# The SIP School
## Learn and Qualify



**Wednesday, November 04, 2015**

# THE SIP SURVEY 2015

## Survey and Thought Leadership:

*Businesses around the world are migrating to SIP trunking. Most experiences are good but some are not. Let's find out what's happening!*

**Editorial and Research**
Graham Francis
CEO The SIP School

**Introduction**

This fifth year of the SIP Survey has proven to be even more popular with **1098** professionals responding.  This continues a year on year increase that really does help strengthen the accuracy of the survey's results.

One BIG change from all the previous surveys to note is that the majority of results are not from ITSPs (Internet Telephony Service Providers or 'simply' the providers) but in the main from companies that are users of SIP trunking services.  We have done this to really focus on what the customer is experiencing because when all is said and done, it is they who will decide if a service is successful or will fail. One drawback of this new approach is that it makes it harder to compare results to previous years but we think that a re-focus is needed.

To highlight the questions we've used two colors

**Purple** = Questions answered by everyone

**Green** = Questions answered by non-ITSPs (the Clients)

**Why this kind of Survey?**

The survey's purpose is to take a look at SIP trunking and specifically, to find what the most common  issues during SIP trunk deployment are and what can be done to make these issues occur less frequently, if at all.

Understanding these issues will help companies focus their efforts on improving the 'failing' elements and also ensure that its staff members understand what to do when things go wrong so that they are able to fix problems quickly.  It's not ideal having a service that's feature packed if you can't count on it to not fail when least expected.

Let's be clear, SIP trunking is one of the fastest moving elements of VoIP in the world of Telecommunications and not without good reason.  It offers benefits ranging from low cost calling, centralization of lines into a business, fast disaster recovery (or failover) and much more.

However, as vendors, service providers and enterprise customers are finding out, SIP trunking is not always an easy service to implement and sometimes not easy to support if things go wrong.

As the survey has been carried out by The SIP School, all of the opinions in this report are our own unless clearly stated. We have been able to embellish this report with comments from people who work in this area and believe that their insights can help people understand what is important and actually happening out in the real world.

The first question of the survey has been changed from previous years to help us focus even more on the clients who are using SIP trunks and not so much the companies providing the service.

## Q1: SIP trunks are becoming more and more popular, where do you and your organization fit?
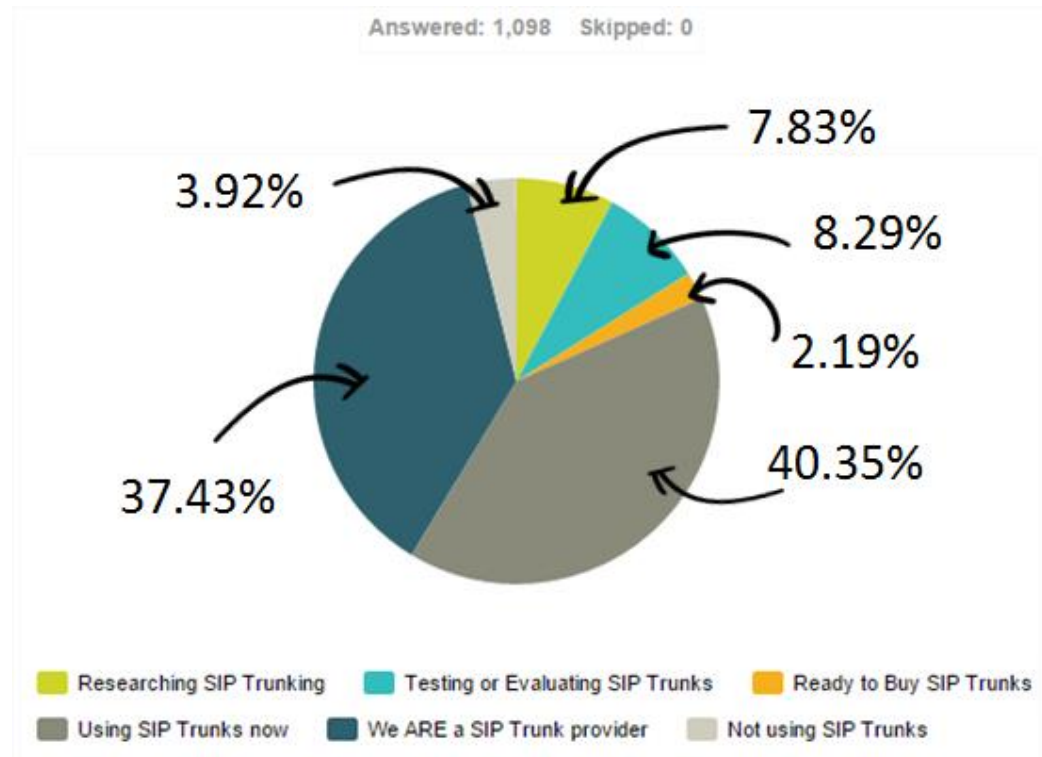


The results here show that almost 38% of responders (to this Survey) are actual ITSPs which means they were subsequently not asked questions from Q2 to Q16. People who told us they are 'not using' SIP trunks were directed to Question 18 which does leave a large percentage of people that are using, testing, researching SIP trunks and whose comments should be taken seriously by the providers and dealers/resellers who are reviewing this document.

Here's the breakdown of '*all*' respondent locations. When we looked at the client locations 'only', the results were so closely aligned we decided just to show these.

- 52.21%    **USA**
- 8.50%    **UK**
- 7.14%    **India**
- 6.63%    **Canada**
- 2.04%    **Germany**
- 23.47%    **Other** countries

Now it's important to know something about the companies that the respondents work for.

## Q2: How many employees are at your company?

Answered: 588    Skipped: 510
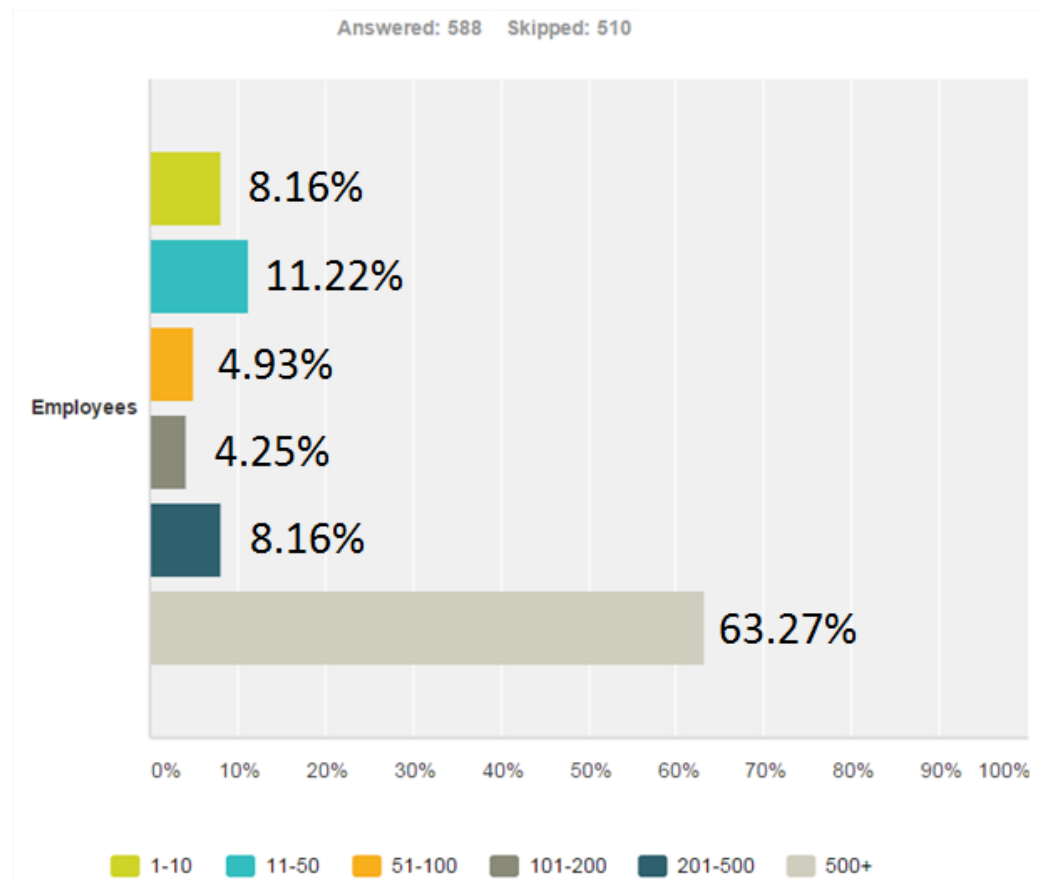


"It is interesting that the biggest set of responses comes from companies of 500 employees or greater. In their case have they adopted SIP trunking across the board or only selectively? The cost benefits would seem much bigger for smaller firms than larger ones."

*Steve Johnson, Ingate*

So, as we are looking at these numbers with fresh eyes this year it's interesting to see that the majority of client responses are from large enterprises.  This doesn't take away from the fact that companies of all sizes are benefitting from SIP trunks.

Note: 510 respondents 'skipped' this question.  This actually means that as they answered that they were a provider (in Question 1) the 'logic function' of the survey system skipped this question (and others) for them.
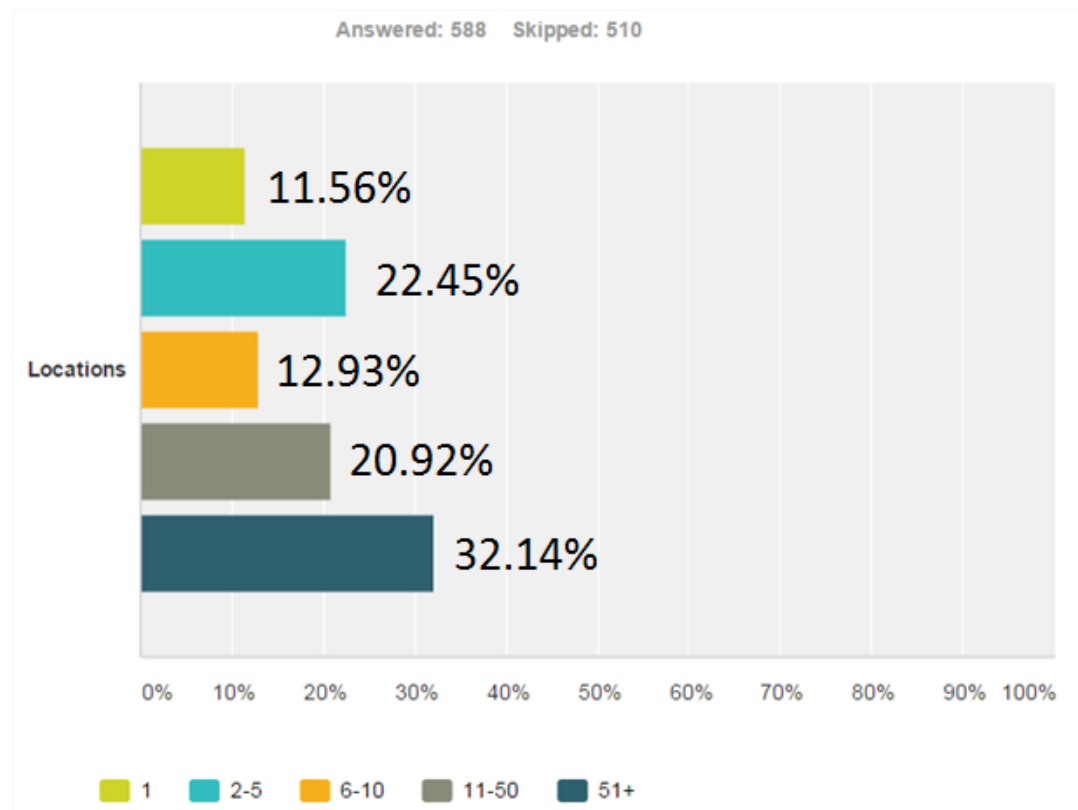
## Q3: How many locations or sites does your company have?

