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# Integrating Your IP PBX with an ITSP

*Leveraging SIP Trunking for Broadband Services*

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Sphere Communications



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# Topics

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- Overview of Internet Telephony Service Providers (ITSP) Service Models
- Tying the Two Together: ITSP + IP PBX
- Important Considerations
- Lessons Learned
- Q&A

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# Internet Telephony Service Models

Business and Consumer Services

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Market Segments	<b>Business</b>			<b>Consumer / Residential</b>	
	Large	Medium	Small		
Business Models	Voice: (Local+LD+Toll Free), Data, Web Hosting			Bundled Services (Triple Play)	Cheap Dial Tone
	Voice over "Managed" IP				
	Internet Dial Tone				
Service Models	<b>Fully Managed</b> Service Level Agreements, performance + fault monitoring, QoS, etc.			• • • •	
	<b>Un-Managed</b> Best Efforts Voice Quality, Capacity Controlled				
Service Types	"Soft" Trunking				
	VoIP Gateway Trunking				
	VoIP Centrex				
ITSP Examples	AT&T, BandTel, Cbeyond, Global Crossing, Starvox, USLEC, XO . . .			Comcast, Vonage, Tomato Vine. . .	

# Internet Telephony Service Models

## Business Services

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- Telecom carriers have been offering VoIP services, to varying degrees, for several years.
- Until recently “Business Class” VoIP services have been focused on 2 models:
  - **VoIP Centrex**: delivering IP dial tone to small businesses over a shared Internet connection. This may be done using an IP phone or an analog terminal adapter (ATA). In some cases, the Internet service is obtained independently of the VoIP service. Sometimes voice and data is bundled.
  - **VoIP Gateway Trunking**: delivering converged voice and data service together, over the same connection, and separating the two at the customer premise using a gateway device. This bundled approach offers QoS advantages for the service provider.
- More recently, Telecom providers have introduced VoIP “Soft” trunking as a means for connecting IP PBX systems to the carrier services.
  - **VoIP “Soft” Trunking** (more recently referred to as “SIP Trunking”) is emerging as a means for Carriers to offer converged services to medium-large enterprises that wish to manage their own voice infrastructure. Most often, SIP trunking will be offered as a bundled and managed service. This positions the Carrier to offer advanced capabilities.



# Internet Telephony Service Models

## Business Services

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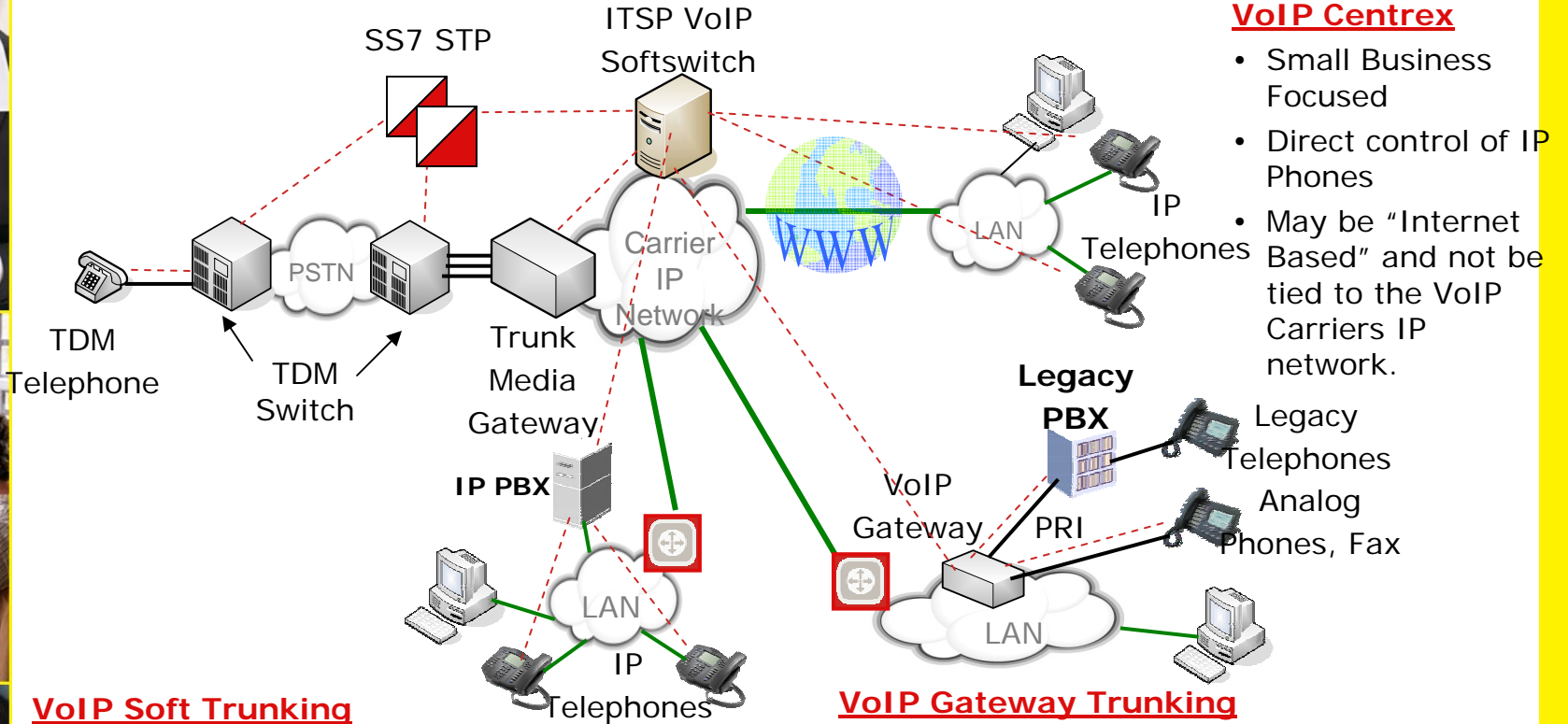


