SIP Trunk 2 IP-PBX User Guide （Asterisk）

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

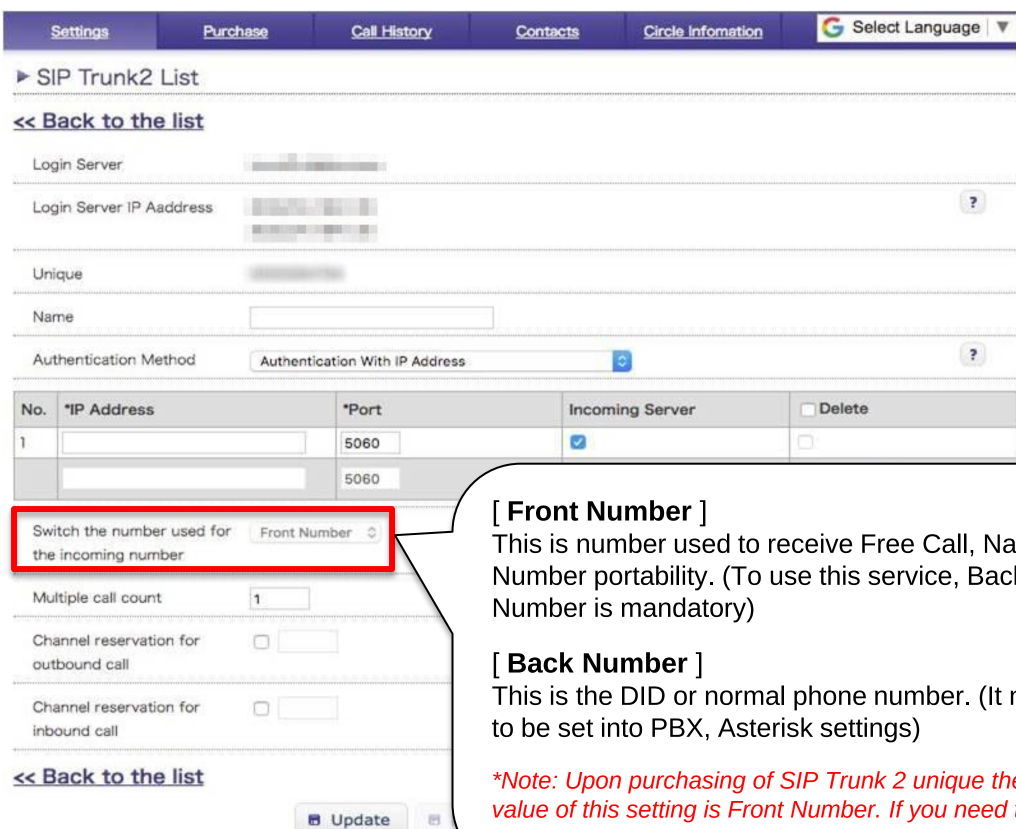
[Sample Configuration SIP trunk2 with IP-PBX]

