Configuration Guide

for the Ingate SBC and E-SBC

SIParator® / Firewall®

Jive Trunking use case

For the Ingate Cloud SIParators using software release 6.2.1 or later

November 2022

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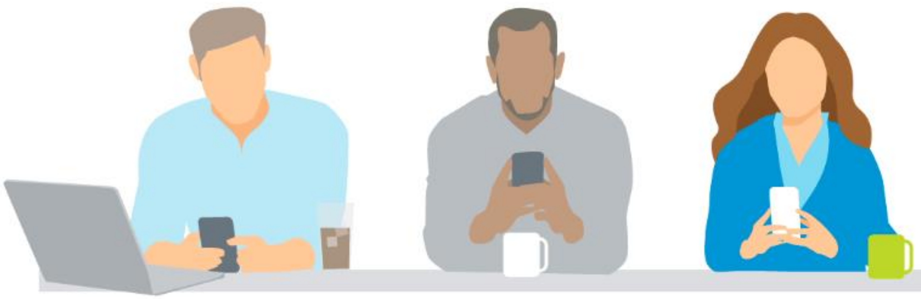
Jive SIP Trunks

Today's workforce doesn't stay in one place—why should your phone system?



Discover the power of Cloud communications.

As a company, you've made a significant capital technology investment in your on-premises communications system and the PBX equipment that powers it. But as your off-premises workforce increases year over year, more and more of your employees can't easily access that system. With Jive SIP Trunks, you can extend the power of your existing PBX to every remote branch office and a workforce on the go. Quickly and easily turn up additional virtual extensions that fit seamlessly into your existing call flow architecture without disrupting your existing PBX. ELiminate the need for one-off, stand-alone solutions in the field and provide employees with a powerful fully integrated mobile communications application.

ANYWHERE ACCESS.

Don't disrupt your premises PBX at your larger locations. Then, add Jive SIP Trunks to seamlessly connect remote offices and road warriors wherever they are into your existing call flow.

FEATURE RICH.

Extend on-premises features, directories, caller ID, DIDs, and E-911 to every system user, regardless of physical location. Include all unlimited inbound and outbound calling.

A UNIFIED EXPERIENCE.

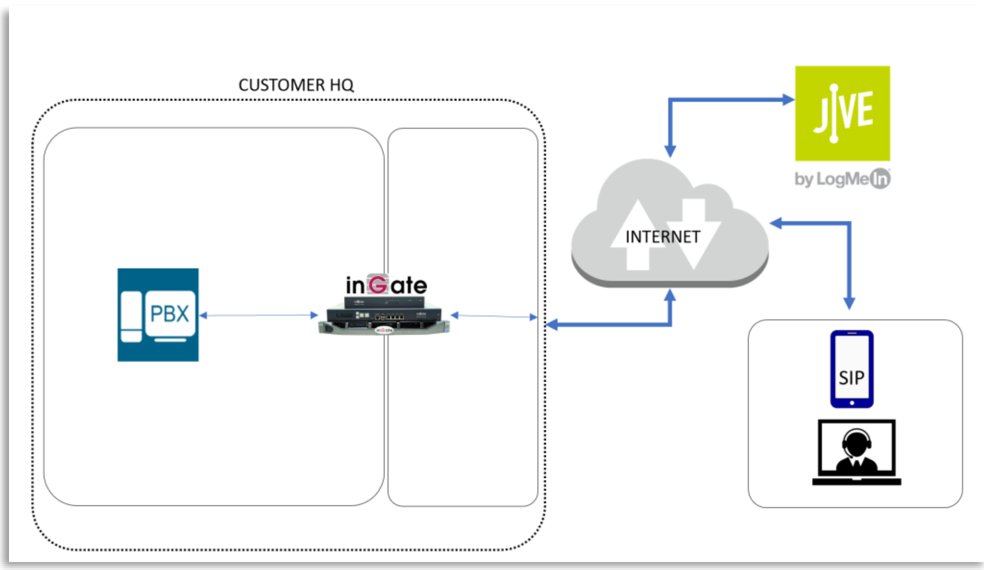
Seamlessly integrate Jive with your existing call control platform, making it possible for remote users to make and receive calls, including extension to extension dialins, like they were in the office.

2 Typical use cases with Ingate SIParator

Jive is a fully hosted voice system (Virtual Service) providing not only the ability to supply all Voice related services including full PSTN connectivity but also integration with customer existing PBX infrastructure (Usually on premise).

When using Ingate SIParator for trunking two use cases are the most typical ones.

