SUCCESS with SIP

Introduction

Voice over Internet Protocol (VoIP) technology and software has been around since the 1990s. While mostly proprietary, many access and signaling protocols have been developed for VoIP. The development of the Session Initiation Protocol (SIP) standard has brought a level of standardization to VoIP. SIP can also support video, Instant Messaging (IM), and many other forms of media. Learning the value and operation of SIP is now important to anyone working in the field of communications and collaboration.

You hear SIP discussed, products introduced, and services offered frequently, but without much detail. What do you need to know about SIP that is important before investigating SIP supported products and services? The most common applications are SIP Trunking, followed by SIP phones. There are other SIP implementations not quite as well known that support a range of devices and interfaces beyond SIP trunks and phones.

Understanding what SIP does and does not support helps to grasp the implementation issues involved. Multiple elements in the SIP path must each be successfully interoperable with each other. However, when implementing SIP trunking, the enterprise can encounter a number of issues. The IP PBX and Session Border Controller (SBC) vendors and the SIP trunk provider can all contribute to the problems.

This paper will provide insight into

- **SIP and related protocols**
- **Session Border Controllers**
- **SIP trunk values, benefits, and economics**
- **SIP phones**
- **Interoperability with SIP Trunks**
- **SIP trunk problems (IP PBX, SBC, trunk provider) and their resolution**
- **Bandwidth calculations for VoIP and SIP trunks**
- **Delivering a successful SIP implementation project**

Session Initiation Protocol (SIP) Defined

Everything encompassed within the Session Initiation Protocol standard is defined in the IETF RFC 3261. SIP supports session signaling and session control. It is not specific to Voice over Internet Protocol (VoIP) and can be used to establish and control voice, video, Instant Messaging (IM), file transfer, games, and other media sessions. The Session Description Protocol (SDP) defines the media carried over the SIP session.

The five SIP functions:

1 **Device Type/Location:**

SIP can locate the user and determine what end system will be used in the proposed session.

2 **Device Presence:**

It can then learn the user availability; can the user be disturbed or is the user busy.

3 **Device Capabilities:**

SIP can determine the capabilities that are available at the user end system for the session.

4 **Establish Connectivity:**

The fourth function for SIP is to establish the session.

5 **Manage Communications:**

The fifth function of SIP is for managing the session such as call termination, call transfer, or changing the session parameters during the call.

The Use of SDP

SIP does not define a phone call. Another protocol, the Session Description Protocol (SDP) defined in IETF RFC 4566, identifies various forms of media that can be carried over the SIP connection. SDP can define a voice call, a video call, or IM event. The IM standard used with SIP is called SIMPLE (IETF RFC 2779).

SDP is used to describe VoIP call characteristics such as the compression technology used during a call. The voice and video packets carried over the session use the Real Time Protocol (RTP), IETF RFC 3550. SIPPING 19 (IETF RFC 5359) defines 19 standard telephony features that can be used with SIP VoIP calls. Many vendors have expanded the telephony feature list with proprietary extensions.

SIP Trunks

If there is any recurring theme in IT today, it is a flat, or reduced, IT budget. One candidate for helping you work within a tight budget is using SIP trunking and retiring the old T1 and ISDN Primary Rate Interface (PRI) carrier connections. In many cases, SIP trunking will produce enough savings to easily offset the cost of the change.

A SIP Trunk is an alternative to the traditional PSTN T1 and PRI connections. The customer's IP PBX can communicate directly without a gateway or a TDM PBX through a gateway over an IP connection. The VoIP to PSTN conversion is then performed by the service provider. SIP trunking can refer to a service provided for connection to the PSTN, a SIP port on a server for interconnection to other server systems (e.g. voicemail), and as a connection between PBX's that replaces tie lines.

The reasons for moving to SIP Trunking are:

- **Lower your cost compared to T1/PRI trunks**
- **Create a flexible service**
- **Offer new services**

While these reasons are valid, SIP Trunking is not as plug-and-play as the traditional T1 and PRI trunks. This may result in a higher number of trouble tickets in the first year for a SIP trunk than for comparable T1/PRI trunks.

The Session Border Controller (SBC)

A major concern with SIP Trunking is security. Another concern is the monitoring of the trunk connection. It is recommended that the SIP trunk operate through a DMZ just the same as any Internet connection. The SBC is the most common solution to the security and monitoring problems. If you are not familiar with the SBC, you should learn more before embarking on SIP Trunking.

The SBC is the alternative to a firewall and is better suited to the task of providing security.

Here are the reasons for selecting a SBC:

- **Provides the firewall rule set while also mapping layer 5 Presentation to layer 7 Application addresses.**
- **Intrusion detection and prevention**
- **Denial of Service (DoS) attack prevention**
- **VPN separation for shared resources**
- **SIP-TLS transmission**
- **Secure RTP support**
- **Possibly supporting IPsec Tunnels**

The SBC can also be used for transcoding and conversion between different VoIP codec technologies.

