

SIP Trunking

DEEP DIVE: The Service Provider

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Agenda

- Overview
- SIP Services
- SIP Transformation Lessons Learned
- Case Studies
- Q & A



Signs of Significant Growth in SIP Traffic

- AT&T plans to move all customers voice communications to new technology by 2020¹
- Frost & Sullivan reported in 2014 that the SIP Trunking services market is expected to grow from **\$2.83B** in 2013 to over **\$9.35B** in 2019²
- The Eastern Management Group reported in 2013 that SIP adoption is expected to grow from **13%** of companies using SIP for all toll traffic to **42%** by 2018³
- Infonetics Research surveyed that **58%** of enterprises will use SIP trunks by **2015**⁴

1 - <http://www.attpublicpolicy.com/tag/ip-transition/>

2 - <http://www.prnewswire.com/news-releases/frost-sullivan-the-advanced-capabilities-of-voip-and-sip-trunking-exceeds-traditional-voice-services-247236161.html>

3 - <http://www.nojitter.com/post/240162594/sip-trunking-research-shows-rapid-growth-through-2018>

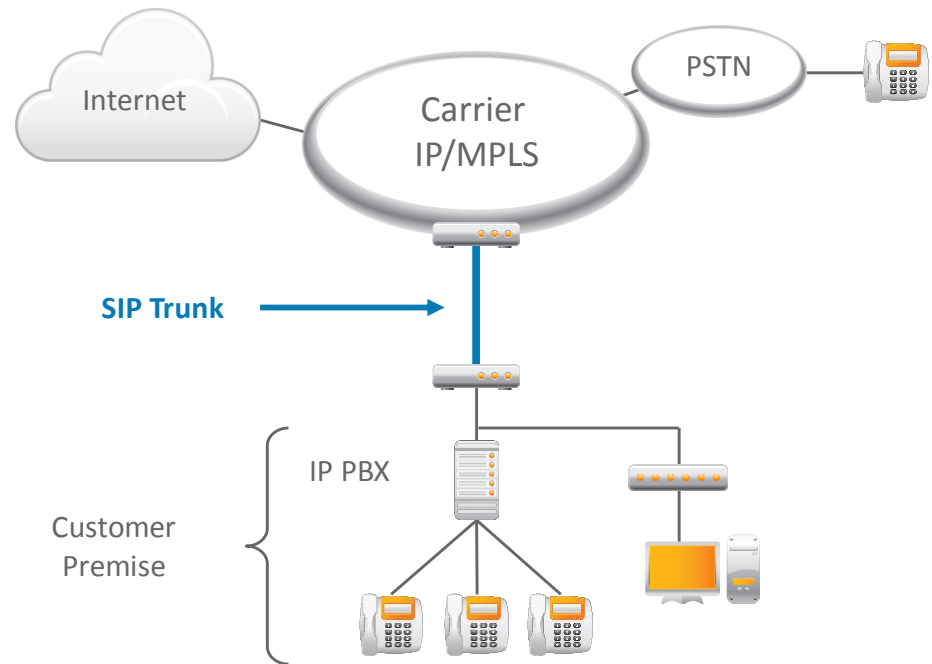
4 - <http://www.infonetics.com/pr/2013/SIP-Trunking-and-SBC-Enterprise-Survey-Highlights.asp>



Why customers are use SIP Trunking?

Benefits include:

- Common protocols for voice and data networks
- Reduced circuit cost delivery
- Network optimization
- Simplified On/Off net dial plan
- End-to-end IP calling
- Dynamic bandwidth allocation
- Scalability and flexibility
- Next-generation IP-based services
 - Contact Center
 - Carrier IMS / Mobility integration
- Open standards



Call Types, Codecs and Bandwidth

Voice

- G.711 a-law/ μ law 64Kbps (about 84kbps per cc)
- G.729, G.729B 8kbps (about 23kbps per cc)
- G.726 32kbps

Fax

- G.711
- T.38

Modem

- Credit card machines, alarm lines, TTY/TDD, and recent-model high-speed dialup data devices
- NOT supported