2.1 Trunking with IPPBX

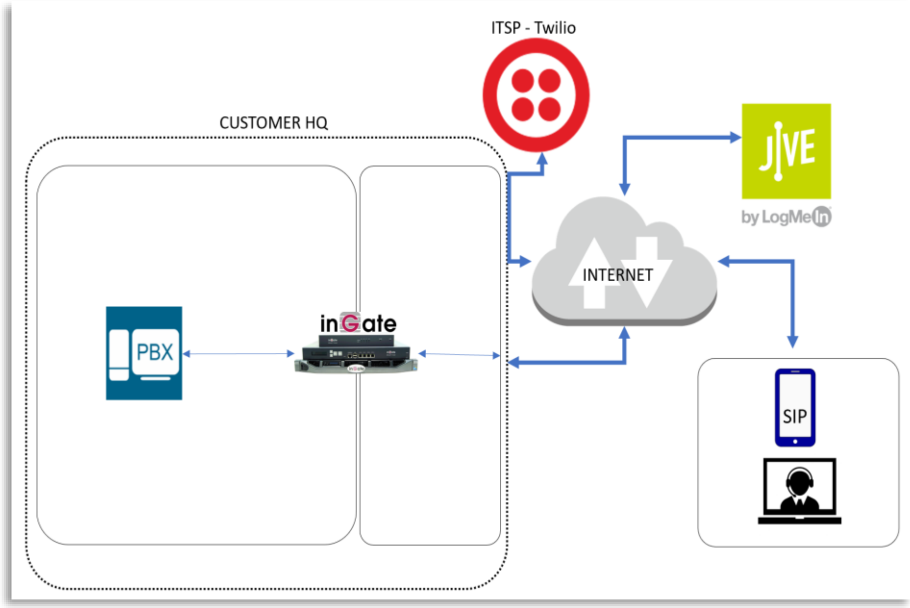


In this case, connectivity will be done using a trunk group registered and pointing to Jive Virtual Service, and connected in the back end with an IPPBX.

Extensions or remote users to the IPPBX will be able to use existing Jive Infrastructure, not only to dial outbound calls to PSTN but also to any user subscribed to Jive Services.

2.2 Customer PSTN existing connectivity integrated with Jive

In this case, a customer with existing IPPBX and PSTN connectivity, when integration with Jive, in addition to the use shown in 2.1 will most likely need to integrated with existing carriers.



In this case, customer with pre-existing SIP trunks with other ITSP's will be able to integrate them with Jive Services.

3 SIParator configuration guidelines for Jive Services

In this section we will present how to integrate with Jive using what Jive defines as a Jive SIP Trunk

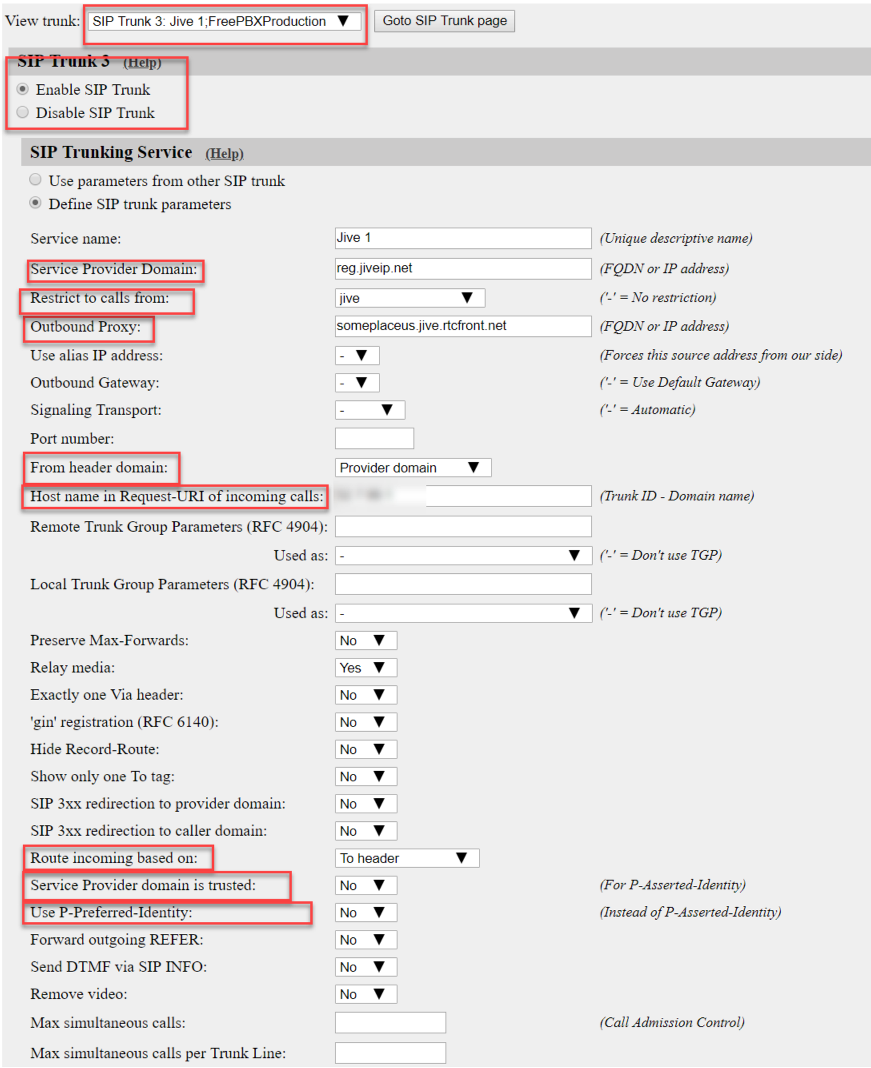
Jive SIP Trunks are provisioned with Credentials (User Name / Password).

