

Enabling Broadband as Commodity within Access Networks: A QoS Recipe

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Abstract. This paper describes the QoS features that will transform the access networks landscape in order to bring “Broadband” as a commodity while setting up the pillars of the “Future Media Internet”. Quality of Experience is obviously key for emerging and future services. Broadcasting services will first need to equal the QoE of their counterparts in the Open-air market (for IP-TV examples would be artifact-free, no picture freezing, fast zapping times) and offer new features often using interactivity (Time-shifted TV, access to more content, 3DTV with feeling of presence). The huge variety of communications alternatives will lead to different requirements per customer, whose needs will also be dependent on parameters like where the connection is made, the time of the day/day of the week/period of the year or even his/her mood. Today’s networks, designed for providing just Broadband connectivity, will not be enough to satisfy customer’s needs and will necessarily support the introduction of new and innovative services. The Networks of the future should learn from the way the users are communicating, what services they are using, where, when, and how, and adapt accordingly.

Keywords: Broadband, QoS, QoE, access networks, resource managers, SLA.

1 Introduction

Quality of Service (QoS) is a key feature in multi-service multi-provider access networks to provide appropriate quality of user experience for the services offered. Approaches that rely on an end-to-end QoS signaling per individual flow (such as IntServ - Integrated Services) have not been deployed widely because of their complexity. Priority based scheduling of traffic per hop or node (e.g. DiffServ - Differentiated Services) is less complex but works best in an

over-dimensioned core network. MUSE [1] therefore elaborated a resource management architecture aided by a simple policy control framework to result in a pragmatic and scalable QoS architecture. [3],[4],[5],[7].

2 The Challenge of Introducing QoS in IP Networks

The current best-effort approach used in IP networks implies not differentiating traffic irrespective of what kind of application has generated it or where the source or the destination points are located. In the best-effort approach, the normal practice is to provide more capacity, when possible, so that there is a reasonable quality of experience using the network. However, this practice leads to an increasing loss of control of both the nature of traffic the network is carrying and the relative importance that it has for the users. QoS should be introduced in access networks, so that requirements of traffic of different nature can be fulfilled by the same network without wasting resources. QoS in IP networks is a problem which can, in principle, be solved in a variety of ways. However there is still no consensus on the best way to tackle the QoS problem. Some solutions (e.g. IPoATM - IP e ATM, IntServ, MPLS TE, ...) are rather complex to be cost-effectively deployed in current access and aggregation networks, as they require radical and non-gradual changes both in network equipment and their control and management whereas others (e.g. DiffServ) may seem too simple or insecure to be useful. Part of the problem seems to be in trying to replicate the traditional carrier class quality. Another difficulty is the multi-service nature of IP networks, hence imposing a broad set of different requirements on the network. However, the main reason for not deploying complex QoS oriented architectures in actual networks is the inherent difficulty for the network operator to appropriately configure and manage them, especially when addressing the residential mass market. That is, one of the main reasons for lack of success of previous proposals is that they did not have a clear business orientation.

3 Pragmatism and Simplicity as the Main Design Principles

There is a need for a pragmatic and simple way to provide services with at least some degree of QoS. Pragmatism is needed to design a solution scalable enough to address the residential market and flexible enough to address the corporate one. By focusing on the needs of triple play offerings, it is possible to design a simple and cost-effective way to provide sufficient QoS for the mass market. QoS will be a way to add value to services that are offered by the network operator and thereby differentiate them from similar services offered over the Internet. The proposed solution should have a wider applicability than just triple-play, in order to adapt to the corporate market needs, although it may not meet the requirements of any conceivable service. Furthermore, triple play services demand a high amount of bandwidth that will require big investments from the

Telecom Operators side. These investments need to be financed. One option is to get the funds from the pockets of its users, but there are signs that they are quite reluctant to pay any more money just for plain connectivity services (i.e. Internet access). However, if the operators are able to differentiate their connectivity resources, they can better sell them to connectivity customers no matter if they are service providers, packagers [8], enterprise users or residential users with specific requirements. Therefore, the key issue is to setup a solution where operators are able to segregate their networks in connectivity resources with specific QoS that can be offered on-demand to “connectivity resource” consumers. This business model can lead to different architecture scenarios where the user, the service provider, the packager (service broker) or all of them simultaneously request resources with QoS for certain applications. Being pragmatic, it is also convenient to take into account the current situation of the broadband access networks already deployed as a starting point. Today’s DSL broadband access architectures (based on TR-101 architecture) [9] consist of aggregating connectivity for mass-market Broadband Access in which all traffic goes to a single aggregation point, the Broadband Remote Access Server (BRAS). In fact TR-101 slightly extends this model, to allow for a second BRAS or Broadband Network Gateway (BNG), which is specifically included to allow video content to be sourced from a separate location. The QoS control is however centered on the BRAS. Bandwidth control is limited to the downstream direction, and is based on the concept of hierarchical scheduling.

