig. 3-29 Average round-trip time for different IP/DVB gateway bandwidth allocation for TCP Traffic

3.3.2. Network Performance Evaluation based on UDP Traffic

The second class of experiments was designed in order to evaluate the capacity of the proposed prototype in providing multicast (one-way) multimedia services, especially when time/delay sensitive services with strict QoS requirements are utilised. In this respect, performance evaluation experiments were designed for IP-multicast services provision, such as IPTV and IP-Radio, in an end-to-end communication basis, i.e. when a user located within CMN2 is accessing/receiving multicast IP-services provided by the ISP/IP multicaster. Towards these, UDP data traffic was used, for emulating the multicast services, while inter-arrival jitter [93], [94], one way delay [95] and total packet losses [96] were selected as the QoS evaluation criteria/metrics [97]. It should be noted that, as in the previous class of experiments for TCP traffic, the total available bandwidth of the common DVB-T stream (i.e. 21.11Mb/s) was exploited as follows:

- 11Mb/s were allocated for three digital television programs (i.e. MPEG-2 live and non-live broadcasts),
- 8Mb/s were allocated to TCP traffic (i.e. for on-demand services provision), and
- 2Mb/s were allocated for UDP traffic (i.e. IP-multicast streams stemming from the active users).

It should be also noted that the IP channel of 2Mb/s (devoted to UDP data traffic in the DVB-T stream), was un-congested, i.e. no other traffic was present in the UDP channel than the one generated during these tests. Performance evaluation under congested network conditions is performed in chapter 4.

The overall architecture of the testbed set is depicted in Fig. 3-30, while the configuration of the ISP/IP multicaster is presented in Fig. 3-31.

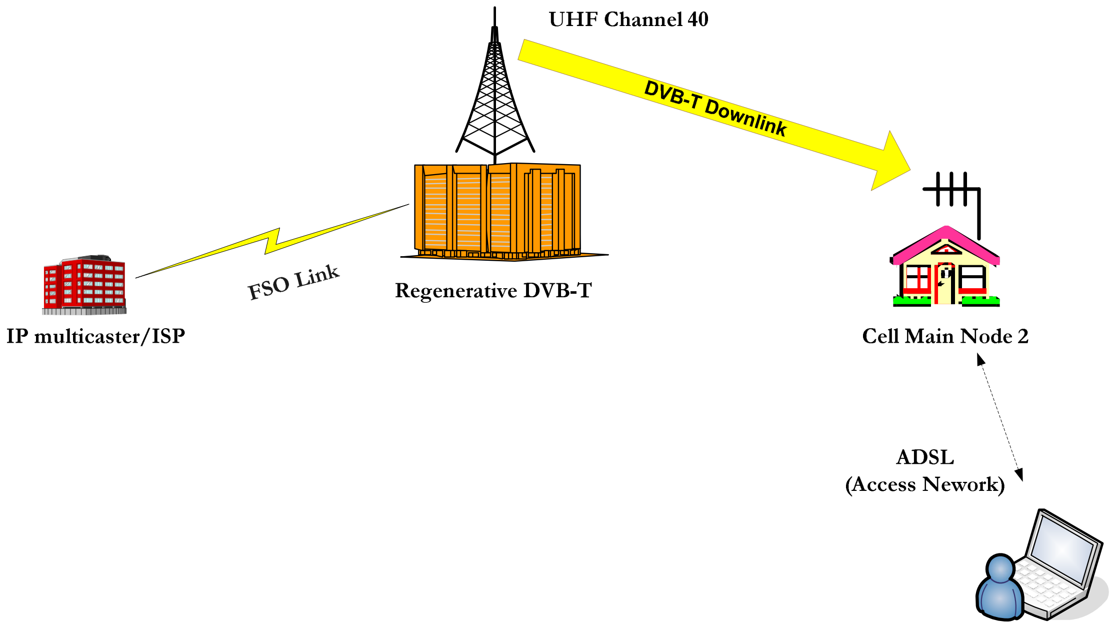


Fig. 3-30 Experimental architecture of the testbed for end-to-end communication based on UDP traffic

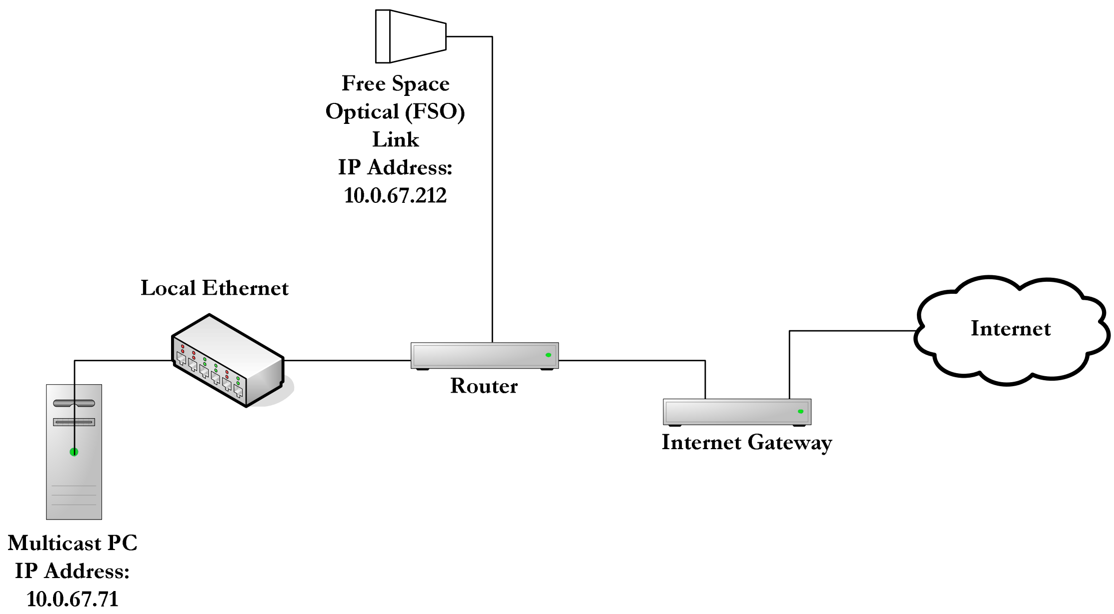


Fig. 3-31 Configuration of the ISP/IP multicaster

In these experiments, the MGEN traffic generator was used for creating a UDP stream, stemming from the Multicast PC of the ISP (IP address 10.0.67.71). This stream was received by the Router at the IP-multicaster's premises and forwarded to the regenerative DVB-T over the FSO link (IP address 10.0.67.212). At the regenerative DVB-T platform, it was encapsulated in the common DVB-T stream by the IP/DVB gateway, prior to be broadcasted to the entire DVB-T coverage area, together with the other services (i.e. digital television programs). Finally, this traffic was received by CMN2 via a COFDM receiver, which forwarded it to the end user's terminal through the ADSL network. The Tcpdump application was used (running both at the end user's terminal and Multicaster PC) for obtaining information concerning the transported packets (i.e. capturing the headers of the transmitted/received data packets), besides storing them as "dump" files. Upon completion of the UDP session, the stored data were processed and analysed, in respect to inter-arrival jitter, one way delay and packet losses. As absolute synchronisation, between the clocks of the Multicast PC and the end user's terminal, was required for obtaining the one way delay results. Network Time Protocol (NTP) [98] was utilised for this purpose.

In this context, the first experimental test utilised one UDP stream of 1024bytes packet size (default value in the MGEN tool) running at 1Mb/s (CBR) for 180s, which was delivered to the entire infrastructure (broadcasting footprint) via the 2Mb/s IP channel of the DVB-T bouquet. The obtained experimental results indicated an average inter-arrival jitter of 6.18ms resulting in good QoS [99]. Fig. 3-32 depicts the graphical representation of inter-arrival jitter during the 180s evaluation period.

On the other hand, the results indicated a maximum one way delay of 52.83ms, a minimum of 22.51ms, and an average of 30.73ms, which, according to [100], results in good quality of service. Fig. 3-33 is the graphical representation of one way delay versus time. Finally, no packet losses occurred during this experimental process, as a result of the un-congested IP-channel in the DVB-T stream.

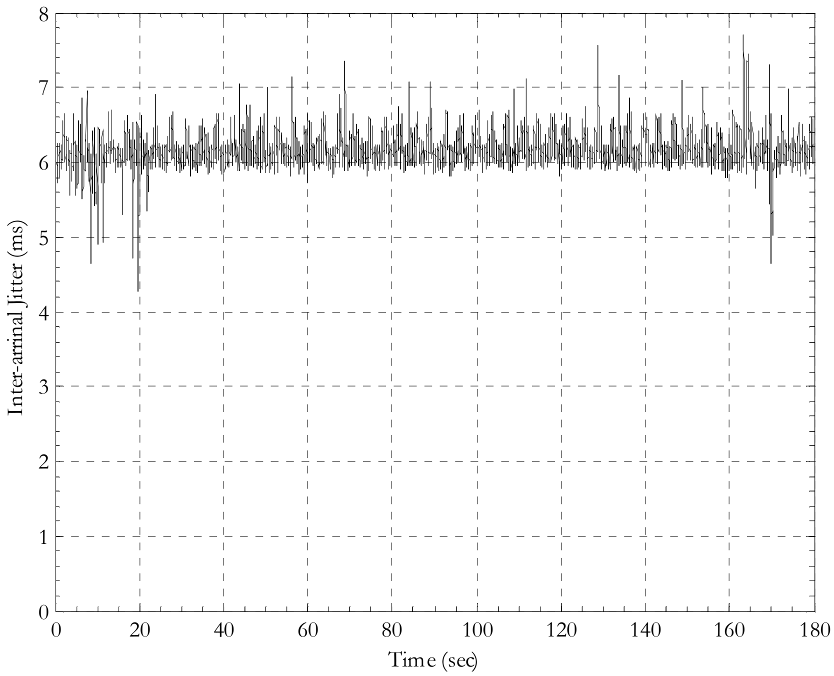


Fig. 3-32 Inter-arrival jitter

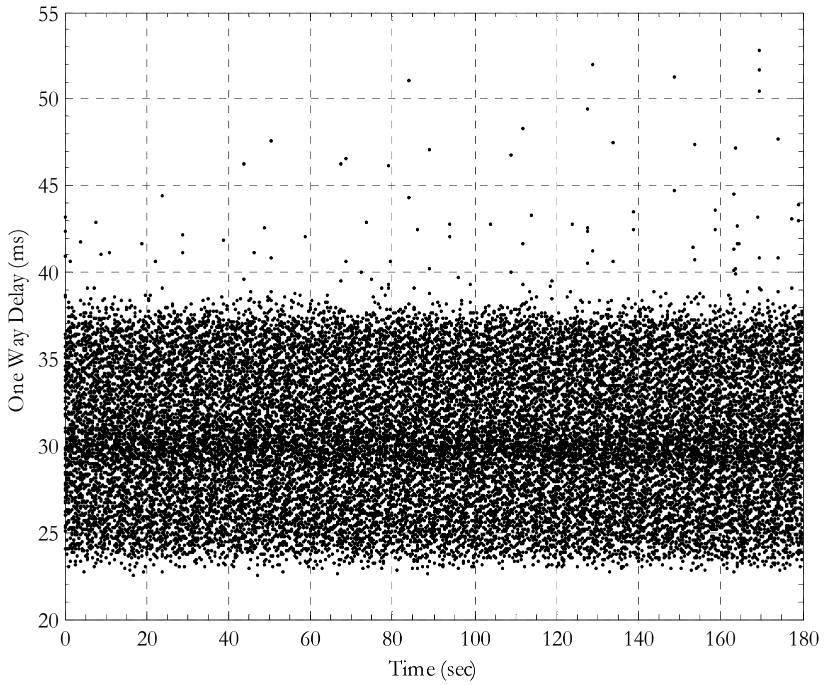


Fig. 3-33 One way delay

Towards evaluating the testbed performance under various multicast traffic parameters, a number of similar experiments were conducted, utilising each time a UDP stream of different characteristics, i.e. bit rate and packet size. Following realistic IPTV services scenarios (as the most challenging delay-sensitive service) and based on the configuration characteristics of the prototype, it was chosen to experiment on bit-rates from 700Kb/s (as a matter of the IPTV stream rates in most ADSL networks) to 1.5Mb/s (as a matter of the DVB/IP channel constraints where a maximum of 2Mb/s was available for UDP traffic). Also UDP packet size was altered from 512bytes up to 1512bytes, examining in this way the proposed system network performance under different UDP stream parameters.

Fig. 3-34 depicts the average values obtained for the inter-arrival jitter, when the UDP stream was utilising bit-rates from 700Kb/s up to 1500Kb/s and packet sizes from 512bytes up to 1512bytes, indicating a minimum of 2.9ms and a maximum of 6.4ms one. On the other hand, Fig. 3-35 depicts the average values of one way delay, versus different bit-rates and packet sizes of the provided UDP streams. It was indicated that the average one way delay varied from 27.55ms (i.e. minimum value) up to 35.87ms (i.e. maximum value), which provide for good QoS [99], [100]. It should be also noted that no packet losses occurred during all these experimental tests, as long as the IP channel in the DVB-T stream was not congested.

Combining the results obtained for inter-arrival jitter, one way delay and packet losses, it comes that delay-sensitive services can be efficiently delivered over the prototype (a matter, of course, of the given network and traffic conditions).

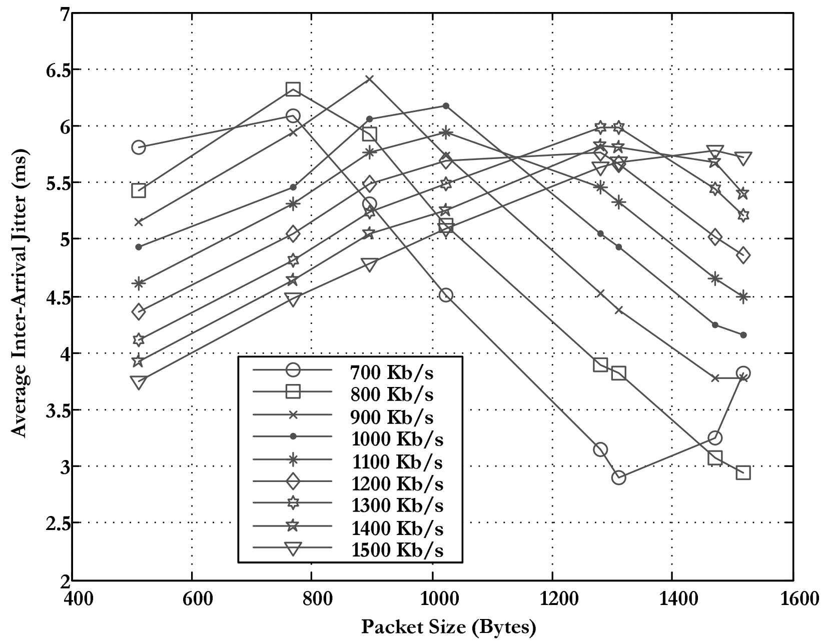


Fig. 3-34 Average inter-arrival jitter for different UDP stream bandwidth and packet size

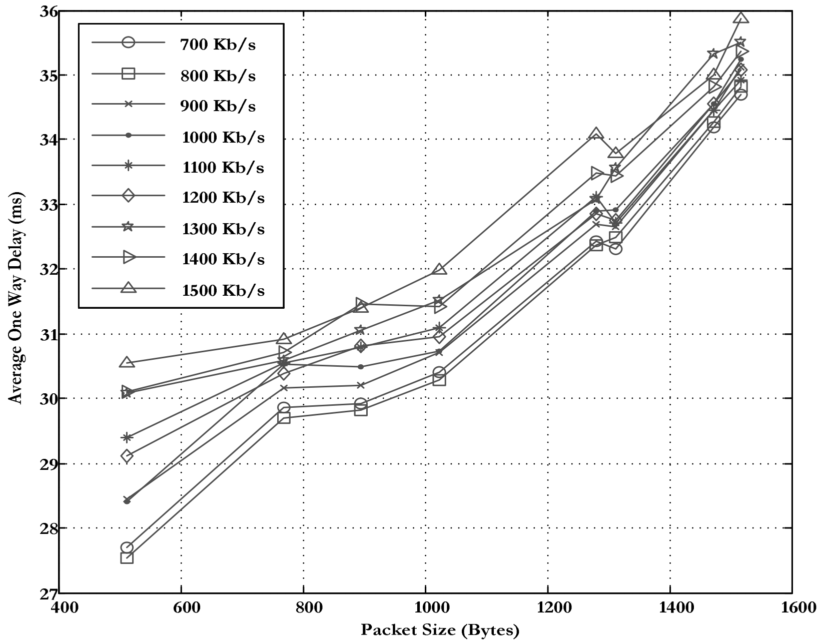


Fig. 3-35 Average one way delay for different UDP stream bandwidth and packet size

3.4. Summary

This chapter presented the realisation of a prototype that conforms to the network design specifications, as these were defined in chapter 2, which exploits the DVB-T stream in regenerative configurations for extending the core-backbone to the entire broadcasting footprint (backhaul architecture), and enabling users of rural and urban areas to be always-online (i.e. always-on connectivity) and access triple-play services carried over the same infrastructure. In this context, it presented the implementation and configuration of the regenerative DVB-T, where the common IP-backbone is created, as well as the realisation of two CMNs; one located in an urban area by utilising WLAN technology in the access network, and another CMN located in a rural metropolitan area (where only primitive PSTN/ISDN lines exist and no access/connection to the core-backbone is available), by exploiting xDSL technology in the access network. This prototype served as a testbed for verifying the validity of the proposed architecture and its capacity in providing on-demand and multicast IP services in the entire metropolitan area. In this respect, a number of preliminary experiments were designed and conducted under realistic transmission/reception conditions regarding TCP for on-demand services and UDP data traffic for multicast IP ones.

More specifically, and in the case of TCP data traffic (i.e. during on-demand services provision), a hierarchical and structured process was adopted, including the evaluation of each individual networking component and the entire infrastructure as a whole, in an end-to-end communication. Instantaneous useful throughput, round trip time (RTT) and retransmitted packets were chosen as the metrics for evaluation, under both un-congested and loaded network conditions. It was experimentally verified that the overall network performance is a matter of the configuration parameters, such as the available resources in the access network, the uplink and downlink characteristics, the bandwidth allocation in the IP/DVB gateway, as well as the utilised protocol attributes. Following the Best-effort approach, it was also verified that the system resources are fully exploited by a single user, while are evenly shared during simultaneous/concurrent access. On the other hand, the preliminary tests concerning UDP data traffic (i.e. multicast IP services provision) were conducted for an end-to-end communication, under un-congested network conditions (in the IP/DVB-T channel), taking into account the inter-arrival jitter, one way delay and total packet losses. It was experimentally verified that, for the given network configuration, a single UDP stream can efficiently be delivered over the entire infrastructure, a matter however, of the stream parameters, such as packet size and bit-rate.

Analysis of these preliminary experimental results, verified the capacity of the proposed architecture in converging Internet, telecommunications and broadcasting technologies, establishing it as an alternative/complementary solution for the realisation of broadband metropolitan networking infrastructures. Furthermore, the preliminary experimental results obtained both for TCP and UDP data traffic, established the capacity of the implemented prototype in providing on-demand and multicast IP services, and verified the validity of the proposed architecture in creating broadband metropolitan infrastructures. However, the Best-effort scheme that was used during the TCP services provision cannot provide any QoS guarantees according to each user's privileges, while the utilisation of a single UDP stream (each time) cannot establish the system's capacity in QoS provisioning, according to multicast services' QoS requirements. Towards these, Chapter 4 elaborates on the issue of QoS provisioning, according to user's privileges and services attributes, by describing the implementation of QoS aware mechanisms, and their incarnation in the implemented prototype, capable to support service differentiation and traffic prioritisation.

The research work presented in this chapter was published in [78], [79], [101].

4. PERFORMANCE OPTIMISATION

4.1. Introduction

Following the implementation of the prototype infrastructure and its preliminary performance evaluation, this chapter elaborates on the design and realisation of QoS aware mechanisms that will enhance the exploitation of the proposed architecture in broadband metropolitan infrastructures. In this respect, it elaborates on the design and implementation of DiffServ-capable modules, by analysing the rules according to which services differentiation and QoS provisioning can be obtained, and describes their deployment (in the implemented prototype) following a decentralised approach. Towards evaluating the overall system performance, and verifying its capacity in QoS provisioning, this chapter elaborates on several experiments that were conducted under realistic transmission/reception conditions, for the provision of delay-sensitive and bandwidth-dependent services (i.e. multicast audio/video). Based on these experimental results, the chapter concludes by elaborating on Perceived QoS provisioning in respect to real/actual MPEG-4 video streams, according to specific services requirements, following a subjective evaluation process.

4.2. Configuration of the QoS aware Prototype

As already established in the previous chapter, the overall network performance of the implemented prototype is a matter of the configuration parameters, such as the uplink and downlink characteristics, the bandwidth allocated to IP traffic in the DVB-T stream, etc. As a result, delay-sensitive broadband IP services such as IPTV multicasts, and prioritised access to on-demand IP-based content cannot be guaranteed, especially under congested network conditions and/or insufficient systems resources, unless QoS aware mechanisms are utilised. Services differentiation technique arises as a very promising solution, if specific QoS rules are adopted both according to user's and services' requirements. In this respect, and by making use of the design specifications derived in chapter 2, this section elaborates on the implementation of DiffServ mechanisms and the respective QoS provisioning rules, both in the core and the backhaul networks. Prior to these, however, the implemented prototype was enhanced with the appropriate hardware modules that would incorporate the DiffServ mechanisms, at the regenerative DVB-T and at each CMN. Fig. 4-1 depicts the configuration of the regenerative DVB-T platform, where a DiffServ Core Router was added, for processing, analysing, shaping and forwarding the IP-traffic according to specific QoS rules, following a Per Hop Behaviour (PHB). On the other hand, Fig. 4-2 and Fig. 4-3 are presenting the configuration of CMN1 and CMN2 respectively, where the PC Routers (as named in chapter 3 – see Fig. 3-3 and Fig. 3-4) are enhanced with DiffServ capable modules (DiffServ Edge Routers).

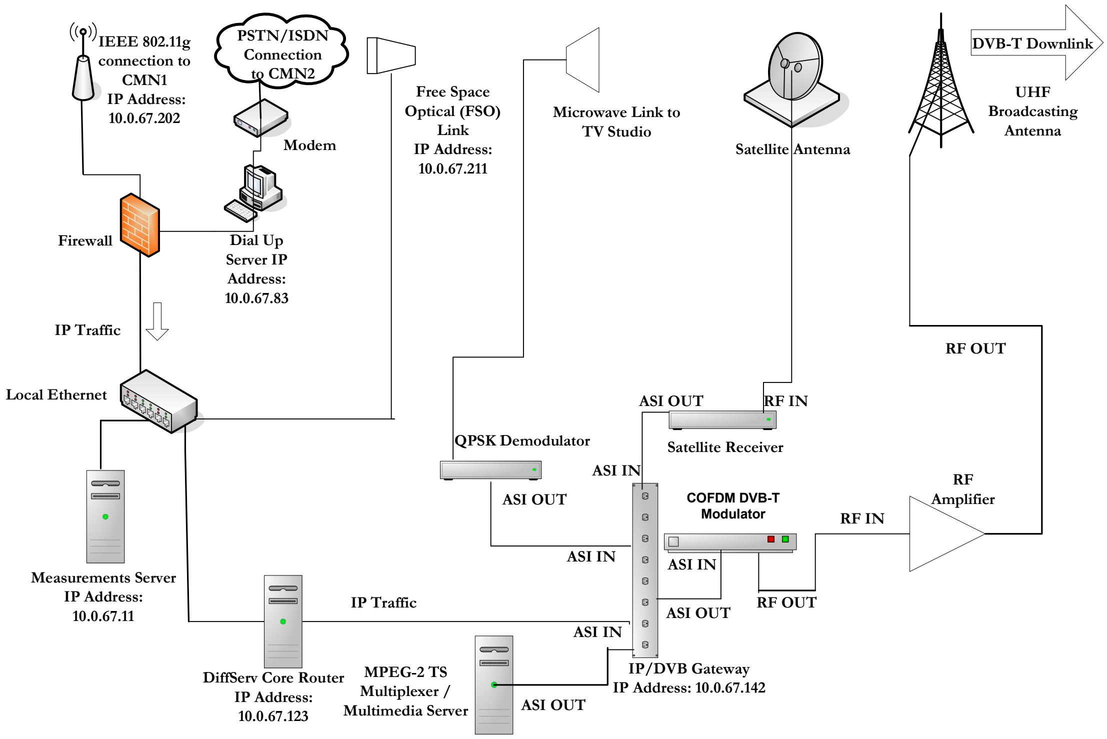


Fig. 4-1 DiffServ aware regenerative DVB-T configuration

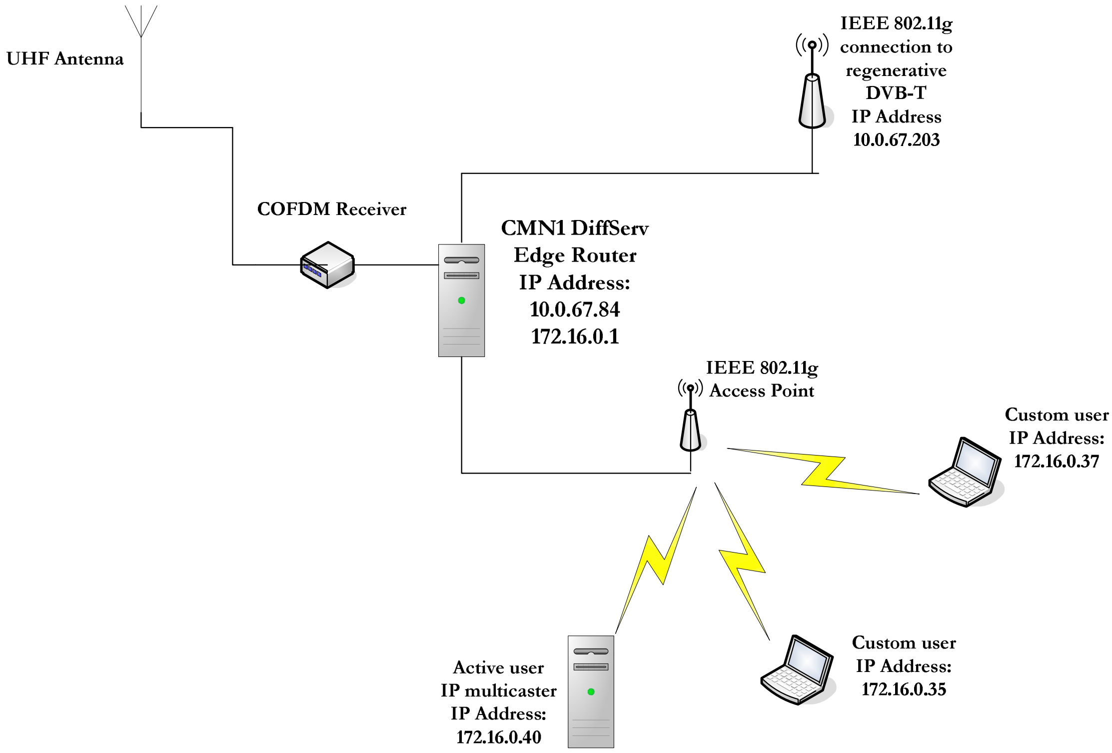


Fig. 4-2 DiffServ aware urban based CMN configuration

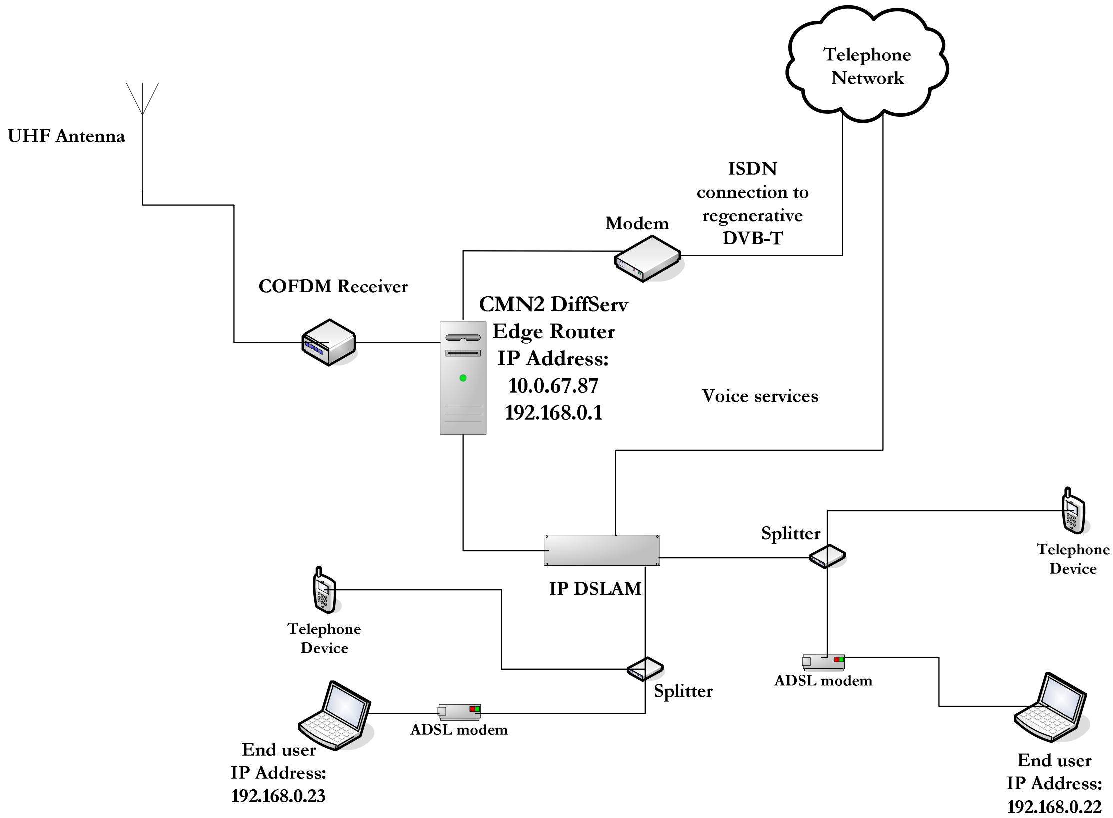


Fig. 4-3 DiffServ aware rural based CMN configuration

4.3. DiffServ aware Mechanisms and QoS Rules

In the described QoS-aware routers (both at the regenerative DVB-T and at the CMNs side) the corresponding DiffServ mechanisms were implemented based on a set of QoS classes and traffic-policing rules, according to which the IP-multicast streams (stemming from the active user in CMN1) were shaped prior to be provided to the entire infrastructure. In this respect, three QoS-classes (and the corresponding rules) were adopted as follows: a Premium service class (i.e. Expedited Forwarding Per Hop Behaviour – EF PHB) [7], an Olympic Model service class (i.e. Gold, Silver, and Bronze services, Assured Forwarding Per Hop Behaviour – AF PHB) [8], and a Best Effort service class. Services classified in the "Expedited Forwarding class", were of the highest priority, receiving therefore the most beneficial treatment from the DiffServ aware modules. On the other hand, services classified in the "Assured Forwarding class", were assigned specific SLAs, defining certain QoS criteria and traffic policing rules, i.e. services classified as "Gold AF1 PHB" received more beneficial treatment from DiffServ aware modules than services classified as "Silver AF2 PHB" and "Bronze AF3 PHB". Finally, services classified in the "Best Effort class" were provided no-QoS guarantees, exploiting the remaining network resources.

Based on these QoS-classes, a representative service scenario was designed for 5 IP multicast streams (as representative cases for bandwidth and delay sensitive services) stemming from the active user in CMN1, according to which the DiffServ mechanisms were implemented in the core/edge routers. In this scenario, the first stream (STREAM 1) was classified as a premium class service, three of them (STREAM 2, STREAM 3 and STREAM 4) as Olympic model class services, and a fifth one (STREAM 5) as a best effort class service. More specifically, the

SLA for the premium class stream was assigned a maximum threshold of 50ms for one way delay and no packet losses. Also, the SLAs for each one of the 3 Olympic model class streams were assigned a maximum threshold of 50ms, 100ms and 150ms for one way delay respectively, as well as a maximum threshold of 0.5%, 1.5% and 1.5% for packet-losses. Finally the last service (STREAM 5), which was associated with the best effort class, utilised the remaining network resources. The configuration of the edge and core DiffServ routers, following this services classification and IP multicast streams scenario, are depicted below in Fig. 4-4 and Fig. 4-5 respectively.

