The Revolution in Telecom Networking:
Using IP Trunking to Maximize Value Creation

Executive Summary

This white paper presents a perspective on the advantages of native Internet Protocol (IP) interconnections for voice and how this is advancing the interconnect model with additional features and reduced costs at comparable quality to legacy methods.

More for Less

Since the end of the last millennium, the focus of the global telecoms industry has been on understanding and leveraging two revolutionary Internet Protocol-enabled services; broadband Internet access for consumers and Voice over IP (VoIP). Even though both of these technologies have roots that extend back into the 1990s, it wasn't until 2000-2001 that broadband adoption reached its tipping point and quickly became the predominant Internet access method and 2003-2004 when VoIP emerged first as a long-distance transport supplement and then as a viable replacement for the Public Switched Telephone Network (PSTN).

Today, with an estimated 56.5 million subscribers in the US and another 124.5 million outside the US and growing at the rate of 9-12 million per quarter[1], broadband has become one of the most quickly adopted technologies in history with VoIP hot on its heels[2].

Accompanying broadband and following VoIP dozens of other innovations have improved and enriched the "Internet Experience":

- More powerful and feature packed Web Browsers
- Melding of voice and video into previously mundane Instant Messenger (IM) services
- Peer-to-Peer file transfer
- Truly viable Web-based commerce in many shapes and forms
- The Google™ and Skype™ revolutions
- Web 2.0 with its powerful social networking and collaboration services

A common thread between these innovations is value: users are getting more and paying less. Businesses, which resolve to figure out how to deliver increasing levels of value by using newer technologies, will aggregate both users and revenues, directly attacking the financial strength of incumbent providers of competing services delivered over legacy technologies.

Obviously, in this highly competitive environment, innovation cannot come with a high cost. Service Providers (now including VoIP-based carriers along with traditional wireline and wireless carriers), eCommunities and Enterprises must rapidly deliver new services to their users while simultaneously driving down all associated costs: design, development, delivery, implementation and total cost of ownership:

- Service Providers who deliver on price, Quality-of-Service (QoS) and feature functionality will take market share.
- eCommunities that incorporate advances in communications functionality will be able to retain users and grow their base, making them a more attractive outlet for advertisers.
- Enterprises able to leverage these innovations to increase productivity or improve customer interaction will find "free money" that can be redirected to achieve other business objectives.

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[1] From "OECD Broadband Statistics to June 2006" study by the Organisation for Economic Co-operation and Development (OECD), Paris, France (Friday, October 13th, 2006).
[2] See "Next Generation Network Development in OECD Countries" declassified by OECD in October 2004.

Less is More

To create value, Service Providers, eCommunities and Enterprises must focus on costs and more often than not, simplification and standardization are the answers. Simplification makes products easier to deploy and operate while standardization improves extensibility and interoperability. For instance, VoIP services delivered via IP transport and SoftSwitch technology in lieu of TDM transport and Circuit Switches have delivered features and savings while at the same time simplifying the service's underlying infrastructure. This shift towards a less complicated service platform has lowered the barriers to entry into the voice space and made sustaining operations more economical, thus allowing innovators to pour resources into new capabilities.

In this way, VoIP has delivered value the incumbents are unwilling or unable to deliver and does so without increasing prices or sacrificing quality. In fact, according to testing equipment vendor Minacom (a Tektronix® company), service quality and network availability of IP-based networks now rivals the legacy PSTN in many key measures: "***VoIP phone service now sounds better and connects faster than the standard public switched phones network*** (PSTN)"[3], while simultaneously providing a platform unavailable using the conventional technologies of the 20th century.

What is H.323?
H.323 is an "umbrella recommendation" developed by the ITU-T for the purpose of standardizing multimedia conferencing over packet-switched networks. Common applications include voice and video over IP. Common generic network element terms used with this standard are Terminal, Multipoint Controller (MC), Multipoint Processor (MP), Gateway Gatekeeper and Border Element. H.323 is a mature, well-supported standard by many equipment vendors and used around the world.

