Choosing a SIP trunk provider

Executive Summary

A guide for UK businesses looking to deploy SIP trunk services, the issues to consider when selecting a network operator and how to avoid the pitfalls of deploying SIP trunks in a business environment.

Graham Lewis
Director of IP Engineering
White Paper – Rev 1
Spitfire Network Services
May 2011

Introduction

This is a guide for UK businesses looking to deploy SIP trunks. It is aimed at a telecoms manager, business owner or anyone who has been charged with looking at or implementing SIP trunks in their business and needs to find a SIP trunking provider commonly referred to as an Internet Telephony Service Provider (ITSP).

The UK has a vibrant and competitive ITSP market some of whom are new entrants to telecoms and others that have been around for many years, and are now offering SIP trunks as a natural extension to their ISP or switched minutes business.

Much of the content is general to a Worldwide audience, some of the content particularly regarding number portability and emergency services provision are specific to the UK market.

What is a SIP Trunk?

Although the original SIP specification [RFC 2543] has been around since 1999 it was never specifically designed to be used for PBX to PSTN connectivity. However it has been adopted as a standard for providing Voice over IP (VoIP) telephony on a new generation of PBXs and connecting those PBXs to the existing PSTN network.

The role of connecting an IP PBX to the rest of the PSTN over an IP network is that of the SIP trunk and in doing so replaces or augments TDM technologies such as analogue lines and ISDN. A SIP trunk is the service that allows your IP enabled PBX to make calls to the Worldwide Public Switched Telephone Network (PSTN) network.

A SIP trunk requires IP connectivity, either over the Internet or within a closed private network, from your SIP enabled PBX to an Internet Service Telephony Provider (ITSP). The PBX is configured with settings provided by the ITSP to send and or receive calls to and from the ITSP. The ITSP provides connectivity to the global PSTN either via traditional TDM interconnects or IP interconnects to all national and international destinations.

PBX and SIP trunk compatibility issues

SIP was never designed to be used for PBX to PSTN connectivity. There are still many areas of potential incompatibility between SIP trunk providers and PBX vendors, particularly with the 70 or so ancillary RFCs that provide additional functionality and feature enhancements to basic SIP messaging. In addition SIP throws in the paradigm shift that end devices have intelligence compared with a traditional TDM network, where the switches have intelligence and end devices do not adding potential complications.

Different vendors use different SIP message types to implement even basic functionality such as call transfers and presenting Caller Line Identification (CLI). Both a PBX vendor and a SIP trunking provider can rightly claim that their service adheres to the SIP protocol specifications but still will not work with each other. Some PBXs are more flexible than others some traditional PBX manufacturers keeping to the maxim that “our way is the right way”.

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Attempts have been made to standardise SIP trunking and PBX interoperability but even the vendors who have signed up to them don’t always abide by what are in effect only guidelines. The only sure way to guarantee interoperability is to check whether your intended provider has actually tested your PBX model and software release against their SIP trunking service.

Acknowledging that there are still some interoperability issues PBX vendors are changing their SIP implementations on a regular basis which occasionally gives rise to further interoperability problems when software is upgraded. Vendor software revisions can make or break SIP trunk features.

So the first question to ask a proposed SIP trunking provider is will it work with my PBX and are there any caveats? Some SIP trunk providers are just reselling a SIP service and will not have actually tested it themselves but just rely on the fact that the upstream provider has done it sometime in the past. If there is any uncertainty ask the SIP trunk provider whether they provide a testing service so that you can work through the issues beforehand.

Who owns the network?

The increasing popularity of SIP trunking has given rise to "virtual operators" and a healthy resellers market who resell capacity on another ITSPs network either with some of their own infrastructure or under a "white label" agreement. Some resellers can add real value particularly if they are experts in your particular PBX or understand the needs of your industry segment better than a larger operator can. However it is as well to know in advance whether they are operating their own network using their own network infrastructure or reselling capacity on someone else's network.

If the ITSP is managing their own infrastructure then they will have control over the whole SIP trunk solution. If things go wrong a reseller will be one step removed and may struggle to get faults resolved. The major ITSPs are interconnected operators with multiple connections to the national PSTN network, a smaller ITSP may just hand over traffic to a larger upstream ITSP. Ofcom publishes a list of what are referred to as annex II interconnected operators.