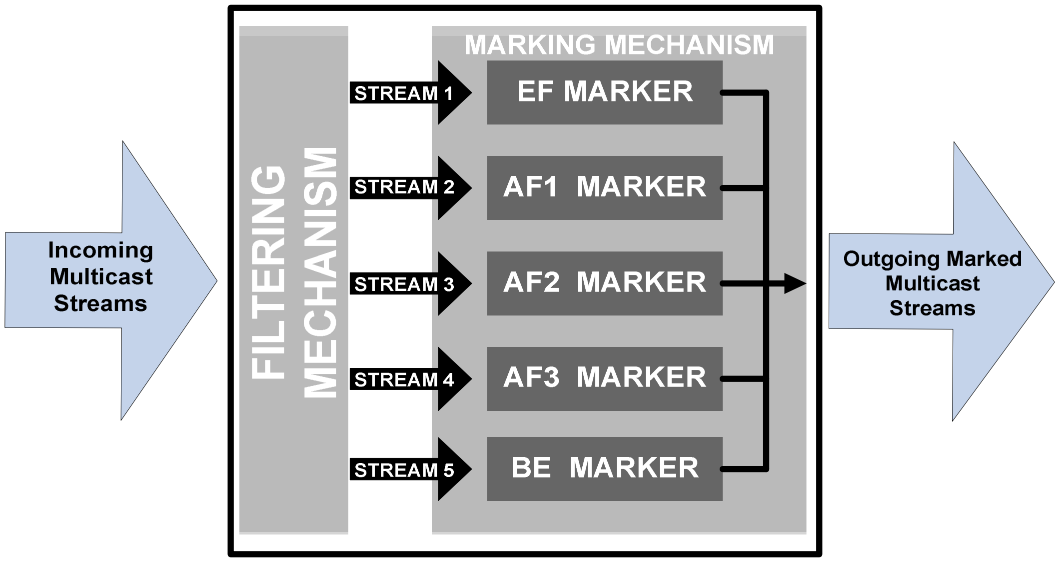


Fig. 4-4 DiffServ edge router mechanisms at CMN level

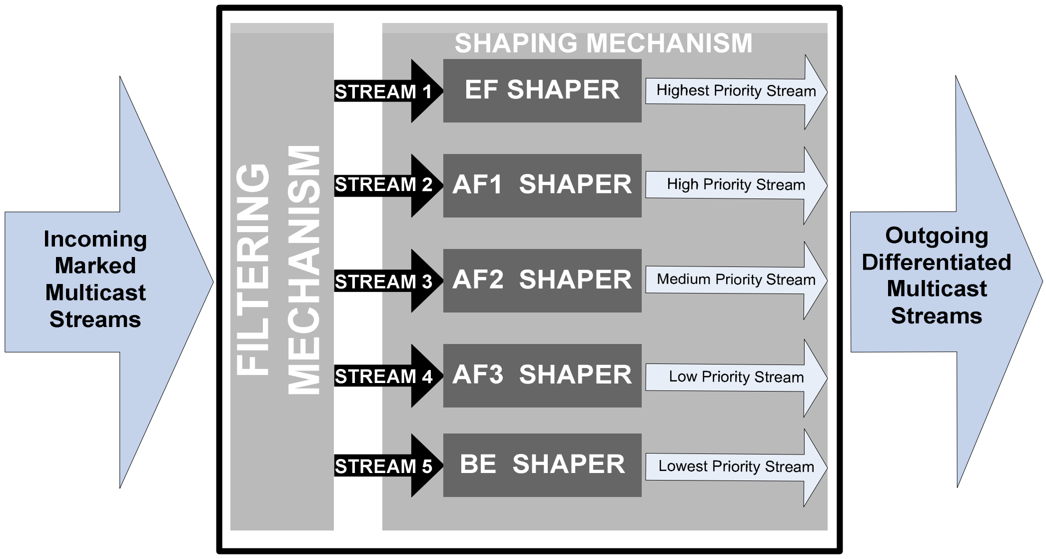


Fig. 4-5 DiffServ core router mechanisms at the regenerative DVB-T platform

In this context, IP traffic stemming from the IP multicaster (in CMN1) passes through the access network to the DiffServ edge router (Fig. 4-4), where it is classified into one of the described QoS classes, before entering the DiffServ domain. More specifically, a filtering mechanism segregates the incoming UDP streams by analysing their UDP segment header field (i.e. port number), and forwards each one to the respective marker. This marker, in turn, assigns a specific DSCP value, into the DS field of each IP packet (i.e. 101110 for EF PHB). This is a standardised value for specific QoS classes, as indicated in [5], [6]. As a result, the output of the DiffServ router at the urban based CMN (CMN1) consists of all UDP multicast streams, each one associated (marked) with a certain PHB before received by the DiffServ core router located in the regenerative DVB-T platform.

The marked UDP streams are received by the regenerative DVB-T platform and forwarded to the DiffServ core router (see Fig. 4-5). This router utilises a filtering mechanism, which segregates the incoming marked multicast streams by analysing the DS field of the IP packet headers and forwards them accordingly to a specific shaper (see Fig. 4-5). Each shaper reserves a portion of the available network resources (i.e. bandwidth allocated for UDP streams in the DVB-T "bouquet") according to the priority level of every IP multicast service as defined in their associated SLAs. The outgoing differentiated traffic is then forwarded to the IP/DVB gateway and broadcasted using the DVB-T transmitter (see Fig. 4-1).

In order to implement the shaping mechanisms of the DiffServ core router, the Hierarchical Token Bucket (HTB) packet scheduler [102] was utilised with five leaf classes (i.e. each one providing different priority level and bandwidth reservation for incoming marked UDP multicast streams). The bandwidth allocated at the parent HTB class was shared among all leaf classes. A pFIFO queuing discipline was utilised in order to implement the first leaf class (i.e. EF class). This queuing discipline treats all packets equally placing them in a single queue and serving them in the same order. Also, four GRED virtual queues were utilised towards implementing AF1, AF2, AF3 and BE classes. GRED queues were configured adjusting the following parameters:

- Q_{max} (1): maximum average queue size after which all packets get dropped,
- BS: percentage of the expected bandwidth share,
- L: maximum desired latency,
- BW: total network bandwidth,
- Q_{min} (2): minimum average queue length after which packets get dropped,
- AvPkt: average packet size,
- B (3): burst value in number of packets and
- Q_{limit} (4): actual queue length which should never be exceeded.

The following values were utilised according to specific system requirements and the QoS rules, defined in the pre-mentioned SLAs of UDP multicast streams:

- total network bandwidth (BW): 4Mb/s
- average packet size (AvPkt): 1024bytes
- maximum desired latency (L): 50ms, 100ms, 150ms and 250ms for AF1, AF2, AF3 and BE classes respectively
- expected bandwidth share (BS): 25%, 23,75%, 23,75% and 2,5% for AF1, AF2, AF3 and BE classes respectively.

Furthermore the Packet Drop Probability in each GRED queue was set to 0.005 (i.e. 0.5%), 0.015 (i.e. 1.5%), 0.015 (i.e. 1.5%) and 1 (i.e. 100%) for AF1, AF2, AF3 and BE classes respectively according to the packet loss ratio defined in their associated SLAs.

$$Q_{\max} = \frac{0,01 * BS * L * BW}{8\frac{bits}{bytes} * 1000\frac{ms}{\sec}} \quad (1)$$

$$Q_{\min} = \frac{1}{2} * Q_{\max} \quad (2)$$

$$B = \frac{2 * Q_{\min} + Q_{\max}}{3 * AvPkt} \quad (3)$$

$$Q_{\text{limit}} = 4 * Q_{\max} \quad (4)$$

4.4. Optimised System Performance

4.4.1. Network Performance Evaluation

The UDP streams scenario previously described was utilised for evaluating the capacity of the prototype in providing QoS guarantees according to the specific service classes. Towards these, the MGEN [92] traffic generator was used, configured so that multiple UDP streams (i.e. emulation of real services) to be delivered simultaneously from the same terminal (IP multicaster in Fig. 4-2) via the same network interface, but over different communication ports. The CMN was configured in order to manage the multicast traffic by utilising the SMCRoute application [103]. Each one of these multicast UDP streams was utilising a bandwidth of 1 Mb/s and was transmitted in Constant Bit Rate (CBR), with packet size of 1024bytes for a period of 180 seconds. It should be noted that during these experiments the total available bandwidth of the DVB-T stream (i.e. 21.11Mb/s) was statically allocated between the MPEG-2 and IP services as follows: 17Mb/s were dedicated to a bouquet of 5 digital TV programmes (i.e. MPEG-2 live and non-live broadcasts), while the remaining 4Mb/s were dedicated to IP services, i.e. one IP channel for UDP streams in the IP/DVB gateway.

The first set of experiments was a preliminary one towards identifying the initial system characteristics, comprising one active UDP stream (1Mb/s CBR with packet size of 1024bytes long) and no-QoS mechanisms (i.e. the DiffServ routers disabled). The experimental results indicated an average one way delay of about 30.7ms and no packet-losses during the 180s evaluation period (as a matter of the un-congested 4Mb/s IP channel in the DVB-T stream).

The next set of experiments was designed in order to evaluate the network performance when all the five UDP streams (each one at 1Mb/s CBR with packet size of 1024bytes long) are simultaneously active, exploiting Best-effort scheme (i.e. the DiffServ routers disabled) for being delivered over the common UHF channel. In this case the IP channel in the DVB-T stream was congested, as long as the allocated 4Mb/s bandwidth was evenly shared among all UDP streams. The experimental results in these set of tests indicated 200ms of one way delay and 23% of packet-loss ratio for each UDP stream, resulting in degraded network performance in comparison to the initial experiment.

The next set of experiments was designed in order to evaluate the network performance when the 5 UDP streams are simultaneously provided over the infrastructure and the DiffServ mechanisms are active. During this experiment, the UDP streams were provided over the same source IP address (i.e. 172.16.0.40, IP address of IP multicaster) but via different source port numbers (i.e. 9500 for EF PHB, 9501 for AF1 PHB, 9502 for AF2 PHB, 9503 for AF3 PHB and 9504 for BE PHB). The segregation of UDP streams was achieved analysing the different source port numbers (i.e. 9500 for EF PHB, 9501 for AF1 PHB, 9502 for AF2 PHB, 9503 for AF3 PHB and 9504 for BE PHB). The marking mechanism in turn assigned different DSCP values into the DS field of each IP packet according to a certain PHB (i.e. 101110 for EF PHB, 001010 for AF1 PHB, 010010 for AF2 PHB, 011010 for AF3 PHB, and 000000 for BE PHB). As a result, the output of the DiffServ router at the urban based CMN was consisting of the UDP multicast streams, each one associated with a certain PHB before received by the DiffServ core router located in the regenerative DVB-T platform.

At the regenerative DVB-T platform, the marked UDP streams were received and forwarded to the DiffServ core router, where a filtering mechanism segregated them according to their DS field in the IP packet headers, before forwarding them accordingly to the specific shaper. Each shaper reserves a portion of the available network resources (i.e. from the 4Mb/s IP channel in the DVB-T "bouquet") according to the priority level of every IP multicast service as defined in their associated SLAs. As already mentioned, the implemented shaping mechanisms were based on the Hierarchical Token Bucket (HTB) packet scheduler [102], by utilising five leaf classes (i.e. each one providing different priority level and bandwidth reservation for incoming marked UDP multicast streams). The bandwidth allocated at the parent HTB class was 4Mb/s shared among all leaf classes. In this respect, a bandwidth of 1Mb/s was reserved for each one of the 1st UDP stream associated with EF PHB and the 2nd UDP stream associated with AF1 PHB. A bandwidth of 950Kb/s was reserved for each one of the 3rd and 4th UDP streams associated with AF2 and AF3 PHBs respectively. Finally a bandwidth of 100Kb/s was also reserved for the best effort UDP stream. Additionally, a different priority level was defined in the HTB for each leaf class (i.e. priority level 1 up to 5 for EF PHB, AF1 PHB, AF2 PHB, AF3 PHB and BE PHB respectively).

The experimental results, regarding the one way delay and the packet loss ratio of each UDP stream, are depicted in Fig. 4-6 and Fig. 4-7 respectively, for the total 180s performance evaluation period. From Fig. 4-6, it can be observed that the average one way delay, when the DiffServ mechanisms were disabled, was about 200ms for each UDP stream. On the other hand, when the DiffServ mechanisms were enabled, the one UDP stream classified as EF PHB (i.e. STREAM1) experienced an average one way delay of less than 50ms, while the three UDP streams classified as AF PHBs (i.e. STREAM2, STREAM3, STREAM4) experienced average one way delays of less than 50ms, 100ms and 150ms respectively, conforming to their respective SLAs. In both cases, the BE multicast stream experienced similar one way delays.

Similarly, from Fig. 4-7 it can be deduced that when the DiffServ mechanisms were disabled, each UDP stream experienced an average packet-loss of about 23%. On the other hand, when the DiffServ mechanisms were activated, UDP streams associated with Premium (i.e. EF PHB) and Gold (i.e. AF1 PHB) service classes, experienced no packet-losses. The UDP multicast streams associated with Silver (i.e. AF2 PHB) and Bronze (i.e. AF3 PHB) service classes experienced packet-losses (i.e. the values observed are below the threshold defined in their respective SLAs), however under the upper threshold (i.e. a threshold of about 2% - 3%

defined in [100]) for a good network performance and network QoS provisioning of real time services. The experimental results also indicated, that the best effort UDP stream (i.e. STREAM5) experienced 95% packet losses when QoS aware mechanisms were enabled, conforming to its respective SLA.

These experimental results indicate that the utilisation of the QoS aware mechanisms by configuring the DiffServ edge and core routers according to the QoS requirements for each UDP stream, results in an optimised performance of the network for the provision of multicast streams through the implemented prototype. Effectively, it can be deduced that the pre-defined SLAs were respected by the entire system, both at the core and backhaul network, and the introduced QoS aware mechanisms improved the overall network performance, setting the basis for end-to-end QoS provisioning. The experimental results of the network performance evaluation regarding the DiffServ aware mechanisms are depicted in Appendix (A.III).

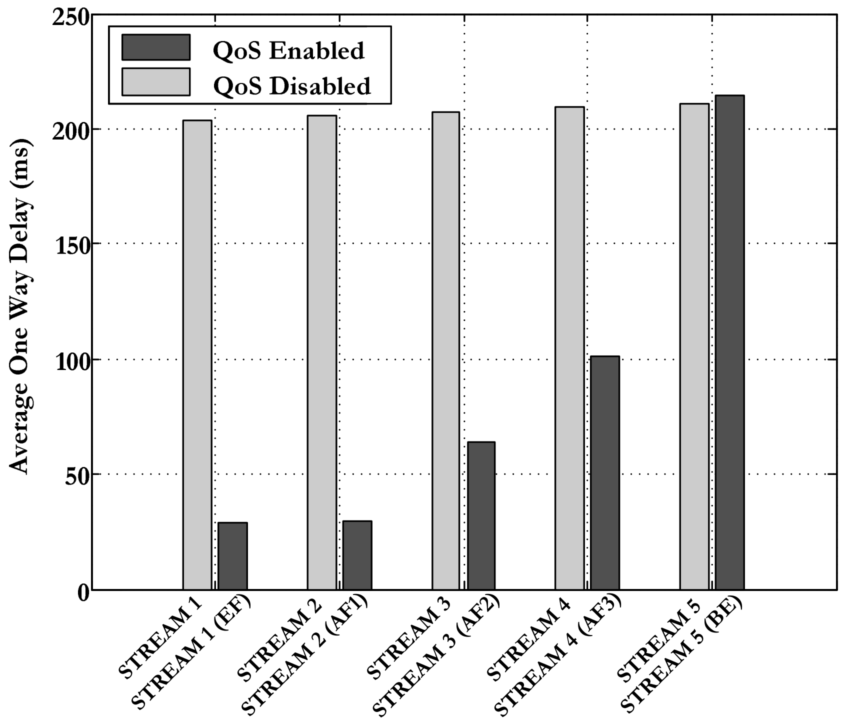


Fig. 4-6 Average one way delay for UDP multicast streams

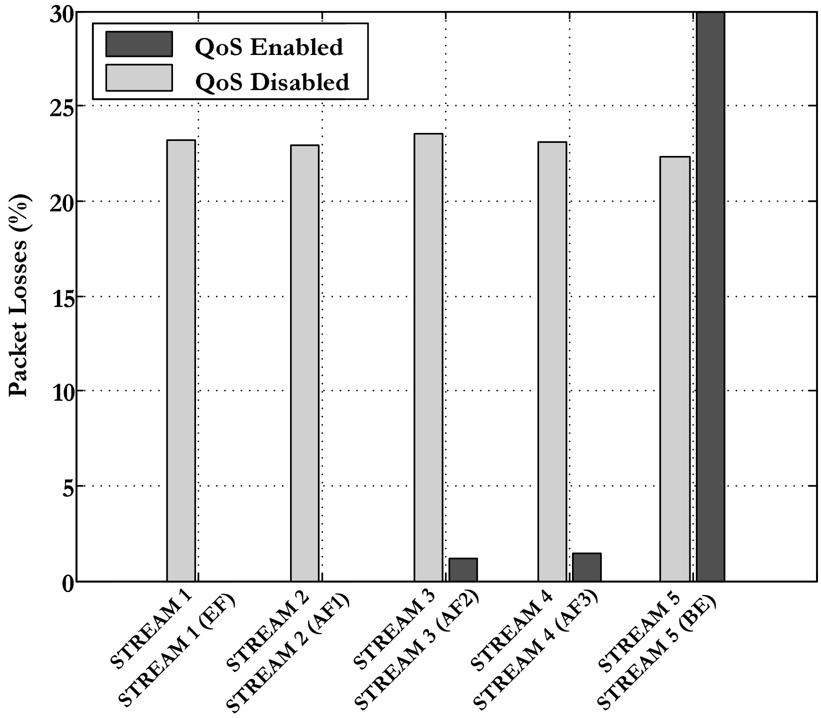


Fig. 4-7 Packet loss ratio for UDP multicast streams

4.4.2. Perceived Quality of Service

The final set of tests in this experimental phase was designed towards evaluating the QoS that rural users experience when accessing real/actual IPTV streams (Perceived QoS), over the implemented prototype. Towards these, a similar experimental procedure as before was followed, where the UDP traffic (previously generated by the MGEN tool) was replaced by a real/actual MPEG-4 video stream, created at the active user's premises (i.e. IP multicaster). It should be noted that this video sequence was initially a video file in AVI format, comprising of medium picture-complexity elements, medium-moving parts and of low-scene transitions (i.e. a documentary). Encoding of this video sequence into MPEG-4 format was achieved by utilising the MPEGABLE Dicas software encoder [104]. Furthermore, it should be mentioned that CIF resolution/format (352 x 288 pixels) was chosen as the display characteristics for representation of this video sequence at the end-user's premises (in CMN2).

According to these specific video characteristics and the chosen resolution/format (i.e. documentary at CIF resolution), the video-encoding bit-rate was selected at 887Kb/s, providing for high image/video quality, while the audio-encoding rate was set at 96Kb/s (for high quality stereo sound), resulting in a total audio/video rate of 983Kb/s (i.e. ~1Mb/s). Fig. 4-8 and Fig. 4-9 depict the video-encoding and audio-encoding interfaces respectively, in the MPEG-4 encoder, while Fig. 4-10 represents the achieved picture quality after the encoding process.

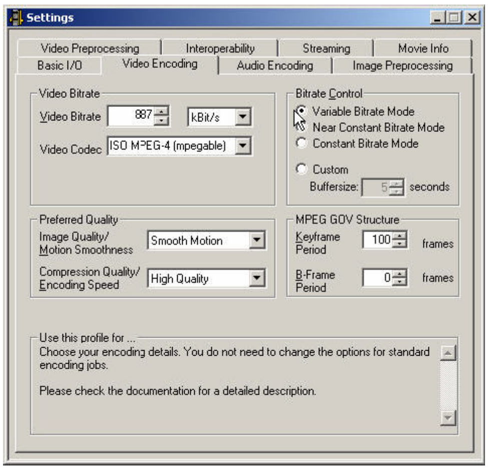


Fig. 4-8 Video-encoding interface at the MPEG-4 encoder

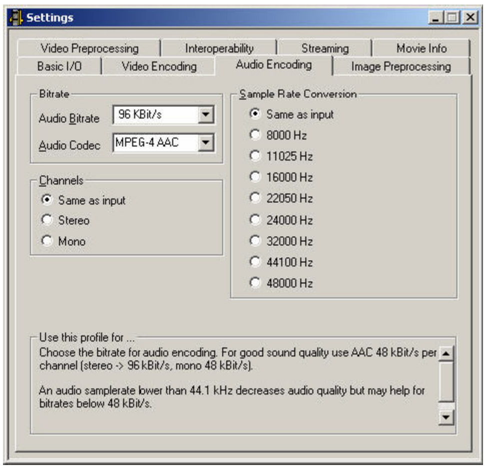


Fig. 4-9 Audio-encoding interface at the MPEG-4 encoder

Fig. 4-10 Picture quality of the MPEG-4 sequence at 887Kb/s

The produced MPEG-4 sequence was saved as a video file at the active user's PC (IP address 172.16.0.40) prior to be multicasted over the UDP protocol (IPTV stream). Towards this, the VLC open-source software tool [105] was exploited for retrieving data of the MPEG-4 video-file (see Fig. 4-11), creating the UDP packets stream, and finally multicasting it to the entire infrastructure (over the UDP protocol) in a specific multicast address and a precise port number (see Fig. 4-12). Reception of this stream via the network and representation of video information on the end-user's terminal (i.e. user's PC screen) was achieved by utilising the MPlayer tool [106].

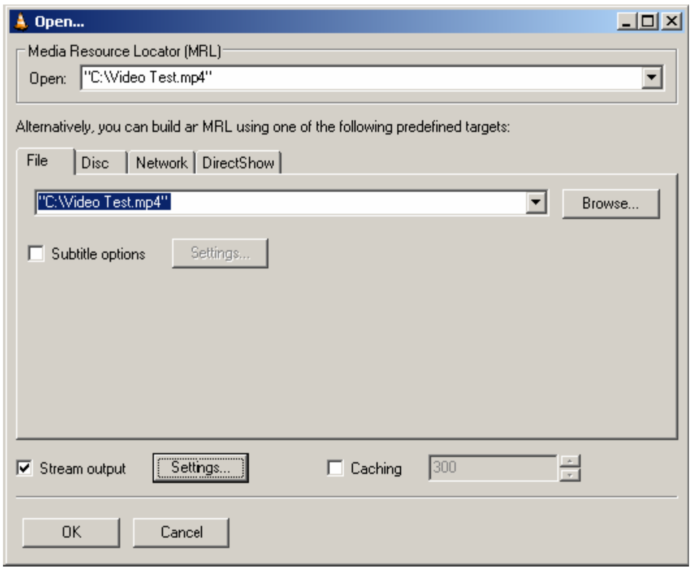


Fig. 4-11 Selecting the desired MPEG-4 file in the VLC tool

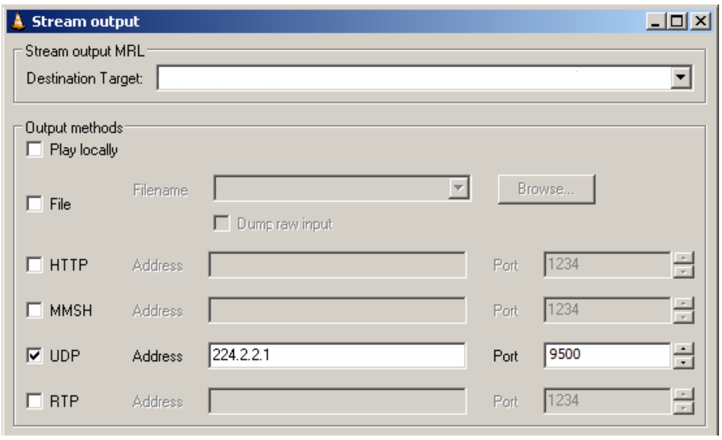


Fig. 4-12 Setting the UDP transmission parameters in the VLC tool

4.4.2.1. MPEG-4 Video Provision when QoS aware Mechanisms are Inactive

The first tests in these final experiments were conducted in order to evaluate the perceived picture quality that an end user experiences (located at CMN2), when accessing an MPEG-4 stream with no QoS guarantees (i.e. when the DiffServ modules are inactive). Towards this, subjective picture-quality assessment was conducted following the ITU-R BT.500 recommendation [107] by utilising a ten-people audience.

In this respect, two sets of tests were conducted, one under un-congested network conditions (i.e. only one MPEG-4 stream of 983Kb/s was delivered over the IP channel of the DVB-T stream), and the other when the IP channel in the DVB-T stream was heavily loaded (i.e. five MPEG-4 streams concurrently competing under the Best-Effort scheme within the IP channel of the DVB-T stream). It should be noted that during these tests, the total available bandwidth of the DVB-T stream (i.e. 21.11Mb/s) was again statically allocated between the MPEG-2 and IP services as follows: 17Mb/s were dedicated to a bouquet of 5 digital TV programmes (i.e. MPEG-2 live and non-live broadcasts), while the remaining 4Mb/s were dedicated to IP services.

More specifically, and in the case of congested network conditions, the VLC software was configured so that the same audiovisual content file (MPEG-4 format, 983Kb/s) to be simultaneously transmitted (five times), over different multicast addresses and port numbers from the IP multicaster in CMN1, while end-user at CMN2 trying to access one of them. Reviewing the opinion of the audience, it was verified that when a single MPEG-4 stream is provided over the prototype, the perceived QoS was characterised as "excellent", while when all five MPEG-4 streams are concurrently delivered, the PQoS was reported as "poor". Fig. 4-13 indicates the perceived picture quality under un-congested network conditions, while Fig. 4-14 indicates the picture quality the end-user experiences when the 5 MPEG-4 streams are concurrently competing in the IP channel of the DVB-T stream.

Fig. 4-13 Perceived picture quality under un-congested network conditions

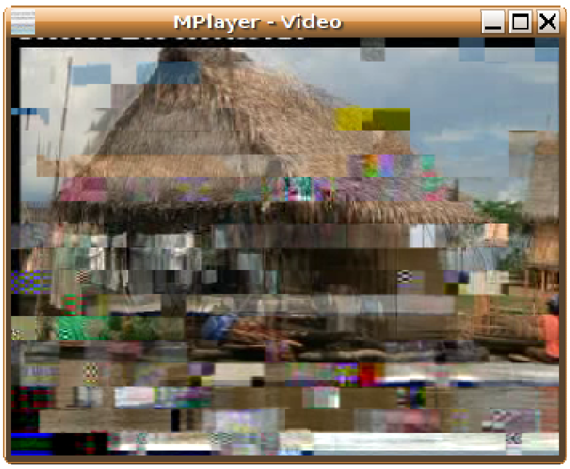
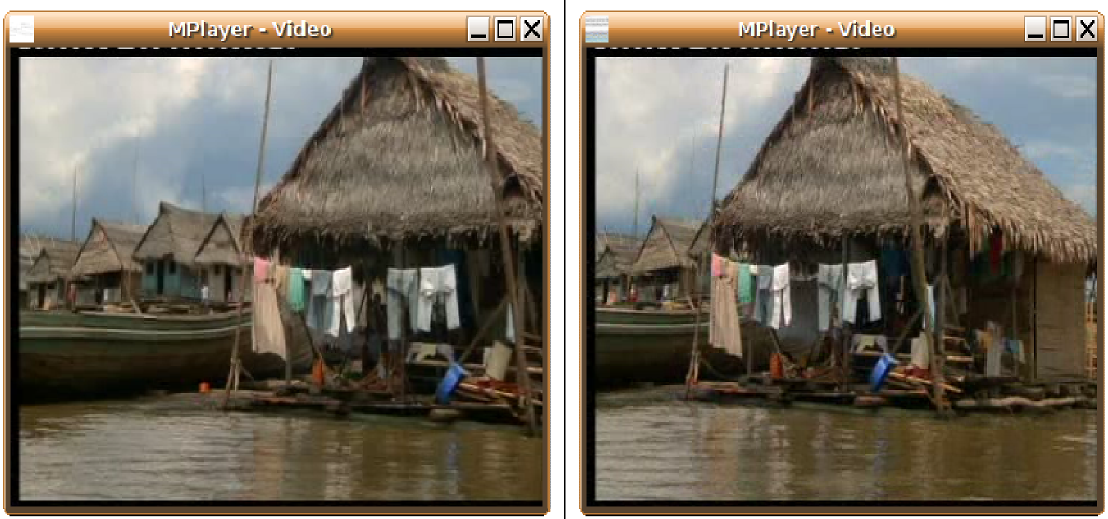


Fig. 4-14 Perceived picture quality under congested network conditions

4.4.2.2. MPEG-4 Video Provision when QoS aware Mechanisms are Active

The next tests in these final experimental phase were conducted in order to evaluate the Perceived picture quality an end-user experiences (located at CMN2), when accessing an MPEG-4 stream with QoS guarantees (i.e. when the DiffServ modules are active). Similarly as before, the picture-quality assessment was based on subjective evaluation, utilising the opinion of a ten-people audience. In this tests, the DiffServ aware modules were activated and configured for supporting all five IPTV multicast streams at three possible quality classes: a Premium service class (i.e. EF PHB) [7], an Olympic Model service class (i.e. AF PHB) [8], and a Best Effort service class. More specifically, the first IPTV service (i.e. IPTV1) was associated with EF PHB for guaranteed "excellent" picture quality, according to a pre-defined SLA, a one way delay below a maximum threshold of 50ms, no packet-losses and assured bandwidth. Three IPTV services (i.e. IPTV2, IPTV3, IPTV4) were associated with one of the three levels of AF PHB (i.e. Gold, Silver and Bronze classes); i.e. services classified as "Gold class AF PHB" were of higher priority than services classified as "Silver and Bronze AF PHB", and therefore receive more beneficial treatment from DiffServ aware modules. The SLAs for these IPTV services were defined as follows; one way delay and packet-losses to be below the maximum thresholds of 50ms and 0.5% for IPTV2, 100ms and 1.5% for IPTV3, 150ms and 1.5% for IPTV4. Finally, the last IPTV service (i.e. IPTV5) was associated with a Best Effort service class utilising the remaining network resources.

Reviewing the opinion of the audience it was verified that the perceived quality of IPTV1 and IPTV2 services (associated with EF and AF1 QoS classes respectively) was of "excellent" QoS, the quality of IPTV3 and IPTV 4 services was characterised as "good", while the quality of the fifth MPEG-4 multicast stream (IPTV5 service) was "very poor". Fig. 4-15 indicates the perceived picture quality experienced for each one of the differentiated IPTV services.



