QoS-enabled IP networks for assured multiplay QoE

Technical White Paper

Nokia Siemens Networks

Insights into ResIP Certified multiplay infrastructure solutions

ResIP certified

Executive summary

The ability to deliver multiplay services over a converged IP network has become a crucial business objective for carriers and service providers dealing with declining revenues from traditional voice services. Multiplay services encompass voice over IP (VoIP), Video on Demand (VoD), IP television (IPTV), best-effort Internet access, content streaming, gaming, informational services, and other services that provide data, voice, or video capabilities.

Delivering multiplay services requires a substantial investment and often complex integration projects to introduce capabilities into the provider's infrastructure. Additionally, it is quite difficult to predict winning combinations of multiplay services, which is why a well-defined deployment strategy is necessary for success. This strategy must address high-quality operations and customer experience as well as the processes and solutions required for an easy-to-manage, highly reliable, secure, and future-proof IP network foundation. In particular, converged IP network infrastructures must guarantee sufficient quality of service (QoS) to give end users the expected quality of experience (QoE) with any multiplay service.

Contents

To help ensure provider success with multiplay services, Nokia Siemens Networks and Juniper Networks jointly established an end-to-end multiplay-enabled IP network infrastructure in the Resilient IP Proof-of-Concept Lab (ResIP PoC Lab) in Munich, Germany. In this comprehensive test environment, a variety of Nokia Siemens Networks SURPASS Solutions and Carrier Ethernet equipment are interconnected with Juniper Networks M-, E-, and J-series routers, SDX-300 service delivery, and ISG 2000 security equipment. The environment emulates realistic carrier-grade IP multiplay networks and enables the investigation of multiplay service performance issues. To address the aspects of administration, management, and the ability to meet service level agreements (SLAs), the ResIP PoC Lab also includes a highly scalable and flexible service deployment system based on the Juniper Networks Session and Resource Control portfolio.

To date, tests done within the ResIP PoC Lab have been based on the end user requirements proposed in the DSL Forum’s Technical Report, TR-126, “Triple-play Services Quality of Experience (QoE) Requirements”. Test results show that ResIP Certified multiplay infrastructure solutions give small to large carriers and service providers a strong foundation for the delivery of multiplay service bundles.

1. Introduction

Operators and service providers must be able to meet end user requirements.

To enjoy the cost-benefits of a converged IP network for multiplay services, operators and service providers must be able to meet end user requirements and expectations with a satisfactory level of service quality. Services must perform the same as, or preferably better than, corresponding services offered over non-IP transport networks. Consumers evaluate performance in terms of the perceived quality, reliability, and the ease of use of the services.

The DSL Forum has made significant progress translating the subjective QoE of end users into the required enabling QoS capabilities for a variety of entertainment related multimedia services involving voice, video, data, and gaming. The two terms QoE and QoS are often confused when it comes to service performance. This paper uses the DSL Forum's definitions of the terms:

QoE addresses the subjective perception of the quality of a service from the point of view of the end user. QoS relates to the network parameters that affect that service.

- **QoE** is the overall performance of a network from the user's point of view. It is a subjective measure of end-to-end service performance from the user perspective and is an indication of how well the network meets the user's needs.

- **QoS** is a measure of performance at the packet level from a network point of view. It refers to a set of technologies that allow a network administrator to manage the effects of congestion on application performance as well as providing a differentiated service to selected network traffic flows or users. QoS metrics include objective network layer parameters such as packet loss, delay, and jitter.

As depicted in Figure 1, TR-126 calls for a network engineering process that follows a top-down approach. This approach addresses two main areas that are strictly related to the aspects of QoE (the "User Space") and to QoS (the "Network Architecture Space"). Both areas must be addressed to meet the TR-126 criteria for multiplay services.

Design target is a QoS-enabled network infrastructure for multiplay services that outperforms end user requirements.

Nokia Siemens Networks and Juniper Networks, as network infrastructure suppliers, extensively focus the network design and validation work in the ResIP PoC Lab on QoS while taking into account the associated end user QoE guidelines. The goal is to develop end-to-end IP network architectures that exceed the end user needs.

As in real-world networks, the last mile from the DSLAM to the subscriber and the subscriber's in-house cabling are particularly prone to noise and other serious interference. Special care must therefore be taken with this aspect of design. The best access, aggregation, and core network designs will be ineffectual and end users frustrated when this last mile performs poorly. For example, bursty bit errors on DSL lines (up to 16 ms due to interleaving) can cause significant IP packet losses that by far exceed the combined effects of the rest of the IP network. These will result in frequent artifacts that are devastating to the QoE of entertainment video applications like IPTV or VoD.

