D. Goderis, H. De Neve, Y. T'Joens, J. De Vriendt, T. Soetens

Next generation network architectures will have to provide extensive support for IP Quality of Service to meet the needs of advanced services.

Towards an integrated solution for multimedia over IP

Introduction

Today's Internet attempts to deliver all traffic as soon as possible within the limits of its abilities, but without any guarantees related to throughput, delay, delay variation (jitter) and packet loss. This "best effort" forwarding paradigm has worked well so far because most applications running on the Internet Protocol (IP) are low priority, low bandwidth data applications with a high tolerance to delay and delay variations. However, this is changing rapidly. Real-time (interactive) multimedia Internet applications (telephony, video) and virtual private networks require a guaranteed bandwidth and/or strict timing requirements. Moreover, the number of Internet users is growing exponentially, further increasing the probability of congestion and leading to even higher network delays and packet loss. The key challenge is to extend existing IP networks with scalable, multi-service capabilities, while still providing the advantages of IP networks that made the Internet possible. On such multi-service networks, operators need to honor complex Service Level Agreements (SLA), covering different types of traffic in terms of bandwidth requirements and other quality parameters. This is what IP Quality of Service (QoS) is about.
IP QoS profoundly affects the way that network elements are designed. Alcatel has extensively investigated the consequences of QoS deployment and is continuously upgrading its product portfolio with QoS capabilities. A series of Alcatel papers treats specific QoS-related issues or particular Alcatel products, for example, the scalability of multi-service IP backbones [1], traffic engineering [2], the Alcatel 7770 core router [3] and the Alcatel IP Management Platform [4].

Alcatel's Vision for QoS in IP Networks

Alcatel's vision for providing QoS in IP networks can be summarized in five key observations.

1. IP is the preferred end-to-end multi-service layer

Providing end-to-end QoS for IP networks has been a hot topic both in the standardization bodies and the industry for the past decade. All of us have witnessed the IP versus Asynchronous Transfer Mode (ATM) debate. The tremendous success of the Internet has resulted in IP being chosen as the end-to-end multi-service layer. IP "bridges" several distinct layer 1 and layer 2 technologies and provides a common layer 3-network interface towards services and applications (so-called *horizontal integration*). Thus QoS guarantees on throughput, delay, jitter and so on must be expressed at the ubiquitous IP layer 3.

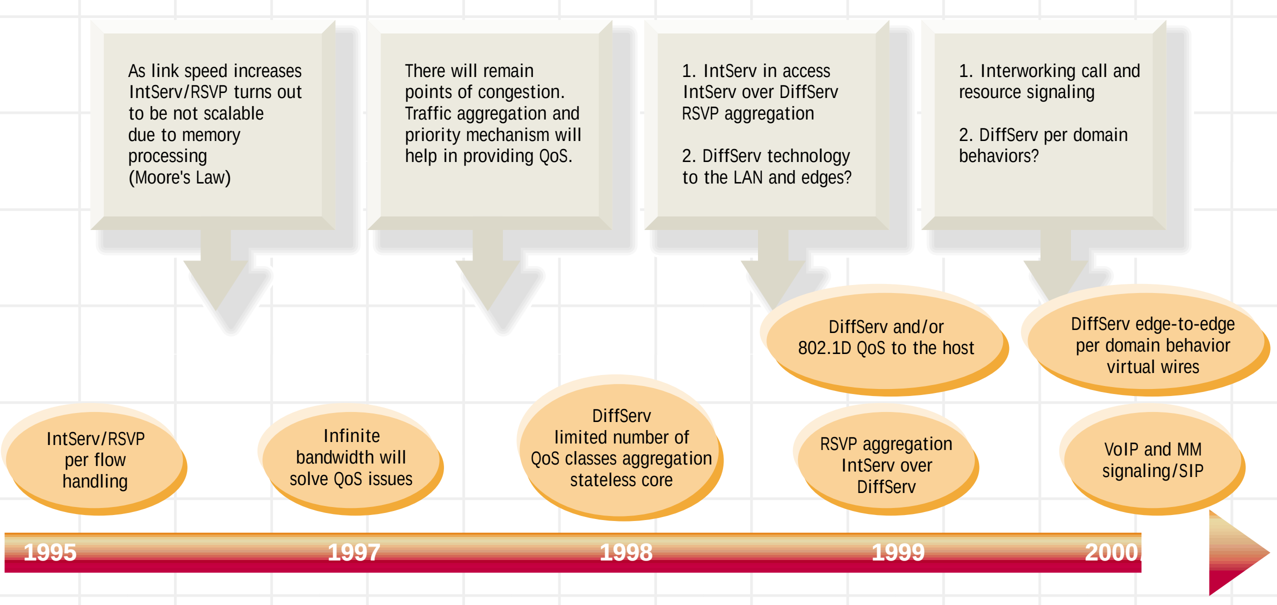
2. Bandwidth brokerage is the key to resource provisioning in IP networks

Providers and users negotiate SLAs relating to the services to be provided. The network provider must then ensure that sufficient resources are provisioned to support the SLAs committed by its domain. Therefore every router has to be configured so that sufficient resources, in terms of bandwidth and buffer space, are available to support the SLAs committed by the provider's domain. This requires a tight and smooth interworking of the data, control and management planes. *Vertical integration* of this type is realized by the bandwidth broker, which controls the domain's network resources and integrates service, resource and network management.