Type	Description	Pro	Con
VoIP Centrex	<ul style="list-style-type: none"> <li>• Providing managed telephone service to a business over an IP connection.</li> <li>• This may be wither a “fully managed” or “un-managed” IP connection.</li> </ul>	<ul style="list-style-type: none"> <li>• Full service</li> <li>• No expert staff needed</li> <li>• Low initial investment</li> </ul>	<ul style="list-style-type: none"> <li>• Little to no “control”</li> <li>• Restricted Capabilities</li> </ul>
VoIP Gateway Trunking	<ul style="list-style-type: none"> <li>• Providing voice and data services to a business over a “converged” network connection.</li> <li>• <b>Narrowband</b> Voice Services are broken out separately at the customer premise using a “media gateway”.</li> </ul>	<ul style="list-style-type: none"> <li>• Economics of V+D convergence</li> </ul>	<ul style="list-style-type: none"> <li>• Narrowband service – limited services</li> <li>• Personnel costs</li> </ul>
VoIP “Soft” Trunking	<ul style="list-style-type: none"> <li>• Providing <b>broadband</b> voice and data services to a business over a “converged” network connection.</li> <li>• Voice services are broadband or “all IP” to the telephony device.</li> </ul>	<ul style="list-style-type: none"> <li>• <b>Economics of V+D convergence</b></li> <li>• <b>Broadband capabilities</b></li> <li>• <b>Enhanced Services</b></li> </ul>	<ul style="list-style-type: none"> <li>• <b>Personnel costs</b></li> </ul>

# Internet Telephony Service Models

## Business Services

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**Legend:** VoIP "Aware" Firewall/Router (Application Layer Gateway) 

