

# SIP Trunk 2 IP-PBX User Guide (Asterisk)

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Ver1.1.1 2017/11/21  
Ver1.1.2 2017/12/14  
Ver1.1.3 2017/12/19

# 1. SIP Trunk 2 Overview

[Sample Configuration SIP trunk2 with IP-PBX]

Unique: 0000123456

SIP Trunk2 Server: [xxx.xxx.xxx.xxx](#)

Your IP-PBX : IP Address 000.000.000.000

DIDs: **0312345678** , **0312123434**

\*If SIP trunk2 Unique was purchased BEFORE March 9, 2017, In case of Japanese toll free numbers such as prefix 0120, 0800 and 0570, you should set its background number showing in Phone Number List of the web portal.

ex.) A number enclosed in parentheses is its background number.

0120\*\*\*\*\* [03\*\*\*\*\*]

\*Customers who use multiple SIP trunk 2 Unique together ( the one purchased before Mar, 9 2017 and the one purchased Mar, 10 2017 afterwards) can check each SIP trunk2 unique setting on the SIP trunk2 Unique Details page as below

Settings Purchase Call History Contacts Circle Information Select Language

► SIP Trunk2 List

<< Back to the list

Login Server [Redacted]

Login Server IP Address [Redacted] ?

Unique [Redacted]

Name [Redacted]

Authentication Method Authentication With IP Address ?

No.	*IP Address	*Port	Incoming Server	Delete
1	[Redacted]	5060	<input checked="" type="checkbox"/>	<input type="checkbox"/>
	[Redacted]	5060		

Switch the number used for the incoming number: Front Number

Multiple call count: 1

Channel reservation for outbound call:

Channel reservation for inbound call:

<< Back to the list

Update

**[ Front Number ]**  
This is number used to receive Free Call, Navi Call, Number portability. (To use this service, Back Number is mandatory)

**[ Back Number ]**  
This is the DID or normal phone number. (It needs to be set into PBX, Asterisk settings)

*\*Note: Upon purchasing of SIP Trunk 2 unique the default value of this setting is Front Number. If you need to change this setting please contact our support.*

## 2. Purchase/Settings in Web Portal

### < SIP Trunk 2 Detailed Settings ▪ Password Authentication >

The screenshot shows the 'SIP Trunk 2 List' configuration page. The settings are as follows:

Step	Setting	Value
1	Login Server	[Redacted]
2	Login Server IP Address	[Redacted]
3	Unique	[Redacted]
4	Name	[Redacted]
5	Authentication Method	Password Authentication
6	Terminal Password	update password
7	Switch the number used for the incoming number	Front Number
8	Multiple call count	1
9	Channel reservation for outbound call	<input type="checkbox"/>
10	Channel reservation for inbound call	<input type="checkbox"/>

① Login server name of SIP Trunk 2

② SIP Server IP Address

Please configure it as [peer] in sip.conf on your Asterisk.

*\*Please refer p.15 for details.*

③ Unique is used as client user ID of your user PBX end.

④ Item "Name" is where you can name/rename your SIP Trunk account.

⑤ Select authentication method as "Password Authentication"

⑥ Enter your terminal password is used as client user password of your PBX end.

⑦ In case you use Free Call or Navi-Dial number, please designate format of incoming number configured on your SIP server.

**Front Number:** This is number used to receive Free Call, Navi Call, Number portability. (To use this service, Back Number is mandatory)

**Back Number:** This is the DID or normal phone number. (It needs to be set into PBX, Asterisk settings)

*\* Changing of this setting is not allowed.*

*\* Upon purchasing of SIP Trunk 2 unique the default value of this setting is Front Number. If you need to change this setting please contact our support.*

⑧ Set multiple call count. It's 1 by default. Purchase "Additional 1 channel for SIP Trunk 2" if you need more than 2 concurrent calls.

⑨ If this option is checked, you can specify channel reservation for multiple outbound calls.

⑩ If this option is checked, you can specify channel reservation for multiple inbound calls.

