FTA5120
SAS User’s Guide

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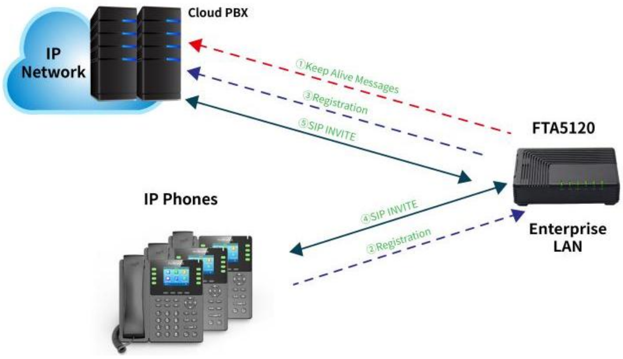
1. SAS Introduction

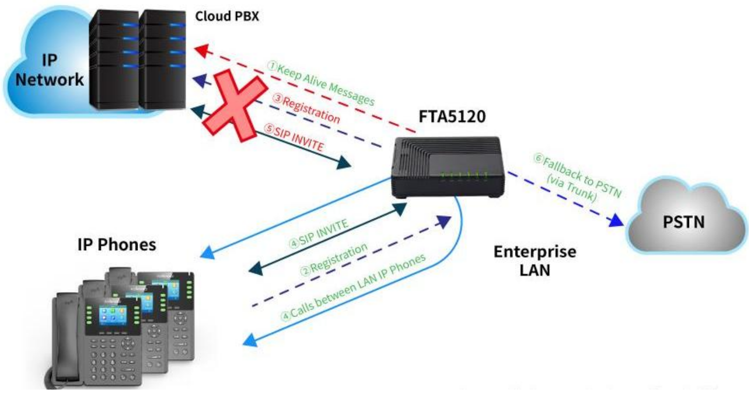
The full name of SAS is Standalone & Survivability, a feature designed to provide a redundancy solution for PBX or a low-cost internal communications solution for small and medium-sized businesses without a PBX. This guide describes the configuration and use of SAS in local networking mode and with Cloud IP PBX mode. The following operations are recommended to work with FTA5120 Gateway and Flyingvoice IP Phones.

1.1. SAS Principle and Working Logic



In Normal state, FTA5120 receives REGISTER request and SIP Invite from UAs in the network and will forward them to the external proxy (i.e., outbound server, proxy server), and send SIP Option message to the cloud PBX to detect its status. Once the proxy replies with a SIP 200 OK, the device records the Contact and Address of Record (AOR) of the UAs in its internal SAS registration database. FTA5120 will maintain a database of all the registered UAs in the network.

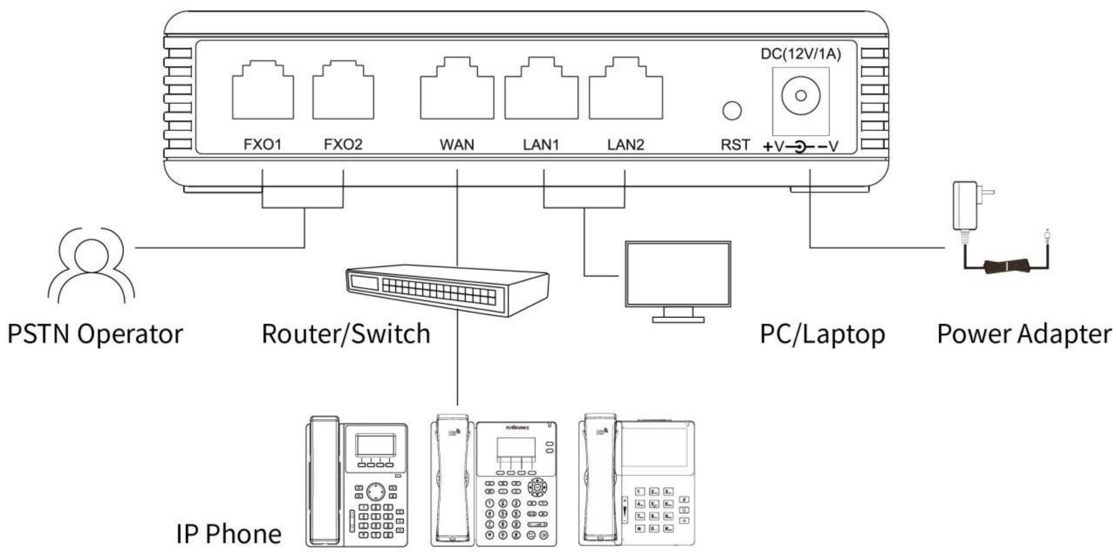
In Emergency state, FTA5120 will server as a proxy for UAs in the network, handle all the Register request and SIP Invite from UAs. Enterprise can carry internal call when the Cloud PBX fails.



When FTA5120 receives calls, it will search the internal SAS registration database to locate the destination address (according to AOR or Contact). If the destination address is not found, FTA5120 will forwards the call to itself. FTA5120 has two FXO port, enterprise can preserves its capability for outgoing calls (from UAs to the PSTN network) or convey IP calls through PSTN line (Via SIP Trunk).

1.2. Hardware Connection

The following operation is applicable to new devices, we recommended that you reset the device to factory settings if used for better performance. Please make sure your devices are connected correctly, the following picture describes the recommended connection topology of FTA5120.



Note:

1. SAS function is currently work only on WAN port, please make sure that you FTA5120 is connected to the network through WAN port.
2. Please make sure your IP Phone is in the same network segment with FTA5120.

2. Configuration of SAS

Please remember to Save&Apply the changes when you are about to leave any page.

When the following prompt appears:

WARNING:Please save&apply the setting to make the last setting effective!

Please Click Save & Apply to make changes take effect

When the following prompt appears:

Please REBOOT to make the changes effective!

Please reboot the device for the changes to take effect.