Answered: 588    Skipped: 510

**Locations**

- 11.56%
- 22.45%
- 12.93%
- 20.92%
- 32.14%

0%  10%  20%  30%  40%  50%  60%  70%  80%  90%  100%

Legend: 1 · 2-5 · 6-10 · 11-50 · 51+

Cost savings can be substantial for a well implemented SIP trunking infrastructure so it's great to see organizations with many locations taking advantage of all that SIP trunking can bring

And remember, that with SIP trunking you do not have to buy more lines than you need. You want 130 trunks; you can have 130 trunks – exactly!

Let's move on and get some information about what equipment people are using so we'll start with the PBX. If fact, since last year we've added more options to reflect how companies may be using multiple systems and possibly even transitioning to the cloud.

## Q4: Does your company have for its own 'Internal' use?

Answered: 580    Skipped: 518

An IP based PBX? — **44.14%**
A TDM based PBX? — **1.55%**
Voice service 'In the cloud'? — **5.52%**
A 'mixture' of above? — **45.34%**
None of the above? — **3.45%**

Responses

0    100    200    300    400    500

"The results kind of surprised me – the Voice service in the cloud seemed like a lower number than the customers that I know of that have abandoned their old PBX or Key systems and moved to the cloud or the consumption model type solutions. The mixed environment did not surprise me especially for large corporations with numerous locations; very hard to migrate every location throughout the world at the same time."
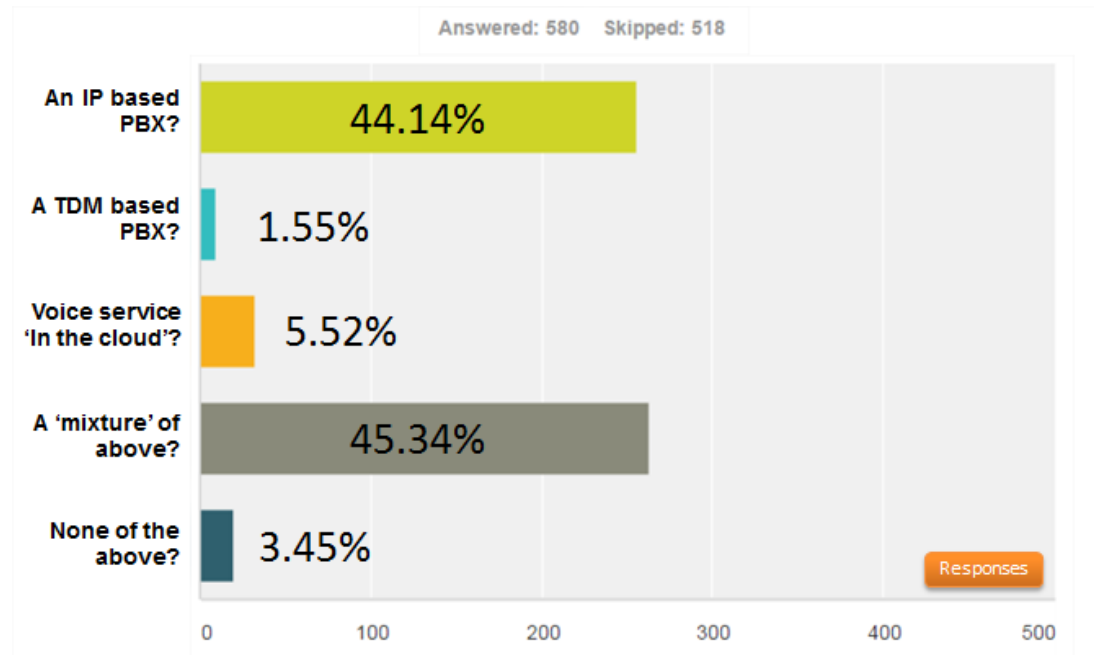
*Ron Zobell, Verizon*

"The Cloud seems to be getting such hype that I'm surprised that only a small number of respondents are using it as the primary provider of SIP trunks."

*Steve Johnson, Ingate*

89.48% for IP PBX implementations (including the 'mixture' responses) indicates that companies have come a long way down the path in the migration from TDM only based systems. Of course a mixture *could* mean that companies still have an old TDM system that's taking time to de-commission or that they have a fully-fledged IP based PBX but using the cloud for other services such as 'cloud based' Call Center services, Call Recording etc.

Ron makes a good point in that it would take time to migrate multiple locations around the world to cloud services with the migration possibly being stymied by any 'country specific' rules and/or regulations.

Most of the 'None of the above' responses (where people can add in their own option via the survey form) should have actually been answered (in our opinion) by selecting 'A mixture of above' and thus that figure should have been higher.

Our next question relates to the manufacturer of any installed PBX system.

## Q5: Who is the manufacturer?

Answered: 481    Skipped: 617

| Manufacturer | Percentage |
|---|---|
| Alcatel-Lucent | 5.82% |
| Asterisk | 9.15% |
| Avaya | 47.19% |
| Broadsoft | 3.95% |
| Cisco | 30.77% |
| Genband | 3.33% |
| Microsoft (Skype for… | 11.02% |
| Mitel (inc. Aastra) | 18.09% |
| NEC | 4.57% |
| ShoreTel | 5.20% |
| UNIFY (inc. Siemens… | 2.29% |
| I prefer not to say | 5.61% |

Multiple selections were allowed on this question

Responses
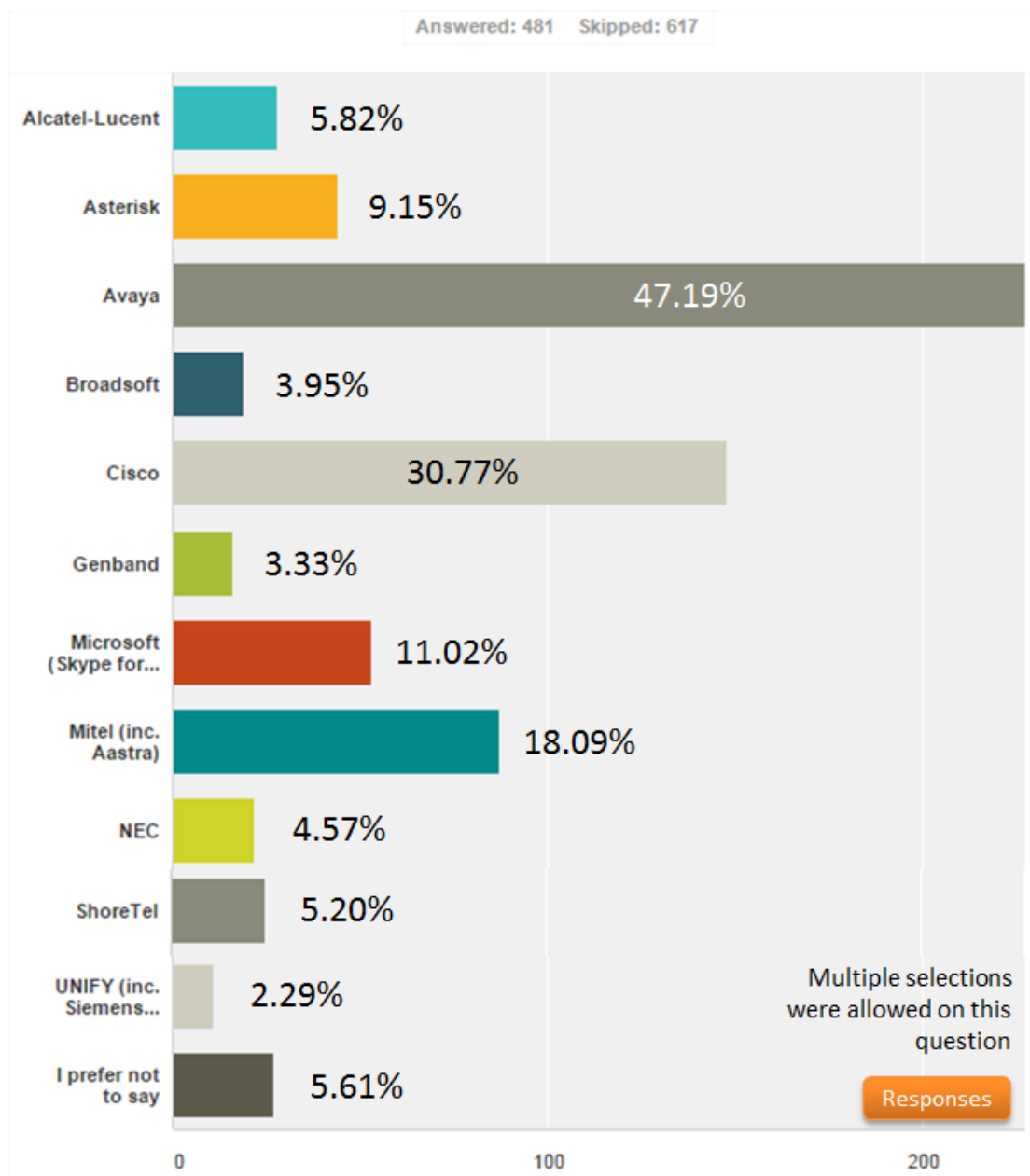
Any options that were made available to Survey responders that dropped under the 1% mark we made the decision not to display them.

Interesting that even in previous surveys where it's been 1: Avaya, 2: Cisco, 3: Mitel that it's the same when we're focusing on the client only. I *can* say that compared to previous years (with ITSP responses included), Avaya has fared even better and also Microsoft's presence has increased.

As we have previously asked if a company is using a cloud based service we thought we'd better ask.

## Q6: If you use a 'Hosted/Cloud' provider for your 'Primary' Voice service, who are they? If your prime company location is not in the USA then please use the 'other' option to tell us.

"It will be good to find out how many are using cloud communication providers for other than primary voice service (such as Voice mails, Call Center, Multi-site SIP trunk consolidation, WebRTC, etc.)"

