VoIP

Addressing QoS Beyond the Provider Network

This White Paper explores the issue of voice quality in VoIP networking and current status of QoS technologies and implementations in today's IP networks. We discuss the QoS mechanisms SmartNode™ employs to ensure the best-possible voice quality over the network-access link.

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Introduction

Today's Internet does far more then email and file transfers. Initially designed for non-real-time (NRT) data applications, the Internet has matured far beyond these tasks. In addition to web browsing, online imaging, and chat rooms, we now expect the Internet to deliver such real-time (RT) media as streaming music, video, and Internet phone calls directly to our homes and offices.

Presently, Voice-over-Internet Protocol (VoIP) is all the buzz—but it's more than just talk. VoIP technology has matured. Not only is it the latest hot new Internet application. Today, VoIP has emerged as a reliable technology that is commercially viable, competing (and winning) against traditional phone services in business and consumer-class markets.

As a real-time application, VoIP—also known as packet voice, packet telephony, or IP telephony—places increased demands on the evolving Internet. VoIP users expect the Internet to deliver toll-quality voice with the same clarity as the traditional Public Switched Telephone Network (PSTN). To meet those expectations, the Internet connection must be more than merely reliable, it must be time-sensitive. Each and every voice packet must be delivered without significant delay and with consistent time intervals between packets.

Advanced Quality of Service (QoS) technology is the key to achieving voice quality that measures up to today's high standards. With SmartNode and Patton's advanced QoS, toll-quality voice on every call is the benchmark.

Quality of Service (QoS)—In a successful VoIP deployment, the perceived voice quality of VoIP calls must satisfy user expectations. Specifically, voice quality in the VoIP system must compare favorably with that of a traditional phone call. To ensure VoIP users hear the best possible voice quality, Patton's SmartNode employs a unique combination of varied QoS mechanisms. Network administrators can tune SmartNode's QoS for optimal performance in a broad range of networks giving packet voice the clarity and sound quality we expect on every call.

QoS and voice quality

From a technical perspective good voice quality involves minimizing delays and interruptions ("blips") in the communication stream. For voice communications over an IP network, packet delays and losses in the VoIP network must be reduced to the levels of established benchmarks. To achieve a level of perceived voice quality that most users find acceptable, end-to-end packet delay must be reduced to a target of 120ms or less. In most IP networks some degree of packet loss may be inevitable. However, for a successful VoIP deployment, we must reduce packet loss to well below 1%. For Fax-over-IP connections, the packet-loss target is especially critical.

Packet loss and packet delay may accumulate at multiple locations within the IP network. Every router, switch and transmission line is a potential culprit for harboring these enemies of voice quality. Technology standards such as TOS or DiffServ are designed to achieve the goal of establishing QoS in each and every node, thus enforcing QoS mechanisms throughout the network from end to end. (These standards will be discussed later in this paper.) The reality today, however, is an Internet that does not differentiate between real-time (RT) and non-real-time (NRT) packets. As a result, VoIP networks require alternate methods for ensuring voice quality.

Addressing QoS at the edge of the network

In VoIP systems that traverse the Internet, the bottleneck typically occurs at the access link—the low-bandwidth connection between the high-speed Internet-backbone (WAN) and the user network (LAN). Both networks typically run at 100 Mbps or above. A typical access link may easily run about 200 times slower than the LAN residing in the home or office (say, 512 kbps for example). Congestion, queuing delay, and queue overflows (resulting in dropped packets) are most likely to occur on this link. Depending on access-link bandwidth, packet size and burst size (the number of packets arriving at once), queuing delay can be especially significant.

In a typical installation, a single access link serves both voice and data traffic, so special measures must be employed to ensure good voice quality. Consider the case of a 256 kbps access link from the Internet's edge router to the user's LAN. Suppose the Internet's transmit queue contains five 1500-byte data packets, followed by one time-sensitive voice packet. It will take roughly 270 ms to send those data packets over the 256 kbps link. When the voice packet follows, it arrives with a delay longer than 120 ms (our target), resulting in degraded voice quality.

Introducing class-of-service in the packet layer addresses queuing delay by ensuring voice packets receive priority treatment—in much the same way that separate queues at airport check-in counters ensure priority service to first-class customers. By creating separate queues for Real-Time and Non-Real-Time traffic, we can assign higher priority to the RT queue and serve that traffic with higher priority.

Upstream and downstream traffic present different problems, and are best addressed by different Quality of Service (QoS) mechanisms. Upstream traffic flows from the (home or office) user to the Internet, while downstream traffic flows from the Internet to the user.

