1.4.14 Voice over Internet Protocol (IP) Transport Service (VoIPTS) [C.2.7.8]

Agencies will be able to interconnect voice communication systems in as many as 48 countries worldwide using high-quality, reliable, and comprehensive voice transport service from our secure voice over Internet protocol (VoIP) products. The service's ability to provide a variety of connectivity options for both new and existing communications equipment will give Agencies connectivity options that ease Agency transition to VoIP.

1.4.14.1 Technical Approach to Transport/IP/Optical Service Delivery [L.34.1.4.1]

1.4.14.1.a Approach to Service Delivery [L.34.1.4.1.a]

(a) Analyze the service requirements specified in this solicitation and descr be the approaches to service delivery for each service. [L.34.1.4.1.a]

With the introduction of voice over Internet protocol (VoIP) into the business enterprise, a need has arisen to provide Internet protocol (IP)-based voice interconnectivity for both IP-ready and traditional private business exchange (PBX) systems. During this period of industry transition to VoIP, Agencies need a solution that will help tie VoIP and traditional PBX systems together, as well as interconnecting to the public switched telephone network (PSTN). To better support Agencies in this transition, special features that have been developed in the PSTN, such as 911 access and private dialing plans, are retained in the AT&T VoIP architecture. **Figure 1.4.14.1-1** shows the basic voice over Internet protocol transport services (VoIPTS) architecture with Agency and PSTN interconnects.

[redacted]

Figure 1.4.14.1-1: AT&T's VoIP Network Approach Interconnects PBX Systems and PSTN. [redacted]

By 2008, there will be more than 40% of corporations using VoIP and nearly nine million consumers with VoIP service.

--USA Today
March 1, 2005

The AT&T approach to VoIP, described in **Table 1.4.14.1-1**, is for a system of application intelligent servers and routers that provide the AT&T IP-based voice service, which is similar to PSTN functionality using the benefits of an IP network. Using this self-contained, network-based VoIP approach, Agencies need only to connect to the customer edge (CE) router to gain access and operate their PBX network over IP. This approach also lends itself to PBX convergence with other IP voice-based products, such as telecommuting.

APPROACH	DESCRIPTION
Standards Compliance	Deliver service with applicable Internet Engineering Task Force (IETF) Request for Comment (RFC), International Telecommunications Union-Telecommunications (ITU-T), and National Institute of Standards and Technology (NIST) standards
Connectivity	• Both IP/Ethernet and T-1 interfaces and protocols are supported in CE • Agency transition to VoIP is made easier using access protocols and PBX support that allows connectivity to both IP PBX systems and existing traditional PBX systems • Provides local long distance VoIP and PSTN call routing, access to international calling, as well as access to Agency locations on virtual private voice network (VPVN) arrangement • AT&T VoIP to PSTN gateway allows calls to function in both directions and is integrated part of PSTN • Contiguous United States (CONUS) PSTN areas that are not directly connected to VoIP are accessed using existing AT&T voice network
Distributed Systems	Using combination of multiprotocol label switching (MPLS) network and distributed high-availability VoIP service nodes throughout network provides Agencies with PSTN-like call quality, including good call completion
Full Dial Plan Support	Support for both public and private dial plans built into VoIP network. This provides full call routing capabilities, supporting both public and private dialing plans without end point modification.
Complete PSTN Interconnect	• Service is fully interconnected to PSTN • CONUS PSTN areas that are not directly connected to VoIP are accessed using existing AT&T voice network • Architecture allows Agency calls in both directions as integrated part of PSTN • Links to services, such as 911, E911, Operator Services, Directory Assistance, and [redacted], are automatically included
Local and International Reach	• Assigned numbers (or numbers gained through local number portability [LNP] capability) are provided and routed through licensed interconnect in [redacted] • VPVN capabilities are provided in over 48 countries worldwide, allowing Agencies to use VoIP calling to offices in foreign lands
Service Quality and Reliability	• VoIP service is built on MPLS network that is extended to Agency customer premises equipment (CPE) • End-to-end VoIP quality that is similar to circuit-switched voice quality
Built in Protocol Conversion	• Protocol conversion and translation is built into network edge • Multiple protocol support, including H.323, session initiation protocol (SIP), integrated services digital network (ISDN) (Q.931), and media gateway control protocol (MGCP) provide access for several PBX system types

APPROACH	DESCRIPTION
Secure	• Security provided using combination of best practice IP network security, SIP network authentication, pinhole routing, and MPLS network separation • Agencies are provided with secure VoIP transport service • Defined security policies and practices safeguard denial of service (DoS) attacks, intrusion, and invasion of privacy

Table 1.4.14.1-1: Approach to VoIP Transport Service Provides Flexibility and Transition. *Agencies will be able to transition to VoIP more easily using the tools and capabilities that are available in the AT&T VoIPTS offer.*

The AT&T Voice over IP Any Distance service delivery network provides the capability to interconnect both traditional and IP-based PBX systems, as well as providing access to local, long distance. Access to international calling services is also available. The Voice over IP Any Distance product is based on the same core network and principles as the other AT&T VoIP products. This means that not only can Agencies use the service to interconnect PBX systems, but also they will have the ability to integrate this solution with other VoIP offers, such as IP-Centrex and Remote Teleworker services.

1.4.14.1.b Benefits to Technical Approach [L.34.1.4.1.b]

(b) Describe the expected benefits of the offeror's technical approach, to include how the services offered will facilitate Federal Enterprise Architecture objectives (http://www.whitehouse.gov/omb/egov/a-1-fea.html). [L.34.1.4.1.b]

AT&T's Networx services and VoIP services, in particular, support the Government's vision of transformation through the use of the Federal Enterprise Architecture (FEA) by providing the technologies that contribute to the Agency's mission objectives. **Table 1.4.14.1-2** describes each service in relation to FEA, summarizes its contribution, and/or provides an example of how it facilitates FEA implementation.

APPROACH	BENEFIT	FEA FACILITATION
Connectivity	Both IP/Ethernet and T-1 interfaces and protocols are supported in CE.	Using variety of interfaces allows Agencies to integrate VoIP with both IP-based and traditional telecommunications topologies. **FEA Link:** TRM/Service Interface and Integration/Interface/Service Description/Interface
Distributed Systems	VoIP components distributed across MPLS network offer PSTN-like call quality.	Combination of MPLS network and distributed high-availability VoIP service nodes provides Agencies with good voice quality as well as good call completion rates that rival traditional PSTN-based services. **FEA Link:** TRM/Service Platform and Infrastructure/Hardware Infrastructure/Network Devices/Standards

APPROACH	BENEFIT	FEA FACILITATION
Full Dial Plan Support	Support for both public and private dial plans built into VoIP network for Agency calling without change.	Service is integrated with VoIP network that offers both private call routing for voice VPNs and public dial routing worldwide. **FEA Link**: TRM/Service Interface and Integration/Interface/Service Description/Interface
Complete PSTN Interconnect	Service is fully interconnected to PSTN. This removes complexity of connecting to PSTN from Agency.	Architecture allows Agency calls to function in both directions as integrated part of PSTN. Links to services, such as 911, E911, and [redacted] are automatically included. **FEA Link:** TRM/ Service Access and Delivery/Delivery Channels/Extranet
Local and International Reach	Service built into more than [redacted] footprint for wide local number availability.	Gateways are deployed throughout network, providing Agencies with direct local access covering over 90% of U.S. population. Remaining CONUS interconnect capability is covered by extensive circuit switched systems. VoIP transport is available in over 48 countries, allowing Agencies to call offices in foreign lands without leaving VoIP network. **FEA Link:** TRM/Service Access and Transport/Service Transport/ Service transport
Service Quality and Reliability	Calls are made to and from PSTN or on-net with PSTN-like quality.	VoIP service is built on MPLS network that is extended to Agency CPE. This provides end-to-end VoIP quality that is similar to circuit-switched voice quality. **FEA Link:** TRM/Service Access and Transport/Service Transport/ Service Transport
Built in Protocol Conversion	Support for multiple PBX protocols allows Agencies to use VoIP with wider variety of systems.	Agencies' transition to VoIP is made easier using multiple access protocols and variety of supported PBX systems. This allows connectivity to both IP PBX systems and existing traditional PBX systems. **FEA Link:** TRM/Component Framework/Business Logic/Platform Independent
Secure and Reliable	Use of dedicated VPNs and security practices reduces Agency risk.	Combining VPN access and VoIP core service networks provides better security and reliability than using Internet for VoIP. Agency telephone traffic is not placed in vulnerable networks without performance guarantees. **FEA Link:** TRM/Component Framework/Security

Table 1.4.14.1-2: Agencies Receive Several Benefits Using AT&T VoIPTS Instead of Internet for VoIP. *Agencies will receive a secure and provides a high level of performance. An Internet-based calling service cannot provide comparable performance.*

AT&T's development of net-centric technologies supports solutions based on service oriented architecture (SOA), which uses standardized, web-adapted components. Our approach ensures that the criteria listed below are followed:

- Technical Reference Model capabilities are fully met and linked to the Service Component Reference Model (SRM) and Data Reference Model (DRM).
- These links are structured to support Business Reference Model (BRM) functions and provide line-of-sight linkage to mission performance and ultimate accomplishment per the Performance Reference Model (PRM).
- AT&T operates as an innovative partner through Networx to help achieve the vision of the FEA to enhance mission performance.

