White Paper

Is SIP Trunking on Your Horizon?

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This white paper from Integrated Research discusses how understanding your call flows, PSTN trunk capacity and usage today will prepare you for a successful SIP trunking implementation in the future.

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White Paper on a Page™

Is SIP Trunking on Your Horizon?

Adopting a SIP trunking service is not a simple process. A profusion of choices and decisions awaits on pricing, service availability, capacity, service level agreements, and much more. That's in addition to selecting the equipment you'll need and sizing your SIP trunking implementation correctly.

This white paper discusses one of the critical considerations of migrating from TDM to SIP trunks – that of understanding current call activity and volumes across your existing VoIP environment. Before you adopt SIP trunking it's wise to analyze your existing VoIP and UC environment to ensure your implementation is as successful as possible.

Assess your VoIP environment today

At a minimum you should understand your existing call flows, trunk capacity, usage, and carrier costs. Reducing branch-level PSTN trunk capacity means you may need to revisit even the basic question of your network's capacity to support the extra traffic that will flow across the WAN to the SIP trunk or trunks.

Is your network ready?

Reducing the branch-level PSTN capacity means you're placing increased dependency on the WAN to carry real-time voice traffic. If you're prepared to take the risk and achieve the benefits then you must put 100% of your effort into making it work. The good news is that now you are a few years down the VoIP track, you are in a much better position to understand what to look for and how to find the answers. You should now have the tools to analyze and properly understand your VoIP environment. This gives you access to valuable information that will position you well to design and negotiate the best rates, contract and service level agreements with carriers and select the right blend of cost and telephony services.

It's also important to understand your gateway usage and how much you are paying to service users' requirements. You can compare utilization and capacity and make informed decisions about your SIP trunk requirements. You can save on the costs of media gateways because in a SIP implementation gateways needed to connect to the PSTN will reside at the ITSP. As many branches are frequently over-trunked, you can monitor and report on their call volumes, which will give you insight into call flows traversing in and out of the infrastructure and plan for any network upgrades or consolidation that may be necessary.

Good detailed reports are important to prepare you fully for the ITSP RFP process or for entering into new carrier negotiations. Reports also help you plan for cost-effective growth. Once you've moved to SIP trunking, real-time voice quality monitoring and reporting will provide insight to the availability and performance of the SIP trunk aggregation.

Leveraging Least Cost Routing

Creative use of SIP trunking can allow you to reduce calling costs even further if you employ the use of least cost routing (LCR) and multiple ITSPs. If you have information that allows you to assess the types of calls going through your network, and the locations, duration and voice quality, you'll understand your existing call patterns prior to implementing LCR.

Managing a multi-vendor SIP trunked environment

AS SIP provides enterprises with more options for multi-vendor integration, management of a combined environment becomes even more critical. Successful interoperability will reduce complexity for users but for administrators the system is now truly global with many more moving parts to manage. The flexibility of monitoring any multi-vendor PBX directly as well as the SIP trunks enables you to manage individual device availability, interface status, capacity and usage and see the impact of one device failure on another.

Continue reading the full version of this white paper or visit www.prognosis.com/sip

Introduction

Adopting a SIP trunking service is not a simple process. A profusion of choices and decisions awaits you on pricing, service availability, capacity, service level agreements, and much more. That's in addition to selecting the equipment you'll need like SIP-enabled IP-PBXs, session border controllers as well as sizing your SIP trunking implementation correctly.

In fact you may experience a sense of déjà vu as many of the questions you need to ask are those that you already asked (or should have asked!) when replacing TDM with VoIP all those years ago. It's not surprising therefore, that many enterprises need support and guidance. This white paper discusses one of the critical considerations of migrating from TDM to SIP trunks – that of understanding current call activity and volumes across your existing VoIP environment.