4 End-to-End QoS Solution

Starting point for QoS implementation was the so-called preferred QoS implementation of MUSE phase 1. This solution has been described extensively in [10], so that here only the key features are summarized. Four QoS classes were defined which differentiate by their timing behavior (see figure 2). More precisely, to be compliant with MUSE/PLANETS QoS classes, a node must comply with the respective maximum jitter, packet loss and burst size values listed in Table 1. The QoS solution can be realized over existing ATM access networks, but more conveniently in course of the introduction of IP based access networks. The per-hop behavior of Table 1 can be requested in respective tenders. It is easy to realize in typical IP DSLAM, home gateway and aggregating Ethernet switches.

The novelty of this QoS classes is the specification of discrete jitter values. Other approaches such as DiffServ (RFC2474) only agreed on relative per-hop

Table 1. QoS Class per-hop Behavior

QoS Class	Max. Jitter	Max. Loss rate	Max.burst size
Low latency (LL)	1ms	10^{-11}	200 byte
Real Time (RT)	30ms	10^{-11}	1500 byte
Elastic (EL)	900 ms	10^{-7} (hi part) ≤ 1 (lo part)	9000 byte
Best Effort (BE)	-	≤ 1 (lo part)	-

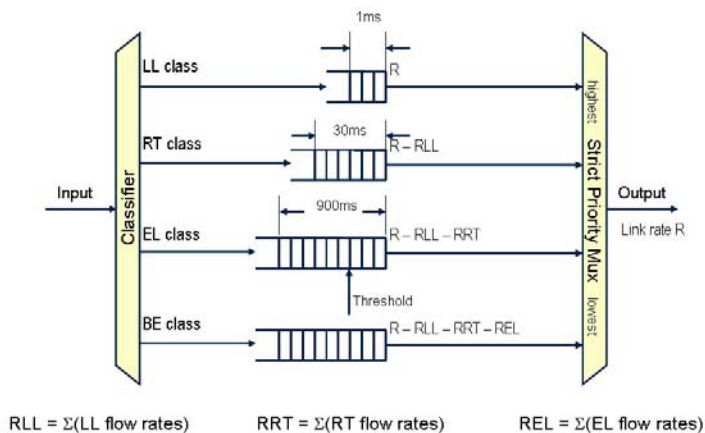


Fig. 1. Scheduler

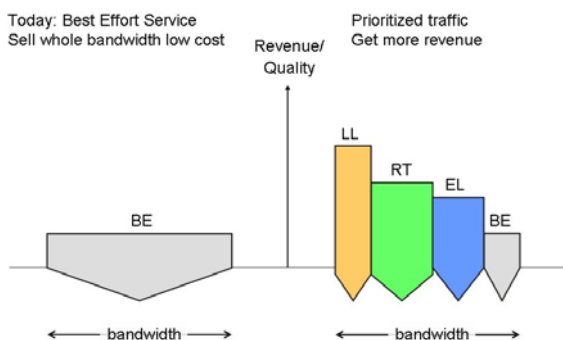


Fig. 2. Operator benefit/Customer perception

behavior. From the discrete values, end-to-end delay values can be calculated. For this purpose a maximum of 10 nodes has been assumed. As an example, a subscriber with 2Mb/s upstream and 10Mb/s downstream has a maximum delay of about 100ms over 1000km distance RT service, while LL service over 100km has a mean delay of 3ms. These figures show that RT class is well suited for telephony, while the LL class is reserved for very demanding applications such as distributed computing or client-server applications. Another benefit of the discrete jitter values is the possibility to calculate queue sizes for the scheduler shown in Figure 1. In this figure RLL, RRT and REL denote the sum of reserved rates for the respective QoS class. Thanks to the per-flow reservation these rates are known and the minimum size needed to support the jitter values can be calculated. The PLANETS web site [2] lists the queue sizes depending on the load. For example an output port serving a link with 90 % load needs 23.6kB for the LL queue, 177kB for the RT queue and 677kB for the EL queue.