In addition to the benefits and FEA facilitations cited earlier, AT&T will assist specific departments and Agencies to meet mission and business objectives through a comprehensive VoIPTS offering.

1.4.14.1.c Major Issue to Service Delivery [L.34.1.4.1.c]

(c) Describe the problems that could be encountered in meeting individual service requirements, and propose solutions to any foreseen problems. [L.34.1.4.1.c]

In transitioning into any new service delivery model, whether it be task-based or fully outsourced, unforeseen issues may arise. Therefore, it is important that GSA selects a service provider, such as AT&T, which brings the depth and background that minimize an Agency's risk during transition. Our experience has enabled us to develop proven methods, processes, and procedures applicable to the simplest or the most complex projects.

Table 1.4.14.1-3 presents six risks involved with transitioning to VoIPTS. As with all large projects, we enter each of these risks and others (after identification and characterization) into our risk-tracking database, and immediately take steps to mitigate them before they become an issue. Because risk management is more effective when all stakeholders are active in the process, AT&T engages the GSA, the client Agency, and other Government solution partners for success with risk mitigation activities.

Risk	Risk Description	Risk Mitigation
Product Maturity	Since VoIP products are new in industry, Agencies might not have equipment that is supported by all VoIP carriers.	[redacted]
Lack of Standards	No specific standards cover VoIP interconnection for all PBX systems.	[redacted]
Local Area Network (LAN) Configuration	Converting to VoIP transport could require some Agency LAN and firewall changes.	[redacted]
Access Capacity	Lack of bandwidth in VoIP access system can affect overall call quality.	[redacted]
Security	VoIP is IP service and can be vulnerable to typical attacks and transmission anomalies in Internet.	[redacted]

RISK	RISK DESCRIPTION	RISK MITIGATION
Product Longevity	New products are often eclipsed by next generation of service leaving early adopters with useless equipment.	[redacted]

Table 1.4.14.1-3: Typical VoIP Risks are Mitigated by AT&T Engineering and Implementation Personnel. *The implementation of any new service or topology carries some risk. Using the services from AT&T to complete the transition to VoIP will help Agencies complete their transition without experiencing the typical pitfalls.*

> *Past implementations of VoIP transport have been crippled by poor packet delivery or lacked connectivity to the PSTN. Other solutions have limited customers to using a single VoIP protocol, CPE vendor, or operating architecture. Agencies can have combined access and high quality multi-protocol VoIP in a single solution from AT&T that includes a full PSTN interconnect.*

AT&T has taken steps to identify risks and provides mitigation for risks that are associated with the VoIPTS service offer. AT&T is committed to service excellence and will work with the Agency to identify and resolve potential problems that might occur during service delivery.

1.4.14.1.d Network Architecture Synchronization [L.34.1.4.1.d]

(d) Describe the synchronization network architecture to support the offeror's access and transport networks. [L.34.1.4.1.d]

AT&T is a leader in the area of network synchronization, by virtue of our active role in the international and domestic standards organizations. We have an existing industry-unique, dedicated timing and synchronization network for distributing Stratum 1 traceable timing to our own national and international telecommunications networks.

Synchronization for access and transport networks begins with the Federal Government's cesium-based standard signal, which is distributed to a series of global positioning satellite (GPS) systems. AT&T derives synchronization from those GPS systems as the primary clock source. [redacted] VoIPTS derives its synchronization from the synchronous optical network (SONET) infrastructure on which those services ride.

[redacted]. A more detailed discussion on network synchronization is provided in Section 1.3.6.1, Network Architecture Synchronization.

1.4.14.2 Satisfaction of Transport/IP/Optical Performance Requirements [L.34.1.4.2]

1.4.14.2.a Service Quality and Performance [L.34.1.4.2.a]

(a) Describe the quality of the services with respect to the performance metrics specified in Section C.2 Technical Requirements for each service. [L.34.1.4.2.a]

To support Agency VoIP services, the AT&T network is built to perform at a level of quality that is required for acceptable voice quality, along with reliable call completion rates; network impairments are controlled network-wide. **Table 1.4.14.2-1** lists the key network performance indicator levels, as compared to the requirements in the RFP.

The use of MPLS exclusively in the AT&T VoIP core, combined with high capacity inter-router links, provides VoIP packets with very good routing performance characteristics.

Key Performance Indicator (KPI)	Service Level	Performance Standard (Threshold)	Proposed Service Quality Level
CONUS Round Trip Latency	Routine	<200 ms	[redacted]
End-to-End Packet Loss	Routine	<0.4%	[redacted]
Availability	Routine	>99.6%	[redacted]
	Critical	>99.9%	[redacted]
Packet Jitter	Routine	<10 ms	[redacted]
Time to Restore	Without Dispatch	4 hr	[redacted]
	With Dispatch	8 hr	[redacted]

[redacted]

Table 1.4.14.2-1: Service Quality and Performance. *AT&T's performance levels exceed the minimum requirements.*

Combining the high-capacity efficient network, with high-availability VoIP call processing equipment, provides Agencies with a VoIP service that has both PSTN-like sound quality, and high call completion rates. AT&T VoIP network call quality, as compared to a typical IP network, such as the Internet, is shown in **Figure 1.4.14.2-1**.

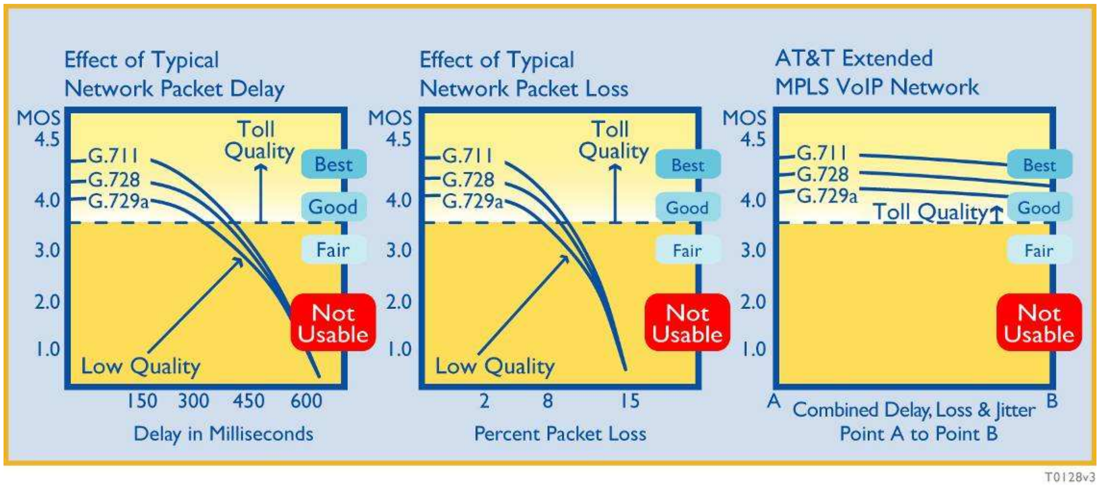

Figure 1.4.14.2-1: AT&T Network Outperforms Typical Networks for VoIP. *Excessive packet loss, delay, or jitter can affect overall call quality. AT&T VoIP network carefully controls impairments to supply a higher quality VoIP service.*

Overall call quality is a combination of call completion and sound quality. Since network errors can affect both call completion and sound quality, AT&T provides Agencies with a service that operates within a dedicated MPLS core routing system, which provides both real-time protocol (RTP) and signaling packets with the highest possible quality transport. This, coupled with the distributed multiserver call handling environment, provides an overall high quality of service.

1.4.14.2.b Approach to Monitoring and Measuring Performance [L.34.1.4.2.b]

(b) Describe the approach for monitoring and measuring the Key Performance Indicators (KPIs) and Acceptable Quality Levels (AQLs) that will ensure the services delivered are meeting the performance requirements. [L.34.1.4.2.b]

Of equal importance to identifying the KPIs for a service is the method by which the KPIs are captured, measured, and monitored. Agencies will receive the most accurate assessment of the service when the KPI measurement and monitoring methodology replicates the real performance experienced by Agency personnel. To provide the Agencies with the most accurate representation of the service performance, AT&T has deployed a separate service performance measurement infrastructure to collect network performance information. AT&T's measurement methodology, therefore, more closely captures the real performance that end users experience by measuring the data path that is very similar to the paths that the end user data would follow. **Table 1.4.14.2-2** lists the methods used to measure the various IP key performance indicators (KPIs).

KEY PERFORMANCE INDICATOR (KPI)	APPROACH TO MONITORING AND MEASURING
Ability to measure network and service specific metrics	[redacted]
Availability measured (service delivery point [SDP]-to-SDP)	
Time to Restore (TTR)	[redacted]
[redacted]	[redacted]

Table 1.4.14.2-2: Monitoring and Measuring IPTelS. *Agencies can easily manage the VoIPTS with easy to access and use data delivered through AT&T **Business**Direct.*

In the AT&T VoIP network, call quality and network transport quality are monitored at each network element using task specific element management systems (EMS), as shown in **Figure 1.4.14.2-2**.

[redacted]

Figure 1.4.14.2-2: Management Network [redacted]

Using the dedicated management network service, performance is monitored and displayed in units that are appropriate for each product. Agencies are provided access to the performance data, using links within AT&T **Business**Direct.

The first time the service is provided through the Networx contract, the performance must be verified. The KPIs will be monitored to certify that the service performance complies with the AQL. [redacted] The service verification process is presented in greater detail in Section 1.3.2.d, Approach to Perform Service Delivery Verification.