Implementing QoS in the upstream direction is relatively easy. SmartNode™ ensures that outbound voice packets get served before other packet types to prevent the Internet access link (the bottleneck) from becoming overloaded. SmartNode also provides tuning mechanisms for additional parameters like packet segmentation and overhead optimization, so network administrators can further fine-tune the upstream transmission for optimum voice quality.

Implementing QoS in the downstream direction is more complex. Typically, customer-premise equipment (CPE) at user locations exerts no control

Figure 1. Internet Access

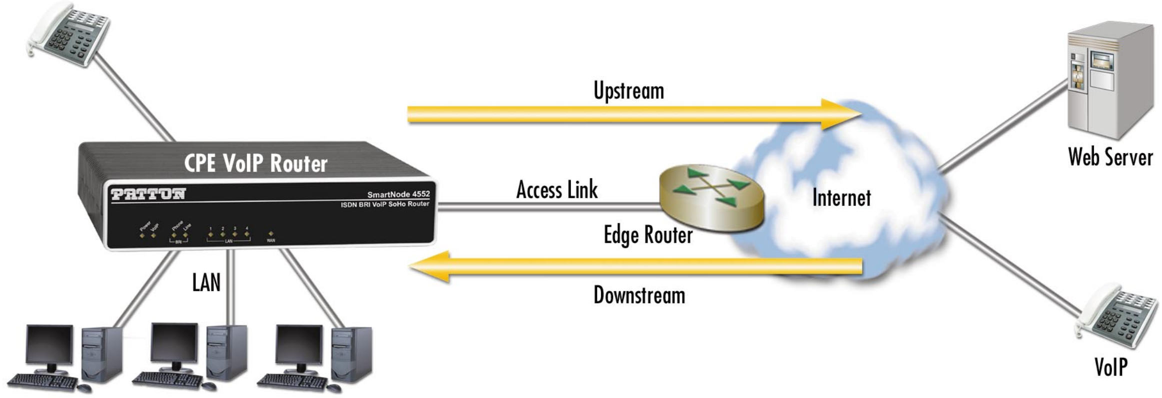
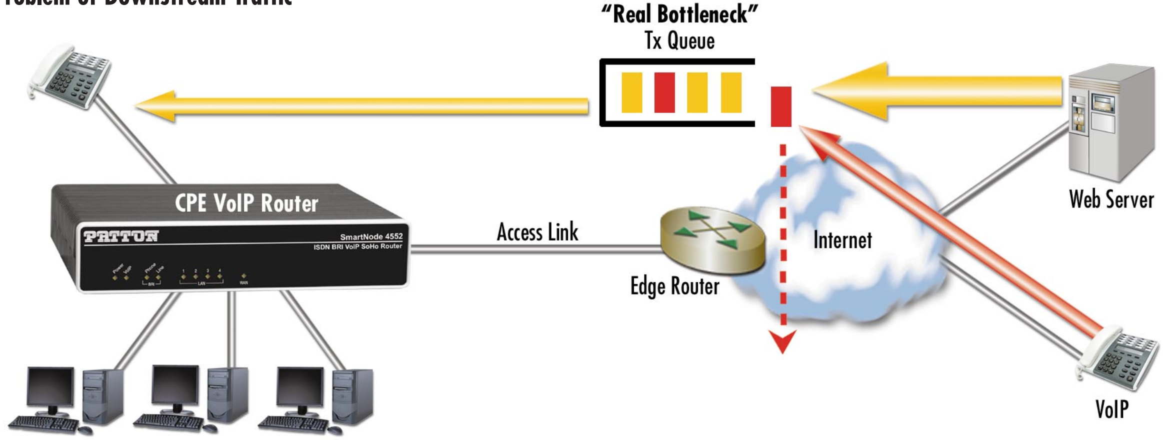


Figure 2. The Problem of Downstream Traffic



over incoming traffic. For traffic flowing downstream from the Internet, the access router cannot control the volume nor the sending users. In addition, local users who share the LAN with you can initiate file transfers or download their email at their convenience. The servers handling these requests may be located anyplace. Since the downstream has not been controllable, the ISP's edge router commonly responds to overloading by discarding VoIP packets with the same probability as any other packet type. These factors, or a combination of them, may degrade voice quality to a degree that users find objectionable. Data traffic, on the other hand, can be retransmitted so the impact on the user experience is simpler slower service response.