SIP Services

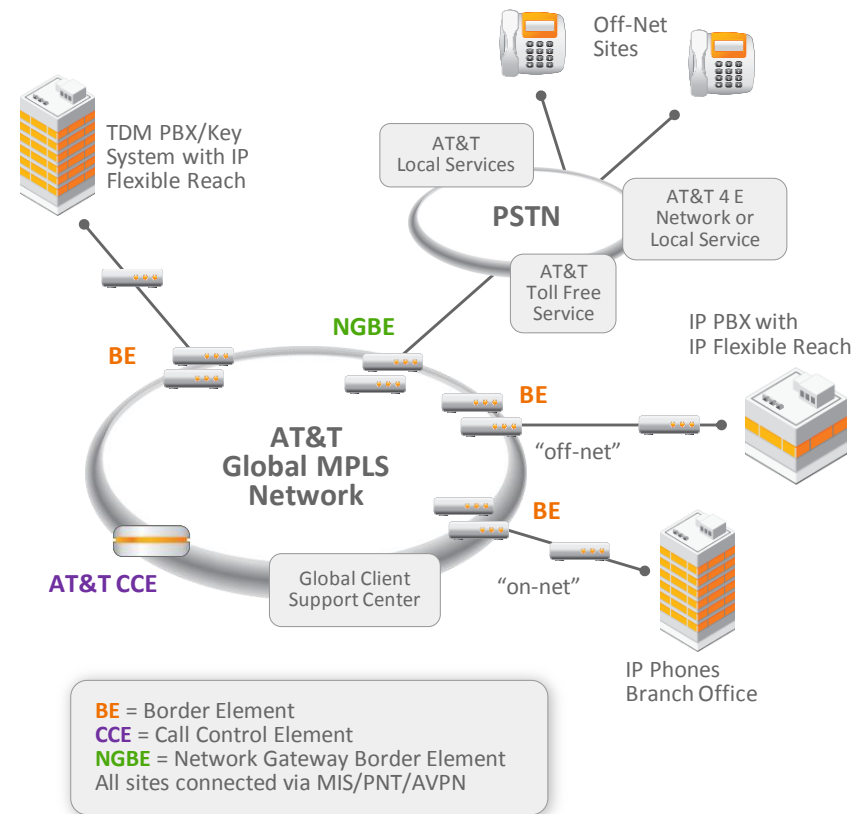


IP Flexible Reach SIP Trunking on MPLS

- SIP “trunking” service delivering integrated access for customer PBX systems
- True “All Distance” calling –National and International Long Distance from 60 countries with Voice & Data on a single network

Highlights

- VoIP calling solution with unlimited local & on-net calling with competitive off-net plans
- Access and bandwidth savings with voice and data on converged transport replacing local TDM PRI’s.
- Available with IP PBX, TDM PBX, Key Systems (analog telephones)
- Global VoIP Call quality SLA’s



AT&T SIP Trunking Features

Voice Quality of Service

- Class of Service with 25 different profiles optimizes voice & data application performance
- Dynamic bandwidth allocation supporting bursting of data during voice idle periods
- Silence suppression for up to 50% reduction of per call packets
- Industry leading call compression capabilities

Interoperability with traditional and next-generation PBXs

- Traditional PBX/Key system interfaces
 - Support CAS, PRI & analog signaling
- IP PBXs interfaces
 - Cisco, Avaya, Microsoft, Siemens & Nortel
- Additional PBX certification testing is on going.

Multiple Call Types

- IP On-Net to IP On-Net
- IP On-Net to PSTN Off-Net
- Inbound /Outbound Local Calling
- International off-net

Virtual Telephone Numbers*

AT&T Management

- Centralized Dial Plan administration and maintenance
- Network QoS monitoring & management
- Network Performance Reporter – Web portal for Call Detail Reports
- Service Level Agreement - Site Availability

* For customers selecting VTNs (or BOE TNs), the AT&T VoIP network will always send 10 digits to the PBX



