SUCCESS WITH SIP 2.0

by Gary Audin

Introduction

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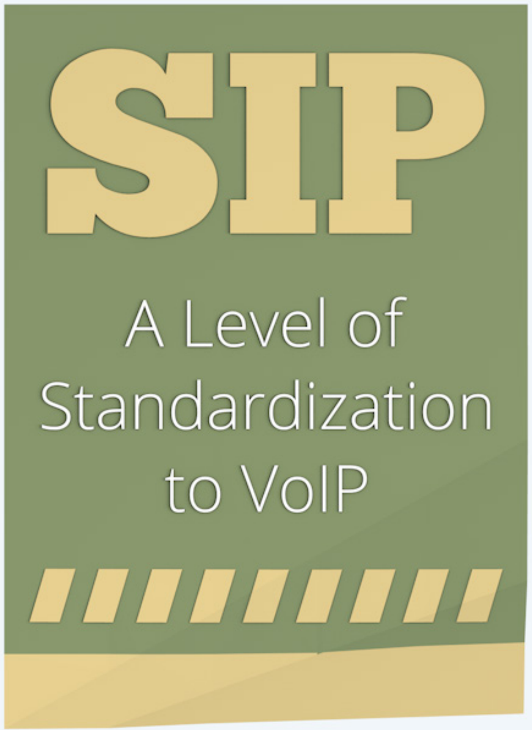
Voice over Internet Protocol (VoIP) technology and software has been around since the 1990s. While mostly proprietary, many access and signaling protocols have been developed for VoIP. The development of the Session Initiation Protocol (SIP) standard has brought a level of standardization to VoIP. SIP 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 frequently hear SIP discussed, products introduced, and services offered, but without much detail. What important information do you need to know about SIP before investigating SIP supported products and services? The most common applications are SIP trunking, followed by SIP phones. SIP can also be used for WebRTC session control, and there are other lesser known SIP implementations that support a range of devices and interfaces beyond SIP trunks and phones.

Understanding what SIP does and does not support helps you 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, an enterprise can encounter a number of issues. The IP PBX and Session Border Controller (SBC) vendors and the SIP trunk providers can all contribute to the problems.

This paper will provide you with 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

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Session Initiation Protocol (SIP) Defined

The Session Initiation Protocol standard is defined in the IETF RFC 3261 and updated in RFC 6878. 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 SPD

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

SDP is used to describe VoIP call characteristics such as the codec and 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) updated in RFC 7088 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

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

A SIP trunk is an alternative to the traditional Public Switched Telephone Network (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 IP PBX's that replaces tie (T1 and PRI) lines. SIP trunks can also be used for inter-site communications.

Reasons for moving to SIP Trunking are:

- Lowers your cost compared to T1/PRI trunks
- Creates a flexible service
- Offers new services such as free on-net (SIP to SIP) calling
- Flexible business continuity

While these reasons are valid, it should be noted that 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's operation than for comparable T1/PRI trunks.

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Session Border Controller (SBC)

Security is a major concern with SIP trunking, affording protection from malicious intrusions. Another concern is the monitoring of the SIP trunk connection. It is recommended that the SIP trunk operates through a Demilitarized Zone (DMZ) the same as any Internet connection. The SBC is the most common solution to these security and monitoring issues, so it is recommended that you learn more about the SBC before embarking on SIP trunking.

The SBC is the alternative to a firewall and is better suited to providing security for VoIP connections. It can also perform transcoding, the converting of one form of VoIP to a form compatible with the SIP trunk provider interface.

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

SIP Trunks Economics

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

The number of channels can be rapidly expanded or reduced, which is helpful for the many enterprises that have seasonal or other variations in their communications requirements. For instance, a business with major Christmas holiday sales has approximately a three-month period during which channel requirements can easily be 300 percent 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 due to limited competition. Today, SIP trunking offers a number of other provider choices, creating significant competition. This competition allows enterprises to obtain more favorable agreements, at lower prices, and also gives enterprises far more leverage with their incumbent carriers, forcing them to price services lower for customer retention purposes.

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 places the two providers in direct 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, due to the economy of a scalable solution. The more bandwidth you buy, the lower the cost per unit of bandwidth. Additionally, SIP trunking allows the aggregation of multiple connections into one or two connections.

SIP trunking can also provide savings to enterprises in the following areas:

- VoIP Gateways

  When an enterprise moves to an IP PBX, VoIP gateways are required for connecting to the legacy PSTN. With SIP trunking, the PSTN gateway is unnecessary. Besides avoiding the purchase cost of the gateway, the enterprise eliminates another device in the reliability chain that must be maintained and its software managed.

