SIP trunking for business voice

yes

Learn about SIP Trunking, the benefits in may bring and things to consider

While most organisations understand the value of Voice over IP technology many are still connecting their primary voice communications systems using legacy approaches. The time is right to consider new technologies that can help rationalise disparate legacy networks and reduce costs – an approach that can also better position your network for productivity enhancements such as unified communications and multimedia. This paper explains this newer approach called SIP trunking, the benefits it may bring, and some things to consider for introducing it as a gradual evolution or an immediate upgrade for your business.

What is SIP trunking?

SIP trunking is a protocol for making voice calls over a data network, and when combined with IP PBXs and IP VPNs has the ability to transform the way businesses handle telephone calls. SIP stands for 'Session Initiation Protocol', and is an international standards-based protocol defined by the Internet Engineering Task Force (IETF). SIP has further capabilities for multimedia applications such as video and unified communications, but for this discussion we will look at the use of SIP trunking to connect an organisation's IP PBX to the Public Switched Telephone Network (PSTN) to make and receive telephone calls.

With SIP trunking organisations may be able to:

- Replace ISDN and analogue telephone lines with a simpler and more cost effective alternative.
- Centralise IP PBXs to reduce equipment and carriage costs, and provide a more consistent experience to end users.
- Converge voice and data networks which may reduce costs and make more efficient use of bandwidth.

How does SIP trunking work?

SIP trunking carries telephone calls from an organisation's IP PBX to an internet telephony service provider using an internet or IP connection, and provides connection to the PSTN from the service provider – allowing telephone calls to be made and received.
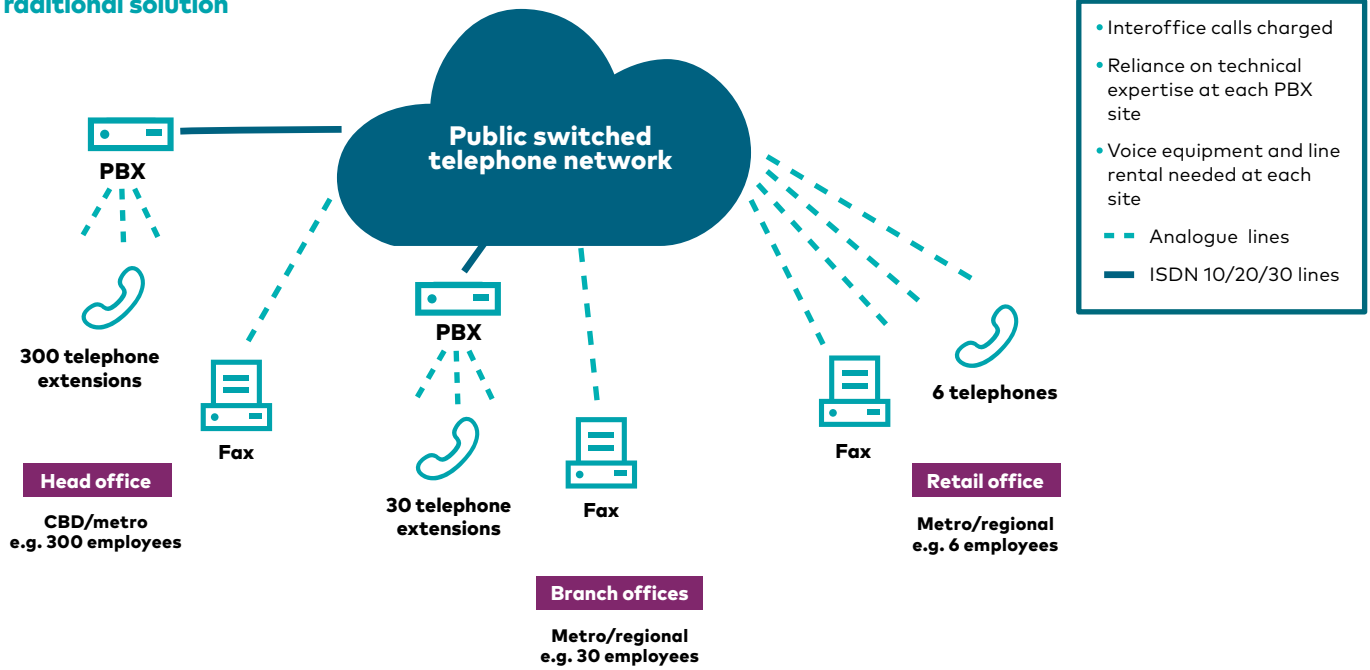
The IP connection can also be used to carry data traffic if desired, provided quality of service (QoS) is implemented on the IP circuit to ensure that voice traffic has priority. SIP can also be configured to transmit faxes, EFTPOS transactions or even dial up modem calls in addition to voice calls.