Typical data provided by Jive will look like this (Using fake credentials for demo purposes):

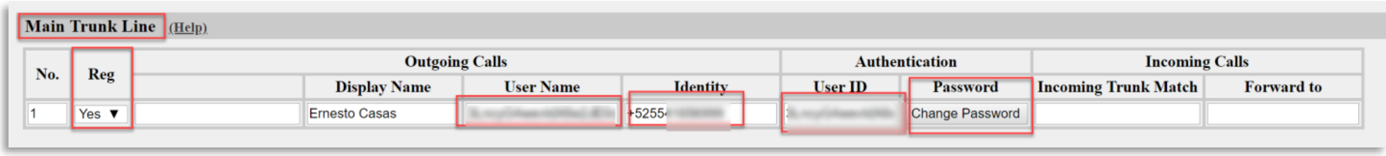
SIP Trunk Extension Number:	6015
SIP User Name:	~~3LncyG4wevtdX6e2JE0qbsedrt52IXK~~
SIP Password:	~~WURoyUGbBtaimmeb~~
Register Server:	reg.jiveip.net
Registrar Port:	5060
Outbound Proxy:	someplaceus.jive.rtcfront.net
OP Port:	5060

3.1 SIP Trunk configuration.

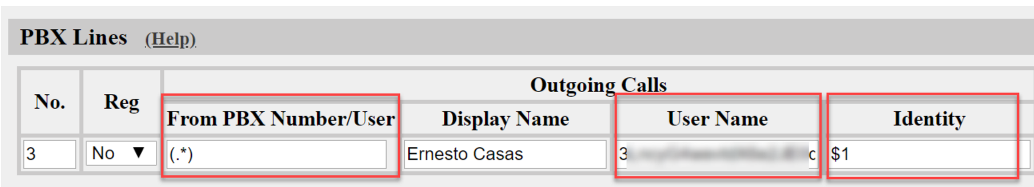
When pointing to Jive Service using a Trunk Page and Jive playing the role of ITSP Trnk Group should look like this:



- Make sure you enable a new SIP Trunk
- Proceed to define sip trunk parameters
- Service Provider Domain will be what Jive calls *"Register Server"*
- Restrict calls from, select the network name explained later as "jive". See section 3.2
- Outbound Proxy will be what Jive calls *"Outbound Proxy"*
- From Header Domain, must be selected as Provider Domain. This will make outbound calls to use Jive Domain as the from header domain.
- Host Name in request URI can be keep blank or just add the public IP address of your SIParator
- Jive uses P-Asserted Identity to identify caller ID for calls inbound to Jive (Outbound in Ingate Trunk). In order to do so, Keep Service Provider Domain as non-trusted, P-Preferred-Identity not selected.



- As Jive Authentication is Credentials based you will need to enable Reg. in the Main Trunk Line
- You need to use Jive's provided User ID as the User Name and User ID in the Main Trunk Line
- Use Jive's provided password to setup the Password when clicking Change Password
- If you will always present the same user information including Caller ID on this trunk to Jive, Populate Identity Field with the value for such DID. If you need to present different DID's and DID's can be provided in the o~~p~~utbound calls from the IPPBX, leave Identity blank and follow instructions in the next section (PBX Lines)