1.4.14.2.c Performance Level Improvements [L.34.1.4.2.c]

(c) If the offeror proposes to exceed the Acceptable Quality Levels (AQLs) in the Key Performance Indicators (KPIs) required by the RFP, describe the performance level improvements. [L.34.1.4.2.c]

The AT&T core VoIP network is based on MPLS routing between all entry and exit points. The performance characteristics of MPLS, combined with high capacity router-to-router links, create a high-performance network that [redacted]

Table 1.4.14.2-3 lists the key AT&T network performance targets, as compared to the RFP performance targets, and denotes the percentage improvement over the Networx performance threshold marks.

METRIC	NETWORX AQL THRESHOLD	PROPOSED SERVICE QUALITY LEVEL	IMPROVEMENT PERCENTAGE
[redacted]	[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]	[redacted]
[redacted]			[redacted]

Table 1.4.14.2-3: [redacted] **Performance Specifications.** [redacted]

Using the VoIP core network with the above set of packet performance criteria, VoIP calls rival the quality of PSTN calls. This quality applies to both the speech quality and call completion rate, as the network performance criteria apply to both the talk path network and the call setup and signaling network. A network that degrades the signaling path in favor of the talk path can cause incomplete and dropped calls. AT&T provides complete call quality through networks that support high call completion rates and call sound quality.

End-to-end VoIP performance depends on both core network performance and access network performance. AT&T staff members calculate the customer access bandwidth requirements, based on recommended site-specific services and usage. For Agencies to receive the needed levels of performance, the AT&T site-specific bandwidth recommendations for access arrangements must be followed.

1.4.14.2.d Rationale and Benefits for Additional Performance Metrics [L.34.1.4.2.d]

(d) Describe the benefits of, rationale for, and measurement of any additional performance metrics proposed. [L.34.1.4.2.d]

AT&T uses the [redacted] to measure the quality of each call in the VoIP network as it passes into or out of the core. The output is expressed in [redacted]. AT&T proposes the [redacted] as summarized in Table **1.4.14.2-4.**

AT&T is the only carrier that offers R-Factor that is an industry standard VoIP performance measure that is equivalent to mean opinion score (MOS).

PROPOSED KPI	DESCRIPTION OF PROPOSED KPI	BENEFIT OF PROPOSED KPI
[redacted]	[redacted]	[redacted]

Table 1.4.14.2-4: Performance Level Improvements. *Incorporating the proposed KPI into the VoIPTS provides comprehensive performance monitoring, allowing Agencies to migrate to performance-based contracts sooner.*

[redacted] (**Figure 1.4.14.2-3**).

[redacted]

Figure 1.4.14.2-3: Call [redacted]

By monitoring the call quality, more information is gained about how a call is actually performing than is gained by monitoring packet loss, jitter, and latency independently. In addition, monitoring call quality at every edge point gives a directional indicator as to which network or component is at fault in a poor quality call.

1.4.14.3 Satisfaction of Transport/IP/Optical Service Specifications [L.34.1.4.3]

1.4.14.3.a Service Description [L.34.1.4.3.a]

(a) Provide a technical description of how the service requirements (e.g., capabilities, features, interfaces) are satisfied. [L.34.1.4.3.a]

The VoIPTS offer is based on the AT&T layered VoIP architecture. The VoIP architecture uses the upper layer of the network to route calls throughout the network. The midsection of the network is where end devices are tracked and protocol differences are handled. The components closest to the customer provide quality routing through VPN access as well as links to the PSTN.

Each layer provides a vital function to the overall call completion process. Each layer also operates within a particular network layout. **Tables 1.4.14.3-1** through **1.4.14.3-3** describe the functions of the network that is shown in **Figure 1.4.14.3-1**.

The first layer of the VoIPTS network is typically called the access layer. This layer contains the access and VPN systems that provide connectivity to the AT&T VoIP network and control the transport quality of the service. Included in the layer are VPN services, links to the PSTN, and peering to other VoIP networks. These networks are all constructed with security and quality of service built in.

LAYER ONE - ACCESS

Service Element	Description	Benefit
VPN	VoIPTS is provided over MPLS network-based VPN.	Access is through VPNs that use MPLS and provides PSTN-like quality.
Class of Service (CoS) Routing	Packets are marked at edge for differentiated preference routing.	CoS is created for all traffic to and from CPE; VoIP packets are routed in highest priority queue, which helps create consistent packet delivery end-to-end.
Managed Access	Access VPNs and termination equipment are provided as managed service.	Access service is monitored and managed by AT&T to provide consistent results and fault management.
VoIP-specific CPE	Agency PBX systems are connected to network using VoIP-enabled routers.	Access termination equipment is specially optioned to handle PBX and IP PBX systems and configured to perform well in VoIP environment.
VoIP-specific Interfaces	• Ethernet RJ-45 • Analog Trunk, 2 Line • Analog Trunk, 4 Line • Digital Trunk, T-1 • Digital Trunk, ISDN PRI • Digital Trunk, E-1	Agencies can connect to virtually any type of PBX system, including both existing TDM-based, and newer IP-ready PBX systems.

Table 1.4.14.3-1: Access – Layer One. *Access strategies play an important role in the quality of VoIP services, including initiating transport quality of service and providing security.*

The second layer contains the border edge (BE) elements that help control access, quality, and security, along with providing protocol conversion, as needed, to operate differing devices that are attached in the access network. The types of BE systems in the network are the customer-facing border element (BE) devices, the PSTN gateways (network gateway BE [NGBE]), and other access gateways (such as the IP gateway BE [IPBE]). These gateways provide secure access to the VoIP core transport and switching network as well as some session control for the session-based signaling protocols, such as SIP.

LAYER TWO – BORDER CONTROL		
Service Element	**Description**	**Benefit**
Network Security	BE is an application layer gateway (ALG) that acts as a security device.	• Tracks call setup information and only allows authenticated call traffic to authorized end points. • BE is able to block: rouge packet streams, typical network based attacks, and attempts to use service in unauthorized manner.
Network Address Translation	Partial BE function is to perform as network address translator.	• IP addressing is more freely allocated within Agency VPN, including RFC-1394 addressing. • Allows VoIP interconnect from differing IP networks.

Multiple Protocol Support	BE acts as protocol converter for VoIP signaling protocols.	• BE provides translations for and supports:
		  • SIP
		  • Media gateway control protocol (MGCP) and media gateway control (MEGACO)
		  • H.323
		• Allows differing Agency devices to access service and interconnect with each other.

Call Quality Measurement Points	[redacted] measurements provided for every call.	• Call quality tracked at every BE • Call quality also tracked at every NGBE • Two points of call quality checking, measuring call quality in and out of core network • Performance data allows Agencies to track VoIP network performance
PSTN Interconnect	PSTN gateway (NGBE) converts VoIP into traditional, circuit-switched voice and provides appropriate ingress/egress point for voice traffic.	• NGBE works in conjunction with call control engine (CCE) and signal system 7 (SS7) gateway to pass and route VoIP calls properly to and from PSTN • Allows Agencies VoIP service to be fully integrated with worldwide PSTN • Processes all call traffic so Agencies don't have to know network
Special Services Interconnects	Gateways to special telephony functions.	These gateways include access to: • [redacted] • Geographically proper public safety answering points (PSAPs) for 911 call handling. • Operator Services • Directory Assistance

Table 1.4.14.3-2: Border Edge – Layer Two. *The border between the VoIP domain and the remaining networks is secured by gateways called border elements. These devices set up access for voice streams into and out of the VoIP network and monitor calls using the R-Factor measuring system.*

The third layer, the VoIP core network, operates within the MPLS core. It is primarily based on the AT&T call control engine (CCE), which routes VoIP calls to the proper end points, based on routing control and numbering information. These elements perform IP to PSTN number translation, manage gateway access, and provide private and public dial plan capabilities. This layer also provides security as well as call completion rate quality for VoIP customers.

LAYER THREE – VOIP CORE		
Service Element	***Description***	***Benefit***
Worldwide Reach	VoIP core's reach provides voice VPN service to m[redacted]	Agencies can communicate over private voice networking in countries around world.
Call routing for local numbers	Local number access to CONUS PSTN [redacted]	• Agencies can transition existing local numbers to VoIP as well as acquire new local numbers in new locations. • VoIPTS uses LNP so there is no need to generate new numbers to transition to VoIP.
Interconnected with AT&T time division multiplexing (TDM) voice network	Outlying CONUS PSTN locations carried by AT&T TDM to local handoff.	Agency interconnection to worldwide PSTN is augmented by existing AT&T participation in overall PSTN.
CoS	CoS routing is maintained through core using MPLS.	• Agency VoIP traffic is passed end-to-end with CoS tags intact. • Differentiated routing is applied in ingress access network. • Differentiated routing acts in egress network. • Provides overall high sound quality for calling.
Scalable call routing infrastructure	CCE call routing engine uses SIP signaling that matches PSTN's E.164 telephone number routing hierarchy for scalability and completes PSTN interoperability.	• Agencies do not need to provision special routing into their communications equipment to use VoIP network. • AT&T VoIP network automatically converts all E.164 public and private numbering to proper IP addressing, including dynamic IP addressing. • This call routing infrastructure scales to very large numbers automatically with growth of service so Agencies do not need to make network changes as service usage grows.
Special routing functions	CCE provides special call routing functions including private dial plans and access to 911, [redacted] etc.	• Agencies can depend on calling features, such as 911 access, operator services, and directory assistance. • Supports ability to use abbreviated dialing plans that are already in place in many offices.
High quality packet routing	VoIP core provides very low latency and high-packet delivery reliability.	• MPLS network provides Agencies with quality transmission for all VoIP calls. • For overall quality service. (Refer to Section 1.4.15.2.c for actual performance metrics.)

