Communications Transformations 2: Steps to Integrate SIP Trunk into the Enterprise

The Changing Landscape

IP-based unified communications is widely deployed in enterprise networks, both for internal calling in the campus and between branch offices of the sites comprising the network. Such enterprise networks increasingly use IP-capable user endpoints (IP phones or softphones), and although interconnections of older time-division multiplexing (TDM) private branch exchanges (PBXs) and key systems through voice-over-IP (VoIP) gateways still exist in many networks, adoption of IP user endpoints in the typical enterprise network is increasing rapidly.

Access to the public switched telephone network (PSTN) from the enterprise network is, however, still predominantly TDM-based. Typically located at each network site, VoIP gateways provide connectivity from IP user endpoints to the traditional PSTN.

The next stage in advancing business communications, which is beginning to occur in enterprise networks, takes unified communications traffic destined beyond the enterprise also to IP. Similar to enterprise networks, service provider networks have widely deployed VoIP inside their own networks and although the enterprise's PSTN interconnection points are still almost exclusively TDM-based, the PSTN backbone has in many cases already deployed VoIP traffic. This situation makes it technically possible to use VoIP also as the method of voice interconnection between enterprise and service provider networks. As unified communications Session Initiation Protocol (SIP) trunk offerings from service providers mature over the next several years, IP interconnection for unified communications presents a new business opportunity for an increasing number of enterprises.

Benefits and Implications of IP Interconnection

Integrating an IP interconnection through unified communications SIP trunks from the enterprise for calls destined beyond the enterprise provides both a new mechanism to connect to traditional PSTN endpoints and access to new services and applications not possible with TDM interconnections and endpoints.

The benefits of a unified communications IP interconnection into the enterprise include:

- New services and applications, the result of almost unlimited bandwidth on IP trunks and end-to-end IP connectivity; the new services include:
    - Inter-enterprise rich-media collaboration applications
    - High-fidelity voice, enabled by wide-band codecs
    - High-fidelity video
    - Presence
- Increased worker productivity as a result of application convergence, traditionally encompassing only other enterprise colleagues, but now also with vendors, consultants, and customers
- Simplified provisioning and management of dial plans within the enterprise

- Simplified capacity addition

Benefits of accessing traditional PSTN endpoints and services through an IP interconnection include:

- Alternative physical access methods, including cable, DSL, and wireless
- Scalable capacity addition because the call volume on a unified communications IP trunk is not constrained by the availability of time slots
- Less dependency of the interconnection point for services (such as local calling or long-distance calling) on the physical point of access into the service provider's network

To realize these benefits, your enterprise should carefully consider how and when to integrate a unified communications IP interconnection (a SIP trunk) into the internal network. Adding an IP interconnection for accessing the PSTN and new unified communications services involves more than simply configuring a SIP trunk from the Cisco® Unified Communications Manager to the service provider's IP connection point. You should consider several network design and implementation variables when adding this new access method, including:

- Location and number of unified communications IP interconnection points from the enterprise
- Dial plans and call routing, including how emergency calls are handled
- Security
- High availability
- Call-traffic capacity and bandwidth regulation, monitoring, and control (Call Admission Control [CAC])
- Interconnecting different VoIP protocols, vintages, and implementation variations (such as early offer and delayed media)
- Interconnecting myriad VoIP and IP-video media encodings, including dual tone multifrequency (DTMF), Real-Time Transport Protocol (RTP), codecs, and transport of fax and modem traffic
- Interconnecting myriad unified communications IP endpoints of widely varying capabilities
- Troubleshooting and billing tools and methods

These network design and implementation variables, and Cisco solutions to address them, are discussed in more technical detail in the white paper "Communications Transformations: Implementation Considerations when Enhancing Enterprise Communications Solutions with SIP Trunks", please visit http://www.cisco.com/go/cube.

Steps to Integrate IP Interconnect

Integrating an IP access method (SIP trunk) for unified communications between your enterprise network and destinations external to your network involves the following steps:

- Evaluate new services (for example, high-fidelity voice and video) and rich-media collaboration applications that would become possible over unified communications SIP trunk access and how, when, and where these services would extend your business opportunities or enhance worker productivity.
- Evaluate unified communications SIP trunk offerings from the providers in the geographies in which your network operates. A cost analysis of these offerings will determine whether you should start with a centralized model (a single SIP trunk into the enterprise for

potentially limited call patterns to a subset of your user base), or a distributed model (multiple SIP trunks into different sites with “local” services for each site on its own SIP trunk).

