Technology solutions for business challenges - SIP calls over internet - blueAPACHE

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How does placing SIP calls over the internet affect its quality? In order to answer this, we must first understand how SIP trunking works. With SIP trunks (Session Initiation Protocol or IP based phone and messaging), voice is first converted into IP packets. These IP packets are then sent to the SIP provider and transmitted over a data service (or virtual phone lines) instead of a traditional physical phone line. Voice is real-time sensitive. In order for SIP trunks to be a reliable service, the voice packets need to arrive within a certain amount of time and in the right order (along with a number of other factors like jitter, packet loss etc.). If some voice packets are dropped along the way or if they are sent too slowly, noticeable audio quality issues will occur. The quality of a SIP call is influenced by many factors some of which are explained below.

Contention

Contention occurs when shared internet bandwidth over a single connection causes all internet traffic to slow down during periods of peak usage. Similar to cars on a freeway, when the traffic is light, all packets (or cars) are transmitted at optimum speeds. However, as the volume of packets (or cars) increase beyond available bandwidth, all traffic is forced to slow down. If we consider that voice packets are red cars on the freeway, we will need to ensure that red cars are not slowed down. When this occurs with voice packets, it will affect call quality. In Australia, the average contention ratio of most Internet Service Providers (ISP) is 50:1. What this means is that up to 50 broadband customers are sharing the same bandwidth at any given time. In the previous example, this would be similar to a large group of motorists sharing the same road. Some ISPs may promote a lower contention ratio. However, whilst traveling from the ISP to a SIP Trunk provider, there are various hops along the route where traffic traverses over other ISP networks in order to get to the intended destination. What this means is that it is not just your own internet usage, but other customers of your ISP and even other ISP traffic, that have the ability to affect your voice quality. Contention is often outside your sphere of influence if you are using the internet for sending voice traffic.

Quality of Service

QoS is a feature that allows flagging of voice packets as a higher priority when compared to all other forms of traffic that are sharing the same connection. Without QoS, all traffic on a network is viewed as the same. Thus, streaming a YouTube video would have the same claim to the available bandwidth as an ongoing voice call. Unless you have massive amounts of bandwidth, this is bound to impact voice quality. For QoS to be supported, the three key requirements are:

The ability to mark packets (usually a router that supports QoS)

The ability to honour the markings (all routers from source to destination must support QoS)

The ability of the network itself to support QoS

Essentially, QoS can be guaranteed only when you have full control over both ends of the circuit and everything in between. The public internet does not support QoS. While some providers may claim to provide QoS with their internet service, it is effectively impossible to guarantee voice quality across an internet connection.

Codecs

Codecs are used to compress voice streams and send more over a smaller bandwidth or a small link. Encoding and decoding the voice stream at each end may introduce latency (delays). A loss of voice quality may also result from the use of low quality codecs. While some modern codecs work well over the internet, it still adds a layer of latency. This is because the voice stream has to be encoded by your provider before it is transmitted to your handset and decoded on the way out to the PSTN. The level of latency, and whether it is noticeable or not, largely depends on the codec and the provider's equipment. SIP calls are also significantly impacted by the type of connection between the customer's phone system and the SIP provider. This can be one of few types – over the internet, dedicated internet connection, or a dedicated private MPLS network.

Over the internet

The internet is essentially a network of networks in which all types of data traffic can travel. It is considered a 'best effort' form of delivery because over the internet, voice packets are co-mingling and competing with all other forms of traffic such as video, email and web. It is impossible to guarantee voice quality across an internet connection as public internet does not support QoS.

Dedicated internet connection

A dedicated internet connection between the customer network and SIP provider is generally considered to be a better method for delivering SIP trunks. However, if that connection is going to be used to send other types of traffic such as video or email, there has to be a way for voice traffic to be prioritised over all other traffic. Having a dedicated connection for voice traffic would solve the problem of contention at the local network level (not at the ISP level), and although this is a better solution, QoS still cannot be ensured and you may encounter performance issues.

Dedicated private MPLS network

The only way to have guaranteed voice quality with SIP trunks is to have a dedicated business grade private MPLS network link provided to your business. While this may be an expensive option, it does address both Contention and QoS. The same link can be used to access the internet and other public or private cloud based services, therefore creating a consolidated network while establishing a high quality SIP trunk service with all its benefits. What is the best sip trunk solution for your business? While there is no one-size-fits-all solution for SIP trunks, the best option for your business will depend on your organisation's size, internet traffic and call volume.

Small business with light phone traffic

In Australia, many small businesses (usually 5 or less trunks) or branch offices provision their SIP trunks over the same broadband internet connection as standard internet traffic. When voice packets share bandwidth with internet traffic, the volume of internet traffic and the bandwidth available become critical factors that determine quality of voice calls. This also happens at different times of the day based on internet usage. If internet and voice traffic are light, a high bandwidth / low cost connection can be a very good solution for small businesses. While you may experience an acceptable grade of

service most of the time, during periods of peak usage you run the risk of dropped packets and delayed transmission which will result in a noticeably poor quality of service for your end users and customers.

