PacketSaver[™]

More Efficient, More Reliable VoIP



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PacketSaver: More Efficient, More Reliable VolP

Voice over Internet Protocol (VoIP) offers a wide range of benefits to both enterprises and communications/network service providers. These include lower costs, unified management of voice and data infrastructure, and – perhaps most importantly – the ability to deploy a new generation of converged voice/data applications.

Although use of VoIP is growing rapidly, several factors have inhibited more rapid adoption across all market segments. These factors include concerns about maintaining consistent voice quality over IP networks, especially during periods where other types of traffic on the IP network suddenly "spike" – potentially putting the squeeze on voice packets and momentarily threatening voice quality.

Quintum Technologies has directly addressed this issue with its multi-switching VoIP architecture, which instantaneously re-routes voice traffic from the IP network to the public switched network if conditions on the IP network threaten to compromise call quality.

Now, Quintum has gone a step further in ensuring the quality of VoIP with its innovative new *PacketSaver*[™] technology. *PacketSaver* multiplexes multiple individual VoIP sessions into consolidated IP packets, significantly reducing the total amount of bandwidth needed to support voice calls over IP networks. This more efficient approach to transporting VoIP results in three key benefits:

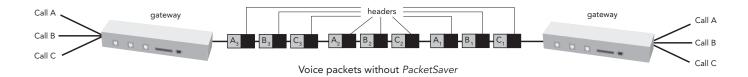
- 1) By minimizing the bandwidth required for VoIP traffic, *PacketSaver* lowers the likelihood that other congestion on the network will threaten voice quality.
- 2) Because *PacketSaver* reduces the total number of packets used for VoIP traffic, it also reduces the chances of packet loss another factor that can affect voice quality.
- 3) In minimizing the amount of bandwidth required for VoIP, *PacketSaver* contributes to the overall efficiency of the IP network reducing congestion and lowering overall infrastructure costs.

This last benefit is particularly appealing to VoIP service providers, whose profitability is largely contingent on their ability to deliver high-quality voice services over IP networks at the least possible cost.

Quintum's *PacketSaver* technology is equally applicable to fax-over-IP (FoIP) traffic, enabling enterprise customers and service providers alike to achieve similar cost-efficiency gains for document-based communications.

How PacketSaver Works

In the case of voice over IP, packets are usually created as a VoIP gateway receives a voice stream. The gateway compresses the voice and digitizes it into a packet payload, adding a header with destination information. With conventional gateways, packets are created as each individual voice stream hits the gateway. This results in the creation of tremendous header overhead. The network – which usually has lots of data packets flowing over it already – can easily become saturated, causing congestion and even lost packets. This can compromise the voice quality, as well as the overall health of the network.



Quintum's *PacketSaver* technology is a packet assembly technique that queues up several voice and/or fax packets and multiplexes them into one or more larger packets, without introducing unacceptable delays in packet processing. In other words, packets from one or more sampled voice conversations are multiplexed (or "packed") together into one larger multiplexed voice packet headed for the same destination device on the other side of the network. When the packet is full, or the pre-determined time limit for packet construction is reached, it is sent to the common destination point.

When this multiplexed packet reaches the destination device, it is then de-multiplexed, allowing the packets for each original voice conversation and/or fax transmission to be distributed to their appropriate end-point destinations.

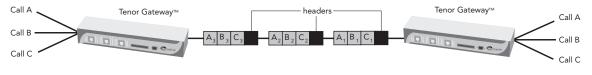
A sophisticated, patented algorithm controls exactly how packets are identified for multiplexing and then effectively de-multiplexed at the other side of the wide-area network (WAN connection). This tunable multiplexing control mechanism also ensures that PacketSaver does not introduce unacceptable processing delays into the network.

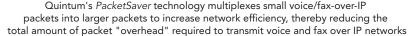
How PacketSaver Conserves Bandwidth

The bandwidth savings that *PacketSaver* provides is a result of the fact that multiple voice and fax packets heading to the same destination device share a common packet header. This greatly reduces the packet "overhead." It thus also reduces the possibility of packet loss that occurs with packet based calls in congested data networks.

One way to picture this savings is to think about mailing letters. A letter acts much like a voice/fax packet. The letter itself contains the information that you want to send from point A to point B. The envelope, on the other hand, is there only to carry the letter. But you need the envelope, since it has the address and the stamp required by the postal service for delivery. The envelope is therefore a lot like an IP packet "header," which also provides necessary addressing and validation data.

