APPLICATION NOTE

Voice Over IP Carrier Transport: Assuring Voice Service Quality over Converged Networks

ALCATEL

Introduction

VoIP is out there — Competitive local exchange carriers (CLECs) are already using voice over IP (VoIP) technology to offer end users more cost-effective voice services across the carrier's data infrastructure. These services are typically delivered to customers through a broadband access network, such as a digital subscriber line (DSL) access network. VoIP technology is clearly the foundation for new multimedia communication services that will address mobility and cost-reducing convergence challenges, both of which are on the wish lists of consumers and business customers. However, VoIP can also optimize the operators' PSTN network costs when used to transport bulk voice traffic over a shared IP data network.

Voice revenues are at stake. Many incumbent operators are seeking to protect voice revenues in the face of extreme pricing pressures. To avoid the price war, they need to offer new, more attractive, differentiating communication packages for both consumers and business customers. VoIP-based services are a key component of those offerings. Typical applications include IP Centrex, video telephony, video conferencing and "push-to-talk."

Incumbent operators are also looking at ways to further control the costs imposed by voice interconnect, for example, (which is being replaced by long-distance bypass over IP networks) in order to guard their margins in this saturated and highly price-competitive market.

IP networks are out there — Operators also need to remain competitive in the market for fixed-line services. In the data arena, leased line and frame relay services in particular, are being challenged by more flexible, cost-effective and scalable IP virtual private networks (VPNs) and Ethernet VPNs. This competitive landscape is forcing operators to look at a new infrastructure that will help them retain their existing customer base while growing their market share by adding VPN-based data services to their portfolios. This infrastructure will be an IP network that supports different types of VPNs and enables network interworking (and even service interworking) functions to leverage the existing infrastructure.

Current IP networks, however, do not handle VoIP-based services well. Building a data network architecture to support mass market VoIP-based services or broader multimedia services involves adding extra functionality to solve a number of issues. These issues are inherent in voice service deployment and evident in voice networks, but no longer obvious in data networks:

> Service performance and availability assurance: Guarantee service quality.

> Service scaling: Ensure that the network can handle the millions of consumers now on the PSTN with no delayed or denied call setups.

> Service control and security: Ensure that only those customers who pay for a premium service have access to that service.

> Service maintenance, provisioning and troubleshooting: Create service-awareness in the data network by enabling service operations, administration and maintenance (OA&M).

Thus, besides data VPNs, VoIP carrier transport represents another demanding application for the data network. This paper will discuss the requirements for this application, and explain how Alcatel addresses them.

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VoIP-Based Services: Why (Now)?

What has happened in recent years in the deployment of voice over IP (VoIP)-based services? Many vendors have brought products that support VoIP to the market — softswitches to accommodate call setup, voice gateways to interface between the VoIP network and the PSTN, IP phone sets and IP private branch exchanges (PBXs), among others, are readily available.

A number of companies around the globe have started offering VoIP services — Skype (22 million downloaded clients for free VoIP service between PCs or paid accounts to make VoIP calls to the PSTN), Yahoo! BB Japan (4 million VoIP subscribers) and Vonage (200,000 VoIP lines). These operators are pursuing a niche market of innovative consumers and business customers.

As indicated in the quotes below from *Converge! Network Digest*, incumbent operators acknowledge the trend towards VoIP-based services and are preparing their networks to gradually evolve to an architecture that should guide them through the next decade.

▲ *"… MCI plans to migrate its US-based international gateway traffic from traditional circuit switching to VoIP. MCI expects to begin transitioning traffic by mid 2005…"*

CONVERGE! NETWORK DIGEST, AUGUST 2004.

"… Wyoming ILEC to Launch VoIP … The move follows the company's decision to deploy … [VoIP] to serve its existing 8,000 subscribers. TCT West will initially be deploying services including IP Centrex, both to meet customer demand in-region and to expand into other markets. The carrier also plans to take advantage of the increasing availability of SIP interfaces from incumbent and alternative interexchange carriers (IXCs) to reduce long distance trunking costs, as well as leveraging the flexibility of SIP-based application servers. In addition to significant cost savings, TCT West expects the adoption of VoIP to double its subscriber base to 16,000 over the next few years."

CONVERGE! NETWORK DIGEST, AUGUST 2004.

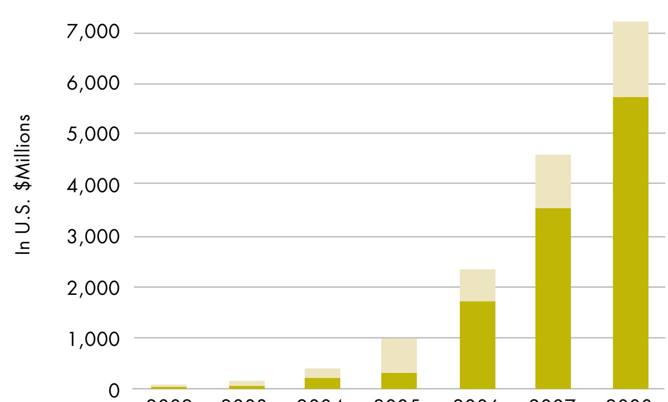
"…While less than 1% of fixed lines worldwide currently rely on softswitch-controlled VoP technologies, Probe predicts that by the end of 2008 nearly 27% of the world's fixed lines will have been converted to packet…" ▶

LINE-SIDE VOIP – WHAT IS DRIVING DEMAND, PROBE GROUP LLC, FEBRUARY 2004.