*Ashish Jain*
*Genband*
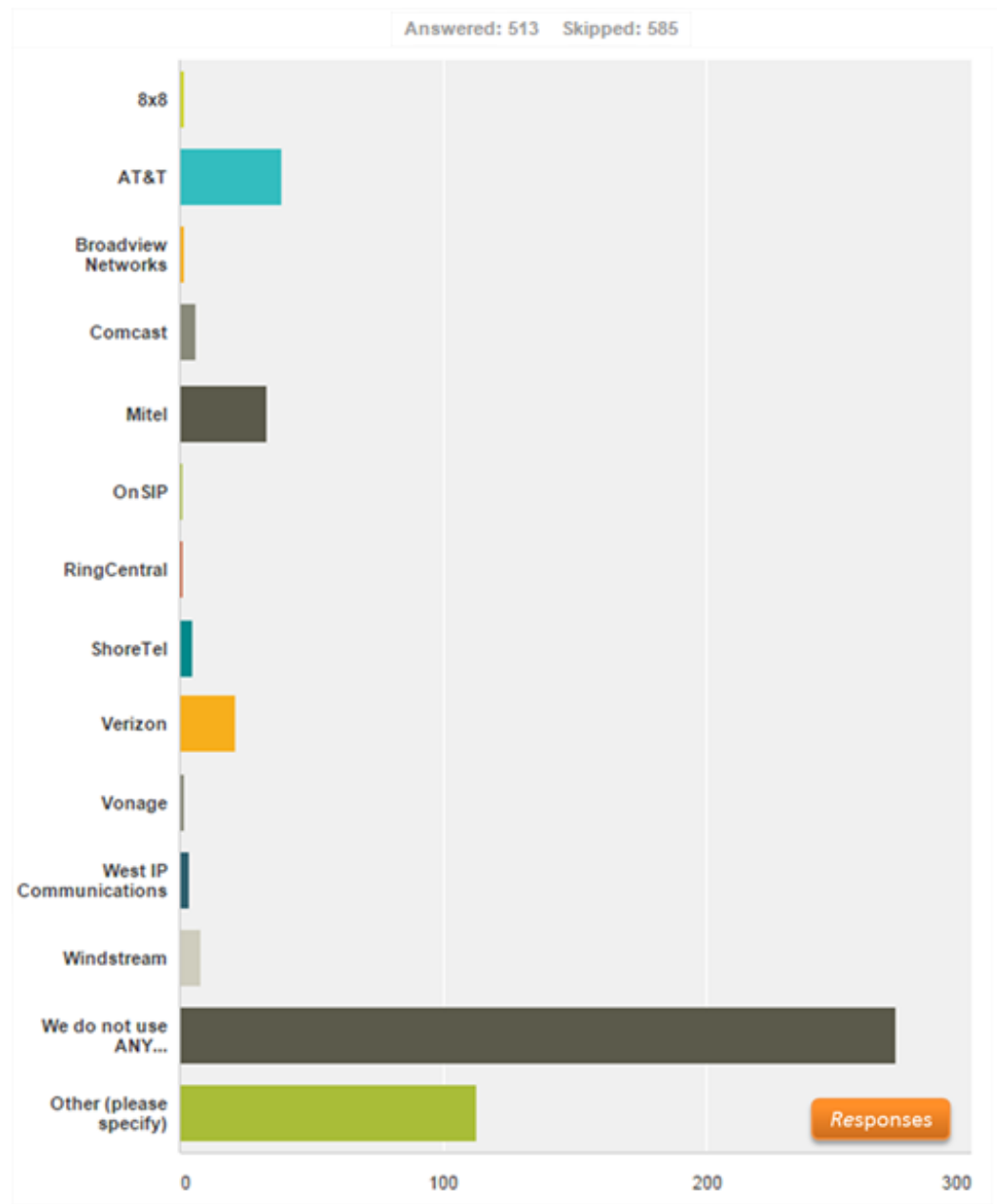


A resounding response for not using anything shows that a lot of people, even though they may be thinking of utilizing the cloud, may not have started that transition. Remember, that a lot of time, effort and money has been invested in

Enterprise systems that simply cannot be discarded and migrations are done (sometimes) at glacial speeds.

The responses in 'Other' of course showed a few other providers but a lot of people mentioned that they used *multiple* hosting providers!

We focused on the US for this question (as there are far too many global providers to list) and you can see that AT&T with Mitel and Verizon did the best. Of course there are a lot of companies offering these services and it will be interesting to see how they can transition traditional 'PBX purchasing' clients away from that model to new cloud based services. Also, with so many players here will there be a lot of consolidation as they (may) find that they cannot gain enough market share to compete?

Here is a word cloud with all the responses from the 'Other' section. The larger the text, the more people that are using their service.

"It is very surprising and concerning that so many respondents don't use an SBC. The Session Border Controller is truly a work-horse element of a good SIP trunking environment. Not only does it provide NAT traversal functionality, allowing the PBX to be located on the LAN with a private IP address, it also provides signification routing capabilities, normalization of SIP signaling between the PBX and the service provider, and not least security. This should be a key element of any SIP implementation and a best practice that should be made available from every PBX manufacturer, reseller and service provider."
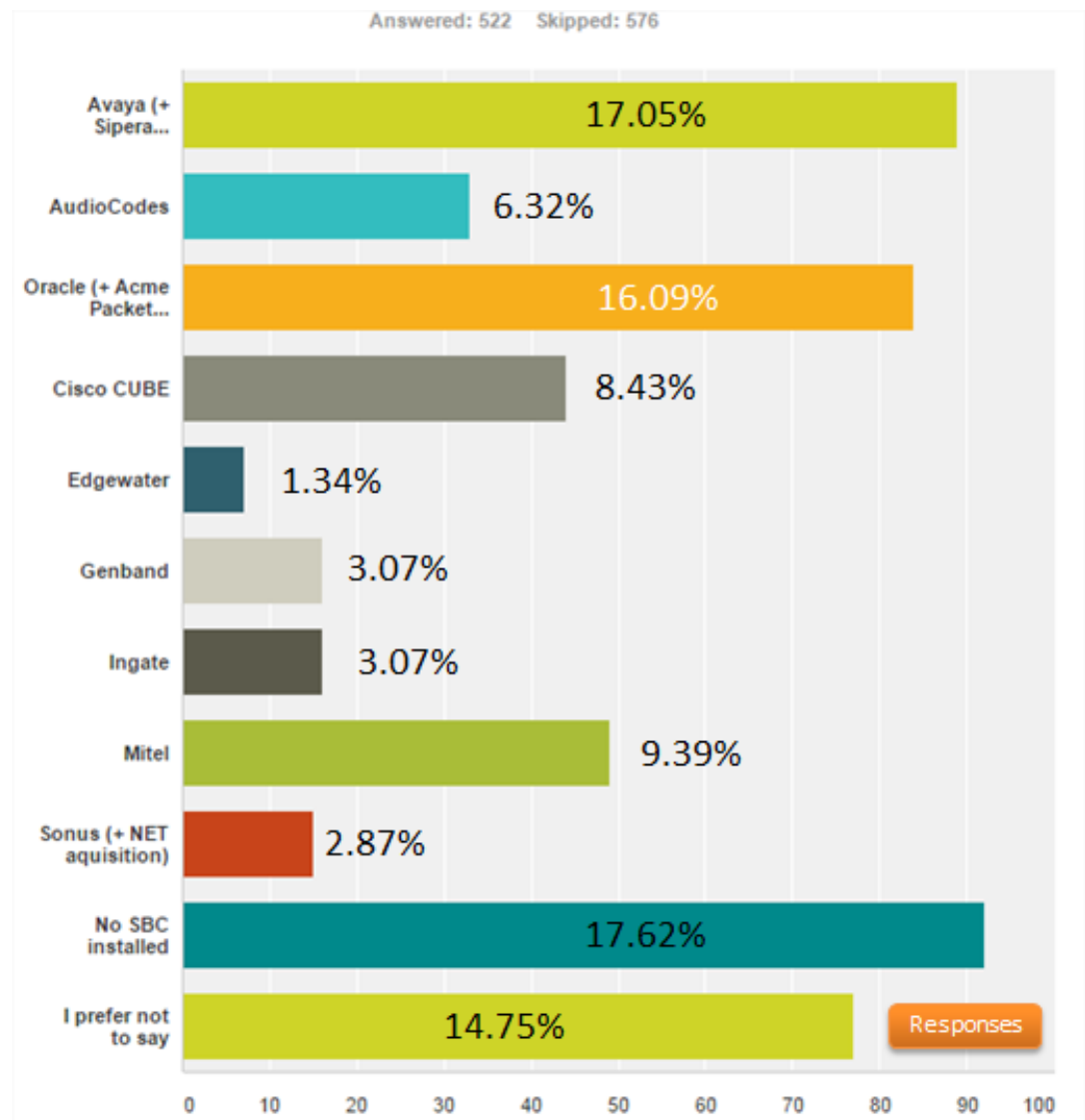
*Steve, Johnson, Ingate*

"Having no SBC is likely ok for cloud/hosted if the clients are line side only. Otherwise it speaks to a gamblers mentality."

*David Chavez, Avaya*

Next we asked about the Session Border Controller (if any) people were using.

## Q7: If you use a Session Border Controller on the Edge of your network, who is the manufacturer?



So we have Avaya at the top closely followed by Oracle with the (quite unnerving) NO SBC in the third spot. Preferring not to say is ok as why give clues on your security setup?

Another new question for this year as we are a curious bunch here….

## Q8: Did you get your SBC....
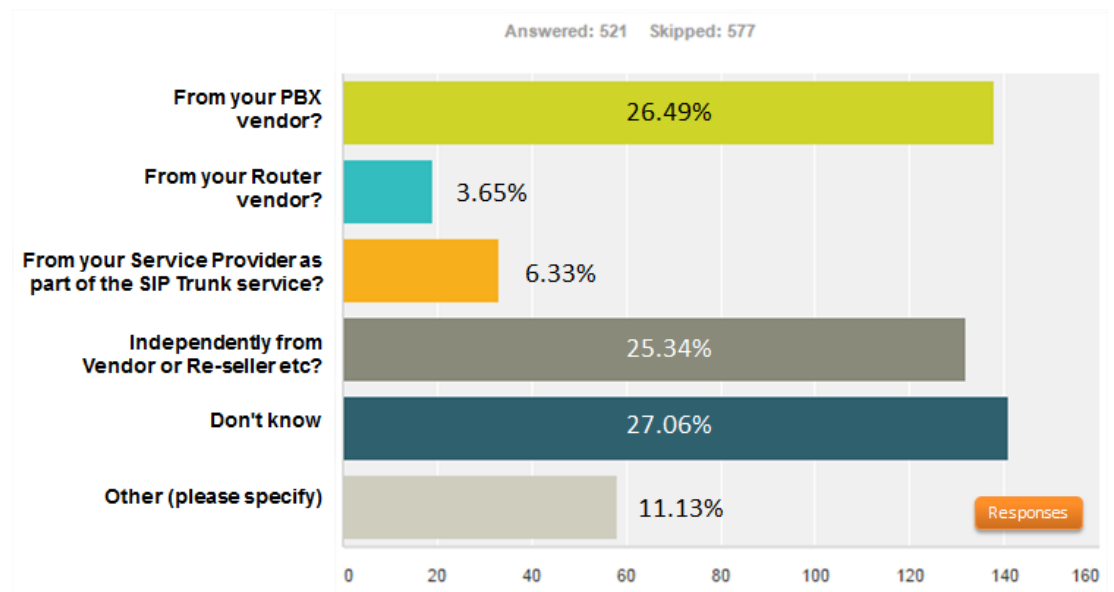
Of course not all respondents would be aware of how things are acquired / purchased but we wanted to try to see how SBCs were obtained by clients.

**From the PBX vendor**, is this because it was part of a 'package'? Or does it offer 'extensions' to the PBX that other SBCs cannot due to a 'Vendor specific' issue or feature?

