Researching New SIP Carriers

How to make your own Kerio Operator how-to document for a new SIP carrier

The aim of this document is to provide some generic guidance for the situations when you need to connect the Kerio Operator PBX to a SIP provider for whom no how-to document exists yet or if you are given the task of creating the how-to for the said provider.

The document assumes you are working with Operator 2.4 or newer. Version 2.4 came with several improvements that make the configuration of SIP interfaces simpler and at the same time give the Operator administrator a finer control of some parameters.

Connecting Kerio Operator to a SIP carrier is usually a straightforward process but we will mention some of the possible catches as well. Now and then, you can encounter a SIP carrier who is quite far away from the “Keep it simple!” ideal. The description of complexities tends to be complex as well, hence some parts of this text might be quite long. Here's hoping the document stays useful despite its length.

The explanation herein is divided into three parts. First of all, we focus on SIP servers, Domains, and user names. After that, we add phone numbers to the mix. We conclude with the discussion of call forwarding and non-standard SIP headers.

SIP Servers, Domains, and User names

Let’s start with a brief explanation of how SIP works with identifiers and server names. This should give you an understanding of what information to look for in the documentation of a given SIP provider.

User Names and Domains

The authors of SIP have designed the protocol with the idea that SIP users should be identified by their e-mail addresses, or at least that user IDs should be formatted very much like an e-mail address. To give an example, if Joe wants to call Jane, this would be represented as user joe@somedomain.com calling user jane@somedomain.com or perhaps even jane@anotherdomain.com. This leads us to the first two pieces of SIP terminology: if you take “joe@somedomain.com”, then “joe” is the SIP Username, and “somedomain.com” is the SIP Domain.

Both the Username and Domain can be reused separately throughout the configuration so it’s quite practical to enter them as two strings, instead of a single “joe@somedomain.com” field. (That’s how we do it in Kerio Operator’s administration console.)

In addition to the Username and Domain, there’s another identifier you need to know, and it’s the “Authentication username”. It is used in SIP’s challenge-response authentication. Don’t ask me why you need two user names, maybe it's because the authentication system was borrowed from the HTTP ex post when someone realized that sending passwords in plain text is not a good idea and something better is needed. Most SIP carriers have nevertheless decided to simplify things by keeping the Authentication username the same as the Username.

SIP Servers

Let's discuss SIP servers now. In the most complex situation, a SIP client (and your PBX is such a client) needs to know the addresses of the registration server (its role is to authenticate you and remember the IP address you are available at), and one or two proxy servers. If you have just one proxy, it handles both incoming and outgoing calls. In the other case, one proxy routes your outbound calls, the other one handles your inbound calls.

There are probably SIP carriers out there who ask you to configure all the three servers, or even more, they've got multiple server addresses for each role. Most of the carriers, though, run a high-availability cluster of servers that can fulfill all the three roles, that is registration, inbound proxy, outbound proxy. As far as you are concerned, the whole cluster is represented by a single host name or IP address. And additionally, in the ideal case, the SIP Domain mentioned above and the single server address are identical. If the provider's documentation includes no "Domain" and instead they mention just a single SIP server address, simply enter the server address into the Domain field in Operator and you are done.

Some SIP carriers are trying to simplify the configuration by having SIP SRV records defined for their Internet domain name. This should be in theory an equivalent to having the server cluster hidden behind a single host name but there's a catch. Kerio Operator is using Asterisk as the VoIP engine, and Asterisk (in all its currently existing versions) is only using SIP SRV resolution for outbound calls. It's not using it for inbound calls. The solution is to ask the provider for a list of their proxies or simply look into their DNS records. Then enter the list of proxies in Operator (the SIP interface edit dialog, tab "SIP Details", the field "Inbound proxy").

You can use the command line utility **dig** to fetch the SRV records for a given domain. Suppose you want to find that information for the domain nexvortex.com. The command to run in a Linux terminal is as follows (works on a Mac as well):

```
dig _sip._udp.nexvortex.com SRV
```

Look for the Answer section in the command's output. For nexvortex.com, it looks as follows:

```
_sip._udp.nexvortex.com. 1800   IN    SRV       20 0 5060 px5.nexvortex.com.
_sip._udp.nexvortex.com. 1800   IN    SRV       30 0 5060 px7.nexvortex.com.
_sip._udp.nexvortex.com. 1800   IN    SRV       10 0 5060 px1.nexvortex.com.
```

Hence, your "Inbound proxy" field should contain px5.nexwortex.com, px7.nexvortex.com, px1.nexwortex.com.