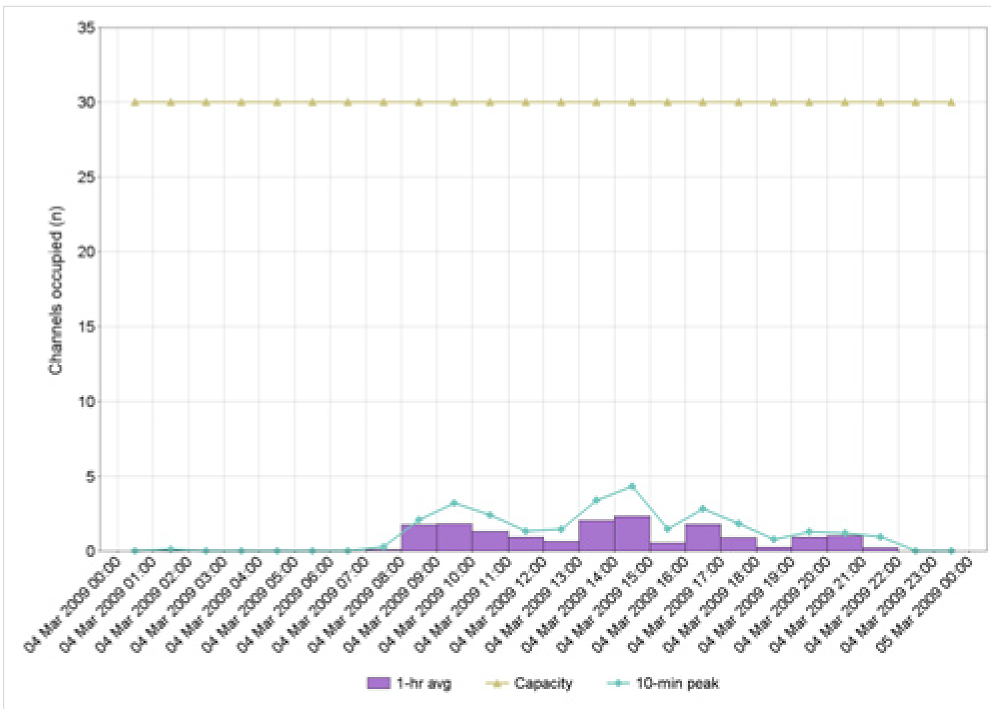
The good news is that now you are a few years down the VoIP track, you are in a much better position to understand what to look for and how to find answers. You should now have the tools to analyze and properly assess your VoIP environment, providing valuable information that will position you well to design and negotiate the best rates, contract and service level agreements with carriers and service providers.

Understanding capacity

Administrators can certainly look at logs and phone bills to try and discover how many minutes they're using per month. Some may even take a random walk through the office showing the number of people on the phone at any given point in time. However doing this for a global enterprise just doesn't stack up. You need to take the guesswork out of capacity planning and know your environment before you change it.

Businesses are wasting enormous amounts of time and money with legacy PRI systems and over-trunking. By collecting information over a day, week or a month you can easily see how much spare capacity you have – that you are probably paying for but not using!

Having this type of information means you can make informed decisions about SIP trunk sizing and network capacity. Below you can see examples of a branch's one-day PSTN call volumes, activity and channel usage. Immediately you can see that this sample location is heavily over-trunked with fewer than five of the 30 available channels being used over the 24 hour period.



24 hour period showing call activity to the PSTN

PBX	Name	Type	Capacity	10-min avg	10-min peak	1-hr peak	Busy hour	In calls	Out calls
\S8300-PBX	1 - OUTSIDE CALL	isdn	30.0	0.7	4.3	2.9	04 Dec 2010 13:40	164	141