It is not a clear cut decision as to which is best but it is best to know who you are dealing with before signing up.

Number Porting

SIP trunking allows geographic PSTN numbers to be routed to any IP address in the World, unlike the traditional TDM network where numbers are tied to a local telephone exchange. It was intended that special VoIP numbers would be used on VoIP services but for businesses the ability to advertise what customers will recognise as 'local numbers" and the need for businesses to keep their existing numbers whenever possible has driven the requirement for number porting between operators.

"Porting in the UK is an archaic and slow process"

Telephone number porting in the UK is an archaic and slow process. If you want to move your existing numbers to your SIP trunk provider check whether they can do this and get a realistic estimate of the time it will take, 30 days is not unusual. Ask what interim arrangements can be made until the transfer date.

Number porting requires bi-lateral porting agreements and not every operator can port to or from every other operator. Ofcom regulations state that this should be the case where technically feasible the process can be a long drawn out affair.

If you are going to get new PSTN numbers from the ITSP check what kind of numbers are available. Most operators can offer a full range of geographic numbers in addition to "VoIP numbers".

Ask what happens to your telephone numbers if you move away. Will you be able to take them with you? Don't always assume this will be possible particularly if there is an upstream reseller involved. When Tesco closed down their VoIP service after the ITSP who was providing the service went out of business all the telephone numbers reverted back to the Ofcom range holder and customers could not port their numbers to new providers.

Fax Support

Attempting to send and receive faxes over IP networks can be problematic as fax is sensitive to jitter and delay. Not all ITSPs support fax over their SIP trunks. Few will actively try to block it but when faxes go missing or come out all skewed and covered in dark lines you won't get a sympathetic ear. There is an ITU protocol - T.38 that is designed to overcome the problems of sending real time fax over IP but problems persist more than ten years since its introduction making fax over SIP trunks a hit and miss affair. If fax communication is critical to your business make this clear to the SIP trunking vendor in the initial discussions.

NAT traversal

Network Address Translation (NAT) is often deployed in small business environments. It provides for an internal network infrastructure to be deployed using what are called private IP addresses and connections out of the local network are translated to globally routed public IP addresses by a network router or firewall. SIP and NAT are not natural bedfellows. SIP assumes that every IP device has native end to end connectivity and assumes this when sending IP information in its SIP messaging. Devices receiving SIP messages unless told otherwise assume this is correct so if a SIP message says that a PBX is on a private IP address this is where the SIP application tries to send replies to but will fail.

If SIP trunk traffic is passing through a NAT device something along the way has to manipulate the IP addresses in the SIP message. There are various ways of performing this either at the PBX or at the ITSP end and an ITSP will support one or more methods of doing this. Without going into technicalities of how each method works you need to agree whether you are going to resolve the NAT issue or are relying on the ITSP to do it.

Some ITSPs may require you to make firewall changes or deploy a special SIP aware gateway which may add an unexpected cost to the deployment of SIP trunks.

Emergency services

An ITSP should state whether they are offering a PAT (Public Access Telephony) service in which case they must be able to route 999 & 112 calls. If the provider is not offering 999/112 services you need to understand the implications of not being able to make calls to emergency services and have alternatives in place and communicate this fact to your staff. An ITSP offering a PATS service must populate the UK 999 database with your numbers to ensure that emergency services are able to respond to “dead air” calls. This is an important regulatory requirement that could have serious consequences.

Company history

“Know thy customer” is a maxim that is often touted but “know thy supplier” is equally important if telecommunications is critical to your business. Check your intended supplier’s credit rating and how long they have been in business. The barriers to entry in the SIP trunking business can be relatively low which makes it easy to walk away from a market segment if it is not profitable leaving customers without telephony service at short notice.

As with any emergent technology lots of new energetic businesses are being born but not all of them may stay the course. Ask how long they have been doing SIP trunking, what investment have they made in their infrastructure to try and guage whether they are in the business for the long run or speculation. Many ITSPs are on their second or third network refresh as the technology has moved on and the market for SIP networking equipment matured.