3. DiffServ is the dedicated IP QoS technology for backbones

Several IP QoS technologies have been proposed within the Internet Engineering Task Force (IETF). *Figure 1* gives an overview of the possible candidates for providing QoS across IP networks.
Alcatel's QoS strategy for high-speed networks is based on the IP DiffServ (Differentiated Services) technology, which uses flow aggregation to deal with scalability issues. The edge and access part of the network involves per-flow management, which depends on the provider's resource provisioning strategy (see section on "Resource Provisioning Strategies") and the available access technology (illustrated in the section on "Voice and Multimedia over IP").
DiffServ is based on the marking of IP packets with priority information, the Differentiated Services Code Point (DSCP) [5]. DiffServ capable routers implement differ-

Fig. 1 IP QoS technology evolution in the IETF

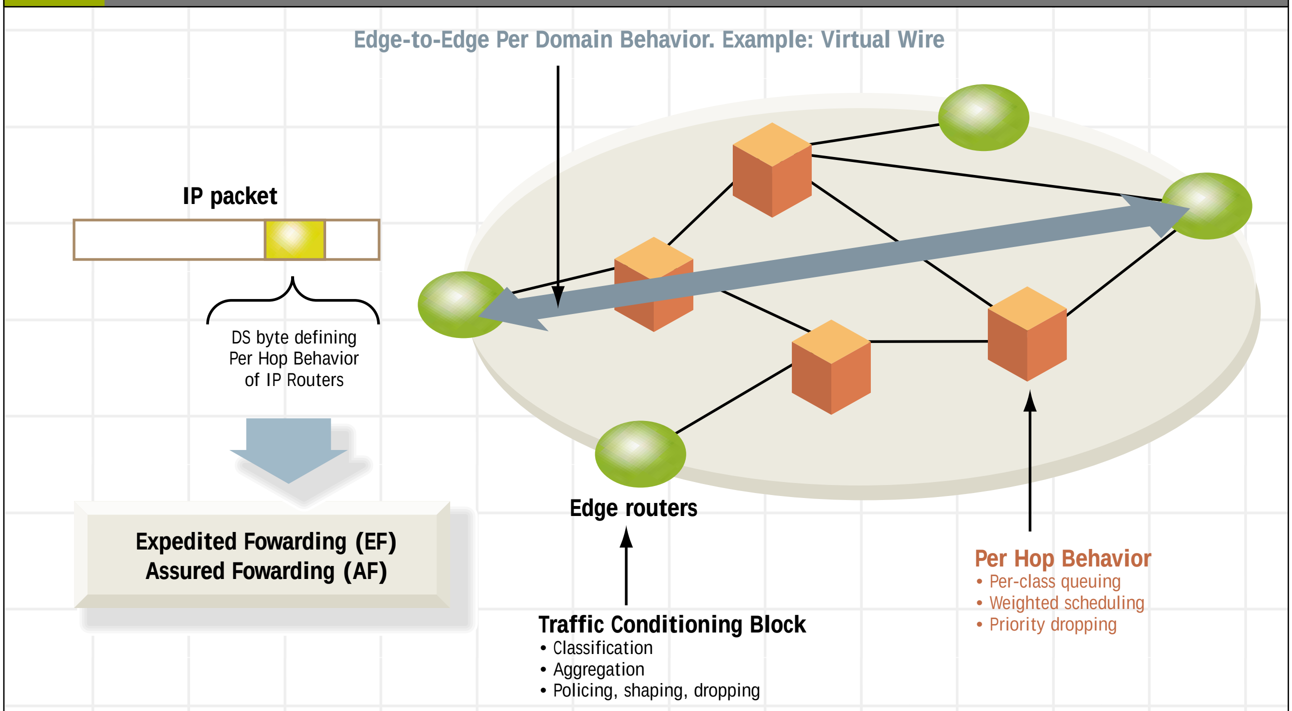


DiffServ: Differentiated Services **IntServ:** Integrated Services **MM:** MultiMedia **VoIP:** Voice over Internet Protocol

ent packet forwarding behaviors, known as Per Hop Behaviors (PHB), for distinct traffic types based on the DSCP value in the IP packet header (see *Figure 2*). This differential treatment of aggregate packet streams (i.e. per DSCP value) makes DiffServ routers scalable, even at Gigabit/s link rates.

The DiffServ architecture offers network operators a series of elementary QoS building blocks, including PHBs and the Traffic Conditioning Block (TCB). In a DiffServ domain, edge-to-edge packet behaviors, known as Per Domain Behaviors (PDB), are realized by traffic conditioning at the edge and by concatenating PHBs in the core network [5]. For example, the Virtual Wire (VW) PDB, which is based on the Expedited Forwarding (EF) PHB, offers a guaranteed maximum edge-to-edge packet delay and a guaranteed throughput for an aggregate IP packet stream, that is, it defines edge-to-edge pipes. More information on the DiffServ architecture and how it is implemented in the Alcatel 7770 RCP core router can be found in [3].

Fig. 2 DiffServ QoS building blocks



4. Existing and emerging layer 2 QoS technologies in access networks enable fast and cost-effective end-to-end multimedia deployment

IP provides the common end-to-end layer 3 network interface towards applications. However, topologies and technologies are much more diverse in access networks than in core networks, hampering the rapid deployment of a universal IP QoS solution. Alcatel's main strategy in the access area is to rely on the dedicated QoS capabilities of the underlying transport layer (as outlined in the section "Voice and Multimedia over IP") for ATM broadband and Universal Mobile Telecommunications System (UMTS) mobile networks. This re-use strategy avoids unnecessary protocol overhead and aims to provide the most cost-effective solution for each type of access network. Of course, in access networks without any intrinsic layer 2 QoS capabilities, and when in-band signaling is really required for per-flow resource reservation, the IP Resource Reservation Protocol (RSVP) could be used.