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

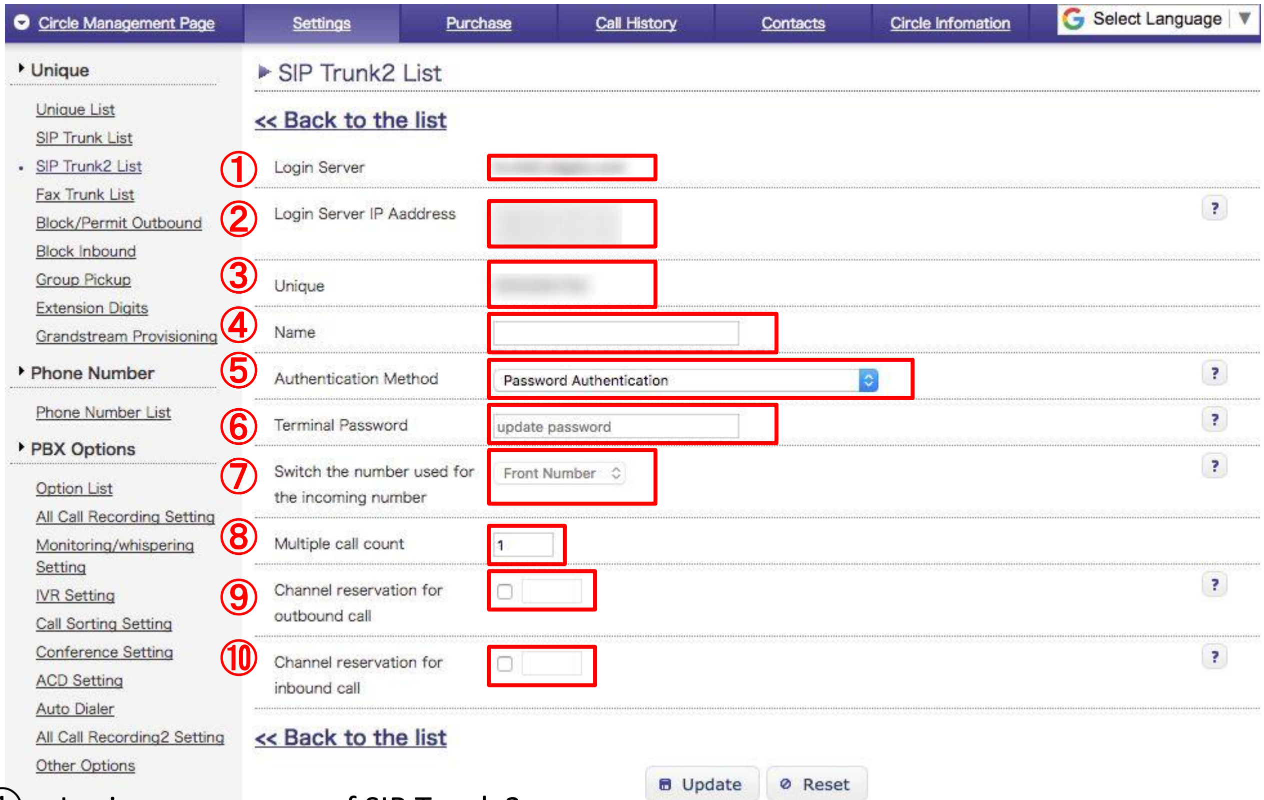


[**Front Number**]
This is number used to receive Free Call, Navi Call, Number portability. (To use this service, Back Number is mandatory)

[**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.*

＜SIP Trunk 2 Detailed Settings ・ Password Authentication＞



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.