Table 1.4.14.3-3: VOIP Core - Layer Three. *Call routing and access to the PSTN are facilitated by the VoIP core components. In addition, number translation between IP and E.164 is provided as well as access to both public and private dialing plans.*

The network described in the above tables is depicted in **Figure 1.4.14.3-1**. This VoIP network is layered on top of the high performance MPLS network that is the core of all AT&T packet network products.

[redacted]

Figure 1.4.14.3-1: VoIP Network for PBX Interconnect. [redacted]

Using the three groupings (layers) of components in the AT&T VoIP service architecture, Agencies will receive a telephone service that is both flexible and reliable. AT&T handles the details and complexities of telephony. Thus, Agencies can concentrate on their own missions.

1.4.14.3.b Attributes and Values of Service Enhancements [L.34.1.4.3.b]

(b) If the offeror proposes to exceed the specified service requirements (e.g., capabilities, features, interfaces), describe the attributes and value of the proposed service enhancements. [L.34.1.4.3.b]

In addition to interconnecting both IP and traditional PBX systems, Agencies can enhance their capabilities with additional features, products, and capabilities that might require an additional fee. **Table 1.4.14.3-4 and 1.4.14.3-5 highlight** additional features and capabilities of the VoIPTS and the VoIP network.

TRANSITION SUPPORT FEATURES		
[redacted]		
[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]

Table 1.4.14.3-4: [redacted]p Agencies Move to VoIP. *These interoperability features give Agencies more choices when transitioning offices and workers to VoIP.*

GENERAL FEATURES		
Service Enhancement	*Description*	*Benefit*
[redacted]	[redacted]	▪ [redacted] ▪ [redacted]
[redacted]	[redacted]	▪ [redacted]
[redacted]	[redacted]	▪ [redacted] ▪ [redacted]
[redacted]	[redacted]	▪ [redacted]
[redacted]	[redacted]	▪ [redacted]
[redacted]	[redacted]	▪ [redacted] ▪ [redacted]

Table 1.4.14.3-5: General features make the AT&T VoIPTS a complete telephone service. *The VoIPTS provides robust interoperability with other VoIP services.*

With these additional features and functionality, Agencies will be able to transition to VoIP, integrate their telecommunications infrastructure, and add additional capabilities to their calling systems. These features are created by the built-in VoIP network capabilities that are all part of the AT&T [redacted] service network.

1.4.14.3.c Service Delivery Network Modifications [L.34.1.4.3.c]

(c) Describe any modifications required to the network for delivery of the services. Assess the risk implications of these modifications. [L.34.1.4.3.c]

Agencies will receive a low-risk VoIPTS solution with no modifications necessary to the [redacted]. Upon award of the contract, AT&T is able to provide the service without modifications to the network or the operational systems.

1.4.14.3.d Transport/IP/Optical Service Experience [L.34.1.4.3.d]

(d) Describe the offeror's experience with delivering the mandatory Transport/IP/ Optical Services described in Section C.2, Technical Requirements. [L.34.1.4.3.d]

[redacted] **Table 1.4.14.3-6** describes AT&T's relevant experience working with a civilian Government Agency.

[redacted]		
Client Need	*Solution*	*Create Value*
[redacted]	[redacted]	[redacted]

Table 1.4.14.3-6: Experience Delivering Managed VoIP Services. *AT&T measures success by our ability to deliver solutions to our customers who create value to their business.*

AT&T offers experience and expertise in networking, VPN technology, and telephone systems. Bringing these three sets of expertise together at the same time provides a unique capability to offer VoIP services that truly rival the PSTN in quality and reliability.

1.4.14.4 Robust Delivery of Transport/IP/Optical Services [L.34.1.4.4]

1.4.14.4.a Network Traffic Utilization [L.34.1.4.4.a]

(a) Given the offeror's current network capacity and utilization, explain how the offeror will support the Government requirements specified in the traffic model. Describe the impact on capacity and utilization, as well as any infrastructure build out contemplated. [L.34.1.4.4.a]

To assess the impact of the Agencies' VoIPTS traffic on the AT&T network, the forecasted traffic in the Networx hosting model has been compared against the growth of AT&T's network. [redacted]. **Table 1.4.14.4-1** describes the impact of the Agencies' forecasted VoIPTS traffic growth on the use of the key elements from Years 2 through 10. ([redacted].)

ESTIMATED USE DATA					
Contract Year	*MOU Per Month*[1]	*MOU Per Day*[2]	*PSTN Trunks*[3]	*Bandwidth*[4]	*% Network Bandwidth*[5]
1	[redacted]	[redacted]	[redacted]	[redacted]	[redacted]
2	[redacted]	[redacted]	[redacted]	[redacted]	[redacted]
3	[redacted]	[redacted]	[redacted]	[redacted]	[redacted]
4	[redacted]	[redacted]	[redacted]	[redacted]	[redacted]
5	[redacted]	[redacted]	[redacted]	[redacted]	[redacted]
6	[redacted]	[redacted]	[redacted]	[redacted]	[redacted]
7	[redacted]	[redacted]	[redacted]	[redacted]	[redacted]
8	[redacted]	[redacted]	[redacted]	[redacted]	[redacted]
9	[redacted]	[redacted]	[redacted]	[redacted]	[redacted]
10	[redacted]	[redacted]	[redacted]	[redacted]	[redacted]

[redacted]

Table 1.4.14.4-1: Use of Network, Based on GSA Estimates. [redacted]

To switch Agency calls to the PSTN without blocking, AT&T has access to over [redacted] interconnect ports [redacted]. In addition, the growth of the IP backbone network from [redacted] will outpace the growth of the VoIP bandwidth usage. The overall bandwidth usage of the Government's VoIP traffic is maintained at [redacted] of the aggregate backbone traffic capability.

1.4.14.4.b System Robustness and Resiliency [L.34.1.4.4.b]

(b) Describe the measures and engineering practices designed to provide robustness of the access and backbone networks, ensure resiliency, and plan for growth. [L.34.1.4.4.b]

Rigorous engineering practices and measurements of the network allow Agencies to obtain a scalable, reliable service to build and operate their mission-critical applications. IP service and backbone capacity planning within the AT&T IP network are driven by three main factors (**Table 1.4.14.4-2**).

MAJOR CAPACITY PLANNING FACTORS	
Business Planning	Annual business planning forecasts of all existing and new AT&T services that use IP backbone network and connected service give combined prediction of use.
Technology Migrations	Capacity for planned technology migrations and insertions are built into system before migration is started.
Historic Growth	Historic traffic growth of existing services, as measured over time, allows for buildout, based on increasing use by AT&T customers.
TYPICAL APPLICATION OF FACTORS	
Call Loads	Call loads to incumbent local exchange carriers (ILECs), competitive local exchange carriers (CLECs), and other inter-exchange carriers (IXCs) are all tracked on a trunk group basis and have additional capacity added based on three principles above.
Use of a Service	Each of above principles is also applied to components and systems that constitute service, such as VoIP.

Table 1.4.14.4-2: Key Capacity Planning Factors. *Network capacity buildout is based on both predictive and measured data. AT&T strives to provide service from a network with more than enough capacity to do the job and grow.*

Because AT&T's IP backbone network is a two-tier architecture with an MPLS network core that does not contain any Internet routes (Internet-route free core), the VoIPTS provided to the Agencies is secure, scalable, and reliable. As the number of Internet routes increased dramatically during the late 1990s, the stability of the IP backbone network was at risk. AT&T recognized this threat and engineered a two-tier architecture that moved the Internet routes to the edge routes, providing a stable, route-free MPLS core. Other IP-based services, such as VoIPTS, leverage the secure and stable, route-free MPLS core for Agencies' applications.

1.4.14.5 Transport/IP/Optical Service Optimization and Interoperability [L.34.1.4.5]

1.4.14.5.a Approach to Optimizing IP-based and Optical Services [L.34.1.4.5.a]

(a) Describe the offeror's approach for optimizing the engineering of IP-Based and Optical Services. [L.34.1.4.5.a]

A detailed discussion of our approach to optimizing the engineering of IP-based and optical services is provided in Section 1.3.6.2.a.

1.4.14.5.b Network Architecture Optimization [L.34.1.4.5.b]

(b) Describe how the offeror will utilize methods such as remote concentration, switching/routing capabilities, and high bandwidth transmission facilities to optimize the network architecture. [L.34.1.4.5.b]

Optimization of the network architecture through the use of remote concentration, switching/routing capabilities, and high bandwidth transmission facilities is described in Section 1.3.6.2.b.

1.4.14.5.c Optimizing Engineering Techniques [L.34.1.4.5.c]

(c) Describe the engineering techniques for optimizing access for improved performance or increased efficiency in areas where large concentrations of diverse customer applications exist (e.g., the use of multi-service edge platforms). [L.34.1.4.5.c]

Optimization of the access for improved performance or increased efficiency through the use of multiservice edge (MSE) platforms is described in Section 1.3.6.2.c.