SIP Trunking Economics

Since most enterprises have some form of IP network, the concept of using IP to connect to the public switched telephone network (PSTN) is attractive.

Flexible Provisioning

One reason to swap out legacy T1 and PRI trunking is their lack of flexibility, which has a cost associated with it. The T1 comes with 24 64 Kbps channels, and the PRI supports 23 64 Kbps channels. So, if an enterprise needs 32 channels, then two T1 or PRI connections are required, which is considerably more than the number of channels actually needed to accommodate traffic loads. The SIP flexibility allows the enterprise to implement just what is required, exactly 32 channels, avoiding overprovisioning.

You can expand or reduce the number of channels rapidly, which is helpful for the many enterprises that have seasonal or other variations in their communications requirements. Those that have major Christmas sales, for example, have about a three-month period during which channel requirements can easily be 300% higher than average. SIP trunking allows the enterprise to increase and decrease capacity without having to add and delete trunks every year – and avoid paying the associated costs to the carriers.

Provider Competition and Enterprise Leverage

Most enterprises have had long relationships with their carriers, because competition has traditionally been fairly limited. Now, with SIP trunking, there are a number of other provider choices. This creates significant competition, which the enterprise can use to obtain more favorable agreements at lower prices. The situation gives enterprises far more leverage with their incumbent carriers, forcing the incumbent to price its services lower for the purposes of customer retention.

For example, an enterprise might wish to provide for a failure situation. It is relatively easy to contract with two SIP trunk providers for backup or, better yet, a shared-load connection. This puts the two providers in competition for price, reliability, and service, all to the benefit of the enterprise.

Other Sources of Savings

Generally, as the number of sites increases, so does the cost advantage of SIP trunking. This is due to economies of scale; wherein the more bandwidth you buy, the lower the cost per unit of bandwidth. SIP also allows the aggregation of multiple connections into one or two connections. There are a number of other areas where SIP trunking services can provide savings, described below.

- ***VoIP Gateways*.**
  When an enterprise moves to an IP PBX, VoIP gateways are required for connecting to the PSTN. With SIP trunking, the PSTN gateway is unnecessary. Besides avoiding the purchase cost of the gateway, the enterprise eliminates another device that has to be maintained and its software managed.
- ***Reduced Conferencing Costs.***
  Audio conferencing can be costly when using carrier legacy trunks. SIP trunk providers also support conferencing bridges and can eliminate the conferencing charge.

- ***International Calling.***
  More enterprises are developing international locations and international customers. The long-distance charges for some enterprises have exceeded their domestic call charges. SIP trunks can be used with VoIP services such as Skype. SIP trunk providers can do a better job of least-cost routing of international calls than the incumbent legacy carrier. This can be very attractive for international mobile travelers and can reduce the call charges by as much as 90%.
- ***On-Net Free Calling.***
  Most SIP trunk providers offer free calls among the enterprise's SIP trunk sites. Calls between enterprises can also be made without extra charges, though this capability might not be available in all countries. India, for example, does not allow toll bypass at this time.

SIP Phones

While SIP trunking won't be a cost-reduction solution for every organization, it will be for most. The enterprise should investigate SIP trunking – if only to eliminate it as a solution.

Traditionally, IP PBX vendors have developed their own proprietary protocols for use with their IP phones. Skype and some other voice services have their own proprietary protocols that perform functions like SIP.

The introduction of SIP phones opens up the IP phone market for third party IP phone vendors. Most IP PBX vendors offer connections to SIP IP phones, however there may be feature or function limitations when third party SIP phones are deployed.

More SIP Applications

We continue to hear about SIP phones, gateways, Session Border Controllers, and firewalls. But did you know that there is a growing world of SIP devices that most enterprises have never heard of that are now available?

SIP is used in other applications such as Computer Telephony Integration (CTI), connecting to servers with software from companies such as IBM and Microsoft. Some IP PBX vendors use SIP to communicate with VoIP gateways. A few IP PBX vendors use SIP for Trunking between their IP PBXs.

Other applications of SIP include:

- Door phone
- Audio Alerter
- Callbox
- Multimedia Intercom
- SIP cameras for video surveillance
- Clocks
- Paging systems

Ensuring SIP Interoperability

In the early days of SIP trunking, each IP PBX vendor supported a different version. The SIP trunk provider had to configure their interface to satisfy each vendor, which led to a number of problems. Providers did not want to support many different versions. This problem forced the vendors and providers to select a common form of SIP trunk interface called SIPconnect. The SIP Forum is the creator of SIPconnect documents. Think of it as an agreed-upon portion of the SIP trunk standard that the participating vendors and providers voluntarily adopt.

