tellabs

# Network Infrastructure Requirements for VoIP

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March 2004

MPLS FACILITATES THE CARRIER-CLASS NETWORK RELIABILITY, QUALITY OF SERVICE, AND CREATION OF CIRCUIT-LIKE SERVICES (USING LSPS) OVER IP NETWORKS WHICH ARE ESSENTIAL FOR VOIP. MPLS VPNS PROVIDE THE NECESSARY LOGICAL SEGREGATION, FOR MANAGED-BUSINESS VOIP APPLICATIONS.

*Today, more than ever, service providers are preparing to migrate voice traffic from a traditional circuit-switched network to a data network using VoIP technology. The technology barriers in VoIP have been lifted with the standardization of a wide choice of open protocols such as H.248/ Megaco[1], H.323[2], SIP[3] and MGCP[4] leading to a wide choice of networking systems such as soft-switches, media-gateways, media-servers, session-controllers and GR-303 packet gateways. These VoIP network systems are readily available for general deployment from many system vendors.*

*The business drivers to converge voice and data services over a single multi-service IP network are more compelling then ever since service providers need to reduce operational expenditures incurred by managing multiple overlay networks. MPLS has emerged as the technology of choice for convergence of voice and data networks. Service providers can use their existing MPLS-enabled IP networks to carry their VoIP traffic to get around the traditional pricing structures which are regulated by the FCC and reduce the total cost of a phone call.*

## Executive Summary

Trends in the mobile phone market reflect the usage of data-applications on cell phones to send and receive instant short messages. By converging voice and data onto a single IP network and using VoIP, service providers can offer similar types of productivity-enhancing applications to their wireline customers.

Voice is the single major revenue source for almost all service providers that want to gracefully migrate to VoIP without disrupting voice revenues. However, service providers need to know the fundamental network infrastructure requirements to support VoIP over a converged IP network. This white paper discusses different VoIP network models relevant to their applicability of use, highlights the functional specifications of the VoIP network elements and exemplifies the key requirements of the underlying network infrastructure to support these VoIP network models.

## VoIP Network Elements[i]

In VoIP networks, the underlying IP network provides the connectivity between the distributed VoIP network elements. Depending on the choice of standards, commonly used VoIP network elements and functionality include:

- Softswitches provide call control, signaling and intelligence, tie together various next-generation voice hardware; reside on a server or a dedicated hardware platform; and control next-generation voice hardware using H.323, MGCP, H.248 or SIP.
- Media gateways provide switching and adaptation between the Public Switched Telephone Network (PSTN) and data networks and have packet/cell and PSTN interfaces.
- Voice application servers provide IP multimedia services such as unified messaging, self-provisioning, voice VPN, IP and Centrex. These application services communicate with a softswitch via SIP.
- Next-Gen voice switches are media gateways that deliver packetized voice services to customer premises by providing a subset of Class 5 switch functions and SS7[5] signaling via an integrated softswitch.
- Remote Access Concentrator (RAC) VoIP gateways are carrier class, Network Equipment Building System (NEBS) level 3 compliant, remote access concentrator-based media gateways using TDM interfaces to connect to the PSTN and packet interfaces to connect to the data network; they support H.323 with future migration to SIP and H.248.
- Voice over broadband gateways provide gateway function between data networks and PSTN using GR-303/V5 interfaces and connect to Class 5 switches and data access networks (e.g., DSL, cable, T1/E1) to enable multiple packetized voice lines over a single data access line.
- Broadband loop carriers are softswitch controlled, packet-based platforms designed for remote terminals; provide Plain Old Telephone Service (POTS), DSL, and packet-based uplinks; feature remote provisioning; and packetize voice traffic on a line-by-line basis.
- Media servers process, manage and deliver media requests made by voice application servers in a packet-based network and are controlled by a softswitch via MGCP, H.248 or SIP.
- IP session controllers control and anchor the multimedia session and appropriate signaling such as SIP, H.323, MGCP or H.248. They perform the native IP interconnection functions required for real-time communications such as access control, Network Address Translation (NAT), and packet processing for QoS.