- Reduced Conferencing Costs

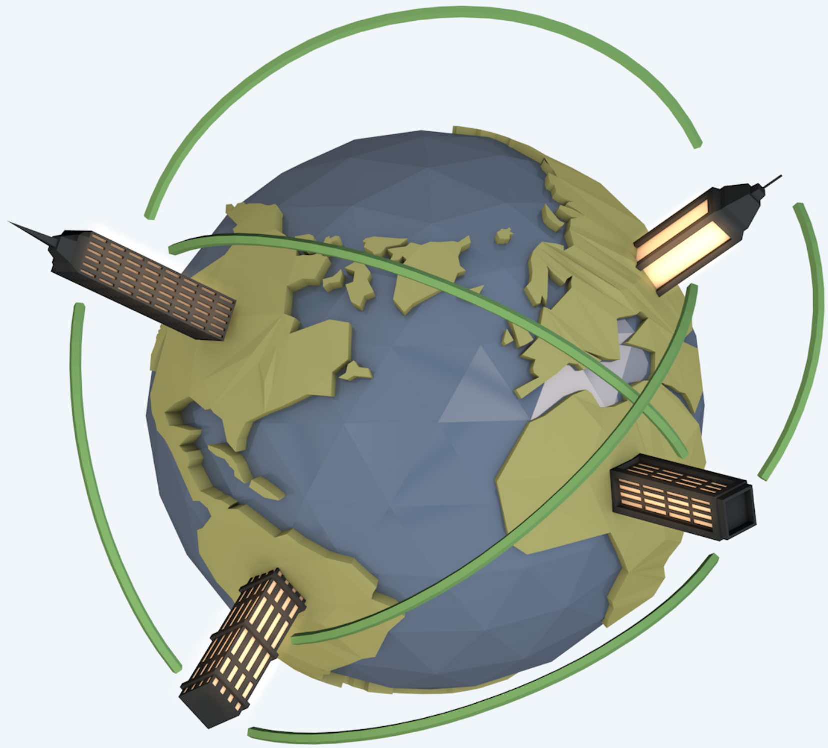
  Audio conferencing can be costly when using carrier legacy trunks. SIP trunk providers also support conferencing bridges and can eliminate the conferencing charges.

- 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 and Vonage. 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 percent (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, although this may not be available in all countries.



SIP Phones

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While SIP trunking may not be a cost-reduction solution for every organization, it will be for most. Investigation into SIP trunking is important– if only to eliminate it as an enterprise 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 SIP-like functions.

The introduction of SIP phones opens up the IP phone market to competition from third party IP phone vendors. Most IP PBX vendors offer connections to SIP IP phones, however there may be limitations of feature or function when third party SIP phones are deployed, so it is best to not assume that all SIP phones are equal.

More SIP Applications

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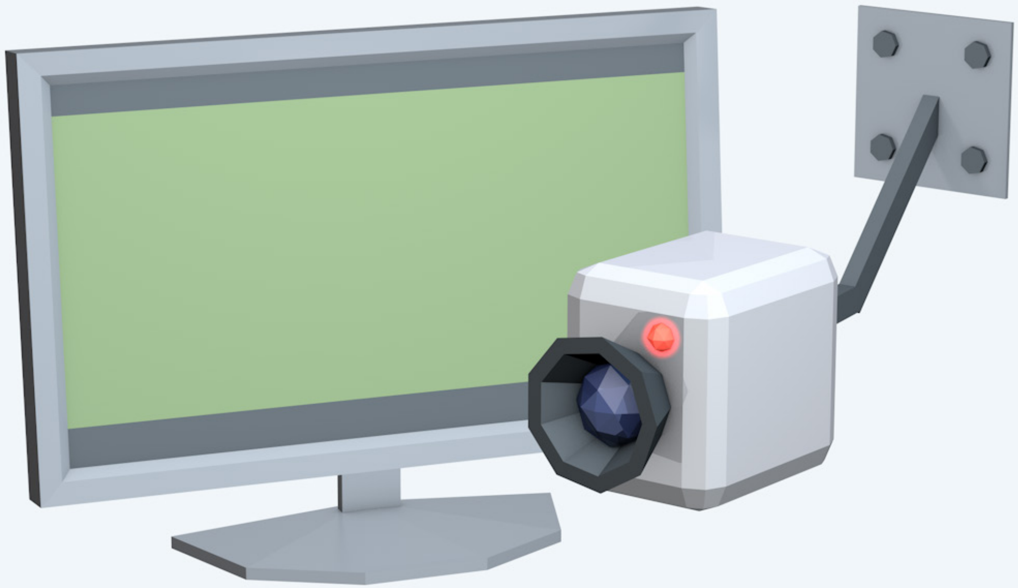
We continue to hear about SIP phones, gateways, Session Border Controllers, and firewalls. But did you know that there is a growing world of SIP devices now available that most enterprises have never heard of?

SIP can be used:

- As the signaling protocol for WebRTC
- For supporting video conferencing systems
- For connecting to cloud communications services
- For Computer Telephony Integration (CTI), connecting to servers with software from companies such as IBM® and Microsoft®
- With The Internet of Things (IoT) range of devices
- By some IP PBX vendors:
  - To communicate with VoIP gateways
  - For trunking between their IP PBXs

Other applications/uses of SIP include:

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



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Ensuring SIP Interoperability

In the early days of SIP trunk connections to providers, 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 and a headache for the IP PBX owner. Providers did not want to support many different versions, so this 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 SIP Forum Board has issued the SIPconnect 1.1 Technical Recommendation which is stable and has 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, so make sure you are working with the same version for all the vendors and providers.