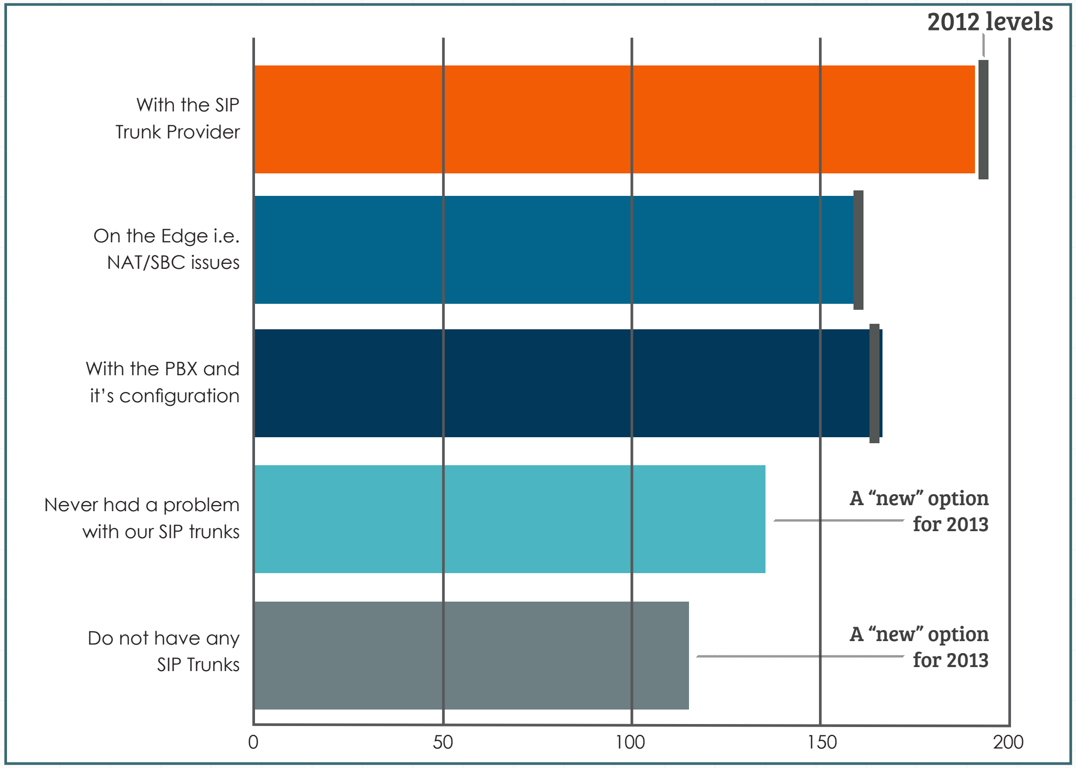
"The SIPconnect Technical Recommendation is an industry-wide, standards-based approach to direct IP peering between SIP-enabled IP PBXs and VoIP service provider networks"

- http://www.sipforum.org/sipconnect

The SIP Forum Board has announced the SIPconnect 1.1 Technical Recommendation that it is stable and is believed to have resolved known design choices. Your provider and IP PBX or SBC vendor may be using the original version or may have adopted SIPconnect 1.1. Ensure you know that you are working with the same version for all the vendors and providers.

Problems with SIP Trunking

Every year The SIP School™ performs international surveys of SIP implementations. "The SIP Survey 2013" www.thesipschool.com/survey2013.html, presents the status and problems relating to SIP trunking. The chart below shows the three problem areas and their comparison to the results of the 2012 survey. You will note they are very much the same. There is no appreciable improvement by the providers or equipment vendors. One hopeful note is that a few enterprises had no problems at all but they are in the minority.