Why should operators consider deploying VoIP-based services? As Figure 1 and the quote from Probe indicate, predictable growth is occurring, mainly due to market momentum. This momentum has several causes:

- Broadband access to serve both smaller office locations and residential users is growing to a point where VoIP services are reaching a profitable deployment scale.
- Service providers are looking to develop new, more competitive approaches to communications services targeted at enterprise and residential customers through service bundles.
- From a network perspective, manufacturers are discontinuing certain PSTN switch types and capacity has been exhausted in places where money must be spent just to maintain current service levels. The new equipment alternatives are VoIP-based.
- Using IP-VPNs, service providers can extend the IP connectivity offering with voice services, answering convergence demand from the enterprise.
- The business package designed as a PBX replacement includes many PBX features such as automatic call distribution, interactive voice response (IVR), auto attendant and management tools such as an operator console and web management for end users.
- Pressure from mobile voice offerings is accelerating fixed voice substitution.

Figure 1 - Enterprise IP Voice Services in Europe 2002-2008



	2002	2003	2004	2005	2006	2007	2008
VoIP VPN Services	14	52	125	268	586	1,028	1,450
Hosted and Managed IP Voice Services	22	81	236	681	1,730	3,553	5,739

Source: Probe Group LCC

Capitalizing on the Momentum with VoIP Carrier Transport

Service providers can capitalize on the market momentum by evolving to a packet-based infrastructure to offer voice-based services. Doing so, however, introduces a number of challenges that Alcatel is equipped to address:

- Maintaining the carrier grade qualities of traditional voice networks by implementing a service-aware data infrastructure to transport and operate VoIP-based services, leveraging existing asynchronous transfer mode (ATM)/frame relay or synchronous optical network (SONET)/synchronous digital hierarchy (SDH) networks where possible
- Safeguarding the investment by re-using this new, single infrastructure for voice services for both residential and business customers, and for data services like IP-VPNs or Ethernet VPNs
- Building the necessary service intelligence in the network to control service availability and protect revenue streams

"...How voice, data and video traffic streams are scaled to provide efficiency in the loop while meeting service-specific quality of service (QoS) requirements will play an important role in the evolution to softswitch-controlled networks..."

LINE-SIDE VOIP – WHAT IS DRIVING DEMAND, PROBE GROUP LLC, FEBRUARY 2004.

Consolidating Data and Voice on a Converged Infrastructure

VoIP carrier transport addresses the VoIP transport requirements of service providers and enables them to deploy multimedia-based applications across a converged data (IP) network. This network will enable service providers to support applications such as IP Centrex, contact centers, multimedia conferencing and unified messaging, while capitalizing on the optimized cost model of a converged infrastructure.

A converged network by definition supports multiple services with multiple levels of service quality, which requires service isolation and differentiation. For voice, this means that characteristics such as service availability (always-on conversation) and service quality (clear, continuous voice, because of low packet delay variation) must be addressed in the network to transport voice.

By adding QoS, reliability, availability and OA&M features to the data network, service providers build the required foundation to offer multimedia services that have the same perception of quality as traditional voice service. A carrier-grade network translates into increased customer satisfaction for VoIP-based services and reduced customer churn.

DEFINING CARRIER GRADE

For many years, carriers have provided voice services and leased line services. The quality requirements for these services are so stringent that they have become a reference for the industry. In this case, quality is distinguished by these characteristics:

Reliability: Provides the necessary features and implements the necessary redundancy to guarantee 99.999 percent availability and predictable, guaranteed performance

Scalability: Provides an architecture that allows a solution to scale in performance and interface density, by gradually growing the infrastructure

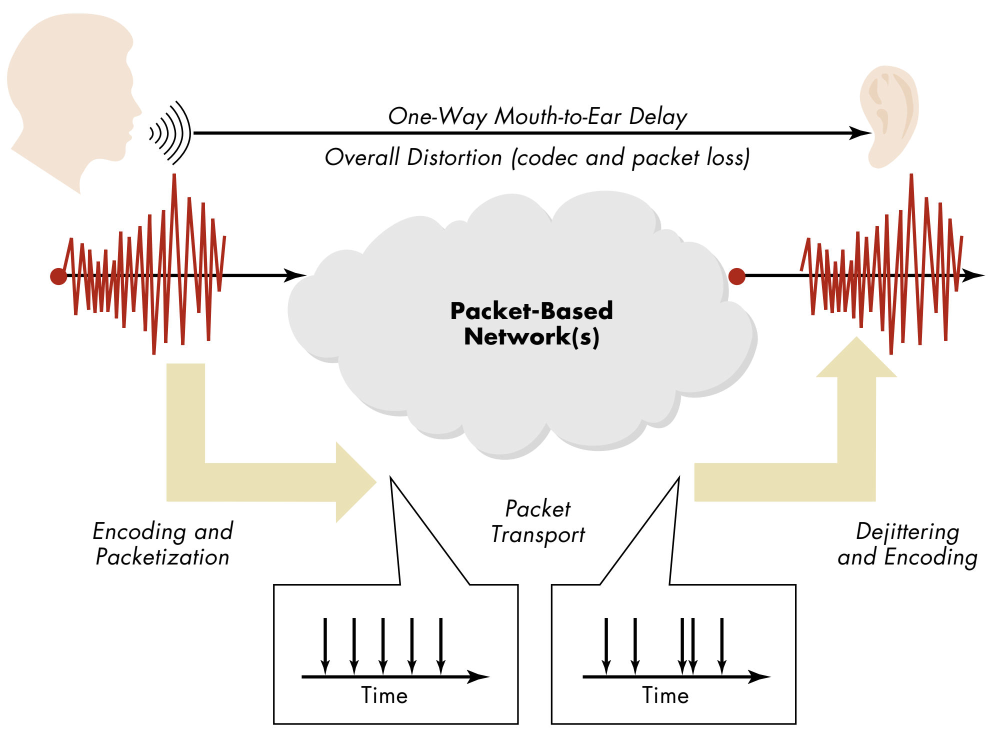
Maintainability: Provides a management environment that assures accurate service levels and includes the necessary OA&M tools to assure the service

Requirements for toll quality VoIP

For traditional (wire-bound) PSTN calls, which do not suffer from distortion, the key factor that determines the quality is the mouth-to-ear delay, defined as the delay incurred from the moment the speaker utters the words until the instant the listener hears them. In the case of packetized voice calls, distortion may be introduced by the codec that compresses the voice signal or by the loss of voice packets. Controlling both the mouth-to-ear delay and distortion is the key to offering high-quality packetized voice calls.