5. Application-based call signaling is the glue for network interworking

Alcatel's strategy is to include inter-domain signaling and call admission control as part of the application plane functionality. The strategy is to fully exploit existing protocols, like the Session Initiation Protocol (SIP) or H.323, for call control, routing and admission. The choice of SIP or H.323 will depend on the operator's available technology. The gatekeeper (or call server), which is part of the application plane, is responsible for per-call admission control for the IP network:

- SIP messages such as INVITE typically contain information for call routing (e.g. userB@domain_name), the host address (IP address and port number on which the message sender would like to receive the multimedia stream), user identity and credentials, and media capabilities such as the supported codec types. The indication (in the SIP message) of the codec type, for example, allows capability negotiation between the end points.
- H.323 is an umbrella standard that encompasses a set of other standards: H.225 for registration and authorization, H.225 for call setup and H.245 for capability exchange and to establish a media session.

Resource Provisioning Strategies

The network provider may adopt several provisioning strategies for supporting real-time (interactive) multimedia services, depending on the (expected) ratio between real-time and non-real-time network loads, and how this load is apportioned across the link and network capacity. From this point of view, the operator might opt for one of three provisioning strategies: over-provisioning, loose control or strict admission.

Over-provisioning

When no QoS technology is available for differentiating between multimedia and best-effort packets, real-time applications are supported by provisioning a network capacity that is far in excess of the total offered load. This might, for example, be a valid solution for Tiers1 providers whose business consists in selling bandwidth pipes to other Internet Service Providers (ISP). Resource management consists in controlling the network through measurements and monitoring. In this way the operator can upgrade the capacity over time, or drop the degree of over-provisioning needed to offer the promised statistical guarantees.

Loose Control

A second strategy is to over-provision the network for real-time multimedia traffic only. This requires a packet prioritization mechanism in the network elements based, for example, on the DSCP. Such a strategy builds on the premise that interactive traffic is a relatively small proportion of the best effort traffic, so no admission control is required per real-time flow. However, it requires real-time traffic streams to be policed at the edges of the network to protect scarce network resources. A monitoring and measurement architecture may further enhance this solution. High priority traffic is monitored on the links; in the event of an alarm threshold being passed, new multimedia SLAs should be refused or new capacity added to the network.

Strict Admission

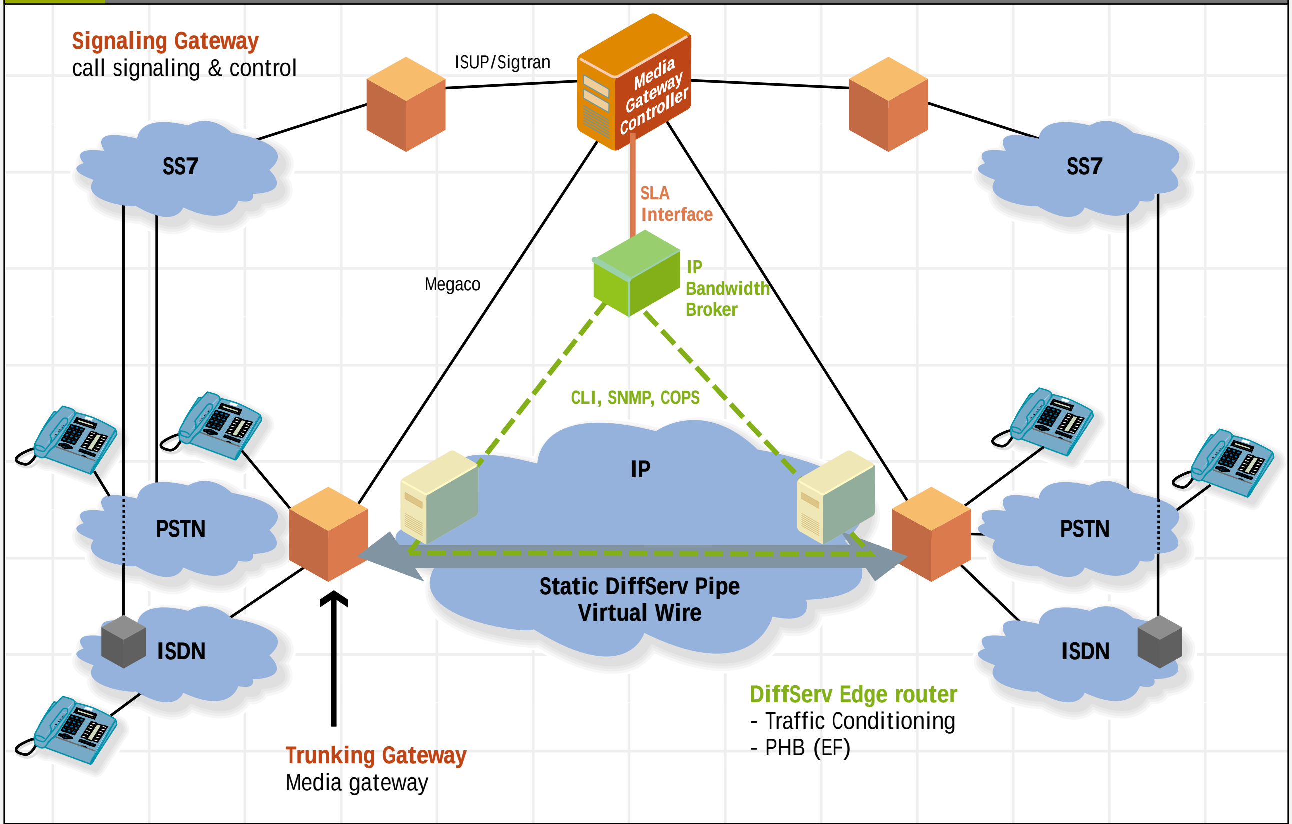
The third strategy is based on strict resource provisioning and admission control per real-time multimedia flow. This solution is needed in scenarios where the real-time multimedia load is a substantial part of the total capacity, such as in UMTS radio access networks. As in the loose control strategy, it also needs QoS-based packet differentiation in the routers and policing of real-time traffic at the edges. In addition, the strategy requires interworking between call signaling and resource provisioning in order to execute the per-flow admission control. Two strategies can be chosen to implement strict admission: *bandwidth brokerage* and *in-band signaling*.