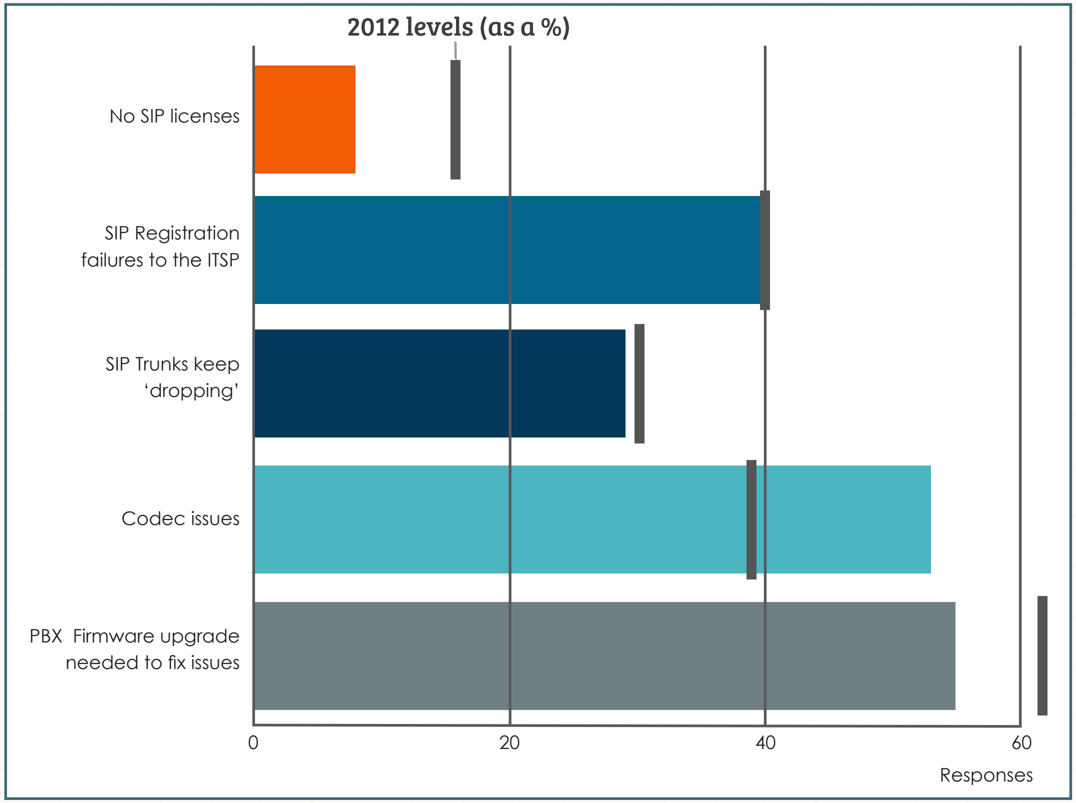


Survey Question: "If you've had problems, where have the issues been?

Source: "The SIP Survey 2013"

IP PBX Problems

There are five areas of concern with the IP PBX vendors SIP trunks support. Most have improved or remained the same since 2012.



Survey question: "If the problems were found to be with your SIP/ VoIP based PBX, what were they?"

Source: "The SIP Survey 2013"

- ***"PBX Firmware Upgrades" have become less of a problem.***

  The upgrades remain the biggest problem. Check with the vendor to determine if there are updates that may affect SIP trunk deployment and operation.

- ***"SIP Registration failures to the ITSP" have remained the same as the 2012 survey.***

  No improvement there. This is probably improper configuration on the part of the IP PBX enterprise staff or more likely the VAR. The questions are the effectiveness of enterprise and VAR training as well as the vendor documentation. Is it just poor typing skills and/or unverified data entry?

- ***"SIP Trunks Keep Dropping" has declined slightly as a problem.***

- ***"No SIP Licenses" is hard to believe.***

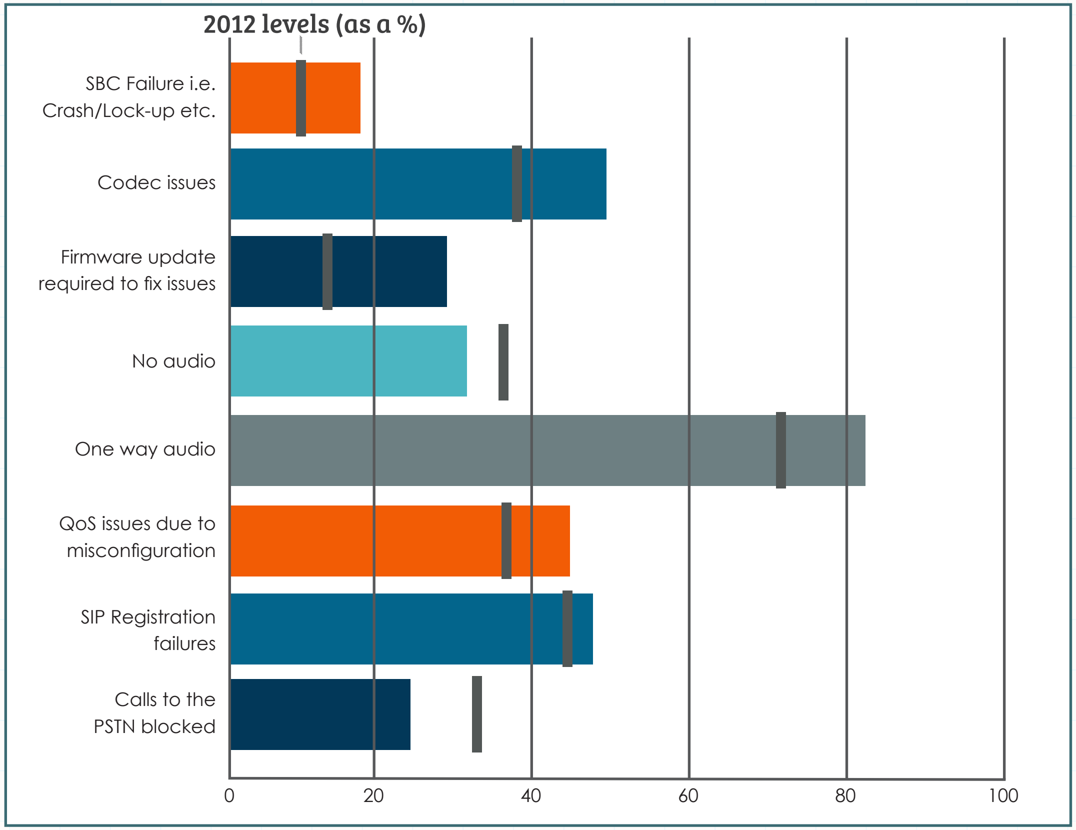
  This begs the question "How much knowledge do enterprises have of the IP PBX they have implemented?" This should be a no brainer. It should not occur at all. Check first with the vendor before deploying SIP trunks. The SIP license may not be included in the procurements.