**From the Service provider** is a good option as (hopefully) it will be optimized for that particular provider, maybe even remotely managed.  It should be noted that some providers can now offer SBC services 'in the cloud' as well.

The '**Independently**' response shows that there is a healthy market for 3$^{rd}$ party vendors to offer extra a 'fuller' range of features for clients such as SIP normalization, QoS control, Security and more…

Again, a lot of people may not even know where their device came from and this is supported by a comment from Kevin Pitts (Oracle Communications),

He said: *"Often the team that administers the gear does not know who sold it to them, but that is how it should be….delivered well and working well for the group that supports it and the end user customers."*
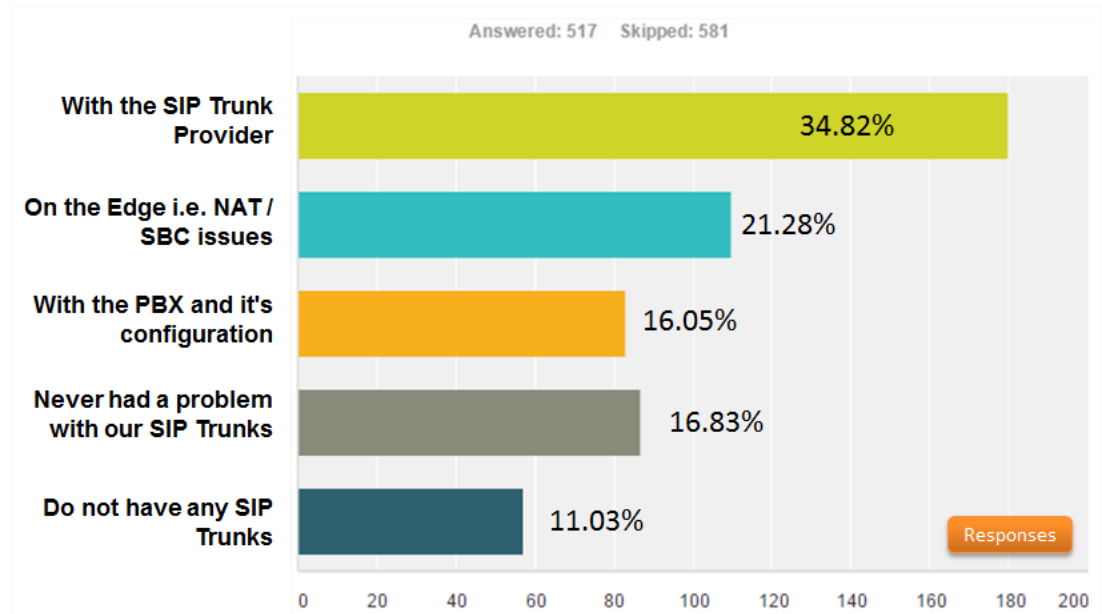
S

Now, if SIP trunks are installed and all works fine, then that's great and your business is reaping all the rewards promised. But what if things go wrong?

## Q9: If you've had problems, where have the issues been?



Now we can see what happens when we just focus on the client's responses to the Survey. For the last number of years the 'blame' for issues has been spread evenly across the SP, PBX vendor and SBC vendors.

**'Never had a problem....'** is a new option for selection this year and good to see that people have had no issues. One of the questions in the survey asked was what people thought were the reasons behind their error free experience, here are the most common and ALL of them should be familiar with you during a SIP trunk installation – if not, why not?

- Good connectivity
- Good hardware configs + Good gear (IP-PBX) Good Planning, Good support
- Good design - including failover support
- Certified Configurations
- Interoperability matrix
- Smart people
- Excellent Techs
- Good support plus SIP knowledge
- Testing
- All of the above and good documentation by the manufacturer

We did particularly like a comment 'Good Luck' … wondered what happened there?

Now, let's look at the three elements that may be having the issues in isolation, starting with SIP trunks.

## Q10: If you've had problems that were found to be on the SIP trunk provider side, what were they?

Answered: 175   Skipped: 923

| | Percentage | Responses |
|---|---|---|
| Codec Mismatch | 29.71% | 52 |
| One way Audio | 43.43% | 76 |
| No Audio | 23.43% | 41 |
| Trunks 'Dropping' intermittently | 42.86% | 75 |
| Trunk Registration failure | 16.57% | 29 |
| Poor Quality i.e. Delay/Jitter/Packet Loss | 33.71% | 59 |
| ITSP SIP Server failures | 19.43% | 34 |
| Incoming Call transfer failure | 17.14% | 30 |
| Call conferencing with 'external' caller fails | 11.43% | 20 |

*Responses, not percentages*

*Multiple selections were allowed on this question*

These items all highlight issues that can be avoided.  For example 'Codec mismatch'.  If the ITSP supplies the client with all the settings it needs for successful service installation/setup then these settings can be put into the PBX/SBC configuration forms to ensure successful setup.

Provision of good documentation for clients to use re: any settings such as Codec, RTP rate, Port numbers, DNS address, Specific IP addresses etc. etc. should ensure a really good chance of getting the service up and running without issue.  BUT, even if great documentation is available how can it be guaranteed that it is used and read as sometimes people just want to install and run to the next job? We all know that 'rushing' is a dangerous game.

Alan Percy also had a comment that 'lines up' with Chakra's:

 *"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."*

So, onto the 'Edge' with the next question.

## Q11: If your problems were with your SBC / Edge devices, what were they?
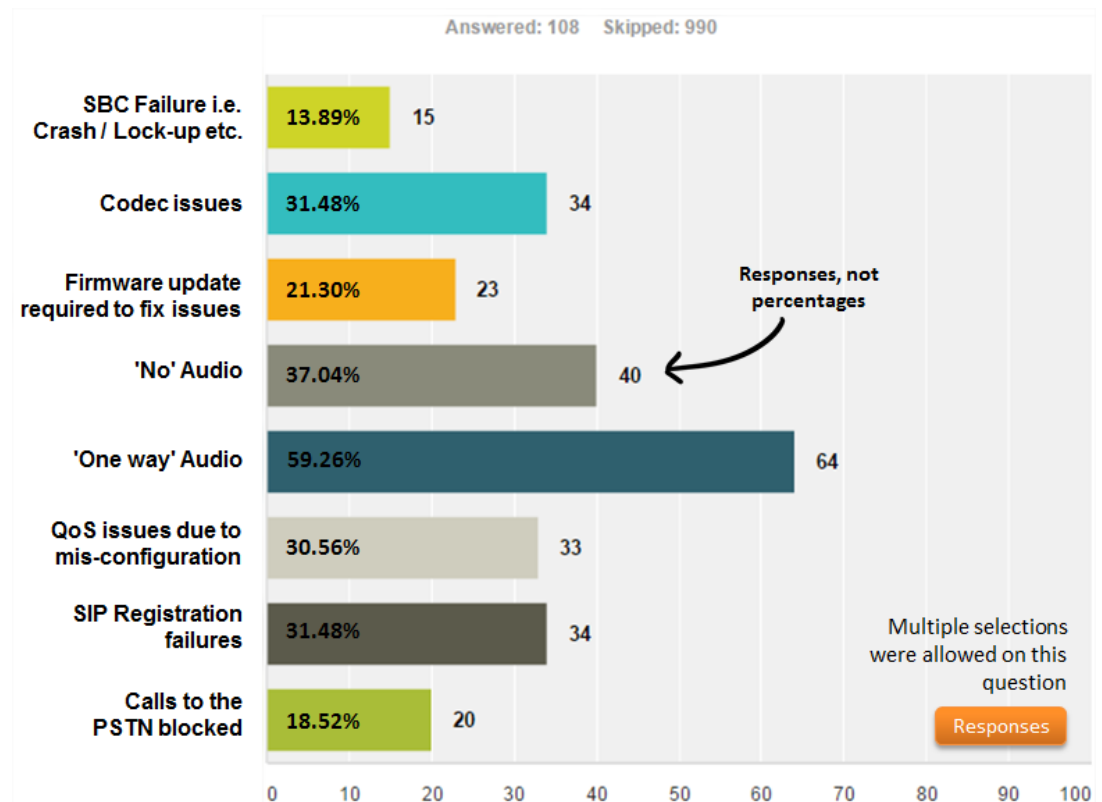
'**One way**' audio again is still the biggest issue here and fixing 'one way' audio is one of the (many) reasons people actually purchase an SBC, so why the problems? Misconfiguration probably... No Audio is also a concern.

'**Codec issues**' is still too high and I will repeat that this is something that shouldn't really be a major issue as people working with this type of 'specialized' equipment should have a good understanding of Codecs and be able to work with others involved in an implementation to ensure that Codecs are configured correctly and tested thoroughly.

'**SBC failure**' results are quite concerning as a device with such a fundamental role on the network should not crash or lock up. Again, test thoroughly and work with the manufacturer closely to see if there are any known issues with products before you 'leave' the testing cycle and also be extra careful afterwards when needing to apply updates to an 'active' system.

Good suggestions for questions next year (from David Leon-Guerrero / Cox) are:

*"How long did it take for problem resolution hours? Days/ Weeks? Who was key for root cause determination and ultimately resolution? Vendor Field Services personnel, Vendor Escalated Support teams, your own customer Network staff? etc."*

Let's move onto the PBX.

## Q12: If the problems were found to be with your SIP/ VoIP based PBX what were they?
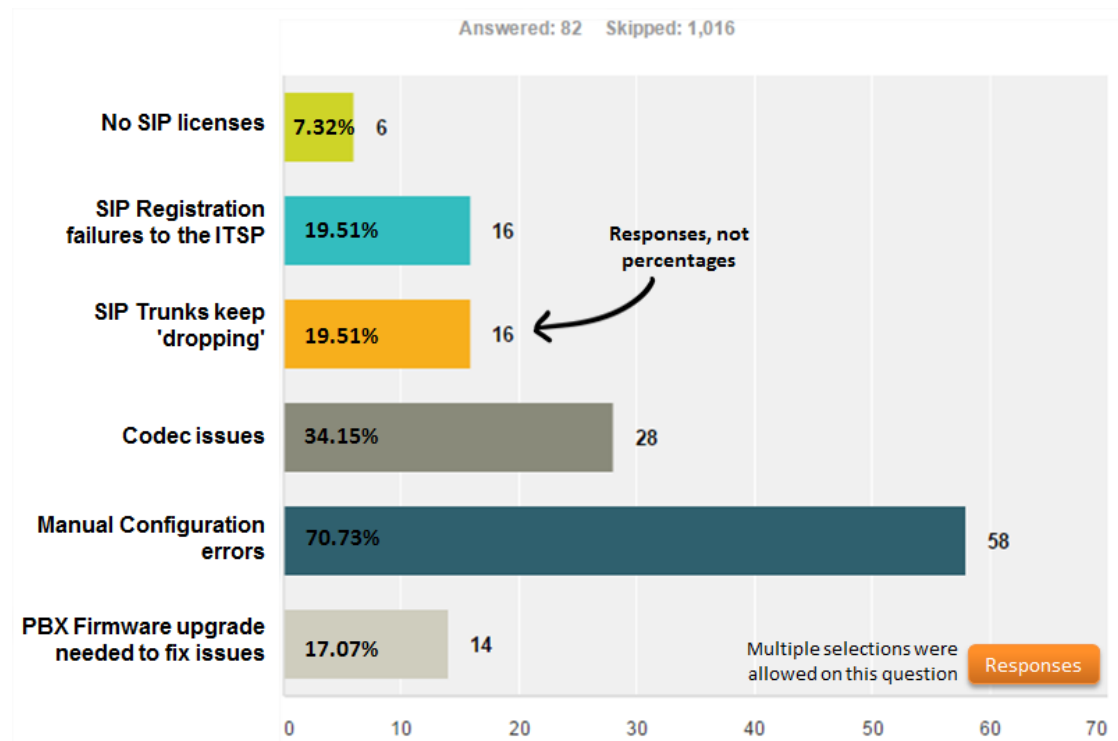
As ever, it's always frustrating to get problems but it should be easy to sort out the majority here?

