SIP Trunking Deployment Models:

Choose the One That Is Right for Your Company

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Executive Summary

Transitioning to Session Initiation Protocol (SIP) trunking is a strategic decision for enterprise organizations. If implemented correctly, the SIP trunk architecture can provide a solid foundation for today with flexibility to adapt more easily and cost-effectively to tomorrow's collaboration needs. This white paper will discuss various SIP trunk deployment models and factors that should be considered when evaluating and designing a SIP trunk voice network.

In the past few years, innovations in collaboration services have delivered significant improvements in employee productivity. By every indicator, new collaboration services will continue to be introduced, providing additional benefits to the enterprise. Although no one can predict exactly what these new services will be, we expect them to incorporate existing fundamental elements of collaboration like voice, video, mobility, instant messaging, and virtual desktop technologies.

However, the problem facing most businesses is that their underlying network resources were not designed to support these new services. For example, most networks are not designed to support a significant volume of real-time video communication. However, real-time video increasingly is becoming a critical element in these new collaboration services. This problem will worsen as these collaboration use cases begin to move beyond the enterprise to suppliers and customers.

To address this problem, service providers are offering real-time communication services for voice and video communication delivered through their packet-based broadband services. The network capabilities needed to support these new real-time services are based on a network protocol technology referred to as Session Initiation Protocol (SIP), which enables virtual connections from the business user to the service provider over SIP trunks. Businesses are starting to replace their time-division multiplexing (TDM) trunks with SIP trunks to take advantage of cost savings and a greater variety of collaboration services.

SIP trunk services are inherently flexible in how they can be deployed, and they follow three basic architectural models: centralized, distributed, and hybrid. For a business focused on voice services, any of these deployment models can be used. But, for businesses trying to take advantage of more advanced collaboration services, the distributed or hybrid SIP trunk deployment models offer advantages. With this in mind, businesses also need to look beyond today and consider which architecture will give them the flexibility to evolve their network easily as their collaboration needs change. This white paper discusses the factors that businesses should consider as they evaluate SIP trunking.

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Collaboration Technologies: Enhancing Productivity and Growth

New collaboration technologies are enabling people to use an increasing array of options and allowing them to conduct complex, multiparty business communications with surprising ease. Only a few years ago, many such collaborative interactions would have been difficult or even impossible. These technologies are helping to break down communication barriers to accelerate problem solving and decision making and improve employee productivity.

Because collaboration services are constantly improving, it is difficult to predict how they will affect the network infrastructure required to deliver these services to the end user.

To understand the dilemma, it is worth recalling a few innovative services that have only recently been introduced and are growing rapidly in their adoption. These services include high-definition telepresence, hosted conferencing services such as Cisco WebEx® conferencing, mobile smart phone integration into corporate voice and data networks, and hosted telepresence conferencing such as Cisco TelePresence® Callway. Each of these services is putting pressure on IT departments to reevaluate their network design.

An additional challenge for IT departments as they plan their network is building in flexibility to cost-effectively transition the network to support future collaborative services. Today, we can only predict that these "future" services will likely involve some combination of video, mobility, voice, instant messaging, presence, and desktop sharing, with one-to-one, one-to-many, or many-to-many user interactions. But at this point their exact effect on the network is difficult for IT departments to assess.

Despite these uncertainties, IT departments are gaining confidence in one aspect of collaboration services: SIP trunking. SIP is now the principal technology for distributing real-time communications over IP networks. The transition from TDM to IP communications that began in the private network more than 10 years ago with IP-private branch exchange (PBX) technologies is now underway in the service provider network through SIP trunk services. As such, both inter- and intra-company voice and video communications can now share the same digital protocols for reliable end-to-end communication across contiguous IP networks. Therefore, SIP trunking has become a critical enabler of collaboration services by extending voice and video real-time IP communications between the private network of the business and the service provider public network.

Role Of Session Border Controller In SIP Trunk Deployments

Regardless of which SIP trunk deployment model is selected, businesses will need a new gateway function performed in their network that is different from the traditional TDM gateway. SIP trunking requires a network function that adapts or translates differing VoIP formats between the enterprise network and the service provider network. This new gateway function is called a session border controller (SBC).