EF IPTV1

AF1 IPTV2

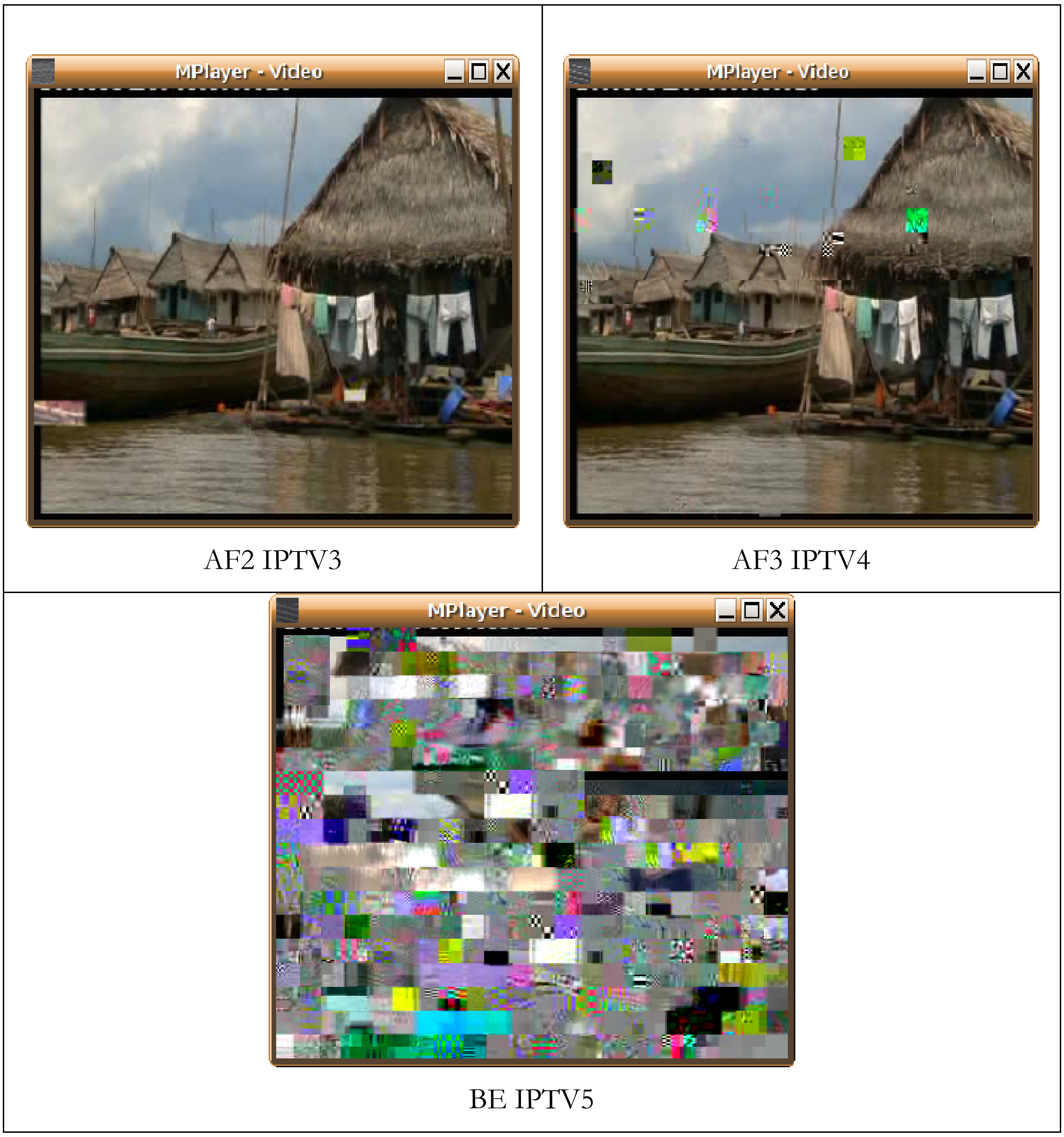


Fig. 4-15 Perceived picture quality when the DiffServ mechanisms are activated

4.5. Summary

This chapter elaborated on the capacity of the proposed architecture in providing QoS guarantees according to specific users' privileges and services attributes. In this context, it described the design and implementation of QoS aware modules that make use of DiffServ capable mechanisms. It also analysed the rules according to which services differentiation and QoS provisioning is obtained, and presented their deployment (in the implemented prototype) following a decentralised approach. Towards evaluating the overall system performance, and verifying its capacity in QoS provisioning, a number of experiments were conducted under realistic transmission/reception conditions, in respect to delay-sensitive and bandwidth-dependent services provision (i.e. multicast audio/video). More specifically, several use/services scenarios were examined, which were emulated over unformatted UDP streams (i.e. row UDP data provided by a traffic generator), based on network performance metrics and network characteristics, including one way delay and packet losses. Following these emulated traffic scenarios, the chapter elaborated on the Perceived QoS provisioning (i.e. the quality that an end-user experiences) when real/actual MPEG-4 video streams, are provided over the prototype. Subjective perceived QoS evaluation and picture quality assessment verified the capability of the implemented prototype in guaranteed QoS provisioning according to user's privileges and services attributes, establishing therefore the proposed architecture as an alternative solution for broadband metropolitan networks that enable for the provision of guaranteed services.

The research work presented in this chapter was published in [108].

5. CONCLUSIONS

5.1. Overview

This final chapter of the thesis concludes it by resuming the research efforts, its scientific results and contribution to knowledge, as well as by identifying fields for future exploitation. In this context, section 5.2 summarises the work carried-out towards the design, implementation and performance evaluation of the proposed networking architecture, which can enable for the realisation of metropolitan infrastructures, providing ubiquitous access to broadband heterogeneous services at guaranteed QoS. Section 5.3 elaborates on the validity of the proposed architecture, which was established via a series of experiments that were conducted on a prototype testbed (conforming to the design specifications), while section 5.4 elaborates on issues for future research.

5.2. Innovation and Contribution to the State-of-the-art

This Ph.D. thesis presented the design, implementation and performance evaluation of a metropolitan network that enables users/citizens to access linear, interactive and on-demand broadband services via a single infrastructure at guaranteed QoS. Building upon the issue of technology and services convergence, it initially studied the existing solutions for the realisation of such unified environments, currently based on two main approaches; the convergence between Internet and telecommunications, and the solutions proposed by the interactive broadcasting sector. Review of the relevant technological solutions indicated that while Internet/telecommunications technology convergence arises as a very promising technological solution, economic and/or geographical issues prohibit their deployment, especially in rural and dispersed areas. On the other hand, interactive broadcasting solutions can hardly contribute to the deployment of broadband unified metropolitan infrastructures, as they cannot efficiently support on-demand heterogeneous services provision and guaranteed QoS provisioning.

Following this technology review, it was anticipated that if convergence is applied among all these sectors both at technological and services levels – and not partially among some of them – a novel networking environment can be realised, complementing existing infrastructures besides enhancing them towards a more dynamic stage: the establishment of a unified/single metropolitan environment that enables urban and rural citizens to receive and distribute heterogeneous services at maximum possible QoS. In this respect, the thesis presented the design and system configuration of a prototype infrastructure that exploits DVB-T technology for the realisation of backhaul connections, which interconnect all citizens within the broadcasting footprint to each other and to the core backbone. Services reception and delivery was achieved via intermediate communication nodes (CMNs) that make use of wired and/or wireless technologies in the access network. In such a decentralised architecture, QoS provisioning was established by exploiting services prioritisation techniques and mechanisms, according to specific services attributes and users' privileges.

More specifically, the proposed design exploits the DVB-T stream in regenerative configurations, towards establishing a common IP-backbone that is present and available within the entire broadcasting area. This IP-backbone acts as a backhaul/middle-mile connection, which extends the core-backbone (mainly present at urban territories) to reach every user within the broadcasting footprint. Users, in turn, access this IP-backbone (and therefore the core-backbone), via the corresponding neighbouring CMN, utilising wired and/or wireless links, both for delivering and consuming broadband services. Efficient network operation, as a matter of dynamic resource allocation and maximum possible QoS provisioning, was confronted by designing, implementing and deploying in a decentralised approach a number of DiffServ capable modules that were making use of QoS aware mechanisms/rules.

As a result, the proposed architecture constitutes an innovative approach for Internet, telecommunications and interactive broadcasting convergence, and provides an alternative solution that complements existing ones towards the deployment of metropolitan infrastructures (supporting broadband heterogeneous services access with guaranteed QoS provisioning). Also, the adopted prototype exploitation of DVB-T stream in regenerative configurations, provided for the utilisation of DVB-T technology not only for linear TV services provision, but also (and most predominant) for the creation of IP-backbones, which are commonly shared among existing (e.g. broadcasters, telecom operators, ISPs) and potential services content providers (active users/citizens who create, maintain and deliver their own services). Furthermore, the novel use of this regenerative DVB-T stream in backhaul connections for extending the core backbone, provided for the fast deployment/unfold of existing metropolitan infrastructures to reach every user/citizen within the broadcasting footprint. Finally, the novel design and use of IP QoS-aware mechanisms in a broadcast-oriented technology (until recently used for custom linear and interactive services provision), allowed for the exploitation of DVB-T technology as part of a unified metropolitan infrastructure where QoS-provisioning is guaranteed, according to services prioritisation and users' classification process.

5.3. System Capabilities and Limitations

Following the system design, the Thesis elaborated on the validity of the proposed architecture via a series of experiments that were conducted on a prototype testbed (conforming to the design specifications) under real/actual transmission and reception conditions. The obtained experimental results verified its capacity in establishing broadband metropolitan infrastructures for heterogeneous services access with QoS guarantees, while drew-up the systems potentialities and limitations.

Towards these, it presented the implementation and configuration of a testbed comprising of one regenerative DVB-T platform, where the common IP-backbone is created, and two CMNs. One CMN located in an urban area by utilising WLAN technology in the access network, and another one located in a rural metropolitan area (where only primitive PSTN/ISDN lines exist and no access/connection to the core-backbone is available), by exploiting xDSL technology in the access network. On this prototype, a number of experiments were conducted regarding both emulated data traffic (i.e. IP data provided by traffic generators over the TCP and UDP protocols) and realistic services scenarios utilising actual IPTV streams (MPEG-4 multimedia content). More specifically, and in respect to the emulated on demand services provision (i.e. TCP data provided by the traffic generator), it was experimentally verified that the overall network performance is a matter of the configuration parameters, such as the available resources in the access network, the uplink and downlink characteristics, the available bandwidth allocated for IP data in the DVB-T stream, as well as the utilised protocol attributes. On the other hand, the experimental tests concerning emulated multicast real time services (i.e. UDP data provided by the traffic generator) indicated that a single UDP stream can be efficiently delivered over the entire infrastructure, a matter however, of the stream parameters (packet size and bit-rate). Furthermore, it was also verified that under the Best-effort scheme (i.e. under no traffic prioritisation and QoS provisioning guarantees) the system resources are fully exploited by a single user, while are evenly shared during simultaneous/concurrent access. However, during services provision under the Best-effort scheme no QoS guarantees could be provided. On the contrary, when services prioritisation is utilised, the system can efficiently support QoS guarantees, following specific rules according to users' privileges and services attributes. This was also experimentally verified via a series of tests utilising real IPTV streams, where subjective assessment of the received picture quality was carried-out (Perceived QoS).

5.4. Fields for Future Research

The work carried out within this Thesis resulted in a prototype architecture that allows technology and services convergence, constituting the basis for the efficient and fast deployment/establishment of broadband metropolitan infrastructures. Prior to these, however, a number of issues have to be taken into account and confronted, ranging from networking matters (such as the exploitation of the available system resources among the various services) to the end-to-end QoS provisioning, as well as issues concerning services and data security, billing and accounting and for some specific use-scenarios, users' and services mobility.

For example, the total available bandwidth of DVB-T-stream in the implemented prototype was statically allocated between MPEG-2 and IP services. In actual/real implementations, however, where time-varying MPEG-2 traffic is usually utilised (e.g. VBR MPEG-2 TV programmes), this static bandwidth resource sharing may result in un-efficient spectrum exploitation with direct impact on the overall system performance. In such a case, a bandwidth management system is required, capable to dynamically re-allocate (in real-time) the available system resources and statistically multiplex the provided services in the regenerative DVB-T stream. Furthermore, and especially in the case of bandwidth sensitive services (such as IPTV streams and on-demand services provision), this resource allocation mechanism should also take into account and be aware of the users' QoS privileges and the Perceived QoS attributes, prior to the changes in the regenerative DVB-T stream multiplex. In this respect, a feed-back loop QoS-provisioning mechanism is essential, acting not only in the core/backhaul connections, but also in the access network. Such a mechanism must be able to control and adapt in real-time all layers of the system, enabling for a cross-layer optimisation in respect to the available network resources and services/users' QoS requirements. In this context, the proposed architecture and the work described in this Thesis, can constitute the basis for future research in the field of end-to-end QoS provisioning, as a matter of real-time resource allocation and dynamic cross-layer system optimisation for guaranteed Perceived QoS.

Another potential field for future exploitation that arises from the conducted research is the issue of data security and content integrity, especially when critical personalised services access is considered. For example, on-demand access to personal content with billing/accounting issues requires a sophisticated mechanism capable to retain data security, especially in the wireless/broadcasting medium (e.g. DVB-T stream). While many techniques and protocols are nowadays available for providing such data and services security (e.g. Conditional Access for digital TV broadcasts, IPSec protocol for IP services provision, etc.), the exploitation of the IP Multimedia Subsystem (IMS) seems to be the most challenging solution for converging Internet, telecommunications and broadcasting data integrity issue into a concrete and unified secure shell. As IMS is a control plane technology, utilising the Session Initiated Protocol stack (SIP), the appropriate modules have to be designed and implemented, both in the backhaul and access network, in order to be efficiently encompassed within the IP-based and broadcast medium.

The above mentioned fields for future research become more evident and obvious when users' mobility is in place, i.e. when a wireless user of the described infrastructure is moving from one CMN to another (inter cell handover) or when moving from one base station to another within the same CMN (intra cell handover). While the latter is mainly an issue of the access network characteristics and configuration attributes (e.g. traffic balancing when the user leaves the cell and enters the new one, seamless services access, QoS maintenance, etc.), the inter cell handover still remains an issue, mainly related to the data-routing and forwarding process taking place at the regenerative DVB-T side. The need for such a routing policy and traffic-management mechanism becomes more apparent when, apart from users' mobility, services mobility is also required, i.e. when a user accesses personalised services (e.g. favourite IPTV stream) from a different CMN than the one his service provider has registered to be the accessing node. Fore example, the IPTV multicaster registers "user-x" as his customer who accesses the IPTV programmes via CMN1, which in turn takes the responsibility of accommodating the provided service at the agreed QoS. In case that this user wants to access the specific service from another CMN, a novel mechanism is needed for transparently conveying the users' QoS-privileges to the new CMN, besides providing the requested service at the agreed QoS. Of course, billing and accounting issues are also applied, a matter however of the proposed IMS modules.

Nevertheless, it should be noted that even if all the above issues will be well confronted and the corresponding mechanisms are successfully deployed, the limited total available bandwidth within the DVB-T stream (i.e. up-to about 30Mb/s per UHF channel) will still constitute a major bottle-neck, unless more resources are provided. Towards this, it is anticipated that a very promising solution can arise if the "regenerative DVB-T" concept is adopted for the entire UHF band, and not only for a single UHF channel. Applying, for example, the regenerative DVB-T approach/design to all 40 UHF channels of a city/region, a virtual medium is automatically created, providing more than 1Gb/s total available aggregated bandwidth. Exploitation, however, of this medium requires research to be carried-out in a number of scientific fields such as bandwidth management and spectrum sharing techniques, adaptive modulation and coding schemes, multiple antenna and user detection schemes, and cross-layer design, in order to efficiently use the UHF band as part of the entire broadband metropolitan infrastructure.

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ABBREVIATIONS

A	
ADSL	Asymmetric Digital Subscriber Line
AF	Assured Forwarding
B	
BA	Behaviour Aggregate
BE	Best Effort
BW	Bandwidth
C	
CBR	Constant Bit Rate
COFDM	Coded Orthogonal Frequency Division Multiplexing
CRC	Cyclic Redundancy Check
D	
DCR	DiffServ Core Router
DECT	Digital Enhanced Cordless Telecommunications
DER	DiffServ Edge Router
DiffServ	Differentiated Services
DS Field	Differentiated Service Field
DSCP	Differentiated Service Code Point
DSL	Digital Subscriber Line
DSLAM	Digital Subscriber Line Access Multiplexer
DSM-CC	Digital Storage Media Command and Control
DVB	Digital Video Broadcasting
DVB-C	Digital Video Broadcasting - Cable
DVB-H	Digital Video Broadcasting - Handheld
DVB-S	Digital Video Broadcasting - Satellite
DVB-T	Digital Video Broadcasting - Terrestrial
E	

EF	Expedited Forwarding
ETSI	European Telecommunications Standards Institute
F	
FIFO	First-In, First-Out
FSO	Free Space Optics
G	
GPRS	General Packet Radio Service
GRED	Generalized Random Early Drop
GSM	Global System for Mobile Communications
H	
HDTV	High Definition Television
HTB	Hierarchical Token Bucket
I	
IEEE	Institute of Electrical and Electronic Engineering
IETF	Internet Engineering Task Force
IntServ	Integrated Services
IP	Internet protocol
IPTV	Internet Protocol Television
ISDN	Integrated Services Digital Network
ISP	Internet Service Provider
L	
LLC/SNAP	Logical Link Control/Sub-Network Access Point
LMDS	Local Multipoint Distribution System
M	
MAC	Media Access Control
MPE	Multi Protocol Encapsulation
MPEG-2	Moving Picture Experts Group
MPLS	Multi Protocol Label Switching
MSB	Most Significant Bit
N	
NPA	Network Point of Attachment

NTSC	National Television System Committee
P	
PAL	Phase Alternating Line
PDU	Protocol Data Unit
PHB	Per Hop Behaviour
PRBS	Pseudo Random Binary Sequence
PSTN	Public Switched Telephone Network
Q	
QAM	Quarter Amplitude Modulation
QoS	Quality of Service
QPSK	Quadrature Phase Shift Keying
R	
RF	Radio frequency
RS	Reed Solomon
RTT	Round Trip Time
S	
SDTV	Standard Definition Television
SFN	Single Frequency Network
SLA	Service Level Agreement
SNDU	Subnetwork Data Unit
T	
TCA	Traffic Condition Agreement
TCP	Transmission Control Protocol
TPS	Transmission Parameter Signalling
TS	Transport Stream
U	
UDP	User Datagram Protocol
UHF	Ultra High Frequency
ULE	Unidirectional Lightweight Encapsulation
UMTS	Universal Mobile Telecommunications System
V	

VHF	Very High Frequency
W	
WLAN	Wireless Local Area Network
WWW	World Wide Web

APPENDIX

A.I Experimental Results of End-to-end Network Performance Evaluation for Different TCP Congestion Avoidance Algorithm