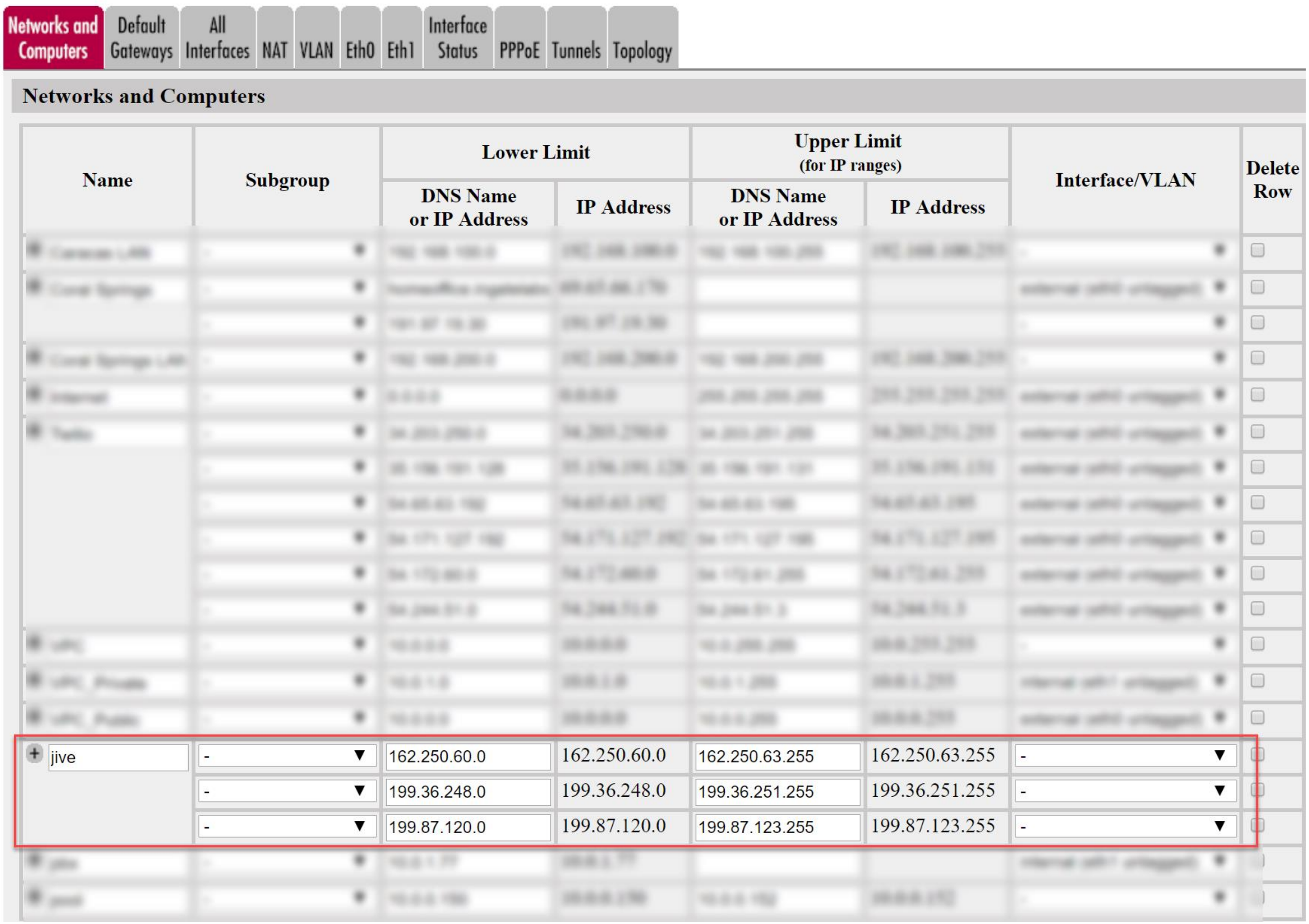
- These outgoing parameters will be used when you want to control Caller ID and other outbound call headers based on IPPBX manipulation.
- The From PBX Number/User fiel you can use regular expressions to capture any section or substring of the User part in the R-uri.
- User Name field must be populated with User Name provided by Jive for this trunk
- Identity can be any expression including a result from capturing in the From PBX number. What needs to be clear is that Jive will require this field in E.164 format.

- In this section you will control how calls coming from Jive to your internal PBX side will be routed.
- The way the user part of the r-uri coming from Jive is usually formatted as <user name><DNIS>.
- Assuming the PBX is configured to route calls for DID, we will need for instance to strip out the user name. That is why we show here as an example capturing anything coming from Jive with the combination of the account username and a DNIS, capture the DNIS and of course prefix will be stripped out.

- In our example we are routing any call matching user name to extension ID 4016 in the PBX, but if instead of using 4016 we use $1, any value captured in the r-uri user will be used as the destination in the PBX.

3.2 Jive Network Valid ip addresses

Jive Services/traffic can be generated (Signaling and media) from a set of known IP addresses. You will need to identify such IP addresses Networks and Subnets to be used in other sections.



4 Additional help or support

If you have questions, suggestions and any other concern feel free to contact Educronix LLC

Web: www.educronix.com

Email: support@educronix.com

Toll-Free: +1 855 866 8854

Ph: +1 954 866 8884

We also provide consulting services as well as remote hands troubleshooting and configuration.