The SBC provides an adaptive interface between the corporate user's internal IP voice network and the service provider IP PSTN. It is essential in enabling real-time voice and, eventually, video connectivity from the enterprise IP network to service provider SIP trunks. The SBC performs four essential functions: session control, security, interworking, and demarcation. These functions have the same purpose whether used in a centralized, distributed, or hybrid SIP trunk deployment architecture.

There are two varieties of SBC devices: standalone and router-integrated SBC devices. The router-integrated SBC simultaneously supports IP routing, IP firewall, and SBC functions. Because the SBC relies on a router for Layer 1 to Layer 3 connectivity to the service provider network, and because the router relies on the SBC to manage the real-time communication signaling with the service provider, in many customer deployments it makes sense to have these complementary functions performed in the same hardware platform. In fact, there are numerous advantages to having the SBC integrated with the router, such as the ability for the SBC to have better awareness of the bandwidth available on a router port connected to a broadband service, thereby enabling more sophisticated Quality of Service (QoS) and Call Admission Control (CAC) mechanisms.

An additional nuance is that the router-integrated SBC typically can be configured to function as standalone SBC, but the standalone SBC is not capable of performing the function of a router.

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How SIP Trunking Will Affect Your Network

When an IT department evaluates SIP trunk services, it is important to assess how deployment of these services will affect both the voice and data network architectures.

The voice network will experience significant changes as a result of the deployment of SIP trunk services. The physical connections supporting Primary Rate Interface (PRI) TDM services, such as T1s and T3s, will transition from being the primary voice service connection between the business and the public switched telephone network (PSTN) to being the backup voice service connection when connectivity loss occurs in the broadband packet network. In fact, depending on the business's requirements and network resiliency strategy, some of these physical TDM circuits will be eliminated altogether.

The PSTN, which was so ubiquitously used as the point of physical connectivity for TDM voice circuits, over time will be replaced by the IP PSTN. The IP PSTN is the service provider broadband network that also supports voice over IP (VoIP) switching and media control.

Using the IP PSTN, businesses can take advantage of their broadband access to public data networks to carry voice and video traffic. SIP will be used as the standard protocol to enable those functions in order to establish session connectivity to the IP PSTN.

As such, the corporate broadband data network facilities will also undergo changes as a result of SIP trunk deployment. These changes will affect data network services for both public WAN connectivity (that is, Internet), and private WAN services. However, these changes will occur mostly at the logical session layer rather than at the physical connectivity layer.

The private WAN of the business, which is another form of a broadband data network that is typically delivered using Multiprotocol Label Switching (MPLS) services from the service provider, will also be affected by the deployment of SIP trunks. How it is affected will depend on the SIP trunk deployment architecture model chosen.

Strategic Choices: Planning the SIP Trunk Architecture

The deployment of SIP trunks allows more flexibility and choice than TDM in designing the network architecture. There are three basic SIP trunk deployment models to choose from:

- Distributed
- Centralized
- Hybrid

These options mean that when a business migrates to SIP trunking, the telecom and IT manager need to choose the SIP trunk deployment model that will best support the collaboration services they are using today. With collaboration technologies continuing to evolve, it is also important to choose a deployment model with flexibility that can adapt to support new services with relative ease.

All three deployment models have merit depending on the requirements of the business. As such, the IT department must address two questions: Which deployment model fits our business? And, what criteria can be used for selecting between these three models?

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Characteristics Of Enterprise SIP Trunk Deployment Models

So, to help the corporate IT department to make the best choice in selecting a SIP trunk deployment model, we have provided a more detailed explanation of each of these three models.

Centralized SIP Trunk Model

In a centralized SIP trunk deployment, only the central hub–usually the headquarters or data center site of the organization–has direct SIP session connectivity to the IP PSTN (Figure 1). All external calls to or from remote sites of the business go through this central hub. Whether or not the central hub has a separate physical broadband connection for the SIP trunking service will depend on whether the same service provider is used for both the SIP trunk sessions and the private WAN. If so, then a single broadband connection may be sufficient; if not, separate broadband connections are needed for the SIP trunk and the private WAN.