A.I.a Instantaneous Useful Throughput

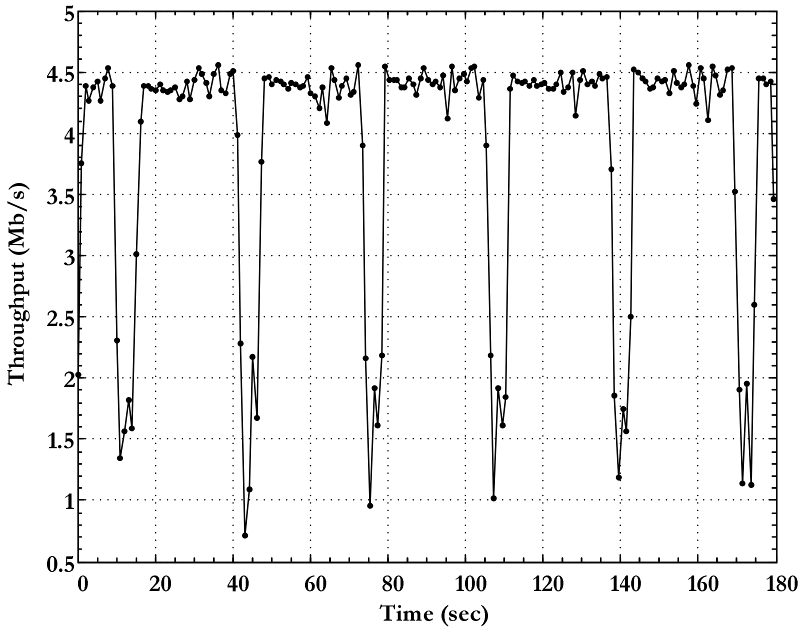

Fig A. 1 Useful throughput for High Speed TCP congestion avoidance algorithm

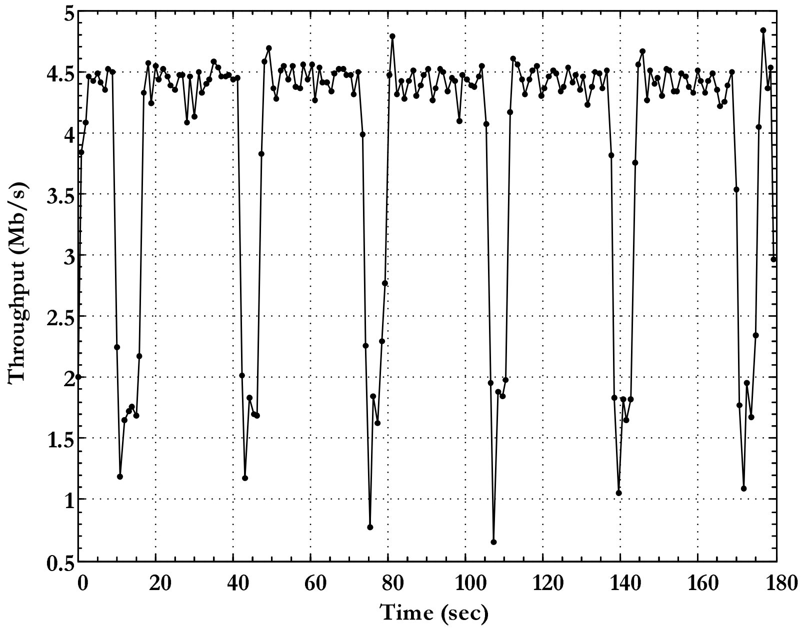

Fig A. 2 Useful throughput for H-TCP (Hamilton) congestion avoidance algorithm

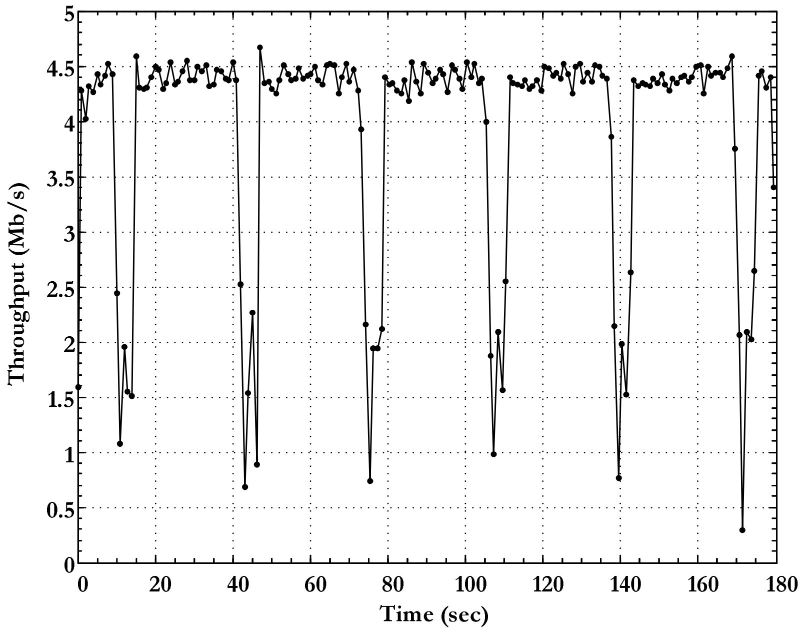

Fig A. 3 Useful throughput for Scalable TCP congestion avoidance algorithm

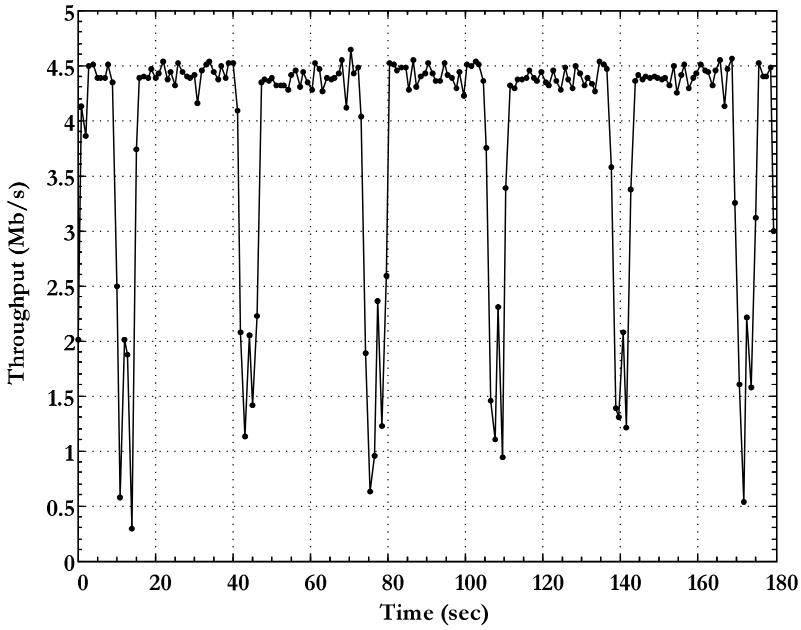

Fig A. 4 Useful throughput for TCP BIC congestion avoidance algorithm

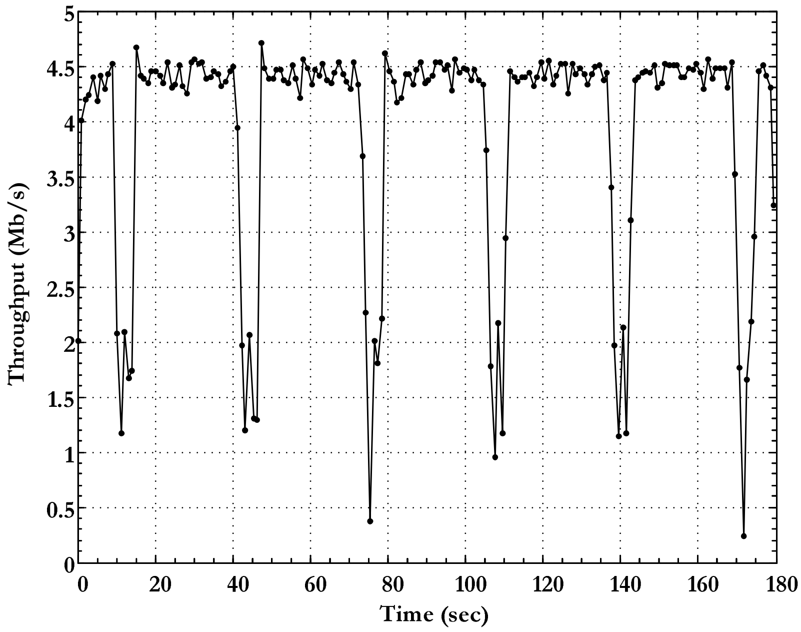

Fig A. 5 Useful throughput for TCP CUBIC congestion avoidance algorithm

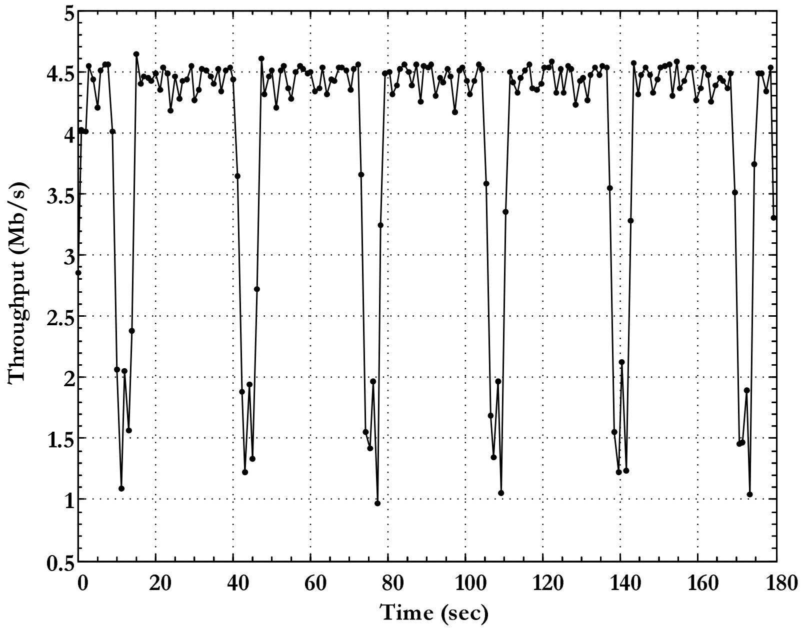

Fig A. 6 Useful throughput for TCP Hybla congestion avoidance algorithm

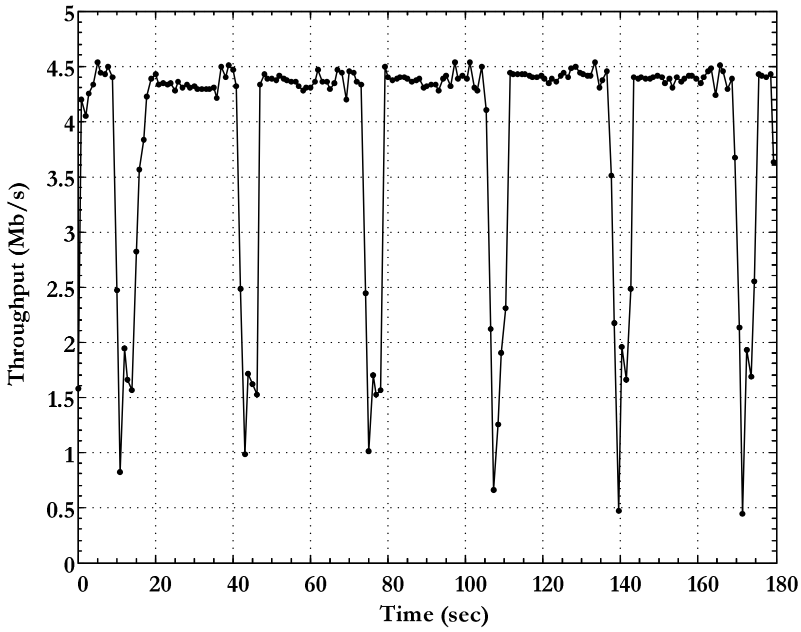

Fig A. 7 Useful throughput for TCP Reno congestion avoidance algorithm

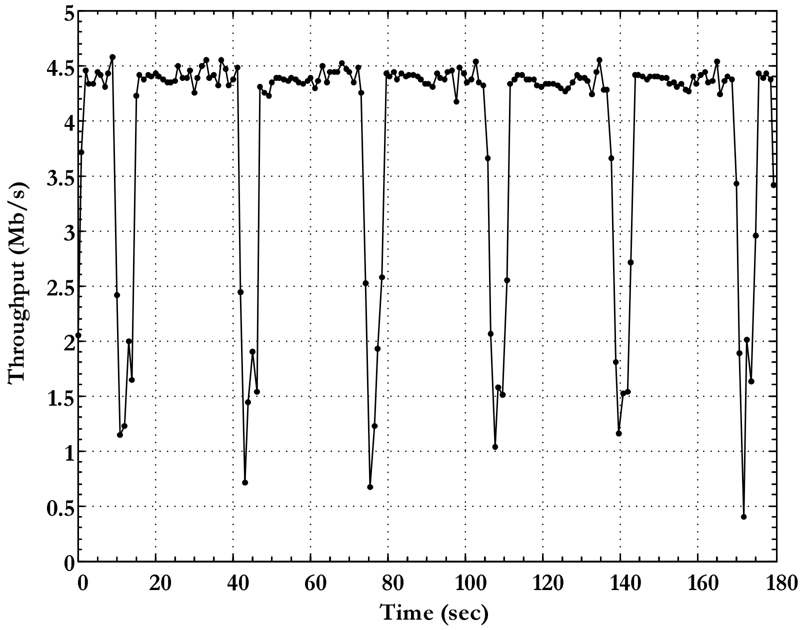

Fig A. 8 Useful throughput for TCP Vegas congestion avoidance algorithm

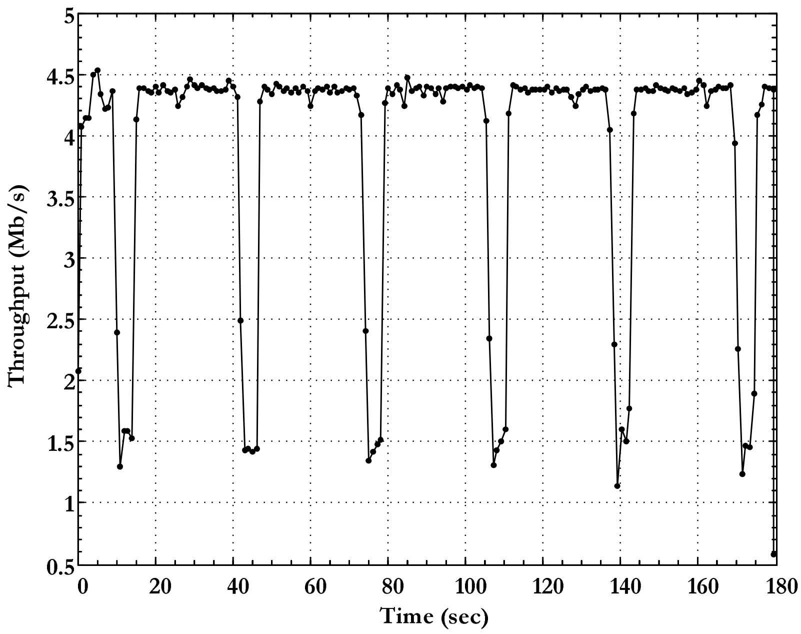

Fig A. 9 Useful throughput for TCP Veno congestion avoidance algorithm

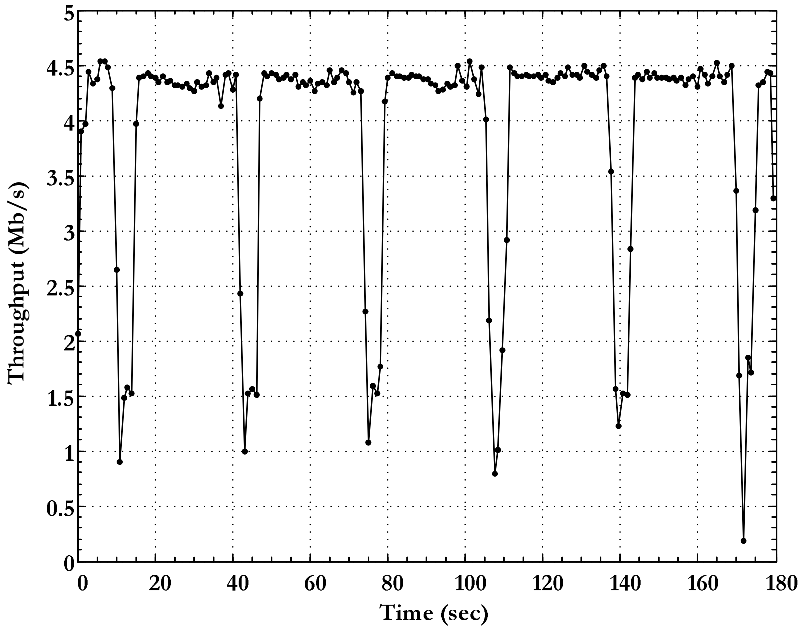

Fig A. 10 Useful throughput for TCP Westwood+ congestion avoidance algorithm

A.I.b Round Trip Time

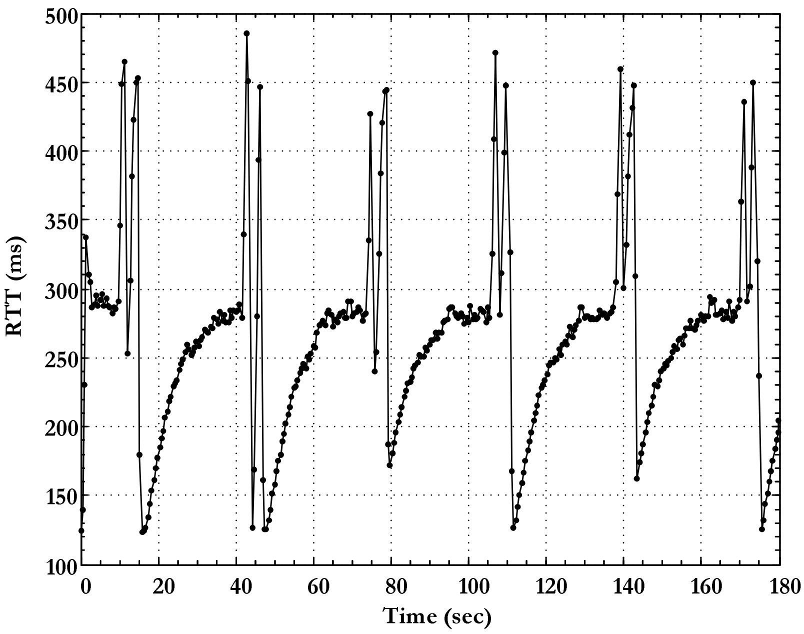

Fig A. 11 Round trip time for High Speed TCP congestion avoidance algorithm

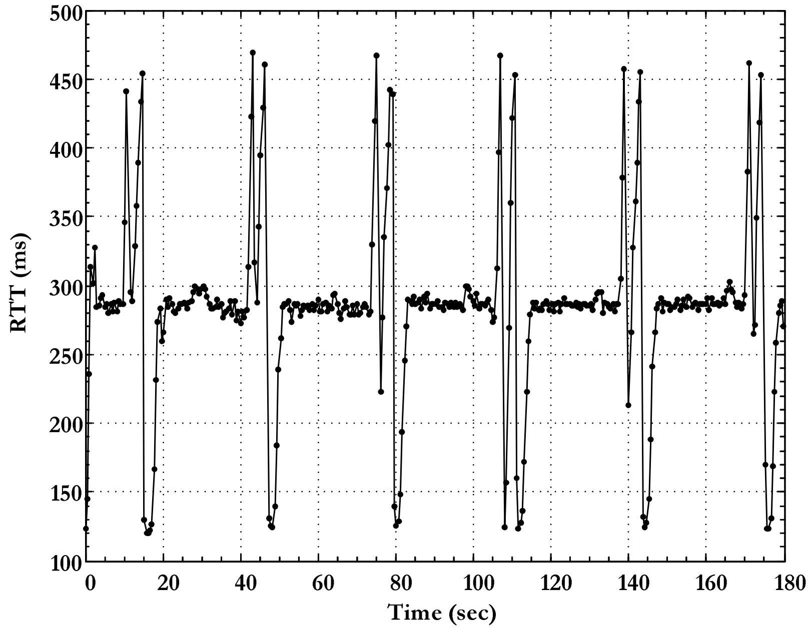

Fig A. 12 Round trip time for H-TCP (Hamilton) congestion avoidance algorithm

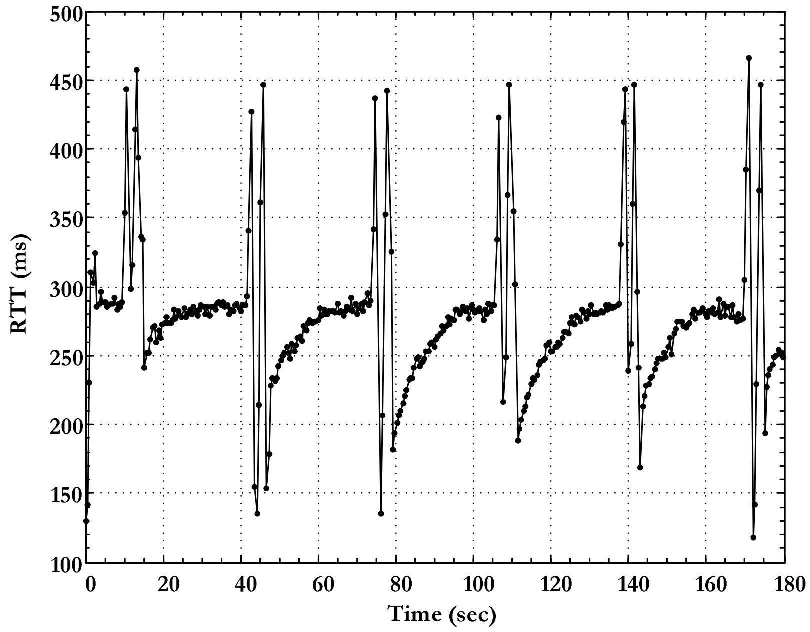

Fig A. 13 Round trip time for Scalable TCP congestion avoidance algorithm

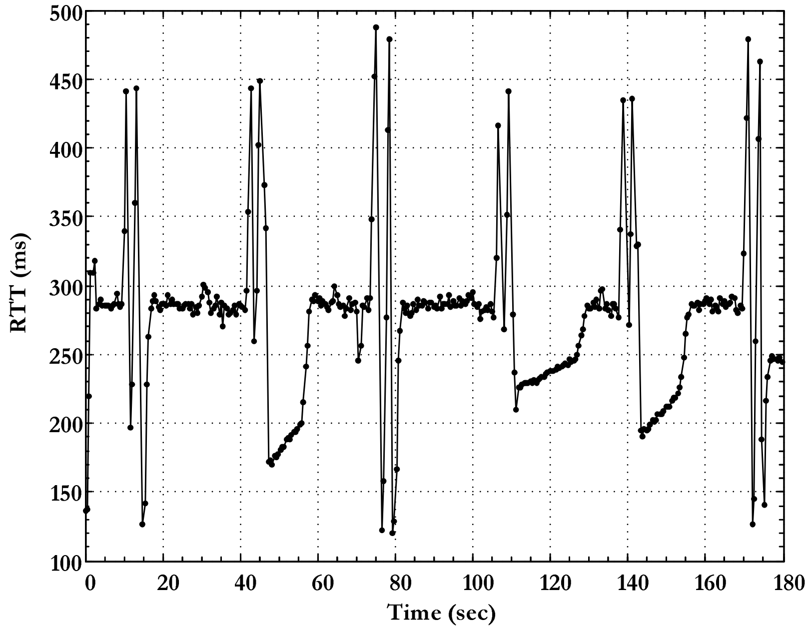

Fig A. 14 Round trip time for TCP BIC congestion avoidance algorithm

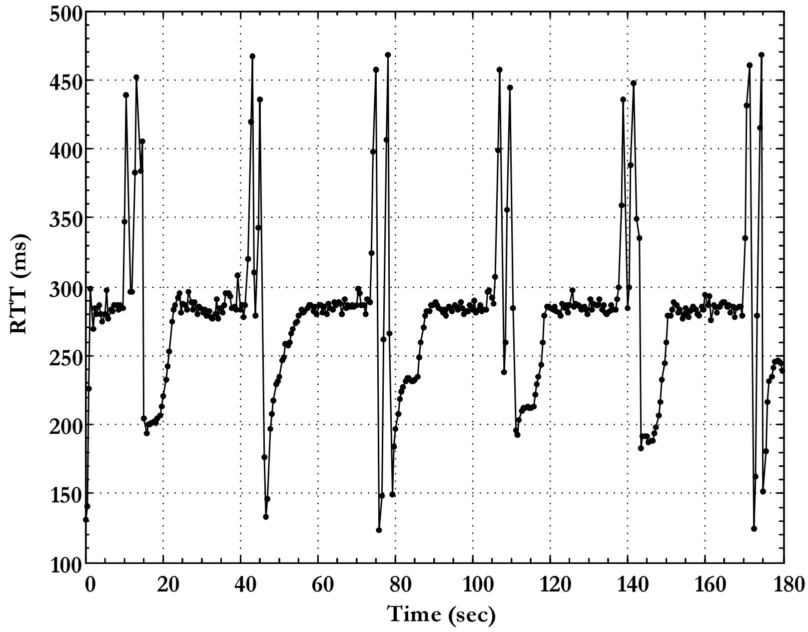

Fig A. 15 Round trip time for TCP CUBIC congestion avoidance algorithm

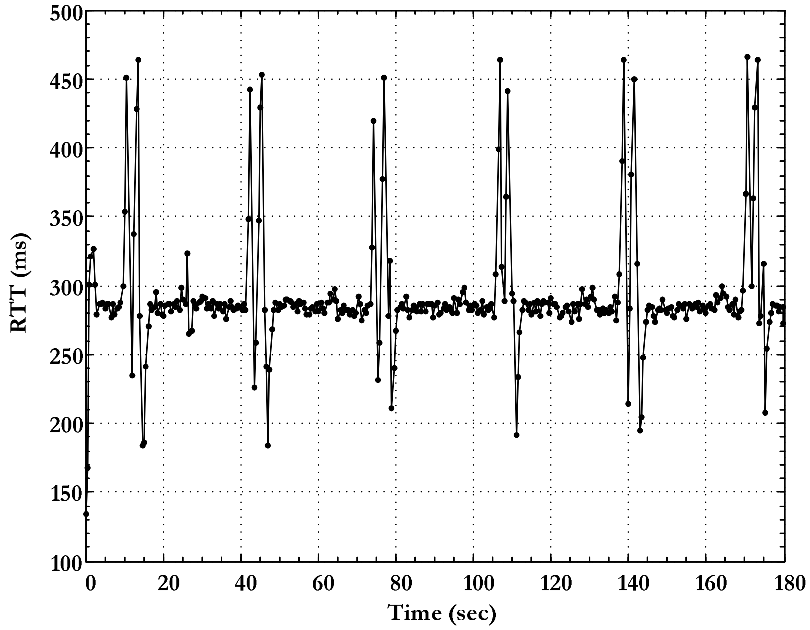

Fig A. 16 Round trip time for TCP Hybla congestion avoidance algorithm

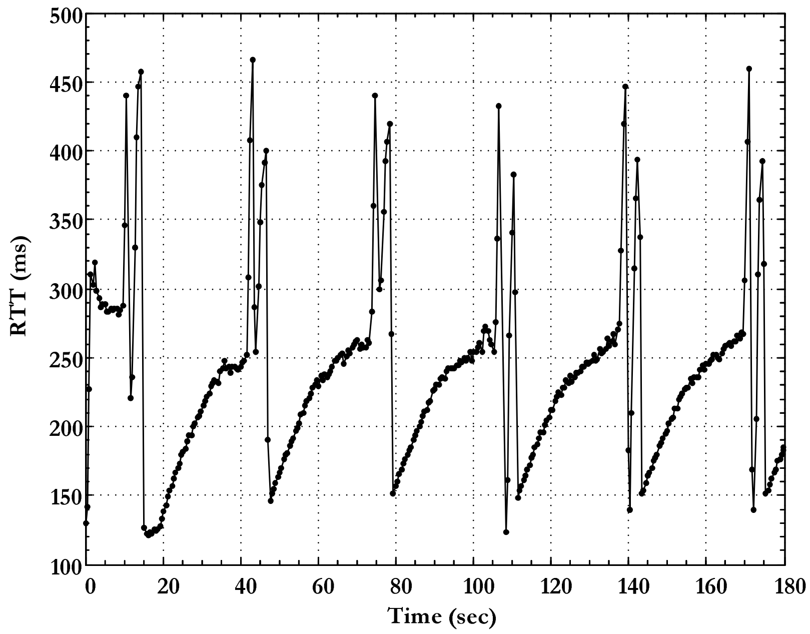

Fig A. 17 Round trip time for TCP Reno congestion avoidance algorithm

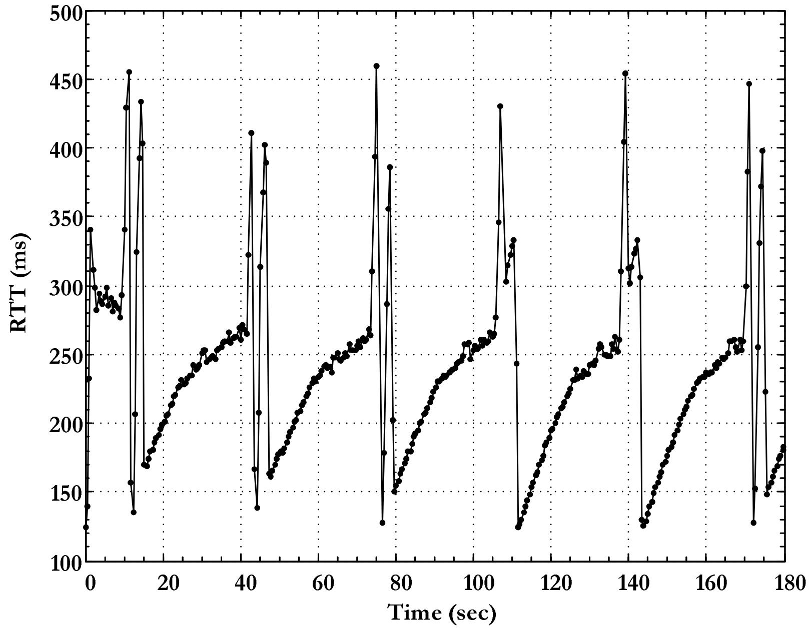

Fig A. 18 Round trip time for TCP Vegas congestion avoidance algorithm

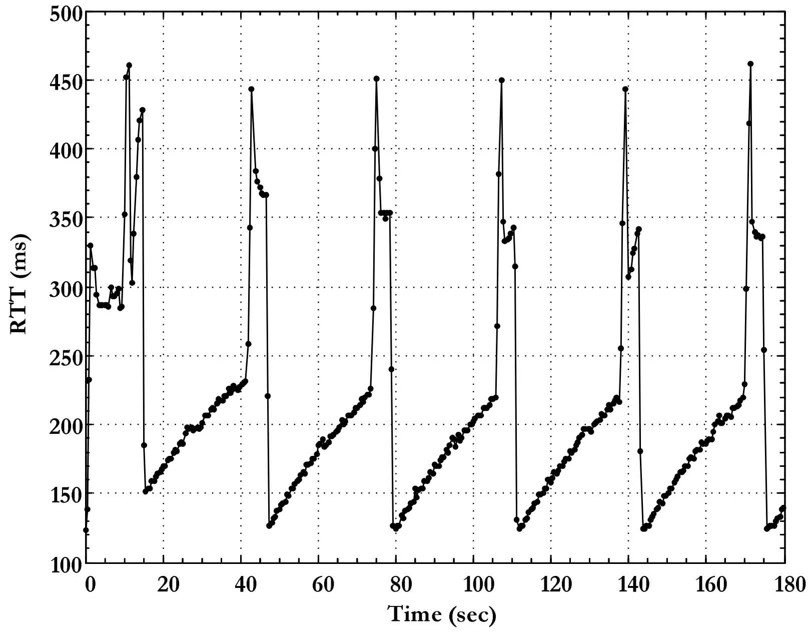

Fig A. 19 Round trip time for TCP Veno congestion avoidance algorithm

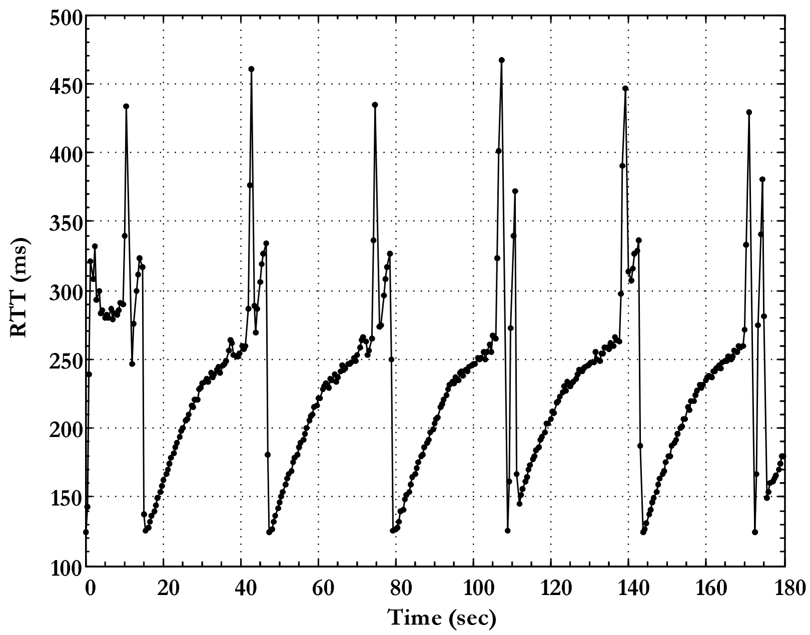

Fig A. 20 Round trip time for TCP Westwood+ congestion avoidance algorithm

A.I.c Loss Events

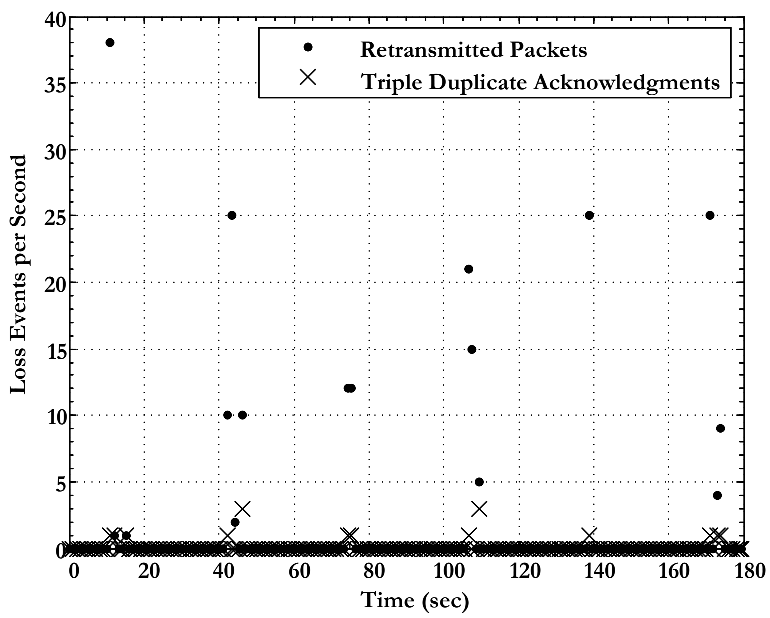

Fig A. 21 Retransmitted packets and triple duplicate acknowledgments for High Speed TCP congestion avoidance algorithm

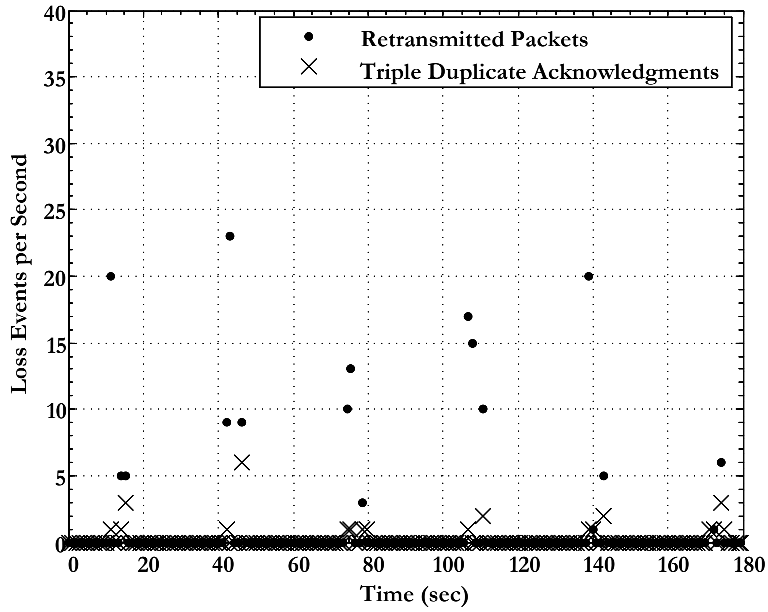

Fig A. 22 Retransmitted packets and triple duplicate acknowledgments for H-TCP (Hamilton) congestion avoidance algorithm

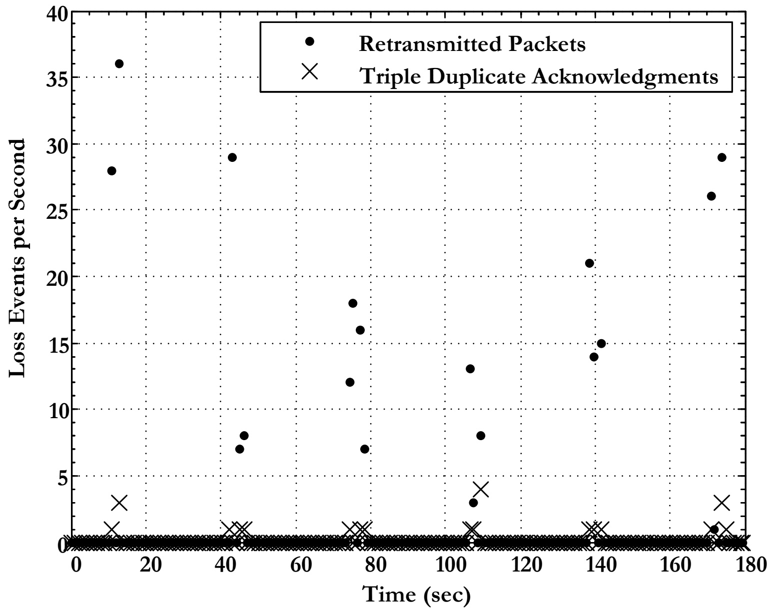

Fig A. 23 Retransmitted packets and triple duplicate acknowledgments for Scalable TCP congestion avoidance algorithm

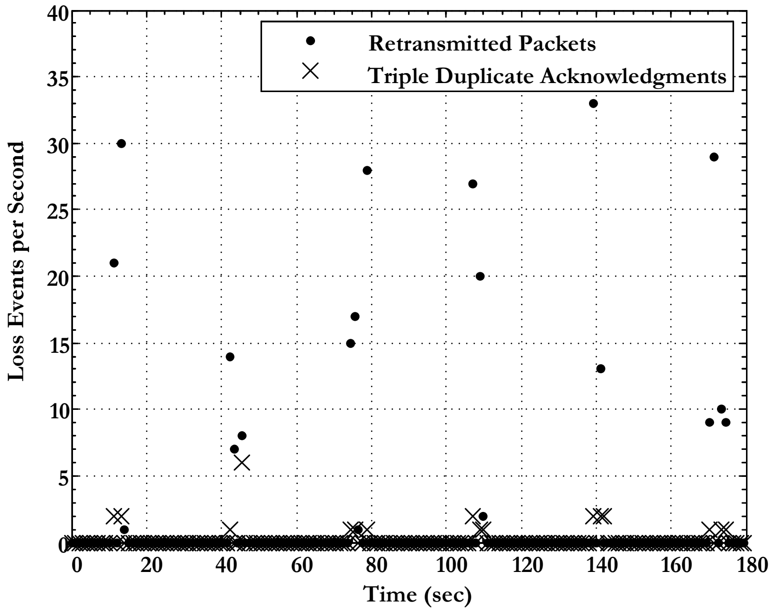

Fig A. 24 Retransmitted packets and triple duplicate acknowledgments for TCP BIC congestion avoidance algorithm

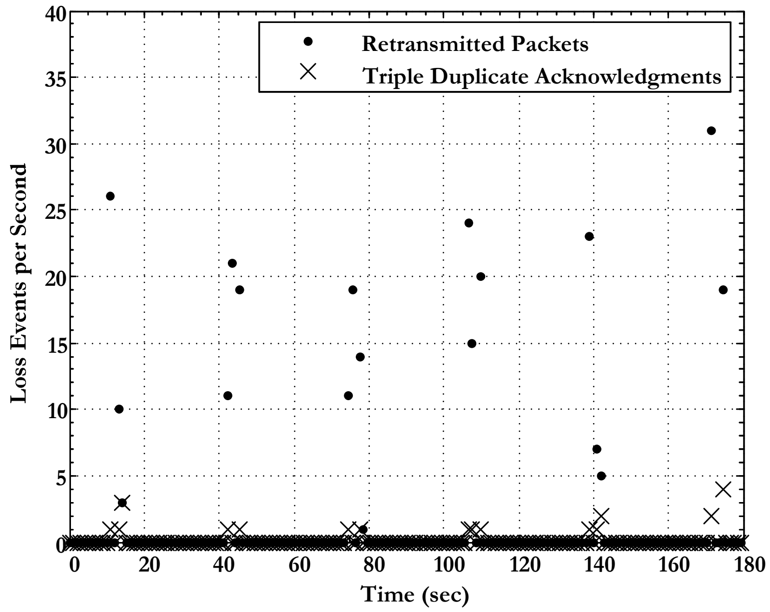

Fig A. 25 Retransmitted packets and triple duplicate acknowledgments for TCP CUBIC congestion avoidance algorithm

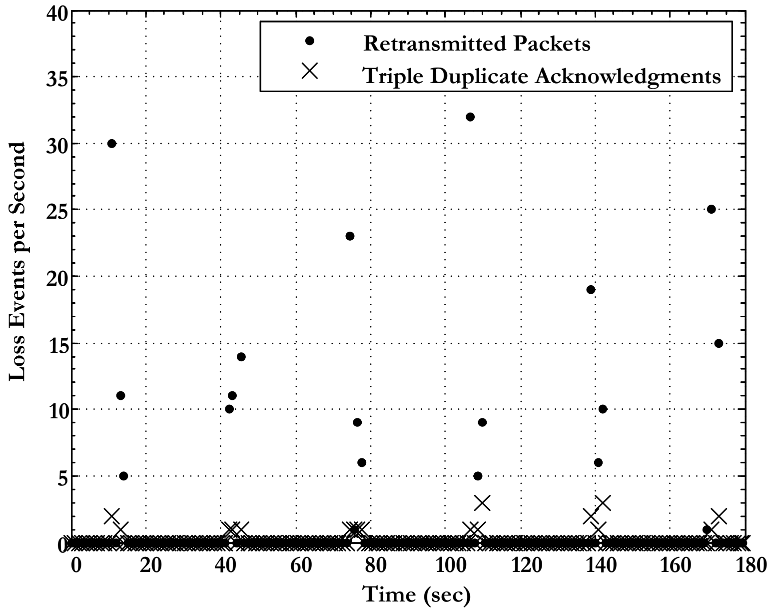

Fig A. 26 Retransmitted packets and triple duplicate acknowledgments for TCP Hybla congestion avoidance algorithm

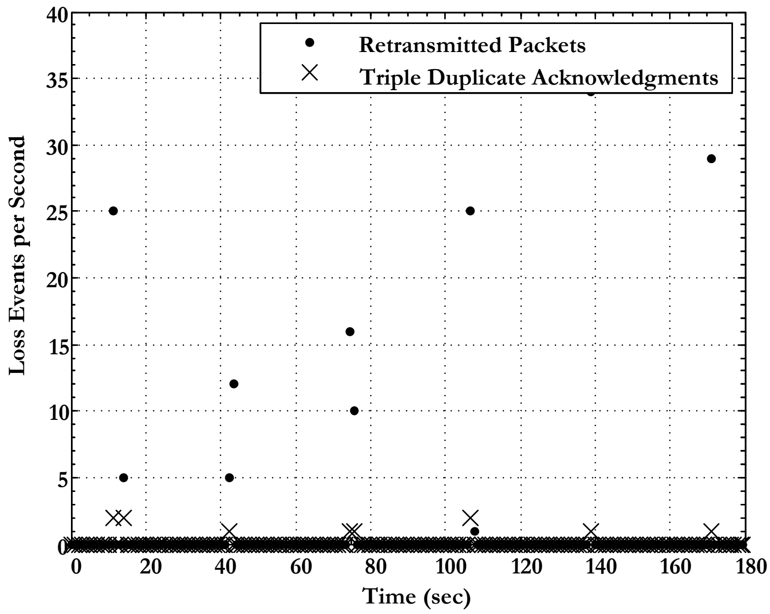

Fig A. 27 Retransmitted packets and triple duplicate acknowledgments for TCP Reno congestion avoidance algorithm

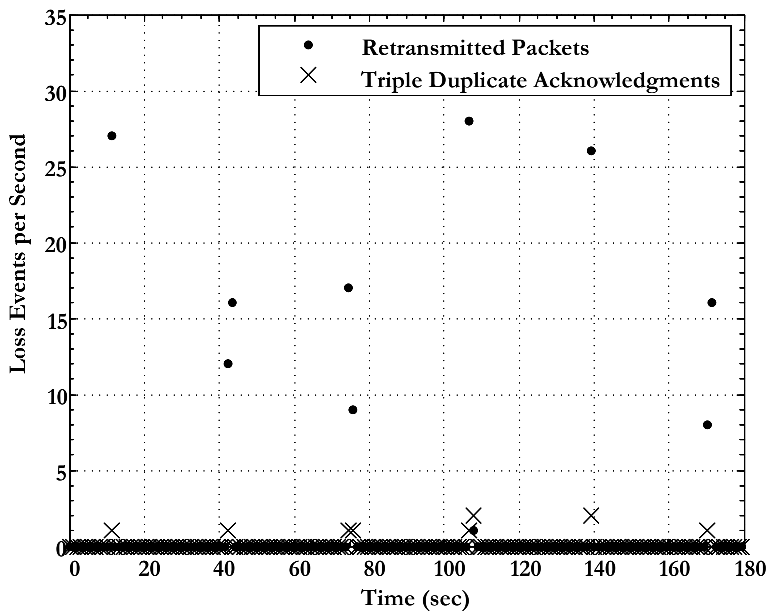

Fig A. 28 Retransmitted packets and triple duplicate acknowledgments for TCP Vegas congestion avoidance algorithm

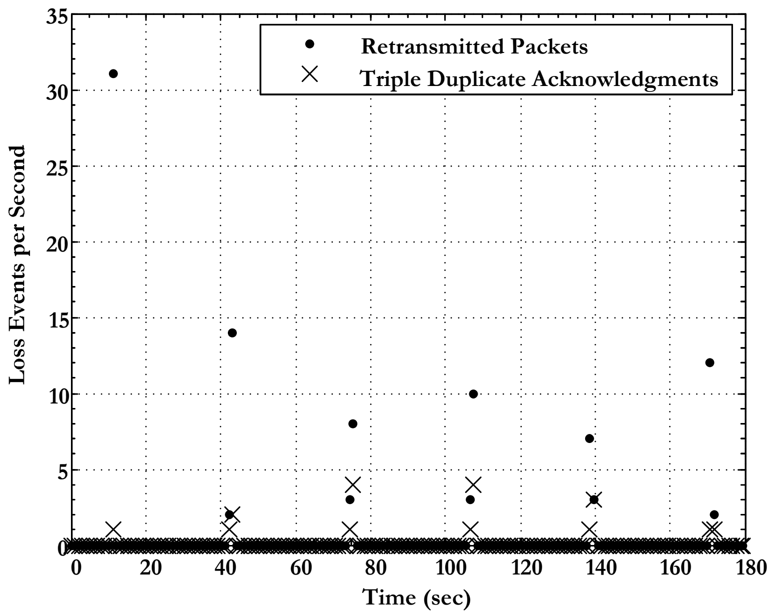

Fig A. 29 Retransmitted packets and triple duplicate acknowledgments for TCP Veno congestion avoidance algorithm