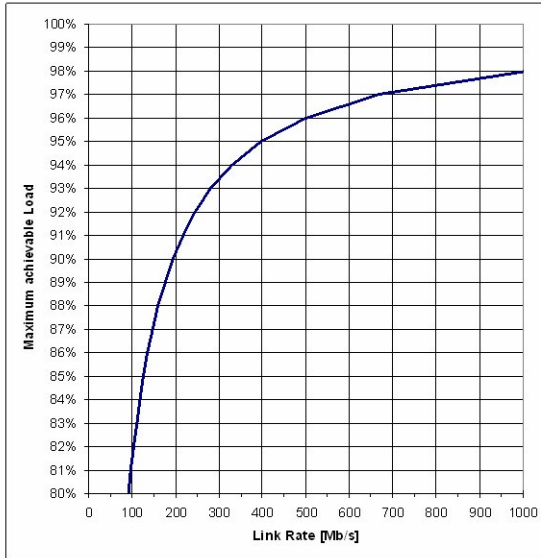


Fig. 3. Maximum Bandwidth Reservation

Queue length calculation assumed statistical arrival of maximum size packets using M/D/1 model [12]. In this model queue sizes go to infinity when load approaches 100 %, independent on the rate. The maximum allowed jitter puts an upper bound to the size of the queues, so that the maximum achievable load is limited. This load limit applies to reserved rate only, as it must be guaranteed at any time. Figure 3 gives the maximum rate which can be reserved on a link. The used rate is always up to 100 % due to the low priority part of the EL class and BE traffic which can go beyond the guaranteed rate. In Figure 3 it can be seen that below 100Mb/s link usage rapidly drops to unacceptably low values. For these link rates the number of independent packet sources is too low to allow a statistical treatment. Hence a deterministic approach is followed with worst case assumption that all sources send packets simultaneously. This limits the number of sources, but allows bandwidth reservation up to 100 %. The deterministic approach is used for example for DSL lines. The previous statistical treatment would remain appropriate for Gbps links higher in the Network architecture.

5 Realising QoS in the Access Network

With the introduction of a control plane, today's basic Internet access will be enhanced by on-demand broadband connections with selectable QoS. The new network element to support selectable QoS is the Multi-service Edge router (Border Gateway). It acts like a large firewall. Connection setup is equivalent to the opening of a pinhole. In particular a pinhole is defined by the subscriber's (residential

gateway) IP address and a destination UDP port. The Multi-service Edge router is connected to a Multimedia Overlay Network which supports QoS. The Multi-service Edge router receives signaling from the Signaling Proxy, but not the Access Node. Terminating signaling on the Access Node is deprecated by network operators, as it leads to higher costs for the respective SW packages, their maintenance and a lot of control traffic. On the other hand, how to reserve bandwidth in the access network without signaling? Over-provisioning can not be the solution, as the 1st Mile is the most significant bottleneck. The solution chosen has been described by Toelle et al. [12]. It uses permanently configured Service Connections (SC) as transport pipes for application flows between Residential Gateway and Multi-service Edge router or Edge Node. A SC is characterized by QoS class and bandwidth. The bandwidth is reserved over the whole Aggregation Network. Examples for application flows are the voice packet stream of an IP phone or the data packet flows generated by a browser running on a PC. Each subscriber has at least one BE class SC to the Edge Node. It provides the basic Internet access as it is offered today. By definition a BE SC uses the unused bandwidth by the rest of the classes, not requiring bandwidth reservation. Any number of application flows can share a BE SC. In addition the subscriber can have up to three SC to the Multi-service Edge router, one for each class. In the example of Figure 4 two SC with RT and EL class are configured between a Residential Gateway and a Multi-service Edge router. Obviously the guaranteed rates of the flows must fit into the SC bandwidth. In addition, the maximum number of flows sharing a SC is limited by the maximum allowed jitter for the respective QoS class according to the deterministic approach mentioned above. Bandwidth and jitter limit are independent. The smaller of both determines the possible number of flows. The acceptance procedure can be done by the Multi-service Edge router if a subscriber is connected to one Multi-service Edge router only. In case of more than one Multi-service Edge router only the subscriber (i.e. the Residential Gateway) knows about all flows. This option is for further study. The Edge Node is not relevant as it receives BE flows only. Finally, note that the commercial exploitation of SC could be via additional flat rates.

