whitepaper

Self-Paced Migration to a VoIP Enterprise

How Enterprise-Managed SIP Trunking Saves Time and Money When Moving to VoIP

For some enterprises, the migration to VoIP can be painful in terms of costs, technology, support resources and time. SIP trunking from service providers is available to ease that pain, giving legacy PBXs connectivity to a global VoIP network. Learn how easy and cost-effective it is to deploy and manage your own SIP trunks, putting your enterprise back in control of its migration to VoIP.

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Introduction

As enterprises continue to migrate to Voice over Internet Protocol (VoIP), service providers are offering complementary voice services that will transform the way we use the broadband network. One such service is Session Initiation Protocol (SIP) trunking. SIP trunking is expected to grow at a rate of 42.8% over the next six years (Frost & Sullivan, 2010). The service will enable an enterprise to use the Wide Area Network (WAN) instead of the Public Switched Telephone Network (PSTN). Using SIP and IP transport, the WAN now supports converged communications including voice, data and video.

SIP trunking from a service provider delivers some important benefits to enterprises. First, SIP trunking enables enterprises to combine data and voice on the same WAN, lowering costs associated with the management of two separate networks. Because multiple services share the same pipe, WAN utilization is improved. Finally, this converged network may now support applications beyond simple voice and data. Applications like video, mobility, conferencing, instant messaging and user presence may all share the same WAN connection.

This paper will expand on the concept of SIP trunking to show how enterprises may deploy their own trunking services to save money, enable new features and provide a "self-paced migration" path from their existing PBXs onto a corporate-wide IP-PBX.

Problem Statement

While SIP trunking is clearly the future for enterprise communications, today's carrier-based service is limited to key metropolitan areas. In addition, the PBX must be certified as being interoperable with a service provider's SIP trunking service. Although cheaper than traditional PSTN communications, there are still costs associated with monthly and per-minute usage fees. The deficiencies with today's SIP trunking services may be overcome when enterprises utilize the power of IP communications platforms and manage their own SIP trunks between locations.

SIP Trunking is expected to grow at a rate of 42.8% over the next six years – Frost & Sullivan, 2010.

Today's Options

Most enterprises currently use the two network models shown in Figure 1. Their PBX is connected to the PSTN and their data network (or LAN) is connected to the WAN. In some cases, the WAN connectivity is simply an Internet connection from an Internet service provider. In other cases, the WAN connection is a high-speed network with the ability to provide Quality of Service (QoS) for various types of traffic. This two network model is inefficient, expensive, overly-constraining, and no longer necessary for today's progressive enterprise.

When more than one location is considered, the two network model is expanded to the example shown in Figure 2. Location 1 and Location 2 independently connect their voice network to the PSTN and their data network to the WAN. If these two locations are of substantial size, the enterprise will pay for more capacity than will actually be required. In addition, this type of enterprise will likely have data connectivity between the two sites, over the WAN, to share a corporate intranet or other business applications. To accomplish this WAN connectivity, a managed data service is often purchased, giving the enterprise "private" data connectivity between the two locations. The enterprise continues to manage two disparate networks.