Calls are encoded as IP data packets using a selected codec, where some codecs can compress calls to reduce the bandwidth required to carry the call (eg G.729), or can provide uncompressed high quality voice (eg G.711).

The service provider uses a softswitch to connect calls to the PSTN, and may use a Session Border Controller (SBC) to mediate the progression of calls between the organisation's IP PBX and the service provider's softswitch. The organisation may also choose to implement an SBC with their IP PBX, depending on the recommendations of their IP PBX vendor.

The SIP protocol is defined in the IETF document RFC3261, and contains many options for signalling and managing calls. Each service provider will specify the detailed implementation of SIP on their platform, allowing organisations to configure their IP PBX correctly and ensure interworking between the IP PBX, SBCs and softswitch.

OPTUS

Traditional solution

- Interoffice calls charged
- Reliance on technical expertise at each PBX site
- Voice equipment and line rental needed at each site
- – – Analogue lines
- —— ISDN 10/20/30 lines

PBX

300 telephone extensions

Fax

Head office

CBD/metro
e.g. 300 employees

Public switched telephone network

PBX

30 telephone extensions

Fax

Branch offices

Metro/regional
e.g. 30 employees

Fax

6 telephones

Retail office

Metro/regional
e.g. 6 employees

SIP trunking as an ISDN replacement

At the most basic level SIP trunking can be used as a direct replacement for existing ISDN voice, as both SIP trunking and ISDN voice have similar call handling features.