- Evaluate media encoding methods (such as voice and video codec choices and DTMF relay alternatives) of the unified communications SIP trunk offering and how they will fit into your enterprise policies. Plan deployment of transcoding or DTMF conversion resources at the appropriate points in your network.
- Determine which users and call patterns will use the unified communications SIP trunk, and what phases of usage patterns make sense (perhaps initially only users at the site co-located with the unified communications SIP trunk will use it, and perhaps only for long-distance calls; or perhaps only contact center agents will use the unified communications SIP trunk, and not the general business users in your network). This decision will determine the call-routing and dial-plan changes you should enter into your network configuration. You could develop several phases to this plan as you expand the pool of users or sites that can access the unified communications SIP trunk, and as you adjust CAC policies between sites to take calls across your WAN to a centralized unified communications SIP trunk entry point.
- Determine how traffic from traditional TDM applications, such as fax, modem, point-of-sale credit card authorization, alarm monitoring, and telemetry, can potentially be carried over SIP trunk access. Some of these applications may continue to use your existing PSTN gateways until IP interconnection offerings reach greater maturity.
- Consider redundancy and availability of services to the users with unified communications SIP trunk service. Discuss failover and load balancing with the service provider and ensure that sufficient measures (such as dual hardware platforms and dual physical terminations) are in place on the enterprise network interconnection points. Maintain call-routing configurations to your existing PSTN gateways for additional backup access.
- Determine the point of demarcation between your enterprise network and the service provider’s network. Discuss what methods and tools you will use to troubleshoot voice-quality complaints from your user community and how you will isolate problems to either your network or the provider’s. Also determine how you will assess billing and how you will reconcile the provider’s billing records with those from your network. It may be useful to enable billing at the demarcation point for these purposes.
- Conduct a testing or certification effort with an enterprise-owned session-border-controller device, such as the Cisco Unified Border Element, and the provider’s offering to ensure interworking for the call flows important to your network, and to mask off the security, traffic regulation, and media interworking considerations of the unified communications SIP trunk from the rest of your enterprise communications servers and endpoints.
- Plan security measures for every point in your network at which a unified communications SIP trunk terminates. Determine what security measures the service provider offers on traffic on this trunk, and which Cisco Unified Border Element features you will use to provide traditional “data” IP traffic protection (such as denial-of-service attacks and firewalling), as well as extra protection specific to voice and video traffic, such as enterprise SIP address hiding.

- Review quality-of-service (QoS) and CAC policies in the network to ensure coverage of new call flows. Potentially, some calls from remote sites will now cross the WAN to use a centralized unified communications SIP trunk on a campus site. Ensure that codecs that violate your QoS or CAC policies are either disallowed or transcoded by the Cisco Unified Border Element.
- Determine call routing for emergency calls to ensure that number and location information is correctly delivered to emergency authorities within the local laws of each site. Discuss the routing of these calls with your unified communications SIP trunk provider and, if appropriate measures are not in place, continue to use your traditional PSTN gateways to provide emergency call access.

Recommendations to Integrate IP Interconnection

Although different phases of deployment must meet your individual business needs and will vary from one implementation to another, some typical deployment stages are common:

- Evaluate the network design and implementation factors in the previous section.
- Implement a pilot deployment of a campus or large-site user pool (such as a contact center), using the unified communications SIP trunk for long-distance or toll-call traffic.
- Add unified communications SIP trunk access for the campus or large-site user pool for local call traffic.
- Add unified communications SIP trunk access for the user pool of some or all remote offices in your network, using the SIP trunk for all long-distance or toll-call traffic.

Upgrading a TDM Gateway to Enable Unified Communications IP Trunking

Enterprise networks typically have traditional PSTN gateways at every site. For flexibility, high availability, traditional applications, the phasing in of new services, and emergency call access, these PSTN gateways will remain in your network as unified communications SIP trunk access is added.

Using a Cisco Unified Border Element to terminate a unified communications SIP trunk into your network is recommended for all the reasons discussed previously. A Cisco Unified Border Element is a software function that can be deployed on the same platforms that currently act as your PSTN gateways. Enhancing an existing PSTN gateway to act also as a Cisco Unified Border Element requires the following:

- Cisco IOS® Software upgrade (only certain images support border-element operation)
- Cisco Unified Border Element license

Summary

Enterprises are beginning to evaluate unified communications SIP trunk access for calls destined beyond the internal network, allowing access to new services and productivity applications, but also having implications on your network design that you should carefully consider and plan for. Establishing an appropriate demarcation point between your network and the service provider's network is critical to ensuring secure and predictable continuation of communications services to your user community.

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