What is SIP?
SIP is an application layer protocol developed by the IETF for the purpose of "creating, modifying, and terminating sessions with one or more participants". Common applications include voice and video over IP with growing use for instant messaging. Common generic network element terms used with this standard are: User Agent Client (UAC) / User Agent Server (UAS), Back-to-Back User Agent: (B2BUA) Proxy Server, Registrar, Server Redirect Server and Feature Sever. SIP is an evolving, but extremely popular standard supported by many equipment vendors and used around the world.

For more details, see ***IP Trunking Reference*** section.

But there's more to this story. During much of VoIP's youthful, formative years, there were artificial boundaries raised to prevent a complete end-to-end implementation from coming to the fore ground. Equipment vendors implementated proprietary platforms or highly customized implementations of open standards. Large incumbent carriers maintained their fixation on legacy technologies and employed next generation architectures only in the network core, if at all. Providers of enterprise voice solutions focused on solutions for the private LAN / WAN environment.

These conditions created an environment where VoIP was marginalized as a niche technology for targeted applications or early adopter hobbyists. It wasn't until broadband adoption reached critical mass and the stampede towards Consumer VoIP services began that the pieces began to fall into place. As start-ups popped-up with paid "over-the-top" offerings and the major Internet Service Providers (ISPs) released free PC-to-PC voice to their subscribers, these new entrants matured VoIP into an end-to-end service delivery capability driven by the need to:

- Simplify - Make VoIP an application instead of another network service. Use IP transport to the edge. Take advantage of smart edge devices.
- Standardize - Avoid the proprietary; embrace open standards.

As with many revolutions, the instigators are often not well known. In the case of VoIP, it was two open standards put forth by competing standards bodies that were key triggers: H.323 (first released in 1996 by the ITU-T, representing the traditional telecoms industry) and SIP (first released in 1999 by the IETF, representing the upstart Internet community). Although coming from different points of view, both standards set out to describe ways and means for managing multimedia communications sessions over IP networks. Both standards leverage the IETF's RTP protocol, which describes how streaming media should be encapsulated for transport across IP networks. By embracing these standards and the network architectures they were meant to support, the new entrants to the telecoms market were able to realize the benefits of simplification and standardization that changed the dynamics of the industry.

[3] From "Internet Phone Quality Improves Significantly and Steadily over the Last 12 Months" press release by Minacom, Montreal, Canada (Monday, August 28, 2006).

Best of all: it's not over. In the future, all networks will interoperate using IP end-to-end to deliver on the promise of better-cost and richer features, including those presently in development as a part of the next-generation communications architecture called IP Multimedia Subsystem (IMS).

Unfortunately, many Service Providers, eCommunities and Enterprises do not yet fully understand the value that can be created by connecting with users and network suppliers using the current generation of IP interconnection architectures. Instead, they have continued to operate using the past century's capabilities (and low value delivery) of the PSTN infrastructure. In some cases, this is because purchasers have selected Carriers that are unable to support direct IP interconnections. In other cases, purchasers are intimidated by the new technology or think that because of their current infrastructure they cannot take advantage of direct IP interconnections without significant human and capital investment.

IP Trunking: The Flexible Choice

What Service Providers, eCommunities and Enterprises need is a flexible solution that allows them to fully leverage their next-generation IP-enabled infrastructure or, if necessary, re-use their existing legacy infrastructure. With IP Trunking, it is possible to replace multiple network links dedicated to TDM voice or IP data with a single IP connection: the IP network continues to carry e-mail, web browsing and other data traffic to and from the Internet as it did previously, but now voice (and even video) is added to the mix as another IP-enabled application.

SIP or H.323 is used to signal voice sessions to and from users supported by Service Providers, eCommunities and Enterprises, and RTP transports the media. Internal calls can continue to be handled as in the past, while external calls to the PSTN or other IP destinations are handed off to the IP Trunking vendor where they use that Carrier's VoIP network to reach their destination. In the case of a call requiring PSTN termination, a Media Gateway converts VoIP to TDM before being handed to a local service provider for the last mile carriage to the called party. In the case of IP termination, the voice traffic remains a VoIP session all the way to the ringing handset or terminal.