E911

Local Calling Plan

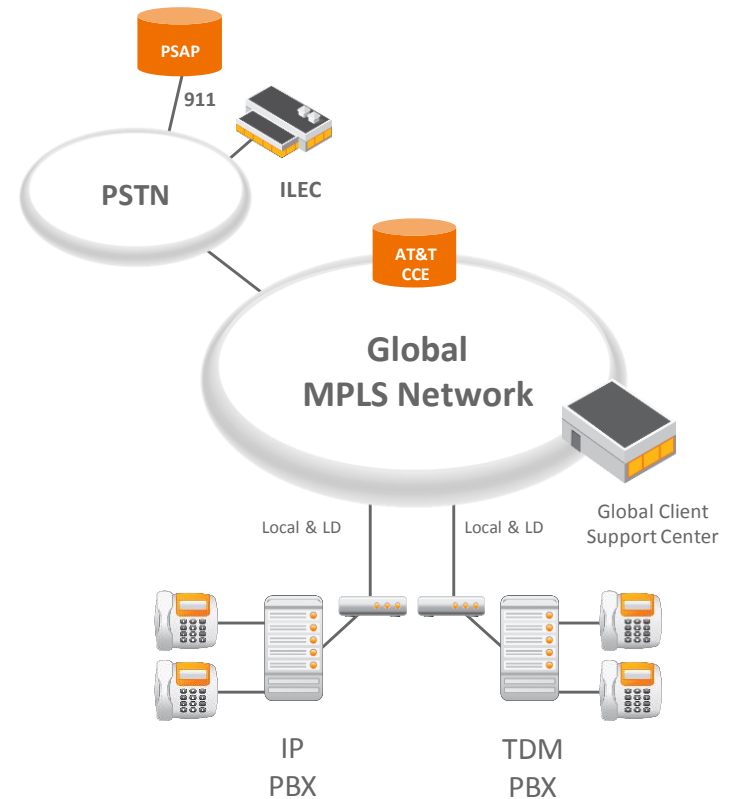
- E911 is supported only for calls made from the registered location within the business VoIP local footprint (including AT&T owned local service facilities)
- Emergency calls are routed to a geographically appropriate PSAP based on the caller's location.
- Key System or PBX (IP or TDM):
 - Only one registered address for a given customer location
- TDM PBX utilizing CAS*:
 - Only one call back number (configured in the router) for all users.
- TDM PBX utilizing PRI or IP-PBX:
 - Station ID is supported as the call back number if provided by the PBX

US LD Only Plan

- 911 calling is not supported. A traditional POTS line needs to be actively maintained to support 911 calling

AT&T strongly recommends alternate means of accessing 911 services at every site

*Channel Associated Signaling

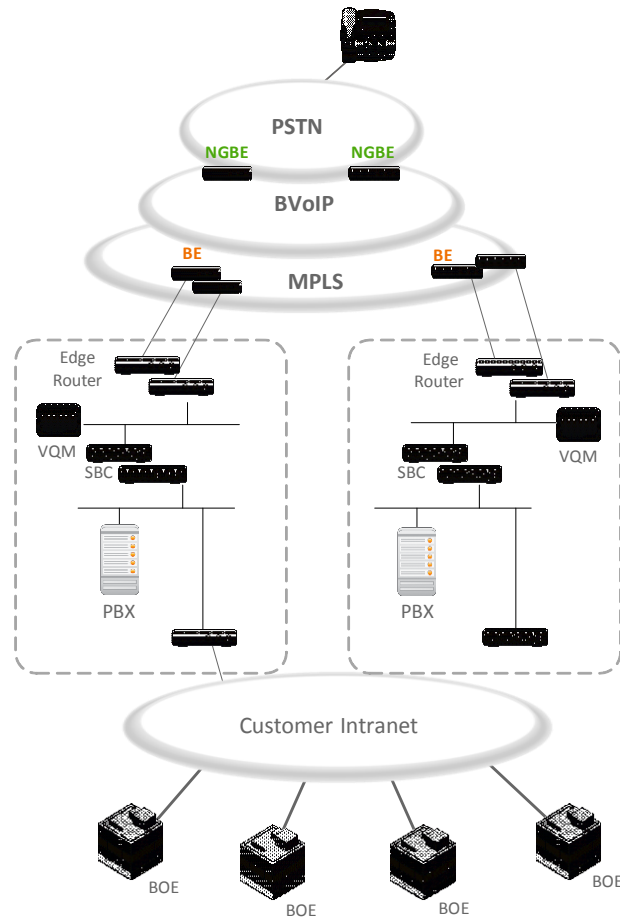


CCE = Call Control Element

PSAP = Public Safety Answering Point



SIP Trunking – Redundancy Overview



Description

- Dual Hub/Dual Trunk
- Active/Active
- Inbound Alternate Routing (IAR)
- Border Gateway Protocol Resiliency (BGP-R) for intrasite failure for call preservation
- Branch Office Extension (BOE)

Call Volume (Example)

- 1000 Total Calls

Class of Service 1 (Real-time)