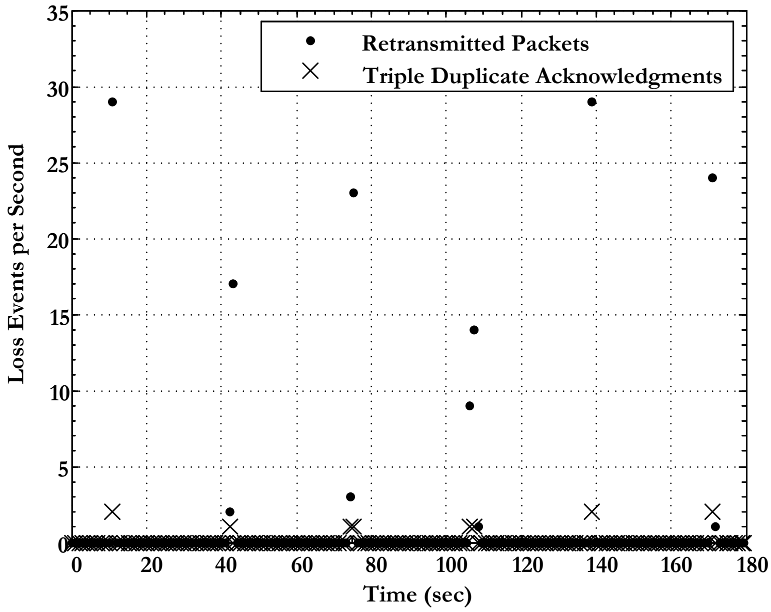

Fig A. 30 Retransmitted packets and triple duplicate acknowledgments for TCP Westwood+ congestion avoidance algorithm

A.I.d Sequence Number (Overall TCP Connection)

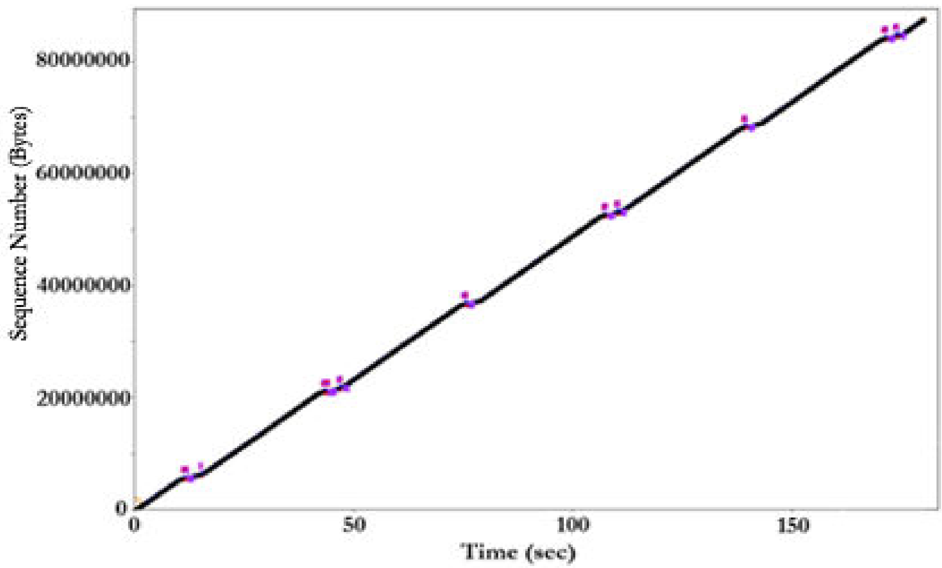

Fig A. 31 Sequence number (overall TCP connection) for High Speed TCP congestion avoidance algorithm

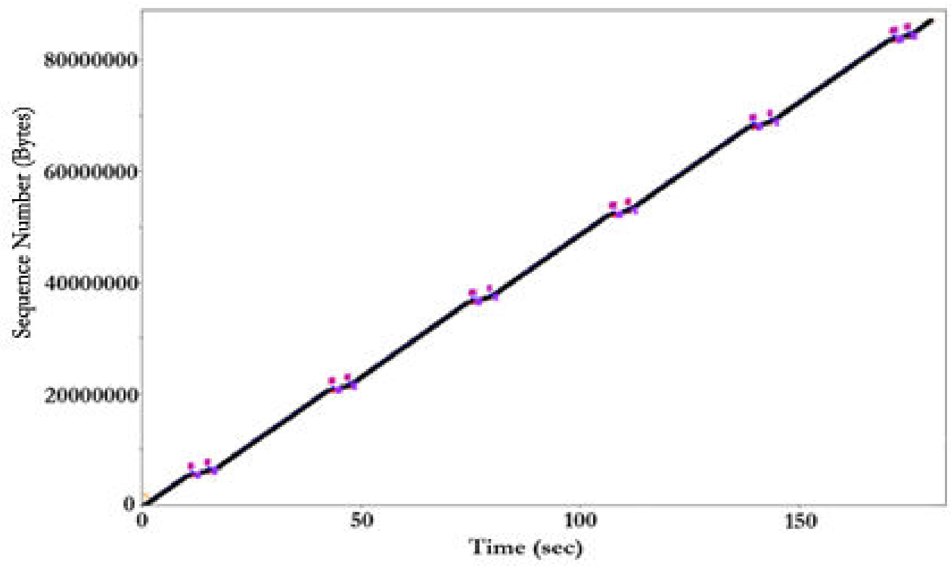

Fig A. 32 Sequence number (overall TCP connection) for H-TCP (Hamilton) congestion avoidance algorithm

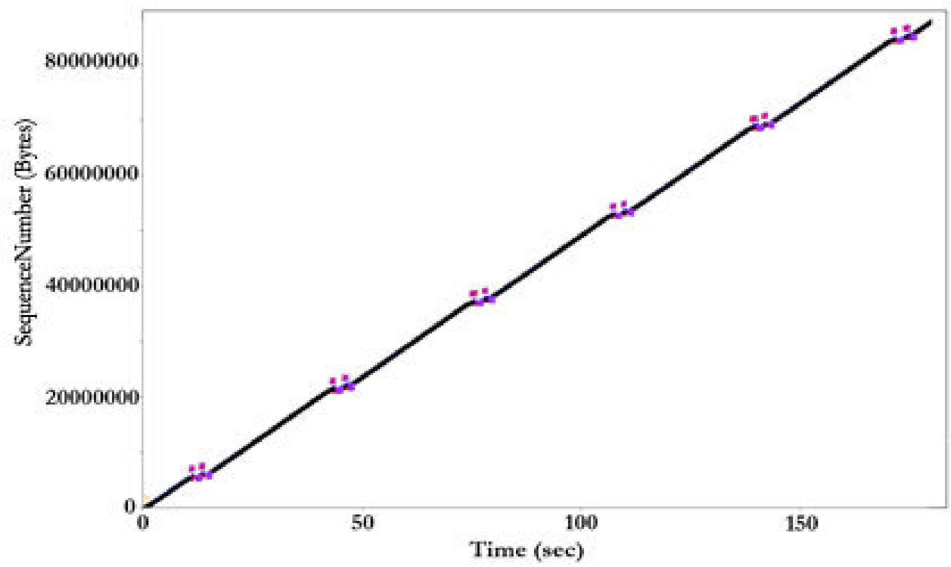

Fig A. 33 Sequence number (overall TCP connection) for Scalable TCP congestion avoidance algorithm

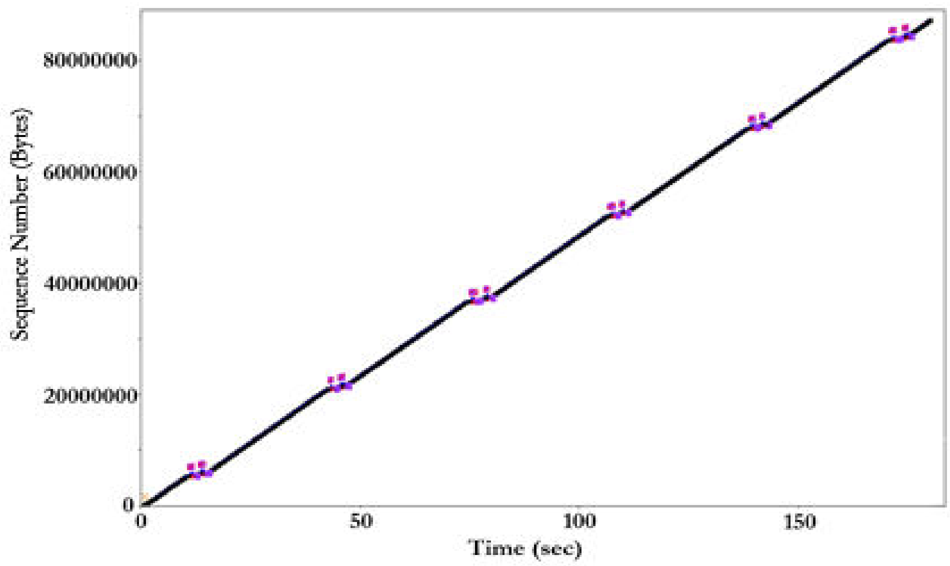

Fig A. 34 Sequence number (overall TCP connection) for TCP BIC congestion avoidance algorithm

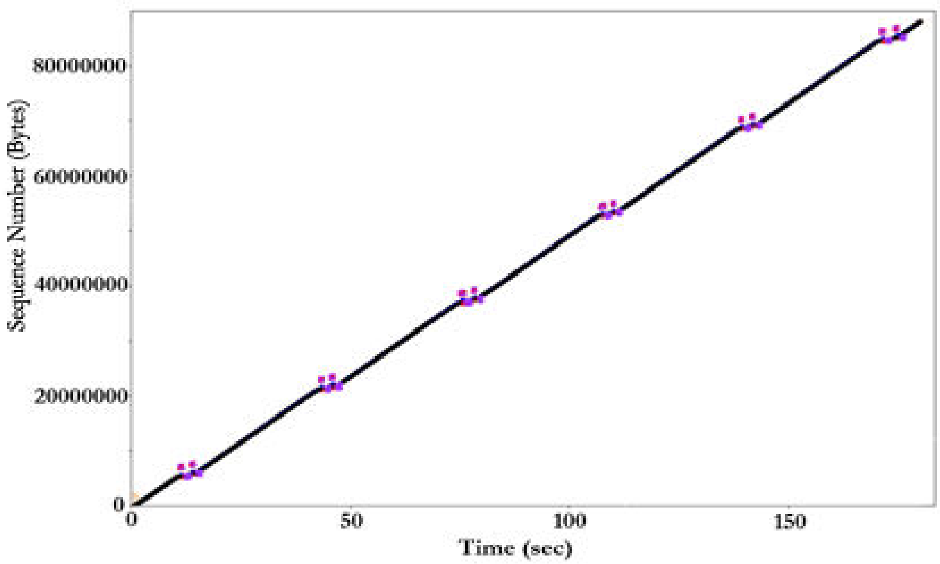

Fig A. 35 Sequence number (overall TCP connection) for TCP CUBIC congestion avoidance algorithm

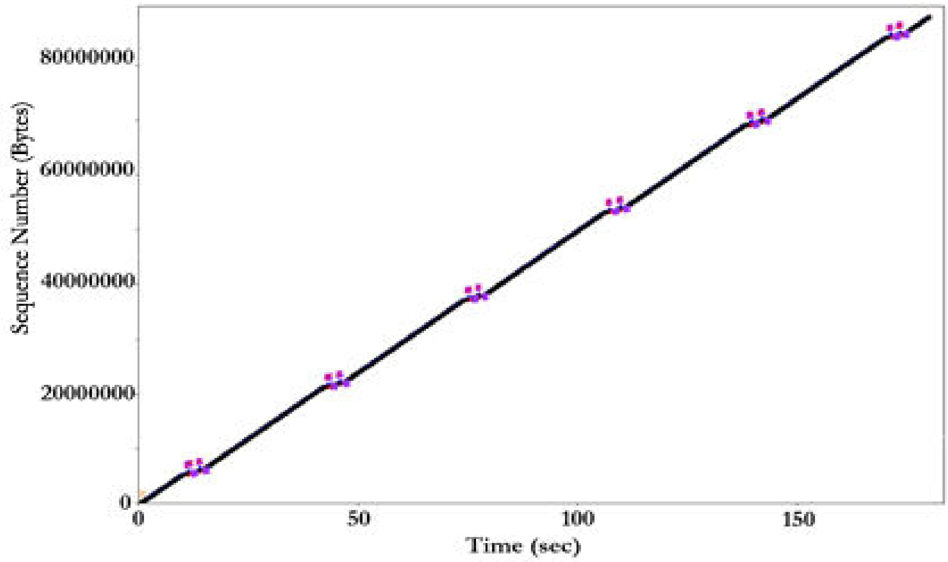

Fig A. 36 Sequence number (overall TCP connection) for TCP Hybla congestion avoidance algorithm

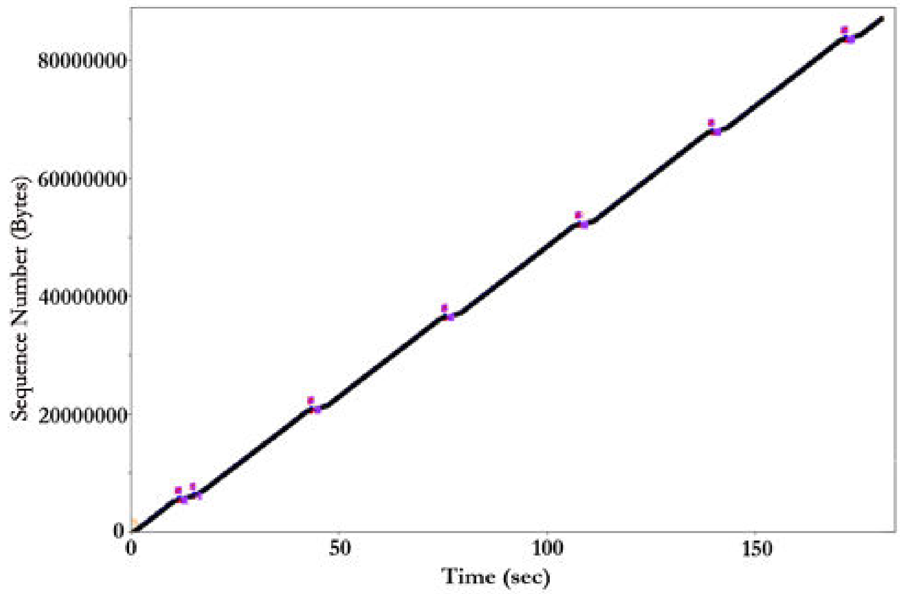

Fig A. 37 Sequence number (overall TCP connection) for TCP Reno congestion avoidance algorithm

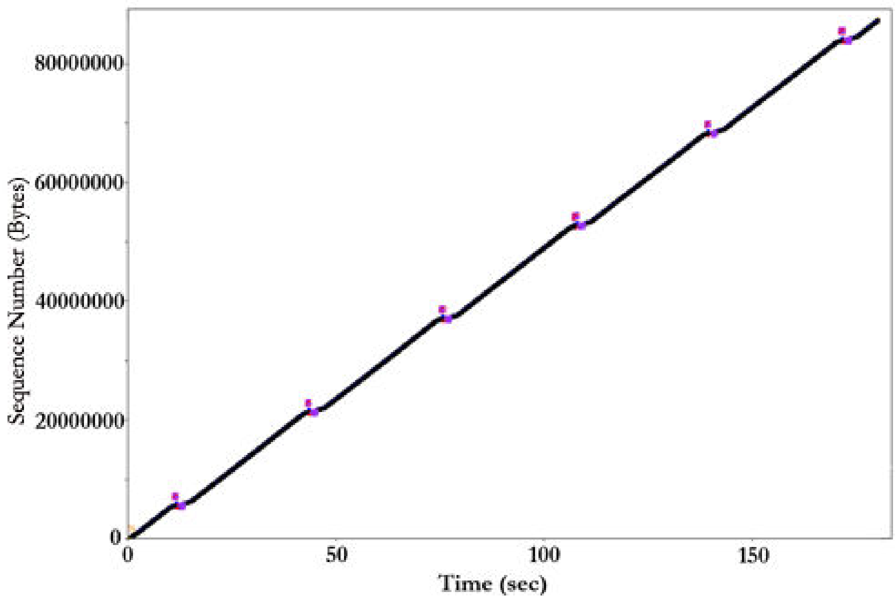

Fig A. 38 Sequence number (overall TCP connection) for TCP Vegas congestion avoidance algorithm

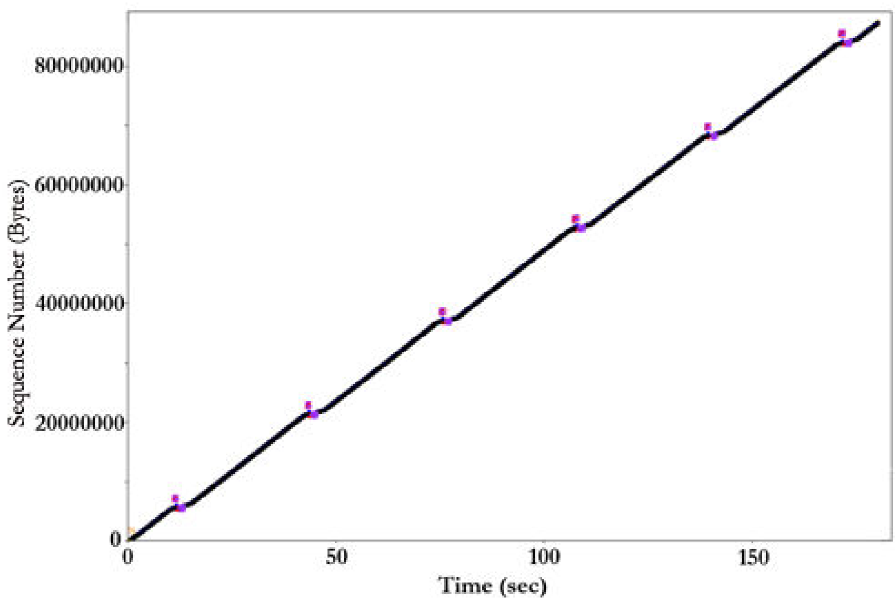

Fig A. 39 Sequence number (overall TCP connection) for TCP Veno congestion avoidance algorithm

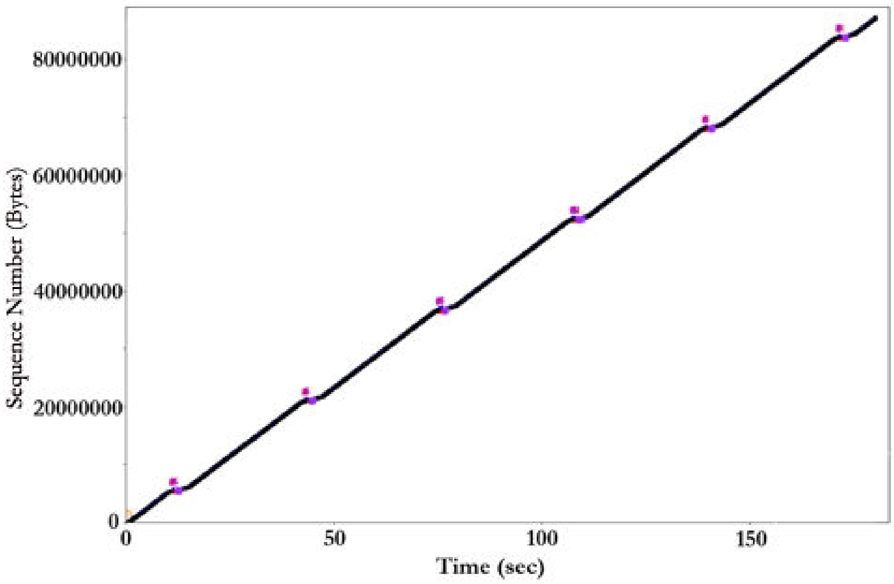

Fig A. 40 Sequence number (overall TCP connection) for TCP Westwood+ congestion avoidance algorithm

A.I.e Sequence Number (Part of TCP Connection)

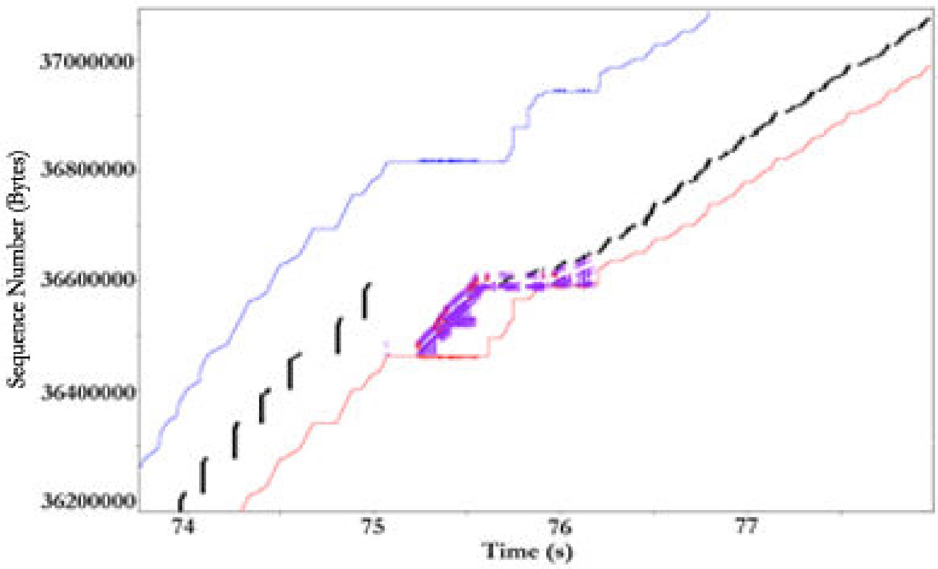

Fig A. 41 Sequence number (part of TCP connection) for High Speed TCP congestion avoidance algorithm

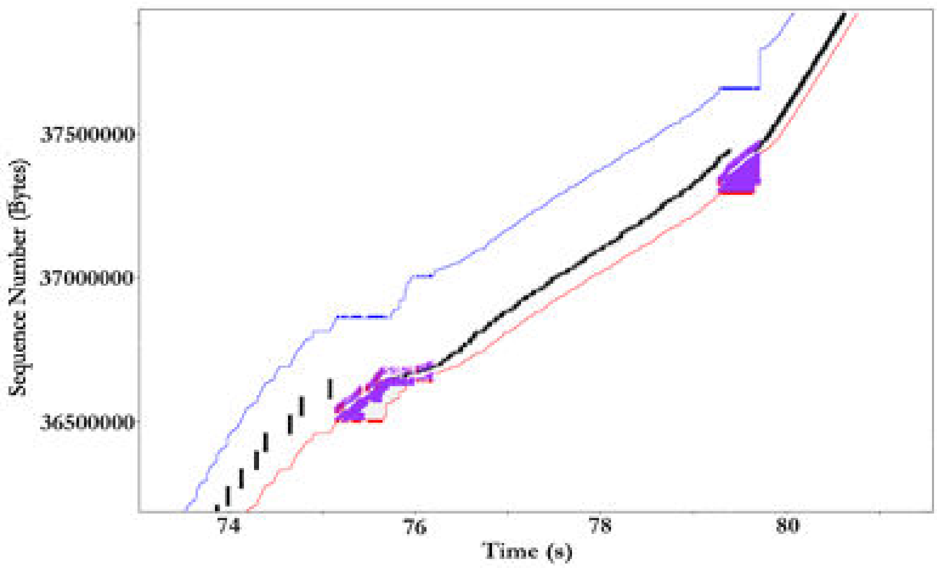

Fig A. 42 Sequence number (part of TCP connection) for H-TCP (Hamilton) congestion avoidance algorithm

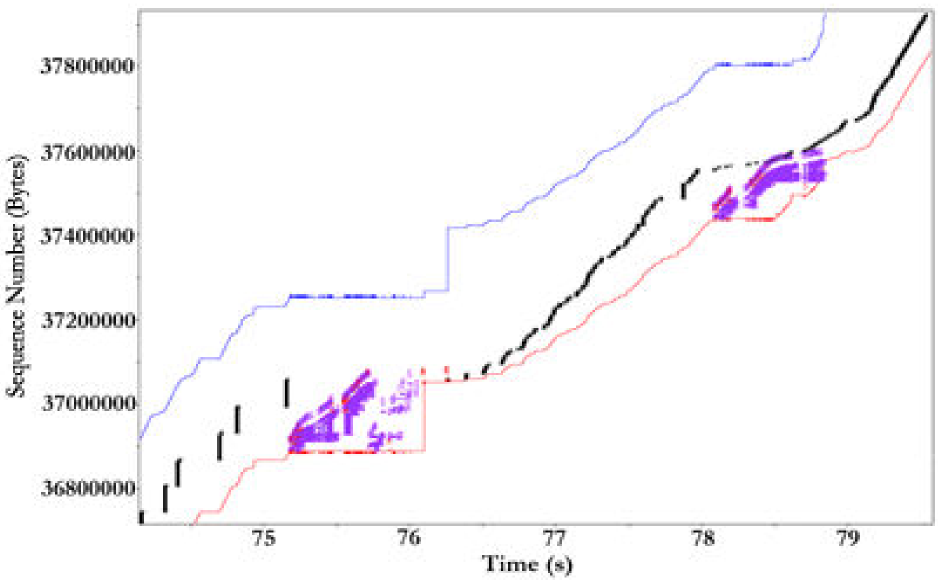

Fig A. 43 Sequence number (part of TCP connection) for Scalable TCP congestion avoidance algorithm

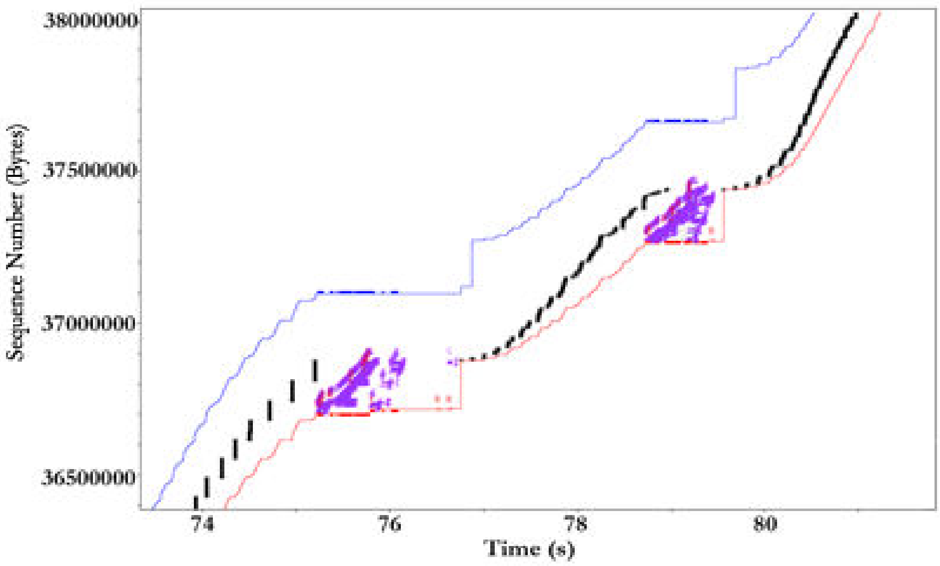

Fig A. 44 Sequence number (part of TCP connection) for TCP BIC congestion avoidance algorithm

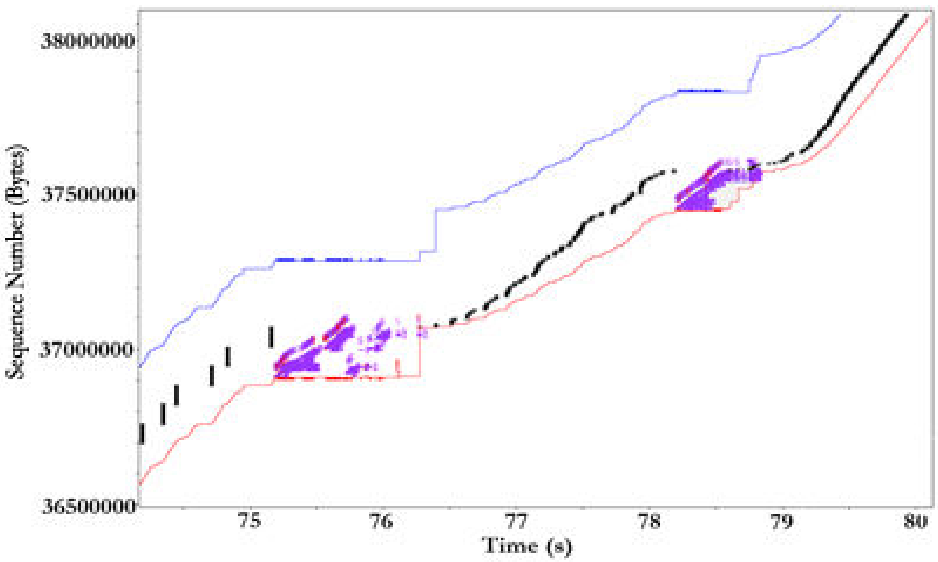

Fig A. 45 Sequence number (part of TCP connection) for TCP CUBIC congestion avoidance algorithm

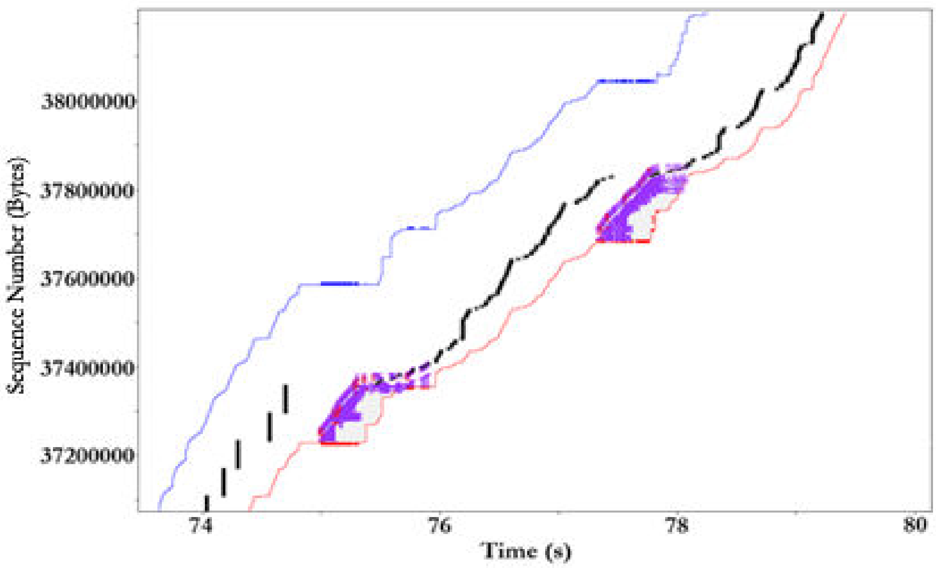

Fig A. 46 Sequence number (part of TCP connection) for TCP Hybla congestion avoidance algorithm

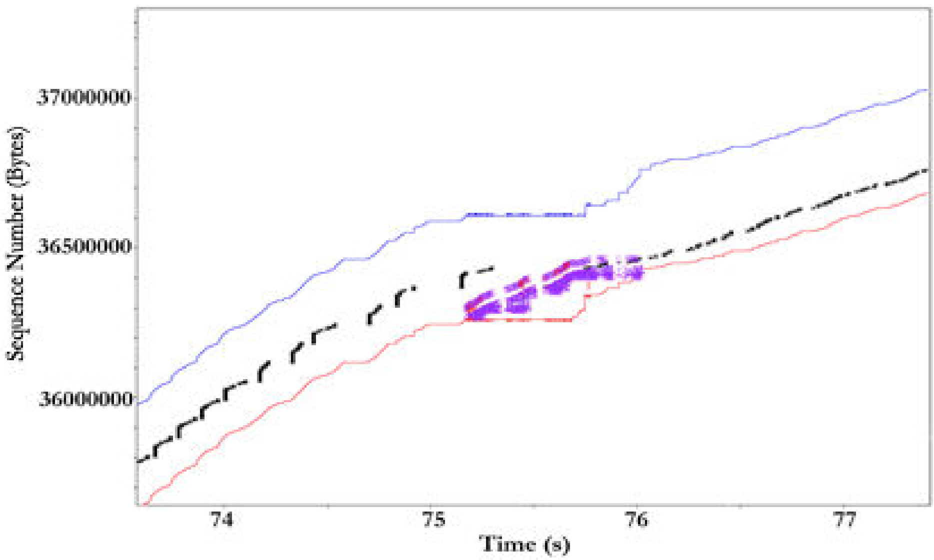

Fig A. 47 Sequence number (part of TCP connection) for TCP Reno congestion avoidance algorithm