> ***"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."***

Problems with SIP Trunking

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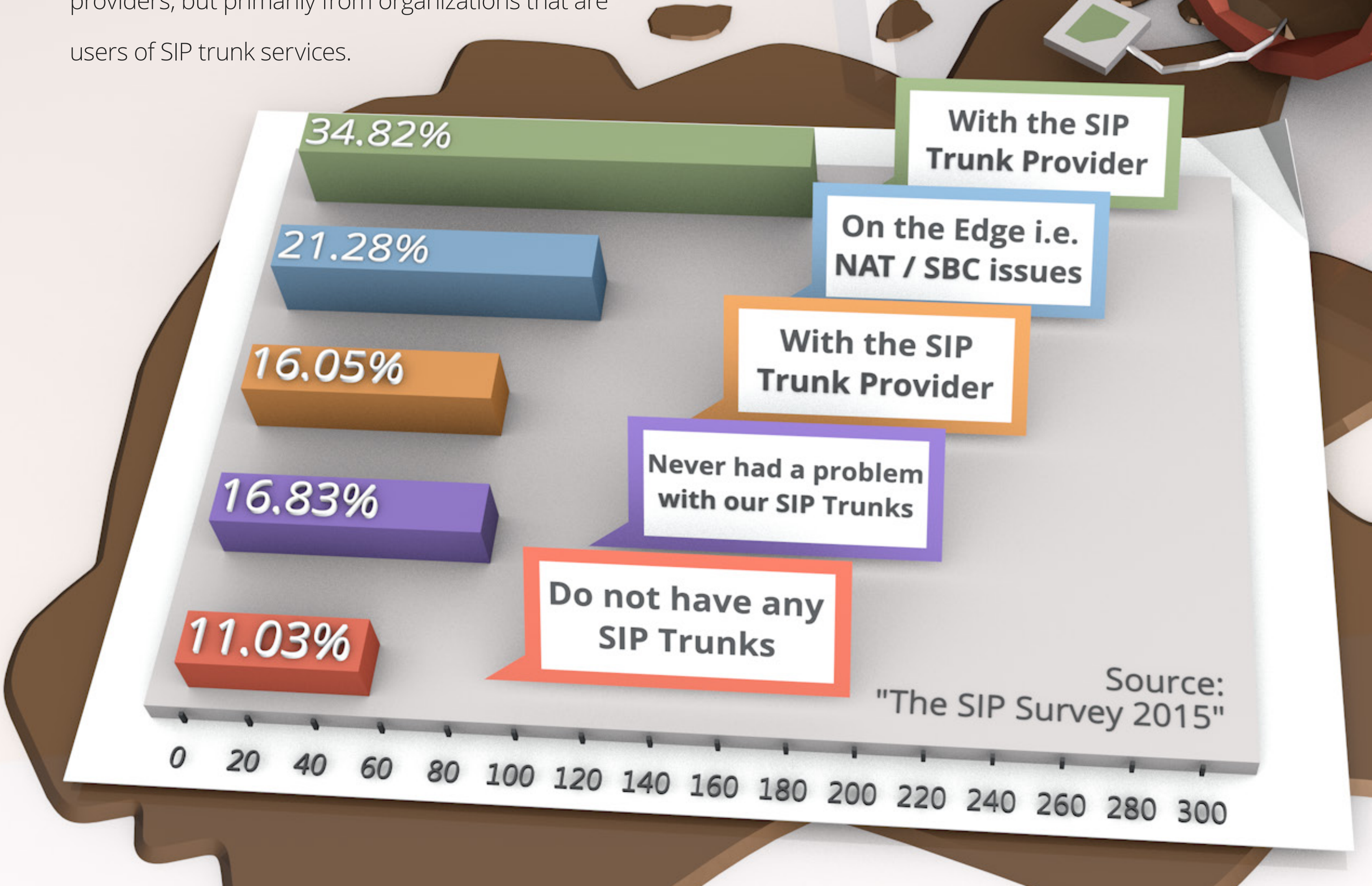
This is the fifth year of "The SIP Survey 2015". The response has been the greatest yet with 1098 professionals responding. This year-to-year increase in respondents helps strengthen the accuracy of the survey's results. A big change from the previous surveys is that the majority of results are not from ITSPs (Internet Telephony Service Providers) or 'simply' the providers, but primarily from organizations that are users of SIP trunk services.

One hopeful note is that, although they are in the minority, about 1/6 of the enterprises reported no problems at all.



An interesting result came from one survey question; it revealed the following reasons for not having problems:

- Good connectivity
- Good hardware configurations + Good gear (IP-PBX), Good Planning, Good support
- Good design - including failover support
- Certified configurations
- Interoperability matrix
- Smart people
- Excellent technicians
- Good support plus SIP knowledge
- Through testing
- All of the above, including good documentation by the manufacturer