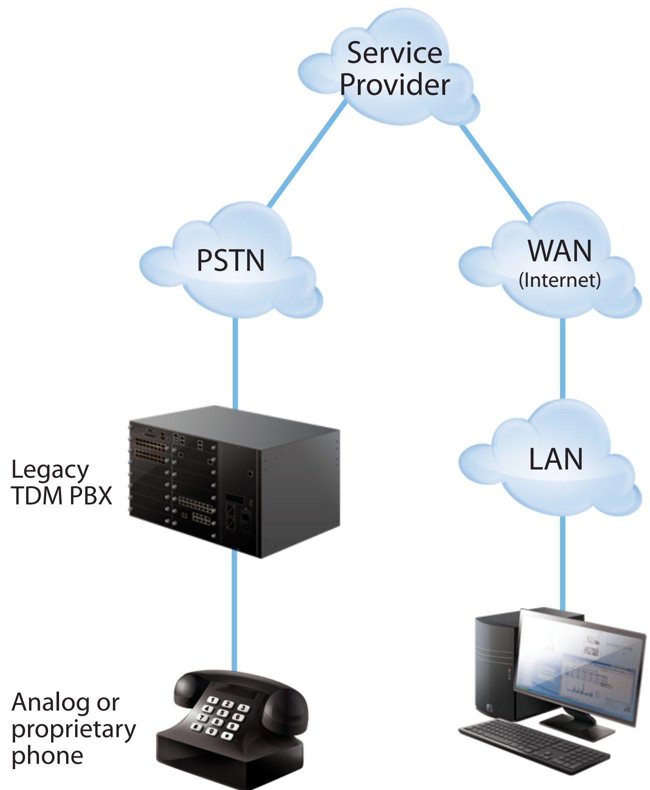

Figure 1. Dual-Network Enterprise.

Figure 2. Multi-site Enterprise.

Figure 3 represents an option for enterprise voice communications. The PSTN is replaced by SIP trunks provided over, or in conjunction with, the WAN. SIP trunks can be viewed as data or IP services that are used to carry voice traffic. In order for enterprises with existing TDM PBXs to take advantage of SIP trunking services, a voice gateway must be installed. This voice gateway interworks voice and signaling between TDM and VoIP networks. Now a single, converged network supports both data and voice, lowering the cost of managing two networks. The enterprise also