Strict admission through bandwidth brokerage

The bandwidth broker architecture, which is Alcatel's preferred implementation, is applied to IP core networks, as shown in *Figure 3*, *4* and *5*. In these scenarios, the ISP sells connectivity services to an Application Service Provider (ASP) through SLAs, each of which describes a mesh of pipes with a guaranteed throughput, packet loss and delay bound (for each established pipe). The ISP mainly controls the transport plane while the ASP is responsible for the functionality of the application plane. Strict control of each multimedia flow is achieved using the following steps:

- The ISP is logically equipped with a central entity (the bandwidth broker), which has an overall view of all the available network resources and topology. In a DiffServ network, for example, the committed SLAs (for real-time traffic) might be configured as a mesh of virtual wires.
- The ASP is also equipped with a central entity, the call server, which might be a SIP proxy, an H.323 gatekeeper or a Media Gateway Controller (MGC). The contracted

Fig. 3 Connecting trunking gateways over an IP DiffServ backbone



SLA provides the call server with a dedicated partial view of the transport provider's network resources. The call server only has a view of its logical overlay network, but knows nothing about the internal network details.

- Admission control per multimedia flow is "outsourced" by the ISP's bandwidth broker to the ASP's call server. The latter uses an application signaling protocol, such as SIP, to monitor all individual ongoing multimedia calls under its authority. The call server knows the capacity of the pipes and the required bandwidth of an individual call, and is therefore in a position to exercise strict admission control.

In-band resource reservation signaling

An alternative strict admission control implementation is based on using an in-band signaling protocol for resource reservation. Examples are Packet Data Protocol Context Activation (PDP-CA) in UMTS and RSVP in IP IntServ. The in-band signaling protocol reserves resources in all network elements along the data path; each network element executes the resource admission control per flow. This method also requires synchronization and interleaving of the call signaling and the in-band resource reservation. As this is the most complex solution, it should only be used in networks, such as the UMTS radio access network (*Figure 5*), that really require this capability. The UMTS network needs this functionality as it is expected that real-time multimedia will make up a substantial part of the overall traffic, radio resources are scarce and mobility requires per-flow state information at the headend of the UMTS access network.

Voice and Multimedia over IP

This section illustrates the principles outlined above for several network architectures and business scenarios. In particular, Alcatel's solution for real-time multimedia over IP is illustrated for network configurations involving IP network connectivity to the Public Switched Telephone Network (PSTN), to an ATM access network and to the UMTS mobile network.

