



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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1 Jive SIP Trunk



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.

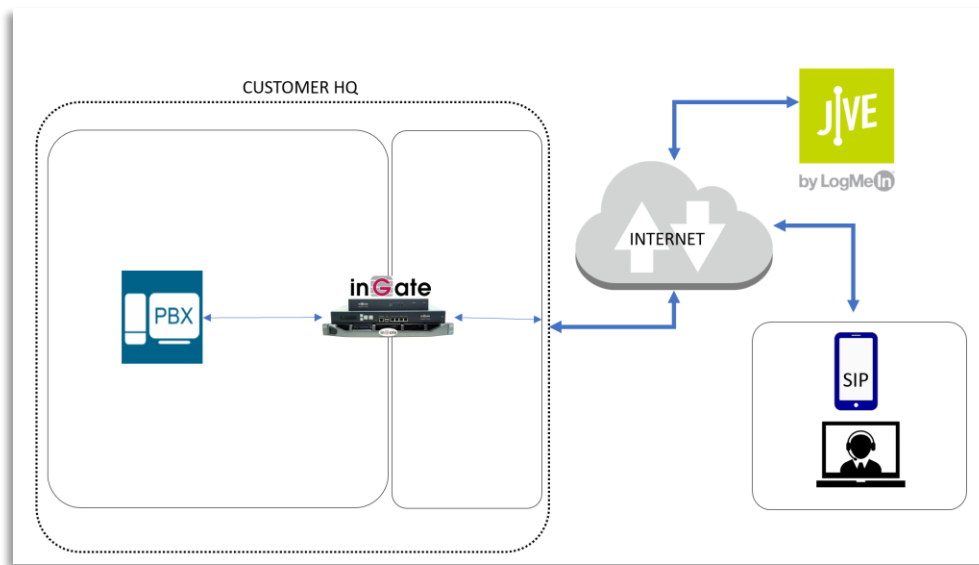
Seamlessly integrate Jive with your existing call control platform, making it possible for remote users to make and receive calls, including extension to extension dialins, like they were in the office.

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

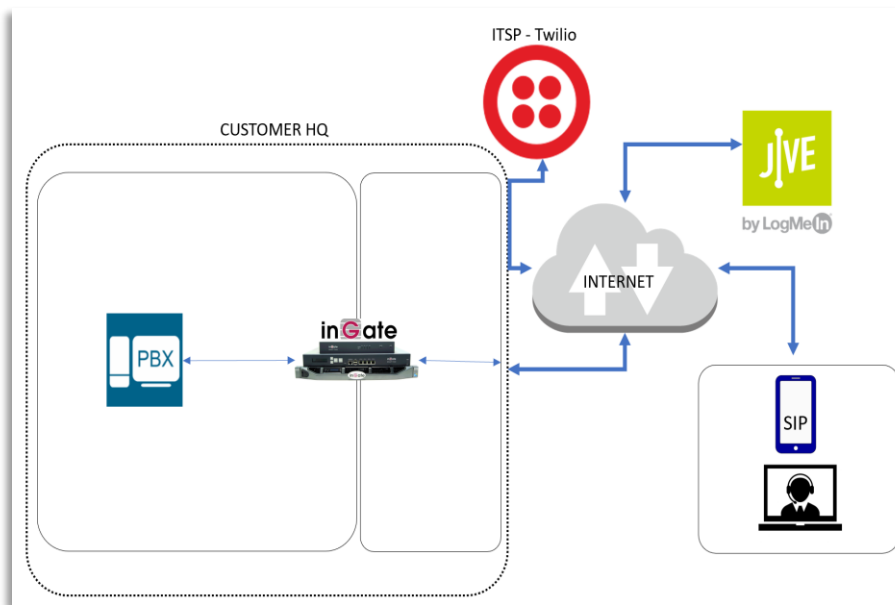


In this case, connectivity will be done using a trunk group registered and pointing to Jive Virtual Service, and connected in the back end with an IPPBX.

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 be 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:	3LneyG4wevtdX6e2JE0qbsedr52LXK
SIP Password:	WURoyUGbBtaimneb
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:

View trunk: SIP Trunk 3: Jive 1;FreePBXProduction Goto SIP Trunk page

SIP Trunk 3 (Help)

Enable SIP Trunk
 Disable SIP Trunk

SIP Trunking Service (Help)