achieves better utilization of WAN infrastructure by combining multiple services on the same pipe. Finally, new IP-based communication services can be more easily integrated with voice services and include video, instant messaging and presence.

While compelling, these benefits alone may not justify the deployment of SIP trunks. Ideally, an enterprise would use a solution that provides the benefits of SIP trunking but also overlays new unified communications (UC) features on the existing infrastructure, further reducing costs and allowing a controlled, self-paced migration to VoIP.

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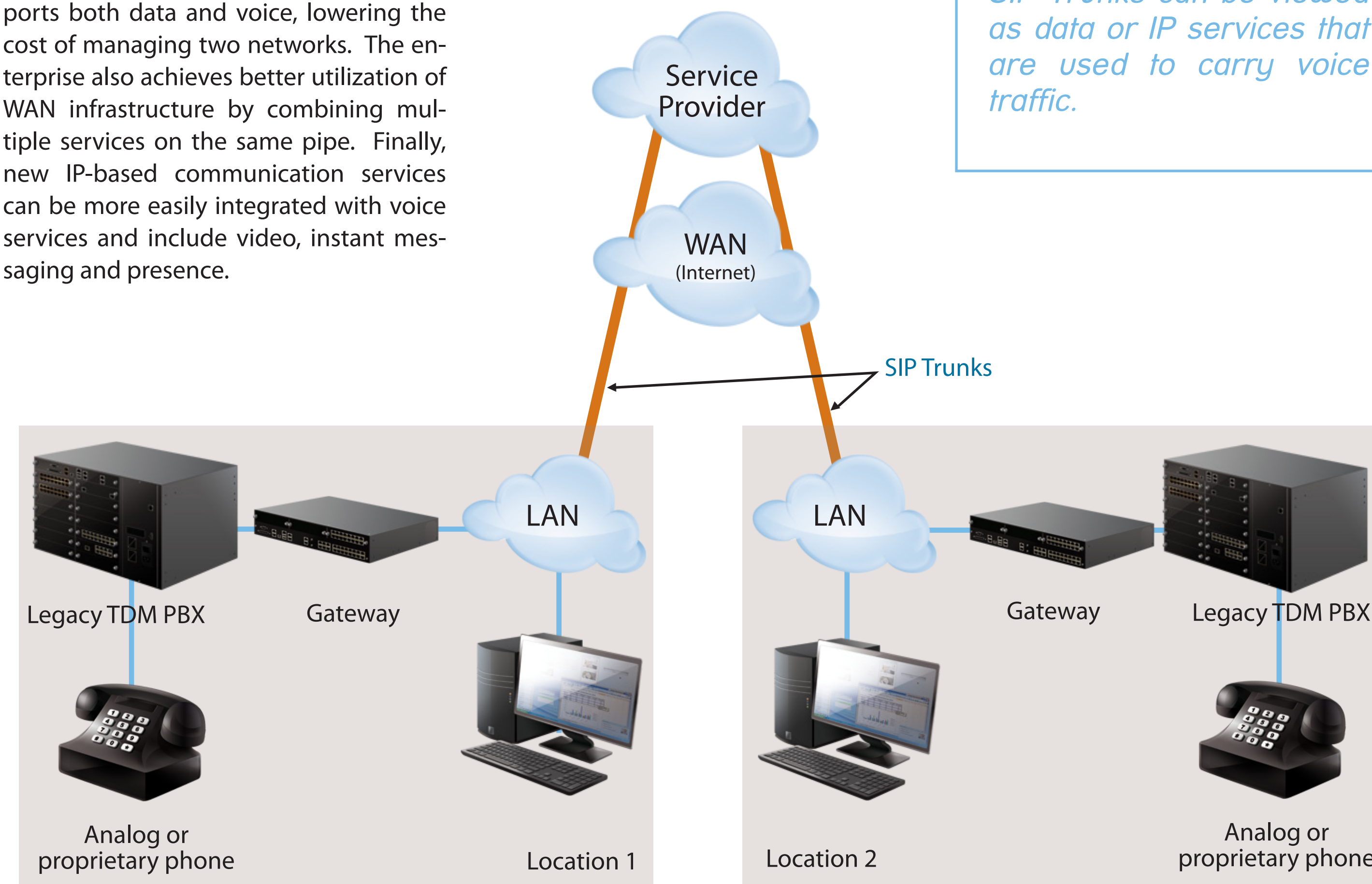