- ***"Codec Issues" has increased since the 2012 survey.***

  This is most likely a mis-match configuration problem. Check the settings carefully.

Edge Device Problems

Let's first look at what has improved since the 2012 survey. The "No Audio" and "Calls to the PSTN Blocked" have diminished as problems. In six other categories, the problems have increased, not decreased from the 2012 survey levels.



Survey question: "If your problems were with your SBC / Edge devices, what were they?"

Source: "The SIP Survey 2013"

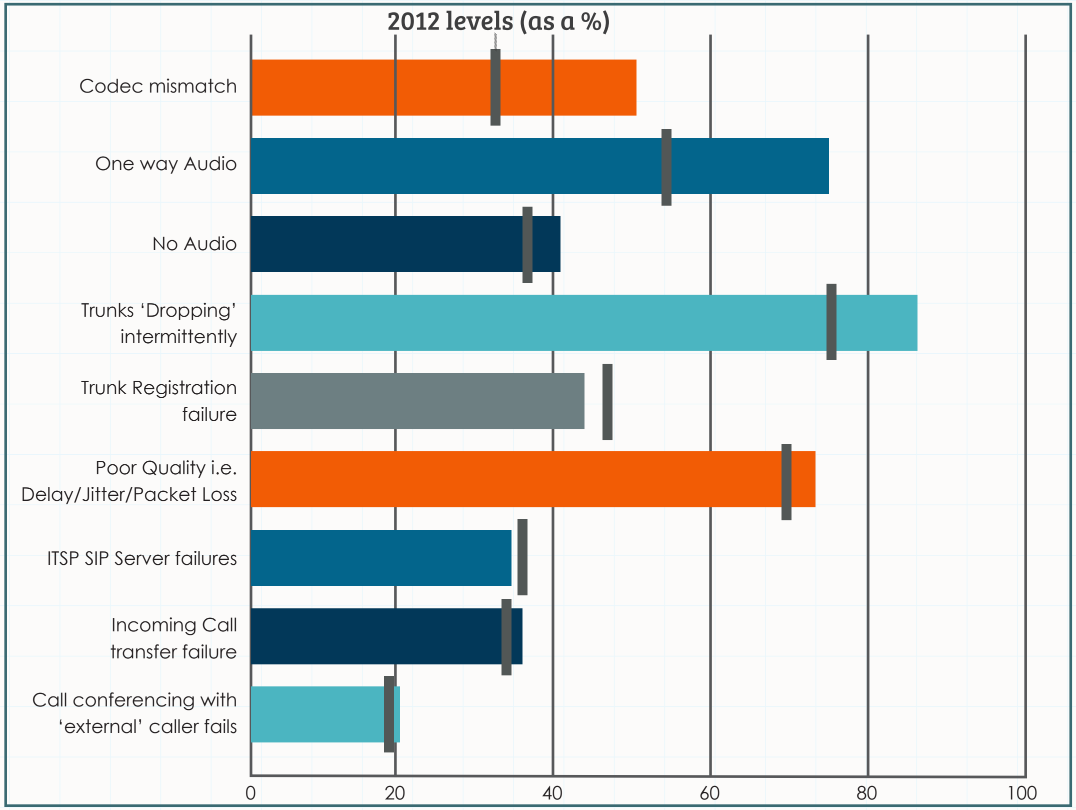
The biggest problem "One Way Audio" remains the primary headache with the SBC. This was the worst problem in 2012 and remains the worst problem in 2013. The rise in this number is significant. This issue strongly highlights the importance of testing the components together before deploying and initializing the SIP trunking service. The results of the testing could influence the purchasing decisions of edge devices.