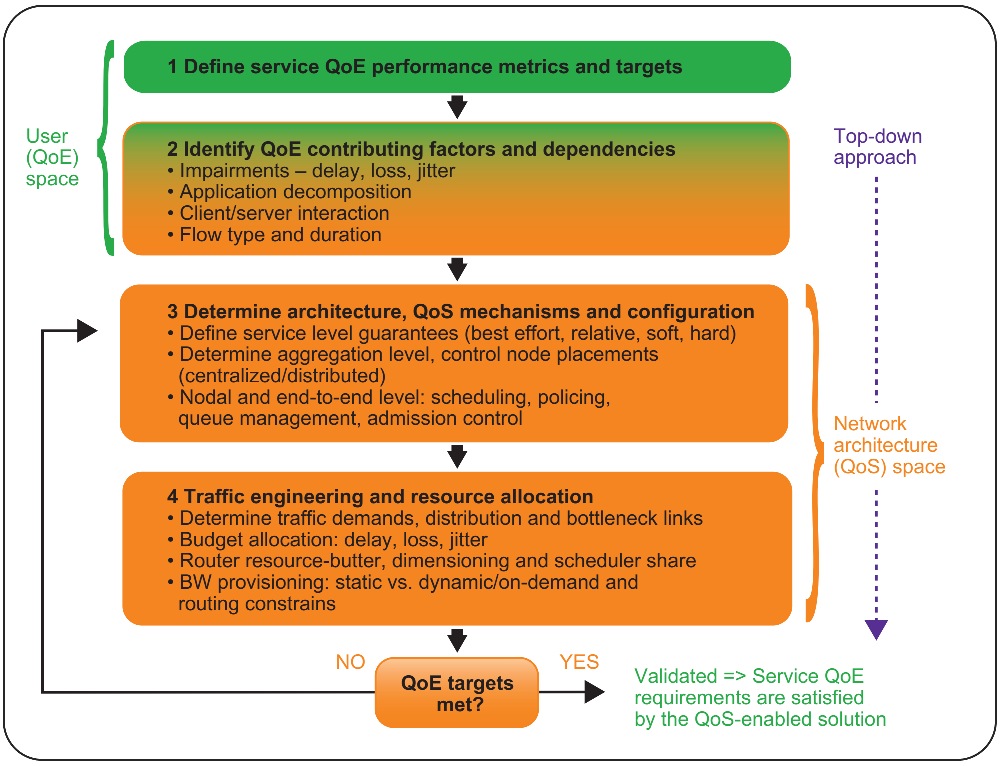

Figure 1: QoE/QoS engineering process: a top-down engineering approach as specified in the DSL Forum's TR-126

The introduction of new services over converged IP networks always involves some technical and commercial risks. To mitigate these risks, service providers must minimize capital and operational costs and evolve a powerful IP network infrastructure that provides secure, reliable, and scalable network services. Multi-play service bundles pose additional challenges, since individual services often have conflicting QoS requirements with respect to bandwidth, delay, jitter, and packet loss for the desired QoE. For example, voice services require comparatively little bandwidth, are sensitive to delays and jitter, but are quite resilient to packet loss. Entertainment video services, in contrast, consume large amounts of bandwidth, are very sensitive to packet loss, but are quite resilient to signal delay and jitter. To tackle this multiplay reality, a powerful resource and admission control solution must be combined with reliable IP access, aggregation, and core network designs.

The introduction of new services over converged IP networks is always associated with some technical and commercial risks.

In the ResIP PoC Lab, Nokia Siemens Networks and Juniper Networks have included resource and admission control features that use Resilient IP to help carriers and service providers successfully deploy QoS-enabled multiplay infrastructure solutions. Both companies bring in-depth real-world experience to the ResIP PoC Lab, making it a venue where service providers can validate, optimize, and certify end-to-end IP infrastructure solutions for multiplay services.

Nokia Siemens Networks and Juniper Networks have demonstrated their combined multiplay services expertise in the joint ResIP PoC Lab.

2. User expectations for multiplay services

It is critical to identify the criteria for end users' QoE when offering multiplay services.

The performance of multiplay services must be evaluated within an end-to-end scenario, with the primary goal being the satisfaction of the end users' expectations. To identify measurable criteria for end user satisfaction, many aspects must be considered. First, the user requirements for setting up and removing a service must be measured, as well as the performance seen by the user once a service is established. Second, many inherent application and transport layer parameters must be taken into account. These contribute to the overall quality of experience in terms of overall responsiveness of an application, data integrity, availability, and reliability, as well as security issues. Finally, the service provider's business point of view is a critical aspect, since providers must offer innovative services to compensate for and potentially outweigh the declining revenues from traditional TDM voice services.

Since customers prefer simple service selection capabilities, a well-defined service deployment strategy must be followed.

Figure 2 shows an example of a multiplay service bundle that could be offered to residential subscribers. Since customers like to build customized bundles based on a comprehensive set of services, it must be fast and easy for both the end user and the service provider to add or remove a service from a user's bundle. Implementations that enable this capability are by no means simple, and can require