Figure 3. SIP Trunking from a Service Provider.

Aastra's Solution

Aastra's Clearspan® is an all-SIP, all IP-PBX built on the same technology deployed by the world's leading service providers to support their SIP trunking offerings. Aastra's solution begins by simply adding the Clearspan core within an enterprise data center as shown in Figure 4. Additionally, Clearspan supports geographic redundancy, so for maximum fault tolerance, a second data center might harbor the redundant system components.

An enterprise would then install a single set of SIP trunks from the service provider. These trunks would support all users that need to place or receive off-net calls. Off-net calls are those calls that are placed to or received from callers outside of the enterprise. From the data center, an enterprise would use existing WAN connections to provide trunking services to all enterprise locations. Using this configuration, an enterprise will recognize four additional benefits beyond service provider SIP trunking:

- Operational savings by self managing SIP trunks between enterprise locations.
- Deferred costs by purchasing and deploying a modular IP-PBX over time.
- Enhanced unified communications features, while still using the legacy PBX infrastructure.
- Managed costs by migrating to VoIP at their own pace.

These benefits, combined with the existing benefits of SIP trunking from the service provider, create a compelling argument for enterprises to begin the migration of voice traffic onto their managed WAN connections. Each additional benefit will be explored in more detail.

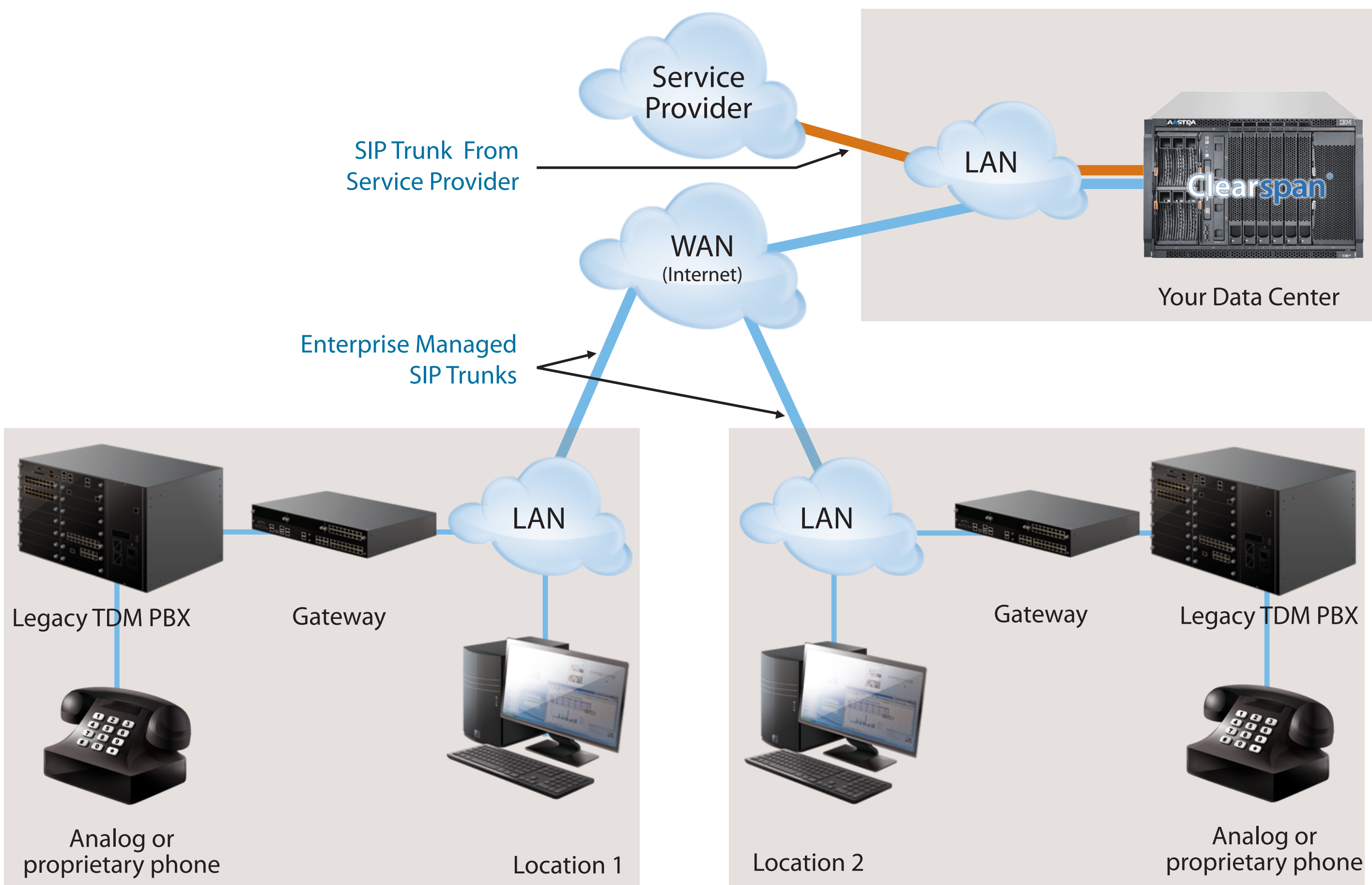

Figure 4. A Better Trunking Concept Using Clearspan.

Operational Savings

With this solution, SIP trunks from the service provider are still required, but because they are installed at the highest level in the enterprise, the trunking capacity is the aggregate of all locations. Therefore, link utilizations and costs are optimized. At the next level of the hierarchy, an enterprise will experience tremendous savings by managing their own SIP trunks across their existing WAN connections. Since the enterprise is self-managing these trunks, there are no usage fees charged by the service provider, assuming that sufficient managed bandwidth exist between locations. All on-net calls (between Location 1 and Location 2) are essentially free. To further maximize efficiency between locations, enterprises may choose to implement a higher compression algorithm, G.729, reducing the bandwidth required and maximizing link efficiency.