While simplifying interconnection architectures, IP Trunks also offer significant financial benefit. By eliminating low-capacity network links with multiple-capacity IP connectivity, base network costs are reduced while at the same time Service Providers, eCommunities and Enterprises gain access to high-quality voice transport and termination capability. Since calls to the PSTN are carried across the Carrier's IP backbone to the Media Gateway with the most economical connection to the PSTN destination, the per-minute toll charges on these calls are greatly reduced due to elimination of numerous "middle men" (each levying their charges) between session originator and terminating end point. Sounds great, but what if you do not have an IP-enabled network or IP-PBX? Some Carriers offer a Hosted Media Gateway option that allows continued use of TDM-based and PRI-connected voice networking gear while exposing the benefits of IP to system users.

IP Trunking: A Buying Checklist

However, not all Carriers supplying IP Trunking today are offering the complete solution. There are a number of areas where IP Trunking, if not properly implemented, can put user service quality at risk. When shopping for a Carrier, Service Providers, eCommunities and Enterprises need to look for a partner with an understanding of, and network solutions for:

- Capacity Management – Measure and control capacity at several levels: overall network, trunk group (or equivalent), device interface and individual trunk equivalent to prevent gross over subscription and possible congestion.
- Quality-of-Service (QoS) – Able to measure, detect and react to issues at Layer 3 (packet jitter, packet loss and latency) and above (audible call quality, traffic congestion and signaling / call flow issues).
- Interworking – Support for a variety of IP-PBX, IP-Centrex (aka Feature Server), SIP Proxy, Session Border Controller (SBC) or SoftSwitch implementations using SIP or H.323.
- VPNs – Support for secure VPN connections into the Carrier network when required.
- Service Level Agreement (SLA) Management – Service provider needs to offer an integrated SLA for both data and voice services.
- Security – Familiar with spoofing, call fraud, eavesdropping, malicious call attacks (aka SIP, RTP or UDP DoS / DDoS) and VoIP VPN integrity preservation as well as measures in place to protect against all, thus ensuring high availability of IP Trunking and protection of traffic.
- Rapid Activation – Scalable processes and systems that allow the Carrier to be responsive to rapid traffic growth by quickly and easily augmenting IP Trunking capacity in the right places at the right times.
- VoIP and TDM Interoperating Expertise – Familiar with and successfully deployed networks using the latest in industry standards for VoIP. Strong candidates should have incorporated peering and addressing technologies such as ENUM along with the ability to transcode IP traffic and map and route traffic between IP and TDM domains.

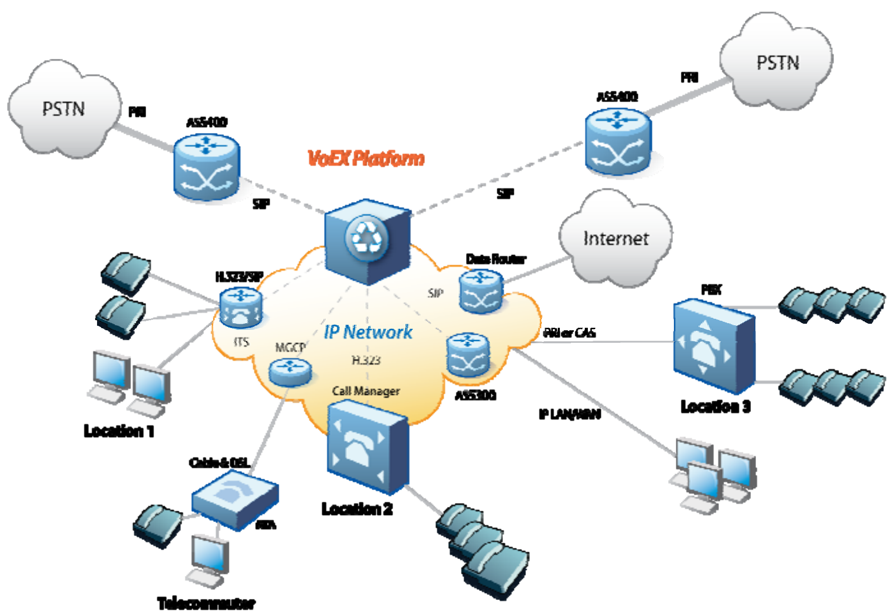
Beyond these basic “table stakes,” the ideal partner should also be comfortable and capable of supporting a true end-to-end managed service (CPE and access link), as well as provide its customers self-directed visibility to service performance and billing information.