The Software Version you obtain from Flyingvoice has Pre-Configuration:

The Phone connected to the network will acquire an extension from 600-649;

Dial 0+Phone number to call PSTN line with FXO1/2 automatically;

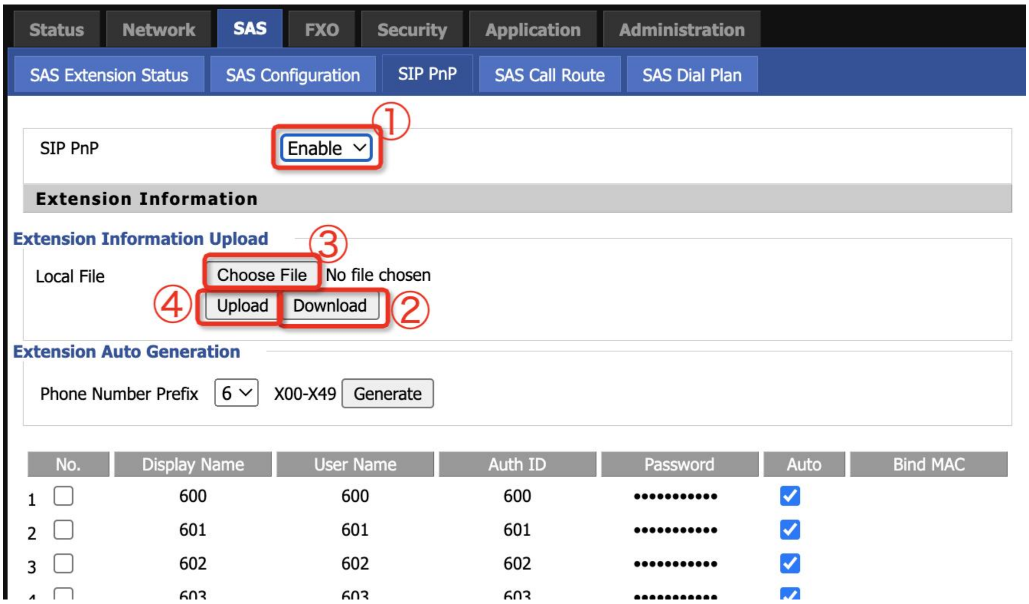
The following steps are for you to configure customized settings.

2.1. Local Networking Mode

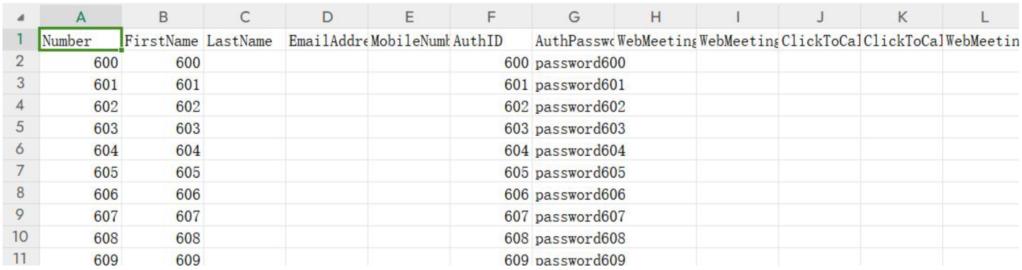
Local Networking Mode is a special application of SAS function, it's the combination of SAS and SIP PnP function. When its registration server (Proxy Server) is the FTA5120 itself or an address that has been unreachable (no response), the system will always be in an emergency state, at this time, its function is similar to a small PBX. With the SIP PnP function, the extension can be created in advance, when you connect the phones, the FTA5120 and the switch, the extension will be distributed to IP Phone automatically, you will get a plug-and-play voice communication system over local IP network in minutes.

2.1.1. Auto Deployment with SIP PnP

1. Login to the Web of FTA5120, If you don't know the WAN port IP address of FTA5120, you can connect the LAN port of FTA5120 to your computer and type 192.168.1.1 in your web browser, the default username and password is admin/admin;
2. Navigate to SAS -> SAS Configuration, select "Enable" for SAS;
3. Navigate to SAS -> SIP PnP, select "Enable" for SIP PnP, click "Save&Apply";
4. Select the prefix of your extension numbers in "Extension Auto Generation", click "Generate", then 50 extension will be added, click "Save &Apply";



5. Or: you can download template file in the "Extension Information upload" to fill in the extension information, upload the filled file, the extensions will be created;



	A	B	C	D	E	F	G	H	I	J	K	L
1	Number	FirstName	LastName	EmailAddre	MobileNumb	AuthID	AuthPasswo	WebMeeting	WebMeeting	ClickToCal	ClickToCal	WebMeetin
2	600	600				600	password600					
3	601	601				601	password601					
4	602	602				602	password602					
5	603	603				603	password603					
6	604	604				604	password604					
7	605	605				605	password605					
8	606	606				606	password606					
9	607	607				607	password607					
10	608	608				608	password608					
11	609	609				609	password609					

6. Plug in the IP phone to your network and wait for the distribution of extension numbers, the IP Phone will reboot after successfully obtaining account.

Note:

1. If you want to distribute an extension number to a specific IP Phone, you can select the extension and click edit, fill in the MAC address of the IP Phone in the blank "Bind MAC";
2. Default password for Auto Generation is "Password+Extension" ,for example, the password for 601 is Password601;
3. Users can add extension with special configuration(e.g. password, number) by clicking "Add" button. The maximum number of extensions is 50;
4. When using PNP, IP phone should be in the same network segment with FTA5120, Otherwise, the configuration may fail to be distributed, we recommend that closing the LLDP and CDP on IP Phone;