Things to Check

So to summarize the above,

- in the ideal case, you will have a single Username (accompanied by a password), and a Domain that is also the only server address you need to configure.
- In not so ideal case, you will have the Username, Domain, Registration server address, and Proxy server address (hopefully this proxy will fulfill both the inbound and outbound proxy roles).
- In an even less ideal case, you will have several IP addresses or host names for each SIP server role.

As another form of a summary, these are the questions you should ask when reading the SIP provider's documentation:

- What is my SIP Username? What is the password associated to that username?
- Is the Authentication username the same as the Username? If they do not mention authentication username in their documentation at all, it is safe to assume the two user names are really the same.
- What is the SIP Domain string they are using?
    - Catch 1: They mention no Domain at all. Simple, dear Watson, it's the case when the Domain and server name are the same.
    - Catch 2: The Domain string may not have the form of the usual domain address. Instead of using something like "mydomain.com", they go for "com.lets.make.it.cryptic.mydomain".
    - Catch 3: They might not be using the term "Domain" at all and instead talk about the "identifier to be used instead of the host name part of the SIP URI". More likely than not, you may find Catch 2 and Catch 3 together, in the same hard-to-read document.
- Do they use the same server name for all the SIP server roles, ie. registration and inbound/outbound proxies? Again, additionally, the simplest case is when the server name and the Domain are identical.
- Are they using several SIP servers?. If so, make a list of all of them.
- Do they rely on SIP SRV for balancing the traffic among multiple proxy servers? If they do, you will need to know the list of proxy servers. Just ask or pull the list from their DNS (the "dig" command-line utility or similar is your friend). If you are not sure you have all the proxies, you will need to try multiple inbound calls. If some of them work and some fail, your list is very likely incomplete.

Phone numbers

As shown above, the situation with SIP servers and user names can be sometimes quite complicated, so we intentionally kept the phone numbers out of the mix. Even though SIP could use e-mail addresses as user identifiers, phone numbers are still important. So let's have a look at how to handle phone numbers in Operator's SIP interfaces.

Number format

The first question you need to ask is, “What format of phone numbers is the SIP provider using?”. Most of them simply use whatever is the national format in their country, for example 9 digits in the Czech Republic (e.g. “377338901”) or 10 digits in the US (e.g. “4084964500”). The carriers in the US may sometimes use 11 digits instead of 10. The 1st digit is then always “1” (“14084964500”).

Some SIP providers are trying to follow the SIP Connect recommendation and hence use the international E.164 number format. This means that the number always starts with the “+” sign followed by the country code, followed by the national number, for example “+420377338901”.

The disadvantage of working with the E.164 format is that you often need to rewrite called numbers to E.164 when calling out. Consider for example that you are in the US and your users are used to dialing “9” to reach outside numbers. If the user dials “94084964500” to call Kerio’s San Jose office, you need to have an outbound call route for the prefix “9” that removes one digit and adds the string “+1” so that the “94084964500” is changed to “+14084964500”. If the same user is used to dialing “9011420377338901” to reach Kerio’s Pilsen office, you need an outbound call route for prefix “9011” that removes four digits and adds the string “+”, thus transforming “9011420377338901” to “+420377338901”.

You might occasionally encounter a custom number format. For example, we know about a SIP carrier in the Netherlands who is using the international format without the leading “+” sign. The solution is the same as above - you will need outbound call routes that translate national and international phone numbers to the format expected by the provider.

The last catch related to number formats is that the format for inbound calls may differ from the format for outbound calls. Let’s use the US as an example: you could encounter a SIP carrier that sends inbound calls to you where your called number is identified with 10 digits but expects you to send outbound calls with the 11-digit format. You can probably already guess what is the solution: use 10-digit format on the inbound side and create custom outbound call routes that translate to the 11-digit format, similar to the treatment of the E.164 above.

Phone numbers as identifiers

When talking about SIP user names above, we have mentioned that the original intention of SIP authors was to use identifiers in the form of e-mail addresses. However, phone numbers are the more usual identifiers when telephone systems are concerned. That’s why many SIP carriers will simply tell you that your chosen phone number is to be used as the SIP username. I’d call this the ideal case, the user only needs to know three pieces of information to configure the SIP interface: the phone number, the SIP password, and the server address.

In this simplest case, you enter the assigned phone number as both the External number and the Username in the External Interface’s tab “General” in the Operator’s administration console.