The VoEX™ IP Trunking Offering: The Ideal Partner

The VoEX™ IP Trunking offering allows communications Service Providers (traditional and next-generation carriers), eCommunities and forward-thinking Enterprises, to realize the many cost and QoS benefits of direct network-to-network interworking while setting the groundwork for support of future IMS-based applications.

The offering combines VoEX™ SuperRegistry™ open-standards Carrier ENUM directory technology with a global carrier-grade IP peering and TDM interconnect infrastructure that enables customers to originate, terminate and share calls or sessions for mobile, fixed, and broadband communications. Calls can originate or terminate over IP or TDM connections (or a Hybrid combination of both) and make use of several popular signaling protocols. Since these capabilities are delivered as a complete service, they easily integrate into any network to deliver immediate cost savings along with the increased capabilities of an IP-based platform.

Any-to-Any Protocol and Device Support Platform



Another important aspect of the solution is edge-to-edge connectivity using IP transport. Through VoEX™ -provided VoIP gateway hardware installed on the customer premise, IP Trunking and SuperRegistry™ services support the conversion of legacy TDM protocols to IP traffic for switching and transport. This offers proven any-to-any interoperability for all carriers and call flows: TDM-IP, TDM-TDM, IP-TDM or IP-IP.

In addition, the SuperRegistry™ platform will allow service providers to create new sources of revenue by deploying innovative SIP-based services such as video, presence, and push-to-talk.

IP-Trunking™ Features and Benefits

Global Reach: VoEX™ provides highly available and reliable services worldwide through our network of business relationships offering a virtual "Tier 1" grade service: reliable, low-latency setup of calls and sessions at a very low cost.

High Quality Service Offerings: End-to-end IP service path management and monitoring with proprietary intelligent routing algorithms assures PSTN-equivalent or better performance between VoEX™ and peer networks.

- High Quality – Real-time QoS Management / Network Optimization
- VoIP Connectivity with PSTN equivalent uptime
- Interoperability with more signaling protocols and support for more codec's than any other provider
- Proprietary Routing Intelligence Discovery Engine (PRIDE) which considers source device protocol, device compression engine, Fax / Modem protocol and Class-of-Service (ESP vs non-ESP) to enable LCR or BQR with only matching qualified vendors.
- VoIP traffic also receives IP QoS service prioritization within VoEX™'s global MPLS network queues (Control Traffic Queue, Voice Queue, Enhanced Queue and Best Effort Queue)
- VoEX™ SuperRegistry™ offering provides Service Providers, eCommunities and Enterprises a means to drive additional savings on their outbound traffic by enabling delivery of inbound calls using VoIP (with conversion to TDM as needed).

Rich Interconnection Capabilities: Highly flexible platform offering an array of interconnection options:

- Options for TDM or IP
- Support for SS7 / C7, ISDN (Q.931), CAS, H.323, H.245, MGCP and SIP
- Support for multiple flavors of G.711, G.723.1, G.726, G.729 and GSMFR

Low Cost: VoEX™ use of IP provides significant savings over traditional PSTN transport and termination offerings and rates. For a typical midsized US regional service provider with less than 2 million subscribers, we estimate realized savings of US$20 million over three years.

- On-Net interconnection via the world's largest Peering and Carrier ENUM solution powering cost efficient IP-IP and IP-PSTN interworking
- Off-Net interconnection for over 1 million unique routes optimized through LCR or BQR algorithms including native LNP dips to guarantee the best routing

Service-based Offering: Everything is available as a complete managed services solution to conserve your investment in human and financial capital making this a "zero risk" choice. Along with no additional capital investment, there are no upfront fees or membership charges. VoEX™ allows flexible inter-working using either IP or TDM protocols for speed in implementing your solution by connecting to what you already have in place.

- Signaling and media transport via an extensive mesh of routing relationships
- VoEX™ -provided local VoIP Gateway hardware included with every solution as the network interface to the SuperRegistry™
- Customized private dial plans available
- Extensive VoIP expertise transfer through Consulting and Integration Services
- Managed service model with low initial investment that can be accounted for as OpEx (vs CapEx)
- Security, audit control and billing with Online Tools
- "Capacity on demand" and multiple footprints for global connectivity

Extensive Knowledge: The VoEX™ team has been working with leading Product and Service Providers to create additional value around IP-based open standards design and interfaces.