- ***"One Way Audio"*** is most often the result of "NAT breaking SIP" which means that since SIP operates at the Application Layer and the NAT is created at the transport layer of the network, media often cannot reach the SIP device being used in the network because it's private IP address is not routable outside the Local Area Network. One of the beneficial functions of the SBC is to resolve that NAT traversal issue and to rewrite the header information so that SIP can reach those devices
- ***"Codec Issues"*** is the next most common problem. This should not be on the rise as those working with this 'specialized' equipment should have a good understanding of codecs. The enterprise staff and VARs that are able to work with others involved in an implementation should ensure that codecs are configured correctly and tested thoroughly (the big issue).

- ***"SIP Registration Failures"*** is next on the list which is most likely a configuration problem.
- ***"QoS Issues"*** is another problem probably due to misconfiguration.
- ***Required "Firmware Updates"*** to fix problems worsened considerably. This appears to be negligence on the part of the VAR or IT department. It is also possible that the SBC vendor is not being diligent in their downloads.
- ***"SBC Failures"*** such as a crash or lock-up have increased but remains a minor problem. This problem is solvable during the testing phase and should not have increased.

SIP Trunk Provider Problems

The 2012 results are shown as a percentage of the 2013 results. Compared to 2012, "Trunk Registration Failures" have improved. "ITSP SIP Server Failures" remained about the same as did "Call Conferencing with External Caller Fails. " Incoming Call Transfer Failure" problems were a little bit worse in the 2013 results. The remaining five areas of concern worsened in the 2013 survey, something that would not be expected since SIP trunk implementations are not new to the providers.



Survey question: "If you've had problems that were found to be on the SIP trunk provider side, what were they?"
Source: "The SIP Survey 2013"

The problem that was worst in the 2012 survey is still the top problem in the 2013 survey, "Trunks Dropping Intermittently." It is even more common than ever. This problem should be addressed during testing / trials as losing a call during a conversation is one of the most frustrating events to occur. This wastes time and the call may be difficult to re-establish. The caller may have to resort to a mobile call which defeats the purpose of the SIP trunk.

- ***"One Way Audio"*** is far worse in 2013 than in 2012 and is the second most common problem. This is very likely a provider configuration issue or it could be attributed to improper configuration of the SBC.
- ***"Codec Mismatch"*** should not really happen as 'configuration settings' should ensure a match. This can be fixed on the SBC to conform to the settings of the provider.
- ***"Poor Quality"*** is an old problem with known established corrections. It is difficult to understand how this problem is increasing in frequency. Delay, Jitter, and Packet loss has been discussed for years. Is it due to poor documentation or negligent configuration practices in the network infrastrutrure?
- ***"Trunk Registration Failures"*** is a little worse in 2013 than 2012 and should not be happening. This could be the fault of improper configuration or a symptom of a provider software problem.
- ***"No Audio"*** also increased slightly. Again this is probably a provider configuration issue or SBC generated problem.

All of the issues in the chart above have been around since the beginning of SIP trunk implementation. It is disappointing that the industry still has not reduced the problem frequency. These basic implementation issues should be eliminated during the initialization of the SIP trunks.

It begs the question of whether the implementation teams are overworked, undertrained or both. Or could it be that the providers have not made these problems high priority on their list of improvements? It would seem that the cost to the provider for resolving these problems could be avoided by further investment in the implementation team along with the necessary testing when SIP trunks are installed.

Preventing the Problems

There will be at least three interested parties to the SIP trunk implementation, IP PBX/UC vendor, SBC vendor, and SIP trunk provider.
Do not try to take the responsibility of coordinating these three without getting them together before and during the SIP trunk implementation.

Testing the three elements thoroughly is very important. Test the configuration, features to be implemented, and the capacity (number of simultaneous calls) that is required. Make no assumptions. The enterprise will be responsible for the assumptions, not the vendors or providers.

The vendors and providers probably have checklists of what they will perform during the implementation. Obtain these checklists and verify they have used the checklist to validate the installations. If there are no checklists, create your own checklists by communicating with other enterprises that have implemented SIP trunks. Use The SIP Survey 2012 to anticipate the issues that can occur. Knowing what has gone wrong at other implementations highlights the issues to be tested.

Some technicians keep tweaking the configurations and settings until it does work. This can be dangerous as the tweaking can turn off features, change security, or produce a liability that will show up later. If the tweaking is performed, ensure that adequate and correct documentation of the changes is produced so there is an audit trail available if problems occur in the future. If not, the tweaking starts all over again and the enterprise will continue to encounter outages or poor operation. For example, Transport Layer Security (TLS) or Secure Real Time Protocol (SRTP) may be turned off and thereby eliminating the security features. Codec changes or more or fewer ports on the SBC could be configured improperly.

When the SIP trunk does not work properly, remember that the problems could also be the result of the IP PBX or SBC improperly configured. This makes it more difficult to pinpoint the culprit. If the corrections performed by the SIP trunk provider do not work, enlist the aid of the SBC vendor and IP PBX vendors before you declare the SIP provider as incompetent. Most of the problems appear to be configuration mistakes. It may be that the provider's implementers do not have enough training, experience, and adequate documentation especially for troubleshooting.