- Each trunk configured with at least enough BW to cover 500 CCs



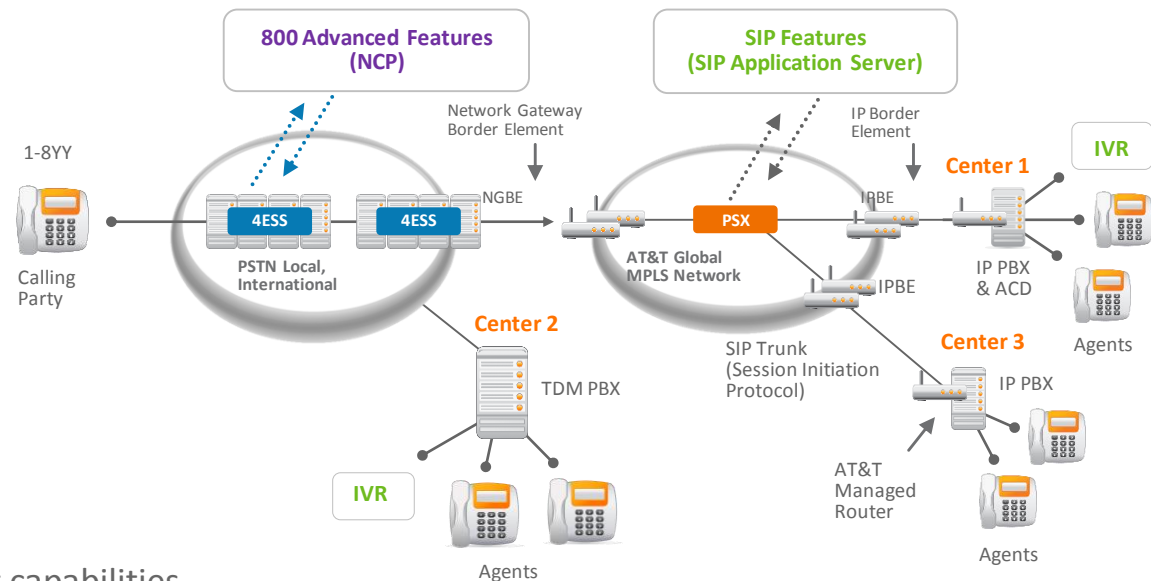
IP Toll Free

Inbound SIP trunking service delivering toll free calls over MPLS network.

Fully integrated routing across VoIP and TDM end points with Advanced Feature capabilities.