1.4.14.5.d Vision to Implement Service Internetworking [L.34.1.4.5.d]

(d) Describe the offeror's vision for implementing service internetworking over a common infrastructure (e.g., IP-centric architecture). Include a view on network interoperability, control plane integration, and optical infrastructure support for IP-Based Services. Describe the benefits and rationale of the offeror's approach. [L.34.1.4.5.d]

The implementation of service internetworking over a common infrastructure, including network interoperability, control plane integration, and optical infrastructure support, is described in Section 1.3.6.2.d.

1.4.14.6 Narrative Text Requirement

1.4.14.6.1 Routing Prioritization [C.2.7.8.1.4 (2)]

The following Voice over Internet Protocol Transport Service capabilities are mandatory:
2. The contractor shall enable a routing prioritization scheme or class of service to distinguish between IP services.

The access links for the AT&T VoIP services are constructed using VPNs that are carried to the network core with CoS markings for most packets. At the edge of the network, the packets are classified and marked by application type for differentiated routing in the network [redacted], as shown in **Figure 1.4.14.6-1**.

[redacted]

Figure 1.4.14.6-1: Differentiated Services [redacted]

[redacted]

- [redacted]
- [redacted]
- [redacted]
- [redacted]

Packet traffic is classified with the CoS tags at the customer premises router. The customer premises router, through extended access lists, identifies the voice traffic using the transmission control protocol (TCP), user datagram protocol (UDP), the network of origination, application port, or socket.

Once classified, the network routers act on the marked packets, according to their level of criticality. [redacted] **Figure 1.4.14.6-2** depicts the classification of packets [redacted]

[redacted]

Figure 1.4.14.6-2: VoIP Packets [redacted]

Refer to Section 1.3.2.e for more details on the overall network routing priority system used in the AT&T networks.

1.4.14.6.2 Private Dial Plans [C.2.7.8.1.4(3)(d)]

Unlike our competitors, AT&T provides CoS across our IP VPN not just at the edge of the network. This offers true end-to-end QoS to our customers and provides a suitable environment for IP voice and converged services.

The following Voice over Internet Protocol Transport Service capabilities are mandatory:
3. The contractor shall provide the following minimum capabilities:
d. The contractor's VoIPTS shall interoperate with private Agency network dial plans.

Private dial plan routing is built into the VoIP call routing network. No special additional systems need to be deployed to support private dial plans. Using the private dial plan support in the AT&T VoIP network enables Agencies to interconnect to PBX systems and retain a comparable level of dialing plans used with the PSTN-based private dialing.

The AT&T VoIP network architecture supports private dial plans in the CCE and network routing engines. The private dial plans are fully integrated with public dial plan calling. The basic features of the VoIP-based private dial plan are listed in **Table 1.4.14.6-1**.

PLAN SCOPE	[redacted]
PLAN ACCESS NUMBERS	[redacted]
DIGITS USED	[redacted]
SITE CODES	[redacted]
EXTENSION CODES	[redacted]
EXTENSION RANGE	[redacted]
PUBLIC NUMBERS CAN BE ALLOWED	[redacted]
PUBLIC NUMBERS CAN BE BLOCKED	[redacted]
PRIVATE VS. PUBLIC	[redacted]
NUMBER MATCHING AND FALL THROUGH	[redacted]
ROUTE OF LAST RESORT	[redacted]

Table 1.4.14.6-1: Private Dial Plan Features. *Dialing plan routing provides a wide range of numbering rules and attributes.*

A customer, for example, has three locations: New York, Denver, and Virginia. New York and Virginia are on an abbreviated dialing plan, while the Denver site is accessed using the 10-digit public number. **Table 1.4.14.6-2** lists examples of dialing sequences that are supported by the public and private dialing plans.

LOCATION	PUBLIC NUMBER RANGE	STATION DIGITS	SITE PREFIX*	PBX EXTENSION	OFFSITE PRIVATE DIALED NUMBER	ONSITE DIALED NUMBER
New York	[redacted]	[redacted]	[redacted]	[redacted]	[redacted]	[redacted]
Virginia	[redacted]	[redacted]	[redacted]	[redacted]	[redacted]	[redacted]
Denver	[redacted]	[redacted]	[redacted]	[redacted]	[redacted]	[redacted]
[redacted]						

Table 1.4.14.6-2: Private Dialing Plans. *Private dialing plans support both abbreviated dialing and carrier select-free public dialing through private networks.*

1.4.14.6.3 Access Gateway [C.2.7.8.1.4 (4)(a)]

The following Voice over Internet Protocol Transport Service capabilities are mandatory:
4. The contractor shall provide transparent access to Agency's Local Wide Area Network (WAN) connections. The contractor's Access Gateway shall provide an Ethernet UNI port to connect with Agency equipment.

For VoIP access, AT&T will provide Agencies with [redacted] routing equipment, depending on interface and bandwidth needs. These routers are configured with the necessary VoIP processing hardware, such as memory and the software required to act as a VoIP gateway. The actual model will be selected to satisfy specific engineering, management, and security requirements. The VoIP access router includes an Ethernet port for connection.

1.4.14.6.4 Reserved

1.4.14.6.5 Reserved

1.4.14.6.6 Agency Firewalls [C.2.7.8.1.4 (6)]

The following Voice over Internet Protocol Transport Service capabilities are mandatory:
6. The contractor shall verify with the Agency that the Agency firewall is compatible with this service.

In processing packets for security, VoIP packets pose a particular problem in that the processing delay of the firewall can acutely affect the quality of the calls passing through it. Due to the industry move toward services over IP, many newer firewall devices are being produced with VoIP capabilities. Given the differences in existing and new firewalls, AT&T has tested the following brands and types of firewalls, listed in **Table 1.4.14.6-3**, and approved usable firewall configurations for VoIP applications.

FIREWALL APPROVAL		
Manufacturer	***Model/Platform***	***Capability***
[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]
[redacted]		

Table 1.4.14.6-3: Approved Firewalls Work with VoIP. *AT&T has tested several firewalls for use with the VoIP suite of products. Approved, managed firewalls allow VoIP packets to operate in the network without excessive delay or other impairment.*

AT&T will verify with the Agency that the premises-based firewall is capable of supporting our VoIP service. This will be accomplished at the requirements determination phase of the project. In addition to verifying and certifying the capability of the router, we will provide expert advice regarding firewall policies and configuration to ensure that the firewall supports the VoIP functionality before implementation.

1.4.14.6.7 PBX Requirements [C.2.7.8.1.4 (8)]

The following Voice over Internet Protocol Transport Service capabilities are mandatory:
8. The contractor shall state the minimum and optimal requirements for Agency owned voice equipment (such as PBX's or other voice systems) to be compat ble and interoperate with the contractor's VoIPTS.

Table 1.4.14.6-4 lists the PBXs that have been laboratory- or customer-tested for VoIP [redacted] The notation of the interface type is important in determining the interoperability of each particular PBX manufacturer. Combinations of supported interfaces are indicated as they apply. All digital PBX systems with standard T-1 and T-1 primary rate interfaces (PRIs) are supported by VoIPTS.

MANUFACTURER	MODEL	INTERFACE	BOARD MODEL/NUMBER
[redacted]	[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]	
[redacted]	[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]	[redacted]
[redacted]			

Table 1.4.14.6-4: PBX Types. *Several types of existing PBX platforms are supported by the VoIP Network.*

Table 1.4.14.6-5 lists the IP PBX systems that have been certified to work with the AT&T Voice over IP Any Distance product as of June 2005. This list will continue to expand as VoIP becomes more widely deployed.

MANUFACTURER	MODEL	INTERFACE	BOARD MODEL/NUMBER
[redacted]	[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]	[redacted]
[redacted]			[redacted]
[redacted]			

Table 1.4.14.6-5: IP PBX Types. *Several IP PBX systems interoperate with VoIPTS.*

1.4.14.6.8 VoIP Security [C.2.7.8.1.4 (11)]

The following Voice over Internet Protocol Transport Service capabilities are mandatory:
11. The contractor shall ensure security practices and safeguards are provided to minimize susceptibility to security issues and prevent unauthorized access.

AT&T provides Agencies with the best possible IP network security using the seven elements listed in **Table 1.4.14.6-6**.

ELEMENT	FUNCTION
Separation	Customer traffic is separated using MPLS VPNs.
Automation	Automated perimeter security tools protect MPLS core.
Monitoring	IP traffic monitoring provides early warning of Internet viruses and worms.
Control	Strict operational security controls are enforced in MPLS core and on-service application platforms.
Testing	Testing, auditing, and reviewing increase security compliance.
Response	Proactive response teams trained in details of networks and security are deployed to places of potential attack or risk.
Innovation	AT&T funds extensive security research.

Table 1.4.14.6-6: Seven Security Elements. *AT&T uses seven elements to provide security across all products.*

Using these elements, along with monitoring and threat elimination tools, AT&T has been able to detect and stop attacks for its customers. (Refer to Section 1.3.1 for a complete description of these elements and the overall network security practices.) In addition to the overall network security practices outlined above, AT&T secures IP and traditional PBX equipment with the SIP VoIP network security model, as shown in **Figure 1.4.14.6-3**.