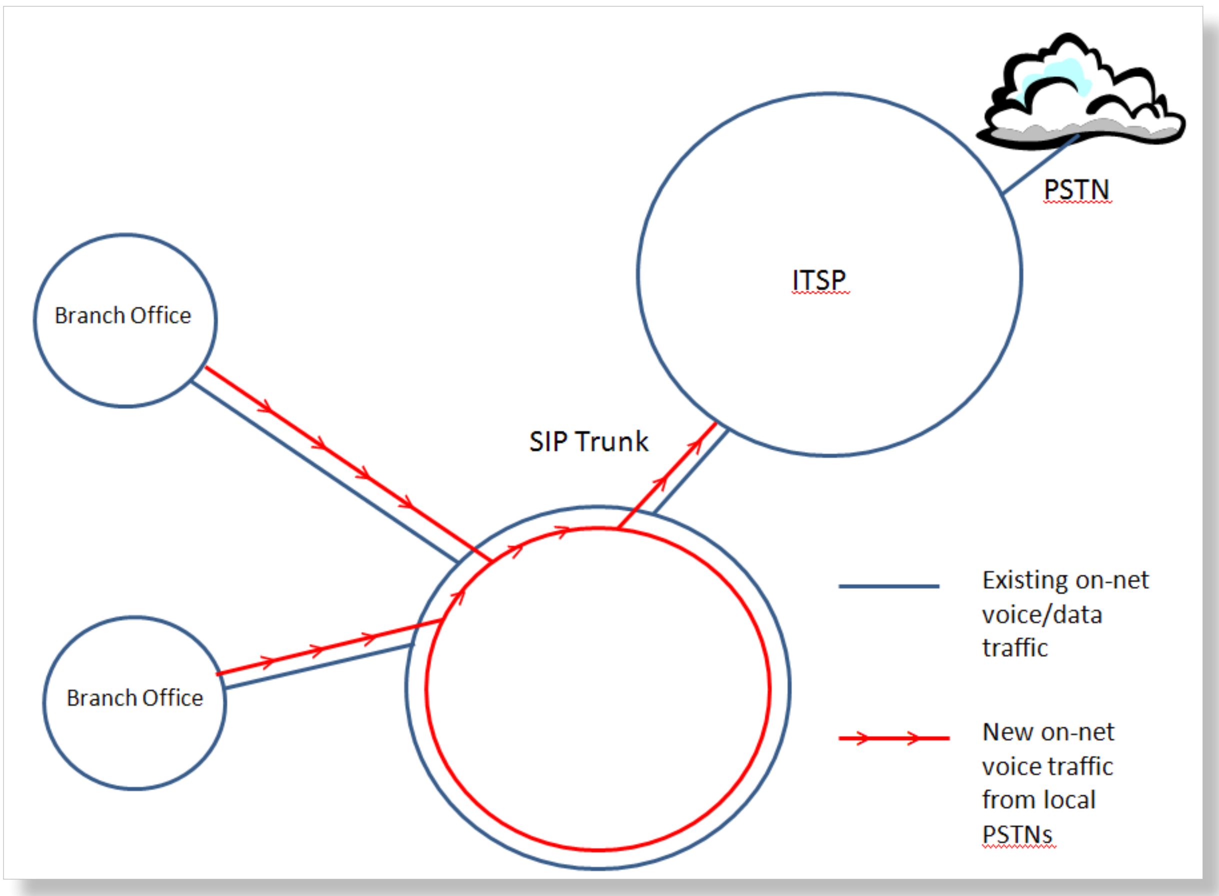
Call activity and PSTN busy hour details

Is your network ready?

In a SIP trunk implementation gateways needed to connect to the PSTN reside at the ITSP, so SIP trunking reduces branch-level PSTN capacity requirements and traditional TDM lines, resulting in cost savings.

However, reducing the local loops' capacity means you're placing increased dependency on the WAN to carry real-time voice traffic. If you're prepared to take the risk and achieve the benefits then you must put 100% of your effort into making it work.

One of the basic questions you need to revisit is that of network readiness. Is your network ready to route the previous off-net branch PSTN traffic across the WAN to the SIP trunk's location? As the figure below shows, the 'off-net' traffic from the branch PSTN links now becomes 'on-net' until it reaches the SIP trunk at Head Office.



Monitoring call flows

If you can monitor and report on availability, capacity, and call volumes, you'll have great insight into call flows traversing in and out of the infrastructure and plan for any network upgrades or consolidation that may be required.

Specialized VoIP monitoring, designed specifically for managing enterprise-VoIP, will show you the number of active calls so that you can establish how many lines are needed. Not everyone uses the phone at the same time. If your company runs a call center, you'll usually require a higher ratio of phone lines to employees but rarely does a business require one reserved phone line per employee.