Highlights

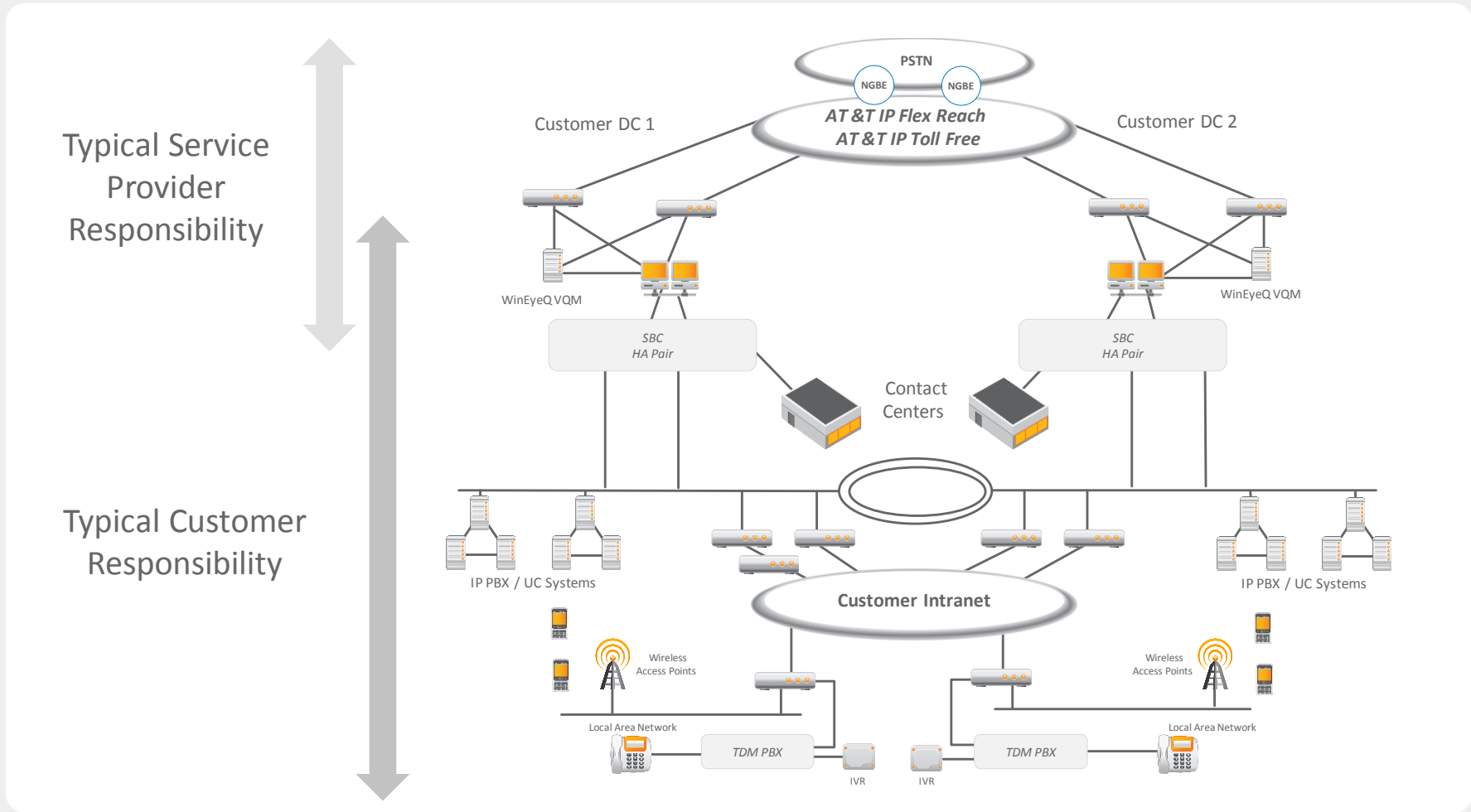
- Enabler for next-gen IP contact capabilities
- Converged voice and data, potential reduced access costs, enable more calls with voice compression
- Connects with new & existing analog, digital, & certified IP PBXs ACD's, IVR's
- Both hybrid (TDM & IP) or fully IP architecture
- Compatible with SIP Trunking