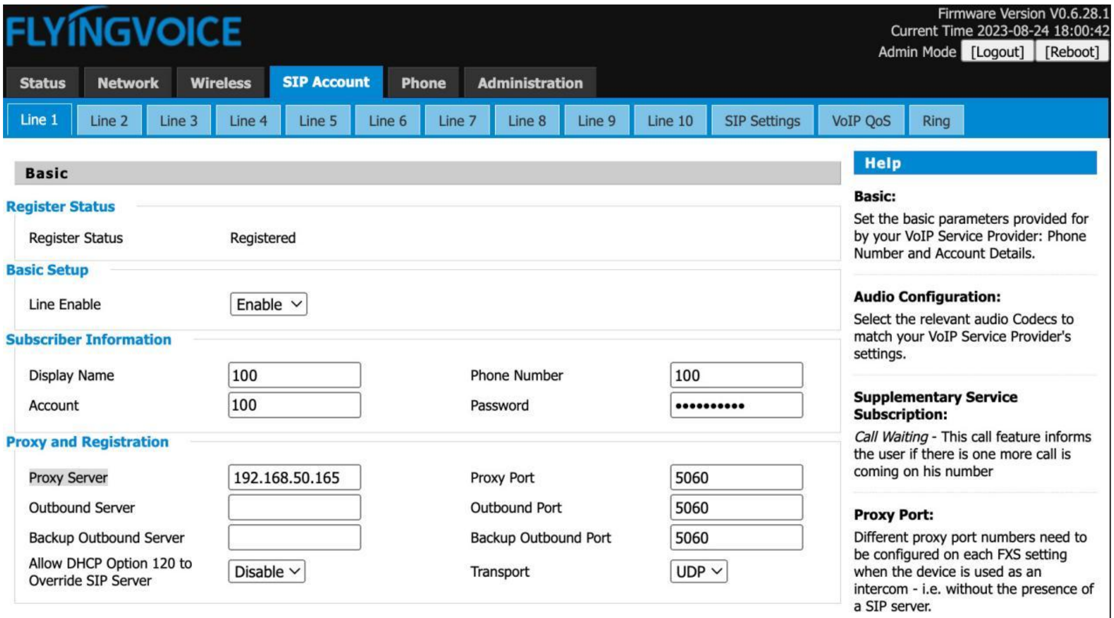
5. If the phone was previously registered for an account, it is recommended that you restore the phone to its factory settings, otherwise it may affect the distribution of the configuration;

6. FTA5120 has been preconfigured with 50 extensions from 600-649. Before customizing the settings, please select all the extensions and delete them before proceeding with the related operations.

2.1.2. Manual Registration

If You are using third-party IP Phone, please refer to the following steps (Diifferent IP Phone may have different configuration steps and configuration item name, the following steps is for reference only):

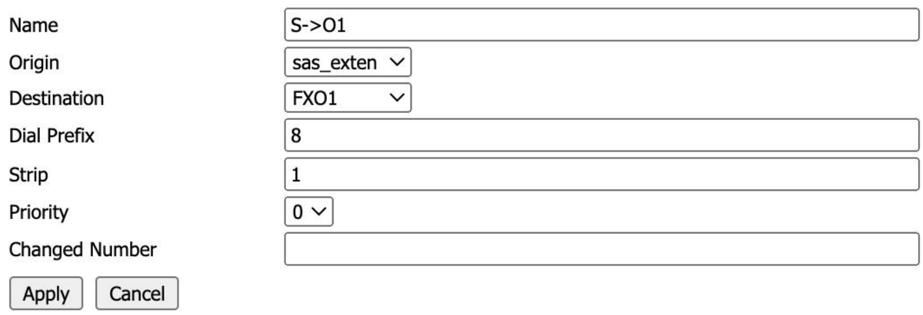
1. Follow the **section 2.1.1** to add extensions on FTA5120;
2. Login to the Web Interface of your IP Phone, Fill in the Display name, User Name; Auth ID(Account for IP Phone), Password according to the information on "SIP PnP" page of FTA5120
3. Fill in the IP Address of FTA5120 in the "Proxy Server" of the IP Phone;
4. Click Save & Apply.



2.1.3. Call to PSTN

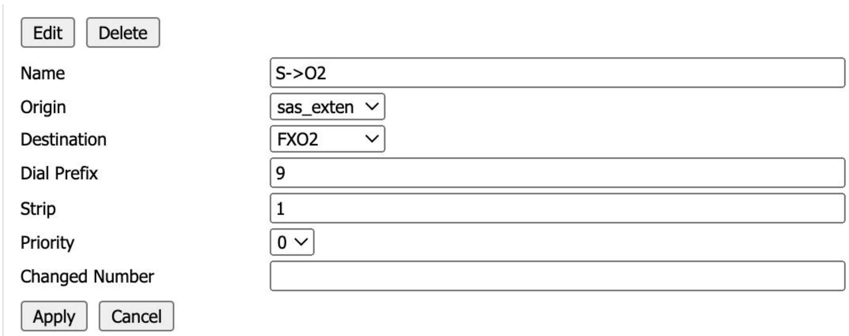
FTA5120 provides two FXO port, you can make a phone call through PSTN by connecting FTA5120 to PSTN or convey an emergency call through PSTN when network fails.

1. Navigate to SAS -> SAS Call Route
2. Select an empty route item, click "edit";
3. Fill in the route name to identify the route for management;
4. Select sas_exten in "Origin", select FXO_1 in "Destination" (please connect FXO 1 to the telephone port before setting);
5. Fill in the 8 for "Dial Prefix", 1 for "Strip"and 0 for "Priority";
6. Click "Save & Apply".



At this time there will be a new SAS call routing, you can dial 8 + external number to call the phone on PSTN:

If you need to use two FXO ports, you need to add another route item:

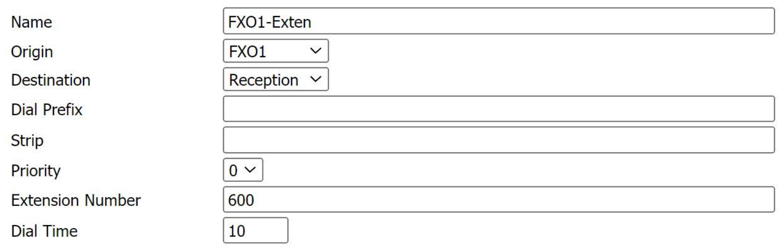


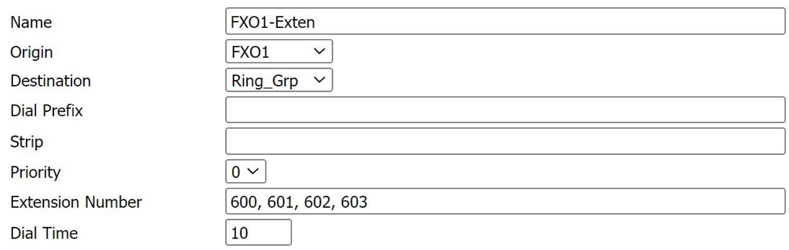
Note:

1. Dial Prefix for outgoing calls should be different from the Dial Prefix for internal calls, if you choose "8" for the extension you create, please choose other numbers;
2. FTA5120 has been preconfigured with 2 Routes. Before customizing the settings, please delete them before proceeding with the related operations.

