Choosing SIP Providers for VoipNow

To offer hosted PBX services, VoipNow providers must rely on good quality call origination and termination services. Choosing the right provider for these services is a vital requirement for delivering high-quality VoIP, Hosted PBX and Unified Communications services.

Overview

To ensure they can rely on stable call origination and termination services, providers will need to take into consideration the availability and redundancy of the SIP services, codecs, the latency between VoipNow and the SIP provider's switch, termination costs, and all provided capabilities (T38, E911, CLI).

Before using VoipNow in operations, it is recommended to perform an Interop Test, which validates the compatibility between the VoipNow platform and the SIP provider's infrastructure. To ensure redundancy and cost-effectiveness, it is also recommended to have several options for call routing (multiple connections with different SIP providers) as well as to use VoipNow's Best Cost Routing feature.

SIP Providers

Choosing the right SIP provider(s) is a complex process, with well established steps. It is recommended to select a local SIP provider (carrier or telco offering SIP trunking services) for routing calls locally and eventually for providing local DIDs. Also, choose two or several SIP providers of A to Z termination; the latter will act as backup for the local SIP provider and will route international calls.

SIP providers are often selected based on their reputation. It is advisable to do a lot of research and ask for references before making a choice, because there are many fraud cases in the industry. Following the initial research, if the references are positive, you can proceed to interoperability testing. Some SIP providers allow you to open an account on their testing platform and do Interop tests that monitor quality parameters such as ASR (Average Success Rate), ACD (Average Call Duration), and MOS (Mean Opinion Score) for calls to various destinations (area codes). Other SIP providers offer several types of traffic quality, on "Standard Routes" or "Premium Routes", which give insights on various parameters. Generally speaking, standard routes are wholesale oriented as they are cheaper; Premium routes are designed for high-end retail traffic (higher ACD and ASR, CLI presentation).

To be able to send traffic through these routes, SIP providers require that a tech-prefix is part of the dialed number, depending on the chosen route. For instance, standard routes have tech-prefix 90, whereas premium routes have tech-prefix 99. Since standard routes have different costs from premium routes, for a better control over costs, you should create two separate SIP trunks in VoipNow; each SIP trunk (channel) should have its own costs and tech-prefix. For this purpose, VoipNow has a separate field that can be set when creating the SIP channel and its tech prefix.

Authentication

There are two ways to authenticate SIP traffic in the provider's switch: one way is to use access credentials (username and password), the other one uses the IP address of the VoipNow server. Ask the SIP provider you are performing Interop testing with about the authentication method they are using and make the necessary SIP trunk settings.

Costs

With local SIP providers, you can negotiate termination costs for national calls that apply for a longer period of time. Because of the dynamics of the market, SIP providers offering A to Z termination tend to change their rates frequently, even as often as once a week. Also, it's important to take into account the biling increment (the way in which the call cost is calculated): 60/1, 60/60, 6/6, 1/1. The gold standard in wholesales is 1/1 (billing per second starting with the first second), but there may be exceptions.

Providers also need to consider the currency they are working with. It is advisable to have a single currency when you work with SIP providers (for instance, if your prices are in Euros, then ensure that the SIP provider's costs are also in Euros), so that you are not affected by exchange rate fluctuations.

Also, ensure that your SIP provider allows you to download the CDRs (Call Details Records) for the calls made from the control panel and/or compare them to the CDRs from VoipNow to ensure there is no inconsistency (call duration, call costs). If there are any differences between the CDRs at the level of call duration, they are most likely linked to inconsistencies in SIP signaling - the provider's softswitch does not make a correct interpretation of the BYE signal, which signifies that the call has ended, and continues to track the call as active.