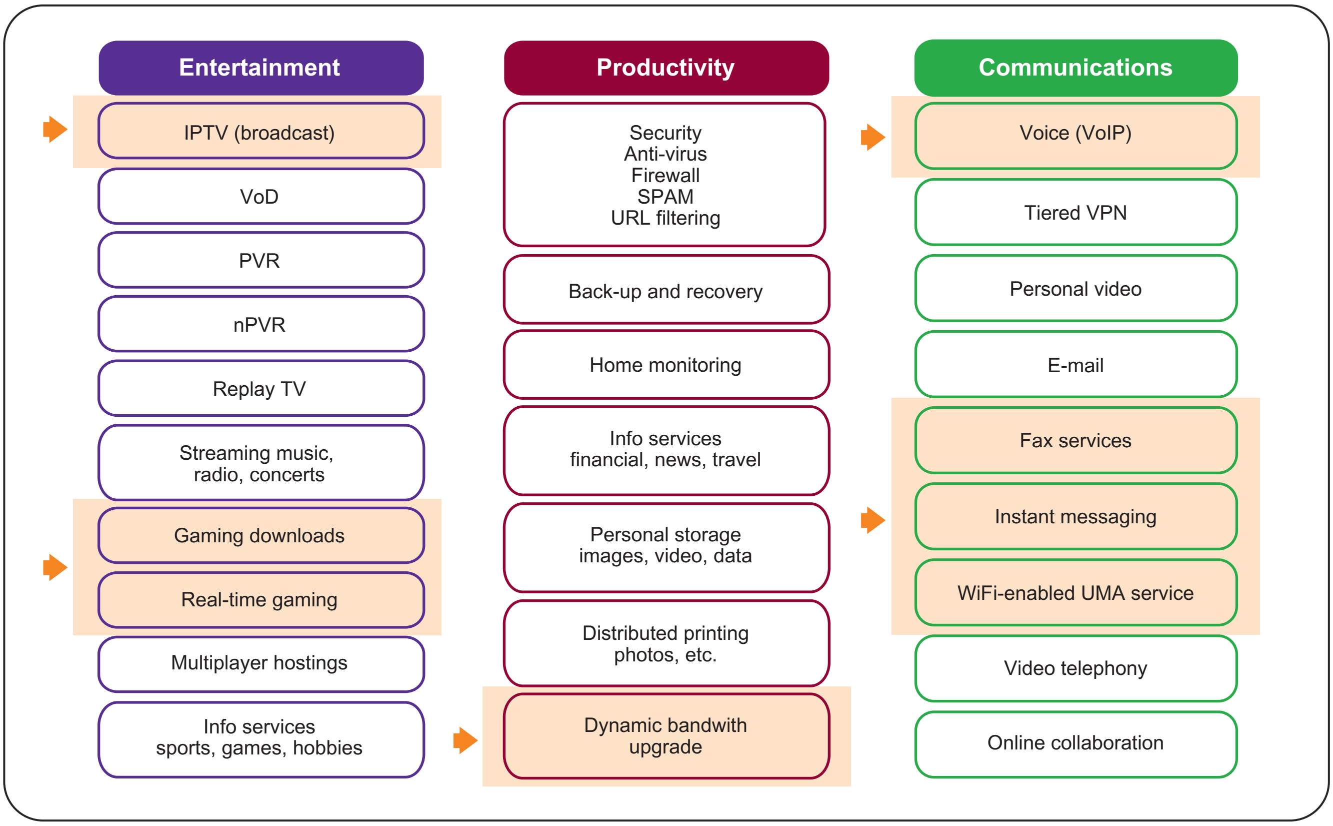

Figure 2: Example of a multiplay service bundle. Customers can select those services they want, enabling the service provider to offer customized packages that generate revenue

substantial investments in equipment and a complex integration process. It can be quite difficult to predict winning combinations of services, making this bundling flexibility even more important to success in the marketplace.

Subscribers have expectations based on experience with traditional service offerings.

Today, many voice and video services are provided over dedicated proprietary infrastructures including traditional TDM voice, terrestrial TV broadcast, broadband cable, and satellite networks. Experience with services delivered over these networks have led to specific subscriber expectations for service behaviors in terms of usability, availability, reliability, responsiveness, and quality. Attracting and keeping customers within this competitive market requires an excellent QoE and more than “me too” services.

Users are focused on QoE and don’t care about the technology behind a service.

As previously mentioned, users do not care about the technology behind a multiplay service unless it causes service quality degradations. To ensure the success of multiplay offerings, technical deployment risks must be kept to a minimum. A converged IP network must provide individual services with sufficient QoS to guarantee QoE for the end users even under worst-case conditions. The challenge from the service provider and carrier point of view is therefore maintaining converged IP networks that can enable:

- Assured QoS in real time (minimum packet delays and jitter) with exceptional interactive responsiveness and only minor visual or acoustic impairments due to congestion or packet loss.
- Support of flexible multiplay service bundles including high-definition video services and premium content.
- High reliability and availability including security and rapid protection switching mechanisms as well as fast network recovery in the event of failures.
- Easy and convenient multiplay services handling.

Apart from delivering a satisfactory QoE to end users, carriers and service providers have to reduce capital and operational expenditures to increase revenues and protect existing assets. Meeting these needs requires:

Carriers and service providers have to protect their assets and reduce investments.

- Cost-efficient networks that enable IP-based unicast and multicast service bundles.
- Advanced management tools and provisioning guidelines to minimize operational efforts.
- Scalable and performance-optimized IP infrastructure solutions that enable fast time to market for innovative multiplay services.

3. Understanding QoE/ QoS requirements

QoE degradations mainly originate from the control and data plane.

The provisioning of multiplay services – such as voice, video, gaming, and data – raises several questions with respect to end user QoE and network QoS. Several layers (service, application, transport) and planes (control, data, usability, content, reliability, security) affect QoE. From the network infrastructure point of view, the control and data plane of the transport layer are of importance to QoE. Therefore, the information in the following sections (summarized from TR-126) focuses on these areas.

3.1. Video services requirements

Packet loss rates need to be less than 10^{-6} for SDTV or less than around 5×10^{-8} for HDTV video services with satisfactory QoE.

Figure 3, taken from TR-126, provides a graphical representation of transport layer packet loss relating to QoE for entertainment video services. For standard-definition television (SDTV) video applications, a one hour loss distance is recommended. Due to higher expectations for high-definition television (HDTV), packet loss should only occur every four hours. Since an IP packet includes seven 188-byte video packets, a loss always produces a noticeable artifact. To achieve packet loss specifications, the converged IP network must also minimize congestion to guarantee that packet loss rates are less than 10^{-6} for SDTV and less than 5×10^{-8} for HDTV. The required bandwidths vary from about 2 Mbps to 18 Mbps.

As the buffer size within a set-top box increases, so does the channel change time; a reasonable compromise must be made.