Figure 2 illustrates the stages of packetized phone call transport. The delay for packetized voice calls, where the most important functions are encoding, packetization, propagation, dejittering and decoding delay, is longer than for a traditional circuit-switched voice call, where the mouth-to-ear delay is mainly made up of the propagation delay and switching delay. Furthermore, mouth-to-ear delays can differ considerably from one direction to the other, which almost never occurs in the PSTN.

Figure 2 - Three Stages of Packetized Phone Call Transport



The sidebar titled *E-Model for Rating Voice Quality* explains how packet loss and delay affect the perception of voice quality. However, choices such as the type of codec and echo cancellation also impact voice quality.

To summarize, when delivering toll quality VoIP-based voice service, providers need to control packet delay to a maximum of 150 ms, and minimize packet loss to no more than five percent. These requirements are addressed by the Alcatel solution for VoIP carrier transport.

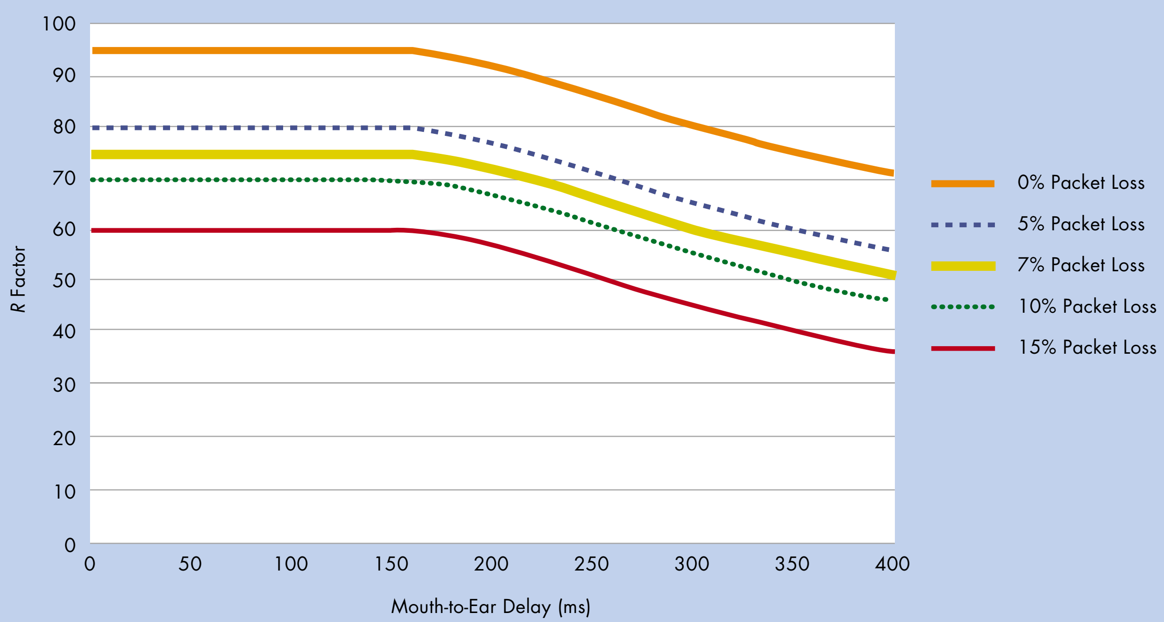
E-MODEL FOR RATING VOICE QUALITY

The E-model (http://portal.etsi.org/stq/presentations/emodel.pdf) is a tool for predicting how an "average user" would rate the voice quality of a phone call with known characterizing transmission parameters. Based on an extensive set of subjective experiments, the E-model's developers defined an additive rating scale R that assesses the quality of a phone call (see figure to the right) by quantifying the various transmission impairments like noise effects, delay effects. Based on the R rating, we can predict subjective user reactions, such as what mean opinion score (MOS) a judging panel would award the call.

R Value Range	90-100	80-90	70-80	60-70	0-60
Speech Transmission Quality Category	Best	High	Medium	Low	(Very) Poor
	PSTN Quality				

The figure below shows the quality rating R, defined in the E-model, as a function of the mouth-to-ear delay for different values of packet loss. It shows that a delay of 150 milliseconds is a general guideline.