Connecting Trunking Gateways

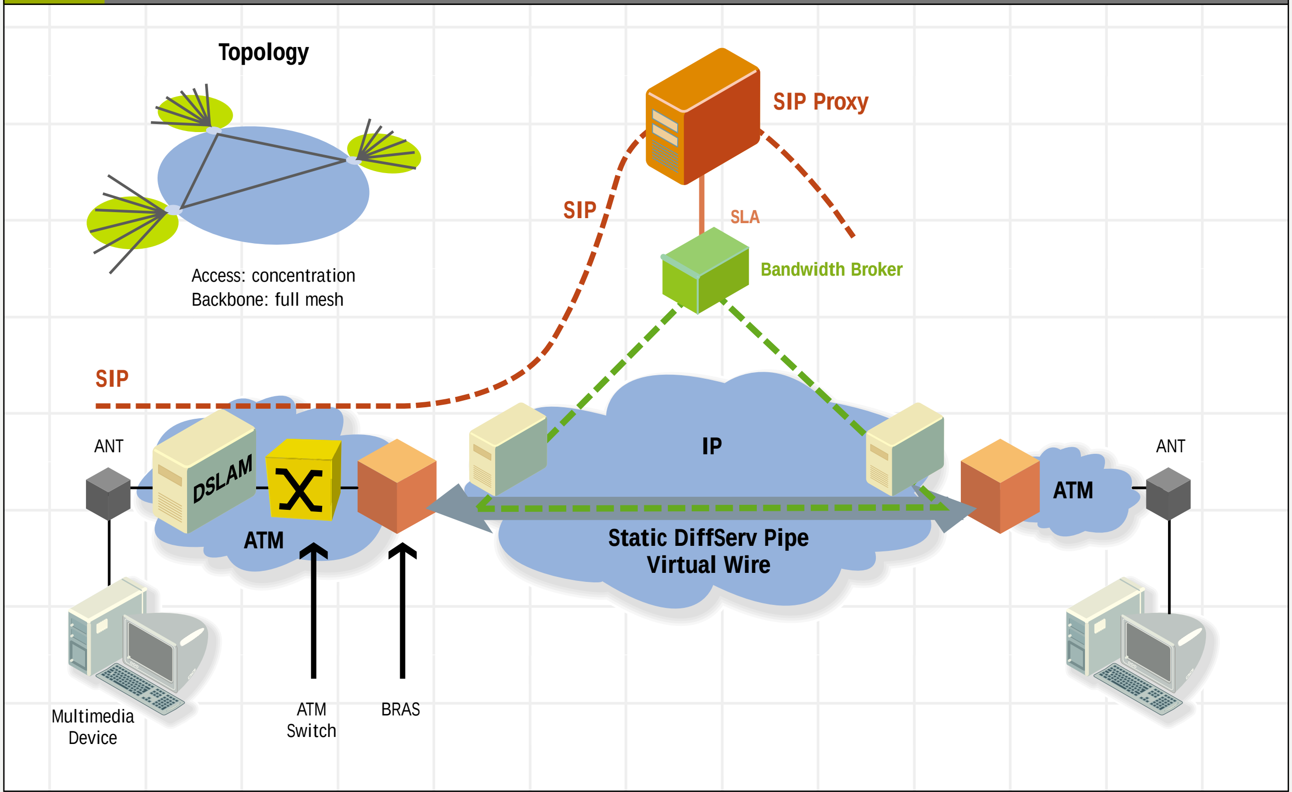
This scenario deals with the transport of voice calls over an IP core network by interconnecting trunking gateways (see *Figure 3*).

IP transport providers (or ISPs) sell transport connectivity services to voice providers. The latter have a restricted number of trunking gateways, signaling gateways and a central call server – the media gateway controller. The GateWays (GW) are logically interconnected by DiffServ virtual wires, yielding a mesh of edge-to-edge pipes; they are physically connected to DiffServ edge routers.

The SLA (between ISP and voice provider) outlines the number of gateways, the expected load between each pair of gateways (capacity of the trunks) and the maximum edge-to-edge delay of each trunk. Estimating the required trunk size is the responsibility of the voice provider. It can be based on various techniques, such as measurement-based traffic forecasting or an Erlang-B type of dimensioning calculus.

The bandwidth broker provisions the network, taking into account all SLAs, by configuring the edge and core routers under its control. Common Line Interface (CLI), Simple Network Management Protocol (SNMP) or the Common

Fig. 4 Providing interactive multimedia over ATM access



Open Policy Service (COPS) protocol can be used to implement the router configuration. Provisioning will typically be undertaken on a granularity of hours or more.
SS7 signaling is captured at the signaling gateway and the information is forwarded as ISDN User Part (ISUP) messages over SIGTRAN – a signaling transport protocol defined by the Internet Engineering Task Force (IETF) – to the MGC. The MGC performs admission control based on its knowledge of all ongoing calls and the size of the provisioned pipes (SLA).