SIP Trunk Provider Problems

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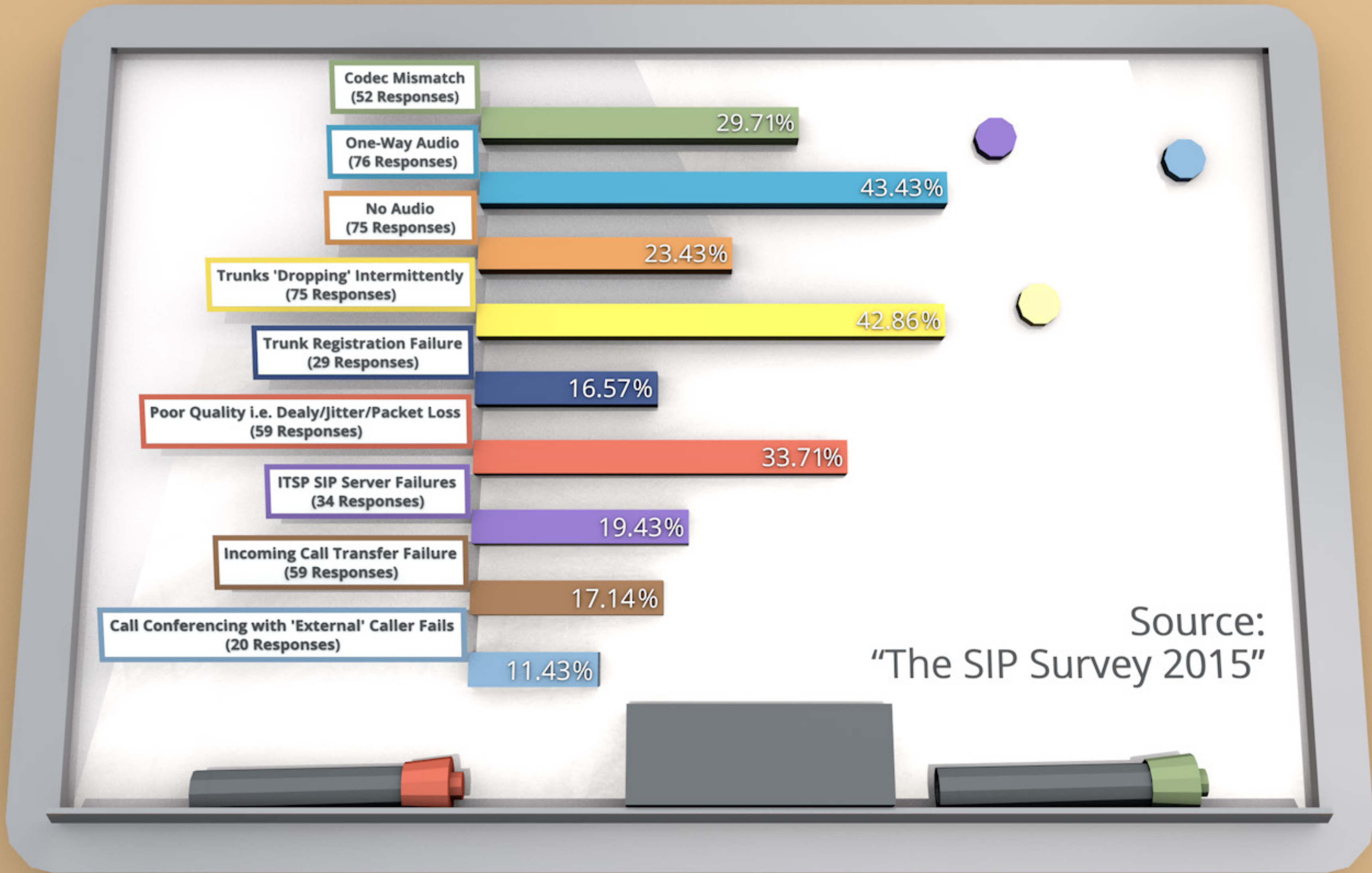
Previous surveys had shown that the percentage of problems were about the same for the providers and equipment vendors; SIP trunk providers, SBC/edge equipment vendors, and the IP PBX vendors had shared nearly an equal percentage of problems.

The 2015 survey found that apparently the equipment vendors have done a better job delivering successful SIP trunking implementation, with over a third of the problems attributed to the SIP trunk provider—and the IP PBX (16 percent) and SBC (21 percent) vendors fared better than in previous surveys.



There is a significant chance you will have one or more SIP trunk provider implementation problems. The graphic on the previous page shows the most common problems, however you can anticipate them by using this information to create a checklist to verify the provider implementation.

The worst problems are “One Way Audio”, “Trunks Dropping”, and “Poor Call Quality”, and it is with some frustration that we have to report the same industry issues, year after year. We start to wonder if the parties involved-- including the enterprises and VARs - will ever learn and adapt.

It appears that many of these problems are caused by incorrect configuration settings. ITSPs provide settings necessary for successful operation, but the enterprise and/or the VAR must do their part to ensure that the settings are correct; these include the IP address, the DNS address, port numbers to be used, RTP packet rate, codec to be used, as well as the NAT configuration.

One issue not covered in the survey was the traffic capacity of the SIP trunk. Even if all the individual tests are successful, you may still have an unexpected problem under “full load”. Since the trunk was ordered based on the number of simultaneous sessions it supports, here is the final, ultimate test: load up the SIP trunk with the maximum traffic to ensure it operates successfully under full load conditions.

The best IT departments will use this survey report to generate testing procedures and carry out troubleshooting measures prior to cutover.

In the 2015 survey results, Alan Percy of Dialogic and a well known blogger spoke to the importance of testing when he commented: “It seems that there is a need for a test suite that should be built into every SBC/PBX that does an initial test of the SIP trunk during the activation process. That would avoid exposing the end-customer to many of these issues.”

Since all the issues in the “SIP Trunk Provider Problems” chart have been around since the beginning of SIP trunk implementation, it is disappointing that the industry has still not resolved the problem frequency. These very basic implementation issues should be eliminated during the SIP trunks initialization process.

It begs the question of whether the implementation teams are overworked, undertrained, or both—or perhaps the providers have not made these problems a high priority on their list of improvements? The high cost of issue resolution to the provider could be avoided by further investment in the implementation team--together with the necessary testing when SIP trunks are installed.

However, before we condemn the SIP trunk provider, we should consider that there are other possible contributors. If you are depending on a VAR/reseller to manage the implementation, then that party may be the culprit, not the provider. Also, if your IT staff attempts the problem resolution, be aware that their configuration tweaking may produce some new problems.