Packet delays and jitter are effectively eliminated with adequate de-jitter buffers in the set-top box (STB). A commercial STB is designed to buffer some 100 ms of video data, making these two parameters a non-issue for satisfactory video QoE. But since buffer size increases are directly related with higher channel change latencies (zapping times), a reasonable compromise has to be made. The recommended values in DSL Forum's TR-126 are:

Jitter	<50 ms
Delay	<200 ms
Zapping times	<2 seconds

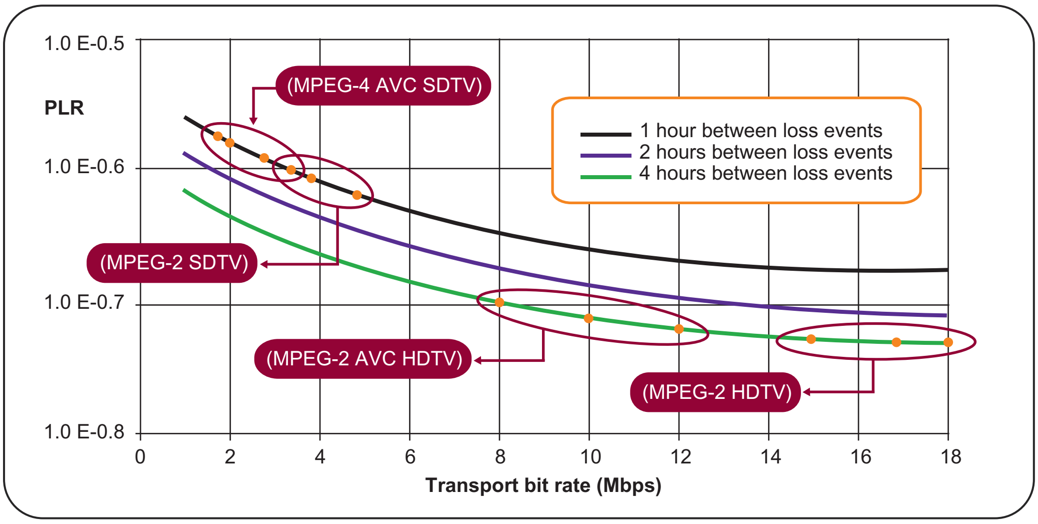

Figure 3: Isolated packet loss rates for satisfactory QoE for entertainment video services

For quality voice services, the choice of an adequate codec is also an important factor.

3.2. Voice services requirements

There are many impacts on QoE for VoIP services, but the most essential factors are packet delay and jitter, packet loss, echoes, and speech codec intrinsic impairments. Depending on the utilized speech codec and compression, the required bandwidth is up to approximately 100 kbps. Note that this is low compared to video services. Apart from providing sufficient performance of the converged IP network, the choice of an adequate codec must be carefully considered to obtain satisfactory QoE for VoIP services.

Standards bodies have investigated the QoS requirements relating to VoIP. The International Telecommunication Union (ITU) measured and published the so-called mean opinion score (MOS) in recommendation ITU-T P.800.1. This work showed that a satisfactory QoE for VoIP can be obtained when the network operates within QoS limits of:

Voice services are quite resilient to packet losses but put stringent requirements on delays and jitter.

Jitter	<50 ms
Delay	<100 ms
(or delay including jitter	<150 ms)
Packet loss	$<10^{-2}$

As can be seen, acceptable packet loss for VoIP services is many orders of magnitude higher than for video services, making it nearly a negligible parameter in converged IP multiplay networks.

It should be noted that the QoS performance targets listed above are valid for an end-to-end voice call. Since calls normally traverse many IP networks, the QoS target values must not be consumed by a single network but have to be distributed across all participating networks.

3.3. Best-effort Internet services requirements

Internet services can be given lower priorities but need to be considered for traffic shaping.

For Web browsing or data transfer applications, work has been done by the ITU to define applicable QoE parameters in terms of delay and error tolerance. From the user point of view, the main performance factor is the time required to display a requested page. In ITU-T recommendation G.1010, a variety of data applications are separately considered. According to this report, delays of several seconds are acceptable, but should not exceed approximately 10 seconds. As summarized in the DSL Forum's TR-126, best-effort Internet Web browsing services should preferably have response times of less than two seconds per page, but no more than four seconds. This translates into data quantities in the range of up to 100 KB, but typically about 10 KB. These data ranges correspond to IP data bit streams of up to 50 kbps or typically 5 kbps. Since these are easily met network performance parameters in comparison to entertainment video or even VoIP requirements, the delivery of a satisfactory QoE for best-effort Internet services is not problematic and requires only minimal attention. Nevertheless, total Internet traffic may add up to a significant amount, and needs to be regarded carefully as part of an overall traffic-shaping scheme.

3.4. Interactive gaming requirements

Demanding gaming applications require short delays and little jitter.

Requirements for interactive gaming are obviously very dependent on the specific game. It is clear that these applications, like other demanding interactive applications, require very short delays in the order of a fraction of a second. As outlined in TR-126, there are different requirements depending on the game category. The most demanding requirements originate from the so-called first-person shooter (FPS) games, and deal with response time and jitter. Data must be distributed among affected players as quickly as possible, requiring a low network delay. To achieve a highly satisfactory QoE for these types of gaming services, according to TR-126, the end-to-end network should at least fulfill the following QoS parameters:

Jitter	<10 ms
Response time	<50 ms

The specified response time is a one-way delay, and therefore results in a maximum allowed round trip time of approximately 100 ms for a gaming action. The system response time is measured from the perspective of the user and includes application layer (game server and game client) and network layer delays. Consequently, the delays introduced within the converged IP network should remain well below the limit of 50 ms.

4. Multiplay infrastructure solutions for assured QoE

Multiplay offerings require high reliability combined with service flexibility.

Within a multiplay IP infrastructure, assured QoE requires highly reliable access, aggregation, and core network designs that enable service flexibility by means of a simplified service creation and management strategy. High reliability is one of the most crucial requirements when it comes to delivering premium IPTV, real-time video, and voice services to the end user. Therefore, the basic principles for all ResIP designs include "no single point of failure" architectures and sub-second restoration times. This includes the Ethernet-based access and aggregation network and the mostly IP/MPLS-based core network as well as duplicated interconnections of content-delivery equipment. To maximize profitability and to reduce cost, multiplay service delivery architectures must combine access, aggregation, edge, and subscriber service delivery into integrated multiplay infrastructure solutions.