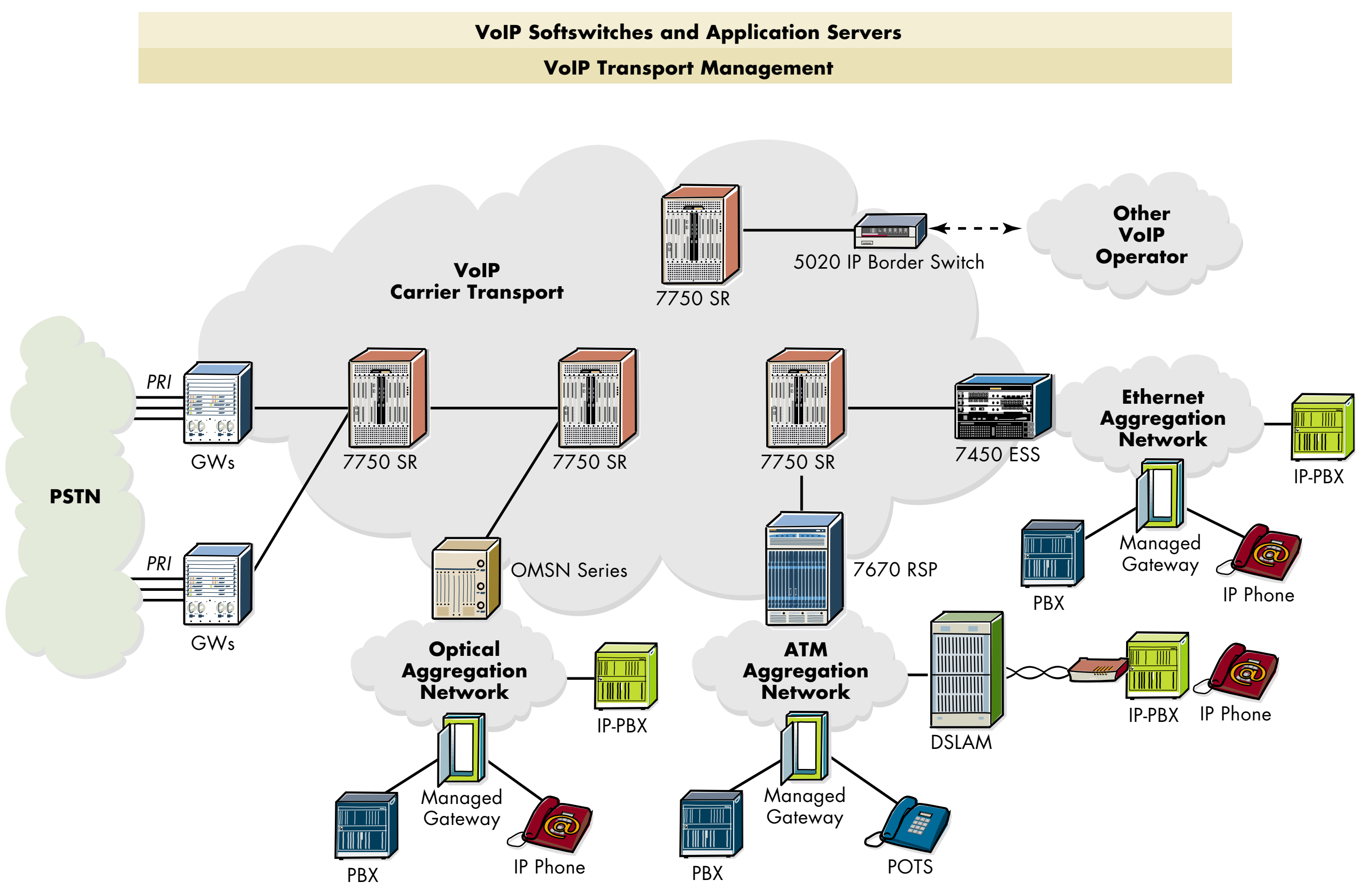
For more details on the E-model, please refer to the Alcatel Telecommunications Review - 1st Quarter 2000, "Quality bounds for packetized voice transport."



The Alcatel Solution for VoIP Carrier Transport

Alcatel's VoIP carrier transport solution is centred around a multiprotocol label switching (MPLS)-based IP network, enhanced with a number of interworking functions to provide end-to-end VoIP transport across different aggregation networks. Figure 3 shows the different Alcatel components involved in this solution. The core of the solution is the MPLS-based IP network, which is built with the Alcatel 7750 Service Router (SR). It is extended with Optical Multi-Service Node (OMSN) Ethernet products, the Alcatel 7670 Routing Switch Platform (RSP) and the Alcatel 7450 Ethernet Service Switch (ESS) to aggregate the voice service from optical, ATM and Ethernet networks respectively. The Alcatel 5020 IP Border Node (IPBN) delivers a secure and controllable environment to interconnect with other VoIP operators. All components are managed by the Alcatel 5620 Management portfolio and the Alcatel 1350 Management suite.

Figure 3 - Alcatel Solution for VoIP Carrier Transport



The Alcatel solution addresses the key issues for VoIP carrier transport:

> **Service performance and availability:** The Alcatel 7750 SR, Alcatel 7450 ESS and Alcatel 7670 RSP have a unique QoS implementation to accommodate the stringent delay and jitter requirements of VoIP traffic. An advanced implementation of MPLS fast reroute (sub 50ms failover) also helps to provide guaranteed end-to-end service quality and availability.

With the advanced MPLS fast reroute capability of the Alcatel 7750 SR, high availability with no single point of failure, and the MPLS-based resilient packet ring (RPR) implementation of Alcatel OMSN and packet ring switch family, service providers have the flexibility to provide different levels of service resiliency for different kinds of customers — with 99.999 percent availability.

Service providers can assign a service to a VPN that relies on IP network restoration timeframes (in the order of seconds), or to a VPN that utilizes the fast reroute capabilities of MPLS (in the order of milliseconds). With the Alcatel solution, service providers have the flexibility to attach a specific restoration scheme to a service depending on the availability requirements of that service. For example, high speed Internet can typically afford a restoration time of a few seconds, whereas voice is less tolerant of service interruption.