## Network Models

Broadly speaking there are three kinds of VoIP networks emerging in today's service provider networks. While they are functionally similar, these three VoIP network models have different topological attributes. *VoIP networks also are being deployed within enterprise networks, interconnecting IP phones and IP PBX's, commonly known as "Enterprise VoIP" and will not be discussed in this document.*

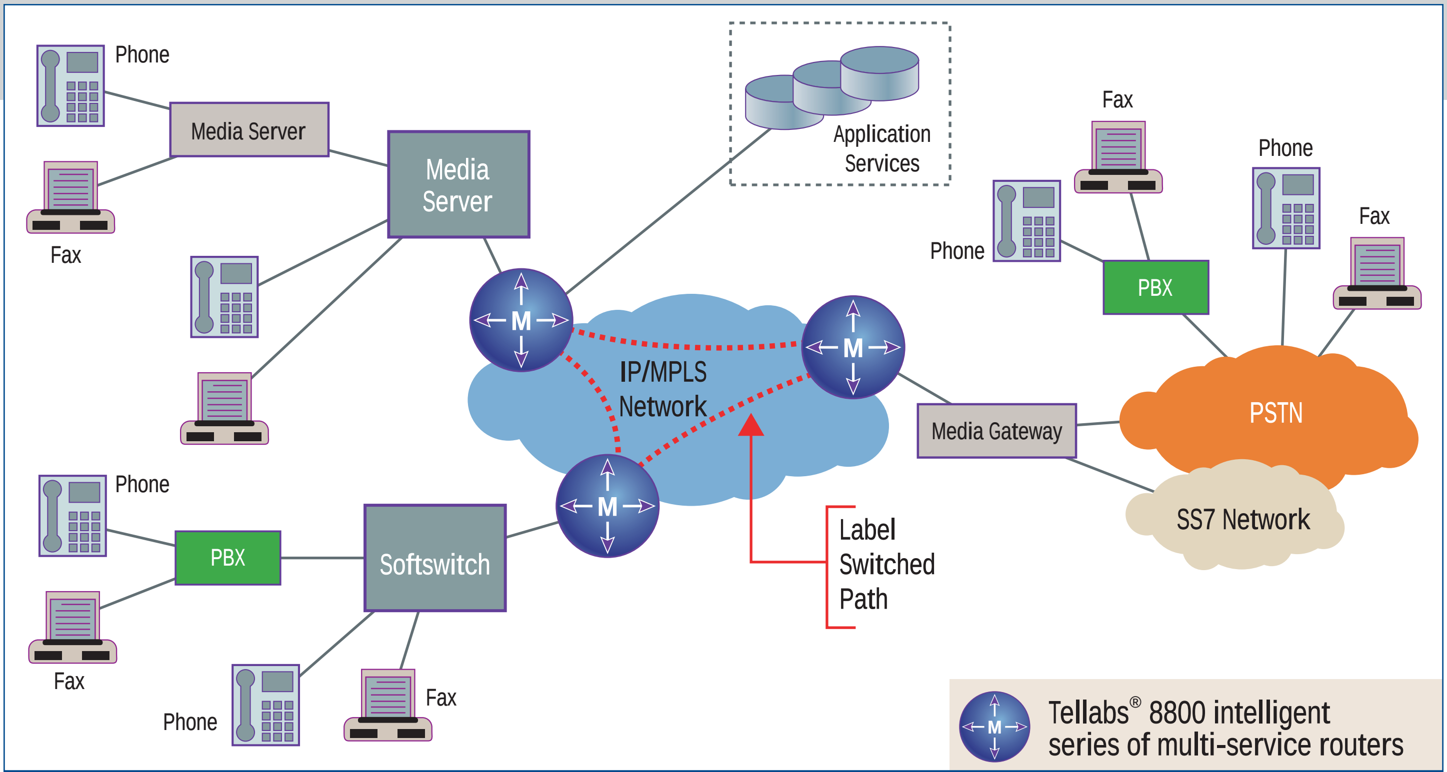


Figure1: Softswitch/Media Gateway VoIP Network

## Generic Softswitch/Media-Gateway VoIP Network Model

Both the Incumbent Local Exchange Carrier (ILECs) and Inter-Exchange Carriers (IXCs) look to VoIP to reduce the cost of delivering voice services, and position themselves to be ready for integrated services of the future. By moving to a VoIP network architecture local telephone companies can reduce the number of traditional voice switches and switching centers with less expensive VoIP network elements such as line and trunk gateways.

In Figure 1, softswitches replace traditional Class-5 switches and provide dial-tone service to PBXs, telephones and fax machines. Multiple softswitches are interconnected via an IP/MPLS network. The interaction between the VoIP network and the PSTN is via a media gateway, which has an IP/LAN connection to the IP network and TDM circuit connection to the PSTN network. The softswitch has the intelligence to coordinate call control and signaling between other softswitches for on-net calls that originate and terminate on the IP network, and between softswitches and media gateways for off-net calls that originate in the IP network and terminate in the PSTN. For off-net calls, signaling translation is done between the appropriate VoIP signaling protocol and SS7 in the media gateway. Similar to the PSTN network, the signaling traffic in VoIP networks is kept independent of the voice-bearer traffic. The softswitches and the media-gateways mark each IP packet with different priority using the diff-serv bits with different DSCP values in the IP header to distinguish between IP packets carrying signaling traffic and bearer traffic. Inherently the signaling traffic has the higher DSCP precedence than the bearer traffic.