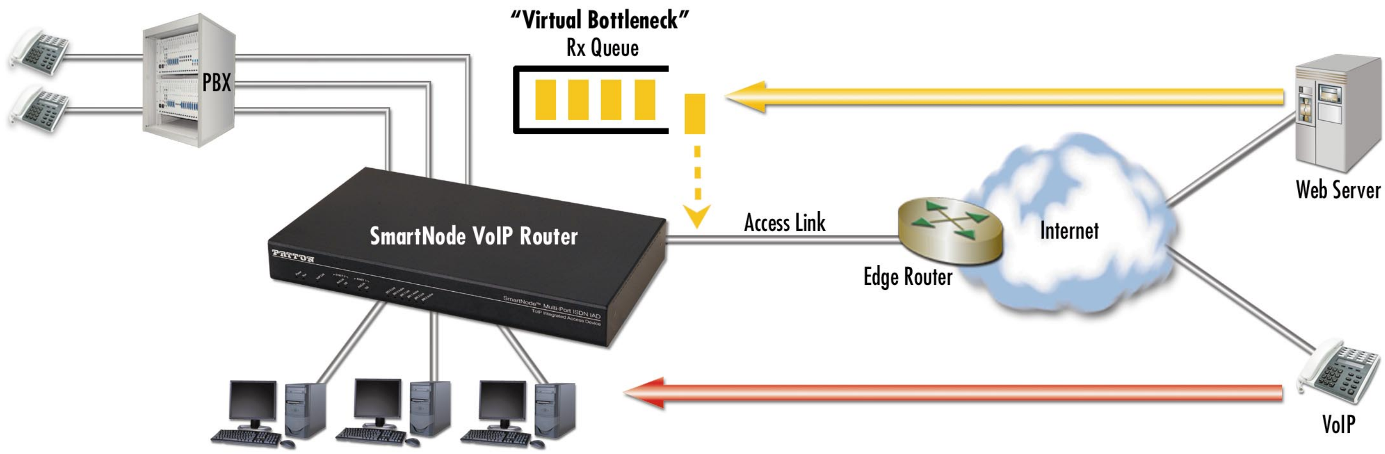
To resolve the problem of degraded voice quality for incoming traffic, Patton has devised a leading-edge technology for SmartNode™ called DownStreamQoS™. Within the SmartNode deployed at the customer premise, DownStreamQoS dynamically creates a virtual bottleneck against the incoming packet stream. This bottleneck can throttle back Non-Real-Time traffic, preventing the edge router from blocking or impeding voice traffic, to ensure voice packets are transmitted freely downstream. DownStreamQoS employs flow-control mechanisms within the TCP standard to create the bottleneck. Because 80% of Internet traffic is transported via TCP, DownStreamQoS is especially effective. The dynamic bottleneck adjusts to varying traffic patterns. So whenever there are no VoIP calls in progress, the full downstream bandwidth is available for incoming data such as file downloads.

Technology standards for QoS

When implementing QoS to ensure good voice quality, VoIP networks may employ a selection of mechanisms from a variety of standard communications protocols. Such mechanisms may include:

- Tag or Label within the Packet or Frame
- Traffic (QoS) Classes
- Traffic (QoS) Conditioning or Packet treatment

Figure 3. SmartNode™ DownStreamQoS™



The table below summarizes the standard protocol mechanisms Patton has employed in our unique implementation of QoS for SmartNode VoIP solutions.

A key point to remember about any QoS-related standard is the following: a standard is only as effective as its specific implementation in a specific network in the real world. For example, almost every router in the world claims to support TOS labeled IP packets. "So why," we want to ask, "is the Internet still a best-effort network? Why has the Internet not implemented QoS?" The answer lies in the additional complexity of administering a network that delivers QoS.

A network operator who wants to offer different service classes must do all of the following:

- Provide Service Level Agreements (SLAs) for the service classes offered (e.g. throughput guarantees or maximum delay values)
- Bill separate service classes individually
- Ensure that service classes are not fraudulently misused
- Dimension the network in a way so that SLAs for each service class can be met, even in fallback and redundancy scenarios

Standard	Layer	QoS Label	Description
IEEE 802.1pQ	2 (Ethernet)	3 bits in Ethernet Header	Extended Ethernet frame for VLAN and QoS. Defines eight Priority Classes.
MPLS	2 (ATM, FR or Shim Label on IP)	Not applicable	Label switching protocol for core routers. Can be used for traffic engineering, VPN and QoS differentiation.
TOS/Precedence	3 (IP)	1 octet in IP Header	Defines three traffic characteristics: low delay, high throughput, high reliability
DiffServ	3 (IP)	Same octet in IP Header as TOS	Defines a 6-bit field for service classes and a number of Per-Hop Behaviors (PHB) on how to treat packets

Administering a QoS network is much more complex than ensuring reliable connectivity in a best-effort network. This complexity is why networks supporting different service classes have not appeared until quite recently, and only a limited scale, and only from single operators. Considering the content of our discussion, one can begin to grasp the complexity of implementing QoS end-to-end across the boundaries of multiple networks.