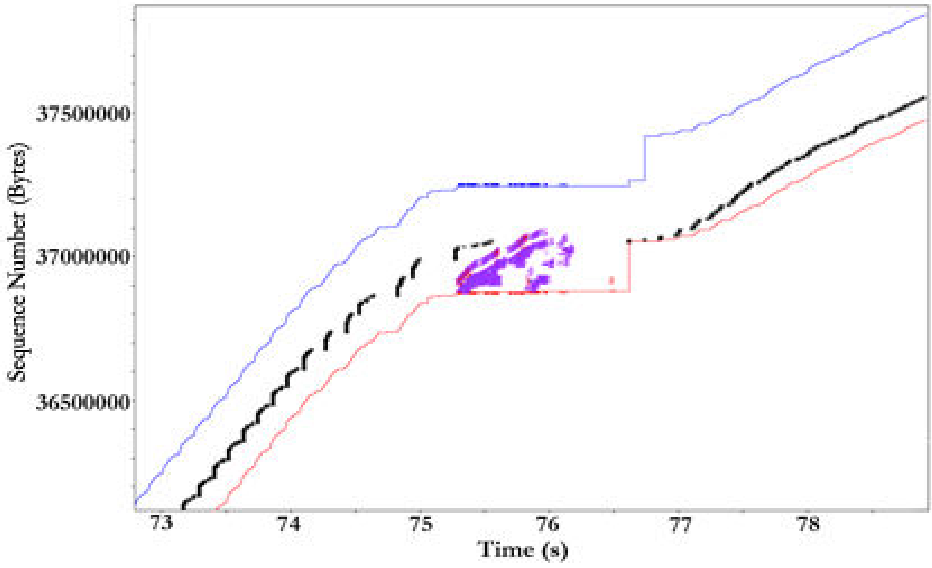

Fig A. 48 Sequence number (part of TCP connection) for TCP Vegas congestion avoidance algorithm

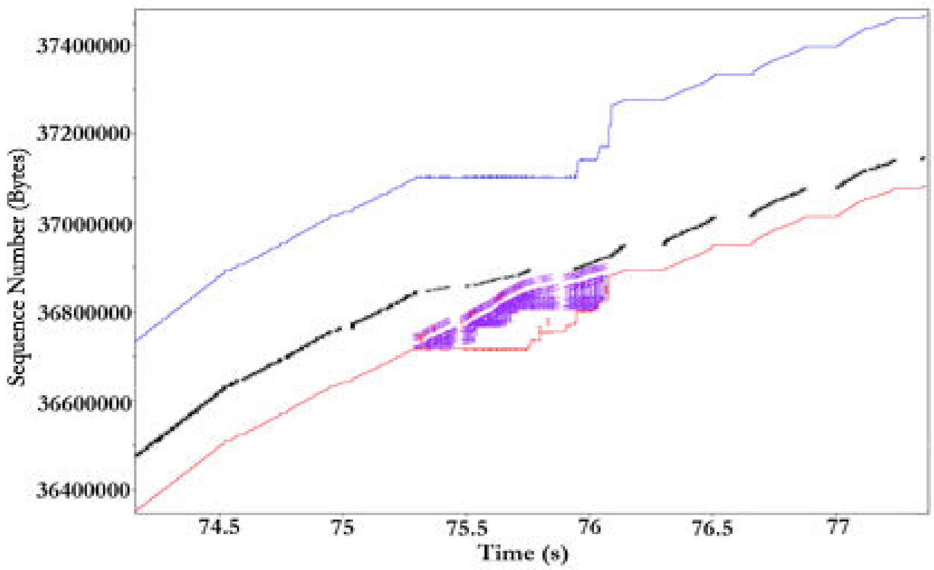

Fig A. 49 Sequence number (part of TCP connection) for TCP Veno congestion avoidance algorithm

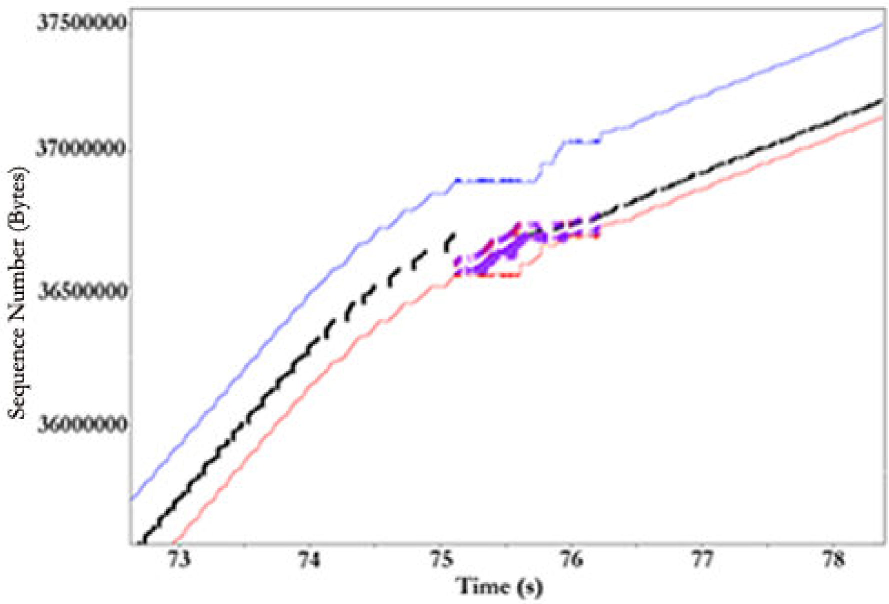

Fig A. 50 Sequence number (part of TCP connection) for TCP Westwood+ congestion avoidance algorithm

A.II Experimental Results of End-to-end Network Performance Evaluation for Different IP/DVB Gateway Bandwidth Allocation for TCP Traffic

A.II.a Instantaneous Useful Throughput

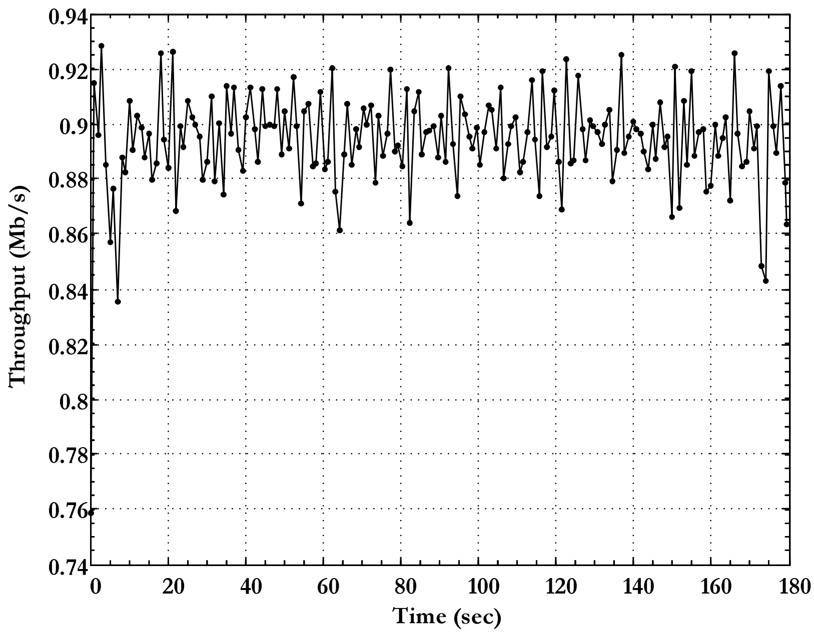

Fig A. 51 Useful throughput for 1Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

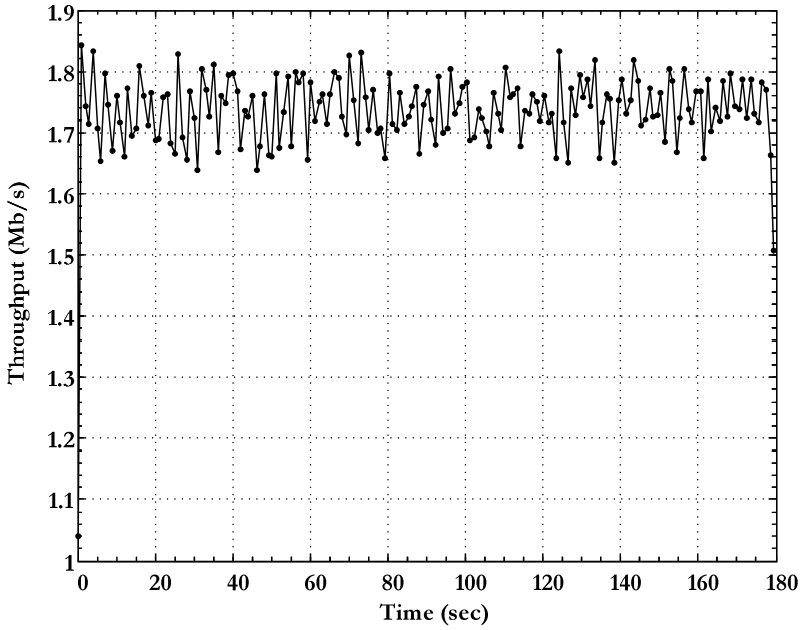

Fig A. 52 Useful throughput for 2Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

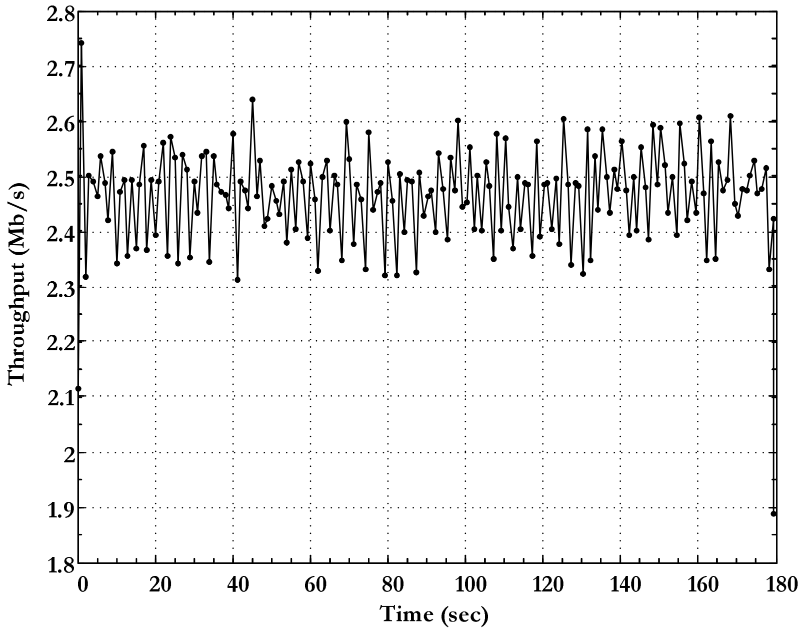

Fig A. 53 Useful throughput for 3Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

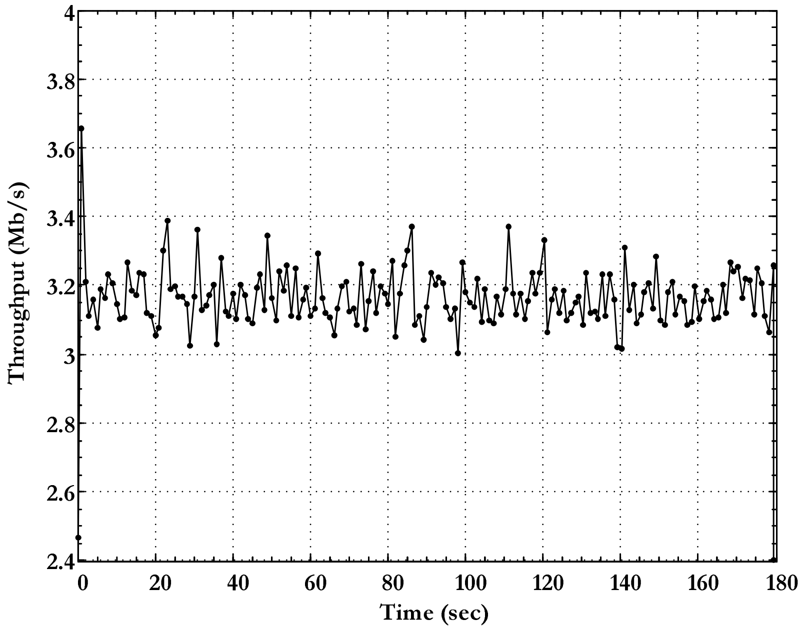

Fig A. 54 Useful throughput for 4Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

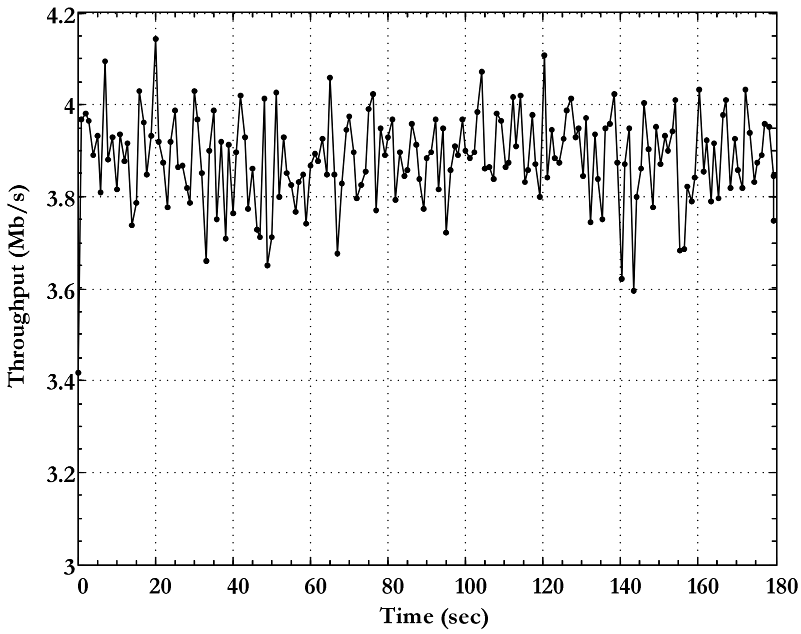

Fig A. 55 Useful throughput for 5Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

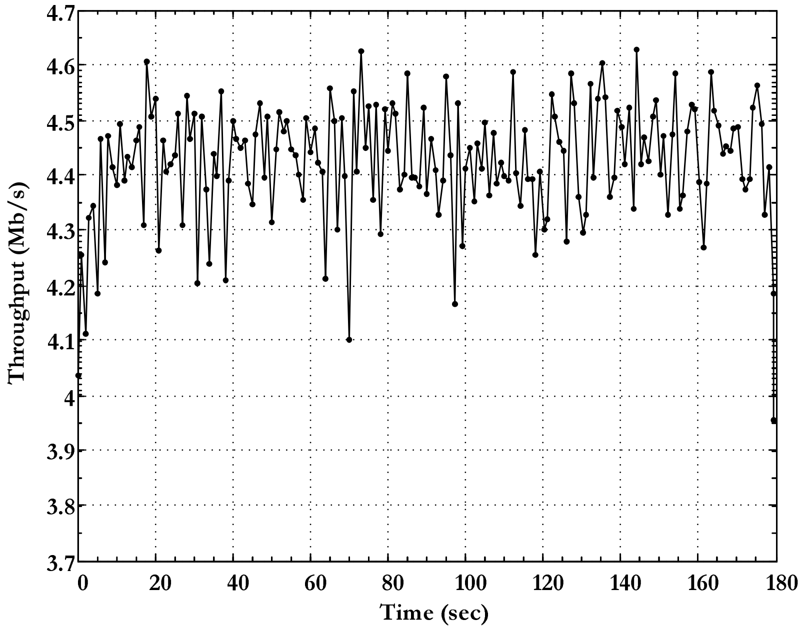

Fig A. 56 Useful throughput for 6Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

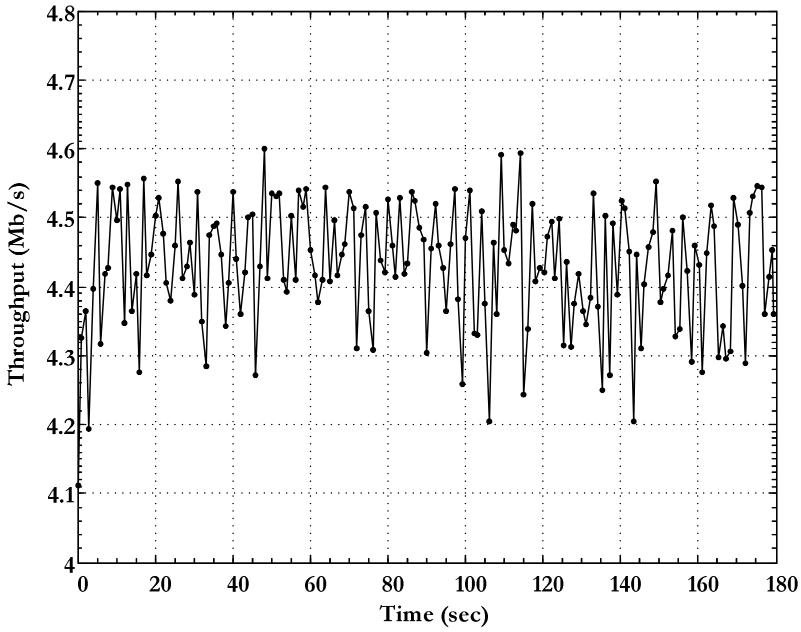

Fig A. 57 Useful throughput for 7Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

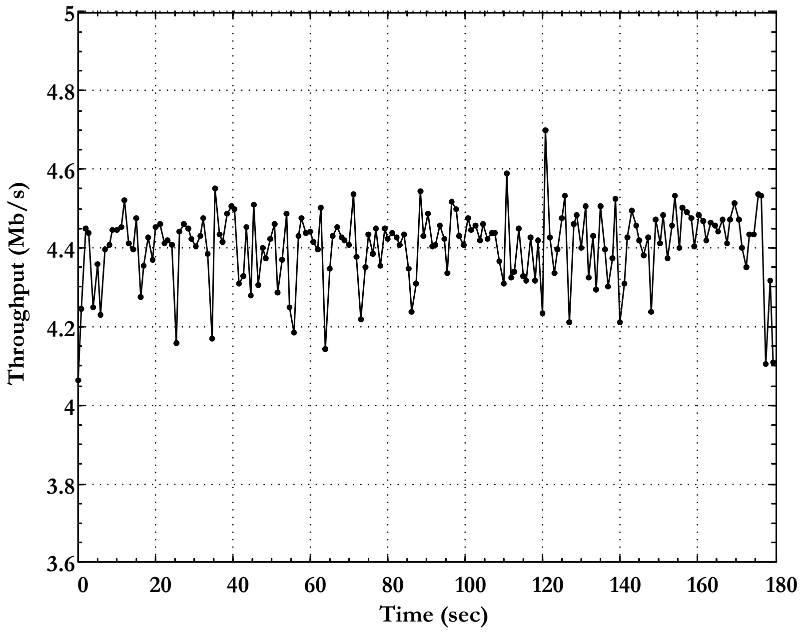

Fig A. 58 Useful throughput for 8Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

A.II.b Round Trip Time

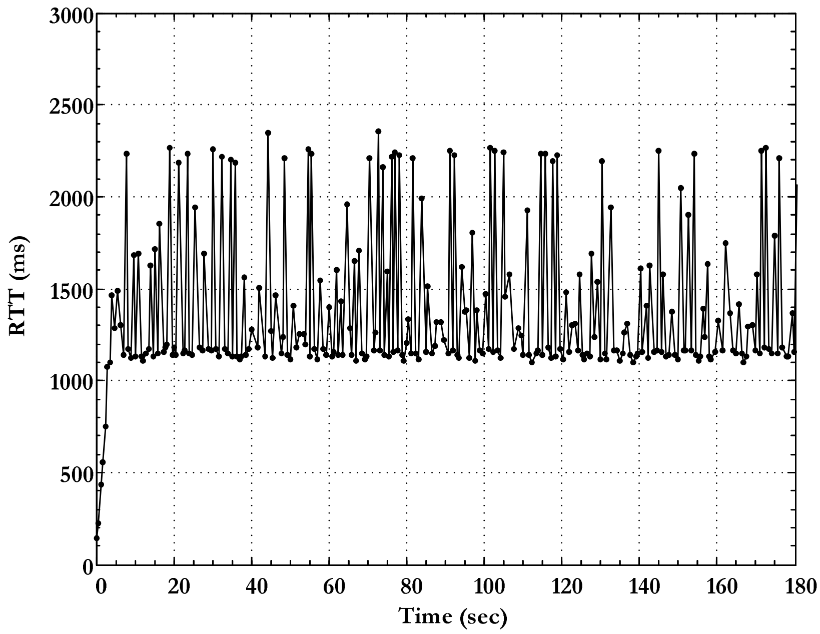

Fig A. 59 Round trip time for 1Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

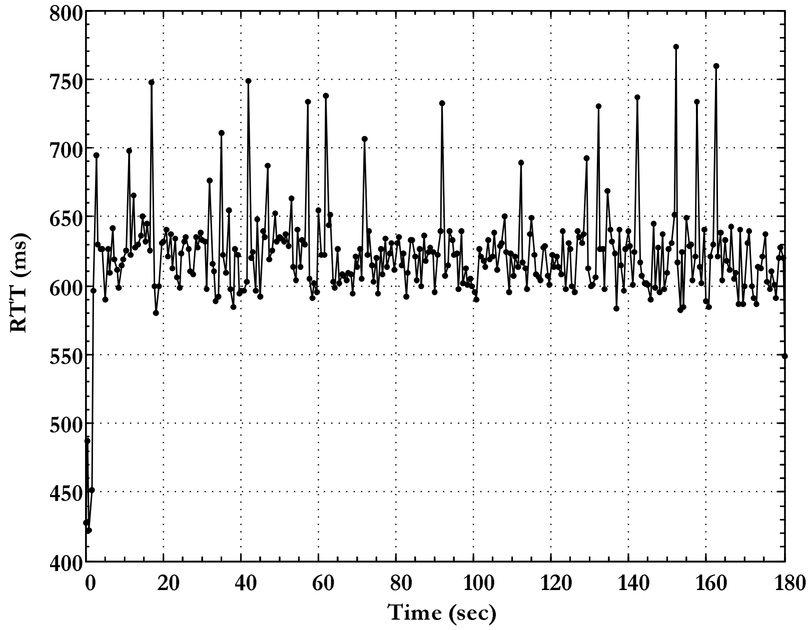

Fig A. 60 Round trip time for 2Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

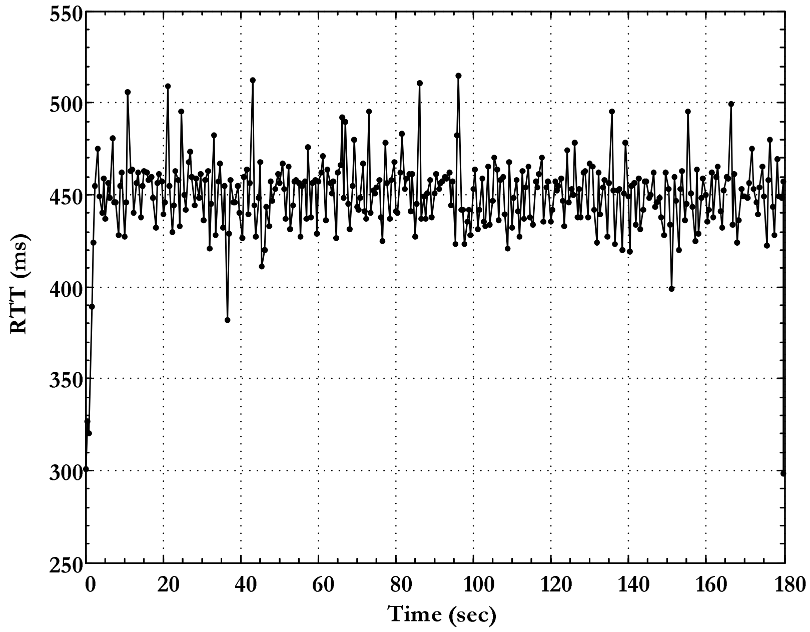

Fig A. 61 Round trip time for 3Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

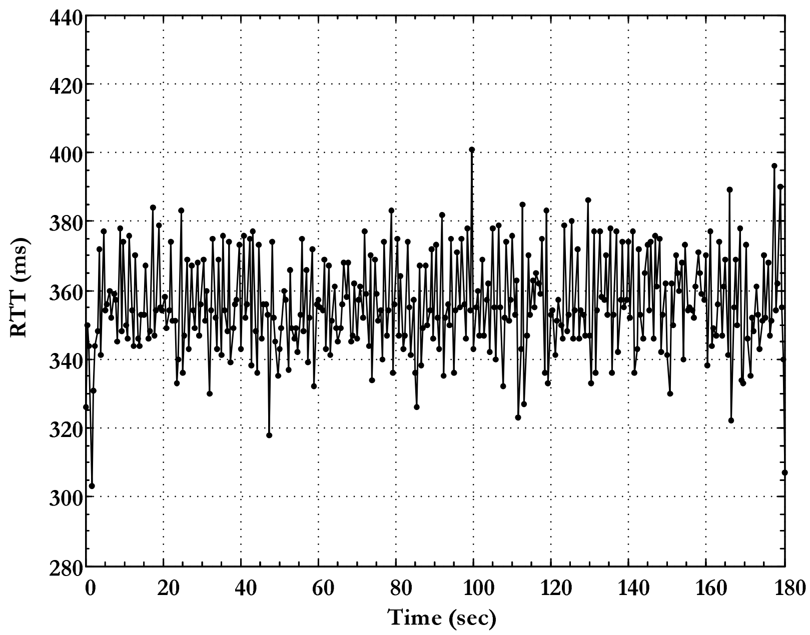

Fig A. 62 Round trip time for 4Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

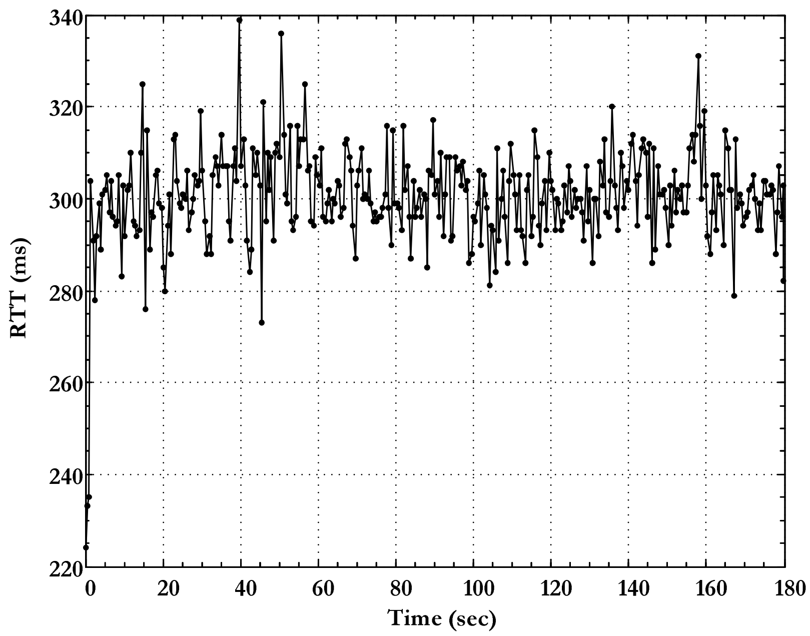

Fig A. 63 Round trip time for 5Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

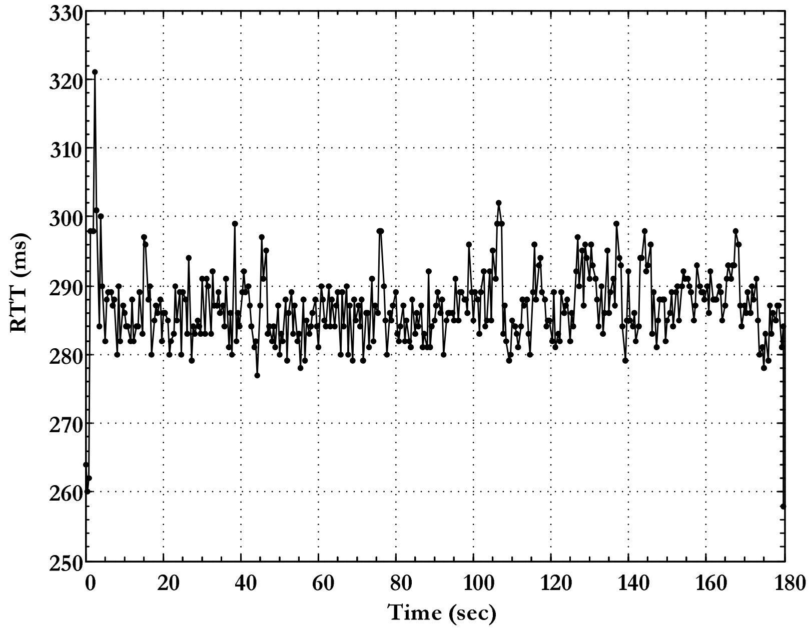

Fig A. 64 Round trip time for 6Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

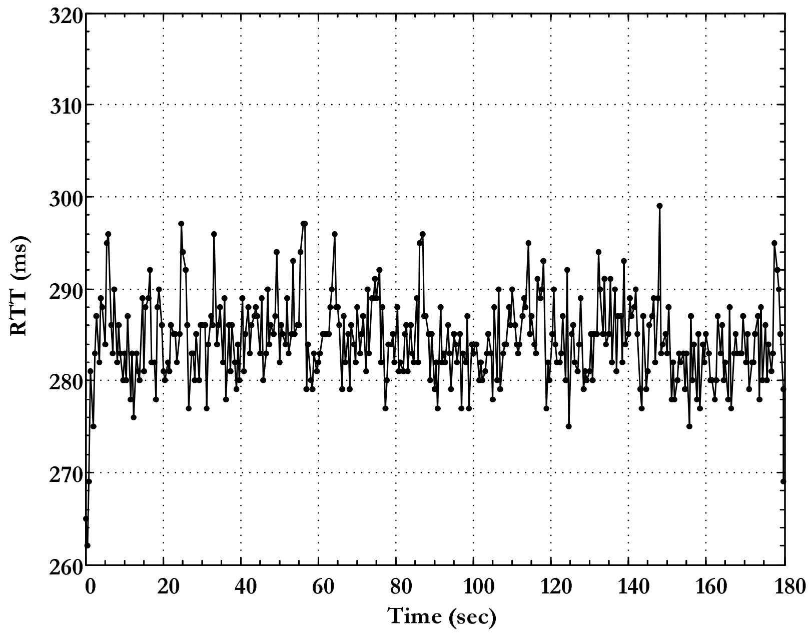

Fig A. 65 Round trip time for 7Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

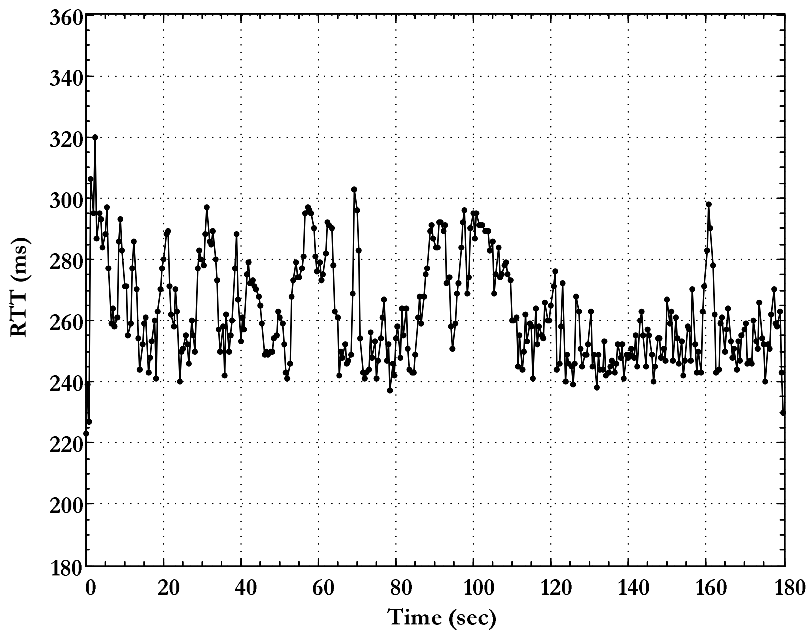

Fig A. 66 Round trip time for 8Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

A.II.c Sequence Number (Overall TCP Connection)