Figure 1 Centralized SIP Trunk Model
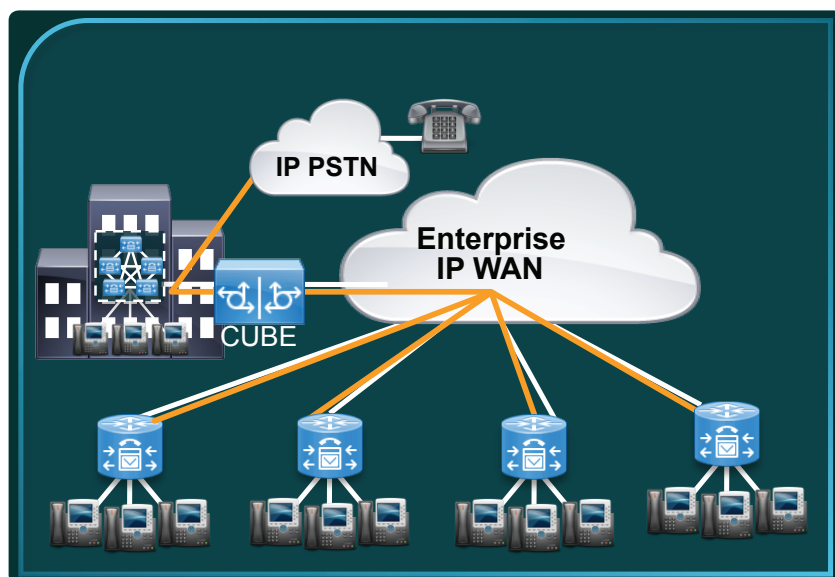
In centralized SIP trunk deployment models, signaling and media for intracompany calls will travel between the corporate sites over the private WAN, but are not processed by the SBC. Only external calls are processed by the SBC at the central hub.

Because the central hub, alone, is responsible for grooming external traffic for placement on SIP trunks connecting to the IP PSTN and for routing calls among company locations, remote sites are highly dependent on this central resource to make or receive external calls. These calls must "hairpin" through the central hub between the IP PSTN and the private WAN, usually MPLS-based, to connect to the remote sites. This method significantly increases bandwidth consumption through the central site and can introduce latency into the SIP session as experienced by the user.

"Hairpin" is a slang term to describe network traffic flow that travels into and out of a network device

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without any local consumption (local to the device performing the "hairpinning") of the content of that network traffic. Although "hairpinning" may be appropriate in some situations, by definition "hairpinning" consumes twice the WAN-link and network device (router) capacity and can potentially create network traffic bottlenecks.

For example, if an employee in Los Angeles calls a customer in San Francisco and must route through a central hub located in New York, the call goes from coast to coast and back again. This travel requires capacity at the central hub for essentially two calls. Long distances may create latency problems, especially if the network includes overseas operations.

Given that all remote sites are linked directly to the headquarters hub and the central hub must support bandwidth for each remote site, this type of SIP trunk deployment has limited flexibility. As more remotes sites are added or as more people are employed at each remote site, the business will need to closely monitor the bandwidth usage at the central hub for its connection to both the private WAN and the IP PSTN.

This bandwidth problem becomes even more problematic as adoption of video services increases. For example, a video call takes approximately 10 times the bandwidth of a voice call. This extra bandwidth requirement will make a centralized SIP architecture more expensive because both the WAN connections to the central hub and the SBC device in the central hub will need significant additional capacity to handle the video calls.

In a centralized SIP trunk deployment model:

- The central hub needs sufficient bandwidth for connectivity to both the IP PSTN and the private WAN to service all remote sites.
- Increasing bandwidth for remote site connection to the private WAN to accommodate video and other collaboration technologies will usually require increasing the bandwidth at the central hub for both the IP PSTN connection and the private WAN connection.
- Remote sites may compete for bandwidth on the private WAN and to the IP PSTN, a situation that can be problematic if the central hub is under-provisioned.
- High availability and strong QoS and CAC capabilities, although always valuable in a well-designed network, will be mission-critical at the central hub.
- Latency of real-time traffic (voice and video) to the remote sites is more likely to be a problem, particularly if the geographic separation between the central hub and remote sites is significant.
- Telephony survivability for remote sites is not provided. A redundancy strategy such as retaining a TDM gateway at each remote site–using either PRI or foreign exchange office (FXO)–as an alternative calling path to support outbound calls is usually needed. Cisco® Survivable Remote Site Telephony (SRST) running on Cisco Integrated Services Routers (ISRs) is an example of such a redundancy mechanism.
- The SBC may be either standalone or router-integrated depending on the number of SIP sessions required for all remote sites and the amount of bandwidth that must be supported by the router function for private and public WAN connectivity.
- Management of the SIP trunks and SBC may be simplified in a centralized deployment.