VoIP Bandwidth Calculations

The bandwidth that is needed for VoIP transmission will depend on a few factors: the compression technology, packet overhead, network protocol in use and whether silence suppression is used.

There are two primary solutions to delivering IP network performance for voice: bandwidth allocation and QoS. QoS is not discussed in this paper. How much bandwidth to allocate depends upon:

- ***Packet size for voice (20 to 320 bytes of digital voice)***
- ***Codec/compression technique (G.711, G.729, G.723.1…..)***
- ***Header compression of RTP + UDP + IP called cRTP, which is optional***
- ***Layer 2 protocol used, i.e., PPP or Ethernet***
- ***Silence suppression/voice activity detection assumptions***

The results of two G.711 and two G.729 calculations are contained in the next table "Minimum Bandwidth requirements" The table illustrates three points:

- ***Bandwidth requirements reduce with compression, G.711 vs. G.729.***
- ***Bandwidth requirements reduce when longer packets (i.e., more bytes per packet) are used, which reduces packet overhead, and bandwidth requirements. The enterprise may have no control over the speech packet size implemented.***
- ***Compressing the RTP, UDP, and IP headers (cRTP) is most effective when the packet also carries compressed voice (G.729).***

CODEC	VOICE BIT RATE	SAMPLE TIME	VOICE PAYLOAD	ETHERNET	PPP	
					RTP	CRTP
G.711	64 Kbps	20 msec	160 bytes	88 Kbps	83 Kbps	68 Kbps
G.711	64 Kbps	30 msec	240 bytes	80 Kbps	77 Kbps	67 Kbps
G.729A	8 Kbps	20 msec	20 bytes	32 Kbps	27 Kbps	12 Kbps
G.729A	8 Kbps	30 msec	30 bytes	24 Kbps	20 Kbps	11 Kbps

Minimum Bandwidth Requirements
(Actual bandwidth requirements may be different for each provider)

The varying designs of packet size, voice compression choice and header compression can make it difficult to determine the bandwidth for a voice call. Many providers have selected 20 ms or 30 ms of speech for the payload size. The SIP trunk provider should be asked to provide a table like the one above for their service, for use in calculating the bandwidth requirements. The provider's bandwidth requirements may be greater than those in the table. A good rule of thumb is to reserve at least 27 Kbps of SIP trunk bandwidth per call for 8 Kbps G.729 compressed voice. If G.711 is used, then reserve at least 83 Kbps of bandwidth per call.

Silence suppression assumes that that both parties of a call do not speak at the same time. On average, a voice call can have as much as 70% of silent time. It is recommended that the designer assume that silence suppression only reduces the bandwidth requirement by 30%. Silence suppression should be used on trunk groups with greater than 24 lines (voice paths).

First, experiment with the silence suppression turned off. Then turn it on and observe if there is voice quality degradation. The degradation may be more than is acceptable. When music-on-hold is in use, then silence suppression may not work well. The use of FAX and PC modems through a SIP trunk will negate the use of silence suppression. Bottom line: Be cautious with the application of silence suppression.

Designers should not calculate the required number of trunks with the minimum number in mind. Always round up to a larger number of trunks. Traffic estimates are just that, estimates. It is better to increase the bandwidth than to have dissatisfied callers. Experiment with voice quality to ensure that adequate bandwidth is implemented.

Every SIP trunk provider appears to support the uncompressed G.711 and most support G.729. However, not every provider may accept compressed RTP packets in all situations. The delivery of compressed RTP may not be supported at the enterprise SIP trunk interface (PBX or IP PBX).

It is very likely that a Session Border Controller (SBC) will be placed between the enterprise and the SIP trunk provider. Investigate the SBC to learn if the SBC can perform the RTP compression and decompression and G.711-to-G.729 transcoding (conversion).

Sizing the SIP Trunk for Voice/Fax/IVR Calls

The proliferation of SIP trunking makes the bandwidth calculations for those SIP trunks an issue that the IT organization must resolve. Too much capacity and money is wasted. Too little capacity and calls will be blocked, callers will abandon the call, the enterprise agents will be less productive, and the enterprise's reputation for service may be harmed.

If the enterprise already has T1 and PRI connections, then calculating the number of inbound and outbound voice paths over the SIP connection is straight forward. The designers just provide the same number of calls paths. However, since T1 connections have 24 call paths and a PRI has 23 call paths, the enterprise may have rounded up to the next larger connection and thereby has oversubscribed. One value of SIP trunking is that the enterprise can request precisely the number of paths needed which can be fewer than previously implemented. Investigate how the voice path calculations were performed. You may find you can work well with fewer call paths than previously designed.