4.1. Reliable access, aggregation, and core network design

To cope with peak rate traffic, a flexible service deployment system helps balance the traffic and provide sufficient bandwidth to each service.

Appropriate measures are necessary to prevent the access network from becoming the bottleneck within an end-to-end QoS-enabled multiplay infrastructure. In addition to the allocation of sufficient bandwidth within the access network, flexible traffic management must be employed to guarantee the required QoS for multiplay services. To cost-effectively provide both the required bandwidth and traffic management, a flexible service deployment system (session resource controller) enhances the broadband access network performance to cope with peak traffic rates. Simultaneously, the service deployment system must ensure that bursty applications do not seriously impact the QoE of concurrent multiplay services by maintaining the required quality levels with respect to packet loss, delay, and jitter.

As an example of an effective design, Figure 4 shows the network setup in the ResIP PoC Lab. This network, like a service provider network, must provide adequate end-to-end performance for simultaneous multiplay services such as best-effort data, VoIP, real-time video, and IPTV.

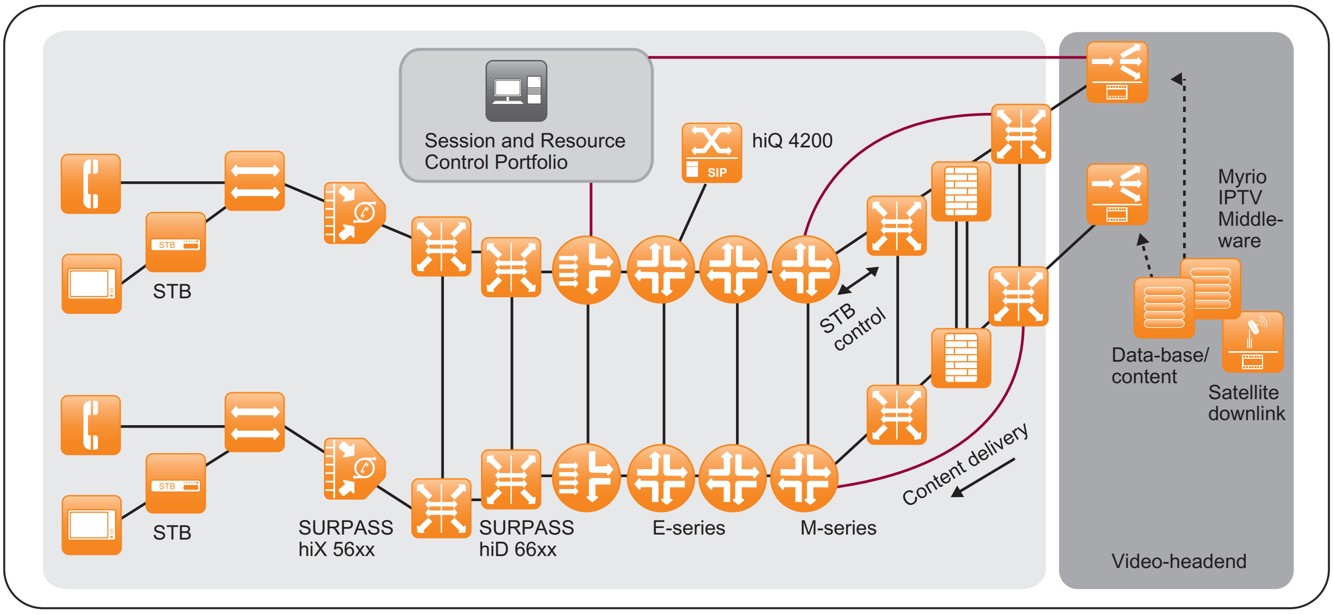

Figure 4: Basic network setup in the ResIP PoC Lab used to investigate multiplay service delivery behaviors

To map subscriber traffic to prioritized traffic groups, the SURPASS hiX and SURPASS hiD access nodes support traffic prioritization through multifield classification, policing, and shaping, and user-defined bandwidth schemes. The prioritization is based on multiple classification criteria derived from Layer 1 (e.g., xDSL-Port), Layer 2, Layer 3, and Layer 4 information.

Hierarchical scheduling for QoS improves the maintainability of the access network.

Hierarchical scheduling for QoS can also improve the maintainability of the access network. This feature allows an assignment of priorities to applications for individual end users, either based on the IP traffic type or Dynamic Host Control Protocol (DHCP) / Point-to-Point Protocol over Ethernet (PPPoE) session. Hierarchical scheduling for QoS also allows differentiation between user types such as residential, business, or premium users. It can be applied for strict priority queuing to give preference to low latency traffic. As can be verified in the ResIP PoC Lab, the Juniper Networks E-series routers support all the features necessary to handle these traffic prioritization schemes and guarantee SLAs at the network edge.

The different traffic types are to be mapped to respective scheduling/QoS mechanisms within the core routers to master competing QoS requirements.

To deliver a satisfactory QoE to the end user, the IP core network must also be classified and prioritized. Due to scalability issues, the prioritization in the IP core cannot be done on a per-subscriber basis. Instead, aggregate classification is carried out with respect to the various traffic types traveling across the core. To master the sometimes competing QoS requirements of multiplay services, the network must be able to differentiate the various traffic flows including network control, voice bearer, voice signaling, video bearer, video signaling, network management, premium data, and best-effort data. Therefore, the different traffic types are mapped to respective scheduling and QoS mechanisms within the core routers. Juniper Networks M-series (and T-series) core routers provide a rich set of features to easily support differentiated service classes for IP and MPLS traffic, as can be verified in the ResIP PoC Lab. In general, a set of primitives is applied to different protocol families to implement superior QoS capabilities. These measures prepare the core routers for traffic policing, dropping priorities, queuing, and scheduling mechanisms. Combined with the industry-leading wire-speed forwarding engine of the Juniper Networks routers, these QoS capabilities are crucial for delivering excellent QoE to end users.