Circuit Connection  IP Connection  Voice Signaling Path  Voice Bearer/Media Path 



# ITSP to IP PBX Soft Trunking Service

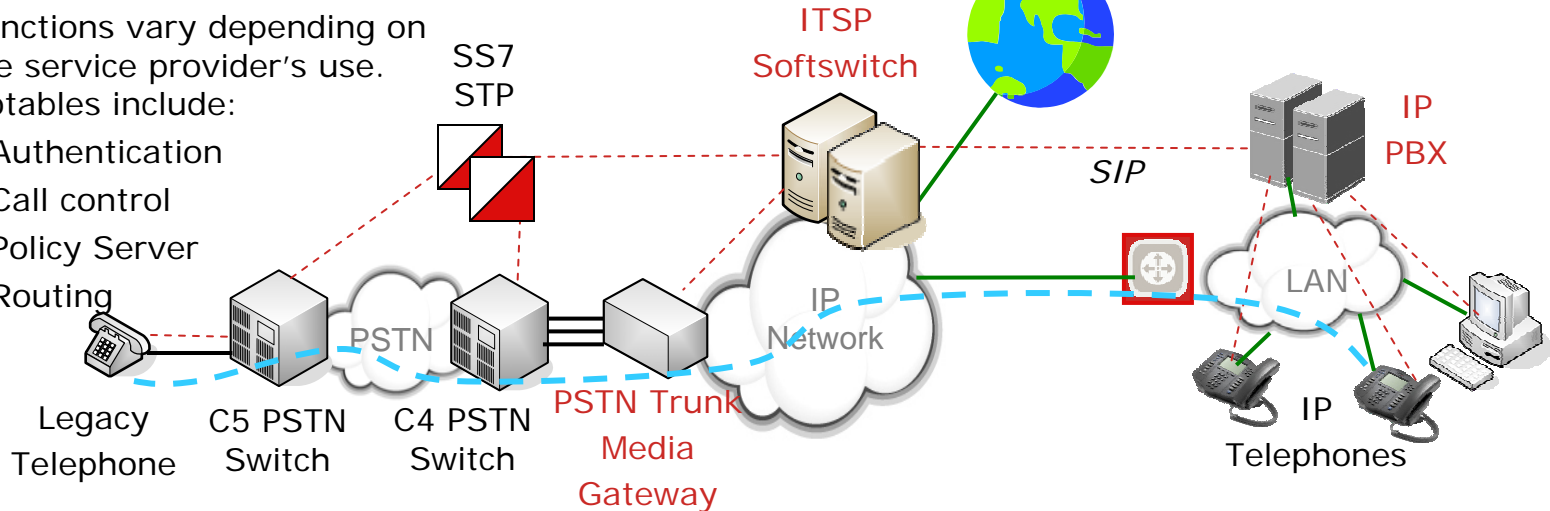
## Key Service Elements

### ITSP Softswitch

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Functions vary depending on the service provider's use. Notables include:

- Authentication
- Call control
- Policy Server
- Routing



### PSTN Trunk Media Gateway

- Converts between IP and TDM formats
- Typically uses Inter Machine Trunks (IMT) to connect to PSTN switches (often C4)

### VoIP "Aware" Router/Firewall

Functions vary widely depending on the service provider's use. Notables include:

- Routing
- Firewall / security
- Public / Private IP addr. translation
- SIP session management
- **NAT traversal**

### IP PBX

- Enterprise wide telephony features and services
- Scalability, Redundancy, High Availability
- Presence Services
- Policy services
- Multi-media communications
- "Application Integration"

### **Legend:**

VoIP "Aware" Firewall/Router (Application Layer Gateway)