Fraud

"Dial through" telephone fraud is on the increase mainly against the new generation of IP PBXs. Most are not sophisticated hacks against a system but rely on the fact that today it is easy for anyone to install a telephone system - no exams needed and no experience of performing basic PBX security checks required. With maintenance passwords and default account settings just a click away to anyone who can use Google they are prime targets for fraudsters who can leave a business with a telephone bill of tens of thousands of pounds over a single weekend.

Discuss what arrangements your intended ITSP has in place to detect and notify you of suspected fraud. Is the first time you are going to know about it when you get your call bill at the end of the month? Most will not discuss the details of how their fraud detection systems work but they should give you some assurance that they are on your side against fraudsters and will make genuine efforts to detect and mitigate losses.

Dispute resolution

A reputable ITSP will have accreditation with a formal industry dispute resolution service such as that run by The Office of the Telecommunication Ombudsman (Otelo). It is rarely invoked but it gives an indication that the company is a serious player in the telecommunications industry. Conversly being the member of an industry body gives no guarantee of the quality of the companies SIP trunks.

Voice quality

SIP trunking is in essence just another application on an IP network and the end points can theoretically be offered from anywhere on the internet just like web hosting, mail hosting and numerous other applications. However the special requirements of transporting voice traffic over IP mean that special consideration must be given to the quality of the IP connection.

The SIP protocol itself is only responsible for completing call set up and tear down. Once the call is established the audio content of a call is carried over a different protocol called Real Time Protocol (RTP) which itself runs over UDP. To successfully carry a two way voice conversation with a quality that is acceptable to both parties an RTP stream of relatively low but constant delay and without any packet loss needs to be available in both directions.

These demanding requirements mean that potentially the most troublesome area of SIP trunking is actually the RTP element.

There are a number of factors that effect voice quality. The first is the actual useable bandwidth of the service – if that is a broadband ADSL line then the upstream bandwidth will limit how many simultaneous calls can be made. If you are considering using an ADSL service then the “Headline speed” of an ISP’s package is no use when choosing one suitable for voice service, a large download speed does not equate to a large upload speed, in fact the reverse is true.

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Voice quality

The contention ratio of the IP service determines how many other customers are sharing bandwidth on the backhaul network which means when the network is busy not everyone can get through. Contention (or statistical multiplexing if you make your living out of it) has been part of telecommunications since the early days. There has never been enough capacity for everyone to phone everyone else simultaneously but what there has been in traditional telephony is Call Admission Control (CAC). This ensures that when there is no capacity call set up is not allowed to be attempted. By contrast on a VoIP network where there is no CAC if you establish more calls than your IP bandwidth can support then all the calls are set up but the voice quality on all the calls degrades.

Although there are protocols available in IP networks for call admission control they are not widely deployed, as to be effective they must span the whole IP network end to end to reserve bandwidth. So at the moment call admission control between SIP trunks and PBX is normally hardcoded – that is you work out how many simultaneous calls can be made over your IP bandwidth and then prevent the PBX from accepting or making more calls than allowed.

Codec choice – there is only one codec that can offer ISDN quality voice and that is G.711 A or U law. Your ITSP must support it and should positively encourage its use.

All other codecs involve compressing and decompressing voice which means at some point or more along the path the voice has to be processed which can result in poor voice quality or some very weird sounding effects.

When calculating bandwidth requirements G.711 uses 85Kbps in both directions over Ethernet and 115kbps in both directions over ADSL.

SIP and connectivity packages

Many ITSPs also offer the IP connectivity element, especially the ITSPs that are aimed at the business market who want to ensure end to end call quality. They will typically offer "voice grade" or "business quality" Ethernet, SDSL or increasingly common ADSL Annex M lines. These are often lines with very low or 1:1 contention ratios and SLA guarantees on packet loss, delay and jitter – all the elements that are important to ensure good voice quality.

"IP connectivity from the ITSP has two distinct advantages."

A decision has to be made as to whether to take this IP connectivity or arrange your own. The two advantages of taking IP connectivity from the ITSP is that there is a single point of contact if things go wrong for the whole service and the connection will be directly into the ITSPs network normally avoiding sending the traffic over third parties network.