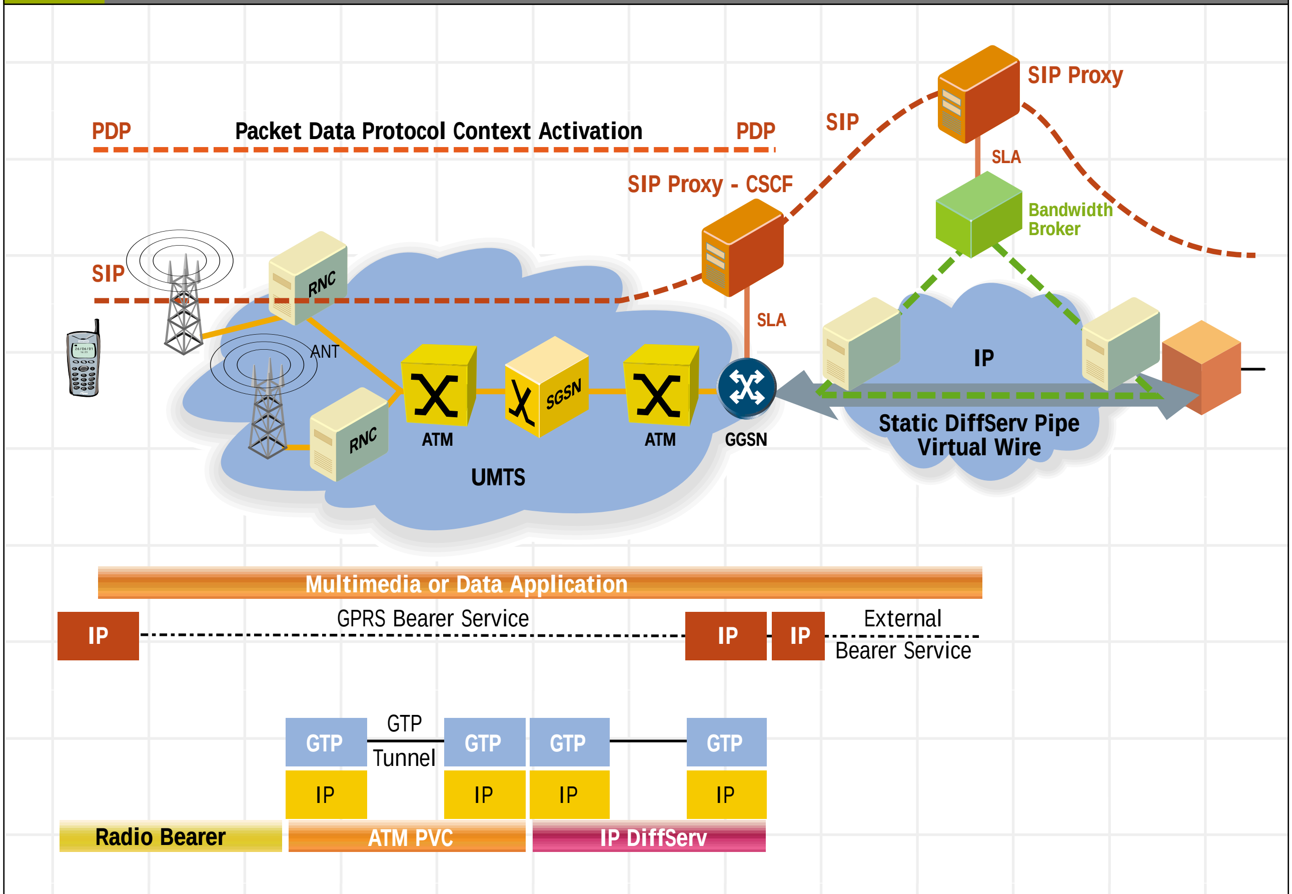
Multimedia over Broadband Access

An ASP offers real-time multimedia and best-effort data services to residential users through ATM broadband access (*Figure 4*). An IP core network, owned by an ISP, interconnects ATM access networks. Eventually the ASPs and access providers, which own the access equipment, might be separate businesses.
The access network concentrates residential users, that is, each Broadband Remote Access Server (BRAS) terminates numerous ATM connections and acts as a Point Of Presence (POP) for many residential users. This centralized BRAS solution predominates in today's networks. It limits the number of BRASs and therefore allows full mesh connectivity between all of them.
ATM, which is now a mature standard technology, is definitely the best existing QoS-capable protocol for broadband access [6]. The concept of Virtual Connections (VC) allows data and multimedia to be differentiated in a simple way. All real-time multimedia packets (or ATM cells) are put onto a guaranteed (constant bit rate) VC, while data packets are aggregated on a best-effort (unspecified bit rate) VC. This allows the Digital Subscriber Line Access Multiplexer (DSLAM) and ATM switches to give voice and video traffic priority over best-effort data traffic. Thus a small delay is guaranteed in the access network. The BRAS acts as a DiffServ router for upstream packets coming from the access network. It marks the packets with the appropriate DSCP value before inserting them on the IP core network. In the case of downstream traffic, the BRAS must put high priority packets on the guaranteed ATM VC and low priority traffic on the best-effort VC. This decision is based on the packet's DSCP value.
Alcatel's resource provisioning strategy in the access network has two elements. Between the DSLAM and the BRAS, the strategy is based on loose control, that is, the network is highly over-provisioned for multimedia traffic. This is possible because, beyond the DSLAM, multimedia calls are aggregated and the majority of traffic is best-effort data. Furthermore, there is no extra cost because the spare bandwidth can be fully used by data traffic.
Loose control is impossible for the "last drop" link between the ADSL Network Termination (ANT) and the DSLAM; strict admission is required here. However, because strict admission is only required for this single link to the user, there is no need for a dedicated in-band resource reservation protocol. Also, resource admission control can be piggybacked on the application signaling protocol (SIP). This can either be implemented at the central call server (the SIP proxy) or at the ANT, which then becomes a SIP proxy.

UMTS Mobile Access

End-to-end QoS support in UMTS networks is currently being standardized in the Third Generation Partnership Project (3GPP). The UMTS architecture should provide end-to-end QoS by concatenating the so-called General

Fig. 5 UMTS access connected to an IP backbone



GTP: GPRS Tunneling Protocol **PDP:** Packet Data Protocol **PVC:** Permanent Virtual Circuit

Packet Radio Service (GPRS) bearer service and the external bearer service (*Figure 5*). A bearer service includes everything that is needed for provisioning a contracted QoS, including control signaling, differential data forwarding and QoS management functionality. The external bearer service depends heavily on the network to which UMTS routes the call. In the case of IP core networks, Alcatel's strategy is based on the DiffServ QoS architecture combined with a bandwidth broker and strict admission per multimedia flow. In this section we concentrate on the QoS architecture of the UMTS access network [7], that is, the GPRS bearer service, which is the main topic of 3GPP standardization.

The GPRS bearer service provides the various value-added (QoS) services that a UMTS operator might offer. It is deployed between the Mobile Terminal (MT) and the gateway to the IP core, the Gateway GPRS Support Node (GGSN) node. 3GPP has standardized four QoS classes: conversational, streaming, interactive and background (best effort) [7]. Each MT negotiates a QoS class and related parameters with the UMTS network, using the PDP-CA protocol defined for GPRS. Some UMTS-specific modifications are currently being standardized in 3GPP.