Distributed SIP Trunk Model

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In a distributed SIP trunk deployment, each corporate site that has routed connectivity to the corporate private WAN also has direct SIP session connectivity to the IP PSTN (Figure 2). This connectivity may be achieved over the same physical connection when one service provider provides both the private WAN (typically MPLS-based) and the SIP trunk session connections. Or, if different service providers are used, then there may be a separate broadband connection in each corporate site for connection to the SIP trunk service.

Figure 2 Distributed SIP Trunk Deployment Model
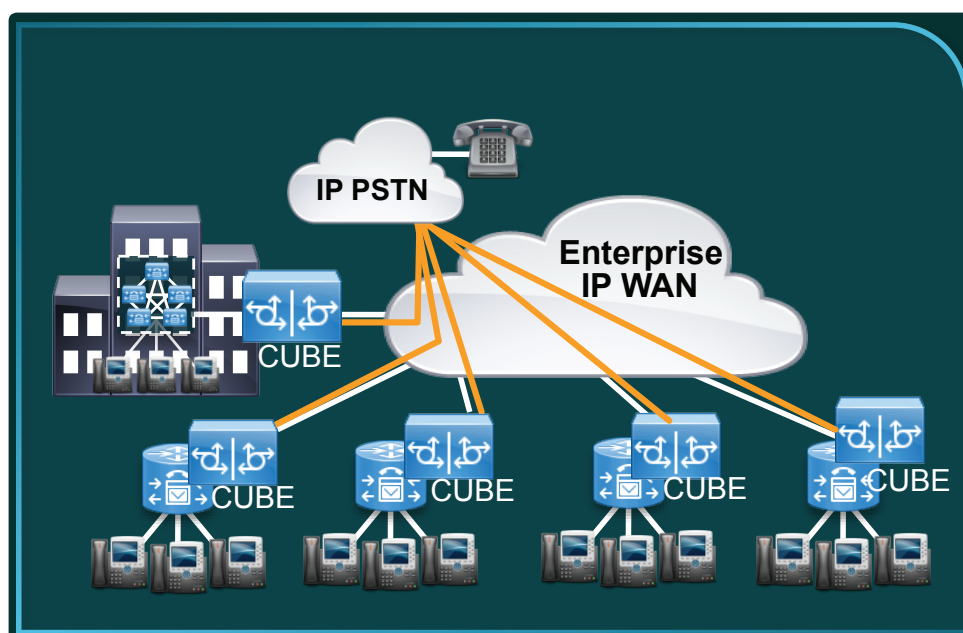
In centralized deployments, signaling and media for intracompany calls will travel between the different corporate sites over the private WAN, but will not need to be processed through the SBC. Only external calls will be processed by the SBC in each site.

When the service provider selected by the enterprise for SIP trunk connectivity supports the concept of pooled SIP sessions between locations, (refer to the side bar for information about pooled SIP sessions), the distributed model for SIP trunking delivers added flexibility. This flexibility allows SIP trunk session capacity to be tailored to the needs of each remote site, even allowing different levels of session capacity per site depending on factors such as time zone or user registration.