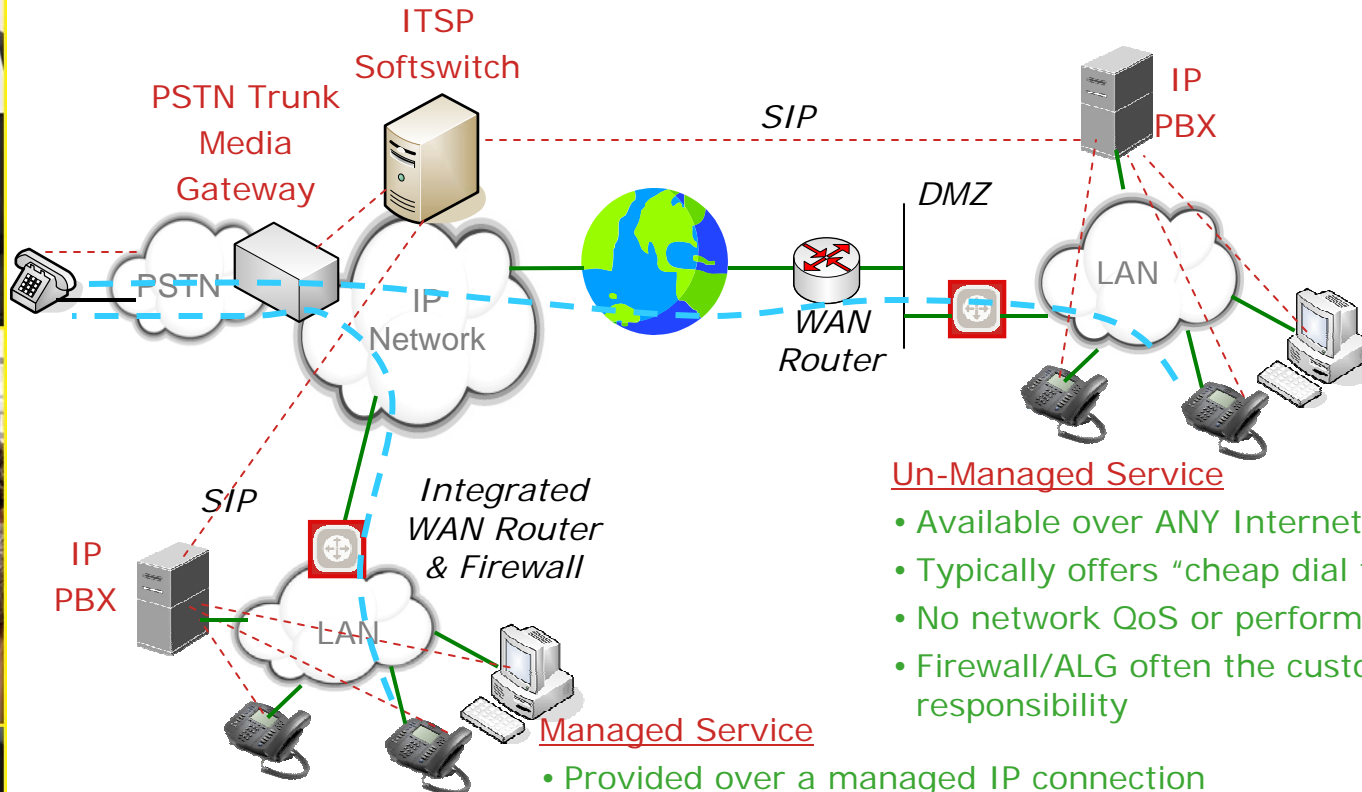
Circuit Connection ——— IP Connection ——— Voice Signaling Path - - - - Voice Bearer/Media Path - - - -





# ITSP to IP PBX Soft Trunking Service Managed vs Un-managed

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### Un-Managed Service

- Available over ANY Internet connection
- Typically offers "cheap dial tone"
- No network QoS or performance guarantees
- Firewall/ALG often the customers responsibility

### Managed Service

- Provided over a managed IP connection
- Often bundled with other "data services"
- Typically includes the WAN termination device
- Service Level Agreements (SLA) offered