4.2. Session and resource control

Offering multiplay services requires finding answers to two questions: How can the service quality and application experience be assured, and how do the different types of traffic need to be treated? The delivery of high-bandwidth and real-time multiplay services over converged IP networks introduces significant challenges, since service providers must cope with millions of subscribers, resist demands at peak-times, and maintain superior QoE. While overcoming these challenges, service providers must also limit capital expenditures to remain profitable. Consequently, network resources must be dynamically allocated and balanced to achieve optimized service delivery.

To meet these challenges, ResIP Certified IP infrastructure solutions include Juniper Networks SDX-300 Service Deployment System, which is to enable highly flexible session and resource control. The SDX-300 is part of the Session and Resource Control portfolio and provides subscriber and policy management. Subscriber and policy management capabilities ensure a superior user experience – even when a service is oversubscribed – by taking the following measures:

- Define bandwidth requirements per multiplay service (guaranteed bandwidth)
- Establish accounts for network resource congestion points like physical ports, virtual LANs (VLANs), or Label-Switched Paths (LSPs)
- Gracefully reject a requested multiplay service if the necessary capacity is not available (like a “busy signal” in traditional telephony)

Available network resources need to be easily managed and balanced with respect to optimized service deliveries through session and resource control facilities.

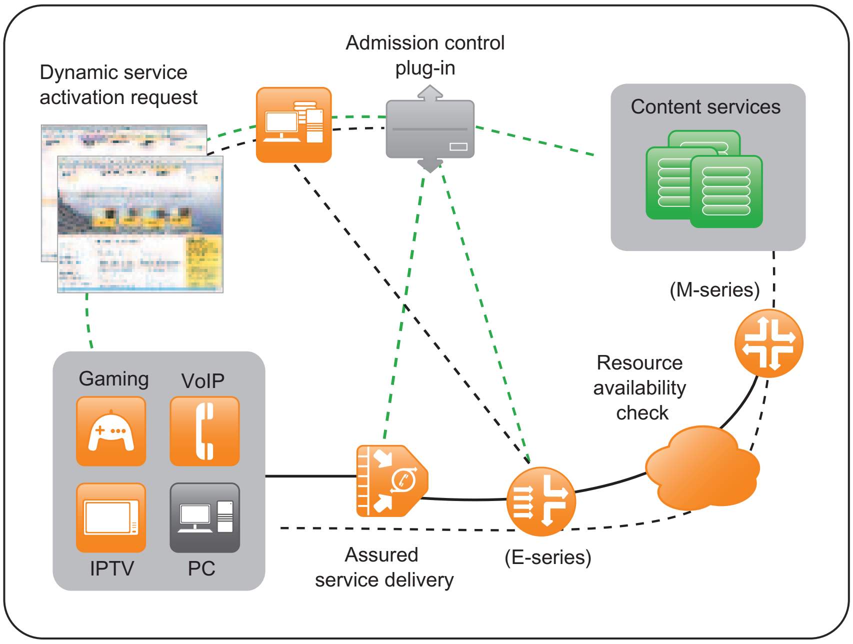

Figure 5: Principle interoperation between SDX-300 and service delivery network

A service request is handled by admission control and resource availability checks.

Figure 5 shows the principal method for SDX-300 interoperation within a service delivery network. Facilities for admission control and resource availability checking dynamically handle a service activation request from a subscriber. The facilities interoperate with access links, network routers, and content servers.

Only if a service request can be approved will it be activated.

Consider an example case of a VoD service that requires a certain bandwidth with sufficient QoS to provide satisfactory QoE. Therefore, before video streaming is started, the VoD server must request that the SDX-300 provision the bandwidth and QoS parameters required for the encoded video. Only if this request can be approved does the SDX-300 activate the VoD service from the content server. Otherwise the request is gracefully rejected and the customer can try again at a later time.

4.3. Securing multiplay services

It is business-critical for carriers and service providers to deploy sufficient counter-measures to protect multiplay service offerings against attacks.

The frequency of security-related press releases reveals that Internet-based attacks are increasing rapidly. The required skills for perpetrating such attacks are very low; numerous malicious scripts are available for downloading. Therefore, it is business critical for carriers and service providers to deploy sufficient counter-measures to protect multiplay service offerings. Subscribers with direct or indirect access to the converged IP network must be denied the ability to sabotage network resources or interrupt service.

To mitigate attacks, a variety of security mechanisms such as packet filtering, deep packet inspection, content security, content encryption, and user authentication must be employed. For example, Juniper Networks J-, E-, M-, and T-Series routers include highly scalable filtering capabilities, unicast reverse path forwarding, and high performance rate limiting facilities to protect against denial of service (DoS) attacks. It is recommended that these mechanisms be enabled in conjunction with filter policies to protect against DoS attacks without compromising the forwarding performance. Furthermore, operators can protect the router's control plane by applying filters to restrict local packets traversing from a physical port to the routing engine.

It is crucial to activate security facilities within the routers (and Ethernet switches) of an end-to-end transmission path, but additional external security equipment is especially important for protecting the Web servers that are content delivery points. For example, at the video head-end and server domain it is best to treat separately the media traffic, signaling traffic, control traffic, and Operation, Administration, and Maintenance (OAM) traffic. For better handling, the traffic types should also be separated into dedicated security domains. This separation into system internal traffic and traffic to untrusted destinations like access networks and the Internet can be performed by router access lists, load balancers, and firewalls.