Edge Device Problems

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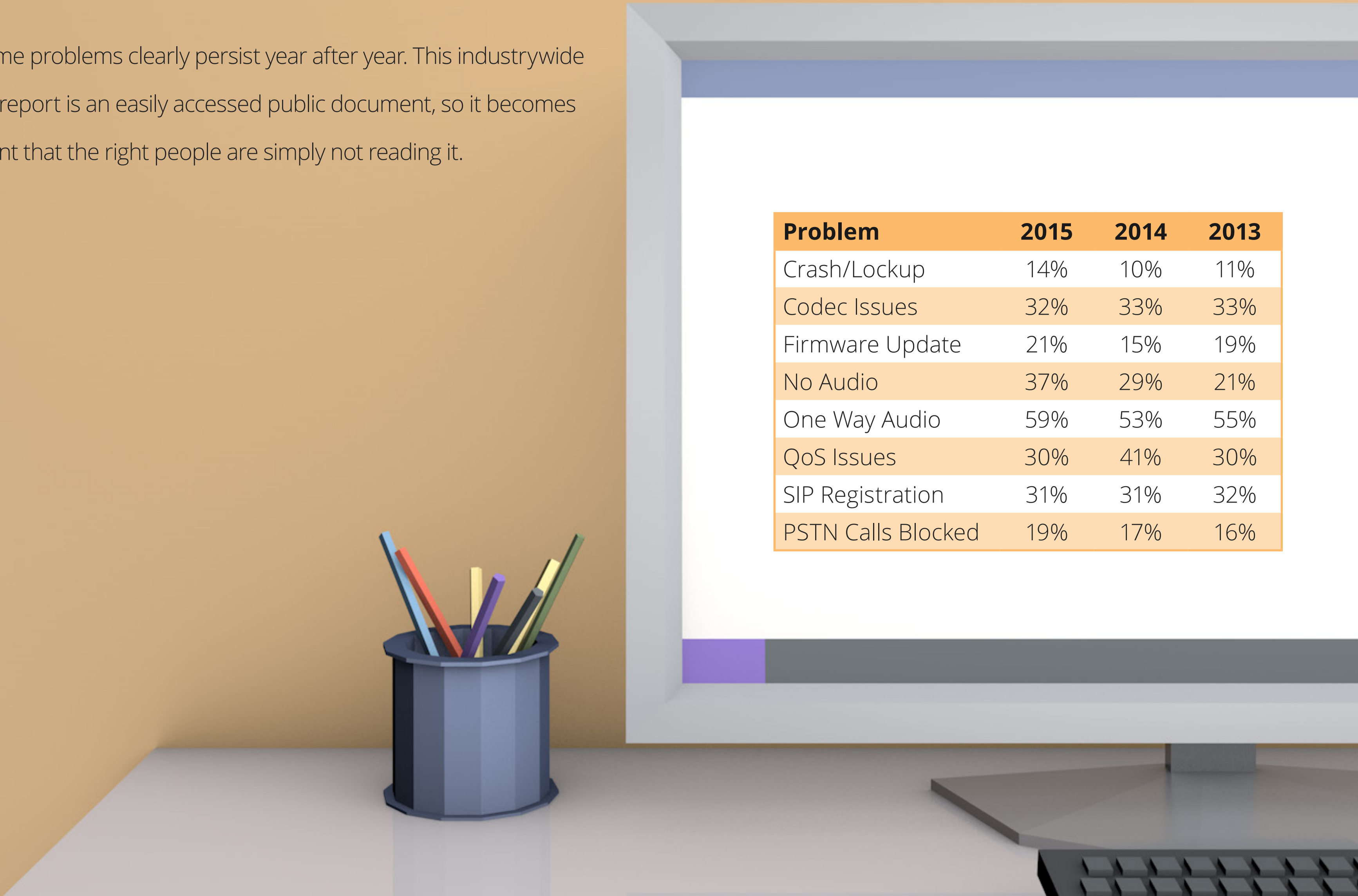
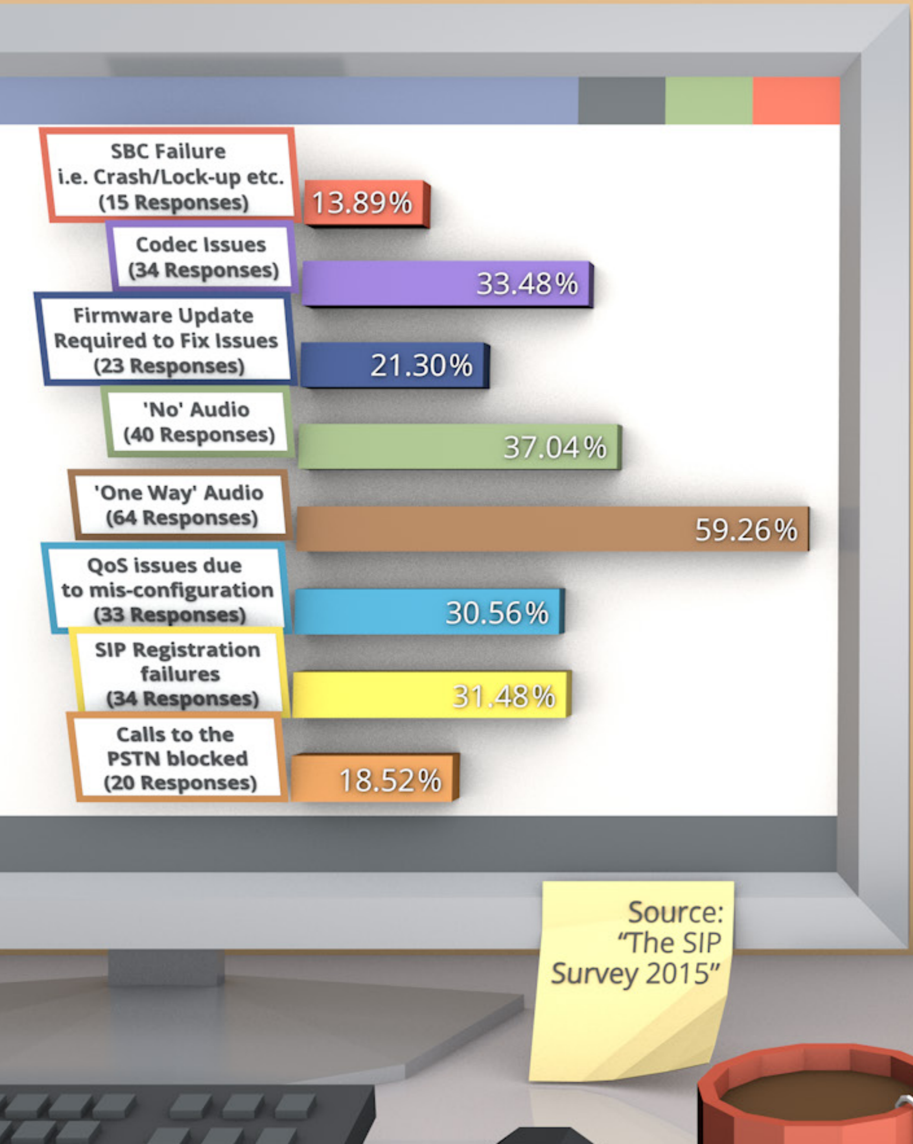
The SBC/NAT equipment problems have not decreased. “No Audio” and “Firmware Update” have actuallty increased as problems over the last three years when compared to the 2014 and 2013 survey data. The table compares the percentage of problems reported.