It is the function of the ingress edge router on the periphery of the IP network to classify these incoming IP packets based on their DSCP values, queue them accordingly and guarantee that the signaling traffic gets higher priority treatment. Once these packets enter the IP/MPLS network, the edge router has to forward this traffic via a pre-established MPLS Label Switched Path (LSP) to the destination. Within the LSP, the edge-router also should have the ability to queue granularly signaling and bearer traffic that have different priorities. To perform these complex and cumbersome functions reliably, the edge router must have a dedicated queue per-IP-flow and each queue must support multiple types of

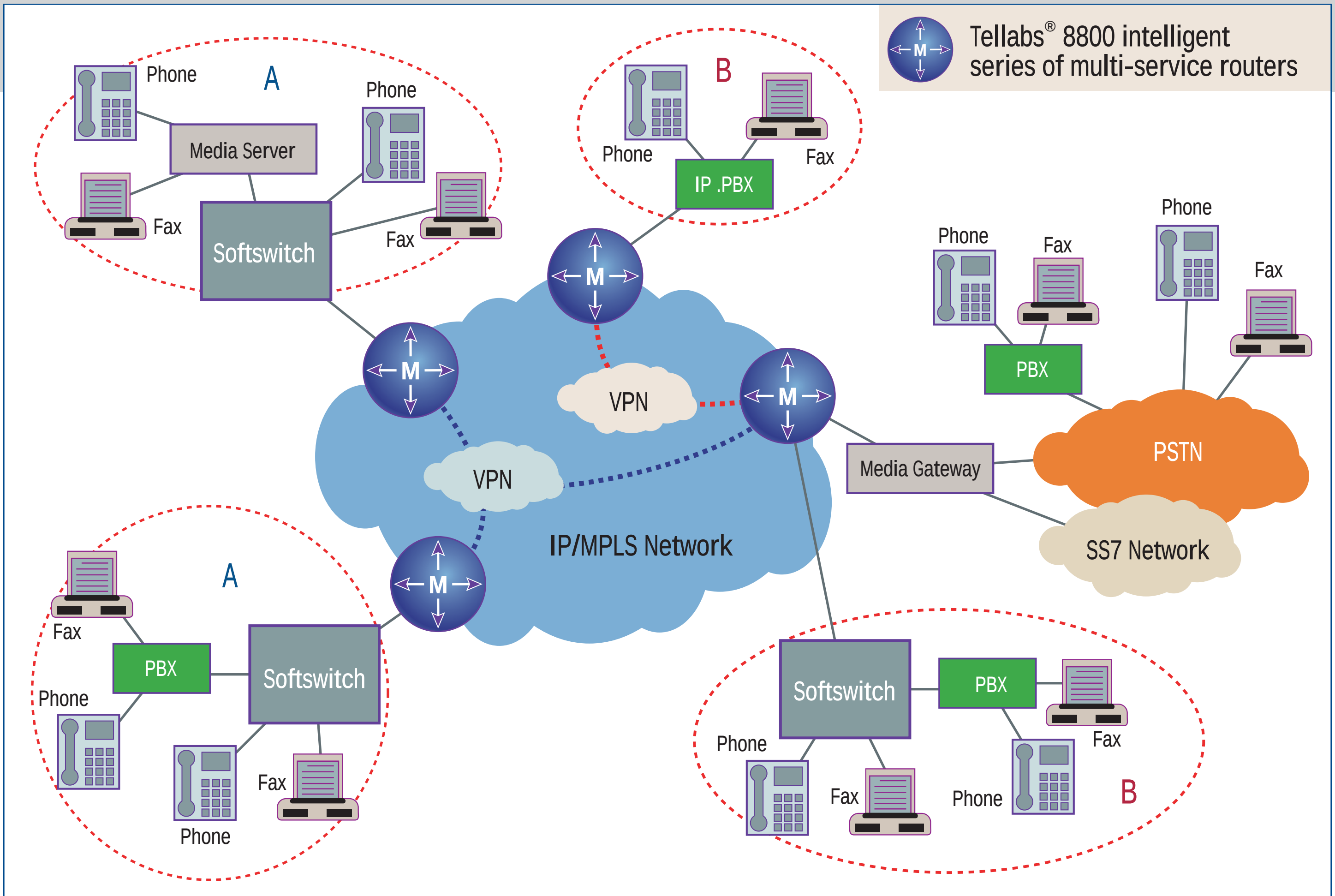


Figure 2: Managed Business Voice Network.

traffic classes. Each traffic class must be governed with upper and lower bound parameters such as min-data rate, max-data rate and burst-size. Also, with an interface that is both incoming and outgoing into the IP/MPLS network, and to preserve a guaranteed min-data-rate across all the per-flow queues, the edge router must have the ability to support connection-admission-control logic and to reject any erroneous over-provisioning beyond the queue limits. The ability to shape and police traffic into each per-flow queue is essential to maintain an end-to-end QoS across the IP network. The current generations of edge routers were not designed to support QoS for differentiated services on a per-flow basis or to support circuit-like attributes for IP traffic such as CAC.