2.1.4. Answer A Call from PSTN

1. Navigate to SAS -> SAS Call Route
2. Select an empty route item, click "edit";
3. Fill in the route name to identify the route for management;
4. Select FXO1 in "Origin", you can select IVR,Reception,Ring_Grp in destination;
5. Fill in the extension numbers in the blank;
6. Click "Save & Apply".





The details of Reception, Ring_Grp (Ringing Group) and IVR are as follows

Reception -> All calls from FXO will be forwarded to this extension。
Example: Fill in the extension number 600 in the "Reception" configuration blank 1001, when receiving an incoming call from FXO port, the 1001 will ring.

Ringing Group -> When receiving an incoming call from FXO port, the extensions in the group will ring in sequence.（use " , " to divide numbers, e.g. 600,601,602, 603）。
Example: Fill "600,601,602,603" in the ringing group, when receiving an incoming call from FXO port, 600 will ring first, 601, 602 and 603 will ring in sequence after the end of ringing of last extension.

IVR -> When receiving an incoming call from FXO port, a voice prompt will be play to help dialing. When caller press 0, the call will be forward to the extension that is filled in "Extension Number" for looking up the number.

Dial Time: Duration time of FXO port ringing, only available when Reception or Ringing group is enable.

Note: When Ringing Group is enable, Dial Time is the duration time of single extension.

2.1.5. Check Status

After the IP phones are registered successfully, you can see all the extension numbers in SAS -> SAS Extension Status. You can view the status of all the currently registered IP Phones, including their IP address, MAC address and working mode.

There are three working modes in total: **Registered (Normal)**, **Registered (Standalone)** and **Unavailable**. The description to different mode are as follows:

Mode	Description
Registered (Normal)	The IP phone is registered to the IP PBX and the PBX is in normal status.
Registered (Standalone)	1. The PBX is offline, FTA5120 enters emergency mode 2. FTA5120 works in Local Mode
Unavailable	The IP phone is offline,removed or broken.

2.2. SAS + Cloud PBX/Local PBX

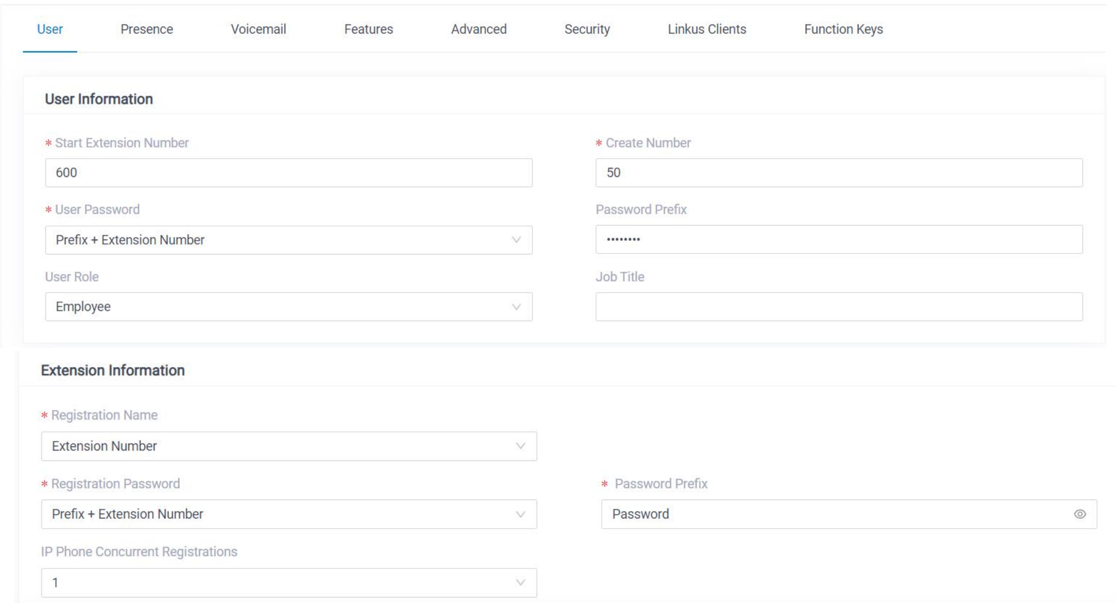
For any operation in the following steps, please make sure your IP PBX is reachable for the network that FTA5120 is in. The following steps may have different configuration steps and configuration item name due to the divergence of IP PBX, the following steps are for reference only.

2.2.1. Create Extension on PBX

In this mode, user should create extension on Cloud PBX/Local PBX first. Here, we use Yeastar P series Cloud PBX for demonstration, for the details of this steps, please refer to User Guide of your PBX or contact your operator:

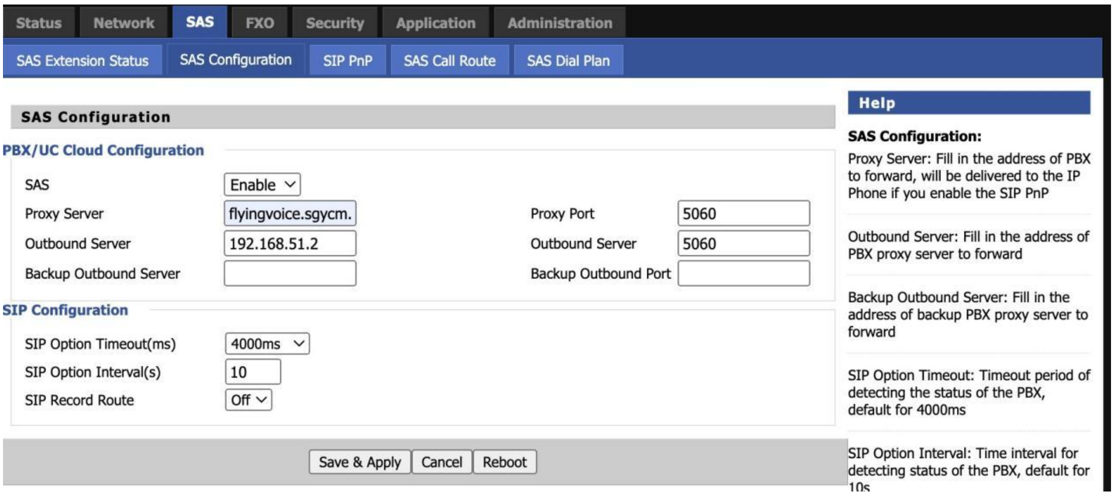
1. Navigate to Extension and Trunk -> Extension
2. Click "Bulk Add";
3. Fill in the configuration information as follows;