\*Note ⑨ and ⑩

*The total channel reservation (outbound + inbound) must not exceed the multiple call count. Please refer p.12 for further details.*

## 2. Purchase/Settings in Web Portal

### <SIP Trunk 2 Detailed Settings ▪ Authentication with IP Address>

The screenshot shows the 'SIP Trunk2 List' configuration page. The left sidebar contains a navigation menu with options like 'Unique List', 'SIP Trunk List', 'Fax Trunk List', 'Block/Permit Outbound', 'Block Inbound', 'Group Pickup', 'Extension Digits', 'Grandstream Provisioning', 'Phone Number', 'Phone Number List', and 'PBX Options'. The main content area is titled 'SIP Trunk2 List' and includes a '<< Back to the list' link. The configuration fields are as follows:

- ① Login Server: [Redacted]
- ② Login Server IP Address: [Redacted]
- ③ Unique: [Redacted]
- ④ Name: [Redacted]
- ⑤ Authentication Method: Authentication With IP Address
- ⑥ \*IP Address: [Redacted], \*Port: 5060
- ⑦ Incoming Server:
- ⑧ Front Number: [Redacted]
- ⑨ Multiple call count: 1
- ⑩ Channel reservation for outbound call:
- ⑪ Channel reservation for inbound call:

At the bottom, there are buttons for 'Update', 'Delete On Checked', and 'Reset', and another '<< Back to the list' link.

- ① Login server name of SIP Trunk 2
  - ② Our SIP Server IP Address  
Please configure it as [peer] in sip.conf on your Asterisk.  
*\*Please refer p.15 for details*
  - ③ Unique is used as client user ID of your user PBX end.
  - ④ Item “Name” is where you can name/rename your SIP Trunk account.
  - ⑤ Select authentication method as “Authentication with IP Address”
  - ⑥ Enter a public IP address / a port number of your IP-PBX \*You can add multiple IP addresses/ports from “+Insert” button.
  - ⑦ Your IP-PBX will receive incoming call if ticked. \*If unticked it will work only for outgoing calls.
  - ⑧ In case you use Free Call or Navi-Dial number, please designate format of incoming number configured on your SIP server.  
**Front Number:** This is number used to receive Free Call, Navi Call, Number portability. (To use this service, Back Number is mandatory)  
**Back Number:** This is the DID or normal phone number. (It needs to be set into PBX, Asterisk settings)
- ⑨ Set multiple call count. It's 1 by default. Purchase “Additional 1 channel for SIP Trunk 2” if you need more than 2 concurrent calls.
  - ⑩ If this option is checked, you can specify channel reservation for multiple outbound calls.
  - ⑪ If this option is checked, you can specify channel reservation for multiple inbound calls.

\*Note ⑩ and ⑪

*The total channel reservation (outbound + inbound) must not exceed the multiple call count. Please refer p.12 for further details.*

## 2. Purchase/Settings in Web Portal

### <SIP Trunk 2 Detailed Settings ·

### Authentication using both IP Address and Password>

Circle Management Page Settings Purchase Call History Contacts Circle Information Select Language

Unique SIP Trunk2 List

<< Back to the list

1 Login Server

2 Login Server IP Address

3 Unique

4 Name

5 Authentication Method Authentication using Both IP Address and Password

6 Terminal Password update password

7 IP Address

8 Switch the number used for the incoming number Front Number

9 Multiple call count 1

10 Channel reservation for outbound call

11 Channel reservation for inbound call

<< Back to the list

Update Reset

- ① Login server name of SIP Trunk 2
  - ② Our SIP Server IP Address  
Please configure it as [peer] in sip.conf on your Asterisk.  
*\*Please refer p.15 for details*
  - ③ Unique is used as client user ID of your user PBX end.
  - ④ Item “Name” is where you can name/rename your SIP Trunk account.
  - ⑤ Select authentication method as “Authentication using Both IP Address and Password”
  - ⑥ Enter your terminal password is used as client user password of your PBX end.
  - ⑦ Enter a public IP address / a port number of your IP-PBX \*You can add multiple IP addresses/ports from “+Insert” button.
  - ⑧ In case you use Free Call or Navi-Dial number, please designate format of incoming number configured on your SIP server.  
**Front Number:** This is number used to receive Free Call, Navi Call, Number portability. (To use this service, Back Number is mandatory)  
**Back Number:** This is the DID or normal phone number. (It needs to be set into PBX, Asterisk settings)  
*\* Changing of this setting is not allowed.*  
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  - ⑩ If this option is checked, you can specify channel reservation for multiple outbound calls.
  - ⑪ If this option is checked, you can specify channel reservation for multiple inbound calls.
- \*Note ⑩ and ⑪*  
*The total channel reservation (outbound + inbound) must not exceed the multiple call count.*  
*Please refer p.12 for further details.*