## Managed Business Voice Network Model

Enterprises that have multiple locations outsource their voice networks to service providers to operate and manage their enterprise-wide voice network services. These networks typically include interconnecting the PBXs at each site of the enterprise through the service provider's circuit-switched voice network with the ability to call from one site to another using the enterprise-wide, 7-digit intercom numbers. As long as the voice traffic originates and terminates within a given enterprise customer's locations, the calls are billed as on-net calls and are part of the Voice-VPN service. Off-net calls outside the territory of the enterprise locations are switched into the PSTN via the service provider's voice network. These managed voice-VPNs were introduced by AT&T and Sprint in the 1980's and maintained their predominance until recently. The main advantage of these services for the end-user is the ability to maintain a virtual corporation with local dialing within the

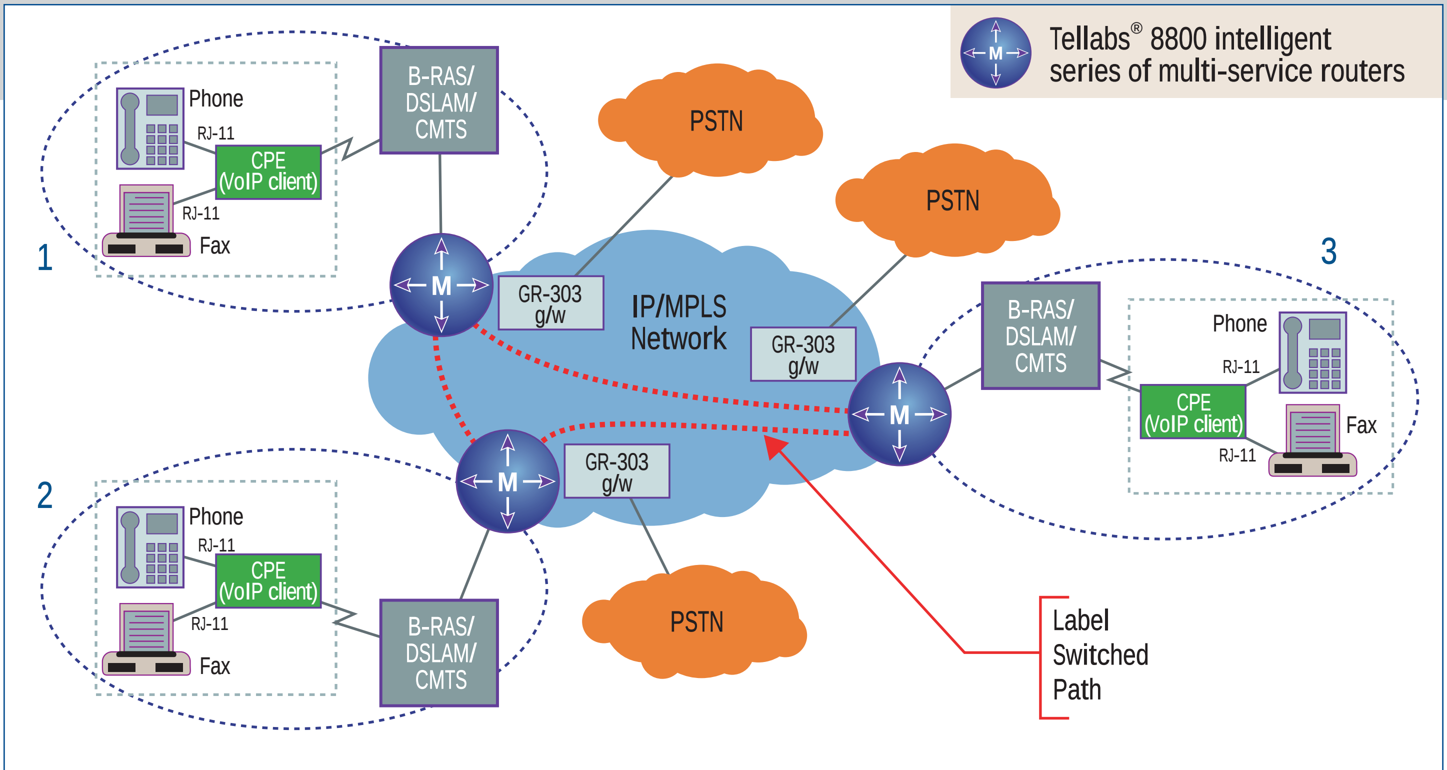


**Figure 3: Residential VoIP Network.**

corporation regardless of the location of its offices, national or international, while gaining value pricing for voice services.