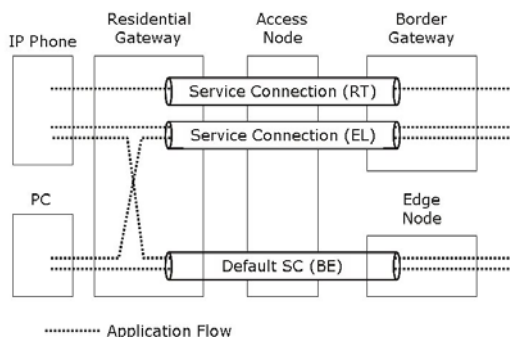


Fig. 4. QoS pipes

6 Realising QoS in the First Mile

In order to efficiently manage the available resources and fully utilize the DSL link capacity, a Copper Plant Manager (CPM) is needed [13]. The CPM tool processes key standardized DSL performance parameters such as Attainable Rate, Actual Rate, Actual Delay, Errored Seconds, and Unavailable Seconds. These parameters are not only monitored, but also statistically processed to support key performance indicators to be interpreted in an appropriate manner to inform the system about actual and/or average behavior of the lines, trends in performance parameters and live changes due to unexpected situations which could lead to service degradation. For example, Attainable Rate when compared with Actual Rate shows the headroom for rate increase in case of more demanding services while Actual Delay, Errored Seconds and Unavailable Seconds determine to a great extent the Quality-of-Service and the Quality-of-Experience (perceived quality by the end user of a given service) that can be expected in a given DSL link. All these parameters are taken into account by the CPM tool. In this way, the Network Management System (NMS) can use this information when determining the policies to enforce for a given service in order to fulfill the Service Level Specifications (SLS) or Service Level Objectives (SLO) defining the required QoS parameters on system level (node, link level).

The SLS is part of the service level agreement (SLA) that represents a formal negotiated agreement between the end-user and service/content providers on a common understanding of the service including service quality, priorities, responsibilities, and guarantee. The SLS, as technical interpretation of the SLA, specifies the end-to-end requirements on the network (nodes, links) to transport the service with the negotiated QoS. The most relevant SLS parameters for typical triple-play services via DSL are listed in Table 2. The resource manager within the NMS plays a central role in mapping the SLS into policies that break

Table 2. Most relevant SLS parameters for Triple Play services QoS and performance monitoring

Service	Parameter	Description
Best-Effort Internet	Peak Info. Rate (PIR)	No guaranteed rate, the rate is restricted with PIR
	Peak Burst Size (PBS)	Maximum number bits that can be transported at PIR
Guaranteed Data Serv.	Committed Info. Rate (CIR)	Guaranteed mean rate, mostly in combination with PIR/PBS maximum restrictions
	Committed Burst Size (CBS)	Maximum number of bits that can be transported guaranteed at CIR
Voice Serv.	Maximum Delay (MD)	Maximum delay caused by processing delay, shaping, queuing
	Delay Variation (DV), jitter	Packet delay variation, filling level in queues
Video Serv.	Packet drop rate (PDR)	Rate of lost packets on the link

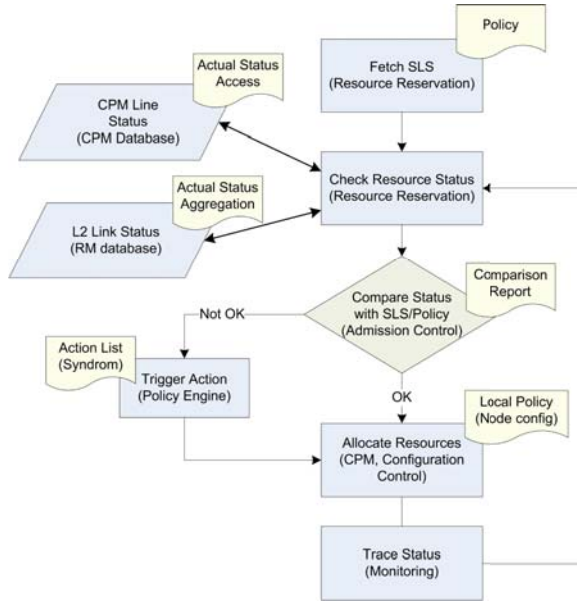


Fig. 5. Simplified CPM - NMS interaction workflow

down to local policies on the network nodes in different parts of the end-to-end service delivery workflow, as described in Figure 5.