Cost Management

Generally speaking, SIP providers update their price lists for each destination (area codes) within a specific time-frame. The current price lists can be downloaded from their website or from the web control panel of the SIP provider where you will log in as user (client). Most of these lists (xls. or csv. files) contain the destinations (area codes) in the ITU E164 defined format (international public telecommunication numbering plan); this is also the recommended format for area codes when uploading costs on each VoipNow channel. If the numbers are in a format other than E164 (for instance, local numbers without country code or international numbers with +, 00, or 011 prefix), such numbers will be normalized throu gh algorithms such as "Process Rules" in the VoipNow's "Outgoing Routing Rules" area.

Codecs

Codecs vary from one provider to another. The most frequently used codecs are G.711 uLaw, aLaw and G.729a (used when the bandwidth is a constraint). SIP providers also use G.723 or G.726. Ask your SIP providers about the codecs they use and set the SIP channel accordingly.

Fallback / Redundancy

Interconnecting with several SIP providers at the same time comes with a lot of advantages. If at some point there is an attempt at terminating a call and the SIP provider responds with an error (server error, timed out connection, traffic congestion, busy, insufficient credit available, etc.), the call can be automatically cascaded to other SIP channels, as predefined in VoipNow's "Outgoing Routing Rules" area.

With VoipNow's "Best Cost Routing" rule, you can automatically select the SIP provider that offers the lowest cost for a certain area code (destination). If the channel responds with a SIP error (busy, congestion, etc.), the next cost-effective SIP channel will be tested, and so on and so forth until the call is placed.

Other Services

T-38 is a standard defined by ITU (International Telecommunication Union) for Fax transmission over IP networks. Certain SIP providers offer routes that are SIP compatible with T38. Use these routes to place Fax calls from the VoipNow server.

CLI - Calling Line Identification - is a service that allows the called party to view the CallerID of the caller. Depending on the quality of the SIP provider's routes, they can be CLI or Non-CLI. Generally, premium routes have CLI, whereas non-CLI routes are usually the "best effort" type, where the CLI service is not guaranteed.

DID Providers

An important stage when configuring the VoipNow server is setting the DIDs that will be allocated to the customers' extensions. The local SIP provider can also be the DID provider; for international DIDs sourcing, there are specialized DID providers.

It's important to analyze the types of services that will be available on these DIDs. If, for instance, they are compatible with Internet Faxing (T38), if they support CNAME services (displaying the name of the caller, this is an additional service to CLI), E911 (localization of emergency calls) or Toll-Free numbers. If you want to have these services, choose VoIP providers that offer DIDs compatible with such capabilities.

The costs of DID lease are also a thing to consider. Some providers request a minimum commitment of up to 12 months, whereas other DID providers do not require such an engagement. There are DID providers that request minimum-activity commitments (leased DID numbers or monthly commitment); others are more flexible. The cost structure for a DID lease includes the setup fee, the monthly lease of the number as well as all related traffic costs (cost per minute for incoming call).

Legal Aspects

The lease of international DIDs refers to the origination of calls from a specific geographic region and the presentation of a CLI from a different location, which may trigger certain legal implications. Some countries have restrictive policies in this regard and ask for documents that certify the identity and the location of the person requesting the DIDs with geographical location from the respective country (for instance, Germany), whereas other countries have a more flexible legislation. Please check these legal aspects with your DID provider before making any purchase.

Toll Free Numbers

The calls to a Toll Free number are free of charge for the callers; the owners of Toll Free numbers or the person who leases them pay for the costs of such calls. Generally, Toll Free numbers can only be called from their country of origin. For instance, a Toll Free number from India cannot be called from the US. Depending on the number and its provider, the costs for the calls made from fixed phone lines or mobile phones may differ, which complicates to a great extent the billing of such numbers. When you load a Toll Free number assigned to a SIP channel in VoipNow, apart from filling in the "Monthly amount paid to provider" field, which represents the monthly lease for that number, you must also fill in the "Incoming call cost" with the call cost per minute. Create a separate charging plan for these numbers and make sure it includes the "Charge Incoming Calls" fees.