As service providers replace traditional circuit-switched voice network elements it is essential to migrate these managed-voice VPNs onto the VoIP network. While the functional operation of these managed VoIP networks is similar to the above softswitched model, the underlying IP/MPLS network must have the capability to support discrete multi-point VPNs that can scale with carrier requirements. Managed VoIP VPNs can be implemented by making use of one of the two following network VPN models depending on the choice of the service provider.

Point-to-multipoint IP-VPNs, based on IETF RFC 2547 standards, can be used to build managed VoIP networks. These IP-VPNs should have the capability to support QoS at an IP-flow level. Alternatively, if the service provider has security concerns using layer-3 IP-VPNs for running VoIP, layer-2 Ethernet VPNs[ii] using Virtual Private LAN Service (VPLS) also could be used. VPLS facilitates support for point-to-multipoint switched Ethernet services over an IP/MPLS network. Either one of these multi-point VPNs are implemented in the edge routers at the periphery of the IP/MPLS backbone. All the earlier QoS requirements for transporting VoIP in a reliable manner also apply to this model. In addition, the edge routers also must have the ability to support QoS on a per IP-VPN or VPLS basis.

## Residential VoIP Network Model

High-speed Internet access using DSL via the ILEC copper plant or via the MSO cable plant is the fastest growing segment in today's service provider market. The primary application is always dedicated Internet access. Although both of these access technologies are asymmetric in bandwidth, the minimum upstream bandwidth that can be supported over these access plants is 128kbps. Within a given region, a group of access loops (both DSL and cable) are aggregated into IP termination platforms B-RAS in DSL networks and CMTS in MSO networks. These IP termination and aggregation platforms are then handed over to an ISP's network, or in case of some ILEC's they have their own ISP network to which the B-RAS is connected.

Broadband ISPs are looking into VoIP as an additional source of revenue, with the success of VoIP players such as Vonage. DSL-modem and cable-modem vendors are introducing products with integrated VoIP clients offering built-in RJ-11 ports and batteries for backups during power outages.

The end-to-end VoIP network includes a GR-303 gateway that terminates the VoIP session from the customer premises over the DSL/cable-modem connection. Multiple GR-303 gateways can communicate to each other over an IP/MPLS network for transporting on-net calls. If the call is determined to be an off-net call, the call is handed over to the locally connected PSTN class-5 switch via the GR-303 gateway. In this network model, the edge router that aggregates multiple B-RAS/CMTS is capable of supporting granular QoS and MPLS LSP with hierarchical traffic shaping and policing features. Packet cable reference architecture recommends MSOs build an IP/MPLS network capable of supporting end-to-end QoS. This model of VoIP network topology also is recommended in the planned massive FTTH roll-outs by the RBOCs.

## Problem Areas in Current IP Networks to Support VoIP

VoIP has stringent delay and jitter requirements, which are defined in ITU G.114 Standards, while the nature of Internet data traffic is bursty. Current router-based IP networks were built to carry best effort Internet data traffic. It is not economical for network operators to build a separate IP network to carry VoIP traffic. Presently, the majority of network operators are exploring how to merge discrete networks into one to optimize operational expenditures. Traditional IP backbones running an Interior Gateway Protocols (IGP), such as OSPF or IS-IS, make use of the shortest path forwarding logic and do not efficiently use the bandwidth resources across the networks. To thwart this inefficiency, service providers are migrating their IP networks to IP/MPLS networks that optimize traffic flows across pre-established routing paths known as LSPs. Service providers are in the process or planning to upgrade their IP backbone to an IP/MPLS backbone to facilitate a circuit-like architecture using MPLS LSPs.

Over the past three years, network operators have upgraded their IP backbone networks for higher speeds. Bandwidth throughput in the backbone is not the problem. Some of the large service provider IP backbone links are only running at 40 percent of their peak capacity. Traffic bottlenecks are found near the edge routers (e.g., ingress/egress routers), where multiple traffic types enter the IP/MPLS networks. The current edge routers are not engineered to support the required QoS to support both VoIP and Internet data concurrently in a scalable manner. The current generation of edge routers were developed 4 to 5 years ago when the primary requirement for routers was to forward bursty IP traffic across the Internet using best-effort metrics.