**Legend:** VoIP "Aware" Firewall/Router (Application Layer Gateway)



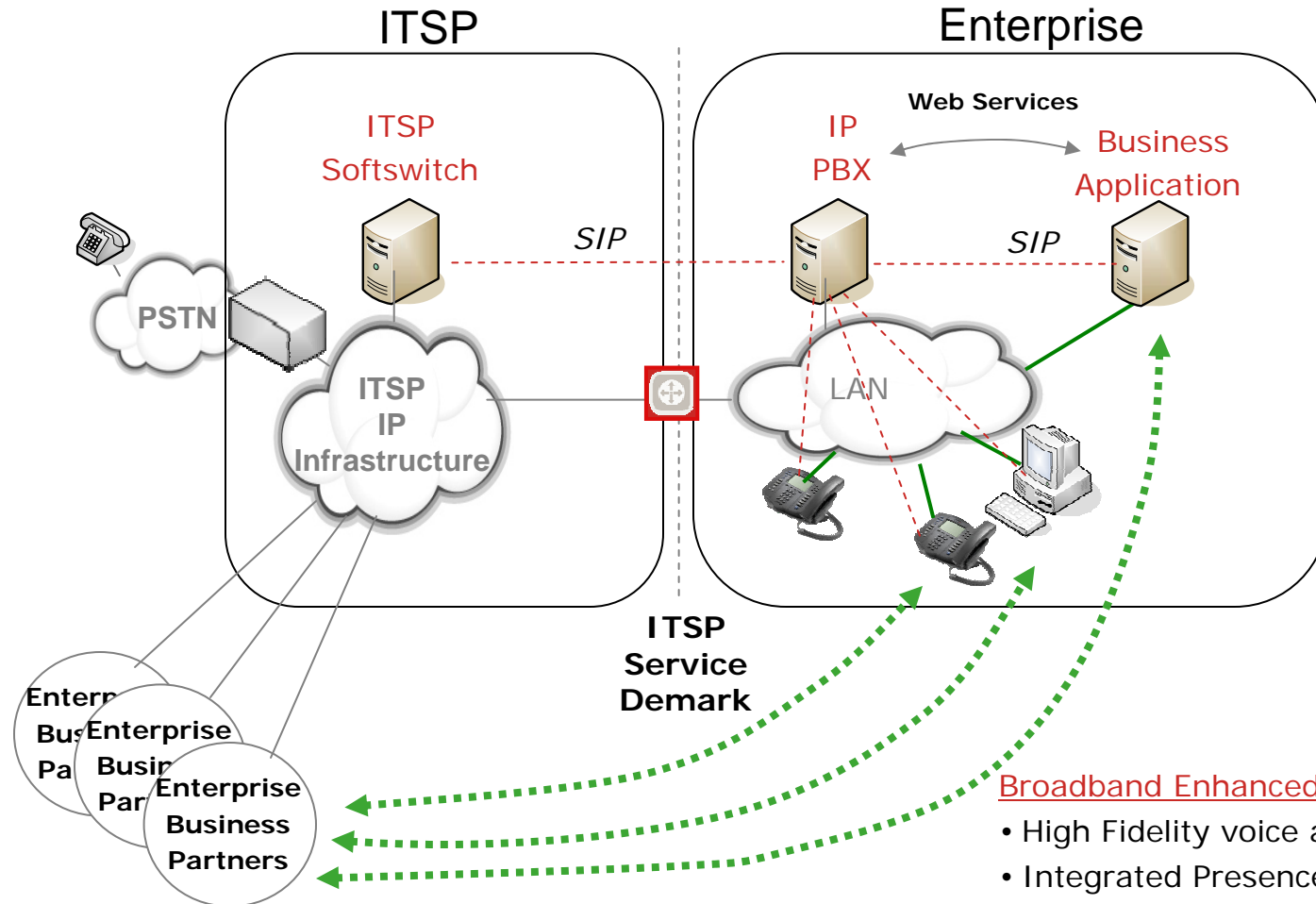
Circuit Connection ——— IP Connection ——— Voice Signaling Path - - - - Voice Bearer/Media Path - - - -

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# SIP Trunking Enables Broadband, All-IP Integration

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### Broadband Enhanced Services

- High Fidelity voice and video
- Integrated Presence and IM
- Etc.