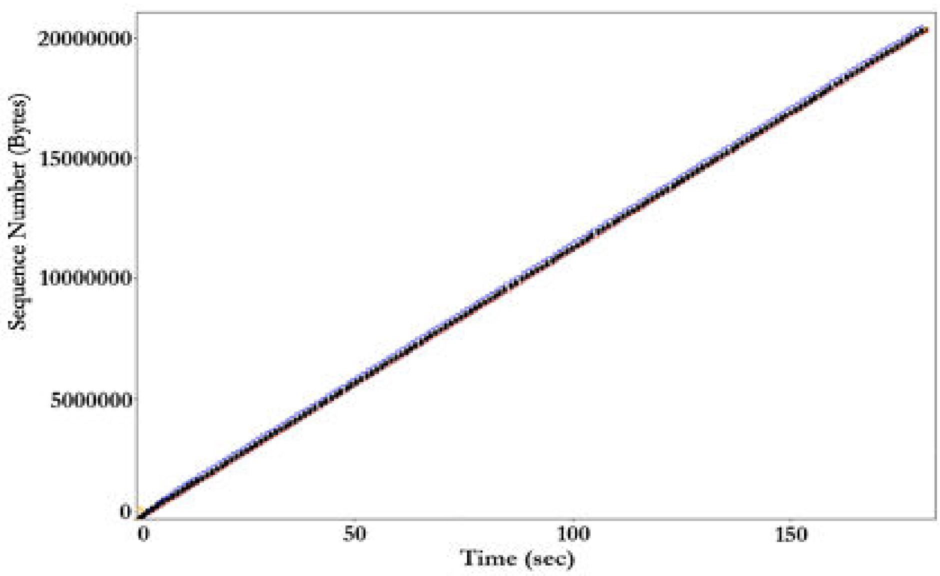

Fig A. 67 Sequence number (overall TCP connection) for 1Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

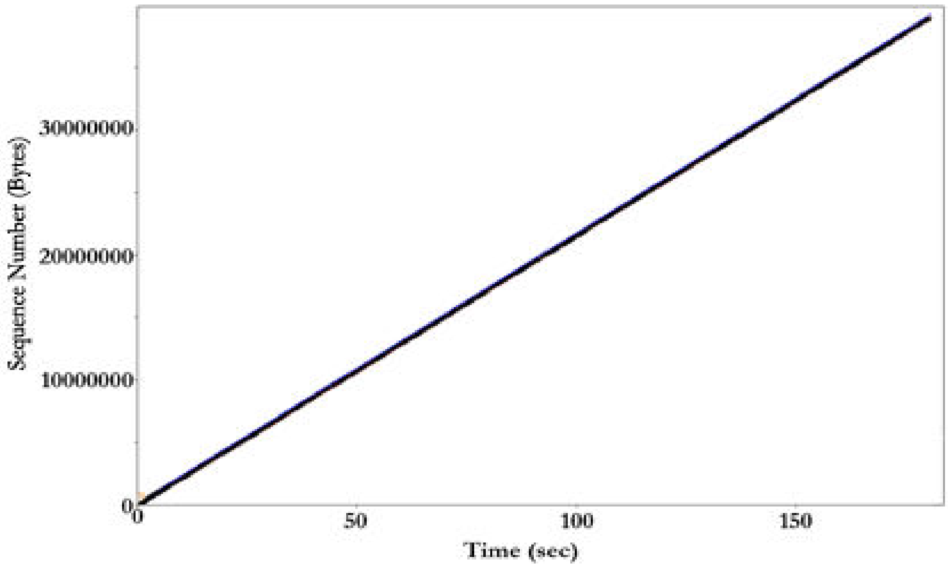

Fig A. 68 Sequence number (overall TCP connection) for 2Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

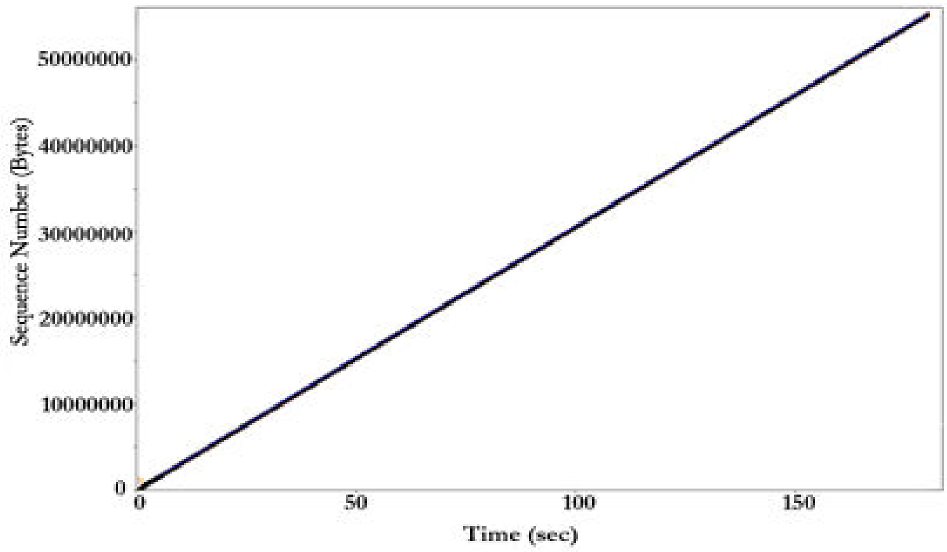

Fig A. 69 Sequence number (overall TCP connection) for 3Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

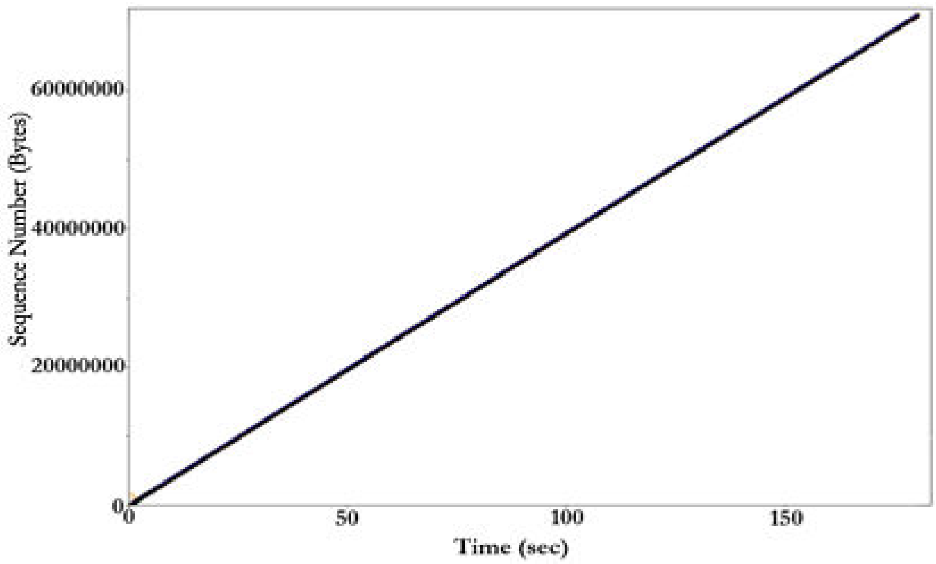

Fig A. 70 Sequence number (overall TCP connection) for 4Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

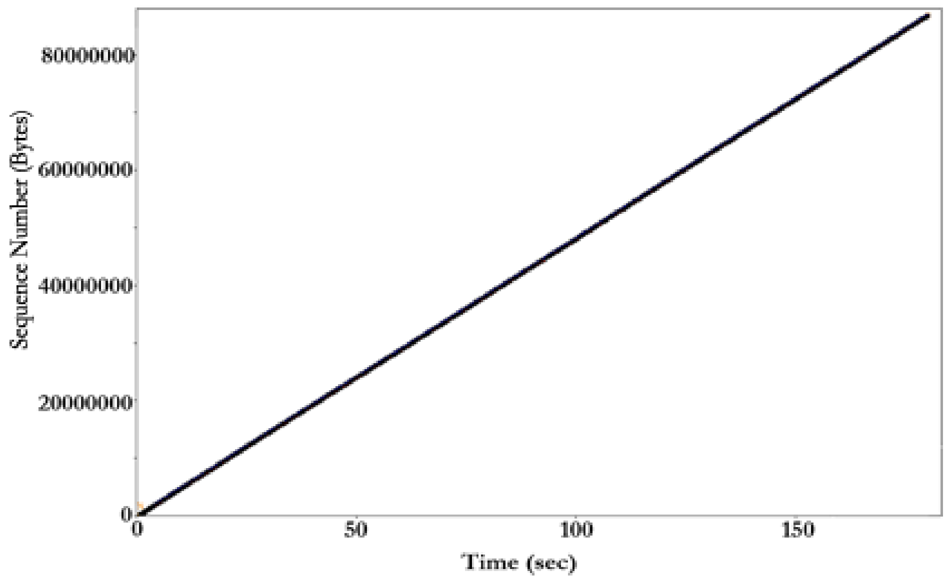

Fig A. 71 Sequence number (overall TCP connection) for 5Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

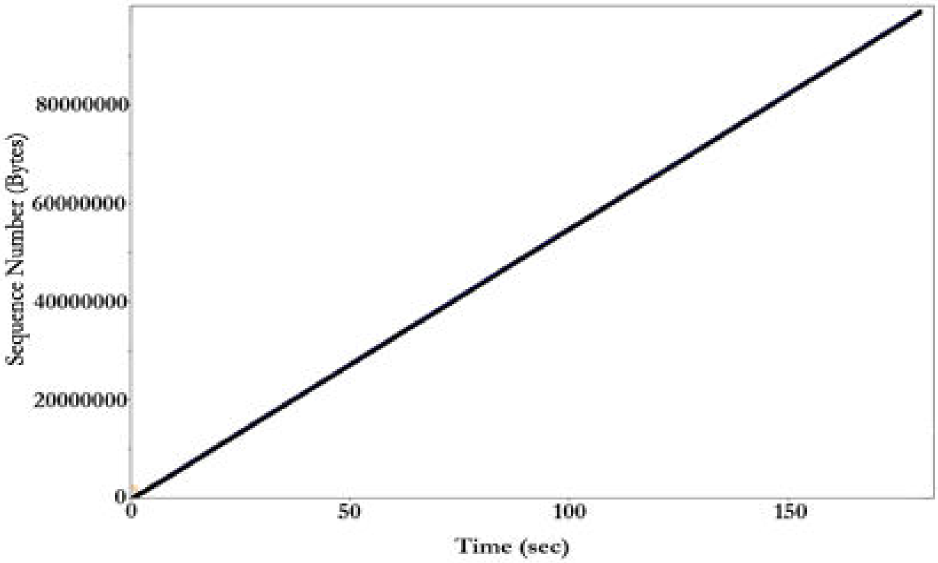

Fig A. 72 Sequence number (overall TCP connection) for 6Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

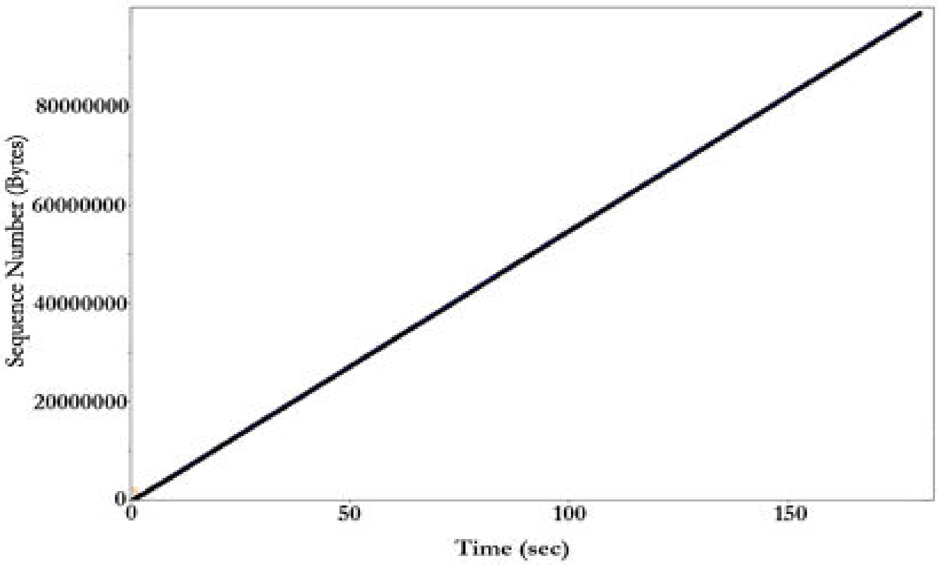

Fig A. 73 Sequence number (overall TCP connection) for 7Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

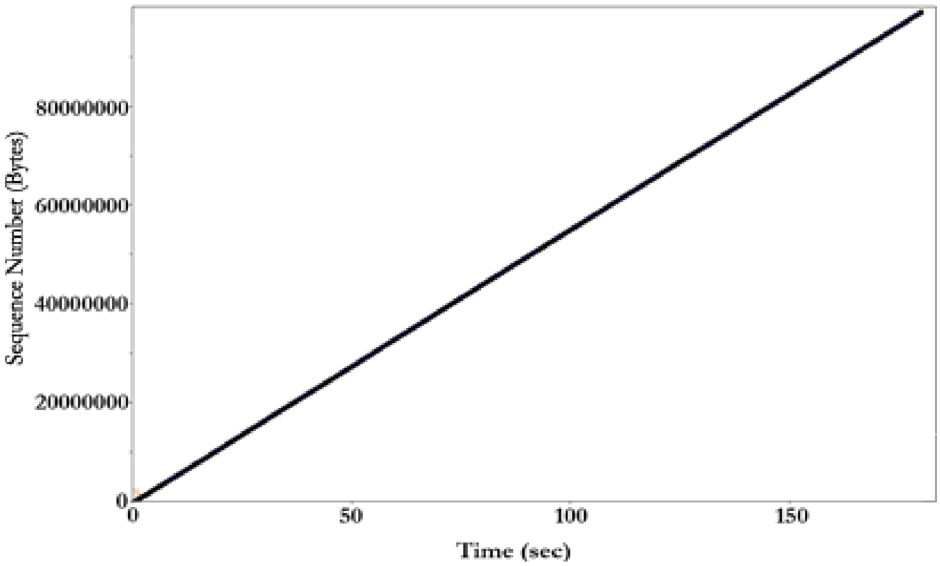

Fig A. 74 Sequence number (overall TCP connection) for 8Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

A.II.d Sequence Number (Start of TCP Connection)

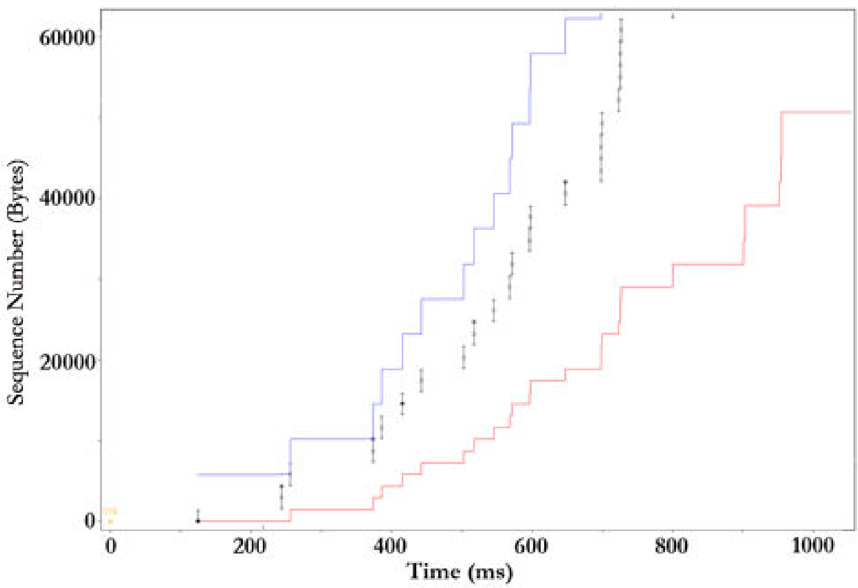

Fig A. 75 Sequence number (start of TCP connection) for 1Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

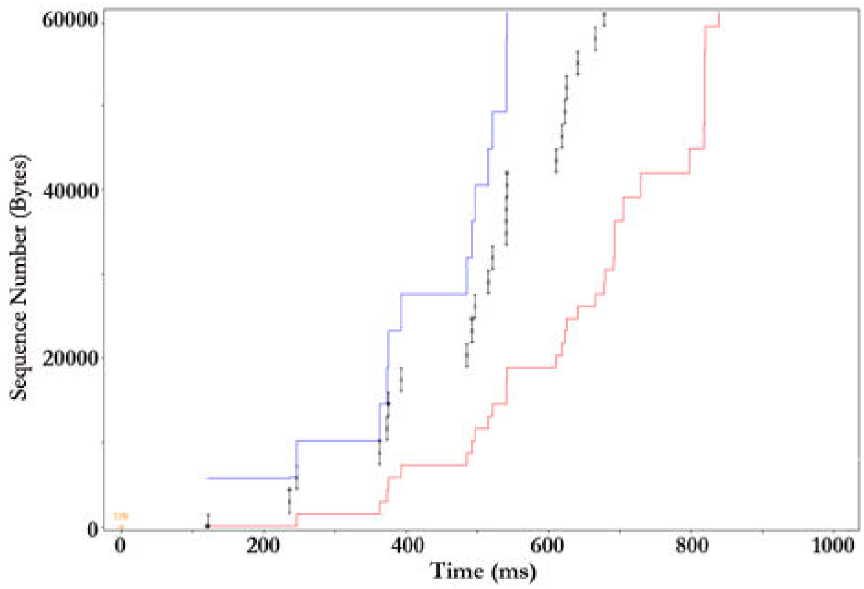

Fig A. 76 Sequence number (start of TCP connection) for 2Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

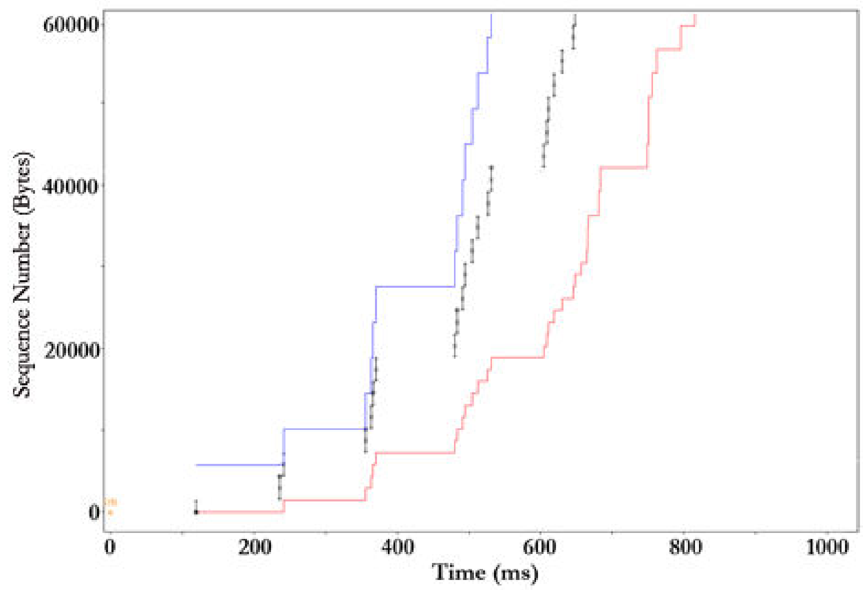

Fig A. 77 Sequence number (start of TCP connection) for 3Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

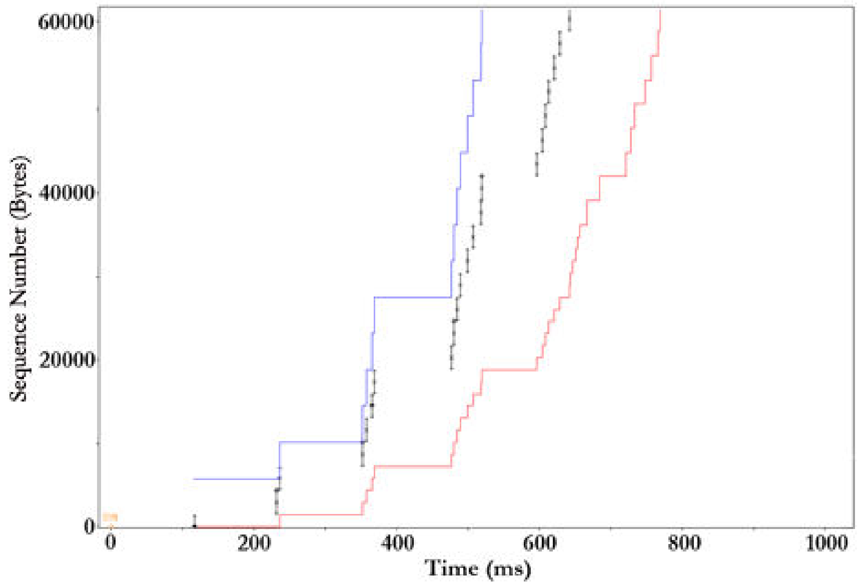

Fig A. 78 Sequence number (start of TCP connection) for 4Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

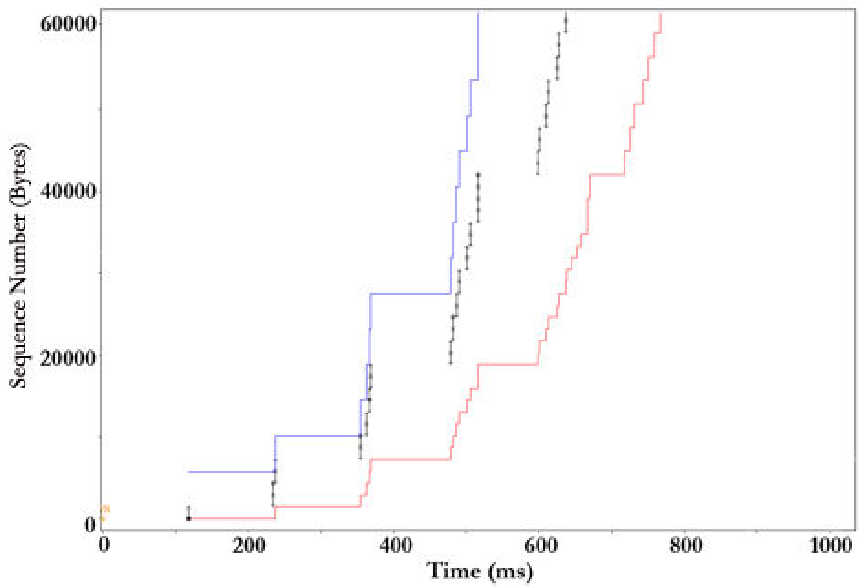

Fig A. 79 Sequence number (start of TCP connection) for 5Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

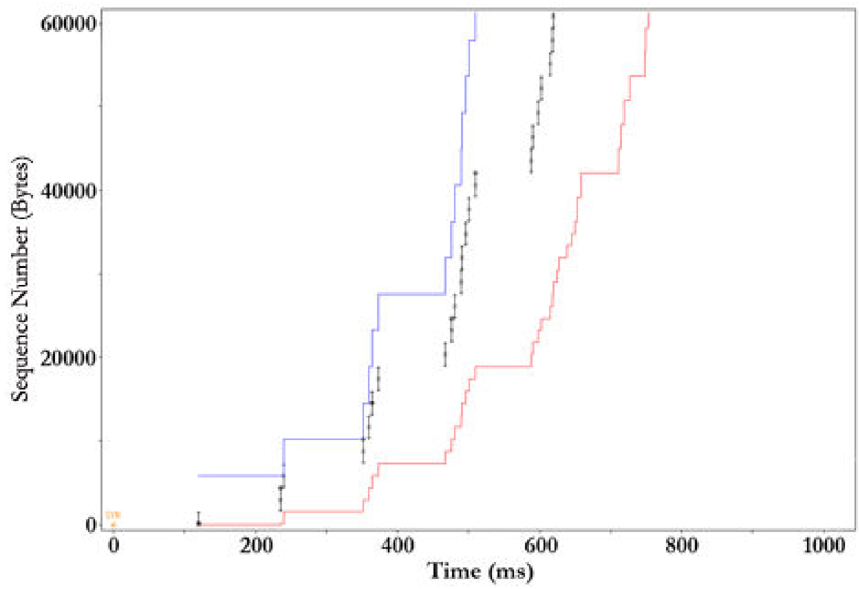

Fig A. 80 Sequence number (start of TCP connection) for 6Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

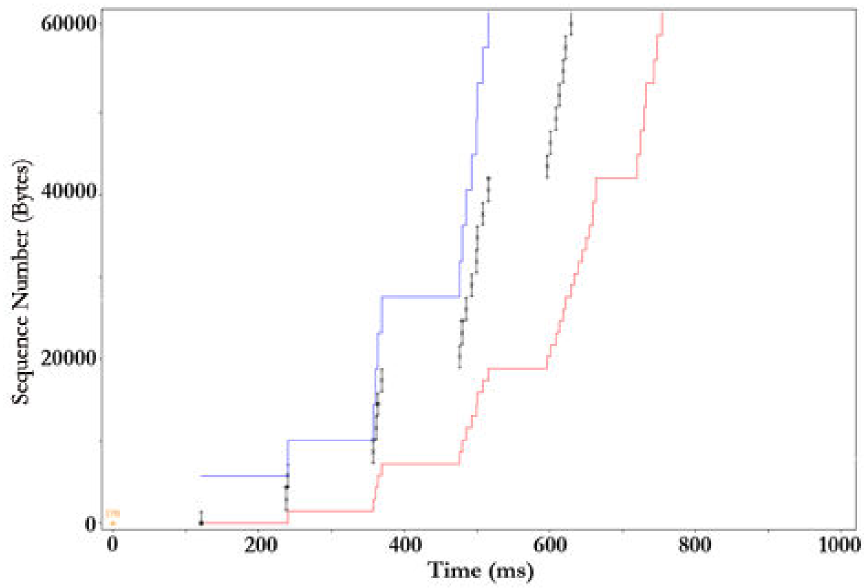

Fig A. 81 Sequence number (start of TCP connection) for 7Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

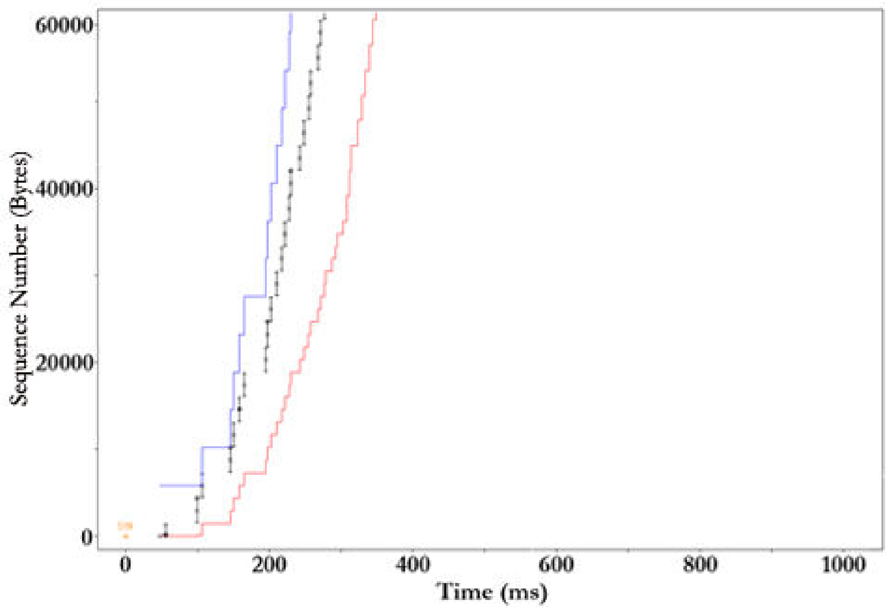

Fig A. 82 Sequence number (start of TCP connection) for 8Mb/s IP/DVB gateway bandwidth allocation for TCP Traffic