The same problems clearly persist year after year. This industrywide survey report is an easily accessed public document, so it becomes apparent that the right people are simply not reading it.



Problem	2015	2014	2013
Crash/Lockup	14%	10%	11%
Codec Issues	32%	33%	33%
Firmware Update	21%	15%	19%
No Audio	37%	29%	21%
One Way Audio	59%	53%	55%
QoS Issues	30%	41%	30%
SIP Registration	31%	31%	32%
PSTN Calls Blocked	19%	17%	16%

As evident in this Equipment Problems graph, “One Way” audio remains the dominant issue in 2015. Installing an SBC will resolve this and many other issues, with misconfiguration being the most likely source of the problems. The same is true for the “No Audio” issue.



“Codec issues” are still high on the problem list. The trained professionals configuring this type of specialized equipment are expected to have a solid understanding of codecs. Since codecs can be easily configured correctly and tested thoroughly, it is most likely a configuration issue due to either ignorance or negligence.

These 2015 “SBC Failures” statistics are frustrating, because the loss of the SBC is the loss of SIP trunks. Unless the SBC configuration is fault tolerant, it can be a single point of failure. Test thoroughly, and work with the vendor to determine if there are known reliability issues with their offered products. Additionally, unless testing of the SBC is ongoing for a year or more, you cannot calculate the reliability of the SBC, and will have to depend on information from the vendor.

IP PBX Problems

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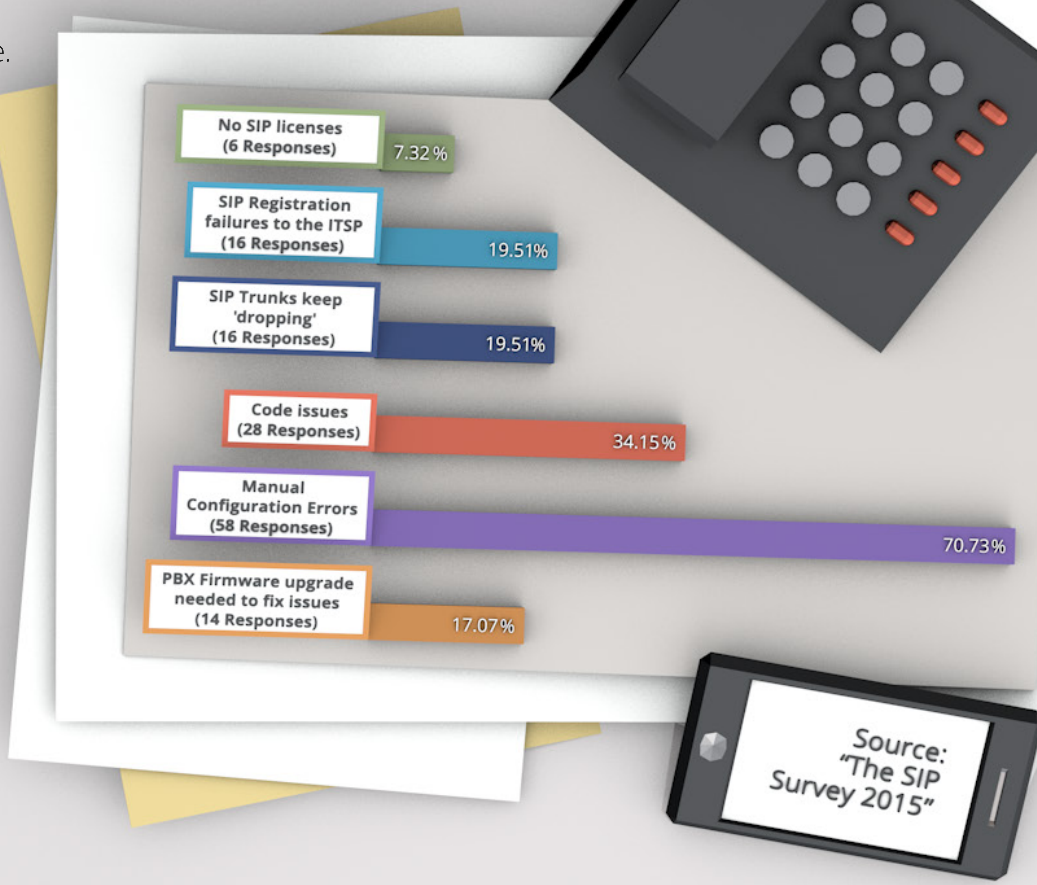
It's gratifying to see the IP PBX is less of a problem source.