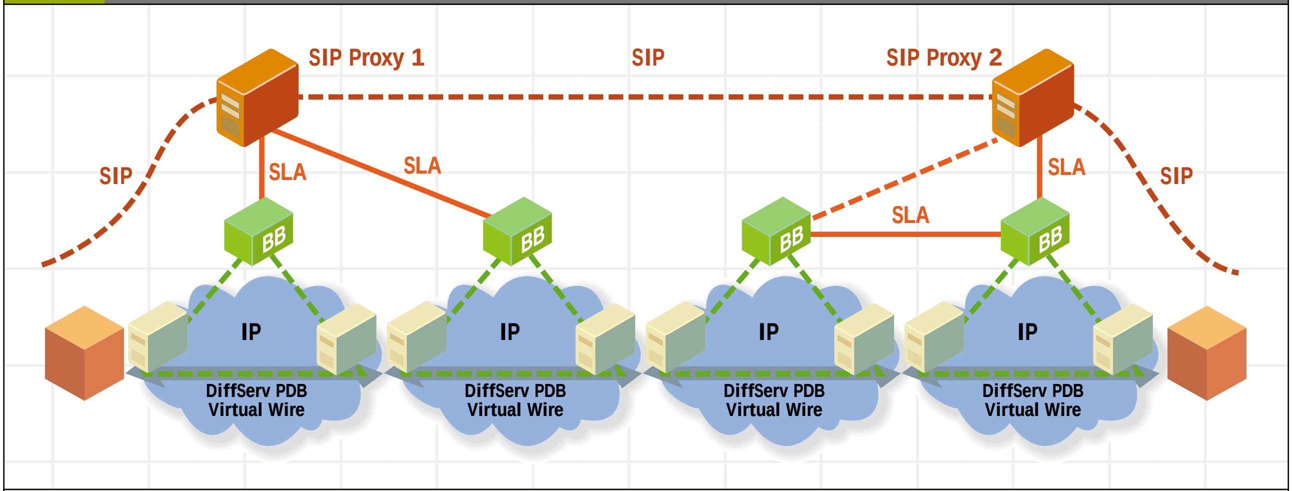
This reservation protocol is required for the strict admission of multimedia flows in the UMTS access part of the network, where resources are scarce. A per-flow tunnel has to be set up (by PDP-CA) because of the mobility aspect, that is, the tunnel might shift from one radio cell and/or Radio Network Controller (RNC) to another. This requires per-flow state information at the Serving GPRS Support Node (SGSN) and at the GGSN. The GGSN gateway must also offer the required mapping between the UMTS QoS parameters and the QoS parameters of the external bearer service. In the case of an IP DiffServ network (*Figure 5*), the GGSN should act as a DiffServ router and must implement similar capabilities to those of the BRAS in the broadband access scenario. There is a need for coordination between SIP call signaling and PDP-CA resource reservation. In 3GPP it is assumed that the GGSN communicates with a dedicated SIP proxy, the so-called Call State Control Function (CSCF) in UMTS terminology.

End-to-end call admission control remains in the application plane and is performed via the SIP protocol. This protocol is the glue between the UMTS access network and the IP core network. Each SIP proxy along the path must admit the call or it will be blocked.

Evolutionary Scenarios

The business scenarios outlined earlier are focused on a single IP backbone connected with the PSTN network and/or multiple access networks. These scenarios cover the vast majority of QoS deployment for the coming years. It is indeed expected that QoS and multimedia will be deployed over a single backbone based on peer-to-

Fig. 6 Evolution scenarios for IP QoS across multiple domains



BB: Bandwidth Broker

peer SLAs between ISPs, ISPs and access networks, or ISPs and application service (voice/video) providers. Alcatel has a strategy for dealing with all of these scenarios and can provide an integrated solution based on a coherent and flexible IP QoS strategy. However, Alcatel is also convinced that in the long term QoS will become a reality in the Internet at large, implying a solution to the problem of end-to-end IP QoS across multiple domains. *Figure 6* shows how Alcatel's strategy anticipates this evolution. The figure shows several interconnected IP core networks, omitting the access networks (or PSTN) connected to ISP A and ISP D.

In this architecture, each ISP deploys a bandwidth broker, controlling its network resources and giving the ISP the opportunity to offer value-added IP services. The bandwidth broker might only have basic functionality or might be sophisticated, depending on the chosen provisioning strategy.

A multimedia ASP, for example, a VoIP provider, implements a central SIP proxy or gatekeeper. Each SIP proxy performs the per-flow admission control for all transport domains under its authority:

- Each SIP proxy may interface with one or more transport domains based on bilateral SLAs. For example, SIP proxy 1 has a partial view of the network resources of domains A and B.
- A SIP proxy may also indirectly have a view of the network resources of a transport domain. This is illustrated by SIP proxy 2 having an indirect view of the resources of domain C. In this case, ISP C positions itself in the market as a Tiers1 provider, selling bandwidth to other ISPs (wholesale service providers). ISP C and ISP D have a peering SLA describing virtual wire bandwidth pipes.
- Each transport domain should be "covered" by at least one application gatekeeper/SIP proxy. This is realized through SLAs; how the SLAs are actually contracted is a business issue rather than a technical problem.

The DiffServ (ingress) edge routers perform the policing for aggregate flows based on the SLA contract. Each SIP proxy performs the per-flow admission control for all transport domains under its authority. Call routing is based on information in the SLAs and SIP. It should be noted that in normal SIP operation, the proxy has no knowledge of the actual route followed by multimedia packets. In the solution described here, the SIP proxy knows all about the ingress/egress routers in the domains under its authority (based on SLA information). Therefore the (sequential) SIP proxies are aware of the Border Gateway Protocol (BGP) path of the multimedia stream.