If there are not T1 or PRI connections or if the designers want to go back and produce a new design, then the designers need to apply some calculations to determine the minimum number of call paths required. The process of determining the required trunk capacity starts with determining the Grade of Service (GoS), which is the probability of a caller hearing a busy signal at a call center, enterprise office or any other location designed to receive calls, whether voice, Fax, or Interactive Voice Response (IVR) calls. Trunks in this case are the number of lines needed to carry the calls.

Erlang Calculations

The number of inbound call paths required for a call traffic load can be calculated using Erlang B calculations. The Erlang formulas have been used for about 90 years for telephone network capacity planning. There are several other formulas with slightly different assumptions.

Erlang B is a formula that can be used in call center scheduling. The formula assumes that an unsuccessful call (the call is blocked, the caller gets a busy signal), is not queued or retried; it is lost forever. The formula also assumes that call attempts arrive independently of the time since the last call.

Because Erlang B doesn't assume calls are retried, it tends to underestimate the number of trunks required. A variation, the Erlang B Extended formula, can account for 10% to 70% of the callers who will immediately retry if their calls do not go through. The Extended formula will produce a slightly greater number of trunks required to carry the call load.

The formulas (Erlang B and Extended) can be used to calculate any one of the following three factors if you know or can predict the other two factors:

- ***Busy Hour Traffic (BHT):***
  The number of hours of call traffic during the busiest hour of operation also called the Erlang load.
- ***Blocking (busy signal Grade of Service (GoS)):***
  The percentage, for example 1%, of calls that are blocked because not enough lines are available.
- ***Lines:***
  The number of lines in a trunk group. One line can carry one call at a time.

The busy hour is the heaviest traffic period during the day. By designing for the busy hour, callers will experience call blocking at the enterprise's desired rate. Other hours of the day will have less traffic volume. This means that the callers will experience fewer blocked calls (busy signals) the rest of the day. The blocking performance will be better for all the hours outside the busy hour, delivering a better GoS. The worst case performance (GoS) is delivered during the busy hour.

This is how to use the Erlang B calculators. The first determination is more of a business issue: how often is it acceptable for the caller to get a busy signal? Most calculations start with a GoS of .01 (1% busy) which means that 99% of the calls are answered and do not receive a busy signal. A GoS of .001 means that 99.9% of the calls do not receive a busy signal. The design for Interactive Voice Response (IVR) systems should have a very high probability that the call is not blocked—i.e., 99.9% or better.

Next, the traffic load must be either measured or estimated in Erlangs/BHT. One Erlang is equivalent to one trunk/line busy for one hour. To calculate the Erlang load, the trunk designer must determine the average length of a call in minutes. The number of calls expected during the busiest hour of the day is also necessary. The Erlang load (BHT) = CAR X H/60 minutes

Where:

- Call Arrival Rate (CAR) is the number of calls during the busy hour
- The average call length or Holding (H) time is measured in minutes

A sample calculator with a calculation example can found at Anaspoint www.ansapoint.com/calculator/erlb/.

A Successful SIP Trunk Project

There are best practices that should be followed to deliver a smooth and problem free SIP trunk implementation. SIP trunking implementations are maturing but not mature.

These eight best practices will help to ensure a smooth implementation.

1. Be clear about your objectives and the key indicators that will demonstrate you have met the objectives.
2. Ask, ask, ask, questions. If you make assumptions you are responsible for them not the vendors and providers.
3. Coordinate with the vendor(s) and provider(s). Have then all meet together with the enterprise staff tasked with the implementation.
4. The problems discussed in the SIP Survey 2012 are the common challenges that will be encountered. Anticipate them. Do not be surprised if they occur during the implementation.
5. Have a good test plan. Assume your implementation is unique no matter what the vendors and providers state. There can be differences in SIP protocol headers, differences in error codes, and DTMF signaling requirements. Verify the software releases to be used. FAX works differently with nearly every SIP trunk provider. Plan extra time for interoperability testing.
6. Ensure you have current and adequate documentation and configuration guides from all parties involved in the implementation.
7. Look for management tools from the vendors and providers that support reporting for capacity planning, voice quality, service levels, and security.
8. Do not expect the time allocated will be enough. Add some time for unanticipated issues.

Resources for More Information

The SIP School www.thesipschool.com has developed a SIP online training and certification program. This program is featured at http://nec.thesipschool.com.

There are several good articles that are archived at the www.nojitter.com site.

IPT and SIP Trunking Slowly Gaining Ground

SIP Trunking Interoperability Update

SIP Trunking: Bullish Calculators Yet Bearish Adoption

About the Author

Gary Audin has more than 40 years of computer, communications and security consulting and implementation experience. He has planned, designed, specified, implemented and operated data, LAN, WAN, and telephone networks. These have included local area, national and international networks as well as VoIP and IP convergent networks in the U.S., Canada, Europe, Australia, Caribbean and Asia.