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In the distributed SIP trunk deployment model:

- Use of bandwidth by the business as a whole can be optimized, because the bandwidth of each site is determined by what it needs, independent of other sites.
- Bandwidth at any site can be increased as needed, setting the stage for video calling and other rich unified communications and collaboration services.
- The "hairpin" of network traffic at a central hub to transport voice or video traffic between the private WAN and the IP PSTN for providing service to remote sites is not necessary.
- Latency of real-time communication is minimized because calls are routed more directly through the WAN (private or public) to the IP PSTN.
- Load on the private WAN is reduced because a lot of media traffic between remote locations and the central hub is eliminated.
- QoS and CAC are handled at each site.
- Redundancy and survivability are inherent in the distributed SIP trunk deployment model. If a session to the IP PSTN goes down at one remote site, external calls to and from that site can be routed by an IP-PBX through another SBC at another remote site. Of course, in centralized call control models, retaining a TDM backup in each remote site is also an option.
- An SBC is required at each remote site connecting to the IP PSTN. In most cases, the router-integrated SBC is the most cost-effective method to satisfy this requirement because a router will also be needed at each remote site.
- Management of the distributed SIP trunk deployment must allow for multi-device configuration, monitoring, and control. For configuration, at minimum, some type of template capability is ideal. For monitoring, traditional Simple Network Management Protocol (SNMP) can be used. However, more advanced voice policy for both monitoring and control can be supported with web-based application programming interfaces (APIs), if available.
- Service provider SIP session pooling also makes the distributed model as economical as the centralized model for buying SIP trunks due to its flexible offering.
- The distributed model also provides a lower risk and easier transition to SIP trunking because it mirrors the TDM architecture.

Service Provider View Of SIP Trunk Deployment Models

It is valuable to understand the service provider perspective on the SIP trunk market. Over time, service providers have come to recognize the inevitability of packet-based real-time communication for both voice and video over the IP carrier backbone. They have also come to recognize that their services must evolve from providing voice circuits to providing collaboration services–or at least the infrastructure to deliver collaboration services. These trends are convincing almost all service providers that pervasive SIP trunking is inevitable, even if the pace at which it displaces TDM trunking is difficult to predict.

Therefore, the adoption of SIP trunking has accelerated rapidly in many areas of the world, most notably in North America and parts of Europe. The service providers in these regions are now offering significant advancements in SIP trunking services.

For example, Verizon and Sprint offer SIP trunking services that allow the enterprise to contract for a "pool" of SIP sessions that can be accessed from multiple locations. The number of sessions used at each location is flexible provided the enterprise does not exceed the total number of sessions in the pooling contract. This model gives the customer significant flexibility over how and where SIP trunk sessions can be accessed.

Service providers are also integrating SIP trunking into other services, such as MPLS-based private WAN services. This integration allows the enterprise to access SIP trunking services through a virtual portal provided in the customer's MPLS-based private WAN, rather than through a separate physical broadband connection. This service allows every customer site connected to the MPLS private WAN to have direct access to SIP trunk sessions through the SIP trunk virtual portal.

In each of these deployment models, the service provider is enabling the inherent flexibility in the use of the SIP technology to allow their customers to use the SIP trunk deployment model that is best suited to their current and future collaboration services needs.

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Hybrid SIP Trunk Model

In a hybrid SIP trunk deployment, some of the businesses' sites conform to a distributed SIP trunk deployment model. In this model each site has direct SIP session connectivity to the IP PSTN, and other sites conform to a centralized SIP trunk deployment, accessing the IP PSTN through a central hub, which has SIP session connectivity to the IP PSTN (Figure 3). The hybrid SIP trunk deployment model may have multiple "central" hubs in different geographic regions, or for specific business functions, such as call centers.