The Alcatel 7750 SR, for instance, enables service providers to differentiate services in the IP network using these capabilities:

- *Hierarchical QoS:* Enables QoS on a per-service basis and at a more granular level, such as per-department
- *Line rate service accounting:* Performs accounting for every service without impacting performance
- *Flexible network recovery schemes:* Allows network to recover in seconds (using IP network protocols) or milliseconds (using MPLS fast reroute)

> **Service scaling:** The transport of VoIP is not only a question of IP network scalability, where the IP routers should be able to route VoIP traffic from millions of users and thousands of business customers at wire speed without delay, but also of extending the transport to the edge of the network via Layer 2 aggregation networks with the same service quality end-to-end. The Alcatel solution supports transport extension in these ways:

- Leverage the ATM installed base for service backhaul and interwork to MPLS networks with the Alcatel 7670 RSP. The ability of the Alcatel 7670 RSP to migrate large scale multiservice networks to MPLS is unmatched in the industry. It is optimized to deliver multiple IP and Layer 2 services reliably and concurrently, including support for these services on the same interface, each routed or switched in their native mode.
- Enhance Alcatel ETSI SDH networks with data-aware, MPLS-based functionality with the OMSN product family, by supporting integrated service adapter (ISA) data-aware functionality. The Alcatel ISA technology optimally adapts the transport infrastructure to new data traffic patterns. The solution supports Ethernet, Gigabit Ethernet, ESCON, ATM and MPLS-based packet ring switching capabilities for improved data stream connectivity and IP services.
- Build an MPLS-based Ethernet aggregation network with the Alcatel 7450 ESS offering the same service guarantees as an Alcatel 7750 SR-based IP network.

> **Service control and security:** The Alcatel 5020 IPBN restricts calls through the network to those that have been paid for and assures the required QoS based on service level agreements (SLAs).

> **Service maintenance, provisioning and troubleshooting:** The Alcatel network management platforms provide per-service provisioning, monitoring and troubleshooting tools such as service ping and round-trip delay, enabling the operator to address potential problems proactively. Alcatel's IP and data-aware transmission management platforms, including the Alcatel 5620 Service-Aware Manager (SAM) and Alcatel 1354 Broadband Manager (BM), create a service-aware element and network management system that provides comprehensive fault, configuration, accounting, performance and security (FCAPS) functionality for Alcatel networks with these capabilities:

- Intelligent alarm management and correlation using per-alarm configuration actions and color-coded active alarms to eliminate duplicate reporting, and alarm logs to analyze trends

- Reduced provisioning time using point-and-click GUI configuration templates and windows for network IP/MPLS, profiles, and services configuration
- Comprehensive set of statistics counters on a per-service or per-port basis to allow operators to accurately measure usage and bill customers for services based on any combination of flat-rate, destination-based, or usage-based models
- Real-time retrieval of current or historical interface statistics or service statistics
- Ability to pinpoint security controls for operator access privileges based on individual or group account settings, and controlled access to the router

Benefits of the Alcatel Solution for VoIP Carrier Transport

The Alcatel solution provides several benefits for carriers in terms of revenue enhancement, capital expenditure (CAPEX) reduction and operating expenditure (OPEX) control, and service flexibility and availability, and addresses the key issues of carrier voice transport:

> **Enhance revenue** by creating differentiating voice services and flexible billing/pricing schemes. For example, VoIP services can be categorized as mission-critical, near mission-critical, and low-cost with different service level guarantees, service characteristics and pricing for each.

> **Reduce CAPEX** by migrating toward a single IP network and aggregation network based on MPLS, for a variety of services (e.g., IP-VPN, video broadcasting, Ethernet and existing data services), but still providing the mediation of existing ATM/frame relay or SONET/SDH networks into the MPLS network. Support for multiple service architectures on a single network architecture optimizes CAPEX, provides service flexibility and mitigates the risk of predicting service uptake.

> **Control OPEX** by providing service-aware network OA&M tools that allow operators to monitor, provision and troubleshoot a service instead of network components. Alcatel's service-oriented and application-aware OA&M tools allow per-customer, per-service administration, management and accounting. This customized management increases visibility for service providers before and after any problems arise. OA&M tools like round-trip delay can help providers proactively test their network for performance and delay requirements and correct possible network issues before they cause service degradation. In today's rapidly changing market, visibility is often a key competitive differentiator and can help enterprises and service providers plan and react quickly. This also helps service providers to increase customer satisfaction and reduce churn.

> **Guarantee service flexibility and availability** by assuring service quality and reliability, ensuring that new services can be accommodated and determining how generated revenue will be secured. The following examples illustrate how Alcatel's solution addresses lawful intercept and controls VoIP-based revenue.

Example 1: VoIP service monitoring meets regional security requirements: The United States' Communications Assistance for Law Enforcement Act (CALEA) requires service providers to be able to obtain itemized connectivity records and access actual customer traffic when legally required. To accommodate this requirement, the operator should be able to copy any number of voice streams to a voice monitoring system without jeopardizing network performance. Figure 4 shows how the Alcatel solution accommodates end-to-end lawful intercept. The Alcatel 5020 Softswitch duplicates signaling data and the Alcatel 5020 Access Border Node (ABN) duplicates media streams towards the intercept mediation device (IMD).

Alternatively, Alcatel implements a feature called service mirroring, which allows service providers to copy a customer's data or data subset to any place in the network (e.g., the NOC). Service mirroring helps service providers by:

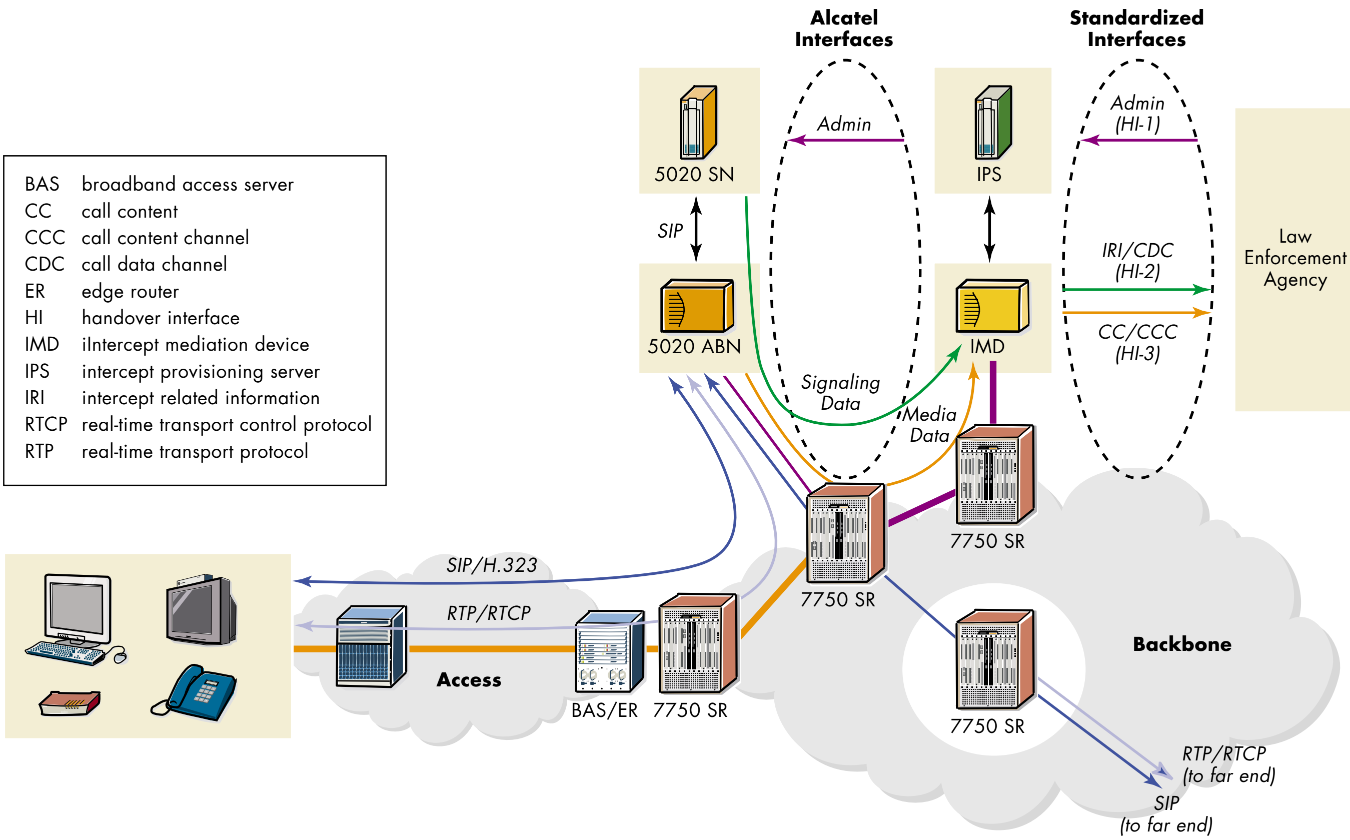
> Allowing for the remote troubleshooting of encapsulation and protocol issues

> Reducing "truck rolls" and costs

> Bringing problems, regardless of location, to the workplace of skilled personnel

Example 2: Security functions support required border control: Secure voice communication services implemented on next generation networks (NGNs) must meet strict QoS requirements. A service provider must allow authorized users into its network with the appropriate service level guarantees, while at the same time protecting its internal infrastructure from illegal uses and denial-of-service attacks. The border can be the VoIP network border between two service providers or between the service provider and its end customers or subscribers. The Alcatel 5020 IPBN is a real-time IP services gateway

Figure 4 - Lawful Intercept



that enables service providers and enterprises to deploy secure, controlled, real-time, enhanced, voice communications services on next generation IP networks. The Alcatel 5020 IPBN is designed to address the VoIP needs of large carriers, service providers and enterprises, meeting their strict requirements for QoS, network availability and security. It provides such essential services as access control, media control, usage control, signaling control, and flow duplication so as to maintain tight security between trusted and untrusted IP data and signaling domains.

Why Alcatel?

Alcatel has a rich history in offering solutions in the fixed voice services arena around the globe. This wealth of experience is a key value for delivering the next generation of voice service, based on VoIP. This leads to the following differentiators:

Global Experience

Alcatel has experience in designing and deploying networks for voice and broadband services in all major markets. As a result, we understand their unique characteristics. We offer a local presence for strong customer support including project management, joint marketing and consultancy services. Our in-depth systems integration capabilities meet the diverse needs of service providers, from supplying a single network element to providing an end-to-end, fully managed solution.