Using SIP trunking instead of ISDN will require a different interface on the organisation's PBX, so the most logical time to start a move to SIP is often when one of the organisation's existing ISDN PBXs reaches end of life and is replaced with a new SIP enabled PBX.
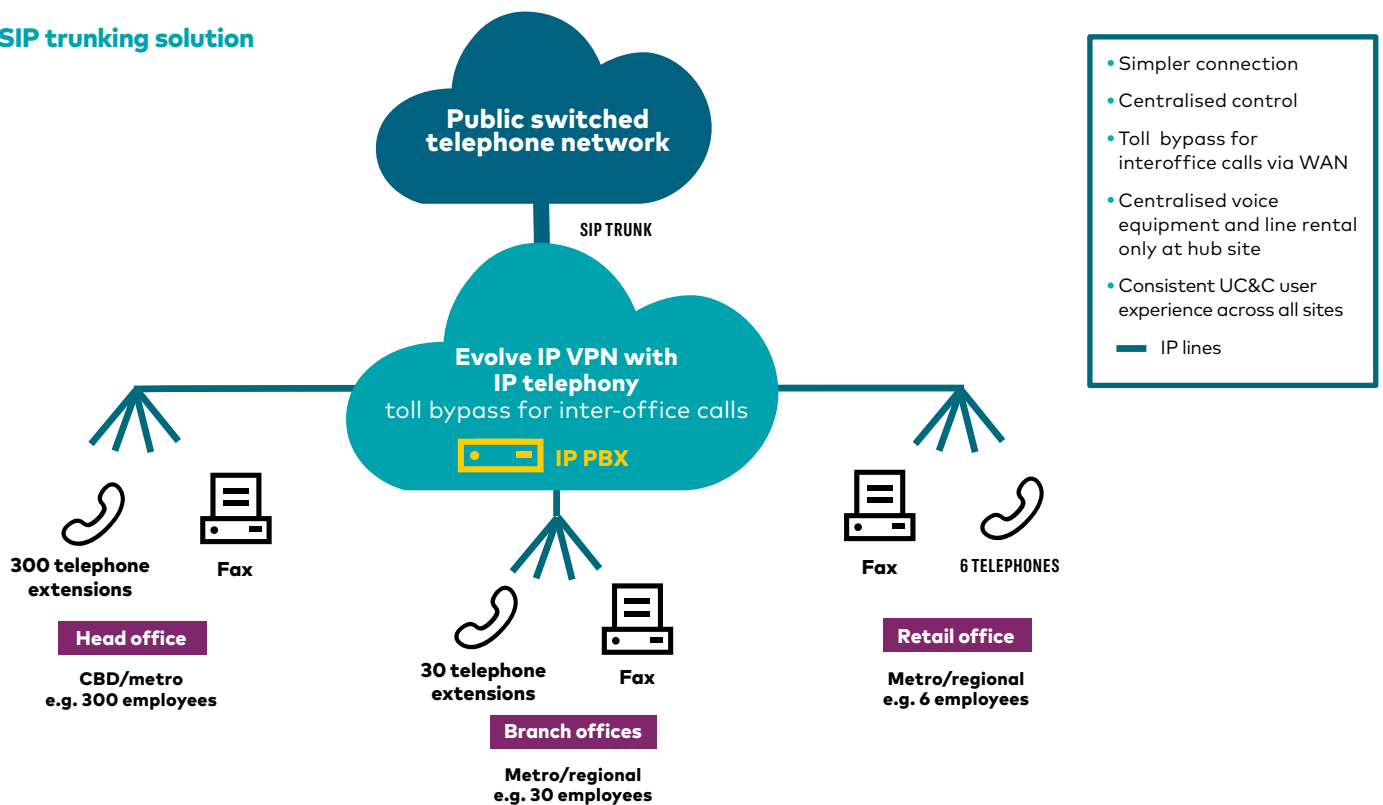
Using SIP trunking as an alternate to ISDN on a PBX can give direct benefits of:

- Simpler connection to the PBX, particularly if the PBX uses more than 30 channels – as a single SIP connection can be used for thousands

of channels if necessary, whereas an ISDN connection will require a separate physical connection (E1) for every 30 channels. For example a 300 channel service will require 10 physical E1 services with ISDN, but only one Ethernet connection with SIP.

- Potentially lower interface costs on the PBX, depending on the relative costs of SIP and/or SBC licenses (where a license is typically required on the PBX for each SIP channel) compared to the cost of E1 ports or cards on an ISDN PBX (where an E1 port is required for each 30 channels).

- The ability to converge voice and data networks to achieve further simplification and cost reduction across an organisation's network, as explained in the next section.

SIP trunking solution

- Simpler connection
- Centralised control
- Toll bypass for interoffice calls via WAN
- Centralised voice equipment and line rental only at hub site
- Consistent UC&C user experience across all sites
- —— IP lines

Public switched telephone network

SIP TRUNK

Evolve IP VPN with IP telephony
toll bypass for inter-office calls

IP PBX

300 telephone extensions

Fax

Head office

CBD/metro
e.g. 300 employees

30 telephone extensions

Fax

Branch offices

Metro/regional
e.g. 30 employees

Fax

6 TELEPHONES

Retail office

Metro/regional
e.g. 6 employees

Converging voice and data networks with SIP trunking

SIP trunking opens up the option for an organisation's voice and data traffic to be carried over a single private IP virtual private network (VPN), rather than having separate ISDN or analogue voice lines and a data-only IP VPN. This convergence can deliver benefits in reducing overall line rental and bandwidth costs, and in simplifying maintenance by having a single network.