For demonstration purposes, we select "Prefix + Extension Number" as the passwords format and configure the password Prefix as "Password" to accommodate the default password for Auto Generate Account on FTA5120, please change your password and set a higher lever password for security consideration.

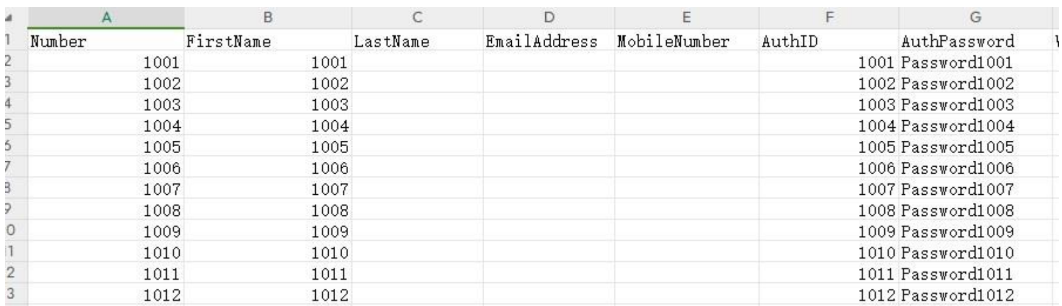


2.2.2. Auto Deployment with SIP PnP

1. Login to the Web of FTA5120, If you don′ t know the WAN port IP address of FTA5120, you can connect the LAN port of FTA5120 to your computer and type 192.168.1.1 in your web browser, the default username and password is admin/admin;
2. Navigate to SAS -> SAS configuration, fill in the domain name(IP address)of your PBX in the "Proxy Server", if you′ re using outbound server, please fill in the blank "Outbound Server" and "Backup Outbound Server" according to your actual situation;



3. Navigate to SAS -> SIP PnP, select "Enable" for SIP PnP;
4. Click "Download" on the "Extension Information Upload"; a csv. Template file will be downloaded automatically;



	A	B	C	D	E	F	G
1	Number	FirstName	LastName	EmailAddress	MobileNumber	AuthID	AuthPassword
2	1001	1001				1001	Password1001
3	1002	1002				1002	Password1002
4	1003	1003				1003	Password1003
5	1004	1004				1004	Password1004
6	1005	1005				1005	Password1005
7	1006	1006				1006	Password1006
8	1007	1007				1007	Password1007
9	1008	1008				1008	Password1008
10	1009	1009				1009	Password1009
11	1010	1010				1010	Password1010
12	1011	1011				1011	Password1011
13	1012	1012				1012	Password1012

5. Fill in the file "SasExtensionConfig" according to the PBX′ s information and click "Choose File" to choose the filled file, then click "Upload" ,the page will jump to prompt page if successfully;

6. Please reboot the device to make the changes take effect;

7. After the device reboot successfully, plug in the IP phone to the network, and the IP Phone will automatically get the extension and register to the PBX.

Note:

1. If you′ re using PBX from 3CX, you can directly export the extension list of the PBX and upload it on the "Extension Information Upload";
2. If the extension number and password of created extensions on PBX is the same with FTA5120′ s default setting, you can also use Auto Generate to create SIP PnP information.
3. You can also manually add extensions by clicking "Add" buttons and fill in the information;
4. The phone may reboot during the SIP PnP, If your phone can not obtain the extension, you may need to reboot the device;
5. For Cloud PBX, fill in the domain name; for Local PBX, fill in the IP address;
6. If you want to distribute an extension number to a specific IP Phone, you can select the extension and click edit, fill in the MAC address of the IP Phone in the blank "Bind MAC";
7. When using PNP, IP phone should be in the same network segment with FTA5120, Otherwise, the configuration may fail to be distributed, we recommend that closing the LLDP and CDP on IP Phone.

2.2.3. Manual Registration

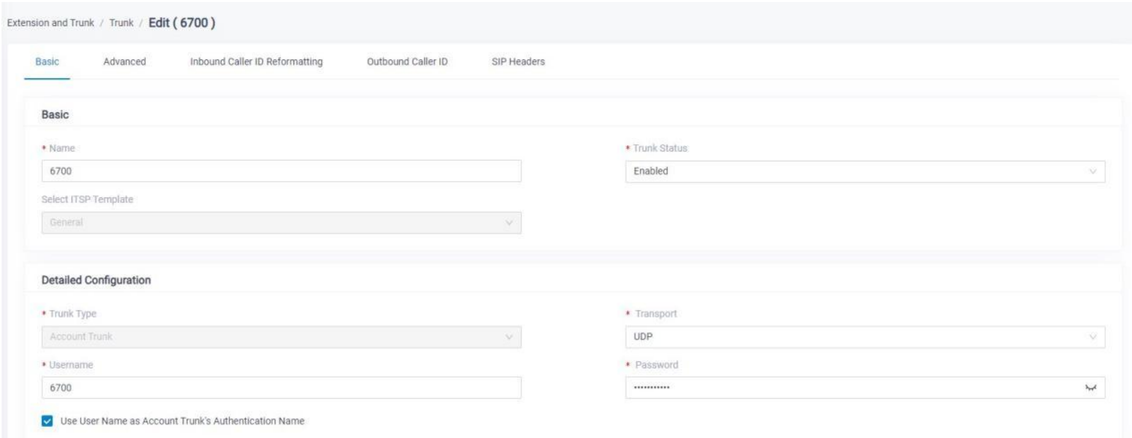
If You are using third-party IP Phone, please refer to the following steps (Diifferent IP Phone may have different configuration steps and configuration item name, the following steps is for reference only):

1. Navigate to SAS -> SAS configuration, fill in the domain name(IP address)of your PBX in the "Proxy Server", if you′ re using outbound server, please fill in the blank "Outbound Server" and "Backup Outbound Server" according to your actual situation;
2. Follow the **section 2.2.2** to add extensions on FTA5120;
3. Login to the Web Interface of your IP Phone, Fill in the Display name, User Name; Auth ID(Account), Password according to the information on your PBX;
4. Fill in the domain name (IP address) in the "Proxy Server", Fill in the IP Address of FTA5120 in the "Outbound Server" of the IP Phone;
5. Click Save & Apply.