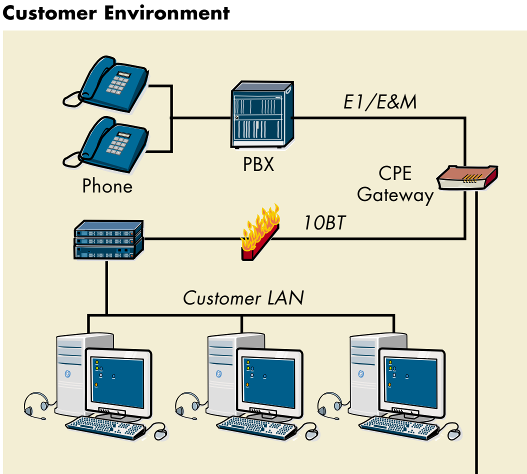
Complete Portfolio of Products for NGNs

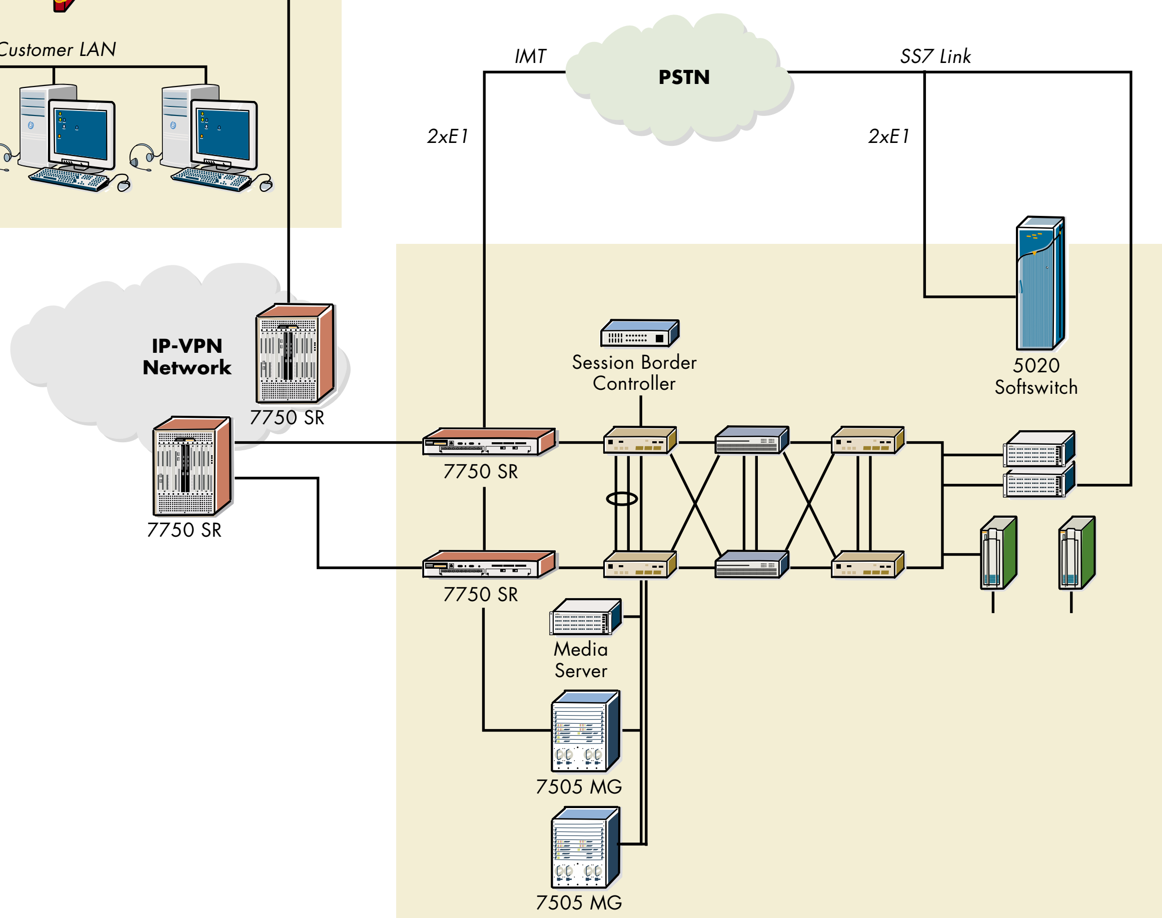
Alcatel demonstrates the broadband access capability that is a key component of an NGN. We offer a comprehensive range of NGN products, including softswitches, media gateways and transport platforms, and we provide an integrated approach to their deployment.

Service providers with an installed base of Alcatel 5020 Softswitches can upgrade these platforms to support feature continuity in an NGN environment. Instead of having to reverse engineer a local country feature set, customers simply migrate to the Alcatel 5020 Softswitch and Alcatel 7510 Media Gateway.

By combining the Alcatel 5020 Softswitch with the Alcatel 7750 SR and Alcatel 7670 RSP, service providers gain an end-to-end VoIP solution that offers industry-leading mediation from ATM to an IP/MPLS VPN and carrier grade VoIP transport.

Figure 5 - Alcatel Solution for Major Operator in South East Asia





Deployment Examples

A major operator in South East Asia provides domestic and international telephone services, ISDN, online and fixed access to the Internet and offers a wide range of value-added services. This telecom operator's VPN is located in different countries in Asia and provides not only leased lines (data and voice) but also voice and mobile services across the region.

The Alcatel solution (see Figure 5) consists of the Alcatel 5020 Softswitch, Alcatel 7505 Media Gateways and an IP core network based on IP/MPLS-enabled equipment, including the Alcatel 7750 SR, which provides QoS in the IP-VPN for VoIP calls. VPN and call routing are the key functionalities provided within the advanced routing module (ARM) of the Alcatel 5020 Softswitch. Call routing is completely VPN-based, which means that it is configurable on a per-VPN basis. The VPN administrator can define individual routing rules for each VPN, which are associated with user groups and the users assigned to each user group. Each VPN can use its own set of associated rules for QoS and routing in the network. Advance routing functions provide least cost routing and QoS through the Alcatel 7750 SR.