Network congestion is the primary contributing factor for voice-quality degradation. It is paramount to have an IP network that can delineate delay and jitter sensitive voice traffic from bursty data traffic. This delineation ought to be incorporated at both the ingress point to the network, i.e., at the edge router and within the backbone network. The edge router needs to have the capability to classify and prioritize the incoming IP packet and schedule the packets for processing in the correct order of priority. To accomplish these cumbersome tasks and to ensure that large data packets in the middle of the data pipe do not block small fixed-sized voice packets, it is absolutely critical that voice and data be put into different queues. To process VoIP traffic from thousands of voice calls entering and leaving the IP network, the edge router must have queues that can scale accordingly. Queues are integral to router hardware design and are built into the silicon. It is not possible to add queues by adjusting or writing new software at a later phase of development. The current generation of edge routers were engineered and developed to handle bursty Internet traffic, and at that time enhanced queuing architecture or queue size were not considered integral components of the router design and architecture. A QoS architecture similar to ATM switches is needed to have dedicated queues per traffic flow. This network evolution calls for a multi-service edge router that has carrier-class reliability and scalable QoS to support both real-time VoIP applications and best-effort Internet traffic.

## Tellabs® 8800 Multi-service Router (MSR) Solutions in VoIP Networks

Carrier-class reliability and scalable QoS are inherently critical requirements of VoIP networks. The current circuit switched PSTN has a phenomenal reliability of five 9's, and service providers will make every effort to make that a reality in future VoIP networks. It is imperative that IP/MPLS edge routers support the two aforementioned network attributes.

Today, Tellabs is the market leader in network products deployed on the edge of current circuit-switched voice networks and voice-quality enhancement products used for echo cancellation in end-to-end voice applications. Over the years, Tellabs has developed expertise in voice networks and can help network operators smoothly transition to VoIP. The Tellabs 8800 MSR is a purpose-built carrier-class edge router that can support the necessary packet classification and Quality of Service that VoIP networks demand.

## The Tellabs 8800 MSR Reliability

The Tellabs 8800 intelligent series of multi-service routers is designed so that the switchover time to backup resources is well within industry-specified ranges in the order of milliseconds. This ensures a non-disruptive service switchover by maintaining 99.999% availability of the system and its resources. The Tellabs 8800 MSR series can detect and correct any single point of failure in all hardware and critical software functions without the need for a two-system redundant deployment.

The combination of the following options makes the Tellabs 8800 MSR a reliable and a robust carrier-grade platform: hot swappable/online insertion and removal; hitless software upgrades/non-stop forwarding; redundant power supplies; redundancy at the interface level using APS or line card level using 1:N backup mechanisms.

The software architecture of the Tellabs 8800 MSR is both modularized and multi-threaded. Each software process is autonomous and independently run. A disruption in one of the software processes does not affect the rest of the system software processes. *(For example, if a software process running an OSPF instance is disrupted for some reason, BGP and other VPNs will continue to function without any interruptions).* The Tellabs 8800 MSR also has configurable software options such as MPLS backup-LSPs, OSPF and BGP graceful restart and will continue to add new and emerging reliability protocols.

## The Tellabs 8800 MSR Scalable QoS

VoIP streams demand atypical timing requirements and lower delay and jitter tolerance across IP/MPLS networks. Supporting end-to-end QoS requires optimizing and enhancing queuing strategies, connection-admission-control mechanisms, traffic-shaping and policing technologies, both at the ingress and within the IP/MPLS networks. The current generation of edge routers does a fine job in best-effort packet forwarding; however, it does not differentiate between different kinds of IP flows or multi-protocol traffic flows needed for VoIP applications.

The Tellabs 8800 MSR was designed with per-flow queuing to provide a highly scalable solution capable of deterministically managing bandwidth and customer SLAs for all flows. The hardware design consists of custom ASICs engineered to enable line rate inspection of every arriving packet.

Each individual flow can be configured to support any of the following QoS classes: CBR, VBR, VBR-RT and UBR+. In the case of a VoIP service, a CBR or a VBR-RT service class might be selected which in turn guarantees the bandwidth and latency of the service across the network. To guarantee the chosen QoS class on a per-flow basis, the Tellabs 8800 MSR was architected with a modified packet-based weighted, fair-queuing algorithm, scaling in support of 512,000 queues per slot.