2.2.4. SIP Trunk Configuration

In this mode, user need to create trunk on Cloud PBX/Local PBX first. Here, we use **Yeastar P series Cloud PBX** for demonstration, for the details of operation of this steps, please refer to the user’s guide of your PBX or contact your operator:

1. Login to the Web management;
2. Navigate to Extension and Trunk -> Trunk, click “Add’;
3. Fill in the name to identify this trunk, select Enabled in “Trunk Status”, select “Account Trunk” in “Trunk Type”;
4. Fill in the user name and password, tick the selection “Use User name as Account Trunk’s Authentication Name”.
5. Click “Save”, and “Apply” the changes



Note: If you need to use two PSTN line, please create another SIP trunk.

SIP Trunk is a method to convey the PSTN call to PBX, so you can answer a PSTN call on IP Phone through a cloud PBX or local PBX or make a call to mobile phone easily. Users can also configure some advanced function, which is depends on the PBX. The following section is the configuration steps of basic function, please connect FTA5120’s FXO port to your PSTN line before the configuration.

1. Navigate to FXO -> SIP Trunk, select SIP 1 in “SIP Trunk”,enable “register”;
2. Fill in the PBX’s domain Name(IP address), If you have an outbound server for this, please fill in the IP address of outbound server in the blank “Outbound Server”;
3. Fill in Display Name, Phone Number, Account and Password according to the configured items in **section 2.2.4;**

4. Click "Save&Apply".

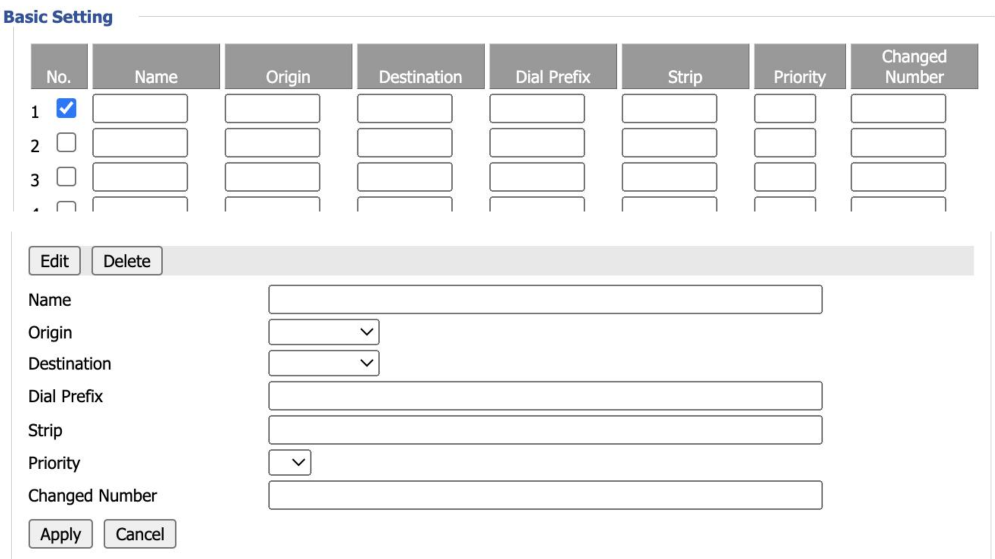
Note:

If you need to use two PSTN line, please select SIP 2 in "SIP Trunk" and register another SIP trunk.

2.2.5. Call Route Setting

On FTA5120:

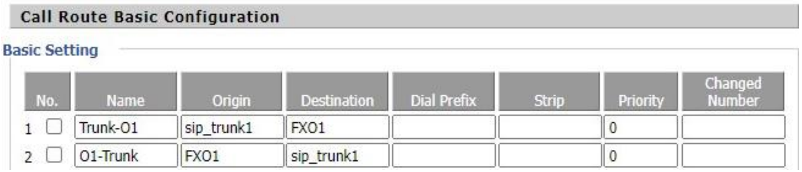
Navigate to FXO -> Call Route Setting



Parameter	Description
Name	Mandatory field, name of the route, unique identifier
Origin	The Source interface of the call, you can configure SIP Trunk or FXO port.
Destination	The destination interface of the call, you can configure SIP Trunk or FXO port.
Dial Prefix	Number with the prefix will be forwraded by this route, available only in route that is to FXO
Strip	Setting the number digits of the call prefix. e.g. When Dial Prefix is set to “89”, you need to set Strip=2 to remove the first two digit. This can be left blank if your PBX support Dial Prefix.
Priority	Select the priority of this route, 0 is the highest priority.
Changed Number	If the route is from FXO to SIP Trunk, this should be filled with an extension on your PBX (this extension should not be registered by any IP Phone); If the route is from FXO to SIP Trunk, but your PBX configure IVR for the FXO calls, this can be left blank; No require for route from SIP Trunk to FXO;

The following steps are for reference only:

1. Navigate to FXO -> Call Route;
2. Select an empty item, click "Edit";
3. Fill in the name to identify the route for management, select "FXO_1" for origin, select "sip_trunk1" for destination, select "0" for priority;
4. Select another empty item, click "Edit";
5. Fill in the name to identify the route for management, select "sip_trunk1" for origin, select "FXO_1" for destination, select "0" for priority;
6. Click "Save&Apply".