Figure 3 Hybrid SIP Trunk Deployment Model
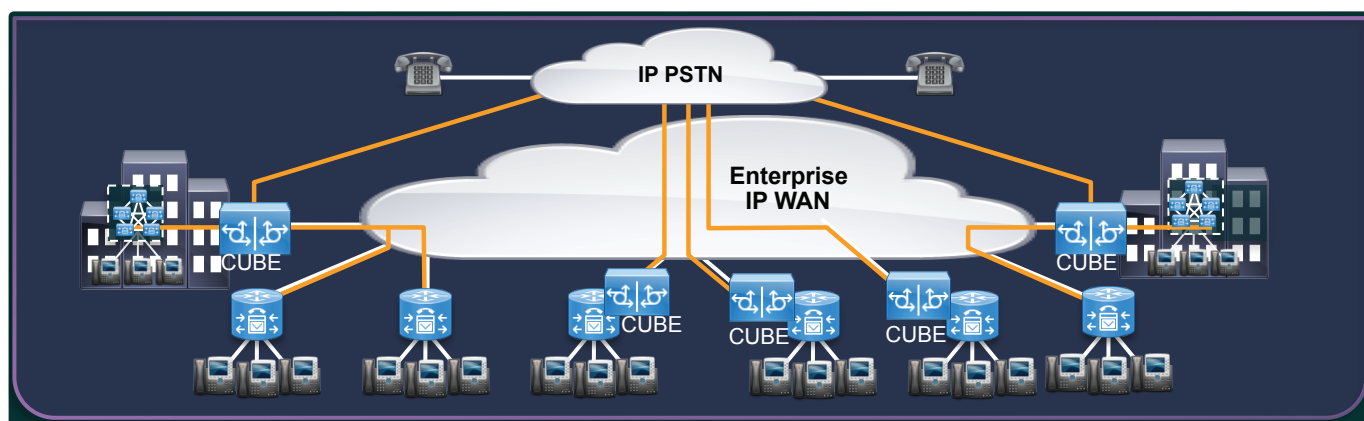
Identifying which remote sites should conform to a centralized SIP deployment model, and which should conform to the distributed model, will depend on various factors for each site. These factors include the number of employees, total inbound call volume, total outbound call volume, and overall bandwidth usage.

For example, remote sites with relatively low call volume and total bandwidth usage will probably be provided SIP trunk connectivity through a central or regional hub. On the other hand, a remote site that has relatively low call volume might still qualify for direct SIP session connectivity to the IP PSTN, per the distributed model if the total bandwidth usage of each site is relatively high (measured in comparison to other remote sites or in relation to the number of employees in the remote site).

Another factor that can affect how each remote site should be given access to SIP trunk connectivity might be the redundancy or survivability of voice services needed at each location.

However, in almost every situation, the largest determinant of which sites should conform to the centralized and distributed deployment model is the use cases for the collaboration services that are or will be deployed throughout the enterprise. For example, as more use cases involve real-time video, remote sites that might originally have been deployed in conformance to a centralized model may need to be converted to a distributed model, with direct SIP session connectivity to the IP PSTN.

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5. Including malware downloads, browser hijacking software, unwanted advertisement software, botnet check-ins, and Trojan (backdoor) connections
6. April 14 until May 31, 2011
7. Data from Q1CY11

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Combining some aspects of the centralized and distributed SIP deployment models, the hybrid model can be the "best of both worlds", depending on the needs of individual sites. It can also serve as an intermediate step to a fully distributed or centralized SIP network.

In the hybrid SIP trunk deployment model:

- Bandwidth is more adaptable to site-specific requirements than in a centralized network, but regionally has some of the same constraints.

- The capacity at sites that connect directly to the IP PSTN is adaptable to regional SIP offerings, perhaps allowing you to add capacity to support collaborative applications more cost-effectively than in a centralized network.

- For those parts of the network that are linked to a regional or central hub, QoS and CAC are managed at those regional hubs; sites linked directly to the IP PSTN manage them locally.

- Survivability is built in for sites linked to the IP PSTN; for those linked to a hub, other survivability strategies must be created.

- Centralized management tools to support both provisioning and policy monitoring and enforcement are critical in this model.

Selection Criteria For SIP Trunk Deployment Model

To summarize the concepts that have been discussed in this document thus far, Table 1 lists criteria that need to be considered and balanced against each other in the process of identifying the "optimal" SIP trunk deployment model for the enterprise.

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Of course, there may be criteria specific to a particular user that we have not included in Table 1, so IT managers should use the criteria in the table merely as guidelines.