Deferred Costs

Clearspan's modular approach allows enterprises to save significant up-front capital by deferring costs until the full migration to UC. For example, purchasing SIP phones for every user typically ranges from one-third to one-half of the price of a new system. Because enterprises keep their existing phones, the purchase of SIP phones may be delayed until the rollout of VoIP-only services to a location. Another significant cost component for a VoIP system is the software license fee. With Clearspan, an enterprise with a single trunk license may support many users behind a legacy PBX. Therefore, up front per-user software license fees are not required.

Figure 5 shows an example of the cost per user for implementing a new IP-PBX with UC capabilities versus Figure 6 which shows the costs associated with adding a UC overlay to the existing infrastructure. As the figures show, up to two-thirds of the costs of implementing the new UC functionality may be deferred.

Clearspan's modular approach allows enterprises to save significant up-front capital, deferring nearly two-thirds of the cost of a new IP-PBX system

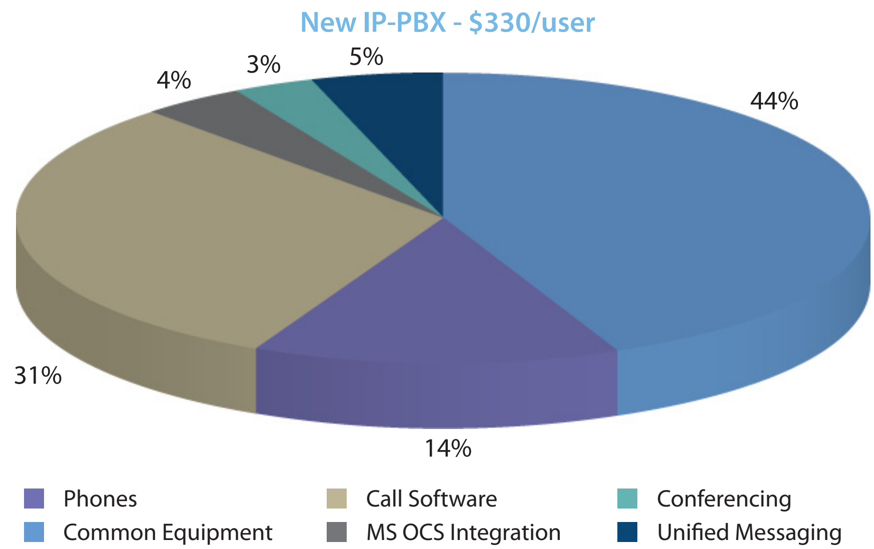

Figure 5. Cost Allocation of a new IP-PBX with UC Capabilities - $330/user

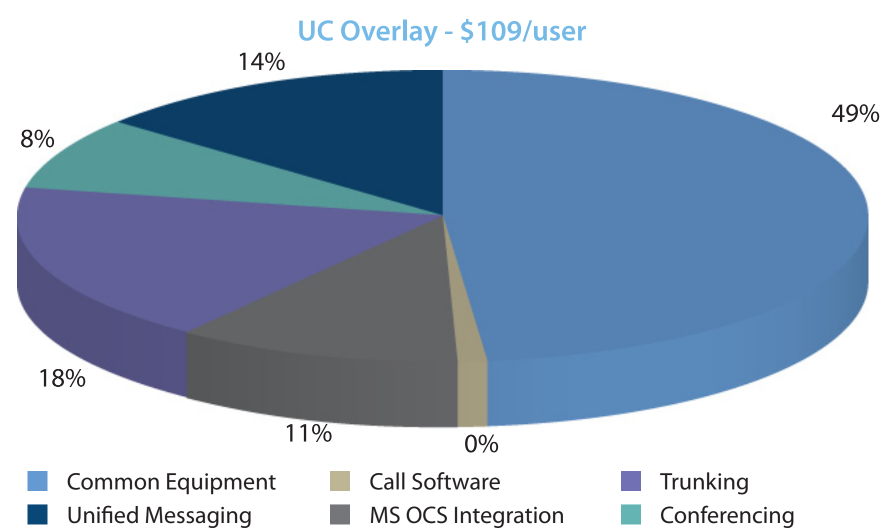

Figure 6. Cost Allocation of adding UC Overlay to existing infrastructure - $109/user

Advanced Features

In addition to operational savings and deferred capital costs, users will receive new, advanced features while still using their existing PBX equipment. These new features include:

✶ **Dial plan aggregation** – Allows enterprises to integrate their dial plans, establish least cost routing rules, and provide manual or automatic call rerouting when needed.