Use parameters from other SIP trunk
 Define SIP trunk parameters

Service name: Jive 1 (Unique descriptive name)
 Service Provider Domain: reg.jiveip.net (FQDN or IP address)
 Restrict to calls from: jive ('.' = No restriction)
 Outbound Proxy: someplaceus.jive.rtcfront.net (FQDN or IP address)
 Use alias IP address: - (Forces this source address from our side)
 Outbound Gateway: - ('.' = Use Default Gateway)
 Signaling Transport: - ('.' = Automatic)
 Port number:
 From header domain: Provider domain (Trunk ID - Domain name)
 Host name in Request-URI of incoming calls:
 Remote Trunk Group Parameters (RFC 4904):
 Used as: - ('.' = Don't use TGP)
 Local Trunk Group Parameters (RFC 4904):
 Used as: - ('.' = Don't use TGP)
 Preserve Max-Forwards: No
 Relay media: Yes
 Exactly one Via header: No
 'gin' registration (RFC 6140): No
 Hide Record-Route: No
 Show only one To tag: No
 SIP 3xx redirection to provider domain: No
 SIP 3xx redirection to caller domain: No
 Route incoming based on: To header
 Service Provider domain is trusted: No (For P-Asserted-Identity)
 Use P-Preferred-Identity: No (Instead of P-Asserted-Identity)
 Forward outgoing REFER: No
 Send DTMF via SIP INFO: No
 Remove video: No
 Max simultaneous calls: (Call Admission Control)
 Max simultaneous calls per Trunk Line:

- 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.

Main Trunk Line (Help)								
No.	Reg	Outgoing Calls			Authentication		Incoming Calls	
		Display Name	User Name	Identity	User ID	Password	Incoming Trunk Match	Forward to
1	Yes ▼	Ernesto Casas		+5255-		Change Password		

- 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 outbound calls from the IPPBX, leave Identity blank and follow instructions in the next section (PBX Lines)

PBX Lines (Help)				
No.	Reg	Outgoing Calls		
		From PBX Number/User	Display Name	Identity
3	No ▼	(.*)	Ernesto Casas	\$1

- 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 field 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.

PBX Lines (Help)									
No.	Reg	Outgoing Calls			Authentication		Incoming Calls		Delete Row
		From PBX Number/User	Display Name	User Name	Identity	User ID	Password	Incoming Trunk Match	
3	No ▼	(.*)	Ernesto Casas			Change Password	4016		<input type="checkbox"/>

- 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.

Networks and Computers							
Name	Subgroup	Lower Limit		Upper Limit (for IP ranges)		Interface/VLAN	Delete Row
		DNS Name or IP Address	IP Address	DNS Name or IP Address	IP Address		
Corporate LAN	-	192.168.100.0	192.168.100.0	192.168.100.255	192.168.100.255	-	<input type="checkbox"/>
Cloud Storage	-	199.36.248.0	199.36.248.0	199.36.251.255	199.36.251.255	external veth untagged	<input type="checkbox"/>
Cloud Storage LAN	-	199.87.120.0	199.87.120.0	199.87.123.255	199.87.123.255	-	<input type="checkbox"/>
Internet	-	0.0.0.0	0.0.0.0	255.255.255.255	255.255.255.255	external veth untagged	<input type="checkbox"/>
VoIP	-	10.200.200.0	10.200.200.0	10.200.200.255	10.200.200.255	external veth untagged	<input type="checkbox"/>
	-	10.198.101.128	10.198.101.128	10.198.101.131	10.198.101.131	external veth untagged	<input type="checkbox"/>
	-	10.65.63.192	10.65.63.192	10.65.63.195	10.65.63.195	external veth untagged	<input type="checkbox"/>
	-	10.171.127.192	10.171.127.192	10.171.127.195	10.171.127.195	external veth untagged	<input type="checkbox"/>
	-	10.172.63.0	10.172.63.0	10.172.63.255	10.172.63.255	external veth untagged	<input type="checkbox"/>
	-	10.200.51.0	10.200.51.0	10.200.51.3	10.200.51.3	external veth untagged	<input type="checkbox"/>
VoIP	-	10.0.0.0	10.0.0.0	10.0.255.255	10.0.255.255	-	<input type="checkbox"/>
VoIP, Private	-	10.0.1.0	10.0.1.0	10.0.1.255	10.0.1.255	internal veth untagged	<input type="checkbox"/>
VoIP, Public	-	10.0.0.0	10.0.0.0	10.0.0.255	10.0.0.255	external veth untagged	<input type="checkbox"/>
+	jive	162.250.60.0	162.250.60.0	162.250.63.255	162.250.63.255	-	<input type="checkbox"/>
		199.36.248.0	199.36.248.0	199.36.251.255	199.36.251.255	-	<input type="checkbox"/>
		199.87.120.0	199.87.120.0	199.87.123.255	199.87.123.255	-	<input type="checkbox"/>
+	gsm	10.0.1.0	10.0.1.0	10.0.1.255	10.0.1.255	internal veth untagged	<input type="checkbox"/>
+	gsm	10.0.0.192	10.0.0.192	10.0.0.192	10.0.0.192	-	<input type="checkbox"/>

4 Additional help or support

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

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