7 Validation

The concept has been successfully verified in a lab trial at T-Systems in Berlin by means of a demonstrator (see Figure 6). This demonstrator implements the entire Multi-service Edge router data plane functionality and part of the control plane using prototypes based on Network Processor subsystems. At call setup the subsystem automatically initiates ARP requests at both aggregation network side and multimedia overlay network side, supporting up to 512 phone calls. For QoS measurements Triple Play scenarios have been set up with video streaming, phone calls and packet generators for background traffic. Zero packet loss has been proven for guaranteed traffic, even in the presence of abundant best effort traffic, as expected.

The concepts explained in previous sections as well as the Service Connections (SC) shown in Figure 4 were validated. This PlaNets demonstrator covered the Home Network and Access Aggregation Network including the First Mile. The First Mile was based on VDSL2 links over telco copper cable. The demonstrator supports IPv4 as well as IPv6. Four Home Networks have been realised in Berlin with typical terminal equipment (e.g. TVs, PCs, videophones). Two IPv6 Home Networks had access to the 'Public Network' via Home Gateway provided by partner Stollmann. In addition two IPv4 Home Networks were connected to the VDSL2 modems via Ethernet CPE switch. The prototype Access

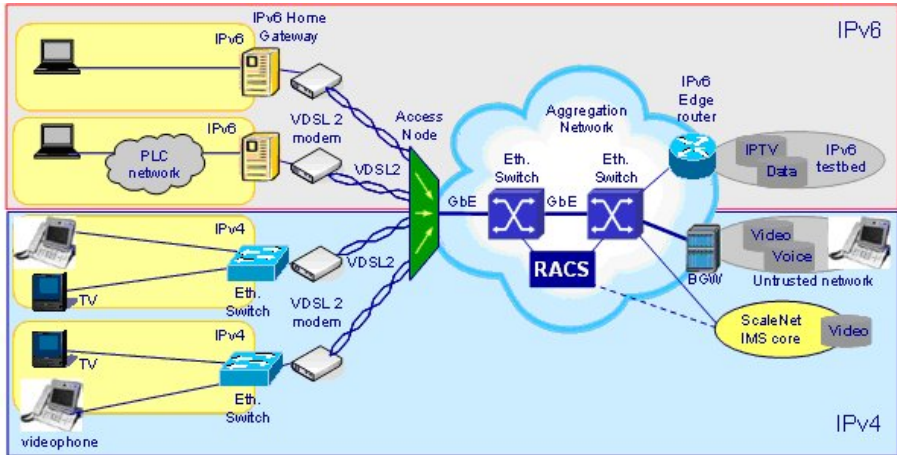


Fig. 6. Demonstrator Network

Node from Robotiker provided 4 VDSL2 lines and two electrical Gigabit Ethernet uplinks. It supported IP forwarding as defined in MUSE [1] and Layer-2 switching as well. The Access Node is connected to the Linux PC based Access Aggregation Network that consists of two aggregation layers. Several Edge Nodes are connected to the Aggregation Network. An IPv6 Edge Router offers access to IPv6 testbed providing IPv6 based Internet Access and video streaming. A cooperation with the German national project ScaleNet (Fixed, Mobile and Wireless IP-optimised convergent Next Generation Access Networks) provides an IMS overlay that enables an IMS controlled RACS implementation. The Border Gateway from Alcatel-Lucent provides conversational services with SIP controlled mechanisms and a trusted boundary to the packet network with NA(P)T traversal support for improved security.

Network Quality testing

Traditional performance tests focus on a per-port basis, addressing the evaluation of the performance of each port such as the maximum throughput, the packet loss and the average latency of the network device. To evaluate the QoS functionality some additional aspects must be considered. Flow-based measuring: Usually different applications are multiplexed onto a single device port, combining different QoS traffic classes. In order to evaluate the QoS feasibility of a network the tests had to address individual traffic flows. A flow-based measuring is the foundation of the QoS testing. The measuring equipment had to be able to provide flow-based measuring procedures.