✶ **Conferencing** – Allows an enterprise to share a single conferencing and collaboration system.

✶ **Enterprise-wide unified messaging** - Allows the entire enterprise to share a common, centralized voice-mail system with modern notification and usability mechanisms.

✶ **Enterprise-wide contact centers** - Allows the enterprise to share a common, centralized contact center, partitioned along individual group needs.

✶ **Authorization / Account Codes** - Provides a secure method of placing long distance calls.

✶ **Find Me / Follow Me** – Allows users to have a single virtual number. When that number is dialed the system routes the call through a user-defined list of numbers that may be called simultaneously or sequentially.

✶ **Fixed Mobile Convergence** – Allows for seamless integration of mobile phones within the enterprise communication system.

✶ **Outlook/browser toolbars** – Gives users the flexibility to manage features and calls from their computers.

✶ **Home office / Teleworker** – Gives remote and teleworkers all of the features and functions as if they were in the office.

✶ **Support for Microsoft® OCS® and IBM® Sametime®** – Allows users to determine the "presence" of others in the organization and then click-to-dial available users.

✶ **Custom Ringback Tones** – Allows a custom audio file (.wav file) to replace the normal ringing tone.

Self-Paced Migration™

Perhaps the greatest benefit of Clearspan is that it gives enterprises the freedom to migrate to VoIP on their time table. As legacy PBXs are decommissioned at a specific location, enterprises simply deploy an edge appliance and SIP phones as shown in Figure 7.

It may be beneficial to maintain the gateway of a location that has migrated to VoIP for two reasons: The gateway provides a local hop-off point for the enterprise needing to make calls to that local area and it provides a failover mechanism for emergency calls if the WAN is down.

The edge appliance is an intelligent device that tracks VoIP voice quality and WAN connectivity. If a problem is encountered, the edge device will redirect calls to the local gateway. Many edge devices also provide a useful, yet limited, PBX feature set that is instantly invoked in case of WAN failure.