Currently, "Manual Configuration Errors" still dominate the problem list. These errors caused 60 percent of the problems in the 2014 survey, and only dropped slightly to 58 percent in 2015. The continued high-level of configuration problems could be due to ignorance, negligence, impatience, and/or poor training, but, whatever the initial cause, the errors can still be reduced by proper testing after the configurations are set.

"Codec Issues" are preventable. Inspect the settings to avoid this problem. (Could we have an automated validation check to match codecs?)

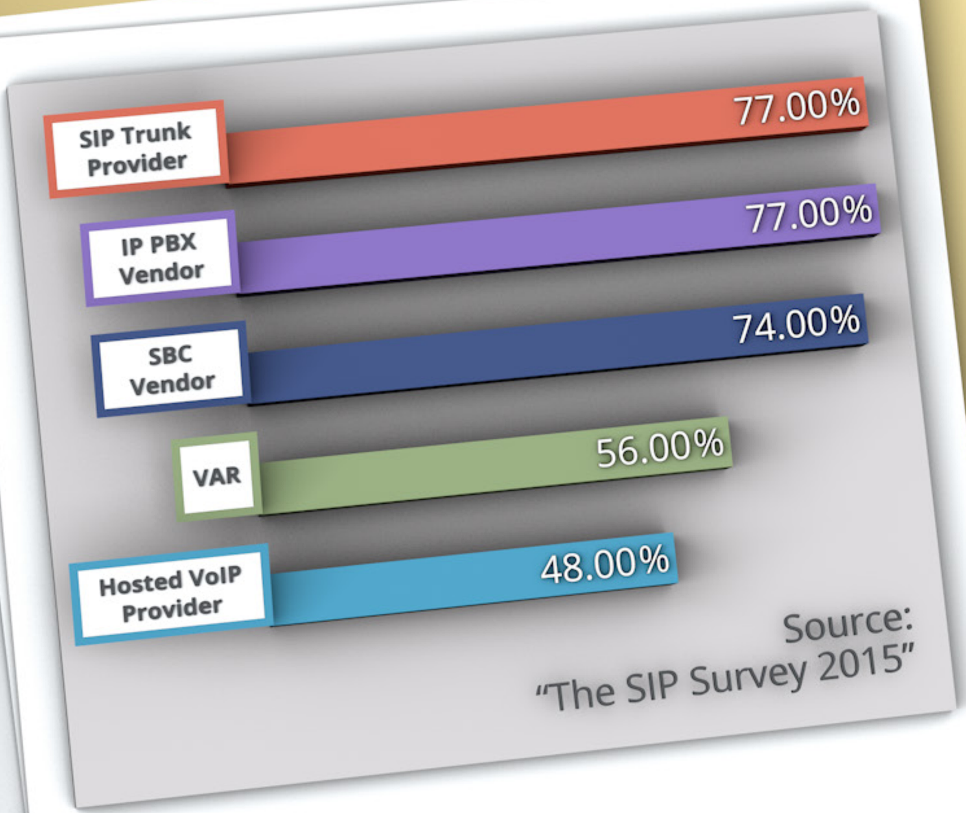
"Registration Failures" and "SIP Trunks Dropping" are attributable to misconfiguration or poor documentation.



"No Licenses"-- This should not happen, and is ridiculous; enterprises should certainly know what they are paying for.

13 Do You Trust the VAR?

The VAR is viewed as the weakest participant in SIP trunk implementation. The "2015 SIP Survey" reported who is the best for solving problems, with results showing a distinct lack of confidence in the VAR and Hosted VoIP provider.



Preventing the Problems

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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 accept responsibility for 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—and make no assumptions; remember, the enterprise will be responsible for assumptions, not the vendors or providers.

Your vendors and providers probably have checklists of what they will perform during the implementation. Obtain these checklists from them, and verify they have used them, to validate the installations. In the event that there are no checklists, create your own by communicating with other enterprises that have implemented SIP trunks. Additionally, using the "SIP Survey 2015" to anticipate issues that may occur is smart; knowing what has gone wrong with other implementations essentially highlights the issues to be tested.