## The Tellabs 8800 MSR IP CAC

The Tellabs 8800 MSR also supports the Connection Admission Control (CAC) algorithm for all IP flows. The CAC algorithms guarantee that the selected QoS class can be delivered within the bandwidth and traffic parameters selected. CAC utilizes the physical access-link bandwidth and QoS capabilities to determine if a connection can be allowed. In Multi-service IP/MPLS networks, the CAC function provides additional features that ensure traffic-engineered, MPLS-based LSPs can deliver the QoS across the network. The available bandwidth and the QoS priority of each traffic engineered LSP is signaled across the network using the standards-based extensions to IP-based routing protocols (i.e., OSPF TE and ISIS TE). Upon request for a connection setup across the network, the CAC checks both the available bandwidth and QoS on the access bandwidth and the LSPs that the connection will transit across the network. If the resources are available, the connection is accepted; otherwise, the connection request is rejected. The CAC algorithm uses the service specific traffic parameters and the particular QoS request to determine the requested sustained bandwidth. If accepted, this bandwidth calculation is decremented from the available bandwidth on both the access interface and LSP.

The CAC algorithm is very sophisticated in its ability to be customized on the basis of QoS, interface and services.

The Tellabs 8800 multi-service router provides the following features as part of its CAC process:

- Configuration of maximum peak, sustained and minimum rates allowed by CAC on a per-interface and per-service basis.
- Overbooking of interface bandwidth
- Link sharing percentage control of bandwidth allocated per QoS class
- Enable and disable CAC on a per-interface basis
- Ingress flow treatments

All flows in the Tellabs 8800 MSR can be configured as either ingress policed or ingress shaped. These two options facilitate support for existing as well as new Service Level Agreements (SLAs) to end customers.

## The Tellabs 8800 MSR Policing

Tellabs 8800 MSR supports industry standard policing mechanisms that match Usage Parameter Control (UPC) standards defined by the ATM Forum as well as the IETF Diff-serv RFCs. Traffic policing is an important part of adhering to SLAs. Policers can be applied on any configured flow and provide a Generic Cell Rate Algorithm (GCRA) based dual leaky bucket policing. Policing is defined relative to the selected QoS class and the respective traffic parameters.

At ingress, the policing mechanism in the Tellabs 8800 MSR utilizes a three-color scheme to mark those packets per-flow that either conform or violate their SLA traffic contract. Packets are marked green if they adhere to the SLA. Packets are marked yellow if they violate the maximum burst size and the traffic rate is below the peak traffic rate. Finally packets are marked red if the traffic exceeds the configured peak traffic rate. An option exists to explicitly drop all packets that violate their SLAs rather than color them red. The packet coloring scheme is used in the case of congestion with the per-flow congestion control mechanisms. Policing can be enabled or disabled on a per-flow basis.

## The Tellabs 8800 MSR Traffic Shaping

The Tellabs 8800 MSR supports per-flow traffic shaping at ingress as well as egress to the network. In particular, the Tellabs 8800 MSR network can shape end customers' traffic to a specific traffic contract in the event their particular CPE or protocol is unable to shape the traffic coming to the network. By using large input buffering schemes, the Tellabs 8800 MSR is capable of absorbing traffic bursts up to line rate and then paces or smoothes that traffic onto the network at the agreed SLA and traffic contract.

This ingress-shaping feature assures that the customer is only allowed to put the amount of traffic on the network that was contracted. It reduces packet drops, rather than using the policing mechanism that either explicitly drops or colors packets for discard in the case of over-bursting. It also benefits the service provider in managing the bandwidth that reaches the backbone, providing an additional mechanism to help maintain the QoS contracts over the switch or network overall.

Per-flow egress shaping is another useful feature offered by the Tellabs 8800 MSR used in smoothing out any potential bursts that the network or networks may create. This feature is beneficial in not over-running CPE ingress capabilities, carrier-to-carrier connections or other infrastructure applications where the Tellabs 8800 MSR egress traffic may be policed at the next ingress point. In addition to traffic shaping, the Tellabs 8800 MSR supports multiple congestion-control schemes that can be enabled and disabled on a per-flow basis.

## Conclusion

To successfully deploy VoIP it is imperative that service providers build a carrier-grade IP/MPLS network that can support multiple services and scale to support granular and coarse iterations of IP service classes required for the VoIP network architectures. In a study conducted by Infonetics Research[i] with service providers in 2003, ninety percent of the respondents said that they will ensure voice quality by enhancing their IP networks with QoS. Sixty-two percent of these service providers said they will use MPLS. It is not only evident from this study that service providers realize the need for QoS enhancements to their IP networks to make next-generation voice reliable, but it is also clear that MPLS- and QoS-enabled IP networks will emerge as the primary technology base for network infrastructure for VoIP. MPLS edge routers that can scale and support multiple services are paramount to a graceful transition from the circuit-switched voice network to a fully converged voice over IP network.