Now, imagine you wanted to send several letters to several people who all worked at the same company office across the country. If you sent each one a separate letter, you would have to spend a lot of money on envelopes and postage – and the post office would have to sort several pieces of mail. But if you sent them all in one envelope, you'd save on your postage and stationery costs. Someone at the destination office would still have to distribute the letters to the right people, but they'd have to do that even if they got separate individual letters from the post office anyway!



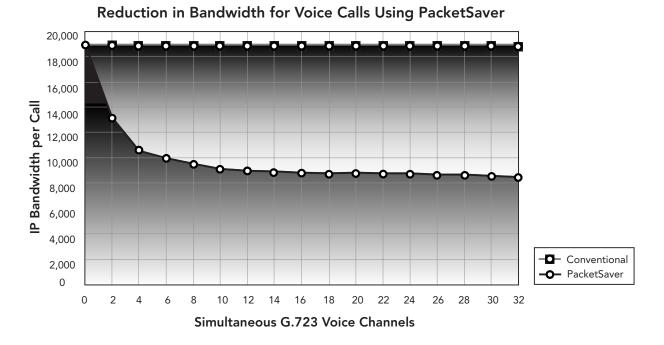


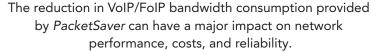
PacketSaver, to follow the post office analogy, uses a single header for a larger, multiplexed voice/fax payload - reducing the total overhead associated with packetization. By reducing the ratio of header "fat" to payload "meat," this multiplexed packetization reduces the total bandwidth consumed by VoIP/FoIP calls and lessens processing loads on WAN routers.

It's important to note that the rate at which a gateway packetizes voice and/or fax traffic depends on the particular codec (code/decode software) it's using. The G.723.1 codec, for example, transmits a packet once every 30 milliseconds. *PacketSaver* does not alter this rate, even as it assembles larger packets containing different voice and fax transmissions. It simply gathers more of those separate transmissions together before sending out its larger packets.

The number of bytes in a voice packet's "payload" is also determined by the codec. For example, a voice packet using G.723.1 encoding will have a 24-byte payload. The size of a packet's header is also fixed, regardless of the size of the attached voice packet. For VoIP or real time media over IP, all the information required – including the timestamp, sequence number, destination, etc. – requires the header to be 46 bytes. Thus, header overhead actually consumes more bandwidth than voice/fax payloads in conventional VoIP networks.

PacketSaver reverses this illogical ratio, allowing VoIP and FoIP to move more efficiently over enterprise and service provider networks without compromising quality in any way. Simply combining just three calls reduces bandwidth by 40%!





Conclusion

PacketSaver is a powerful technology for any organization seeking to better leverage IP networking to support voice and fax, in addition to data. By reducing bandwidth requirements and router processing loads, *PacketSaver* cuts costs and improves reliability. Combined with Quintum's proven multi-switching architecture, *PacketSaver* presents VoIP/FoIP implementers with the most robust and cost-effective solution for building high performance, multi-purpose networks.

About Quintum

Quintum Technologies is an innovator in the voice-over-IP (VoIP) market. The company offers highly reliable VoIP products that deliver superior voice quality and provide an easy, risk-free migration path to the convergent future of networking. Quintum was founded by Cheng T. Chen and Dr. Rajiv Bhatia, both of whom have over 20 years of experience as lead engineers at companies including Bell Laboratories, Teleos, Madge and 3Com. The company's mission is to deliver enterprise-class VoIP solutions that provide:

- Outstanding value to customers
- Ease of installation, ease of use, and ease of management
- Superior quality and reliability
- Open architectures and standards compliance
- Flexible migration to succeeding generations of convergence technology

Quintum's unique Tenor MultiPath VoIP Gateway is the first VoIP gateway that intelligently switches calls over both IP networks and the PSTN in order to ensure high voice quality and provide failover capability. Unlike conventional VoIP gateways that only route calls over IP networks, the Tenor Gateway can transparently switch calls over to the PSTN if IP network congestion or a device failure impacts voice quality. The Tenor Gateway thus addresses the reliability concerns that have heretofore prevented many corporate decision-makers from moving ahead with VoIP and receiving all of its benefits.

Quintum Technologies Inc. is a privately held corporation headquartered in Eatontown, N.J. More information on the company, its management team, and its products can be found at www.quintum.com.



14 Christopher Way Eatontown, NJ 07724 877-SPEAK IP (toll-free) 732-460-9000 732-544-9119 fax www.quintum.com info@quintum.com