A converged IP VPN circuit is generally configured with a maximum number of voice channels and a maximum total bandwidth, and this bandwidth can be dynamically shared between voice and data traffic. Quality of service (QoS) must be implemented on the IP circuit to ensure that voice traffic has priority – but if fewer voice calls are in progress then more bandwidth can be used for data traffic. This can be of particular benefit at sites where peak voice traffic and peak data utilisation are at different times of day, where the same bandwidth can be used for both peaks.

Convergence can also be particularly cost effective at sites where existing IP VPN data bandwidth is under-utilised, where voice channels can be included with no need to expand bandwidth.

Centralising business voice networks with SIP trunking

Combining SIP trunking with IP telephony to centralise an organisation's PBXs can transform the organisation's voice network, and can help deliver major cost and productivity improvements well in excess of the incremental savings discussed so far.

Some SIP trunking providers offer the ability for telephone numbers for anywhere in a state or country to be allocated to a single physical service, which means that an organisation can then put all of its telephone numbers onto a single carriage service and a single PBX. Incoming and outgoing calls can then be delivered from this centralised PBX to each of the organisation's sites using a QoS enabled IP VPN between the sites. Voice, fax, EFTPOS and other call types can be handled through this centralised solution.

This solution can eliminate the need for ISDN or analogue lines at individual sites, and line rental cost savings can often be dramatic as a number of low usage analogue lines at individual sites can be replaced by a single IP trunking channel at the centralised site. For example, it may be possible to replace ten analogue lines spread across a number of sites with line rental of $30-40 per line per month (total $300-$400 per month) with a single centralised IP trunking channel at less than $10 per month.

Centralisation can also eliminate the need for PBXs and PBX maintenance at each site, with the potential for reduced capital costs associated with the purchase of PBXs, reduced maintenance costs with fewer PBXs to maintain, and faster restoration of any issues with the PBXs located at major sites. Organisations will often reduce to two PBXs to ensure resilience and reliability, and this can still deliver significant savings compared to a multisite solution with multiple PBXs.

In summary, the benefits of a centralised SIP trunking/PBX solution can include:

- Significantly reduced line rental costs, through reduction in line counts and reduction in rental rates.
- Reduced calls costs, with the ability to make calls between sites without incurring PSTN call charges.
- Reduced capital and maintenance costs for PBXs.
- A more consistent experience for end users across the organisation, with the same PBX and associated unified communications features used at each site.

Ensuring avaibility with resilience and redundancy options

One potential concern with converging voice and data services and centralising PBXs is the risk of a single equipment or service failure causing an outage across the entire organisation, if all voice and data is carried over a single service and/or all calls are managed by a single PBX.

For high availability sites this risk can be effectively managed by using a range of resiliency and redundancy options, while still gaining overall benefits of cost reduction, simplification and end user experience when compared to a discrete voice and data solution with distributed PBXs.

Some commonly used resiliency and redundancy options are:

- Dual carriage links, with two physical access links delivering parallel IP VPNs to a site, either with both links active or with an active primary link and a secondary link on standby. The same telephone numbers can be associated with both links to ensure that calls can be carried, even if one link is down.

- Dual carrier voice switches, where each of the dual carriage links connect to a different carrier switch – so that a link remains active even if there is an outage at one carrier switch.

- Dual centralised PBXs, located at major sites or at datacentres, so that calls can still be carried with the loss of one PBX or one site. Again the same telephone numbers can be associated with both PBXs by combining dual PBXs with dual carriage links, even if the two PBXs are at different sites.

SIP trunking with Optus

Optus has been providing business grade SIP trunking services since 2008, with implementations ranging from small offices and agencies to major banks, nationwide retailers and government departments. As a Tier 1 carrier, we have observed that this technology is now widely accepted for business voice services.

If you are ready to explore SIP trunking for your business, our experienced voice specialists can help you introduce SIP trunking into your existing communications environment or design, build and implement a complete new solution.

Contact us today to get started.

OPTUS | 1800 555 937 | optus.com.au/business | @optusbusiness | yesopt.us/blog | bit.ly/OBLinkedIn

GMO_N 0405 0816MZ