QoS can be characterized by a set of network performance metrics including:

- End-to-End Throughput
- End-to-End Delay
- Frame Loss rate
- Delay variation or jitter

- Max. burst size
- Packet sequence the ability of the network to deliver packets in the proper sequence
- Service availability the ability to gain access to the network
- Connection availability the ability of the network to complete the required connections (i.e., TCP) with the requested performance characteristics

Perceived QoS testing

The perceived QoS is the quality of a service or application as experienced by an end-user. Recently it has become fashionable to use the term Quality of Experience (QoE) to denote perceived QoS.

Network Quality Measurement

Objective: Measuring the QoS handling of the network for IP flows with different priority in downstream direction.

Test set-up (see Figure 7)



Fig. 7. Test setup for measuring downstream network quality

Procedure:

- Generate two IP flows with different priority (Real time and Best Effort)
 - Set-up a Layer-3 (IP) end-to-end Real time QoS class stream (downstream) using a Traffic Generator
 - Set-up an additional Best Effort traffic stream that fills the whole capacity of the link.
- Increase the traffic load (e.g. from 10 % to 100 % of the link capacity in steps of 10 %)
- Analyze end-to-end Throughput, end-to-end delay, end-to-end Packet loss rate, end-to-end Jitter, and packet sequence of the Real time stream with and without congestion situation using a Traffic Analyzer

Figure 8 and 9 show the QoS handling of the network for two IP flows with different priorities (Real time and Best Effort). The chart on the left-hand side shows the frame loss ratio in case of disabled QoS mechanism. The chart on the right-hand side shows the behaviour with enabled QoS mechanisms in downstream direction. Both charts show a congestion situation for a traffic load of 60 Mbps (maximum VDSL throughput). But with enabled QoS only the frames of the Best Effort flow are dropped and the frames of the Real Time flow are

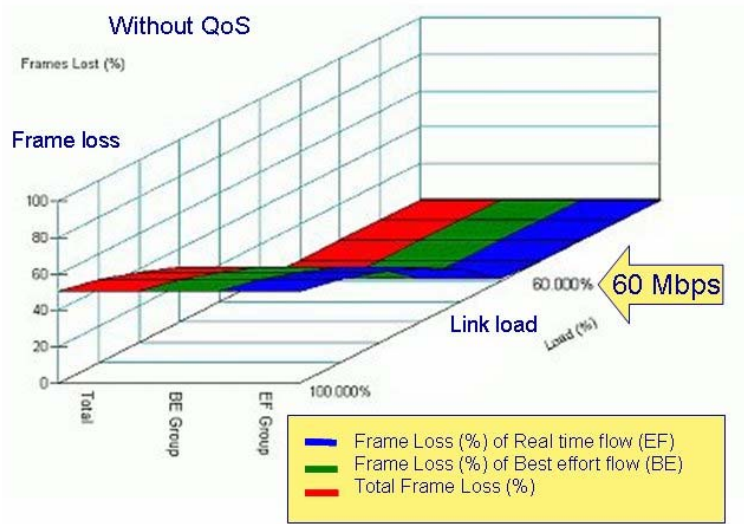


Fig. 8. Measuring the QoS handling of the network (without QoS)

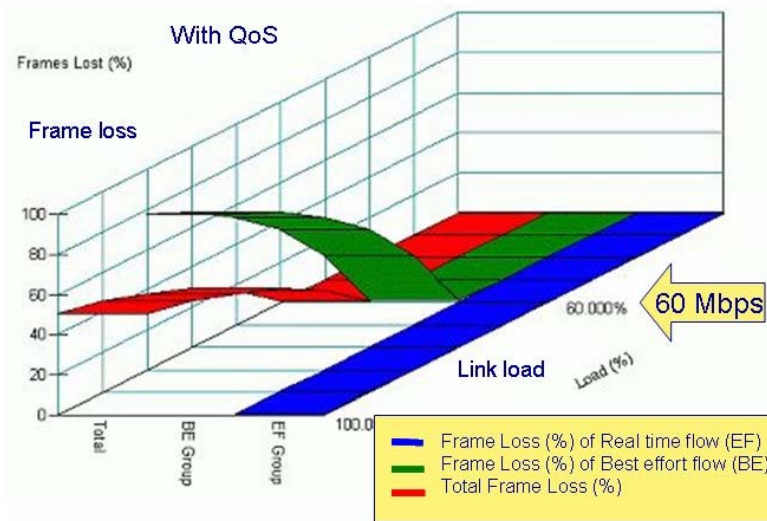


Fig. 9. Measuring the QoS handling of the network (with QoS)

forwarded with priority. Figure 10 shows the average Latency (μs) for Real Time and Best Effort flows with a frame size of 512 Bytes in case of congestion. 60 % link load is equivalent to 60 Mbps.