[redacted]

Figure 1.4.14.6-3: Network Core [redacted]

Using the SIP security model, the AT&T BE provides proxy authentication for the Agency PBX. This allows Agencies to use the service securely, even if the equipment that the Agency possesses is not an IP-ready PBX. Since the rest of the VoIP network is a closed MPLS environment, Agencies do not need to monitor or manage the security aspects of the remainder of the network. Finally, AT&T monitors all its IP networks for activity anomalies. This monitoring helps detect attacks and the setup of attacks before an actual attack occurs. This allows AT&T response teams to thwart most attacks often before they affect the service. Refer to Section 1.3.1 for more information on attack mitigation through monitoring.

1.4.14.6.9 Security Updates [C.2.7.8.1.4 (11)]

The following Voice over Internet Protocol Transport Service capabilities are mandatory:
11. The contractor shall ensure security practices and policies are updated and audited regularly.

[redacted] **Table 1.4.14.6-7**.

STANDARDS WORK	Following standards, tracking threats, and making industry-wide recommendations
DESIGN	Designing security into every component of every product
MANAGEMENT	Managing security best practices in service deployment
TESTING	Laboratory testing systems for vulnerability
MONITORING	Monitoring system against threats or changes in operation that could indicate a new threat
ACTION	Managing threats and mitigating attacks through response team action
ASSESSMENTS	Assessing new security threats and protection mechanisms for emerging products, such as VoIP

Table 1.4.14.6-7: Security Practices. *AT&T security personnel perform tasks that keep the network security up-to-date.*

Refer to Section 1.3.1 for an overall understanding of the AT&T security practices and how they are updated.

1.4.14.6.10 Denial of Service [C.2.7.8.1.4 (11)(a)]

The following Voice over Internet Protocol Transport Service capabilities are mandatory:
a. Denial of service – The contractor shall provide safeguards to prevent hackers, worms, or viruses from denying legitimate VoIPTS users and subscribers from accessing VoIPTS.

The primary defense against denial of service (DoS) attack, worms, viruses, and hackers is the separation of networks (outlined in Section 1.3.1). Using this separation scheme, the VoIP network, although interconnected to other networks on secure BEs, routes as an independent MPLS network. The core network is invisible to the Internet. In addition to being hidden from the Internet, customer networks cannot freely route to other customer networks [redacted]

Figure 1.4.14.6-4: DoS Attacks are Thwarted by MPLS Separation of Networks. [redacted]

Using AT&T VoIP services, Agencies are also protected from internal threats by the VoIP specific security architecture. In addition to the six points of security outlined in **Figure 1.4.14.6-3**, AT&T also follows a strict practice of securing every system in the network through a process of system hardening, as described in Section 1.3.1.

1.4.14.6.11 Intrusion [C.2.7.8.1.4 (11)(b)]

The following Voice over Internet Protocol Transport Service capabilities are mandatory:
b. Intrusion – The contractor shall provide safeguards to mitigate attempts to illegitimately use VoIPTS service.

The security that is built into the AT&T network automatically protects against intrusion threats. **Table 1.4.14.6-8** lists the separated major elements of intrusion control within the AT&T VoIP network. These security mechanisms work in conjunction with the security mechanisms in the VPN service that provides access to the VoIP network.

MECHANISM	EFFECT ON INTRUSION
Separation of Networks	Separation of networks makes it imposs ble for someone on public network, such as Internet, to access VoIP network or any of its customers. Customer networks are shielded from each other as well.
Secure Border Edge	Secure border edge device creates VoIP signaling control point that requires authentication to pass signaling data. Since unauthorized signaling is not passed to call control units in interior of VoIP network, coupled with network separation, signaling access is only granted to authorized VoIP users. Interior systems will only accept signaling from authorized interior systems. The border edge is authorized gateway to interior.
Systems Hardening	Every system in VoIP network is hardened, removing all unused IP ports and unnecessary operating system (OS) and program functions. This eliminates use of functional block as intrusion stepping stone.
Systems Security Standards	Every system in network is operated under strict set of AT&T security standards. (Refer to Section 1.3.1 for complete description of AT&T system security standards.)
Strong Authentication	All legitimate systems access is authenticated using strong authentication and [redacted] This control mechanism prevents outside access and logs all access as well.
Monitoring	All VoIP systems are continuously monitored for any access violation. Access violations create alarms and energize action from incident response team.

Table 1.4.14.6-8: Security Mechanisms. *Several elements and procedures secure the VoIP systems from intrusion.*

For further details on the trusted security domains and protections against intrusion, refer to Section 1.3.1.

1.4.14.6.12 Invasion of Privacy [C.2.7.8.1.4 (11)(c)]

The following Voice over Internet Protocol Transport Service capabilities are mandatory:
c. Invasion of Privacy – The contractor shall ensure VoIPTS is private and that unauthorized third parties cannot eavesdrop or intercept VoIPTS communications.

Network protection against eavesdropping is supplied by the standard VoIP security mechanisms outlined in **Table 1.4.14.6-9**. These security mechanisms are linked in their effectiveness to the Agency LAN eavesdropping protections.

AREA OF NETWORK	SECURITY MECHANISM
Access	Secure managed VPN service eliminates access from outside sources. Transporting IP in MPLS makes address spoofing virtually impossible
Signaling	Signaling is secured in the border edge and will only forward packets to authenticated partner system. All call handling platforms are located in VoIP core and are secured using AT&T's systems hardening and strong authentication methodology.
VoIP Core	VoIP core is 100% MPLS routed. No IP access exists in core. Access to core routers and elements are strictly controlled. Physical security is strong and follows telephone systems standards.
Agency LAN	Best security practices in Agency LAN will provide like-for-like eavesdropping security, as compared to traditional telephone service systems. Strong internal physical security is also required.

Table 1.4.14.6-9: Area of Network. *Eavesdropping protection in the network is provided by a solid VoIP security model (**Figure 1.4.14.6-3**) and practices.*

The industry standard deployment strategy for privacy is to deploy VoIP in a way that achieves invasion of privacy security that is like-for-like with traditional telephone service. VoIP telephones are no more easily tapped than a traditional telephone. If an Agency uses a switched Ethernet environment, and adequate physical and systems management controls are placed on the LAN switch equipment, then the VoIP service is as secure against eavesdropping as a traditional telephone service (**Figure 1.4.14.6-5**).

[redacted]

Figure 1.4.14.6-5: VoIP Offers Like-for-Like Wiretap Resistance. [redacted]

1.4.14.6.13 Voice Over Internet Protocol Transport Service Interfaces [C.2.7.8.3.2 (1)]

UNI Type 1
Interface Type: Ethernet port: RJ-45
(Std: IEEE 802.3)
Payload Data Rate or Bandwidth: Up to 100 Mbps
Signaling Type: SIP, H.323, MGCP or
SCCP [Optional]

The Agency UNI Type 1, RJ-45 Ethernet 10/100, is provided as a standard interface on the service delivery point (SDP) options. The SDP is signaling independent and provides access to the first point of CoS routing and VPN termination. Router options are dependent on access methodologies, such as MPLS, ATM, and frame relay, but all configurations will be able to supply the 10/100 Mbps Ethernet interface in a signaling protocol agnostic manner. The two examples listed in **Tables 1.4.14.6-10** and **1.4.14.6-11** depict the availability of Ethernet interfaces.

CISCO ROUTER CISCO2811-V/IK9 – 2811 VOICE BUNDLE, PVDM2-16, SP SERV, 64F/256D CONFIGURATION	
Part Number	***Description***
[redacted]	[redacted]
[redacted]	[redacted]
[redacted]	[redacted]
[redacted]	[redacted]
[redacted]	[redacted]
[redacted]	[redacted]
[redacted]	[redacted]
[redacted]	[redacted]

Table 1.4.14.6-10: SED for T1. *Agencies with a need for T-1 access and a 100 Mbps Ethernet port are provided equipment such as the Cisco 2811 router.*

CISCO ROUTER CISCO7204VXR – CISCO 7204VXR, 4-SLOT CHASSIS, 1 AC SUPPLY CONFIGURATION	
Part Number	***Description***
[redacted]	[redacted]
[redacted]	[redacted]
[redacted]	[redacted]
[redacted]	[redacted]
[redacted]	[redacted]
[redacted]	[redacted]
[redacted]	[redacted]
[redacted]	[redacted]

Table 1.4.14.6-11: SED for T-3. *Agencies with a need for up to T-3 access and100 Mbps Ethernet port are provided equipment such as the Cisco 7204 router.*

1.4.14.7 Stipulated Deviations

AT&T takes neither deviation nor exception to the stipulated requirements.

1.4.14.8 Local SIP Trunk Service

Agencies using the Local SIP Trunk Service are provided Session Initiation Protocol (SIP) based IP voice trunking that includes access to both local and long distance calling via a single IP access circuit. This SIP-based telephone trunking service is carried to Agency locations over an AT&T IP transport product, such as NBIP-VPNS, which allows an Agency to consolidate voice and data on a single bandwidth access and port. Wide support for Agency call switching systems is provided by the Local SIP Trunk Service as it supports both VoIP call control systems such as the Cisco Call Manager and other IP ready Private Branch Exchange (PBX) systems. The service also can support traditional digital PBX systems [redacted]

The Local SIP Trunk Service is fully integrated with the Public Switched Telephone Network (PSTN) and is provided by systems that are fully integrated into the AT&T IP core network. The service is offered through an AT&T Competitive Local Exchange Carrier (CLEC).