Call Route Basic Configuration

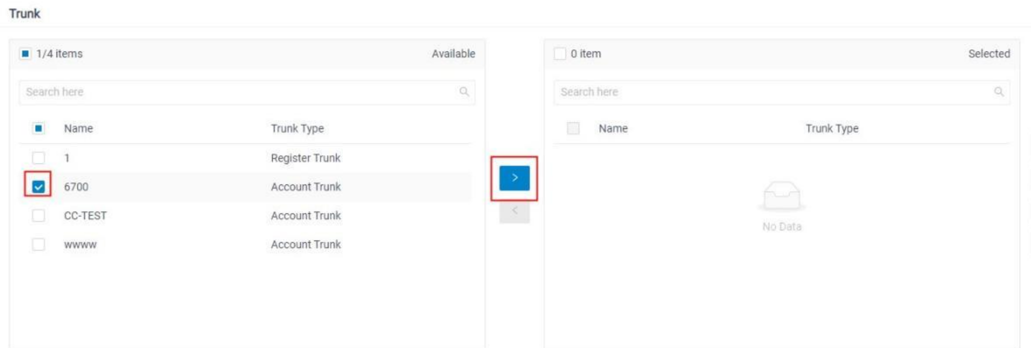
Basic Setting

No.	Name	Origin	Destination	Dial Prefix	Strip	Priority	Changed Number
1	Trunk-O1	sip_trunk1	FXO1			0	
2	O1-Trunk	FXO1	sip_trunk1			0	

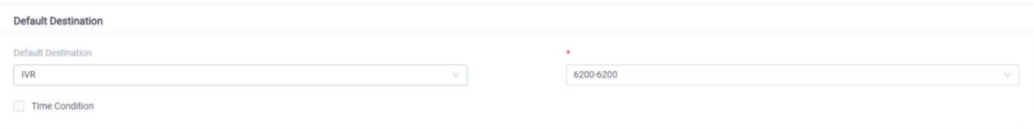
On PBX:

Inbound Route Setting:

1. Navigate to Call Control -> Inbound Route, click "Add";
2. Fill in the name to identify this route;
3. Select the trunk 6700(Created in the section 2.2.4), and move it to the "Slected " blank;



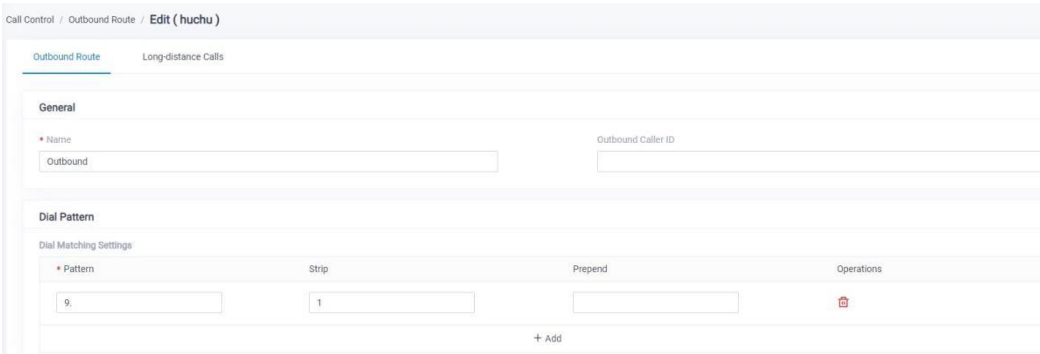
4. Select IVR in the "Default Destination" ,select a number for IVR (In Yeastar P series PBX, you need to configure one previously in Call Feature -> IVR.);



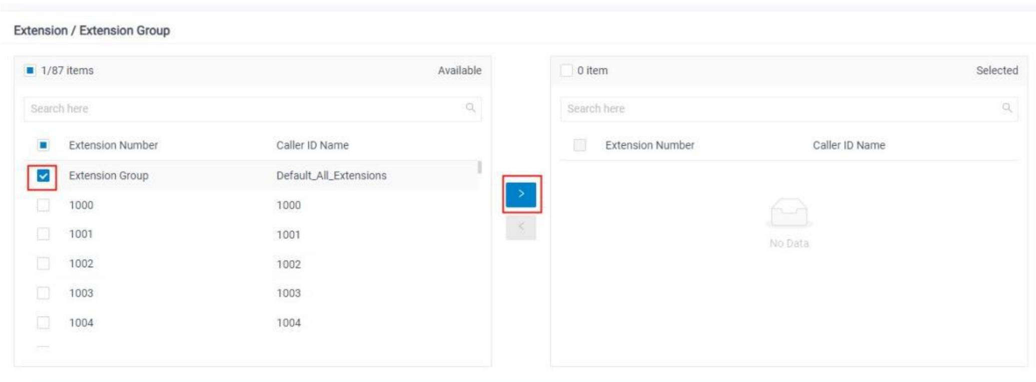
5. Click "Save", and "Apply" the changes.

Outbound Route Setting:

1. Navigate to Call Control -> Inbound Route, click "Add";
2. Fill in the name to identify this route;
3. Fill in "9." in "Pattern", "1" in Strip (When IP Phone dial 9+PSTN Number, it will be carry through this route, and the first number 9 will be removed to deliver correct PSTN number to the FXO port of FTA5120.);
4. Select the trunk 6700(Created in the section 2.2.4), and move it to the "Slected " blank;



5. Select the extension/extension group in this route, for demonstration we select "Default _All_Extensions" and move to "Selected" blank;
6. Select Always in "Time Condition -> Available Time".



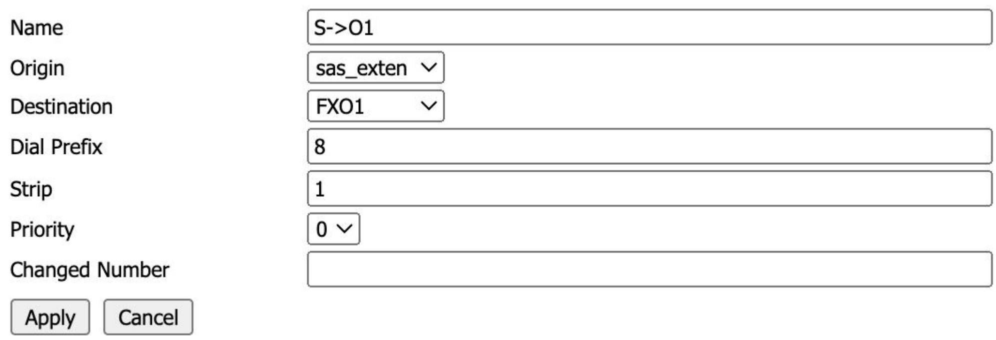
Note:

1. Please make sure there is no conflict route on FTA5120 and your PBX, otherwise the function may work abnormally.
2. If you need to use two PSTN line, please create another group of Inbound and Outbound route that use different SIP Trunk and FXO port.