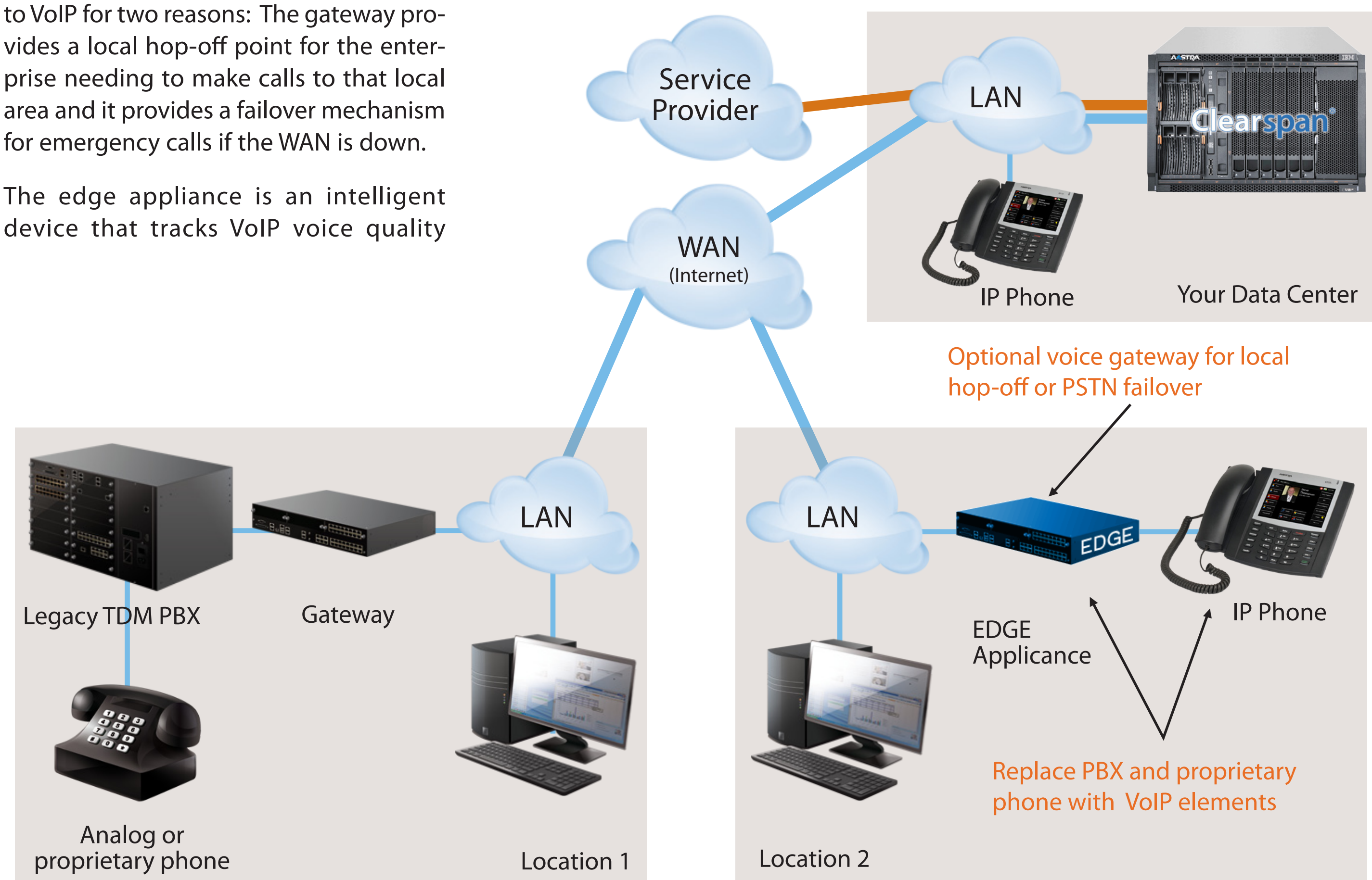

Figure 7. Self-paced Migration to VoIP using Clearspan.

Summary

SIP trunking is an important evolution for VoIP. The service allows enterprises to move to a converged network that supports data, voice and video, alleviating the need to manage two separate networks. Enterprises should look at the business case for moving their PSTN trunks to a SIP-based service. Included in that business case should be consideration of self-managed SIP trunks across the WAN and how much more value might be expected.

Clearspan takes SIP trunking to the next level, allowing enterprises to manage their own SIP trunks, reducing operating and capital expenditures, and enabling new features on their existing network.

Most importantly, Clearspan gives enterprises the platform to migrate to VoIP at their own pace.

For additional information, please visit www.aastraclearspan.com/trunk.

About Aastra USA

Aastra USA Inc. is the US business unit of Aastra Technologies Limited, a company at the forefront of the enterprise communication market. Headquartered in Concord, Ontario, Canada, Aastra develops and delivers innovative communications products and applications for businesses. Aastra's operations are truly global with more than 50 million installed lines around the world and a direct and indirect presence in more than 100 countries. Aastra is entirely dedicated to enterprise communications and offers IP telephony and Unified Communications solutions individually tailored to satisfy its customers' requirements. These range from feature-rich call managers for small and medium businesses and highly scalable ones for large enterprises, integrated mobility, call center solutions to a wide selection of terminals. With a strong focus on open standards, Aastra enables enterprises to communicate and collaborate more efficiently.

For additional information about Clearspan and Aastra, visit us online at www.aastraclearspan.com

Aastra USA
2811 Internet Blvd.
Frisco, Texas 75034
T 1 800 468 3266
F 1 469 365 3006
www.aastrausa.com

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