Because the Local SIP Trunk Service is a built- in component of the greater AT&T IP network, and because it is offered through [redacted] Agencies are provided with key voice systems functions set forth below when the service is coupled with CIPS or NBIP-VPNS and connected to an Agency PBX or IP call manager.

- Local Telephone Numbers
- Local inbound and outbound calling
- Access to low cost IP based Long Distance
- On-net to on-net calling to any AT&T VoIP location [redacted]
- Interfaces to either SIP based or traditional PBX systems
- Combined voice and data on a single access arrangement
- 60%-70% lower voice bandwidth usage
- E911 call routing [redacted]

Using the Local SIP Trunk Service will enable Agencies to access the features of new IP based telephone systems, including the consolidation of call control and PSTN access over a [redacted]

1.4.14.8.1 On-Net and Off-Net Calling

This SIP-based trunk service provides calling to both on-net and off-net locations. On-net calls are calls that route within the AT&T VoIP network or customer virtual private IP network. This on-net network could be the Agency's own IP-VPN or the greater AT&T VoIP core network, including other AT&T VoIP customers. Off-net calls are calls that are routed through the AT&T VoIP core network and into the PSTN. For this SIP-based trunk service, all on-net and local calling is [redacted]. **Figure 1.4.14.8.1-1**, below, illustrates how callers can make calls and have those calls remain on-net, by staying either on the Agency's IP-VPN, or within the AT&T VoIP core network. On-net calls include calls to any telephone that is served by any one of the [redacted]

[redacted]

Figure 1.4.14.8-1: [redacted] includes all calls [redacted]

The SIP-based trunk service provides Agencies with combined local and long distance access over a single IP VPN. The ability to have many calls stay on-net allows Agencies to take advantage of low cost IP bandwidth.

1.4.14.8.2 Service Features

AT&T's Local SIP Trunk Service is a fully featured local telephone service offered utilizing [redacted]. This allows AT&T to offer [redacted]

[redacted]

- Direct inbound calling from the PSTN to the [redacted]
- [redacted]
- Local telephone numbers (new and ported) and PSTN local telephone number registration,
- [redacted]
- E911 calls routed to the correct PSAP with location data
- [redacted]
- N11 (211, 311, 511, 711) calling
- Calling Name (CNAM) and Caller ID (CID)
- Directory Listing
- Directory Assistance (411)
- Originating Toll Free calling
- Standard NANP 7 and 10 Digit Dialing

Along with the basic set of local telephone service features listed above, the Local SIP Trunk Service also provides Agency selectable site specific outbound call blocking options. These features allow Agencies to selectively block telephone services that are typically [redacted] Optional blocking for both outbound and inbound calls is shown in **Table 1.4.14.8.2-1**.

[redacted]	[redacted]
[redacted]	[redacted]
[redacted]	[redacted]

Table 1.4.14.8-1: [redacted]

The Local SIP Trunk Service directly supports VoIP based PBX and call control systems, thus providing Agencies with an alternative to traditional telephone trunking services. The service seamlessly maps its SIP based features to complete PSTN interconnectivity to provide the government with a single source for voice systems interconnect.

1.4.14.8.3 Local Number Assignments

[redacted]. Adherence to local number portability regulations allows AT&T to migrate existing DID number ranges to the AT&T Local SIP Trunk Service. The SIP trunk service can also be used to assign additional or replacement numbers as required by the Agency. In addition AT&T's ability to [redacted] Number assignment scenarios are shown in **Table 1.4.14.8-2**.

CAPABILITY	ACTION
Number Allocation	AT&T provides telephone numbers through LNP or through new number assignment. This allows a customer to easily move existing users to AT&T VoIP as well as expanding existing facilities and locations or starting up new facilities in new locations.
[redacted]	[redacted]
[redacted]	[redacted]

Table 1.4.14.8-2: North American Numbering Plan (NANP) number assignments. *North American Numbering Plan (NANP) number assignments typically use LNP, but can be assigned as new numbers as well.* [redacted]

1.4.14.8.4 IP, PSTN and Long Distance Capability

The Local SIP Trunk Service provides local calling, IP to IP calling and connectivity to PSTN based long distance calling. Local calls, those within the same Local Calling Area (LCA), and all IP calls that do not leave the AT&T [redacted]. The calling plan for the service is shown in **Table 1.4.14.8-3.**

[redacted]	[redacted]
[redacted]	[redacted]
[redacted]	

Table 1.4.14.8-3: Local SIP Trunking. [redacted]

1.4.14.8.5 Trunk and Transport Capabilities

To better serve its customers AT&T has built VoIP functionality into the AT&T core IP infrastructure. As part of the core IP infrastrure, functional equipment, such as the AT&T [redacted], routes E.164 calls within the VoIP network in addition to providing connectivity to the many PSTN gateway units that deliver two-way access to the world's telephone systems. [redacted] provide security and keep private networks private.

Using the integrated network approach to offering VoIP services provides several key functional advantages over independent component built voice systems such as number mobility, call control consolidation, and combined voice and data transport. **Table 1.4.14.8-4**, outlines the major technical advantages to providing VoIP from the AT&T integrated VoIP network infrastructure.

AT&T BUILT-IN VOIP NETWORK FUNCTIONALITY	
Built-In Feature	***Built-In Benefit***
Call Quality	Overall call quality is maintained throughout the VoIP network using DiffServ based Quality of Service (QoS) and MPLS packet forwarding. These mechanisms confirm that VoIP calls are made at toll quality to other VoIP locations and to the PSTN. QoS also provides highly reliable delivery for business class data. The QoS metrics are customer selectable in the Networx IP-VPN services (NBIP-VPNS and CIPS).
Voice Data Integration	Voice and data integration provides customers with a lower cost option for overall operation. Only a single access mechanism is needed for both voice and data. Voice bandwidth requirements are greatly reduced using IP codecs. The access and transport can be configured to support video conferencing on the same IP access as well. This allows for newer IP based communications systems to operate from a single access infrastructure further reducing cost.
Capacity Expansion	The call capacity on the SIP trunks is easily expanded. Generally no new access needs to be ordered when additional VoIP call capacity is required. Typically the Voice/Data QoS ratios are the only changes required when a site needs additional capacity.
Bandwidth Usage	Bandwidth savings are typically 60%-70% over ISDN PRI Voice. The savings varies depending on how many G.711 (high VoIP bandwidth) applications are required, such as analog full duplex speaker phones.

AT&T BUILT-IN VOIP NETWORK FUNCTIONALITY	
Codec Support	The AT&T SIP Trunk Service supports multiple codecs in order to provide the und users with a telephone service that is both useful and cost effective. Low bandwidth codecs such as [redacted] are supported for basic calling and can consume as little as [redacted] of bandwidth. For applications where full bandwidth is required, such as the use of an analog Polycom speakerphone, the G.711 codec is supported. T.38 for facsimile systems is also supported in the service. This support for multiple codecs is integrated throughout the entire AT&T VoIP network infrastructure allowing users to take advantage of VoIP on-net to on-net, or to and from the PSTN.
Number Translation	Most private dialing plans can be supported by the AT&T CCE units. Up to 35 digits per dial string are supported. This allows for maximum flexibility when PBX systems are used to provide digit fill for private dialing plans between locations. Since the CCE's are built into the entire network VoIP network, private dialing can be supported to any of the customer's sites that are on the VoIP network.
[redacted]	AT&T provides the SIP trunking service i[redacted] depending on telephone configuration, when a low bandwidth codec is used. All bandwidth on the IP transport for VoIP is allocated on an as needed basis only. Other traffic, such as business data, can utilize any of the bandwidth space when VoIP calling is not at full capacity. Since the [redacted], is calculated for high-day-peak-usage, most of the time there is additional bandwidth left for business data.
Interface Choices	The AT&T SIP trunking service is built to support various types of customer telephone equipment. Customers no longer have to upgrade [redacted] of VoIP. The Local SIP trunking service supports direct SIP trunking to IP enabled systems and media [redacted]
VoIP Service Interconnectivity	The AT&T VoIP network has been built so that there is interconnectivity to existing IP and non-IP services such as conference calling and video conferencing. This allows customers to use these services from their WAN/VPN access without building yet another network.

Table 1.4.14.8-4: AT&T Built-in VoIP Network Functionality. *A high level overview of several key functional elements that are provided by the AT&T integrated VoIP infrastructure. The functionality gained through built-in features provides customers with a high quality, easy to use VoIP service.*

In order to support varying sets of needs for our customers, AT&T has built a VoIP network infrastructure with technology is based on years VoIP and telephone experience. Using the AT&T Local SIP Trunking Service, Agencies receive a local telephone service that operates in a similar fashion as PSTN based services, supports new VoIP call control platforms, allows for a consolidation of voice and data, and provides a mechanism to consolidate and centralize communications systems.

1.4.14.9 Custom Local SIP Trunking

The custom SIP trunk solution provides various trunking solutions into many Agency call processing configurations. The custom solution allows for multiple types of failover configurations, mixed trunk type environments, and remote hosted call control support. The custom SIP trunks can also be configured with fixed trunk to station ratios that allow more economical resource sharing across an enterprise, and custom calling [redacted]

Figure 1.4.14.9-1 shows one possible custom solution that employs a [redacted] circuits operating under a [redacted] configuration.