IP Trunking Reference

Acronyms Used in this Paper

BQR	Best Quality Routing
CLEC	Competitive Local Exchange Carrier
CPE	Customer Premise Equipment
DoS	Denial-of-Service attack
DDoS	Distributed Denial-of-Service attack
DNS	Domain Name System
ENUM	T**E**lephone **NUM**ber Mapping or **E**164 **NU**mber **M**apping or **E**lectronic **NUM**bering
E.164	The International standard for defining PSTN telephone numbers
ESP	Enhanced Services Provider
H.323	An ITU-T standard for managing sessions on packet networks
IETF	Internet Engineering Task Force
ILEC	Incumbent Local Exchange Carrier
IMS	IP Multimedia Subsystem
IMT	Intermachine Trunk
IP	Internet Protocol
ITU-T	International Telecommunication Union - Telecommunication Standardization Sector
LCR	Least Cost Routing
LERG	Local Exchange Routing Guide
MPLS	**M**ulti**P**rotocol **L**abel **S**witching
PSTN	Public Switched Telephone
RTP	Real-time Transport Protocol
SIP	Session Initiation Protocol
TDM	Time Division Multiplexing
VPN	Virtual Private Network

Recommended Reading

You can learn more about H.323 @ ITU-T Home Page and through review of the current ITU-T standard describing H.323:

- H.323 System Implementors' Guide

You can learn more about SIP @ IETF SIP Working Group Status Pages and through review of several IETF Requests for Comment (RFCs) describing SIP:

- RFC 3261 – SIP: Session Initiation Protocol
- RFC 3550 – RTP: A Transport Protocol for Real-Time Applications
- RFC 3824 – Using E.164 numbers with the Session Initiation Protocol (SIP)
- RFC 4566 – SDP: Session Description Protocol

3rd party information about both H.323 and SIP is abundant and readily available on the Internet. Two excellent and reputable resources are http://www.h323forum.org/ and http://www.sipforum.org/

Also, to better understand ENUM, please review VoEX™'s white paper on ENUM entitled "Bridge Over Troubled Waters: An Overview on ENUM for Connecting VoIP Islands". That white paper presents a perspective on the current state of VoIP interconnectivity and proposes a solution for advancing the interconnect model with improved quality and reduced cost.

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About VoEX™

VoEX™, Inc. is a global IP managed service provider using a pure VoIP peering infrastructure and an advanced carrier grade ENUM and phone number registry connecting Any-to-Any TDM, IP and Hybrid carriers. With the migration to IP-based telephony, carriers as well as private exchanges (such as business PBXs, universities and call centers) increasingly demand interconnect services comparable in quality and scalability to those of traditional PSTN LECs. VoEX™'s software-based interconnect network enables current LECs, wireless carriers, cable providers, PSTN private exchanges, and VoIP-based private exchanges to realize dramatic price reductions, find additional revenue opportunities and deploy advanced inter-party application functionality not offered by traditional carriers. Depending on a customer's on-premise equipment, VoEX™ interconnects either directly to existing IP-enabled hardware or via a media gateway provided and managed by VoEX™. VoEX™ customers, including the largest global IP-based voice communities as well as Top Tier carriers, eliminate domestic and international long distance charges. Furthermore, VoEX™ enables private exchanges and edge service providers to earn access fees from inbound long distance calls creating newfound revenue streams. On-demand software solutions include web-based billing, network management, and customer usage analytics. For VoIP-based carriers and private exchanges, VoEX™'s interconnect platform resolves incompatibilities between IP PBX systems, including interconnection, signaling, multiple codec support, media translation, and gateway services for advanced features (such as shared voicemail, directory services). VoEX™ offers universal call completion capabilities with support for both SIP signaling to IP endpoints and SS7 signaling for PSTN call termination. Finally, for VoIP carriers seeking to deliver application specific enhanced telephony applications (such as presence, mobility and email & contact management) VoEX™ provides custom developed services and a Web Service-based application development platform.

Pure VoIP