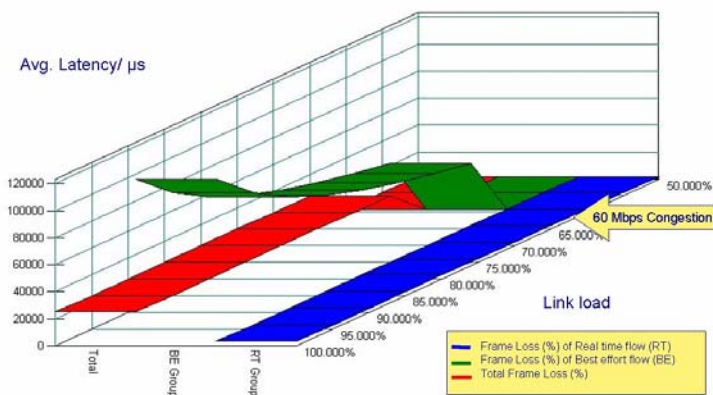


Fig. 10. Avg. Latency (μs) for Real time and Best Effort flows in case of congestion

The following table (Figure 11) shows the average Latency (ms) for Real Time and Best Effort flows with a frame size of 512 Bytes in case of congestion.

Link load in % of 100 Mbps	55 %	60 %	65 %	70 %	75 %	80 %	85 %	90 %	95 %
Best effort (ms)	1.68	1.74	48.32	53.26	58.73	65.49	74.00	85.03	100.44
Real time (ms)	1.62	1.67	2.34	2.35	2.42	2.50	2.53	2.53	2.53

Fig. 11. Average Latency (ms) for Real Time and Best Effort flows with a frame size of 512 Bytes in case of congestion

Perceived Voice quality end-to-end testing

The aim of the test described in this section was to determine the perceived QoS of a VoIP call between a SIDE A and a SIDE B including the end-to-end test network as depicted in Figure 12.

Procedure: In order to determine the conversational quality of a voice link a Voice Quality Tester was used. The test suite consisted of the following four steps:

1. One-way listening, speech quality in both directions with inside video-signal-transmission (video-telephony switched on) Measurement conditions
 - VQT Release 4.300
 - Voice Phone Adapter

- Method PESQ
 - 10 samples
 - Direction A → B and B → A
 - IP VideoPhone Innomedia
2. One-way listening, speech quality in both directions without inside video-signal-transmission (video-telephony switched off) Measurement conditions are the same as for step 1.
 3. One-way interaction, delay in both directions with inside video-signal-transmission (video-telephony switched on)
 4. One-way interaction, delay in both directions without inside video-signal-transmission (video-telephony switched off)

In addition a traffic generator was used to generate background traffic in the network for a more realistic scenario.

The MOS-LQ (Mean Opinion Score-Listening Quality) -values between 3,8 and 4,0 represent a good voice-quality. The directional difference of the values is caused by the different videophones. In both directions the measured delay time does not influence the inter activity of the conversation. The delay is marginal

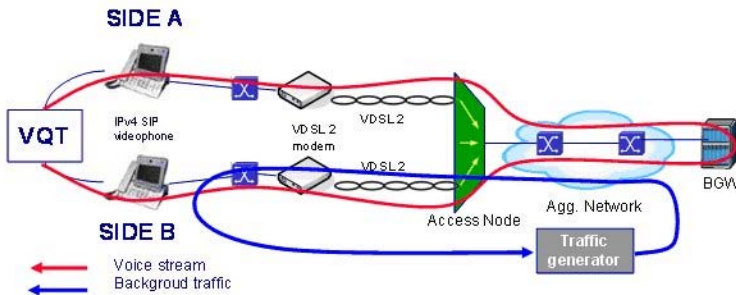


Fig. 12. Test set-up to determine the perceived QoS of a VoIP call

Direction	MOS-LQ (Average)			
	with video-signal		without video-signal	
A → B	3,82		3,80	
B → A	4,03		4,04	

Direction	Delay			
	with video-signal		without video-signal	
	minimum delay	maximum delay	minimum delay	maximum delay
A → B	130 ms	132 ms	121 ms	122 ms
B → A	101 ms	105 ms	92 ms	93 ms

Fig. 13. Perceived Voice quality end-to-end testing

influenced by the video signal transmission. But the difference is small compared to the whole delay (< 10 ms). The video signal does not influence the voice-quality.