Small to medium-sized business with moderate phone traffic

Businesses that need over 5 trunks and have moderate phone traffic will benefit by having a dedicated private network connection. This can be provided using a private MPLS link which supports QoS and eliminates contention. If that does not suit your business, the next best type of connection would be a dedicated internet connection for voice. Most call quality issues arise in the last mile or the connection between the customer and ISP. Keeping standard internet traffic and voice traffic separate will resolve the issue of voice packets having to compete for the same bandwidth as all other internet traffic.

Medium to large business with large phone traffic

For businesses that experience large phone traffic, the optimal solution would be to install a dedicated connection between their premise and the SIP provider. The same connection can be used to provide internet access and a QoS supported voice solution. Although such a solution could be more expensive than typical DSL or cable connections, it has the added benefits of long distance savings, elimination of copper trunks at remote offices (with SIP, those numbers can port to the corporate office) and better use of bandwidth.

Our emPOWER SIP Trunks advantage

We own the core MPLS network infrastructure and as such we have absolute levels of influence and complete accountability. Unlike resellers and aggregators of third party services, we can change everything and be truly accountable for our services. We also offer you a single point of contact for all issue resolution, without any opportunity for different vendors to blame each other for issues. If there is a problem, we can fix it. Being an internet service provider, hosted PBX service provider and a carriage service provider, we have the broad breadth of services required to ensure that your voice traffic is not routed over a public network if you are on our network. This translates to a more secure, reliable, QoS enabled network for all voice customers All our Voice customers have access to support from our local specialist service centre and to the Voice and Network engineering teams. We empower Voice:

Does not use any low bandwidth codecs from end to end across the SIP core

Can guarantee end to end QoS for all voice traffic within the emPOWER Voice network

Involves active monitoring of voice traffic across the emPOWER voice network to ensure quality

Voice networks are business grade and uncontended

Can be delivered over the Internet if required

Assured technical excellence in all voice and network solutions

For more information on how you can leverage SIP Trunks for a competitive advantage, contact our local Voice Engineering team .

How does SIP trunking convert and send voice traffic?

With SIP trunks (Session Initiation Protocol or IP-based phone and messaging), voice is first converted into IP packets, which are then sent to the SIP provider and transmitted over a data service (or virtual phone lines) instead of a traditional physical phone line.

What factors affect the quality of a SIP call over the internet?

SIP call quality over the internet is influenced by contention, Quality of Service (QoS), and codecs, as well as the type of connection used between the customer's phone system and the SIP provider (internet, dedicated internet connection, or dedicated private MPLS network).

What is contention and how does it affect SIP call quality?

Contention occurs when shared internet bandwidth over a single connection causes all internet traffic to slow down during periods of peak usage, similar to cars on a freeway slowing down as traffic volume increases. In Australia, the average contention ratio of most ISPs is 50:1, meaning up to 50 broadband customers share the same bandwidth at any given time, which can affect voice quality if voice packets are slowed.

Why can't the public internet guarantee Quality of Service (QoS) for SIP calls?

QoS requires three things: the ability to mark packets, the ability to honour those markings across all routers from source to destination, and the ability of the network itself to support QoS. QoS can only be guaranteed when you have full control over both ends of the circuit and everything in between — something the public internet does not support, making it effectively impossible to guarantee voice quality across an internet connection.

How do codecs impact SIP call quality?

Codecs compress voice streams to send more data over a smaller bandwidth or link. Encoding and decoding the voice stream at each end can introduce latency (delays), and low quality codecs can cause a loss of voice quality. Even modern codecs that work well over the internet still add a layer of latency because the voice stream must be encoded before transmission and decoded on the way to the PSTN.

What are the three types of connections used for delivering SIP trunks?

The three types of connections are: over the internet (a 'best effort' delivery where voice packets compete with other traffic), a dedicated internet connection (which reduces local contention but still can't fully guarantee QoS), and a dedicated private MPLS network (which addresses both contention and QoS but can be more expensive).

Which type of connection guarantees voice quality for SIP trunks?

The only way to have guaranteed voice quality with SIP trunks is a dedicated business grade private MPLS network link, which addresses both contention and QoS. The same link can also be used to access the internet and other public or private cloud services, creating a consolidated network.

What SIP trunk solution is recommended for a small business with light phone traffic?

Many small businesses in Australia (usually 5 or fewer trunks) or branch offices provision their SIP trunks over the same broadband internet connection as standard internet traffic. This can work well when internet and voice traffic are light, but risks dropped packets and poor call quality during peak usage periods.

What solution is recommended for small to medium businesses with moderate phone traffic?

Businesses needing over 5 trunks with moderate phone traffic benefit from a dedicated private network connection, ideally a private MPLS link that supports QoS and eliminates contention. If that isn't suitable, the next best option is a dedicated internet connection for voice, since most call quality issues arise in the last mile between the customer and ISP.

What does your emPOWER Voice service offer for SIP trunk customers?

We empower Voice does not use low bandwidth codecs from end to end across the SIP core, can guarantee end-to-end QoS for all voice traffic within the emPOWER Voice network, actively monitors voice traffic to ensure quality, uses business grade and uncontended voice networks, can be delivered over the internet if required, and provides assured technical excellence in voice and network solutions. We own the core MPLS network infrastructure, giving it full accountability, and offer a single point of contact for issue resolution along with support from a local specialist service centre and Voice and Network engineering teams.