The Tellabs 8800 MSR series was designed, engineered and purpose-built as a multi-service edge routing platform to provide support for both bursty data traffic and delay- and jitter-sensitive voice traffic over IP networks. The platform has over one-half million available queues (512,000) built in hardware. Service providers can custom tailor the use of these queues for a mix of voice, data and video applications on both the ingress point from which these applications enter the platform and the egress point into the IP backbone network.

The Tellabs 8800 MSR series supports a wide range of interfaces from DS3/1/0, OC3, OC12, OC48, OC192, Fast Ethernet, Gigabit Ethernet and 10 Gigabit Ethernet. All of these interfaces not only support line rate forwarding of IP packets, but also multi-protocol formats that are prevalent in today's service provider networks such as Frame-Relay, ATM, X.86, SONET and 802.1Q Ethernet. The Tellabs 8800 MSR can support IP/MPLS VPNs using the IETF RFC 2547bis standards and traditional layer-2 VPNs using Frame-Relay, ATM and emerging Ethernet VPNs using IETF-draft-martini and VPLS standards. A wide array of mechanisms to create Virtual Private Networks gives a service provider the flexibility to choose the flavor of VPNs of its choice to deploy VoIP networks.

## Footnotes

1 H.248 is an International Telecommunication Union (ITU) standard that defines the "Gateway Control Protocol." In the joint collaboration between the ITU and the IETF, it also is referred to as MEGACO in IETF RFC 2885. Both of these standards define a centralized architecture for creating VoIP applications. It should be noted that H.248 extends the MGCP specifications.

2 H.323 is an ITU standard that defines a distributed architecture for creating VoIP applications.

3 Session Initiation Protocol specified in IETF RFC 2543 defines a distributed architecture for creating VoIP applications.

4 Media Gateway Control Protocol specified in IETF RFC 2705 defines a centralized architecture for creating VoIP applications.

5 Signaling System 7 is an out-of-band packet-based network used for signaling and call control in the present PSTN.

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ii Halabi, S. (2003) *Metro Ethernet Applications.* Cisco Press

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74.1465E Rev. A 03/04

## Tellabs® 8800 Series of Intelligent Multi-Service Routers

### Superior next generation intelligent multi-service routers

Tellabs delivers a series of intelligent multi-service routers for next generation multi-service networks. Tellabs believes multi-service packet- and IP-based networking is the natural migration path for today's IP-based networks and legacy overlay networks. The Tellabs 8800 MSR enables connection-oriented network characteristics such as QoS and security with our powerful traffic engineering capabilities and multi-label tagging algorithms while maintaining the superior scalability and flexibility of pure IP networks.

The Tellabs® 8820 multi-service router (MSR) is a high-performance networking platform supporting both emerging and legacy carrier services simultaneously. Placed within the service provider network or a service provider-owned premise location, it serves as a multi-service creation and aggregation platform. The Tellabs 8820 MSR provides 16 Gbps of port density, occupies only 3RU of telco rack space and consumes minimal power.

The Tellabs® 8860 multi-service router combines Layer 2 switching and QoS capabilities with the flexibility and intelligence of Layer 3 routing. The Tellabs 8860 MSR is a carrier-class, high-speed networking platform offering business-class IP, Frame Relay, ATM, Ethernet and TDM/Private Line services. It combines unprecedented performance with a greatly simplified "build once, sell many times" provisioning paradigm to significantly increase profit margins as soon as deployed. Service providers can reduce capital expenditures by offering a broad range of services with fewer network elements.

The Tellabs 8860 MSR combines both IP-routing and time-tested ATM-based Quality of Service (QoS) contracts with the efficiencies of MPLS traffic engineering. Providing superior scalability with over 6000 DS-3s in a rack, it scales to 320 Gbps non-blocking full-duplex performance in the shelf and provides investment protection through the ability to scale in a multishelf configuration to 2.5 Tbps.

The Tellabs® 8890 management system is a sophisticated software application suite that provides advanced multi-service domain management, and is architected to scale as you grow. It manages both the Tellabs 8820 MSR and the Tellabs 8860 MSR.

### Tellabs. The future of your business. Starting now.

Tellabs (NASDAQ: TLAB) delivers technology that transforms the way the world communicates. Tellabs experts design, develop, deploy and support our solutions for telecom service providers in more than 100 countries. More than two-thirds of telephone calls and Internet sessions in several countries, including the United States, flow through Tellabs equipment. Our product portfolio provides solutions in next-generation optical networking, managed access, carrier-class data, voice quality enhancement and cable telephony.