Conclusion

The Internet Engineering Task Force (IETF) and the networking industry have defined a number of technology standards and methods for offering levels of IP-network service that exceed the typical best-effort service levels currently available. Yet the reality today is that IP Networks—and the Internet in particular—continue to lag behind in implementing these existing technologies in order to deliver QoS to subscribers. The reason for the delay lies in the substantial complexity involved in administrating a QoS network.

SmartNode has implemented QoS mechanisms that significantly improve service quality for voice and data in a crucial section of the network—the subscriber access link. Further, SmartNode's QoS mechanisms operate with complete independence from the Internet or wide area network. Finally, SmartNode is flexibly designed for future adaptation, ready and able to support leading QoS technology standards as network providers begin to deploy them.

Today's enterprises, carriers and service-providers are eager to realize the cost-savings and flexibility of VoIP. Patton's SmartNode™ family of VoIP solutions offers a complete line of VoIP Gateways and Broadband Internet Access Routers that deliver those cost and flexibility benefits now, while offering a smooth migration path to tomorrow's fully-converged voice-data networks. Scaling from dual-port analog gateways up to Quad-PRI digital Gateway-Routers, all SmartNode™ models combine standards-based signaling (SIP, H.323 and MGCP/IUA) with advanced Patton's feature-rich SessionRouter™ software to deliver robust Telephony-over-IP with industry-leading voice quality. SmartNode VoIP solutions integrate seamlessly with legacy communications infrastructures to deliver end-to-end voice services over any IP network.

Key SmartNode features to make your VoIP project a success:

- **Standards Compliance**—Protecting your investment in VoIP, SmartNode implements key industry standards for proven interoperability with third-party equipment and networks.
- **Advanced Call Routing**—SmartNode's flexible SessionRouter software seamlessly integrates VoIP with existing telephony systems and numbering plans.
- **Feature Transparency**—SmartNode forwards PBX and PSTN features transparently to the user, preserving the benefits of advanced calling features in both traditional telephony and digital packet-voice networks.

SmartNode Product Line Summary

Product Line	Segment	Series	Description
Residential		S-ATA	Residential Smart Analog Telephone Adapter
		S-DTA	Residential Smart Digital (BRI) Telephone Adapter
Branch Office SOHO		SN41XX	Multi-Port Analog VoIP Gateway
		SN402X	Analog VoIP SOHO Router
		SN455X	ISDN BRI VoIP SOHO Router
Enterprise	Medium Enterprises	SN452X	Multi-Port Analog VoIP IAD
	Medium Enterprises	SN48XX	Multi-Port Analog IAD with Integrated WAN Access
	Medium Enterprises	SN463X	Multi-Port ISDN VoIP IAD
	Medium Enterprises / Large Enterprises	SN465X	Multi-Port ISDN VoIP IAD with Integrated WAN Access
Carrier	Medium Enterprises / Large Enterprises	SN1200/1400	ISDN BRI VoIP Gateway
	Large Enterprises / Carrier	SN2350	Modular VoIP Gateways
	Large Enterprises / Carrier	SN2300	Modular VoIP Routers
	Large Enterprises / Carrier	SN4900	IpChannel Bank
	Carrier	SN2450	4-Slot Modular VoIP Gateways
	Carrier	SN2400	4-Slot Modular VoIP Routers

Product Highlights

Low Cost Analog & ISDN Adapters

- Low Cost simplified VoIP Gateways
- Plug and Go Feature Set
- Two Models for both Analog and ISDN Phones
- Single and Multiline with full FAX support

Branch Office Routers with VoIP

- SOHO VoIP with seamless connection to enterprise voice and fax
- Analog or digital options for one or two lines
- Network compatibility features allowing use virtually any IP service network with VPN, QoS, and Firewall features
- Single box LAN/WAN integration

Mutiport Analog & ISDN with integrated WAN

- Small enterprise; branch/satellite office
- Voice extension and PBX voice and data networking; any-access-to-any-network
- PSTN access with FXS and FXO and FXS/FXO combinations; or ISDN BRI So
- Complete LAN/WAN routing with QoS, VPN, and network integration for a single-box solution
- Integrated WAN options for full routed connections with PPP/Frame Relay

IpChannel Bank

- Analog trunking VoIP
- Multiport analog connections for call centers
- Expansion slot for integrated WAN uplinks using PPP/Frame Relay
- Reliability with redundant power and redundant Ethernet
- Low to high density (from 12 to 32) analog ports

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