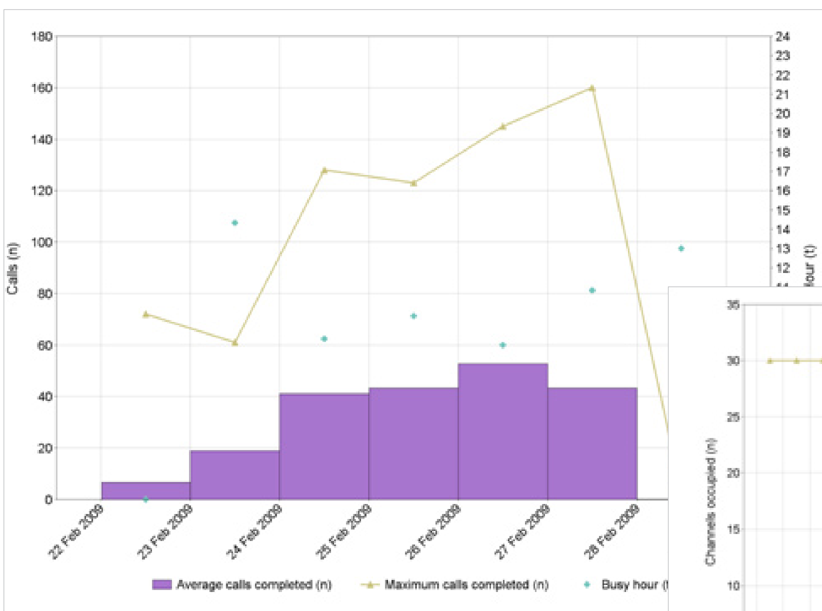
You can also conduct route pattern analysis in real time and historically to establish trends and determine the calls offered and carried and any queuing overflows that occurred.

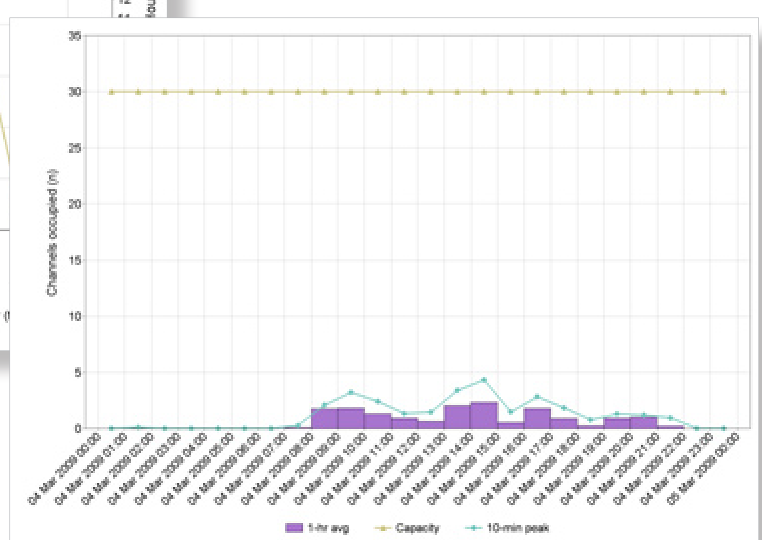
Another benefit of SIP trunking is in bandwidth utilization. VoIP usage is generally characterized by several hours a day with many calls, several hours a day with very few calls and what's known as the 'busy hour'. The busy hour is the busiest daily hour measured by attempted calls. Granular details of call volume can be broken down into voice stream and Erlangs, by PBX or across all PBXs.

This information is useful when assessing the aggregate busy hour as well as the busy hour for each PBX. For example, a company has two PBXs, one in Denver and one in Sydney. The aggregated busy hour totals are calculated when people working in different time zones cross over.

If it is 9-10 am in Sydney, it will be 3-4 pm the previous day in Denver, depending on the time of year. Reports indicate that the afternoon is the busiest time for Sydney and morning is the busiest for Denver. This information lets you know when people are using the telephony system and how far it is pushed during peak time.

In this way you'll know exactly the usage, capacity and head room on your network for the busiest period of each day and for each region. The graphs below illustrate some of these key metrics, in easy to understand format.





Graph showing average 10 minute peaks compared to available capacity

Additionally, if you know how many concurrent calls occur per site, you can take advantage of call load balancing. You can instruct your ITSP or configure your Session Border Controller to help manage capacity in real time, using excess trunk capacity from other locations if needed.

PBX	Name	Description	Calls in	Calls out	Average call length	Total Erlang
\S8300-PBX	566	Peter Johnson	14	12	12.69 min	2.8
\S8300-PBX	614	Lani Jones	3	6	22.22 min	2.2
DVR-PUB-Cluster	SEP0015FADCA747	Bill Allen	3	5	20.87 min	1.7
\S8300-PBX	507	Peter Benn	6	6	15.15 min	1.6
DVR-PUB-Cluster	SEP001647F23657	Wilma Stein	4	10	5.78 min	1.5

Ensuring call quality

By now you should certainly have good QoS policies in place, so you will know if you are achieving and maintaining consistent voice quality. When considering ITSPs, you'll need to confirm they offer support for your preferred codec or codecs, which in turn can influence SIP trunk sizing.