A.III Experimental Results for QoS Provisioning

A.III.a DiffServ Mechanisms Disabled

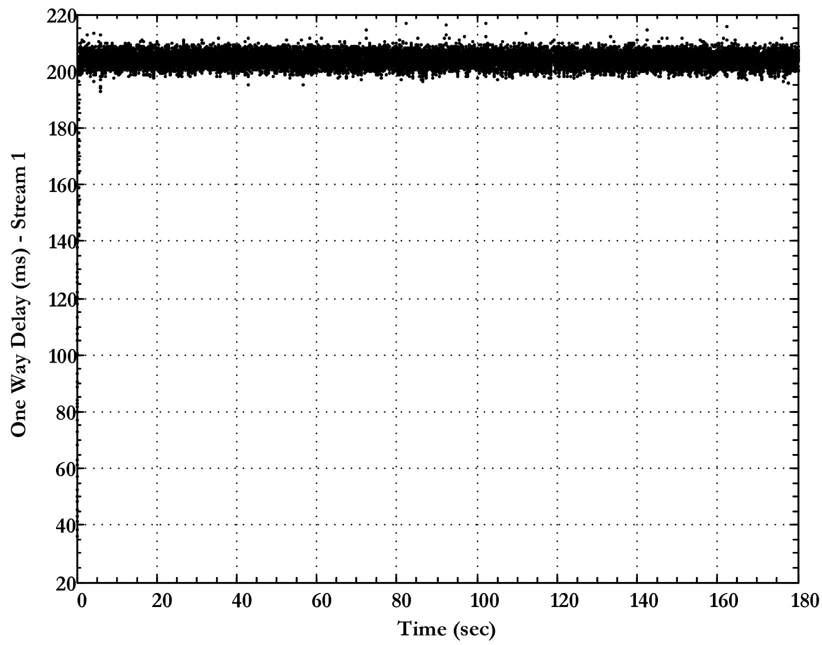

Fig A. 83 One way delay for 1st UDP stream

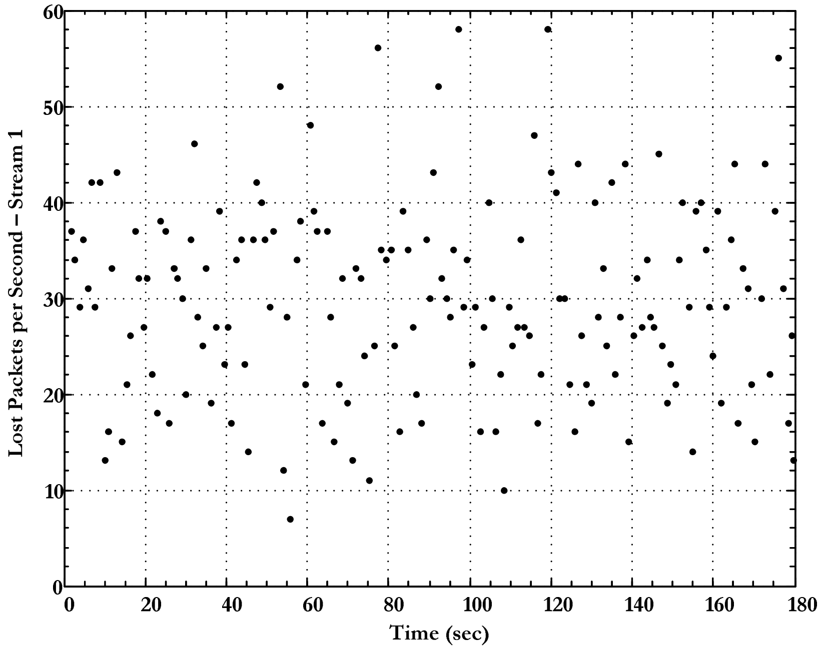

Fig A. 84 Lost packets per second for 1st UDP stream

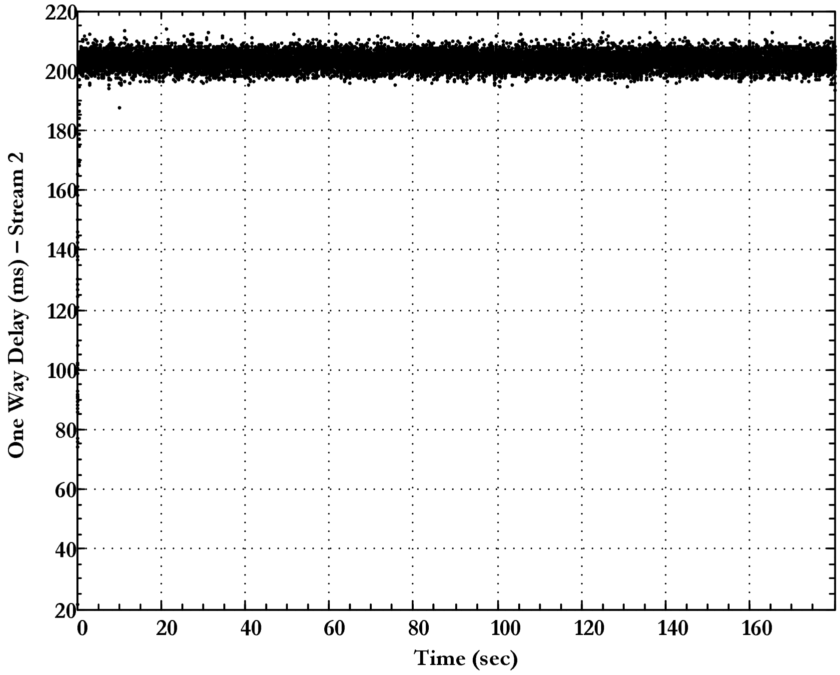

Fig A. 85 One way delay for 2nd UDP stream

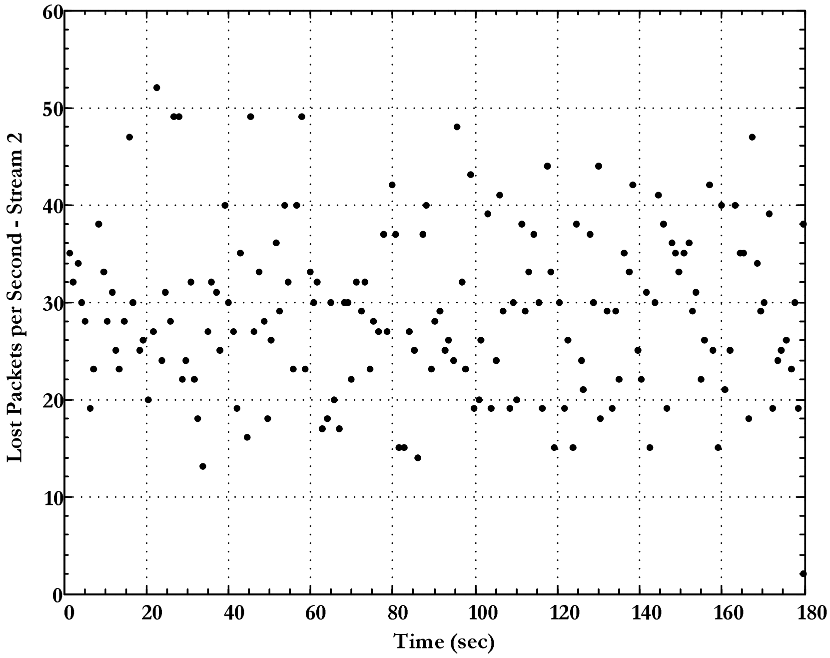

Fig A. 86 Lost packets per second for 2nd UDP stream

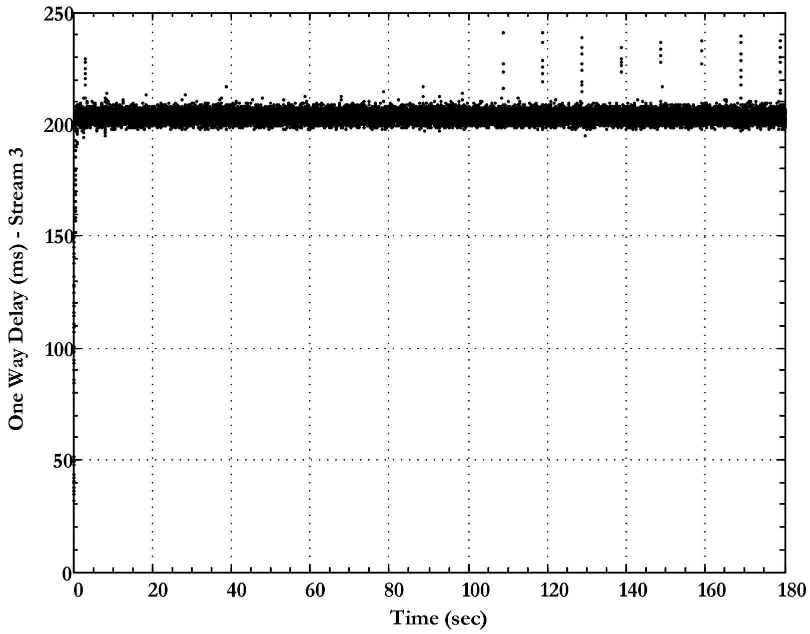

Fig A. 87 One way delay for 3rd UDP stream

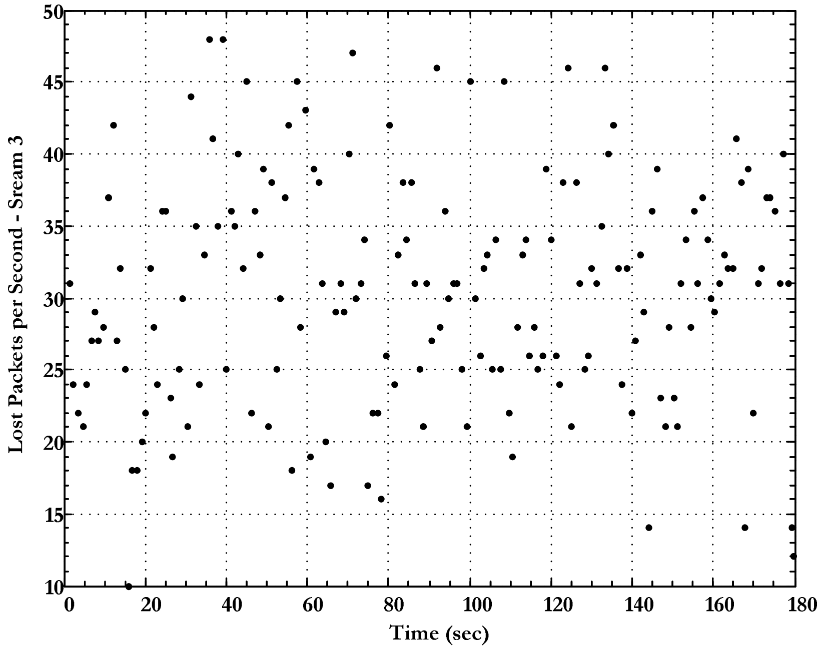

Fig A. 88 Lost packets per second for 3rd UDP stream

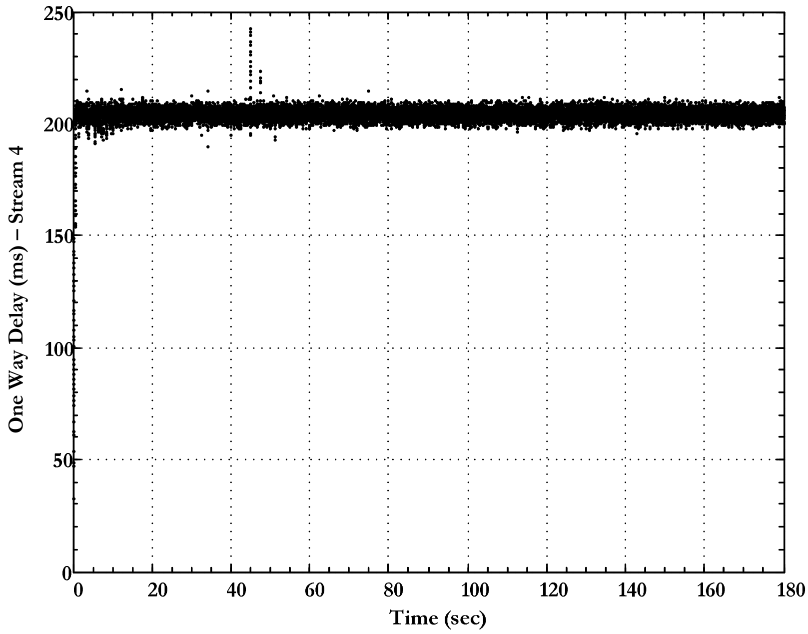

Fig A. 89 One way delay for 4th UDP stream

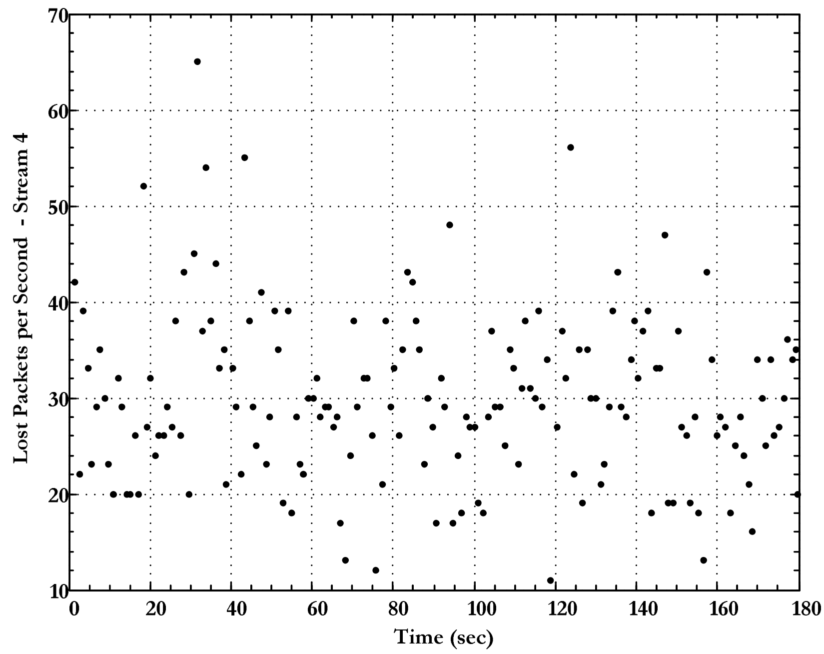

Fig A. 90 Lost packets per second for 4th UDP stream

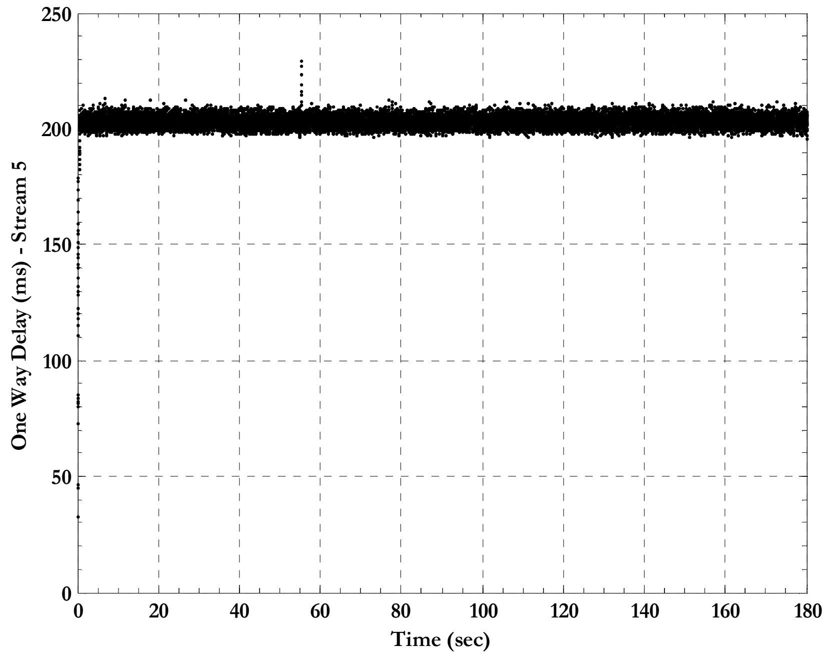

Fig A. 91 One way delay for 5th UDP stream

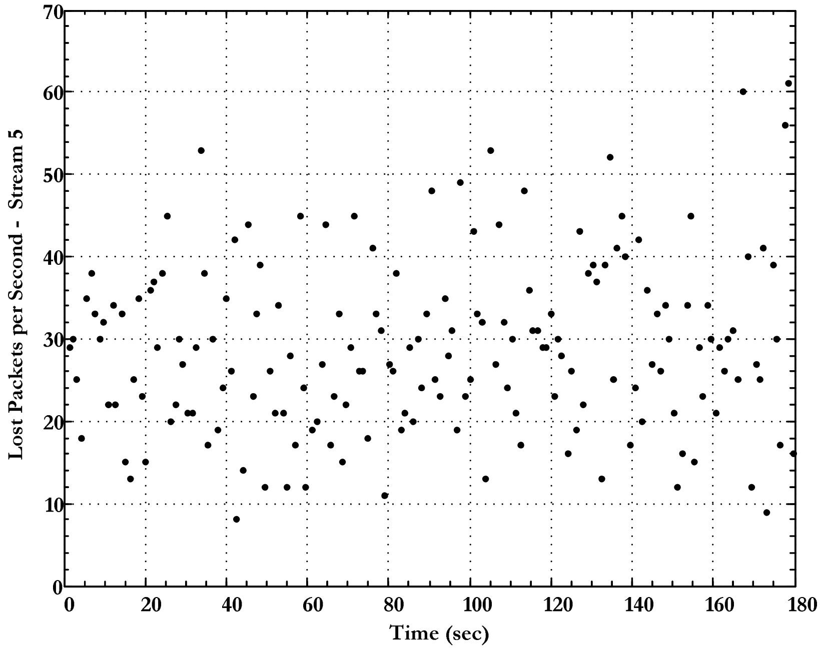

Fig A. 92 Lost packets per second for 5th UDP stream

A.III.b DiffServ Mechanisms Enabled

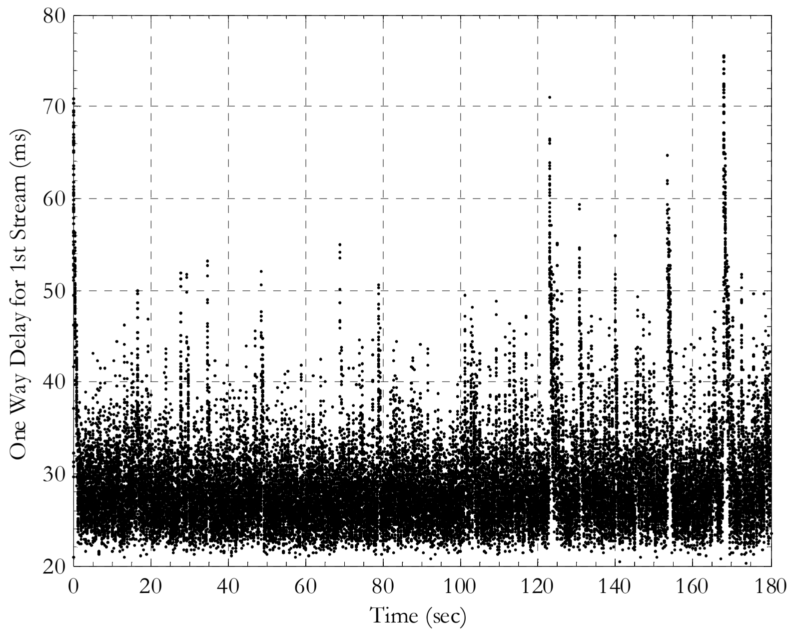

Fig A. 93 One way delay for EF UDP stream

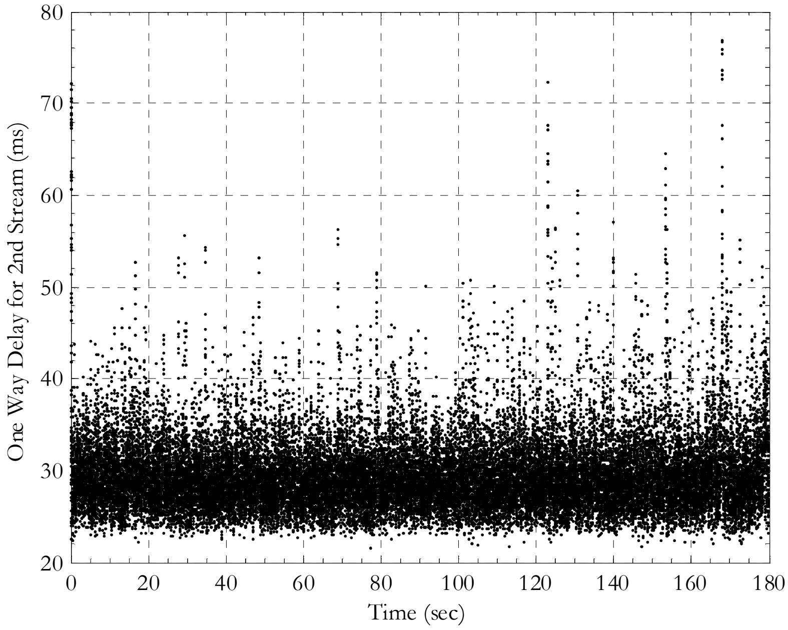

Fig A. 94 One way delay for AF1 UDP stream

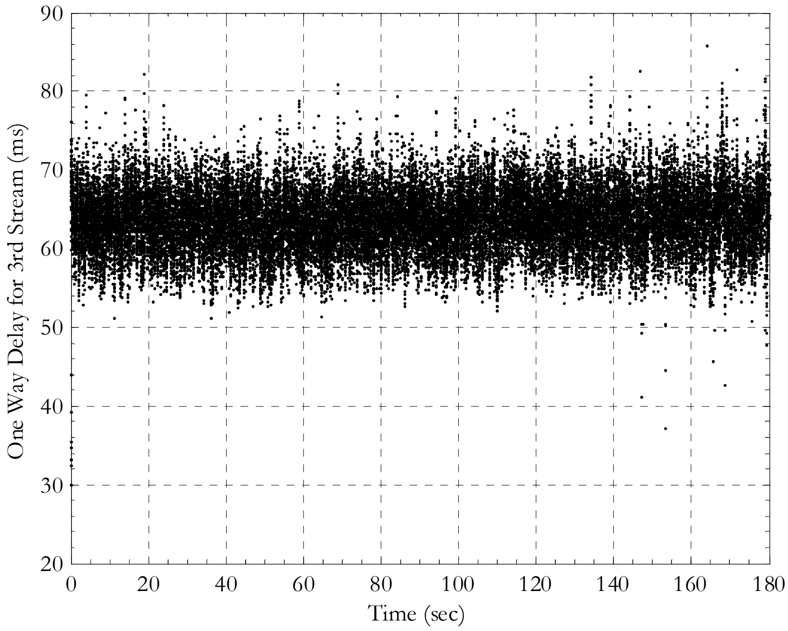

Fig A. 95 One way delay for AF2 UDP stream

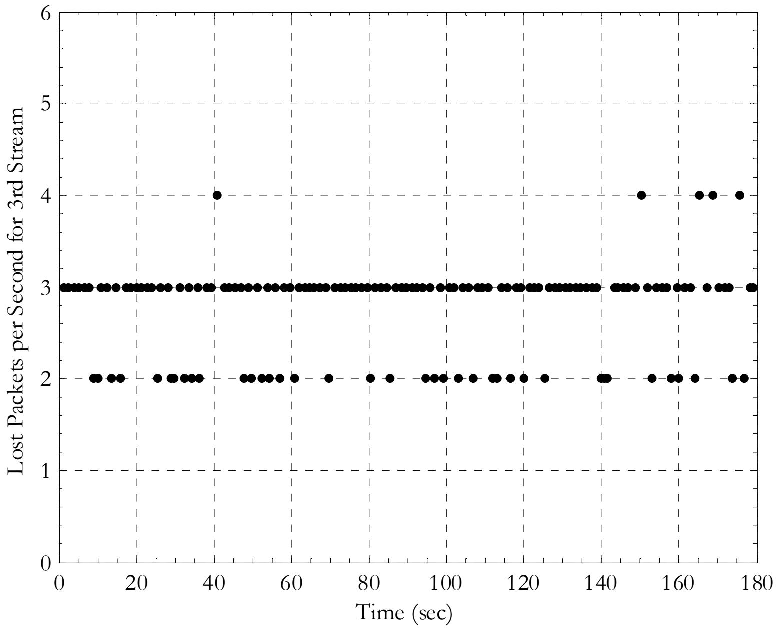

Fig A. 96 Lost packets per second for AF2 UDP stream

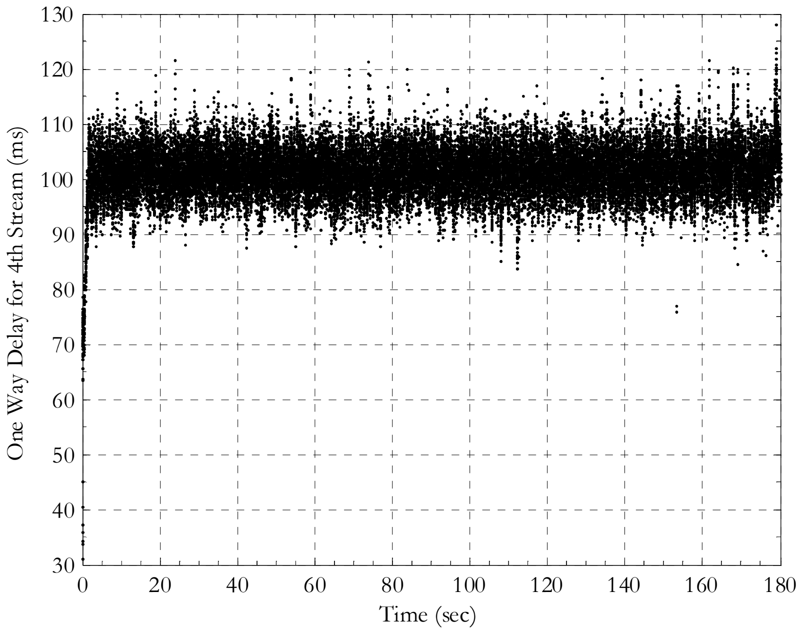

Fig A. 97 One way delay for AF3 UDP stream

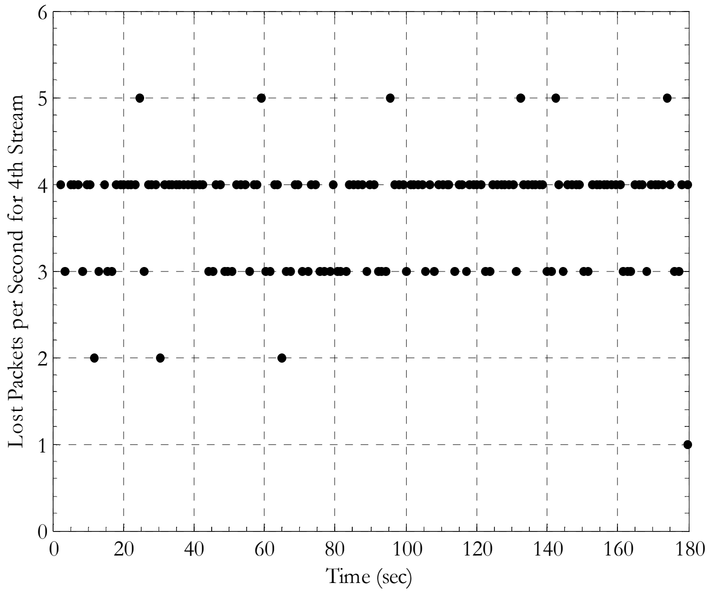

Fig A. 98 Lost packets per second for AF3 UDP stream

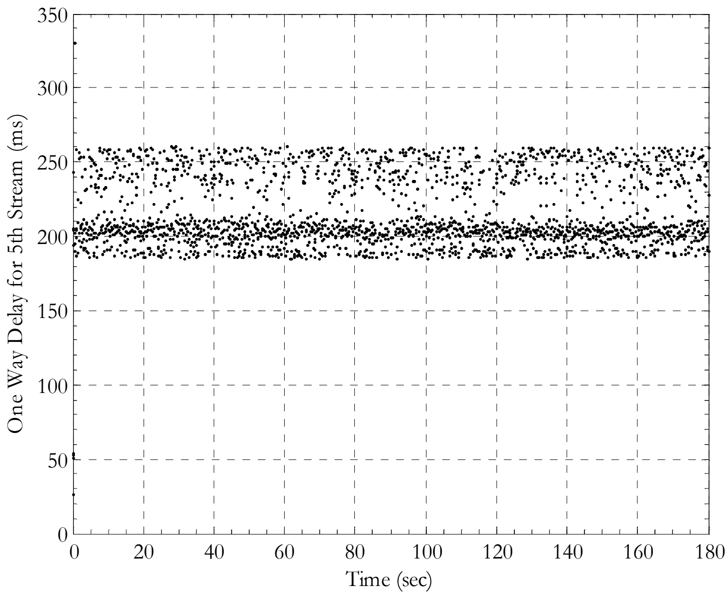

Fig A. 99 One way delay for BE UDP stream

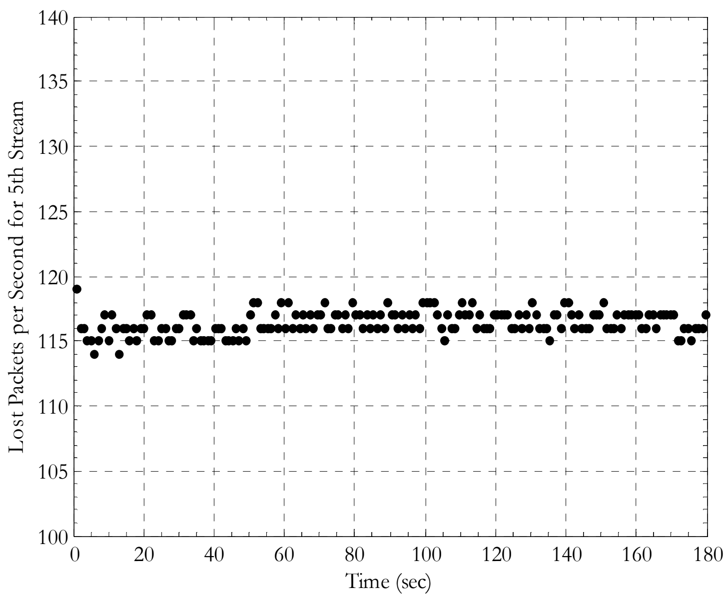

Fig A. 100 Lost packets per second for BE UDP stream

A.IV Perceived Quality of Service

A.IV.a DiffServ Mechanisms Disabled

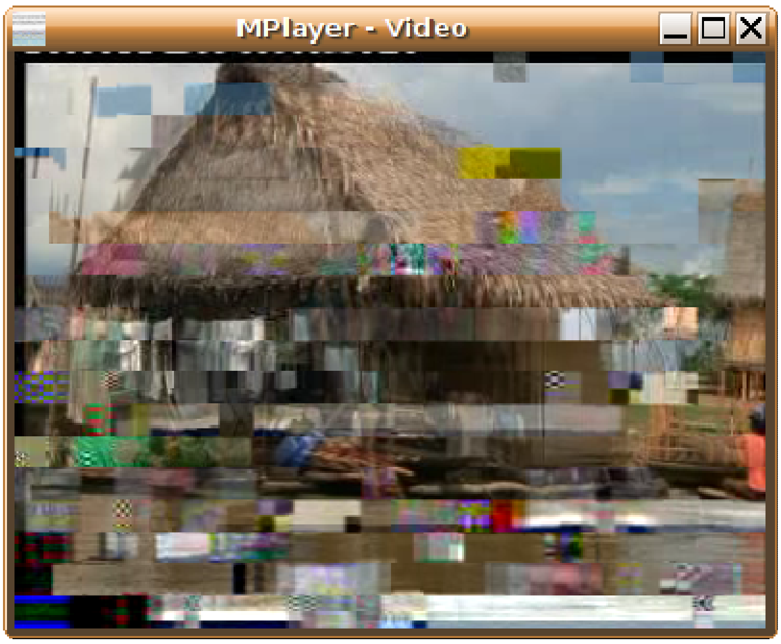

Fig A. 101 IPTV1 multicast service

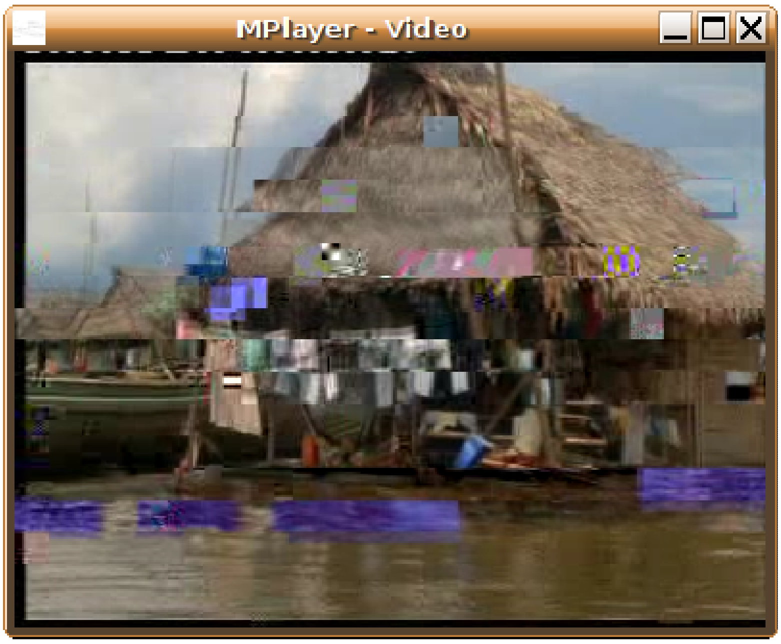

Fig A. 102 IPTV2 multicast service

Fig A. 103 IPTV3 multicast service

Fig A. 104 IPTV4 multicast service

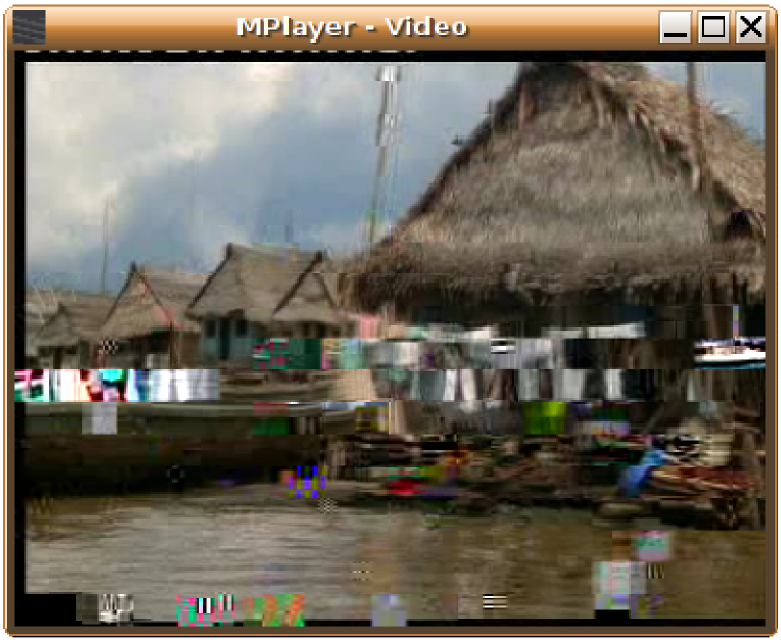

Fig A. 105 IPTV5 multicast service

A.IV.b DiffServ Mechanisms Enabled

Fig A. 106 EF IPTV1 multicast service

Fig A. 107 AF1 IPTV2 multicast service

Fig A. 108 AF2 IPTV3 multicast service

Fig A. 109 AF3 IPTV4 multicast service

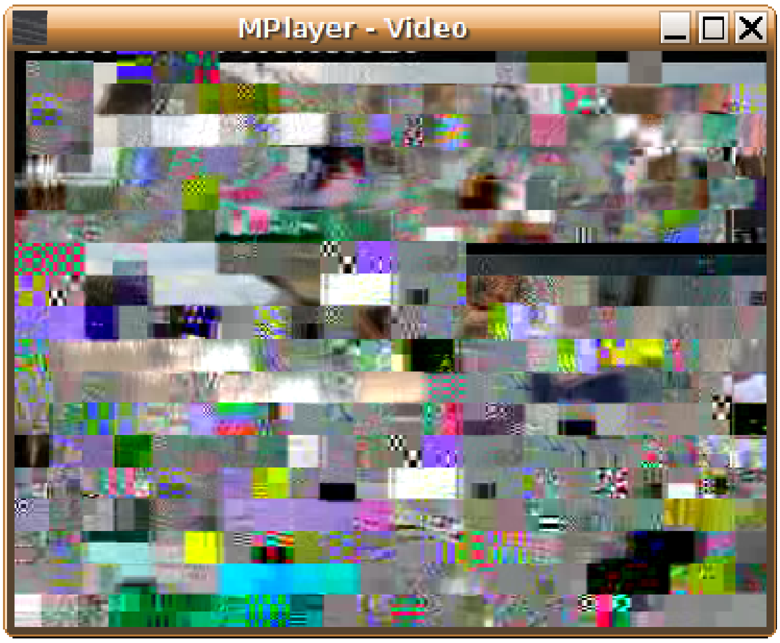

Fig A. 110 BE IPTV5 multicast service