Perceived QoS for IPv6 web browsing

The aim of the test described in this section was to determine the perceived QoS of a web browsing session. Procedure: Determine the WEB_MOS as described in [6]. The test starts by starting the browser on an IPv6 PC and then typing a URL. After this URL is typed and the “enter” button or “go” button is pressed, two stopwatches are started: the first stopwatch measures T1 and the second measures T2. The latter time can be related to the text “done” appearing on the status bar of the browser. If there is interactivity with the web page, the same is done for the second page, measuring T3 and T4. In the test the second page is obtained by clicking a link on the webpage which was loaded before. This procedure is repeated ten times to be able to deal with variations in these times due to variability in network behaviour.

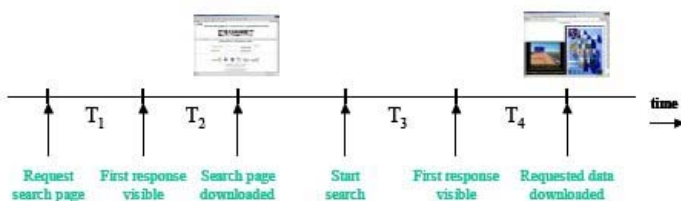


Fig. 14. Timing of a webpage downloading

The next step is to calculate the average values of T1...T4 which will be denoted as $\overline{T1}...$ $\overline{T4}$. Then, the weighted session time (WST) needs to be determined, which is defined in equation 1

$$WST = 0.47 * \overline{T1} + 0.60 * \overline{T2} + 0.71 * \overline{T3} + 2.22 * \overline{T4} \quad (1)$$

Finally the perceived QoS for the web browsing session is calculated:

$$WEB_MOS = 4.38 - 1.30 * \ln(WST) \quad (2)$$

Results: The measurement method using a stop watch is quite difficult to handle because of all the short times below 800ms. It was decided to use the network protocol analyzer “Ethereal” to measure the different times described above. What means that the time between receiving an http packet and displaying the content on the monitor is not considered, leading to better results for MOS values than using the stop watch method. T1 is the time between the first “GET” of the client PC and the first received http packet with content from the server. T2 is the remaining time until the last http packet with content from the server is

Table 3. Definition of MOS scale, based on Opinion Scores

MOS	Quality	Impairment
5	Excellent	Imperceptable
4	Good	Perceptable, not annoying
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very annoying

web page	$\overline{T1}$ (s)	$\overline{T2}$ (s)	$\overline{T3}$ (s)	$\overline{T4}$ (s)	WST	WEB_MOS
www.ripe.net	0.0477	0.7918	0.0688	0.5461	1.7587	3.65
www.deepspace6.net	0.0981	0.5446	0.0958	0.47	1.4843	3.87

Fig. 15. WEB_MOS

received. For T3 and T4 this measurement was repeated after clicking a link on the originally page. The test was carried out with two different web pages.

For the page “www.deepspace6.net” the header icon was requested and received after 2s. This led to a $\overline{T2}$ value of around 2.5s and thus to a WEB_MOS of only 3.1. In the table above an adjusted value is used.

Furthermore some additional details according to the test setup:

- Client PC with MS Windows Vista Home Edition; Intel Core 2 Duo E6400 @2.13GHz; 2048 MB RAM
- Cookies and browser cache deleted before each test step
- Each test step starts from an empty page

The subjective QoE of web browsing is pretty good for the PlaNetS [2] network. The objective measurement according to the test method described above results in nearly good quality for the tested web pages. For the entire test case “QoS for web browsing” it is important to have in mind that a lot of different parameters have an impact on the objective measurement (especially PC and web server performance). The WEB_MOS value can strongly differ between different web pages.

8 Conclusion

The QoS concept presented in this paper is an innovative pragmatic and scalable QoS solution by a resource management system guarded by a flexible Policy Control Framework. With low implementation effort, fully traffic engineered network resources can be guaranteed on demand and real-time on a call-by-call basis. QoS together with QoE will be the “enablers” of the multi-provider multi-service concept within the Next Generation Networks (NGN) Access Network architecture. The implementation work presented in this paper shows a validated solution to achieve quality-based multimedia delivery over DSL broadband access networks.

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