Some technicians will keep tweaking the configurations and settings until it does work, which can be dangerous as the tweaking can turn off features, change security, or produce a liability that will show up later. If tweaking is performed, ensure that the changes are adequately and correctly documented, providing an audit trail if future problems occur. (If not documented, the tweaking starts all over again and the enterprise will continue to encounter outages or poor operation.) For example, during tweaking: Transport Layer Security (TLS) or Secure Real Time Protocol (SRTP) may be turned off (eliminating critical security features), improper codec changes may occur, and (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 being configured improperly, making it more difficult to pinpoint the culprit. If the corrections performed by the SIP trunk provider do not work, then enlist the aid of the SBC and IP PBX vendors before you declare the SIP provider as incompetent. You may find that most of the problems will appear to be configuration mistakes, and that the provider's implementers do not have enough training, experience, and adequate documentation to perform the troubleshooting.

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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 obviously not yet mature.

These eight (8) 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, remember you are responsible for them, not the vendors and providers.
3. Coordinate with the vendor(s) and provider(s). Have them all meet together with the enterprise staff tasked with the implementation.
4. The problems discussed in the "SIP Survey 2015" are the most common challenges that will be encountered, so 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, even blockage of 911 calls. 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. Budget some time for unanticipated issues.

Determining the SIP Trunk Capacity

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Once you have decided to deploy SIP trunks, you will need to determine the necessary SIP trunk capacity. The greater the capacity implemented, the higher the cost. The number of SIP sessions implemented will define the SIP license fees associated with the SIP trunk. The following information and tutorial will assist you with areas such as VoIP bandwidth calculations, needs assessment, sizing the SIP trunk, and Erlang B calculation.

VoIP Bandwidth Calculations

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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 Quality of Service (QoS). (QoS is not discussed in this paper.)

How much bandwidth to allocate actually depends on:

- 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

VoIP Packet	Packet Size	Over Ethernet	Over PPP w/RTP	Over PPP w/cRTP
IPv4 w/G.711 & G.722 at 64kbps	20 ms/160 bytes/packet	87 kbps	82 kbps	68 kbps
IPv6 w/G.711 & G.722 at 64kbps	20 ms/160 bytes/packet	95 kbps	90 kbps	68 kbps
IPv6 + Header Extension w/G.711 & G.722 at 64kbps	20 ms/160 bytes/packet	98 kbps	94 kbps	69 kbps
IPv4 w/G.729 at 8kbps	20 ms/20 bytes/packet	31 kbps	26 kbps	12 kbps
IPv6 w/G.729 at 8kbps	20 ms/20 bytes/packet	39 kbps	34 kbps	12 kbps
IPv6 + Header Extension w/G.711 & G.722 at 64kbps	20 ms/20 bytes/packet	42 kbps	38 kbps	13 kbps

The results for the G.711 and G.729 codec calculations operating over IPv4 and IPv6 are here in the "Minimum Bandwidth Requirements" table.

Minimum Bandwidth Requirements table illustrates three points:

- Bandwidth requirements reduce with voice 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. However, most designs produce small packets. 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) and MPLS is employed.

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/packet 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 percent (70%) of silent time. It is recommended that the designer assume that silence suppression only reduces the bandwidth requirement by 30 Percent (30%). Silence suppression should only be applied on trunk groups with greater than 24 simultaneous SIP sessions (voice paths).

Before enabling silence suppression, 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 will 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. The delivery of compressed RTP may not be supported at the enterprise SIP trunk interface (PBX or IP PBX).

A Session Border Controller (SBC) is 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

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The proliferation of SIP trunking makes the bandwidth calculations for those SIP trunks an issue that the IT organization must resolve. If too much capacity is assigned, then money is wasted. If too little capacity, then 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 call 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 call paths needed which can be fewer than previously implemented. Investigate how the voice path calculations were performed for the T1/PRI trunk connections. 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, help desk, or any other location designed to receive calls, whether voice, Fax, or Interactive Voice Response (IVR) calls.

Erlang Calculations

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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 percent (10%) to 70 percent (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 call paths 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 there are not enough lines/call paths available.

- Lines:

  The number of lines/call paths is in a trunk group. One line/call path 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 percent busy) which means that 99 percent (99%) of the calls are answered and do not receive a busy signal. A GoS of .001 means that 99.9 percent (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 percent (99.9%) or better.

Next, the traffic load must be either measured or estimated in Erlangs/BHT. One Erlang is equivalent to one line/call path busy for one hour 100% of the time. To calculate the Erlang load, the SIP 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

NOTE: A sample calculator with an example can be found at Ansapoint.

SIP trunking is financially attractive. SIP trunk providers also offer other features like least cost routing and bundled national and international services. SIP trunking cannot be ignored, but it is important to size your SIP trunking to support the calling services that your enterprise requires, and that your customers and other users have become accustomed to with the legacy systems.

Resources for More Information

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The SIP School (www.thesipschool.com) has developed a SIP online training and certification program (http://nec.thesipschool.com).

Additional articles available (www.nojitter.com):

- “Peeling Back the SIP Resiliency Layers”
- “Slideshow: How to Acquire SIP Trunks”
- “Implementing SIP Trunking: Best Practice #3”
- “SIP Trunking: Come for the Cost Savings, Stay for More Cost Savings”
- “SIP Trunking -- Victories and Nightmares”
- “How to Size Your SIP Trunks”

About the Author

Gary Audin, President of Delphi, Inc. delphi-inc@att.net has more than 40 years of computer, communications, and security consulting and implementation experience, and 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.