- '**Manual Configuration errors**' is the biggest reason for issues… Why can this be? Simple 'Human error' or just rushing and not taking care?

- '**Codec issues**' Check settings early on to avoid this problem. Maybe in the future SIP trunk 'registration' could take a leaf out of the WebRTC book and specify (somehow) that there should be auto negotiation of codecs i.e. Opus or G.711 (as 1st choices) and manual override if something else is required i.e. G.729. Difficult to achieve?

- '**Registration failures**' and '**SIP trunks dropping**' again will be mainly attributed to misconfiguration or poor documentation

- '**No licenses**'. Easy to fix, ask the vendor how many (if any) you need and then purchase them. Oh and activate them as well…

This next question explores how people approached SIP trunking installations along with their reactions when things didn't quite work out as planned.

## Q13: After your initial SIP trunking 'Trial' period, what did you do?
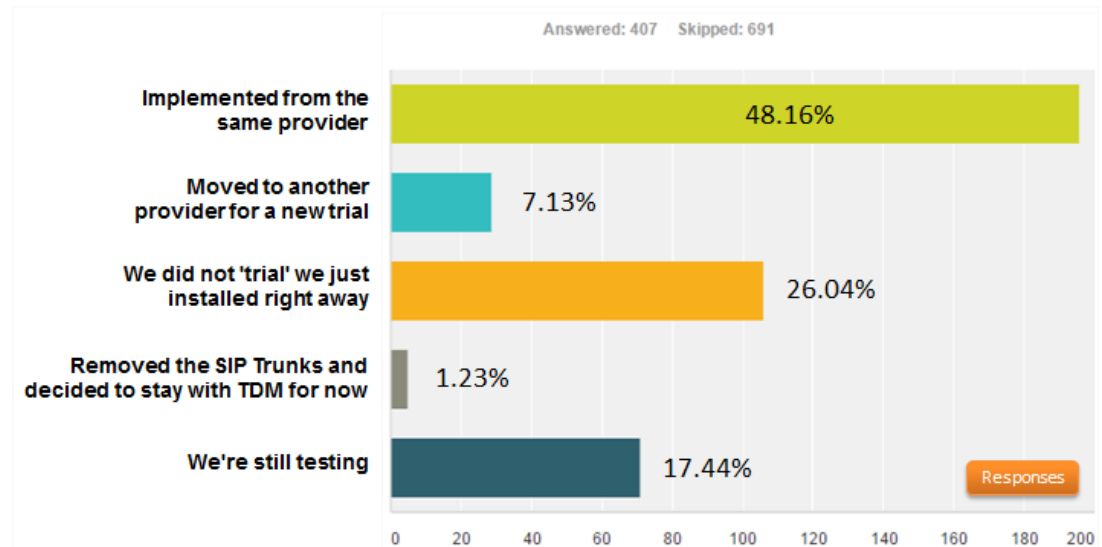
So, it's pretty clear that if things go well a client will (most likely) stick with the solution that they chose for the trial, almost 50% here have done so. There is no bigger reason than this kind of decision by the client for vendors / providers / dealers etc. to work together to make it work for the client. A happy client will stick with you for years thus making the time, effort and expense of offering a trial to them more than worth it.

26.04% jumped in feet first into the world of SIP trunking – fantastic but brave? I continue to think it is usually the best policy to trial SIP trunks before committing to any kind of contract and then finding issues later, maybe due to your testing (for whatever reason) not covering all usage scenarios.

Let's move on a little here and focus on what happens when things *do* go wrong.

### Q14: When things go wrong with the SIP trunks (operationally) and you talk to support staff, how do you rate their ability to fix problems?

Answered: 407    Skipped: 691

Multiple selections were allowed on this question

| | SIP Trunk Provider | Hosted VoIP Provider | SBC Manufacturer | PBX Manufacturer | Dealer/Re-seller |
|---|---|---|---|---|---|
| No Comment! | 9.23% | 40.16% | 18.80% | 12.85% | 26.08% |
| They need education! | 13.72% | 11.75% | 6.79% | 10.28% | 17.47% |
| Average ability | 46.88% | 31.97% | 35.25% | 36.76% | 38.71% |
| Great assistance showing good technical skills | 30.17% | 16.12% | 39.16% | 40.10% | 17.74% |

We didn't ask for specific incidents that support people had to deal with, all we wanted to get from this question was the customer's view of their support experience in general. **Remember -** it is customer experiences and perceptions that can win or lose business regardless of where the fault actually lies!

We fully understand that working in support can be a tough job, though if people at both ends of the conversation have a good technical understanding of SIP then it bodes well for a satisfactory and quick resolution but this is not always the case. You can take the figures as they are and reach your own conclusions though for us, we see that the Dealer/Reseller (year on year) fares poorly in this area – More training required?

Let's take a look at what people are asking for from the ITSP for the (not to distant) future.

## Q15: If you could ask one question of your SIP trunk provider what would it be?

Answered: 407    Skipped: 691

| Question | Response |
| --- | --- |
| When will you support HD Audio? | 11.55% |
| What are you doing to support FAX over IP? | 23.10% |
| Can you provide TLS/SRTP for secure signaling and media? | 25.31% |
| When will you support SIP 'profiles' to match each PBX/SBC manufacturer's configuration? | 32.68% |
| Other | 7.37% |
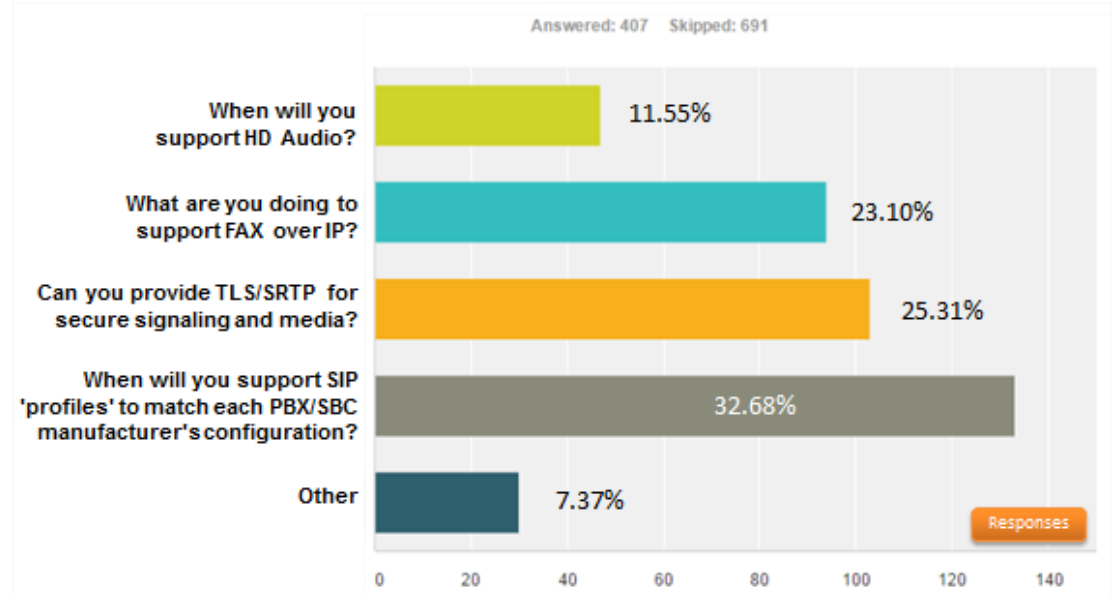
We added a new option for this year and it came out top! People want **Profiles** (or setup guides) that match their own equipment on their site from the ITSP. How many do you know supply profiles for all major vendors of PBXs and SBCs?

**Security** is high on peoples agenda which is not surprising as incidents of hacking are rising. It must be said that securing SIP trunks would be just one element of a corporate wide security policy.

'**Support for HD Audio'** is much the same as later year yet Fax over IP is still a service that not all ITSPs are providing – why is this?

Some good comments from **'Other'** we liked…

- *When will you support Dynamic Failover of Numbers between providers?*
- *What tools do your engineers use for troubleshooting issues and can we have a demo of them?*
- *When will you support HD audio and VoLTE multivendor roaming (VoLTE hub interconnect)?*
- *When will you support HD Audio on an IPv6 capable SIP trunk?*

All good questions…

Another new question for 2015 as we wanted to see if people have visited the providers in whom they will trust their communications to, we asked.

**Q16: Have you ever taken the time to visit your service provider's premise to take a look at their data center / network topology / fail-over / business continuity capabilities etc?**
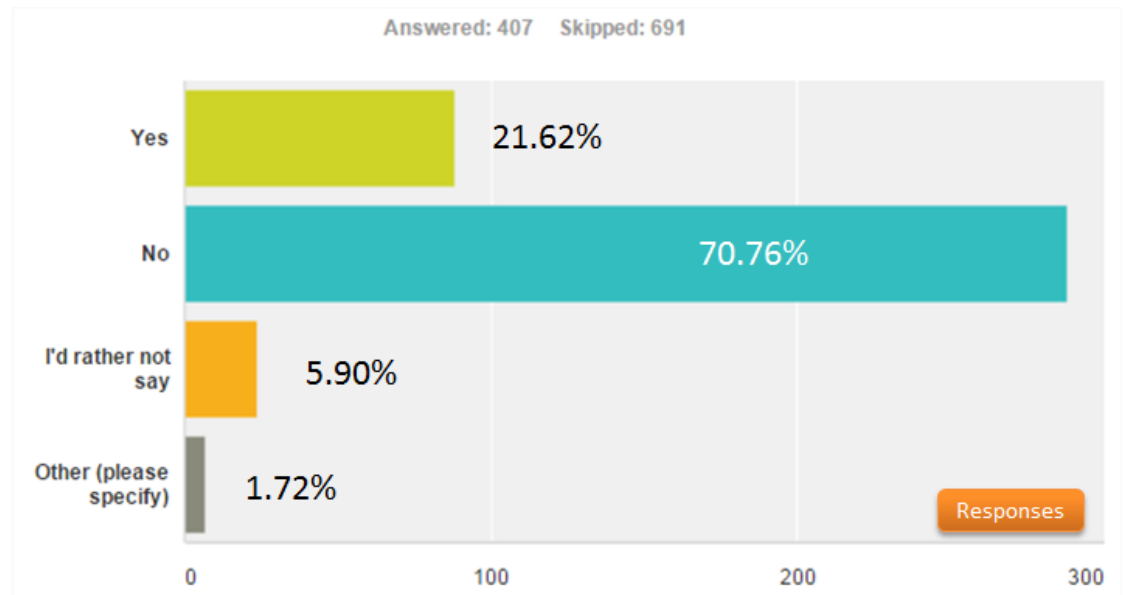
In general it's hard to read much into these results as one person may be responsible for visiting the ITSP yet another 50 people from the same company didn't visit the ITSP yet still completed this Survey.  One thing we will say is to make sure someone **does** make the visit and asks all the right questions about the service you are buying i.e. Support, Disaster Recovery, Re-routing to carrier partners and lots more…

One response did actually say 'Not allowed' - I wonder what that was all about?