If you ask the SIP provider for multiple phone numbers, there are two frequent solutions:

1. One of the phone numbers (most likely the first one) is used as the user name, or

2. they give you multiple accounts, each with a single phone number. Then you have to create multiple SIP interfaces in Operator. The technique you can then use for outbound calls is to assign all the interfaces under the same dial-out prefix (for example “9”, you can also use the empty prefix for prefix-less calling).

In special cases, like when you have a group of 100 numbers (say, 5551200 through 5551299), the username can be something unrelated, like a random string, or the common part of your phone numbers (“55512” in our example).

Special use of SIP headers

This section discusses special/non-standard use of SIP headers for call setup and call transfer.

Call Setup

When a device needs to initiate a SIP call, it sends the SIP INVITE request. The beginning of the message is as follows:

```
INVITE sip:13@10.10.1.13 SIP/2.0
Via: SIP/2.0/UDP 10.10.1.99:5060;branch=z9hG4bK343bf628;rport
From: "Test 15" <sip:15@10.10.1.99>tag=as58f4201b
To: <sip:13@10.10.1.13>
…
```

In this example message, the extension number 15 is calling to extension 13. As you can see, the called number appears in the message twice, in the Request start line (the one starting with “INVITE”, a.k.a Request-Line) and in the “To:” header line. The calling number appears in the “From:” header. When parsing the called number, the SIP client uses the Request start line and ignores the “To:” header. This is the most frequent use of the headers and most SIP carriers adhere to it. When you create a new SIP interface in Operator, it is by default configured to adhere to this standard as well.

You have probably guessed by know that the reason we mention the headers here is that some SIP providers behave differently. To work around that, Operator lets you change the default behavior in the tab “SIP Details” in the SIP interface edit dialog.

The calling number can be read from the “From:” header (the default) or alternatively from “P-Asserted-Identity”, “P-Preferred-Identity” or the “Remote-Party-ID” headers. It’s quite unusual that the provider should not use the default header “From:”, though.

As to the called number, it is somewhat more frequent that the providers use the “To:” header instead of the Request start line. To cover even the very strange configurations, Operator lets you parse the called number also from the “P-Asserted-Identity”, “P-Preferred-Identity” or the “Remote-Party-ID” headers in addition to the Request start line and the header “To:”.

An example when the provider might prefer using the “To:” field instead of the Request start line is the case when you have multiple (100) phone numbers and your SIP user name is the common part of the numbers. So if your numbers are 5551200 through 5551299 and the user name is “55512”, the provider might send the string “55512” in the Request start line of your inbound calls and the exact called number (for example “5551234”) is placed in the header “To:”. The only reason they do this is probably that they’ve chosen this approach when implementing their system and it’s not as easy to switch and migrate all the existing user base to a simpler configuration.

The same SIP provider may further complicate the configuration by requiring that your outbound calls should always use the fixed username (“55512”) in the field “From:” and the actual number you are making the call from should be reported for example in “P-Preferred-Identity”. This is relatively easy to achieve, using the configuration of SIP headers for outbound calls in the “SIP Details” tab.

Call transfer

Using special SIP headers for transferred calls is relatively simple. You usually want to notify the party that receives the call about who was the original caller. The best way to do this is with the “Diversion” header, you just enable the switch in “SIP Details” and leave the default value as is. Some SIP carriers expect you to use the “P-Preferred-Identity” or the “Remote-Party-ID” fields instead, so it's best to ask if you are not successful with the “Diversion” field.

Conclusion

Here's hoping this document was useful for you despite the fact it is quite dense.

If you still have issues making your SIP carrier configuration work in Kerio Operator, we conclude this text with several recommendations about what you can try:

- When communicating with the SIP carrier, ask specific questions. “I'm not sure about what is your number format for inbound calls” is better than “It does not work!”
- Ask the carrier about their terminology. Sometimes reading just a few paragraphs of their how-to gives you a strong signal you should ask. Sometimes the catch can be subtle and it may take longer before you realize you need to clarify a detail (“How come your 'E.164' format does not start with a plus sign?”)
- Use the packet sniffer! Kerio Operator has a built-in packet sniffer that can really be a valuable tool. Just open the packet dump in Wireshark and filter for SIP messages, or use the tool under Telephony → VoIP calls. You don't really need to be 'fluent' in SIP to discover what should be the right IP address or the right phone number format by looking at what packets the SIP carrier is sending to you.