Dynamic SLA/SLS negotiation

One of the drawbacks of the solution described above is the static provisioning of the virtual wire pipes. As ISPs and ASPs negotiate the SLA "off-line", the bandwidth capacity of the pipes is static. Dynamic negotiation of the technical part of the SLA – the Service Level Specification (SLS) – could allow more dynamic provisioning of the pipes. For example, it would allow ASPs to launch "increase" or "decrease" requests for bandwidth capacity, depending on the number of calls handled by the SIP proxy.

However, dynamic negotiation requires a well-defined SLS format. Therefore a further step towards QoS across the Internet is a standard format for the SLA technical specification. Alcatel is currently proposing a standard SLS template at the IETF [8]. The multi-vendor environment requires a standard set of parameters and semantics for the SLSs being negotiated at different locations. A standard SLS template should also allow for a highly developed level of automation and dynamic negotiation of SLSs between customers and providers. Automation and dynamics are helpful in providing customers (as well as providers) with the technical means for dynamic and automated QoS provisioning.

Conclusion: Strengths of Alcatel's Bandwidth Broker Solution

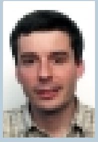
Deployment of QoS-based real-time multimedia services over IP networks is one of the most exciting challenges that service providers are currently addressing. However, it involves a technological paradigm shift in packet networking. One of the main technical challenges is *to tie everything together*, that is, to integrate the different technologies and the data, control and management functions. Alcatel has a clear strategy for meeting this challenge. Bandwidth brokerage plays a key role in Alcatel's solution. While the IP layer realizes the horizontal integration by tying together several layer 1 and layer 2 technologies, the bandwidth broker architectures ensures the smooth vertical integration of service, resource and network management. This management-oriented architecture for providing IP QoS offers a number of strategic advantages for operators implementing this solution:

- It allows strict separation between the transport (IP) plane and application (multimedia) plane functionality. This sound architectural design allows quasi-independent evolution of the transport and application technologies and services.
- It offers the operator a flexible, evolutionary resource provisioning strategy for the network. The bandwidth broker architecture may initially provide service assurance only through SLA monitoring and resource over-provisioning (as is mostly the case today in IP backbones). However, in time it will evolve to a fully controlled solution offering strict QoS guarantees to real-time multimedia flows.
- It solves the scalability problem for IP backbones by adopting a two-level approach for admission control. SLAs are admitted (by the ISP) on a longer timescale and are resource controlled by the bandwidth broker. The SLA offers the ASP (the ISP's customer) a logical overlay network with QoS guarantees for aggregate traffic. Real-time multimedia flows are admitted (by the ASP) on a per-call basis and are controlled by the call server or SIP proxy.
- It is a future safe architecture, enabling IP QoS *today* based on peering SLAs, while allowing for the incremental deployment of end-to-end IP QoS involving multiple ISPs, ASPs and access providers. The architecture is fully open for the (undoubtedly) numerous business models (and SLAs) that will emerge in the near Internet future. ■

References

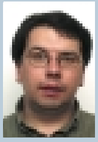
1. "A Vision of IP Multiservice Backbones and Proposed Implementation", *Alcatel Technical White Paper*. October 2000, URL: http://cid.alcatel.com/doctypes/techwhitepaper/pdf/a7770d.pdf.
2. "Traffic Engineering", *Alcatel Technical White Paper,* October 2000, URL: http://cid.alcatel.com/doctypesint/techwhitepaper/pdf/a7770c.pdf.
3. "IP QoS Support in the Internet Backbone", *Alcatel Technical White Paper,* October 2000. URL: http://cid.alcatel.com/doctypesint/techwhitepaper/pdf/a7770b.pdf.
4. "Alma Vision IP, A Service Assurance Solution for IP-centric Networks", *Alcatel Technical White Paper,* October 2000. URL: http://cid.alcatel.com/doctypesint/techwhitepaper/pdf/a7770a.pdf.
5. IETF DiffServ Working Group Charter, RFCs and Internet Drafts: http://www.ietf.org/html.charters/diffserv-charter.html.
6. ATM Forum "Traffic Management Specification Version 4.1", *ATMF specification af-tm-0121.000,* 1999.
7. "3GPP Specifications", URL: http://www.3gpp.org/3G_Specs/3G_Specs.htm.
8. "Service Level Specification Semantics and Parameters", *IETF Internet Draft,* November 2000, URL: http://search.ietf.org/internet-drafts/draft-tequila-sls-00.txt.

Danny Goderis is responsible for IP QoS technologies and IP traffic performance in the Traffic and Routing Strategy Project within the Alcatel Corporate Network Strategy Group in Antwerp, Belgium.

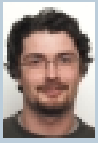


Hans De Neve is project manager of the Traffic and Routing Strategy Project within the Alcatel Corporate Network Strategy Group in Antwerp, Belgium.

Yves T'Joens is Access and Gateway Director within the Alcatel Corporate Network Strategy Group in Antwerp, Belgium.



Johan De Vriendt is project manager of the Mobile Networking Project within the Alcatel Corporate Network Strategy Group in Antwerp, Belgium.



Timothy Soetens is a research engineer in the Traffic and Routing Strategy Project within the Network Strategy Group in Antwerp, Belgium.