SIP Transformation Lessons Learned



Areas of Responsibility for SIP Infrastructure

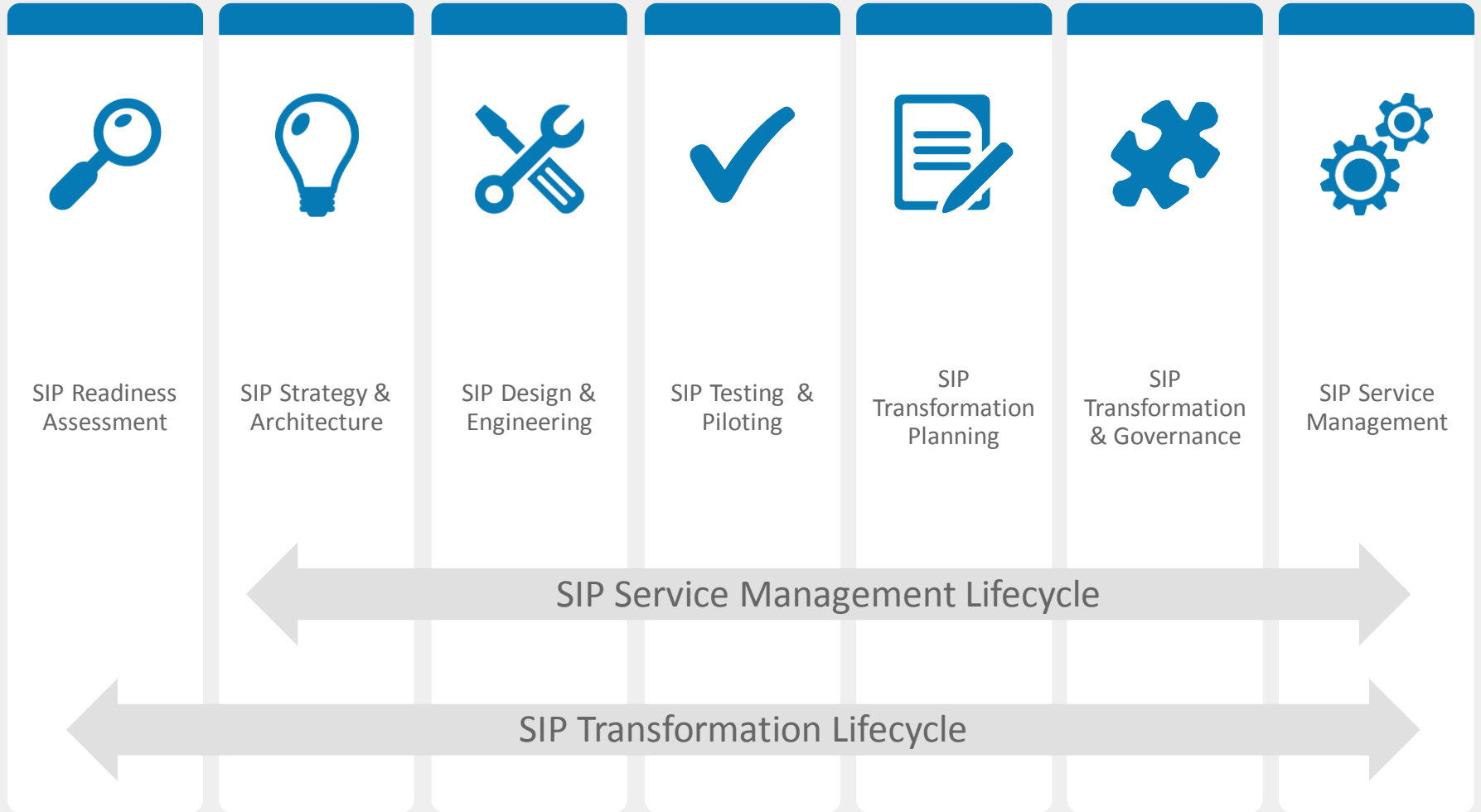


Challenges with SIP Transformation

- SIP deployment is not “plug and play”
- Customers need to understand how SIP will impact their network, contact center and UC environment
 - Availability
 - Performance
 - Resiliency
- Customers require an understanding of how SIP will impact the costs for voice and network services
- Customers may have resource availability or expertise constraints in IP PBX, UC, Contact Center & SBC platform configuration
- Service Providers assume that customers will have internal capability to support
- Customers often require front-end integration testing with existing systems
- Customers often require load testing the environment before shifting production traffic
- Implementation schedules are typically built around circuit provisioning times, not site readiness



SIP Transformation Phases



Healthcare Provider

Improve communications and collaboration with UC and SIP

Customer Challenges

- **Lack of direction** for voice and telephony transformation
- **Aging and failing** legacy PBX systems
- Lack of consistent voice & collaboration capabilities impeding **quality of care**
- **Inefficient utilization** across distributed PBX & voice trunking environment
- Inability to provide effective **operational support** across remote locations

Case Study



The transformation to centralized SIP trunks has provided cost savings due to the elimination of underutilized TDM circuits while increasing redundancy

Industry Benefits

Providing exceptional patient care requires teamwork and **access to information**. A UC strategy provided the roadmap to allow the clinical staff to **effectively communicate** with each other while **accessing data in real time**.

Customer Benefits

- **Comprehensive strategy roadmap and design** for UC and SIP migration
- **Improved operational support** with consistent and simplified architecture
- **Cost savings** with centralized SIP call routing



Government

Reduced costs via Managed UC Service with Centralized SIP

Customer Challenges

- Approximately **350 locations and 30,000 endpoints** with hundreds of distributed PBXs and **associated TDM trunks**.
- **Reduce costs** and deploy **flexible, customized voice services** to their constituent locations while providing high **availability, failover and centralized control**.
- **Complex scheduling requirements** that impacted SIP service **ordering, installation and cutover**.

Case Study



Upgraded voice and UC services to provide a single platform for delivering unified capabilities to all users at a lower total cost of ownership

Customer Benefits

- **Improved operational readiness** & management capabilities through standardization of voice services.
- **Realized cost reductions** by deploying **centralized SIP based voice services**

Customer Benefits

- Achieved SIP Transformation objectives through **effective coordination** of customer and AT&T project teams.
- **Successful pilot deployment** proved out solution capabilities.



Q&A



Thank you.