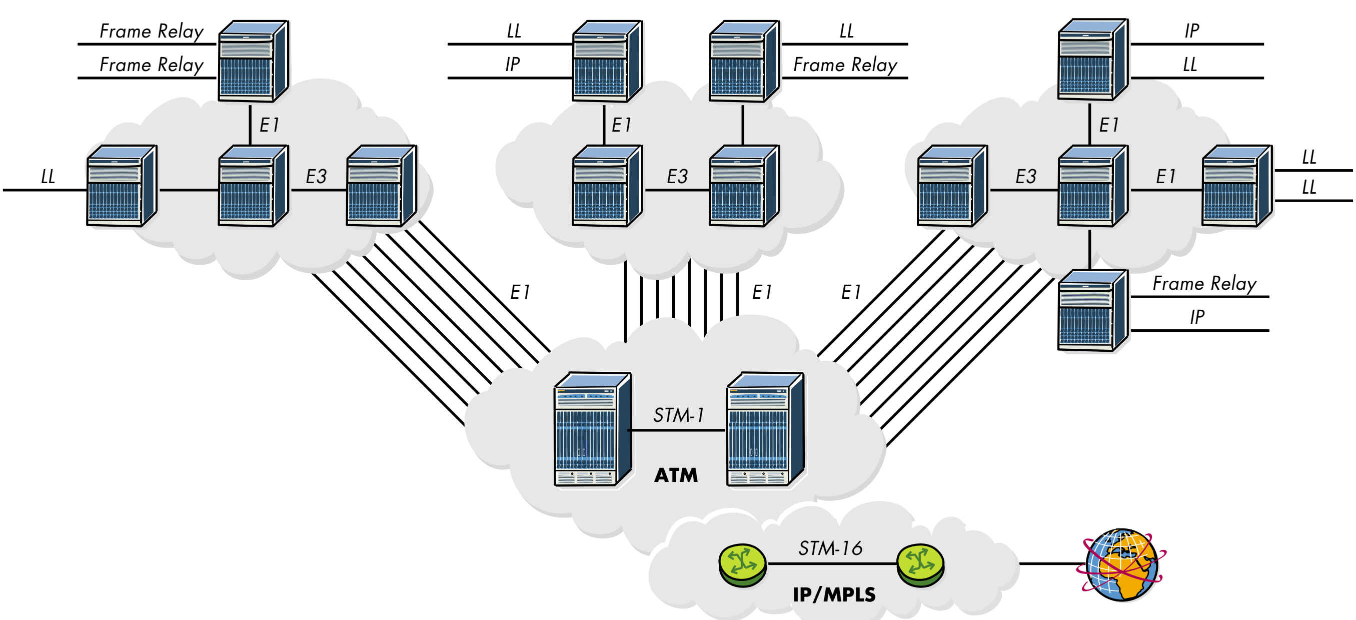
A major operator in Eastern Europe provides domestic and international telephone services, ISDN, dialup and fixed access to the Internet as well as offering a wide range of value-added services. The company offers many Internet access and data services, including IP-VPNs, frame relay, leased lines, LAN and ISDN, and also operates a mobile network. Its leadership was secured through a combination of nationwide coverage, a wide range of products and services and an innovative business strategy.

Data services are currently provided from multiple discrete networks, consisting of existing ATM switches for leased line and frame relay services, a router-based IP/MPLS network for IP-VPNs and Internet access, and an ATM backbone for network interconnection and broadband access (see Figure 6).

The Alcatel 5020 Softswitch is the signaling interface to the SS7 network and provides call control instructions (using SIP, or MEGACO/H.248-based signaling links) to the access gateways and trunking gateways in the network. Alcatel 1540 LiteSpan Multiservice Access Gateways are located in the access portion of the network providing connectivity to residential subscribers via plain old telephone service (POTS) lines. These convert time division multiplexing (TDM) voice channels into VoIP using the standard real-time protocol (RTP), and the real-time control protocol (RTCP) to allow monitoring of the data delivery. Next, Alcatel 7505 MGs are the interface between the TDM-based PSTN voice network and the packet network, where they function as trunking gateways converting TDM voice to VoIP and vice versa.

The Alcatel 7670 Edge Services Extender (ESE) has multiple roles in the network. It aggregates the incoming VoIP traffic carried on E1s from the LiteSpan gateways and the VoIP traffic coming in via 10/100 Ethernet links from the Alcatel 7505 MGs onto STM-1 ATM links for connectivity to the Alcatel 7670 RSP in the backbone.

Figure 6 - Alcatel Solution for Major Operator in Eastern Europe



The Alcatel 7670 RSP forms the IP/MPLS backbone in conjunction with the existing Cisco GSR 12000-based network. The Alcatel 7670 RSP functions as a provider edge (PE) router in this network, converting incoming IP/ATM traffic from the Alcatel 7670 ESE to routed IP traffic. MPLS paths are established between the Alcatel 7670 RSPs and the Cisco 12000 GSR routers, which function as provider (P) routers. The P routers then route the data traffic to the appropriate endpoint where it either remains as IP for an IP-attached phone, or is routed to the PSTN via the Alcatel 7505 MGs for a PSTN-bound call. In addition, the Alcatel 7670 RSPs and Alcatel 7670 ESEs support the delivery of enhanced Ethernet, frame relay and leased line services.

Conclusion

Service providers who want to converge their networks to deliver packet-based voice services must assure service quality and availability for these VoIP-based services end-to-end across the whole data network. This will enable them to provide the same carrier grade qualities that end users are accustomed to with their traditional voice service. In order to achieve carrier gradeness, the data network must be application-aware and service-oriented.

The Alcatel VoIP carrier transport architecture provides these characteristics through these key components:

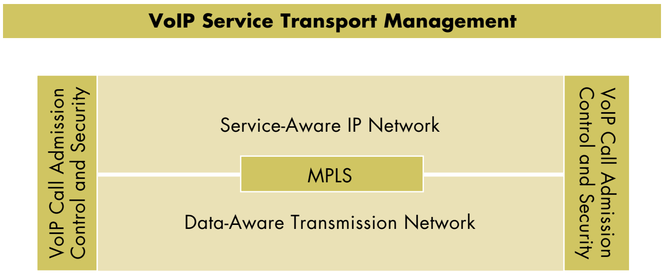
> Service-aware IP network and data-aware transmission network: Serves as the network foundation for high quality VoIP transport. The service-aware IP network is assembled from an IP portfolio that includes the Alcatel 7750 SR, Alcatel 7670 RSP and Alcatel 7450 ESS. The Alcatel 1662 Packet Ring Switch (PRS) or OMSN family are essential in the data-aware transmission network.

> VoIP call admission control: Allows only revenue-generating VoIP traffic to be put onto the network. The Alcatel 5020 IPBN provides this function when required.

> VoIP transport management: Allows efficient service provisioning and per-customer, per-application transport management and troubleshooting. Management is provided by the Alcatel 5620 SAM and Alcatel 1354 BM.

These components are illustrated in Figure 7.

Figure 7 - Alcatel VoIP Carrier Transport Framework



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