A new question again as we wanted to find out what tools people used to test / troubleshoot their communications services

## Q17: If you actively monitor your 'Voice services network', what tool/s do you use?
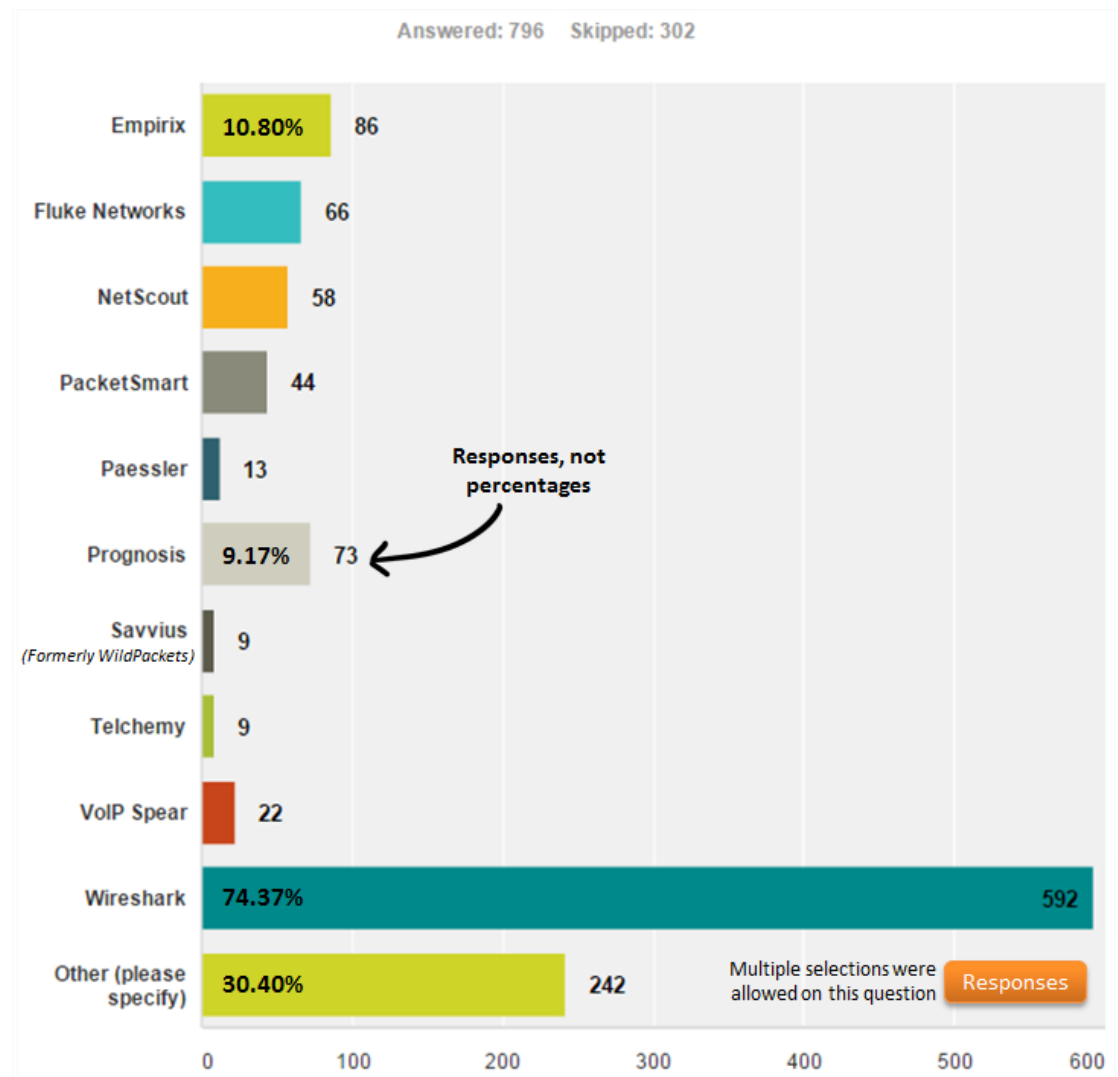
People could select multiple options here and it's no surprise that **Wireshark** is the clear leader. It's a great protocol analyzer and being free is hard to compete against. Of course other tools provide specific services that Wireshark can't with regards to Real-time VoIP traffic analysis, Monitoring, Alerting, Report generation etc. etc.

Some others that were mentioned in the 'other' response were Palladion, Homer, SolarWinds, One Vision and Marwatch.

So now onto SIP and what you know

## Q18: Understanding SIP is important, which most interests you?
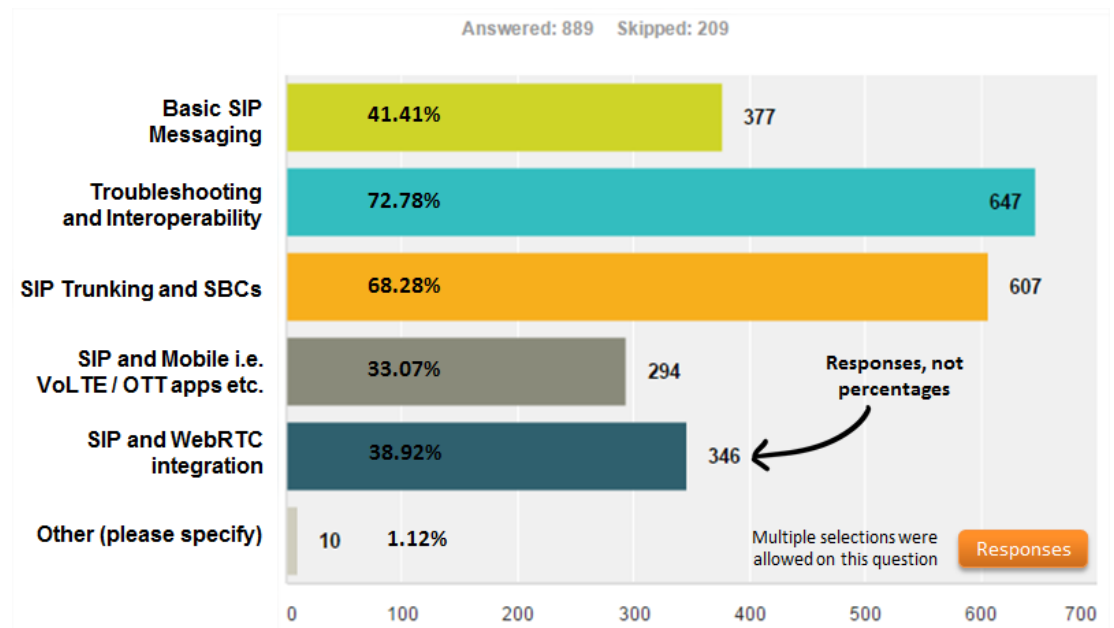
Again **Troubleshooting**, **Interop** and with (we think) a focus on getting the Edge to work with the ITSP providing the SIP trunks are the most important areas. Of course, you want to get the SIP trunks up and running and be able to fix things if there are issues.

**SIP and Mobile** is a growing area of interest so we actually added a whole new section into our training called **SIP in the Cloud, LTE, IMS and VoLTE** this year.

**SIP and WebRTC integration**, is growing in importance as more WebRTC services need to interoperate with existing SIP based infrastructures.

As is always the case with this survey, we look at what people are interested in and work towards improving and complementing our existing training programs.

We even went a step further and asked about what people feel they need to see more of in The SIP School's own training program.

## Q19: What would you like to see in our SIP training program?
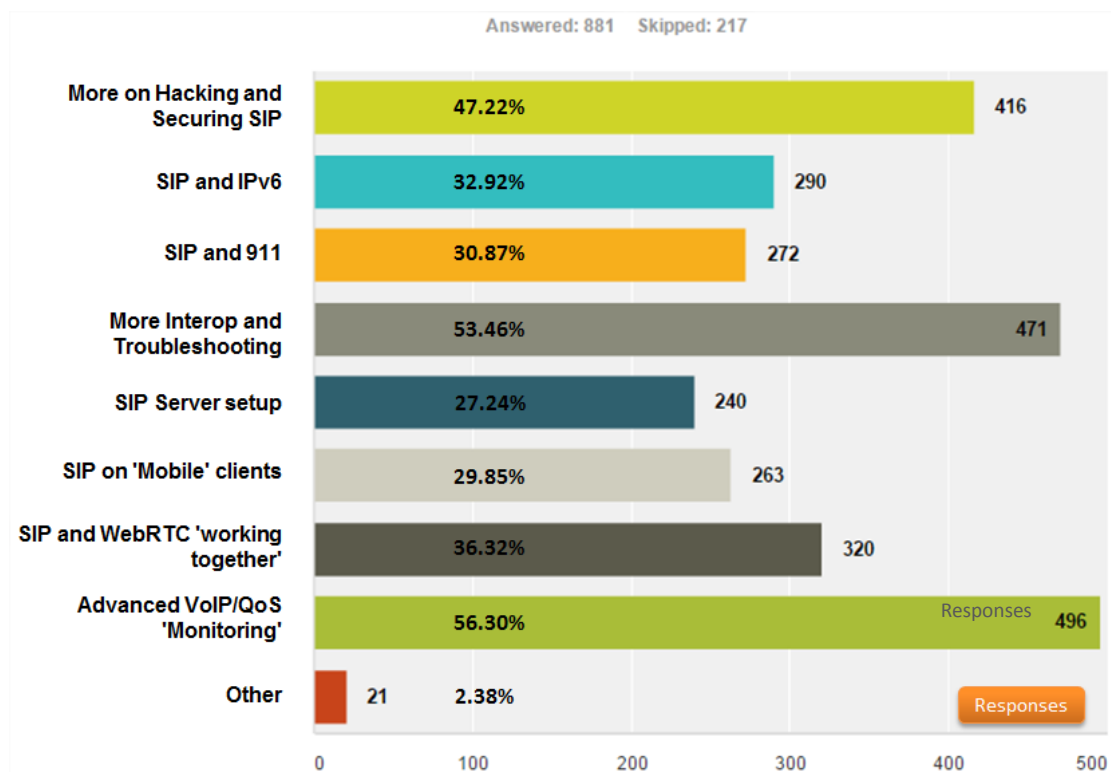
**Advanced VoIP/QoS monitoring** is a great request as we always say that it's best to watch what's happening on your network and spot potential issues early on. We have a WAN Assessment network available to all students so that they can set up SIP trunk call scenarios and run them for real. Students can analyze the results on screen and via a downloadable report. We intend to extend on this area with even more monitoring, testing and alerting features by Q2 2016.

**Security** is always high on the request list and we do cover a lot in the program. During our next update cycle we will be including even more on SIP trunk security.

**SIP and IPv6** is still being request but as of yet, no real offerings from the providers.

**SIP and WebRTC 'working together'** is something that we already cover in our WebRTC programs and more working examples will be included in 2016.

