TATA COMMUNICATIONS

FIND YOUR IDEAL SIP TRUNKING MATCH

Ten questions to ask your prospective SIP Trunking provider

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INTRO: GET THE MOST FROM YOUR SIP TRUNKING PROVIDER.

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INTRO: GET THE MOST FROM YOUR SIP TRUNKING PROVIDER.

Here's how.

The service a SIP Trunking provider offers can make or break your relationships, so choosing the right one is essential. The pressure's on to make the right choice — but how? How can you really get the measure of a SIP Trunking provider and establish which ones will really serve the particular needs of your organization?

We started to explore this in the ebook, **"The Five things you MUST know about Global Voice over IP (SIP Trunking) Providers"**. This next ebook goes an essential step further. Here's a detailed look at exactly what you should be asking any prospective SIP Trunking provider. Consider this a checklist for sourcing the information you need to identify a provider that's the right fit for your organization.

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AVAILABILITY

If your contact center becomes unavailable for any reason, how will your customers' calls still reach a location with skilled people to handle them?

You need to know how the SIP Trunking provider will route your customers' calls during a failover scenario — and what circumstances would cause such a scenario. Get the SIP Trunking provider to make this clear — and be demanding. Your SIP Trunking provider should allow you to have multiple levels of failover for each of your contact centers. So if your primary contact center is unreachable, you should be able to choose a second, third and even fourth option, so that calls are routed to one of your contact centers with the right skill set to handle your customers' inquiries. Also, make sure you're not restricted to a single geographic region, or that you're charged more for these failover routing scenarios. More on this in the next question.

FAILOVER

Does the SIP Trunking provider have the ability to route failover calls both intra- and inter-regionally?

You need to understand the limitations of the provider's network routing capabilities — and how these limitations would affect your ability to provide services to your customers. Is it possible to establish a failover plan that allows for major catastrophic events — natural or man-made — in a geographic region? Are you limited to only having failover locations in the same geographic region, or can you only select certain regions for your failovers?

This could greatly affect your ability to provide services to your customers, especially at a time when they may need them most.

EXTRA COSTS

Are there any extra costs on either per unit call or per minute basis for these re-routed calls? There shouldn't be.

Check to see if there are any hidden costs for calls affected by a failover scenario. Every SIP Trunking provider handles these scenarios slightly differently; in some cases there's a 're-routing' fee to complete the call to one of the backup contact centers. These fees aren't typical for calls handled within the same geographic region, they're usually only applied when the call is routed to another region.

LAST MILE

How are the last-mile circuits connected to the provider SIP Trunking network: directly or via MPLS?

Most of today's SIP Trunking providers use Multi-Protocol Label Switching (MPLS) to connect their customer to their network. What varies between providers is how these MPLS connections are physically connected to the SIP Trunking infrastructure. There are two main methods: the first is a virtual one—and it should be your preference. In this case the physical MPLS connection has a physical port on the provider's MPLS network and a virtual one to the SIP Trunking switch. The second is to have a direct connection, still based on MPLS, directly to the SIP Trunking switch.

You can make a case for both methods, but in terms of positive functions, the virtual connection to the SIP Trunking switch wins hands down. With a virtual connection you eliminate the single point of failure within the provider's network. So if something were to happen in the providers network, such as a card failure or a rare total switch failure, the provider's network will automatically 'move' this virtual connection to a secondary SIP Trunking switch. This allows your business to continue without any lost calls and maintain your customer service standards.

The same can't be said for a physical connection. Yes, it may save some costs, but it also introduces a single point of failure within the provider's network. As this connection is directly to a card in the SIP Trunking switch, if that card or switch were to fail then you'd be down and unable to process incoming or outgoing calls.

What's more important for your organization: saving money on the cost of connectivity or having the ability for calls to be completed in failover scenarios? It's something to consider.

REGIONS

How is region-to-region routing handled?

Let's say you've established how the SIP Trunking provider handles calls in a failover scenario. Now, what about how they handle calls during normal operations? This is just as important. Is their SIP Trunking network architecture flat or hierarchical? Find out, because this will affect how your calls are routed between geographical regions and potentially add more delay into the voice stream. Here's why:

In a traditional TDM network you have edge switches and tandem switches. Customers are connected to the edge switches, which in turn are connected to the provider's edge switches for regional traffic, and tandem switch for inter-regional traffic. This kind of hierarchical architecture works well for TDM networks as the switches are connected to each other via point-to-point circuits call Inter-Switch Trunks (ISTs). However, it does create potential problems with routing as a call will route through the edge switch, then one or more tandem switches and then to another edge switch for call completion. While this architecture has been around for years, it can be an operational nightmare—particularly when you consider the number of ISTs you'd need to cover the world.

This is one reason why flat architecture is usually a better option. With a flat architecture, all the SIP Trunking switches are connected to each other via dedicated or virtual (MPLS) connections. This eliminates that need for a middleman (tandem switches) as the edge switches all talk to each other. Besides decreasing the operational headaches for the provider that a hierarchical network creates, flat architecture also eliminates the costs of interconnecting all the edge switches with one or more tandem switches.

ETHERNET

Does the provider use Ethernet connectivity for its last mile connections to your premises?

You may wonder why it matters what type of last mile connection you have. In terms of call quality, it doesn't really. But in terms of operations expandability, it matters a great deal.

In a traditional network, the last miles are usually T-1/E-1s, DS-3, STM-1 etc., which have very defined bandwidths. So as a network location's voice needs expand beyond the capabilities of the T-1/E-1, your next expansion is to a DS-3. Going from 1.5 Mbps or 2 Mbps (T-1/E-1) to a full DS-3 (45 Mbps) makes the cost of the last mile circuits much higher. If you have an Ethernet last mile, you can start with a 2 Mbps connection and usually increase in 1 Mbps or 2 Mbps increments. This has a much smaller impact on the last mile cost compared to the T-1/E-1 increase to a DS-3.