Table 1 Architecture Selection Criteria

Selection Criteria	Centralized	Distributed	Hybrid
Limitations on central hub bandwidth availability	Requires strong QoS strategies on enterprise WAN	Not affected by hub bandwidth availability	Adaptable to bandwidth availability
Distributed call control	Not recommended	Preferred	Allowable
Variability of branch office service requirements	Recommended with uniform and simple branch office requirements	Recommended with uniform but complex branch office requirements	Optimal when branch office requirements are variable
Variability of branch office capacity requirements	Optimal when branch office capacity is low (<20% of trunks)	Optimal when branch office capacity is high (>50% of trunks)	Optimal when branch office capacity is high but varies from site to site
Videoconferencing and video telephony requirements through service provider	Requires strong QoS strategies on enterprise WAN	Requires adequate bandwidth at each site	Provides flexibility in phased deployment
Desire or need to maintain branch office site IT functions	Allowable	Recommended	Allowable
Maintain consistent latency across voice network	Can cause inconsistent latency	Recommended	Recommended
Degree of centralized services (voicemail and conferencing)	Preferred if branch offices have minimal IT infrastructure elements	Preferred if branch offices support IT infrastructure elements (e.g. voicemail servers)	IT infrastructure may exist in some branches, but not in others
Data center strategy	Optimal if branch offices have minimal IT infrastructure elements	Optimal if branch offices support IT infrastructure elements (e.g. WAAS servers)	IT infrastructure may exist in some branches, but not in others
Voice gateway protocols for TDM access	If MGCP is used on TDM gateway, SIP centralization may be easiest transition	H.323 or SIP used on TDM gateway allows easy transition to SIP trunks	Preferred when both MGCP and H.323 and SIP are used on various TDM gateways
Capabilities of centralized management	Device management may be adequate	Requires strong centralized management	Same as distributed
Standardization of branch office platforms	Branch office platforms (e.g. servers) are not deployed	Recommend standardized definition of platforms and configurations	Standardization may be difficult due to wide variation of branch office requirements
Enterprise WAN capabilities	Requires strong QoS and CAC	Not a consideration	Requires QoS and CAC
Survivability and alternative-path strategy	Requires TDM backup in distributed branch offices	Multiple SIP connection points provide survivability	Multiple SIP connection points provide survivability

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Important Criteria To Be Considered Include:

- Are there limitations on the availability of bandwidth to the central hub?
- How much SIP traffic now originates in how many branch offices overall, and how much in each branch office?
- How much growth is expected in call volume, bandwidth usage, and video traffic?
- Does your company expect to be a voice-centric company or to become a collaboration-centric company, taking advantage of more advanced technologies such as video?

If you have limited bandwidth and numerous branch offices that use a lot of it, the centralized model does not initially look like the best choice.

A guideline followed by some enterprises dictates that if the remote sites of the business consume more than 50 percent of the bandwidth in the WAN, then the distributed SIP trunk model should be considered. If the remote sites consume less than 20 percent of the bandwidth in the WAN, the centralized model may be the preferred deployment model. The hybrid model should be considered for situations where branch office demand is high but that demand varies significantly from site to site.

Another factor is geographic distribution. For example, depending upon the country where remote sites are located around the world, compliance with local regulations may mandate local terminations. This situation will automatically require the business to deploy a distributed or hybrid model SIP trunk architecture.

Another consideration is how quickly the organization needs to transition to SIP trunking. Most enterprise voice networks based on TDM trunks are distributed in nature because of the constraints of the traditional PSTN. Making a direct transition to a SIP trunk centralized model can be significant because of the change of both technology and network topology. If speed of deployment and the desire for lower risk and less incremental change are important to an enterprise, then distributed SIP trunking architecture may be the preferred option.

Finally, choosing the SIP trunk architecture that best simplifies the network evolution when new collaboration services are introduced is likely the most critical decision. If a particular SIP trunk service locks an organization into a specific deployment model that is appropriate for some collaboration service use cases but will not easily adapt to others, then the business may face expensive network upgrades in the future as evolving collaboration use cases create different requirements for the SIP trunk services. This inflexibility could lead to competitive disadvantages for the enterprise, when rival organizations are able to deploy new collaboration services faster. In other words, the SIP trunk deployment model selected should simultaneously provide for future flexibility, in order to accommodate collaboration use cases as yet unforeseen.

User Examples Of SIP Trunk Deployments: What Three Different Enterprises Chose

One company, a large retailer, had more than 1000 locations and a distributed IP-PBX network already in place. Deploying a SIP trunk distributed architecture allowed the company to continue to take advantage of its call control at remote sites and minimized the risk of a big architecture change to a centralized model. The company also chose a distributed architecture because it wanted to prepare for the implementation of collaborative technologies such as video as they become available.

Another company, a health insurance provider with four primary locations and 1500 employees, had a centralized IP-PBX at its headquarters and two call centers that were not linked into this system. It chose the hybrid model.