The SIP School™ is the issuing authority for the SSCA® Certification with over 4300 certified engineers around the world! We know that this survey presents a good opportunity to see if people want or even need a SIP Certification. So, we asked:

## Q20: Is an 'official' SIP Certification important to you?
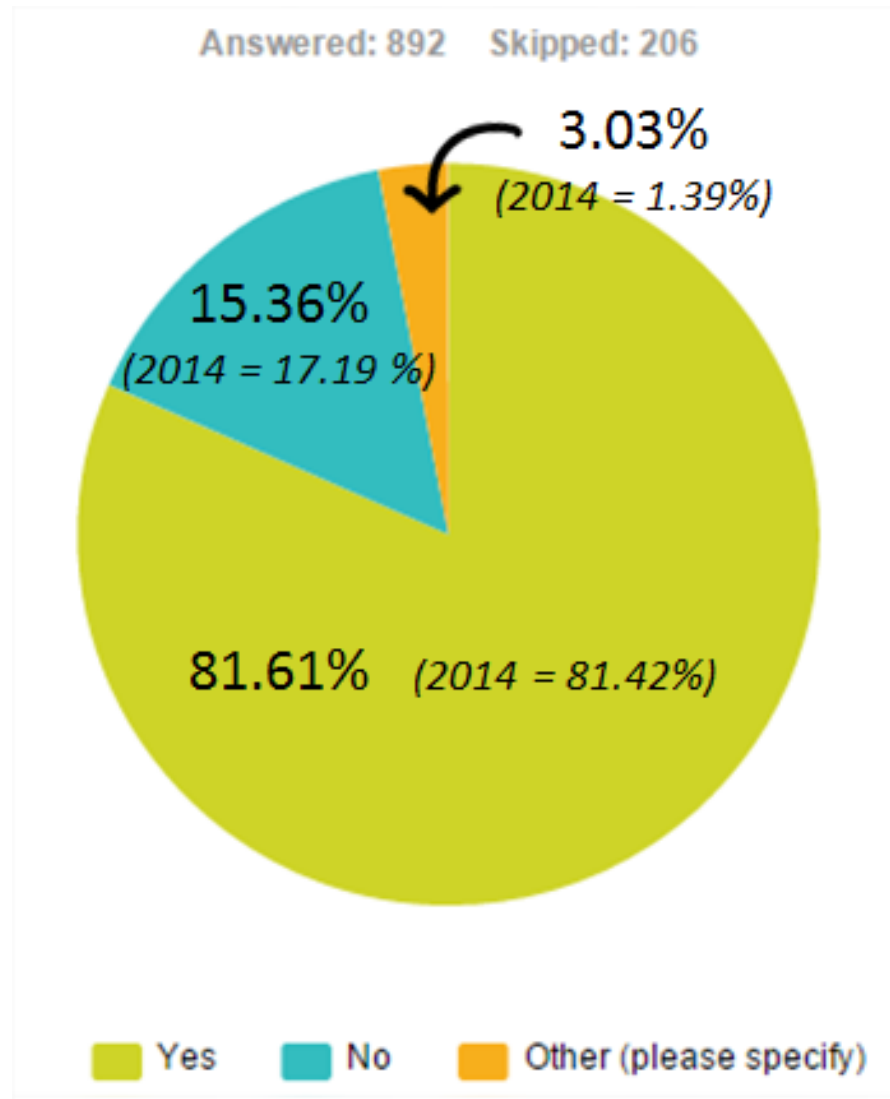
"I agree that while the SSCA isn't a vendor specific certification like CCNP from Cisco for example it does reflect a proven familiarity and knowledge of SIP on the part of the individual engineer."

*Joe Alice, Verizon*

"Again, in line with all the other responses. Certification is highly desirable."

*Hussain Ali, Cisco*

"Certification makes the person a safer bet when hiring or promoting. It is hard to evaluate based only on experience."

*Gary Audin Delphi*

Answered: 892    Skipped: 206

3.03% (2014 = 1.39%)

15.36% (2014 = 17.19 %)

81.61%  (2014 = 81.42%)

Yes    No    Other (please specify)

Small changes from last year but still with over 80% of people saying yes, it indicates that having a standard certification to aim for helps students strive to attain the certification that industry recognizes.

But hang on, this is not over yet!

WebRTC has been making a lot of noise in the world of communications and web technologies and it's set to get noisier. Of course, we wanted to ask a couple of questions to get a general 'feel' for what people are thinking.

## Q21: What do you know about WebRTC?
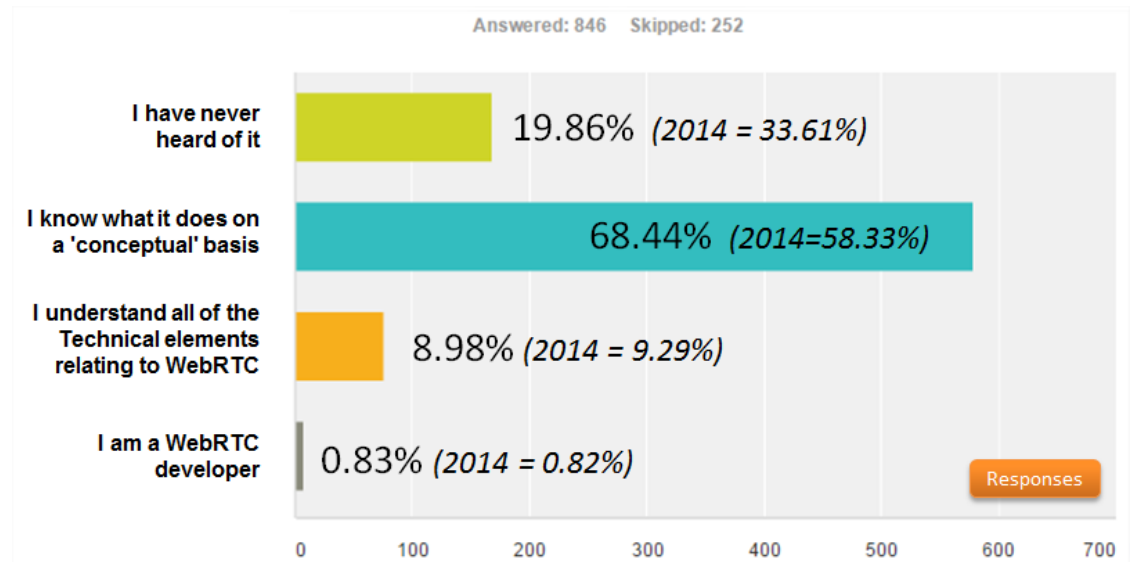
WebRTC to many people is brand new and there are still many people out there who work in Telco that have never heard of it.

It is good to see that over 65% understand what the 'idea' of WebRTC is because they then have it on their 'radar' as the specification develops and more applications become available. It's also a really good time to start 'playing' with WebRTC to make sure that when your company is ready to work with it you are not starting from scratch … Build your experience up now…!

There was also the option for 'Other' and one comment made us want to share. The respondent said:

 *"It's a way of making the Telephony network insecure!"* … ??????

"WebRTC is a tool, and people use a tool for a variety of jobs. It is clear that service providers most likely force WebRTC media flows through media devices (somewhat analogous to SBCs, but not the same) while web developers will utilize the more P2P modes."

*Alan Johnston, Avaya*

"While it's clear that WebRTC will need SBC support to cross enterprise boundaries, usage within enterprises or on the open Internet may not. SIP was originally designed to be peer-to-peer, but because its primary target was POTS replacement it ended up with similar deployment requirements. WebRTC is at least one step closer to being "communications, but not a phone call", making it useful in scenarios where centralized control, federation, etc. may not be as necessary."

*Dan Burnett, Standards Play*

### Q22: Do you think WebRTC will be 'allowed' to flourish as a Peer to Peer technology or do you think it will follow the same path as SIP and require multiple intermediary devices such as Session Border Controllers to work?
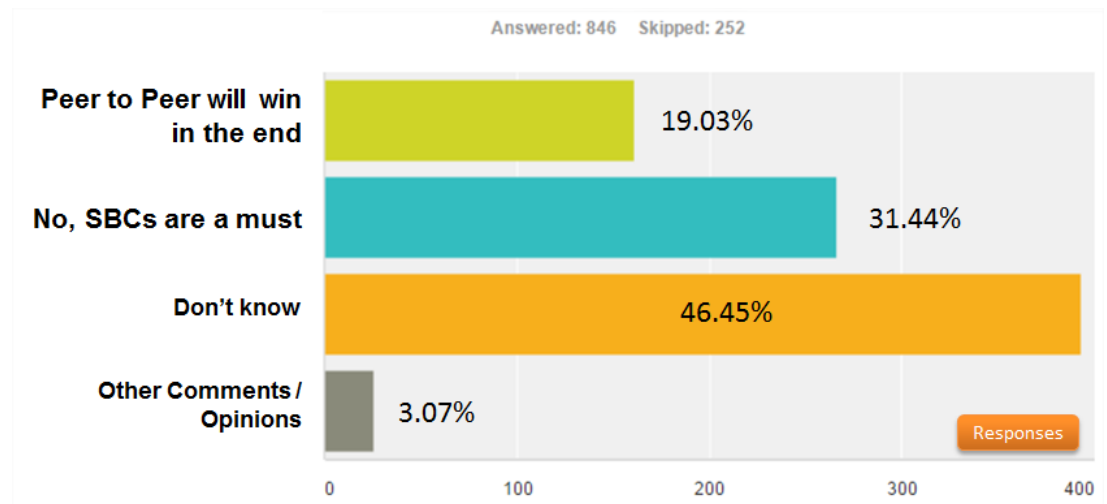


This is a revealing question where the answers will shock / disappoint the WebRTC purists.

WebRTC can be a 'pure' Peer to Peer play that the existing Telco world needs to understand doesn't need any intermediary devices. SBCs will come into play (providing lots of Gateway functionality) when WebRTC sessions need to include services / devices running other protocols such as SIP, Jingle, XMPP as well as non-WebRTC (recommended) codecs. WebRTC utilizes Opus for Audio and VP8/H.264 for Video (at the moment!)…

At this early stage with so many people expecting to have to use an SBC, will this hamper WebRTC's Peer to Peer model or will the sheer volume of WebRTC implementations over the web start to bypass the need for SBCs as all communications move to the Web over time?

This thought is also reflected in another of Alan Johnston's comments, *"What is more interesting is the question of whether telcos will take advantage of the P2P architectures that WebRTC allows."* He said.

A new WebRTC question for 2015.