Having an Ethernet last mile brings other advantages, too: there's no need to switch out circuits when an increase is needed, it's quicker to implement the next bandwidth increase, and the cost associated with the equipment that terminates the respective circuits types is also lower.

TOLL FREE

In what geographic locations can the provider sell you toll and toll-free access for your contact centers?

If your business has contact centers (help desk, telephony sales centers, etc.) you want to ensure you can give your customers a number that's either toll-free or local. With this in mind, make sure the SIP Trunking provider can provide services like **International Toll Free Services (ITFS)** and Local Number Service (LNS), and that these countries/cities they're able to provide these for, align with your current and future business needs. If the best SIP Trunking provider in the world can't provide phone numbers in the countries where you do business, it's of no use to you.

PARTNERS

Who are the partners the provider works with in each of the countries where you need voice services?

No matter what a SIP Trunking provider tells you, they work with partners to deliver your calling services. No single provider has the ability to own the connection into every home and business worldwide, so find out who their partners are around the globe.

There are over 1,600 providers of telephony services globally, but only about 100 top level or Tier-1 carriers, so SIP Trunking providers have plenty of choice in how your calls are routed to their final destination. With this in mind, find out whether your prospective SIP Trunking provider uses a Tier-1 carrier as the primary partner in most (if not all) of the countries where they provide inbound and outbound calling services. Do they have different levels of services for enterprise customers versus carrier customers? They should: enterprise customers should enjoy the highest level of quality, without the provider using Lowest Cost Routing (LCR) to deliver your call to its final destination.

UTILIZATION

What are the circuit utilization metrics the provider uses for both customer access and trunk circuits?

Just as you would ask your data provider about their traffic capacity management policies, ask the same questions of a SIP Trunking provider. It's important to understand exactly how they monitor traffic, and at what intervals. Also, find out the utilization percentages that place a circuit on the 'watch list', and what triggers the movement from the watch list to the upgrade list.

A number of different criteria cause these triggers to become active, and lower cost providers tend to allow higher circuit utilization than their more expensive counterparts. It's also worth finding out the measurement period as this can affect the ability of your calls to be completed. A carrier that averaged 75% over a 1-hour measurement period may sound like it's performing better than the carrier that averaged 95% utilization over a 5-minute measurement period. What you don't see is that the carrier averaging 75% hit 100% utilization over a 10-minute period during that 1-hour. So for 10 minutes, no calls were completed. Compare this to the provider who hit 95% for 5 minutes, but who may only have had a minute or two of 100% utilization.

In other words, it's not enough just to consider utilization; you need to look into the measurement period as well. And ultimately, you want a SIP Trunking provider with key monitoring systems and KPIs in place, both for connectivity and call quality. Learn more from **this short video**.

OUTAGE

In the event of a failure, how is the outage handled and from which locations?

Despite the best effort of any provider, outages do happen. What matters is how they're handled. Any SIP Trunking provider can say they provide support for the outages, but get them to tell you exactly how they're handled, and by whom.

All providers today offer services on a 24/7/365 basis, but are your troubles being handled by an operations center that specializes in enterprise voice, or is it the same center that handles the provider's data network as well? What skill sets are available at these operations centers, especially the people who take that first call? Those people should have training in both data and voice services, have the ability to diagnose what the issue is and in most of the cases (optimally over 80%), resolve the issue during that first call.

Understanding how calls are handled by the provider and who is going to answer that phone call is just as important as having automatic re-routing capability in the network—they both affect how your service is going to work.

IN BRIEF, YOUR IDEAL SIP TRUNKING PROVIDER SHOULD:

- ☑ allow you to have multiple levels of failover for each of your contact centers
- ☑ allow you to establish a failover plan that allows for major catastrophic events, with the option of failover locations in more than one geographical region
- ☑ manage a failover scenario, without charging you extra for rerouting
- ☑ use a virtual connection from the MPLS network to the SIP Trunking switch to practically eliminate call failures
- ☑ use a flat architecture for a smoother, lower cost operation
- ☑ use Ethernet for the last mile connection
- ☑ provide toll and toll-free access for your contact centers, wherever they are
- ☑ use a Tier-1 carrier as their primary partner in most or all of the countries where they provide inbound and outbound services
- ☑ offer enterprise customers the highest level of quality
- ☑ keep utilization levels consistently moderate — and monitor these regularly
- ☑ handle outages by offering skilled support from highly trained people.

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ABOUT TATA COMMUNICATIONS

Tata Communications Limited along with its subsidiaries (Tata Communications) is a leading global provider of A New World of Communications™. With a leadership position in emerging markets, Tata Communications leverages its advanced solutions capabilities and domain expertise across its global and pan-India network to deliver managed solutions to multi-national enterprises, service providers and Indian consumers.

UNIFIED COMMUNICATIONS—ON YOUR TERMS

With our global infrastructure, rich managed services portfolio, best-in-class network quality, and flexible terms, we can meet all the current and future needs of your enterprise. We offer modular solutions for both customers and partners that enable reliable and scalable global voice, video, unified conferencing, managed services, cloud contact center and real time communication APIs, with unparalleled interoperability and flexibility.
Uniquely positioned in key growth markets, we are the only telecommunications company with a strategic presence across six continents, the world's largest wholly-owned subsea fiber cable network, and data center space across 44 locations worldwide. Wherever you want to go with UC, count on us to get you there.

Any platform. Any way. Any time. Anywhere.

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