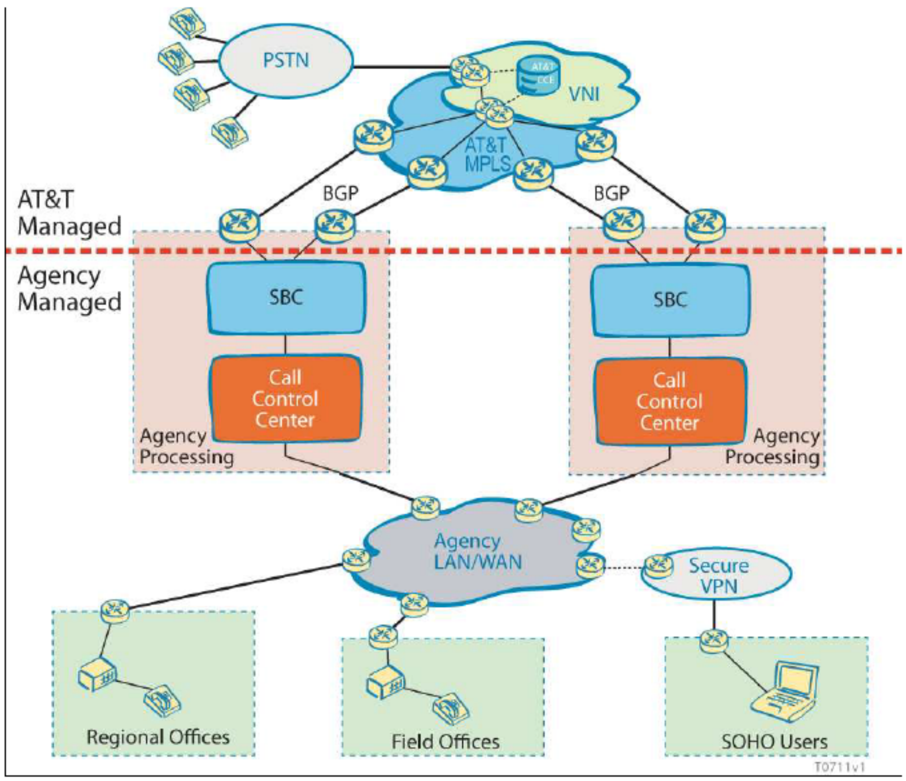

Figure 1.4.14.9-1: SIP Trunk Connectivity Support. *Using custom solution elements this Agency is able to support regional offices, field offices and SOHO users from call processing centers that are* [redacted]. *Using* [redacted] *each call processing center is provided better SIP trunk availability. In this configuration* [redacted] *processing centers can also be implemented.*

In Table 1.4.14.9-1, the key configuration parameters that are used to build custom solutions are described. These basic elements are combined with other SIP capabilities to provide trunking solution that will best support the call control systems configuration that the Agency has selected.

CAPABILITY	DESCRIPTION	USE
Multiple Connection points per trunk group	[redacted]	[redacted]
Redundancy using BGP	[redacted]	[redacted]
Failover SIP Routing	[redacted]	[redacted]
Hosted Call Control Support	[redacted]	[redacted]

CAPABILITY	DESCRIPTION	USE
Fixed Trunk to station ratio	[redacted]	[redacted]
Off-net Calling package	[redacted]	[redacted]

Table 1.4.14.9-1: Basic SIP Trunk Options and Components. *Using variable features and functions within the AT&T VoIP infrastructure, Agencies are able to deploy call management systems in configurations that better suit their network infrastructure and the needs of their users. Additional features such as* [redacted] *features that can be combined in an overall solution as they become available.*

Custom SIP Trunking has the ability to support many different call control configurations. This is done by combining different features of the SIP Trunking service into the solution and varying the number of concurrent calls supported in each link of a solution. For example, in the [redacted] solution shown in **Figure 1.4.14.9-2**, there could be a peak concurrent call load of 300 required [redacted]. In order to support the full concurrent call load [redacted], an additional [redacted] site. This allows the peak number of concurrent calls [redacted] traffic .

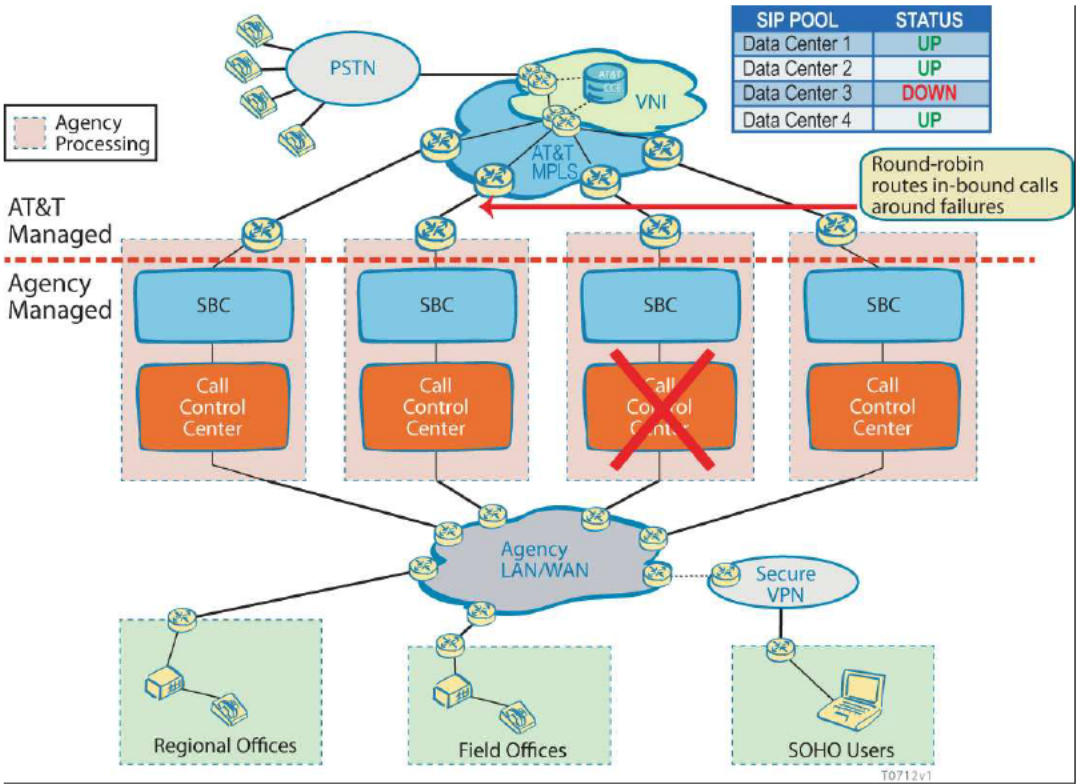

Figure 1.4.14.9-2: [redacted] *Using a configuration that includes multiple sites in a solution, and* [redacted]*, this Agency is able to sustain a failure that* [redacted]*. This configuration supports all four regional offices, the* [redacted] *offices and the* [redacted] *fully from all four processing locations. While the installation and maintenance of the* [redacted] *being fully capable of handling all* [redacted] *can be more costly, the* [redacted] *reduces the number of* [redacted] *that must be configured into each data center.*

In the example shown in **Figure 1.4.14.9-2**, the addition [redacted] at each site that is intended to handle failover increases the bandwidth requirements at each site which [redacted]. Therefore, it is not always advantageous to deploy a [redacted] into every solution as this amount of traffic handling may rarely be needed. Solutions are [redacted] to support the needs of each individual Agency to help provide a robust service while reducing costs, as opposed to employing a [redacted] that may not match the customer's configuration needs.

The overall cost of deploying any custom solution will vary depending on how the solution is configured, and how many trunks are available overall. On a case by case basis, AT&T will help the Agencies find the best solution to fit their VoIP connectivity needs, and allow Agencies to control costs over the lifecycle of their overall VoIP solution . Each solution will be designed with the [redacted], allowing an Agency to grow or shrink the level of service required to meet their needs, without significantly affecting their overall solution. For example, the solution shown above in **Figure 1.4.14.9-2**, is configured as an overall solution that includes [redacted]. For each group of active trunks installed, [redacted]. In this scenario, the [redacted] service would be active and standby trunks in fixed ratio pairs.

Examples of Items that will affect the individual Agency's unique configuration of a Custom SIP Trunk solution are:

- Total capacity versus [redacted], when [redacted]
- Ratio of trunks to users, [redacted]
- Total capacity versus [redacted], [redacted]
- Total number of [redacted] supported
- Number of trunks included in a [redacted] configuration
- Inclusion of [redacted]
- Termination of SIP services on a [redacted] such as a SIP Session Border Controller

In this offer, custom SIP trunking solutions are provided in one of four formats. Within these four formats, a trunking solution can be fashioned to support nearly any configuration or trunking need an Agency might have.

Table 1.4.14.9-2 outlines the four trunk types and how the can be used to support different situations.

TRUNK TYPE	FEATURES AND FUNCTIONS	USE
[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]
[redacted]	[redacted]	[redacted]

Table 1.4.14.9-2: SIP Service Trunk Types. *Using one of the four custom trunk types Agencies can build a service that matches the complexity of their VoIP call control services or systems.* [redacted] *types are used when the Agency is* [redacted] *as a fully managed solution. The two custom resilient solutions can be tailored to most Agency owned call* [redacted] *configurations.*

When Agencies are deploying VoIP within their organizations or outsourcing their call control, the custom local SIP trunk service allows for cost effective configurations that vary as much as the many VoIP solutions that are available in the industry today. Agencies will also benefit from the cost alignment that can be provided [redacted].