Although SIP trunks are capable of transporting multiple media types, most ITSPs currently only offer G729 or G711 for voice. If it's possible to standardize on codecs – for example G711 throughout, it will reduce the challenge of supporting and transcoding across multiple IP-PBXs. Another benefit is that G711 can provide support for Fax over Voice. If you have selected G729 for voice you'll need to be able to flag fax calls and provide G711 for them on an as needed basis.

Real time, historical online data and custom reports provide very useful statistics to understand the call quality that's being achieved prior to SIP trunking. Real time displays and historical reports help uncover patterns of unacceptable voice quality and narrow problems down to particular groups of phones, regions, locations, route patterns or time of day. Furthermore, an end-to-end view of the entire VoIP ecosystem means you can correlate, diagnose and resolve network performance issues that can affect the quality of your VoIP service.

Using customized reporting

Good documentation via detailed reports is important to prepare you fully for the ITSP RFP process or for entering into ISP/ITSP negotiations. Reports can be used for executive and operational purposes to provide quantitative measurement of the actual service delivered. The type of information it's useful to know includes:

Voice quality	Voice quality over time, exceptions, network impairment factors
Service levels	Route pattern availability, IP and PSTN trunk availability
Capacity planning	PBX call load and utilization, trunk utilization, phone counts
Analytics	Call failures, long calls, call types, node utilization, call throughput

Summary report types also provide an "at a glance" view of grouped components' faults, failures, availability and utilization. As well as identifying problems and establishing trends, they help you plan for cost-effective growth.

Helping resolve inter-operability issues

As SIP continues to evolve, a variety of equipment compatibility and interoperability problems has occurred. In order to successfully migrate to SIP trunking, your IP-PBXs, SIP trunk service and border devices must be compatible. If they aren't, you can't complete voice calls.

You can use specialized VoIP management to provide details of your VoIP ecosystem to help identify compatible and non-compatible devices. Equipment inventory, updated in real time, can prove invaluable in assessing the network components that may need attention prior to SIP migration.

Achieving least cost routing

Creative use of SIP trunking can allow an enterprise to reduce calling costs even further if you employ the use of least cost routing (LCR) and multiple ITSPs.

LCR is the process by which a company chooses the path of an outbound communication by the price of the call, opting for the lowest price. If a company uses multiple SIP trunks from different service providers based on geographic locations and time zones, each call can be routed to the cheapest provider based on country codes, saving a significant amount of money on international calls.

Information about the types of calls traversing your network, their location, duration and voice quality helps you understand your existing call patterns prior to planning LCR.

ITSP service validation

Once you've deployed SIP you need to monitor the quality of service during initial SIP deployment and ensure you're getting everything you signed up for. Naturally this will vary between customers and platforms but if you have real time and historical data available you can see details of quality of service, availability, utilization, throughput and performance of your new SIP trunks.

Monitoring session border controllers

Although a border element maintains connection logs you don't want to be opening them for analysis each time you need updated information. Specialized VoIP management will do this easily for you in real time. You can see at a glance the total number of available sessions, broken down to inbound and outbound, the SBC's status and network details such as the maximum burst rate and latency metrics.

Managing multi-vendor integration

As SIP provides enterprises with more options for multi-vendor integration, unified management of a multi-vendor environment becomes even more critical. Successful SIP trunking reduces costs and complexity for users but for administrators the system is now truly global with many more moving parts to manage.

The flexibility to monitor multi-vendor PBXs, different vendors' border controllers and SIP devices enables you to monitor individual device availability, interface status, capacity and usage and see the impact of one device on another.

Real-time monitoring of voice quality, route usage, capacity and failover activity provides a complete picture of a multi-vendor environment making it possible to streamline processes, teams and toolkits for both the enterprise and service provider.

In conclusion, an assessment of your current VoIP environment gives you invaluable information that will position you well to design and negotiate the best rates, contract and service level agreements with carriers and service providers.

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Integrated Research

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Prognosis for IP telephony was launched in 2000, and has been chosen by many of the world's largest companies to manage their IP telephony deployments. These include the world's two largest aerospace companies, two of the largest US electric utility companies, three of the largest US financial services firms, and the largest telecommunications companies in the US, Germany and France.

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