## Q23: What do you see are the biggest challenges with using WebRTC?
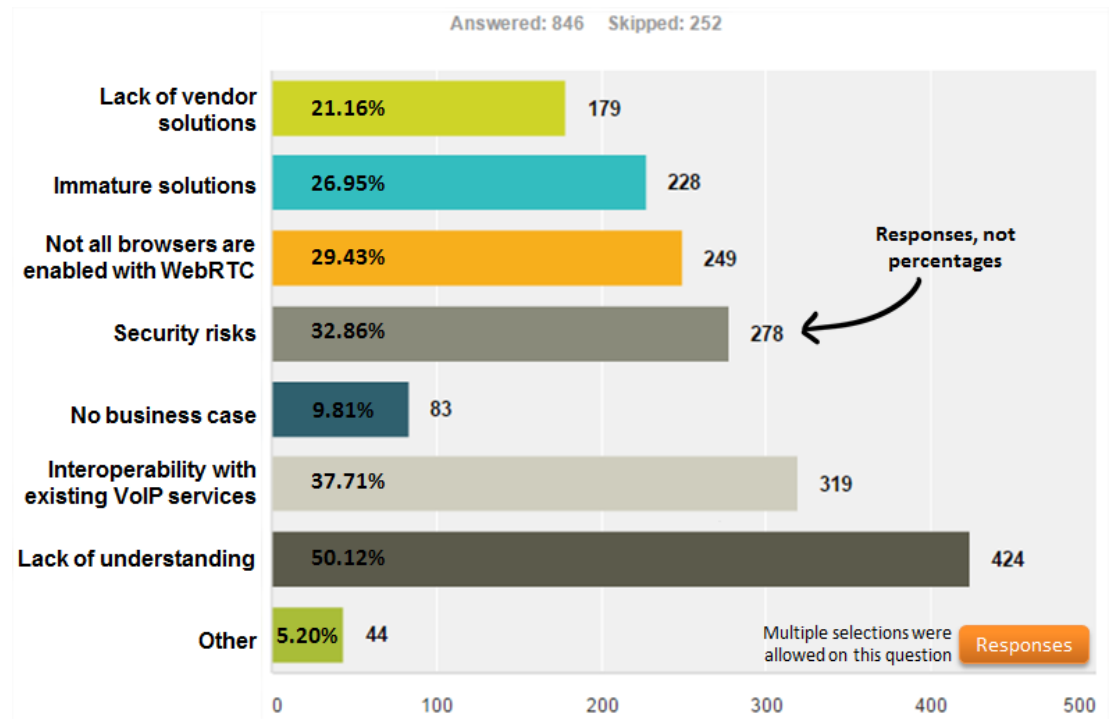
Of course with any new technology there are going to be a lot of questions that need to be answered before enterprises look to using it. Here we wanted to see what people thought may be holding back / hindering the adoption of WebRTC and overall there seems to be plenty of stumbling blocks even though a lot of the above concerns can be addressed easily.

- **'Lack of understanding'** can be solved by education and keeping abreast of developments. These websites are good places to start.
  - http://www.webrtcworld.com/
  - https://bloggeek.me/
  - https://webrtchacks.com/

- **'Not all browsers enabled'** has always been brought up as an issue but with the new Microsoft Edge browser moving toward supporting WebRTC, things are improving.

- **'Security risks'** is always a concern but as all Voice / Video streams are encrypted by default with WebRTC this should help to alleviate these worries.

This new question wasn't meant to be a trick question, just simply a question to see what people's answers were with regards to how WebRTC is perceived with relation to SIP.

**Q24: This question is designed to get your comments and thoughts so we can see what you really think WebRTC can do, here we go:**

**Do you think WebRTC services will pose a threat to SIP trunking in the future?**



"The folks who answered yes may be actually perceiving the real "threat" of WebRTC, the elimination of trunking altogether.

Of course WebRTC will not replace SIP as it is not a trunking signaling protocol....it eliminates the need for trunks altogether in a new paradigm."

*Phil Edholm, PKE Consulting*

"Glad to see that the respondents understand that WebRTC and SIP are complementary technologies and that many of the future applications will leverage both together."

*Alan Percy, Dialogic*

So, **'Not likely'** was a winner here and rightly so as WebRTC and SIP are not the same. SIP can be used as the Signaling protocol that WebRTC needs though it's not the only option. There are multiple applications for SIP yet it's main strength lies in enabling enterprises to leverage IP connectivity for their Voice and Video conversation as well as Service providers using SIP for interconnect and control of an IMS they may have deployed. Therefore, we also like the amount of '**They are not the same thing**' responses….

**'Most likely'** is interesting because is it people thinking that WebRTC is a similar but better/newer replacement for SIP or that over time, even though people have migrated to SIP, WebRTC services will become more popular and people will find that using WebRTC enabled devices and apps means that there is less need for SIP services.

Other comments / opinions is where we were hoping to see some good comments and this is what we got:

- *They are not the same. The ideology between SIP and WebRTC and their usage varies. WebRTC mainly concentrates on delivering Peer-to-Peer rather than Server and Client.*

- *There are certain scenarios where WebRTC can´t compete against SIP trunking; also QoS is an issue.*

- *I view them as an added channel which won't necessarily compete with SIP trunking. They'll coexist.*

- *Not really a threat, especially given SIP's maturity in the marketplace. Market forces and certain use cases might make WebRTC suitable, but SIP seems well entrenched and is adapting to the market needs.*

- *Not in the immediate future, right now I think SIP trunking is still working to overcome PRI and T1 trunking, it will be a long time before that happens, and even longer before WebRTC overtakes SIP.*

- *They're compatible in terms of "horses for courses", WebRTC is great for click-to-talk, SIP for Carrier to Customer and Carrier to Carrier.*

- *I don't think it will pose a threat in the sense that it kills off SIP trunking. I see it more as a complement to SIP trunking. It enables a different type of communication than does SIP trunking and will have different uses.*

- *I see businesses possibly needing to support inbound WebRTC traffic but TDM/SIP/traditional voice services will still be the primary means of communication for quite a while.*

- *I think WebRTC will supplement existing services rather replace them.*

**The Final Analysis**

So, let's take a step back and see what we've found out and hopefully learned from this year's Survey.

Just as we saw in the Survey last year, there are still significant issues that can arise when implementing SIP trunks and these issues must be faced head on in order to make installations go smoothly.

SIP trunking is working in a lot of places where competent companies work together to ensure interoperability and continuity of service but looks like it can still have problems (as also seen with previous year's surveys) when it's installed quickly and without careful thought for Quality of Service and ongoing service management. Customers can get exasperated when they can't make or receive calls from their own clients and can resort to extreme measures.

**Recommendations**

This survey shows clearly that issues occur in the main during the installation and configuration of SIP trunks and in order to make things as painless as possible for all parties involved there are some simple things that can be done and most of these can be done relatively quickly.

Firstly, it's always wise to talk to all parties involved before moving forward. Get case studies from ITSPs and the vendors you are working with. Talk to their people about their installation experiences along with discussing the issues they have come across and how they overcame them. Talk to people about interoperability testing and conformance to standards and recommendations such as SIPconnect from the SIP Forum. In essence, good research and talking to people early on will help you decide which companies to work with.

It should be noted that the SIP Forum is starting work on the next version of the SIPconnect recommendation (version 2.0). This will be a really important update to version 1.1 of the document and I urge all those involved in SIP to take a look at what's happening - even get involved in helping to 'shape' the new recommendations.

Before committing to a SIP trunk installation (or even a trial) it's recommended that you assess your network for suitability. WAN assessment tools are available in the market for you to do this (maybe your prospective provider has some) and these tools should highlight any potential issues such as link instability, router problems, bandwidth issues etc. etc. MPLS networks can deliver on Quality of Service (QoS) but can also be expensive. New 'developments' such as SD-WAN may be able to 'beat' MPLS for cost but needs to deliver an 'assured' service to be taken seriously.

If you are an enterprise looking for a SIP trunking solution to suit your needs then ask ITSPs to respond to your business requests and see if they can cover everything you need from Service Level Agreements (SLAs) to support for the smallest of sites in the remotest of locations, even international locations. You

must ensure that everything can be covered by the ITSP and that they understand everything you need before things move closer to provisioning SIP trunks. ITSPs should also be willing to let you trial SIP trunking for free for a reasonable period of time. Survey responses clearly display that a successful trial will normally result in a full implementation. If so, then do it and test the Trunks using all the call scenarios you can think of such as call transfers, conferencing and so on. Also test the ITSP's support people at various times of the day, why not call them at 6pm on a Friday evening and see what the response time is and the level of knowledge of the support personnel available?

The market is also changing quickly with smaller companies merging and being 'absorbed' by larger organizations, along with the 'giants' of the industry finally coming to market with SIP services.

If you are 'looking' for SIP trunking services then you need to decide who can deliver exactly what you need today as well as being around in the short / long term to continue to deliver these services – not an easy thing to predict with such market activity at the moment.

When it actually comes to installation of the SIP trunks the one thing that really stands out is the need for correct documentation that supports the configuration of the PBX and the SBC/Edge device in order to get SIP trunks to register and work. In our own experiences we've found that installations go way more smoothly if educated and experienced people use documentation that is clear and easy to understand and also based on settings that have been proven to work in the combinations of equipment being configured. For example, if you are installing SIP trunks from AT&T, connecting to an Ingate SBC and then into a ShoreTel PBX it's important to check that these all interoperate and if so, get the configuration documentation into the hands of the installers. Again, make it easy for people to get it right the first time.

Once SIP trunks are in and working it's not the end of the story. To ensure they continue to operate and function at their best it's wise to continually monitor their performance. Whose responsibility is it to run monitoring / alerting software to check for Mean Opinion Score (MOS) values falling, increasing packet loss, increasing Jitter values and so on, YOU or the provider? Someone needs to do this.

Also ensure that whenever any software upgrades are to be carried out on the PBX or SBC that these upgrades do not affect service. Again it's wise to talk to vendors first before making any changes that may affect operation of the SIP trunks.

One more note… Don't forget to read the list of things that people attributed to 'Never having a problem' with their SIP trunks, lots of good advice there.

**WebRTC in the Survey**

WebRTC is an interesting addition to this survey and it's clear that knowledge is growing. A lot of services are available now with many being free and simple to use, yet these hardly scratch the surface of the potential of WebRTC. New ideas and services are springing up daily so please keep your eye on some of the WebRTC blogs and sites available.

WebRTC is an OTT phenomenon but even though it has amazing congestion adaptation capabilities for delivering Video and Voice over unpredictable bandwidth conditions, if all the bandwidth is gone, then the experience will be terrible. The only way to avoid this is for all services providers (globally) to agree on a percentage of their network being dedicated to Real time traffic. But as this may conflict with how they offer / differentiate their own services on the network with guaranteed QoS, this may never happen.

The 'near future' is sure to see Enterprise to SP communications as well as Inter Carrier plus VoLTE services relying on SIP where the 'Web world' will embrace WebRTC more, well it was what it was designed for. And who knows? In 20 years' time will all Comms be web based with not a SIP trunk in sight? Whatever the outcome, you know things will take time with some interesting twists and turns along the way.

**Conclusion**

This survey has again highlighted the need for all parties involved in providing a complete SIP trunking solution to clients to work together and continually test their products and services for interoperability so that it's not the customer's premises that becomes the test bed.

We all know that SIP, Voice and Video over IP services are the future and TDM services will one day be a thing of the past. How quick this all happens is up to the people providing the services.

The SIP School would like to thank all those who contributed with their valuable comments and insights.

**About The SIP School**

The SIP School does not formally recommend any one provider, service or product as we are a friend and supporter of all who are involved in the world of SIP, Voice and Video over IP and now, WebRTC.

The SIP School™ is owned by Vocale Ltd which was founded in April 2000 (Vocale Ltd is also the owner of the WebRTC School). It's SSCA® SIP training and Certification program has become recognized as the globally accepted Certification for VoIP professionals to strive for. Organizations such as the Telecommunications Industry Association officially endorse the program and Bicsi value the program at 19 CEC credits towards their own certifications. Details of more industry supporting companies can be found at http://www.thesipschool.com/industry.html

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