Aside from cost, the main reason not to take IP connectivity from an ITSP is that you already have internet connectivity that you want to share for voice and data.

If you intend using the same circuit for voice and data it is vital to ensure that Quality of Service queuing policy is configured on your router to give voice priority over data. You should also test beforehand what the latency is between your existing ISP and the ITSP at different times of the day. If the service provider is reselling SIP trunks not on their network then they should at least have a direct IP peering interconnect with their upstream ITSP.

Some ITSPs claim that many customers happily make calls over their normal internet bandwidth without any issues. This can be a very hit and miss affair and the difficulty is in getting issues resolved if problems do arise. The quality of your IP connectivity is going to have a direct effect on the quality of your voice.

Hybrid approach to SIP trunks and ISDN

Consideration should be given to whether SIP trunks are going to be a total replacement for ISDN lines or part of a hybrid approach. Initially it was assumed that people would mainly use SIP trunks for outbound calls to save on call costs and retain ISDN for inbound calls. However many companies have done the opposite so they can deliver geographic numbers to a single PBX and retain ISDN lines for outbound calls. Others have adopted more complex scenarios for disaster failover purposes.

If you are considering the flexibility of the hybrid approach find out whether the ITSP can fail over inbound calls from ISDN to SIP trunk.

Cost

The caveat, as with many things in life, is that the cheapest is not always the best in terms of quality of calls and support. However cost remains a powerful driver in moving to SIP trunks.

Most ITSPs aimed at business users charge for SIP trunks based on a fixed monthly cost per available SIP trunk channel and then call costs per minute for calls made. The idea of a "SIP trunk channel" may seem like an abstract one when considering that SIP is designed for initiating end to end media sessions over the Internet, however it arises from the fact that most traffic will still break out to the PSTN and the network operator will be paying for the cost of physical E1 ports to an interconnect somewhere along the line. It keeps the way of paying for and ordering SIP trunks similar to what people are used to when buying ISDN channels – a flat rental fee and call charges on top.

"IP connectivity is often the 'hidden cost' that gets overlooked"

Remember also part of the complete cost of placing SIP calls will be the IP connectivity to the ITSP which will probably be a separate item and contract to be charged in addition to the SIP trunks or will be taking up internet bandwidth paid for elsewhere – so when making comparisions with ISDN costs or between providers this needs to be taken into account rather than the headline per channel cost. This is often the "hidden cost" that gets overlooked.

Cost

Some ITSPs offer fixed monthly calling plans that may be available to business customers though often aimed at residential users. The obvious point of these bundles is that the operator expects to make money from unused minutes and has in their interest that you are on a package that you will never fully use. Rather like "unlimited usage broadband" offerings expect to find a cap on usage somewhere in the small print.

Free "on net" calls, that is calls to customers on the same ITSP network, is generally of limited benefit as most SIP trunk providers are individual Islands of connectivity and the industry is yet to see mass consolidation with large dominant vendors. In a consumer market you may be able to get friends to join your network to save call charges but you are hardly likely to get or even spend time trying to get all your suppliers and customers to join you on your ITSP's network.

The cost of DDI numbers for inbound calls is often an additional extra charge with additional charges sometimes made for easy to remember or so called "golden numbers".

Summary

Choosing the most suitable ITSP for your particular requirements is key to a successful SIP trunk deployment and leveraging the benefits of SIP trunks for your business.

Break down your requirements into defined points and armed with this information research available suppliers before committing to any contracts:

- Is your internal voice system compatible with the ITSP?
- Is the ITSP responsible for the SIP trunking end to end?
- Can the ITSP transfer your current number range?
- Will the SIP trunk work with your current IP network/firewall?
- Has the ITSP got 999 ability?
- What's the ITSP's business history and stability?
- What fraud prevention/mitigation is offered?
- What's the ITSP's dispute resolution procedure?
- What's in the small print regarding voice quality?

About the author

Graham Lewis MSc MIET MBCS is Director of IP Engineering at Spitfire Network Services Ltd a leading UK ITSP.

He is responsible for the technical implementation and management of Spitfire's SIP trunking service. He is a member of the British Computer Society and the Institution of Engineering and Technology.