Traffic should be separated into dedicated security domains.

To demonstrate network security issues in the multiplay solutions with respect to high availability, the Juniper Networks Integrated Security Gateway ISG2000 has been redundantly employed to protect VoD content servers by applying deep packet inspection measures. If an attack is detected, it will take corrective action by shutting down the attack source. The downstream traffic (from the VoD server to the customer) is not monitored, and therefore the quality of the video is not impacted.

Deep packet inspection is recommended to protect the content servers.

The ISG2000 is an ASIC-based platform that handles high-bandwidth traffic with minimal latency, jitter, and packet loss. Therefore, it is ideally suited to effectively secure carrier and data center infrastructures where multiplay applications such as VoIP, VoD, and other services require consistent and scalable performance when it comes to deep packet inspection firewall, virtual private network (VPN), and DoS solutions.

4.4. Scalability and reliability

The ultimate success of a multiplay infrastructure implementation is based on its scalability, reliability, and performance as a carrier-grade environment. However, most equipment vendors only test and publish the scalability and performance capabilities of their products as they relate to standard compliance assessment. Considering the complexity of the protocols involved, scalability and performance issues are often genuine challenges, making scalability one of the biggest obstacles to overcome for service provider networks. To remain competitive, carriers and service providers must understand the dynamics of network growth as new customers are added, as well as the ultimate performance limits of the networks. Without considering these behaviors, network congestion will likely occur and result in poor QoE for the end users. Therefore, these issues directly impact customer satisfaction and churn rates.

Carriers and service providers must understand the dynamics of growth in their networks.

Taking scalability and performance issues seriously can be a competitive differentiator for equipment vendors targeting carriers and service providers that plan to offer multiplay services. Nokia Siemens Networks and Juniper Networks are perfectly aware of this. To address scalability and reliability concerns properly, the design target for all ResIP Certified infrastructure solutions dictates that the quality of a service is to be consistently maintained from small to very large infrastructure deployments. The verification work comprises even the most adverse network conditions to determine the performance limits with reference to the respective service requirement outlined in Section 3 of this paper.

ResIP Certified solutions are always verified for small to very large infrastructure deployments.

5. Lab findings

Engineering rules are available for the configuration of carrier-grade IP networks.

In the joint Nokia Siemens Networks and Juniper Networks ResIP PoC Lab in Munich, manifold investigation work has been performed within the ResIP program. For a variety of multiplay services such as voice, video, IPTV, best-effort Internet, and gaming, the network behavior was scrutinized. The results obtained from this ongoing comprehensive verification work is continuously translated into engineering rules that help providers and carriers optimally configure carrier-grade IP networks.

Live and remote demonstrations are available to showcase the behavior of multiplay services delivered over a ResIP Certified I network.

To confirm the IP expertise of both companies and to share the plurality of positive results with customers, live demonstrations at the ResIP PoC Lab are routinely presented. Each demonstration includes multiple scenarios for IP- or MPLS-based core networks, Layer 3 VPNs, access, SDX-300, and security features to display the related effects of network interferences, interruptions, and attacks on the behavior of multiplay services. In addition, a subset of showcases has been configured into a remote demonstration. This demonstration can be initiated from anywhere in the world to let remote customers take advantage of the ResIP PoC Lab resources without traveling to Germany. In terms of the QoS parameters, the ResIP PoC Lab results are summarized in Table 1, together with the values recommended by DSL Forum's TR-126 to provide for a satisfactory end-user QoE related to VoIP, entertainment video services, best-effort Internet, and gaming. It is quite evident that the results for delay, jitter, and packet loss within the ResIP PoC Lab by far surpass the TR-126-based requirements, confirming that the joint Nokia Siemens Networks and Juniper Networks IP infrastructure solutions are carrier-grade, and the first choice for successfully delivering multiplay services.

	Delay	Jitter	Packet loss	Others
Voip	100 ms	50 ms	$< 10^{-2}$	
Entertainment video services	200 ms	50 ms	$< 10^{-6}$ SD $< 5x10^{-8}$ HD	Zapping < 2 s
Best-effort Internet	< 2 s/page			
Gaming	50 ms	10 ms		
PoC Lab results	**< 1 ms***	**< 0.5 ms***	**None****	**Recovery < 1 s** **Zapping < 1.5 s+**

Table 1: Summary of required QoS parameters from DSL Forum's TR-126 and the related values measures in the ResIP PoC Lab

* The ResIP PoC Lab results do not include transmission delays (about 5 ms/1,000 km) and additional jitters that are introduced on long-distance interconnection links and are valid for a small number of network nodes (<10).

** Packet losses were not observed in the ResIP PoC Lab for link loads with up to 99 % of their capacities.

+ Time observed from pressing the remote control button to a stable video image of the selected channel (using state-of-the-art video codecs).

6. Conclusions

End-to-end IP multiplay infrastructure solutions have been certified in the ResIP PoC Lab. They are based on technologies drawn from the Nokia Siemens Networks SURPASS Home Entertainment, SURPASS Hosted Office, and SURPASS Carrier Ethernet solutions, and the Juniper Networks E-, J-, and M-series routing platforms, Service Deployment System SDX-300, and ISG2000 security equipment. By means of SDX-300, the solutions reduce operating expenses (OPEX) through state-of-the-art subscriber management and provisioning. As a result, the ResIP Certified multiplay solutions leverage the combined expertise to demonstrate that converged multiplay service bundles can both scale cost-efficiently and provide the required QoE to end users.

Engineering rules and design guidelines are available for service providers who want to take advantage of the ResIP engineering excellence.