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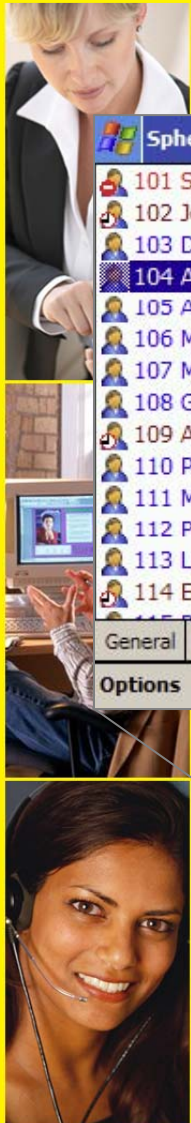
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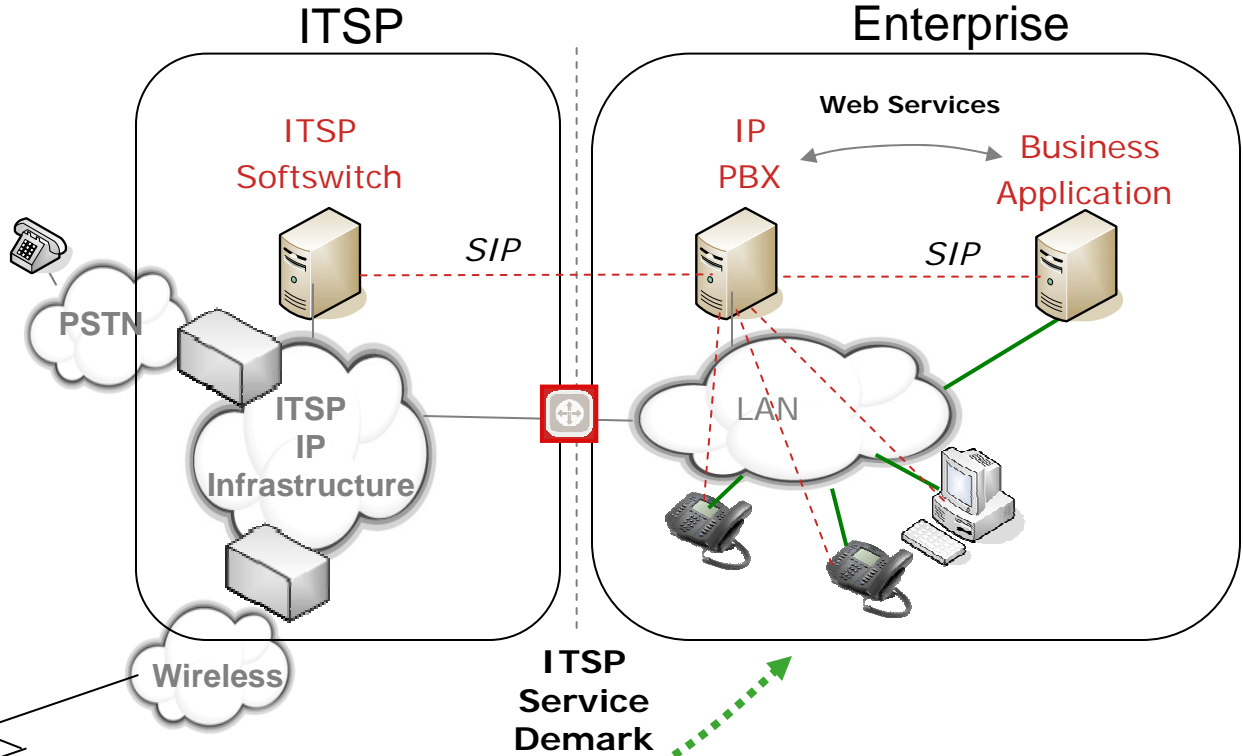
# SIP Trunking

## Enables Broadband, All-IP Application Integration

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Mobile Phone Client



### Broadband Enhanced Services

- Monitor / Screen Calls
- Managed personal communications profiles
- Presence and IM

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# Understanding the ITSP Services (Today and Tomorrow)

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- Managed vs. Un-manged
- On-net vs. Off-net
- Quality of Service Capabilities
- Service Level Guarantees

Carrier	Local / DID	Long Dist.	Toll Free	E911	Centrex	VoIP GW	Soft Trunk
ITSP 1	Yes	Yes	Yes	Yes	Yes	SIP	SIP
ITSP 2	No	Yes	Yes	No	No	SIP	No
ITSP 3	No	Yes	Yes	No	No	H.323	No
ITSP 4	Yes	Yes	Yes	Yes	SIP	H.323	No

# Important Consideration

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- Signaling Methods: (SIP, H.323, Proprietary)
- Standards and “Best Practices” are evolving: see the SIP Forum and “SIP Connect”
- Interoperability and Certification
  - Leading Carrier Softswitch Vendors (ex: Broadsoft)
  - Bilateral arrangements (ITSP and IP PBX)
- Services Demarcation Point: where does the ITSP’s responsibilities end?



# Lessons Learned

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- Using SIP to trunk between an ITSP and an IP PBX **does not** guarantee interoperability.
  - Reference “Best Practices” such as the SIP Forum’s SIP Connect specification.
  - Reference certification / interoperability arrangements.
- Following are some areas to be “aware” of:
- **Authentication:** how does the ITSP perform authentication and is the IP PBX compatible
- **DID call screening:** Some ITSP’s screen ANI information to ensure that the originator number is part of their DID range. This can be a problem if you have mixed DID’s – ex: some from a traditional carrier.