Finally, a services company chose a centralized SIP trunk architecture. This company was a voice-centric company with no plans for high-bandwidth applications such as video. It also had a majority of its traffic originating from its headquarters site with a large call center at that location.

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How Does Cisco Fit Into These Decisions?

The network design concepts discussed thus far are the concepts that have been directing Cisco product development over an extended period of time. Cisco understands that SIP trunking is the basis for supporting the evolution of collaboration services into the future. Cisco can help you meet your objectives for deploying the SIP trunk infrastructure that will allow you to adapt new collaboration services to the needs of your organization.

Cisco Unified Border Element: Cisco's Session Border Controller

Cisco Unified Border Element, also referred to as CUBE, provides SBC features in a router-integrated form factor on a wide range of Cisco routers. CUBE has unique capabilities to help smooth your transition to SIP trunking in ways that are appropriate for your business:

- CUBE supports all three SIP trunk deployment models: centralized, distributed, and hybrid. Our goal is to give you the freedom to choose the model that best suits your company and its future collaboration service requirements.

- CUBE provides the broadest range of SBC scalability of any vendor in the industry today. As such, CUBE delivers competitive price-performance for any size business site and deployment model. It provides price-competitive SIP session connectivity with support for as few as 15 sessions on the Cisco 800 Series Integrated Services Routers, as many as 16,000 SIP sessions on the Cisco ASR 1000 Series Aggregation Services Routers, and any number of sessions within this range on the Cisco 2900 and 3900 Series Integrated Services Router platforms.

Cisco Unified Communications Manager Session Management Edition in Your SIP Trunking Environment

Cisco Unified Communications Manager Session Management Edition is an important element in large enterprise centralized, distributed, and hybrid SIP trunking architectures. Cisco Unified Communications Manager Session Management Edition with Cisco Unified Border Element (CUBE) supports all three architectures, and together they create a flexible foundation to evolve your network to SIP trunking and beyond.

Cisco Unified Communications Manager Session Management Edition centralizes administration, dial plans, applications, PBXs, and IP-PBXs to streamline the enterprise network for faster problem resolution and to extend applications and features to any endpoint. CUBE breaks down the TDM barriers by using SIP trunking to extend these applications beyond the enterprise to customers, partners, or anyone.

Collaboration is constantly improving and changing, and Cisco Unified Communications Manager Session Management Edition and Cisco UBE create the platform to make the evolution faster, easier, and more cost-effective.

- On the Cisco 2900 and 3900 Series routers, CUBE provides the ability to simultaneously support multiple unified communications services such as SBC and TDM gateway functions. This capability provides our customers the flexibility to manage their migration from TDM trunks to SIP trunks according to their business needs.

- CUBE on the Cisco ISRs provides the greatest flexibility for SIP trunk deployment available in the. If you have a Cisco ISR at your various branch office locations being used for other functions, such as a TDM gateway or an IP or MPLS router, you already have the hardware platform needed for SIP trunk connectivity. Often, CUBE can be added easily to an existing Cisco ISR platform with just a simple license or software upgrade, with no hardware upgrade needed.

- The Cisco ISR can support CUBE not only for SIP trunking services, but also for many other unified communications services such as Cisco SRST for resiliency in the voice network. This service flexibility means that the Cisco ISR can be repurposed as your network requirements change.

- Having both routing and session border control capabilities in the same platform makes the footprint smaller and reduces your need for spare units. It also simplifies transmission.

- Integration with Securelogix Voice Policy provides the industry's most comprehensive and flexible solution to help lower costs, improve efficiencies, and secure the enterprise.

- The addition of Cisco Unified Communications Manager Session Management Edition in large-scale deployments also provides greater ability to save, simply, and extend rich-media services in SIP trunking environments. This application can be used in all three SIP trunk architectures. (Refer to the side bar for more information about Cisco Unified Communications Manager Session Management Edition in SIP trunking environments.)

Deciding on the SIP trunk architecture that will be the foundation for the collaboration services in your network is critical to your organization. Understanding how that decision can be made to help your organization's network more easily evolve to support future collaboration services will benefit your organization for years to come. We hope that this white paper has helped you better understand your options in making this decision.