User expectations for innovative multiplay services delivered over a converged IP network stem from experiences with today's dedicated transmission networks. Uninterrupted services are taken for granted and QoE should not be compromised, even when a failure occurs in the network. Entertainment video services put stringent demands on bandwidth and packet loss, whereas voice and gaming services are more sensitive to delays and jitter. Therefore, a ResIP Certified multiplay solution is designed to differentiate between various traffic types, and manages each according to its specific end-to-end requirements across multiple links and network elements. Additionally, a ResIP Certified multiplay solution addresses critical aspects of network security using common security policies across the entire network to protect service bundles from network-based attacks.

Nokia Siemens Networks offers flexible support solutions for the effective and fast integration of new products and applications in today's complex and heterogeneous multi-vendor networks. The Nokia Siemens Networks approach is based on an in-depth understanding of customer needs, priorities, and requirements, thus enabling the development of optimized solutions. Nokia Siemens Networks analyzes each customer's business requirements and customizes products and applications to the specific needs. Prior to implementing any solution, Nokia Siemens Networks performs comprehensive tests such as performance and conformance testing, interoperability checks, and technical verification in a reference system to ensure proven end-to-end functionality. Integration projects carried out by Nokia Siemens Networks result in shorter times to revenue generation, while keeping expenditures to a minimum by delivering the right quality, at the right cost, and at the right time. Should a customer wish to see how a Nokia Siemens Networks IP infrastructure solution would meet their demands, a trial on the customer's site or at one of the Nokia Siemens Networks Integration Laboratories can be carried out.

Nokia Siemens Networks supports customers for fast and risk-free integration of new services on converged IP networks.

For more information about Nokia Siemens Networks IP Connectivity solutions, visit www.resip.net.

For more information about Juniper Networks solutions, visit www.juniper.net.

Learn more!

7. About Nokia Siemens Networks and Juniper Networks

Nokia Siemens Networks is:

- A Juniper Networks Authorized Education Center to enable operators' engineers;
- An official partner in the Juniper Networks Content and Applications Alliance, to develop revenue generating solutions for operators;
- The first Juniper partner to have achieved the status of a Juniper Networks Authorized Global Support Provider, which offers the full range of support services from installation to optimization worldwide.

The solution-based expertise and experience that Nokia Siemens Networks has gained in worldwide installations of voice and packet networks results from a range of unrivalled offerings. Nokia Siemens Networks is a strong and reliable partner that maintains a stable relationship for conducting successful business in the highly competitive carrier market. The company's experience and know-how can also be demonstrated with regard to its customer base, strategic partners, and the worldwide availability of its large pool of certified IP experts. A globally present and strong service organization supports carriers 24 hours a day, seven days a week.

For many years, Nokia Siemens Networks and Juniper Networks have been helping customers build the largest, most reliable, and most profitable IP networks in the world. This blend of world-class installations experience, combined with global presence and expertise, results in carrier-grade converged IP infrastructure solutions from Nokia Siemens Networks and Juniper Networks.

Recently, Nokia Siemens Networks and Juniper Networks have collaborated to develop end-to-end QoS-enabled IP infrastructures that help guarantee end users an excellent QoE for multiplay service bundles. These ResIP Certified multiplay solutions put carriers and service providers in a position for immediate deployment, since stringent testing and verification work has already been done.

Nokia Siemens Networks and Juniper Networks are working together to solve the toughest challenges faced by service providers. The ResIP program has been established to help service providers through the tough times of dwindling voice revenues and declining customer bases. It includes well-defined steps to develop, test, and certify all technology for the next-generation architecture and solutions. The resulting benefits for carriers and service providers are fully tested and complete end-to-end IP network infrastructures that allow multiple certified solutions to be deployed for new revenue opportunities.

- Nokia Siemens Networks is one of the largest global players in the telecommunications industry for converged mobile and fixed network technologies. It is the only provider in the market that offers a full-range portfolio covering equipment for end users to complex network infrastructures for carriers and service providers as well as related services.

- Juniper Networks is the leader in enabling secure and assured communications over a one-shop IP network. The company's purpose-built, high-performance IP platforms enable customers to support many different services and applications at scale. Service providers, enterprises, governments, and research and education institutions worldwide rely on Juniper Networks to deliver products for building networks that are tailored to the specific needs of their users, services, and applications. Juniper Networks' portfolio of proven networking and security solutions supports the complex scale, security, and performance requirements of the world's most demanding networks.

8. Abbreviations

BSR	Broadband Service Router
DHCP	Dynamic Host Control Protocol
DoS	Denial of Service
DSLAM	Digital Subscriber Line Access Multiplexer
FPS	First-Person Shooter
HDTV	High-Definition Television
IP	Internet Protocol
IPTV	Internet Protocol Television
IS	Intermediate System
LSPs	Label-Switched Paths
MOS	Mean Opinion Score
MPLS	Multi-Protocol Label Switching
OAM	Operation, Administration, Maintenance
OPEX	Operating Expenses
PLR	Packet Loss Rate
PoC	Proof of Concept
PPPoE	Point-to-Point Protocol over Ethernet
QoE	Quality of Experience
QoS	Quality of Service
ResIP	Resilient IP
SDTV	Standard-Definition TV
SDX	Service Deployment System
SLA	Service-Level Agreement
STB	Set-Top Box
TDM	Time Division Multiplex
TR	Technical Report
VLAN	Virtual LAN
VoD	Video-on-Demand
VoIP	Voice over IP
VPN	Virtual Private Network
xDSL	All types of Digital Subscriber Lines

Nokia Siemens Networks Corporation
P.O. Box 1
02022 Nokia Siemens Networks
Finland

Visiting address:
Karaportti 3, Espoo, Finland

Switchboard +358-71-400-4000 (Finland)
Switchboard +49-89-515-901 (Germany)

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www.nokiasiemensnetworks.com