## Lessons Learned (continued)

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- **Trunk to Trunk transfers:** Example, an outside caller may try to reach an employee who has their office phone transferred to a cell phone. ITSP's that perform **DID Call screening** will not allow the transfer if the original calling party number is used as the CID. The PBX will need to substitute the originating CID with a DID within the ITSP range.
- **Media Session Restrictions:** some ITSP's will restrict the media sessions between devices. For instance, a B2B scenario could include company to company video calls or high fidelity voice calls.

## Lessons Learned (continued)

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- **Out-of-Band DTMF Compatibility (RFC 2833):** implementations. Is this mandated by the ITSP? Does your IP PBX support it?
- **Telephony Dialing Rules:** Will the ITSP mirror the traditional telecom dialing rules within a given market: For example, 7, 10, 11 digit dialing where appropriate. Some ITSP's require 11 digit dialing even for local calls.



## Lessons Learned (continued)

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- **NAT Traversal:** This has been the **#1 issue** to work through in our experience. A reliable NAT traversal solution is required.
- Traversing between the public and private IP address boundaries presents problems for interactive communications.
- Most commonly, for business applications, SIP “aware” firewall or “Application Layer Gateway is utilized.
- This device must be capable of routing inbound and outbound sessions to the appropriate public/private destination.
- Often this device is provided by the ITSP as part of the service offering. If not, determine if the ITSP offers supported recommendations?
- SIP ALG / Firewall Examples: Adtran, Ingate, Cisco, more . . .
- Other solutions (ex: STUN) have not taken hold in Enterprise environments where firewall configurations are asymmetrical and more complex.

# Summary

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- SIP trunking between and ITSP and IP PBX can offer customers with “the best of both worlds”
  - Broadband convergence (cost savings and features)
  - Control of their own communications systems
- Best Practices are available – verify that your ITSP and IP PBX providers are engaged
- Look for Certification and Service Level Agreements
  - Remember you get what you pay for!

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# Additional Resources



SIP Forum - SIPconnect overview - Microsoft Internet Explorer

File Edit View Favorites Tools Help

Back Forward Stop Home Refresh Search Favorites

Address http://www.sipforum.org/content/view/274/228/

## SIP FORUM

## SOFTSWITCHING

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Home > TECHNOLOGY > SIPconnect > SIPconnect overview

### SIPconnect overview



**The SIPconnect Interface Recommendation**

**A Standards-based Approach for Direct IP Interoperability between IP PBXs and VoIP Service Provider Networks**

The SIPconnect Interface Specification is an important industry initiative that builds on existing IETF standards to define a method for interconnection between IP PBXs and VoIP service provider networks, and specifies a reference architecture, required protocols and features, and implementation rules necessary for seamless IP peering between IP PBXs and VoIP service providers.

**Features & Benefits**

- **A Ubiquitous Approach.** SIPconnect provides a common method for IP peering between SIP-enabled IP PBXs and VoIP service providers
- **Standards Based.** SIPconnect leverages existing SIP and related VoIP standards published by the Internet Engineering Task Force (IETF)
- **Customer Cost Savings.** Peering will lower service provider infrastructure cost and reduce the need for customer premises gateways
- **Richer Feature Support.** SIPconnect helps service providers deliver enhanced, personalized services to IP-PBXs and extends rich-media services enabled by IP-PBXs across service provider networks
- **Quality of Service.** Methods for handling QoS configuration, echo cancellation, DTMF relay, packetization rates, codec support and fax and data traffic are defined

**Recommendation status**

SIP Forum documents go through several phases on their way to completion:

- Drafts (multiple). This is the phase the document is in during development.
- Proposed Recommendation. Once the document has reached a stage where there are no more comments, it enters this stage, and stays here until multiple interoperable implementations exist.
- (Final) Recommendation. Once proven implementations exist, the document enters its final status.

The [SIPconnect Recommendation](#) is currently in the Proposed Recommendation status (as of August, 2004)



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Questions?



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