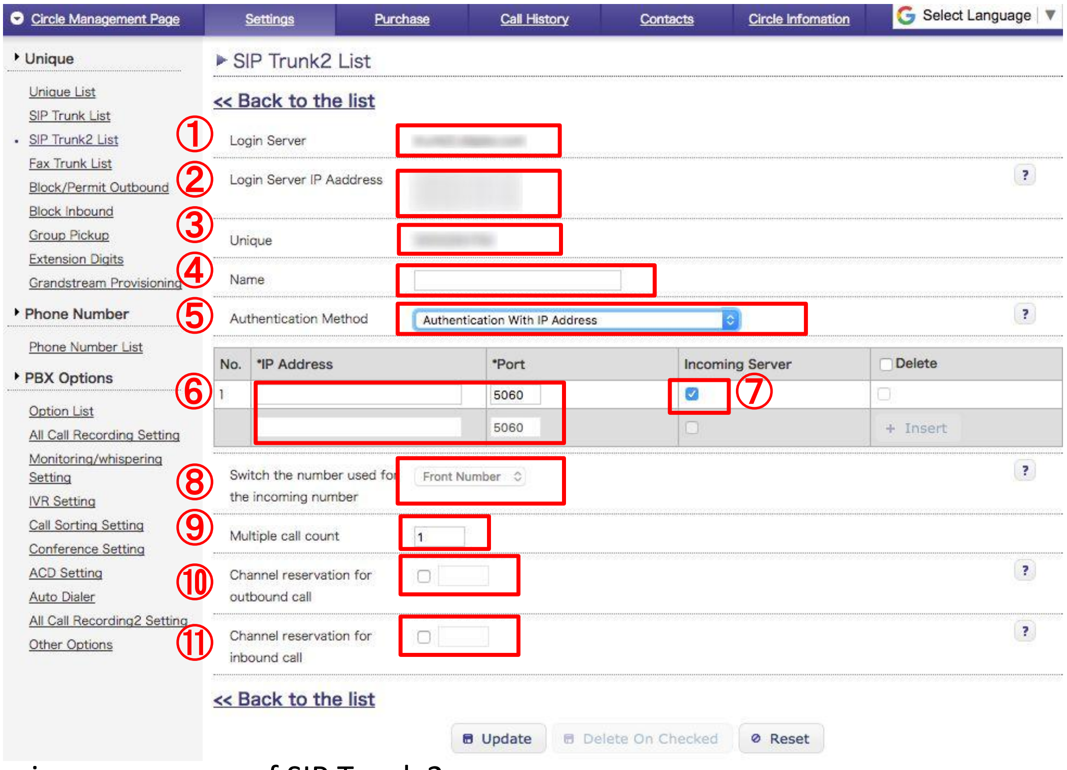
*Note ⑨ and ⑩

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＞



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)

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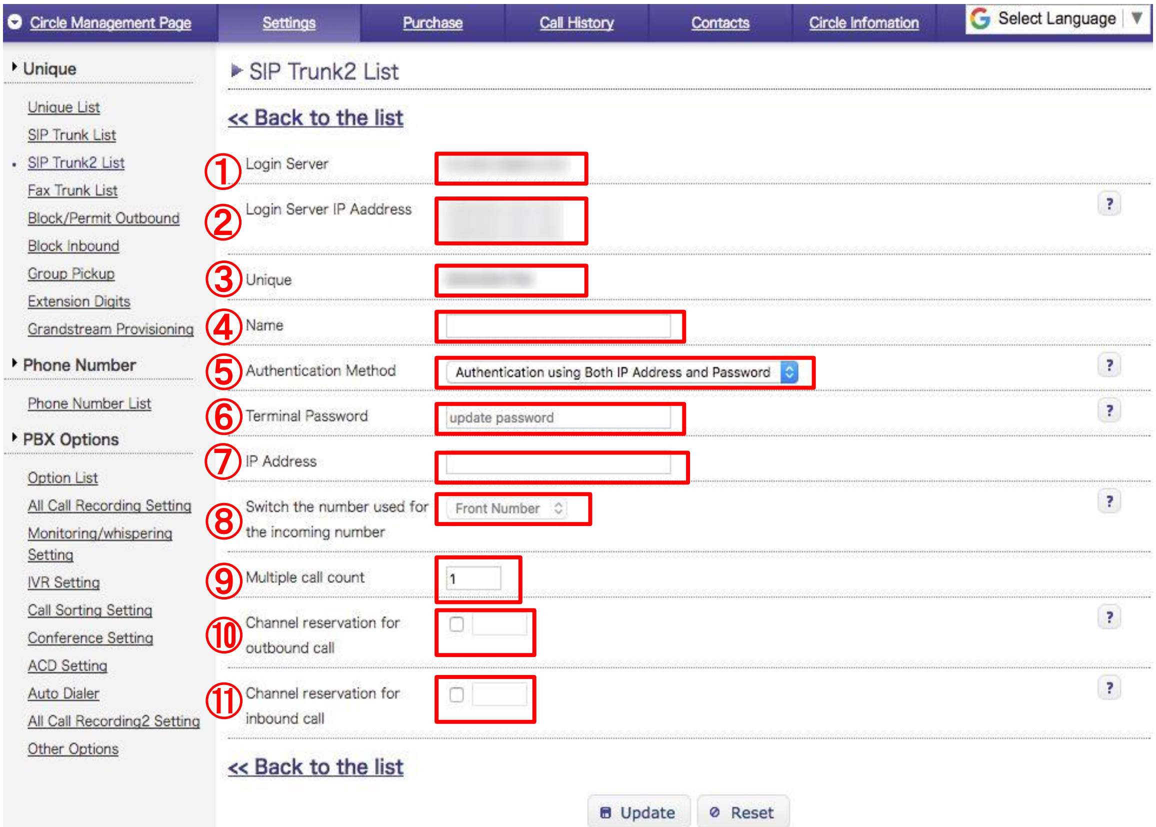
＜SIP Trunk 2 Detailed Settings ・ Authentication using both IP Address and Password>



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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)

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