2.2.6. SAS Call Route

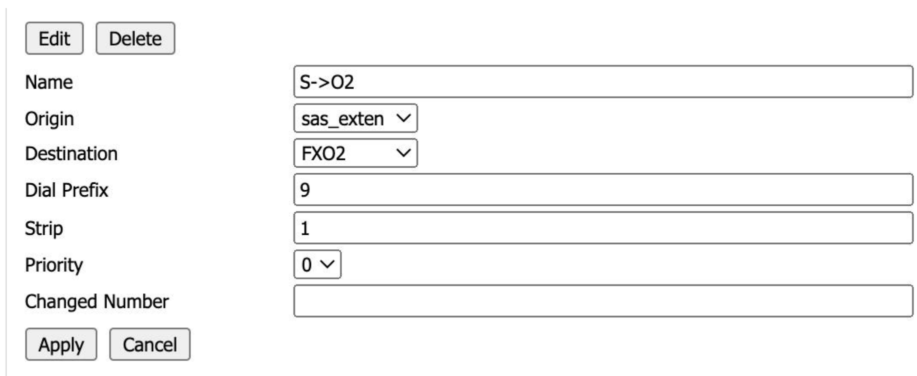
When PBX fails, the device will enter Emergency Mode, under this mode, calls will not processed by PBX, which means the call route in FXO -> Call Route will not take effect. In order to maintain normal communication capability with PSTN line, you need to set SAS call route:

1. Navigate to SAS -> SAS Call Route
2. Select an empty route item, click "edit";
3. Fill in the route name to identify the route for management;
4. Select sas_exten in "Origin", select FXO_1 in origin (please connect FXO 1 to the telephone port before setting);
5. Fill in the 8 for "Dial Prefix", 1 for "Strip"and 0 for "Priority";
6. Click "Save & Apply"。



At this time there will be a new SAS call routing, you can dial 8 + external number to call the phone on PSTN:

If you need to use two FXO ports, you need to add another route item:

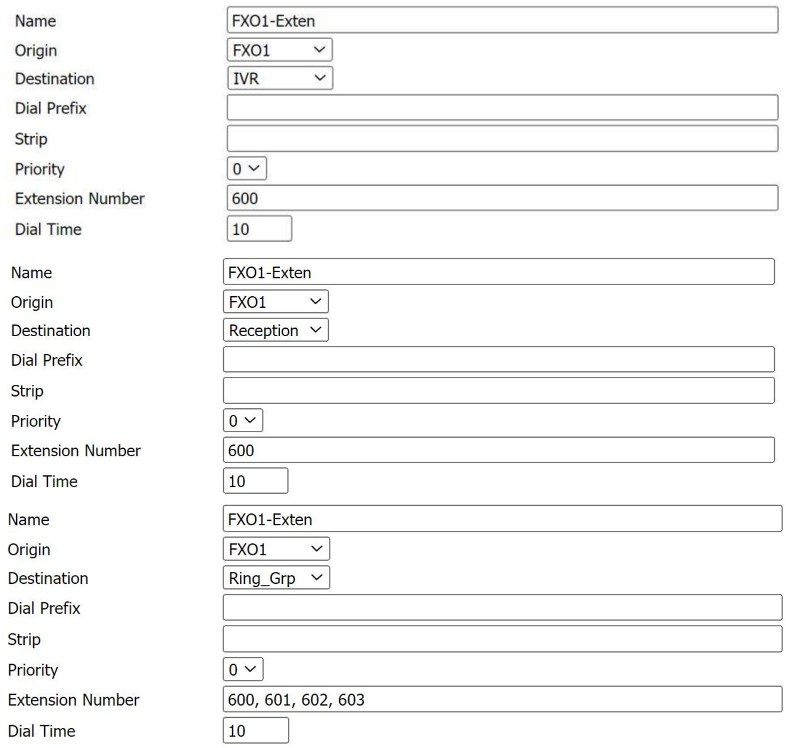


Note:

Dial Prefix for outgoing calls should be different from the Dial Prefix for internal calls, if you choose "9" or "8" for the extension you create, please choose other numbers.

Inbound Route:

1. Navigate to SAS -> SAS Call Route
2. Select an empty route item, click "edit";
3. Fill in the route name to identify the route for management;
4. Select FXO1 in "Origin", you can select IVR,Reception,Ring_Grp in destination;
5. Fill in the extension numbers in the blank;
6. Click "Save & Apply".



The details of Reception, Ring_Grp (Ringing Group) and IVR are as follows

Reception -> All calls from FXO will be forwarded to this extension。
Example: Fill in the extension number 600 in the "Reception" configuration blank 1001, when receiving an incoming call from FXO port, the 1001 will ring.

Ringing Group -> When receiving an incoming call from FXO port, the extensions in the group will ring in sequence. (use " , " to divide numbers, e.g. 600,601,602, 603) 。
Example: Fill "600,601,602,603" in the ringing group, when receiving an incoming call from FXO port, 600 will ring first, 601, 602 and 603 will ring in sequence after the end of ringing of last extension.

IVR -> When receiving an incoming call from FXO port, a voice prompt will be play to help dialing. When caller press 0, the call will be forward to the extension that is filled in "Extension Number" for looking up the number.

Dial Time: Duration time of FXO port ringing, only available when Reception or Ringing group is enable.

Note: When Ringing Group is enable, Dial Time is the duration time of single extension.

2.2.7. Check Status

After the IP phones are registered to the PBX successfully, you can see all the extension numbers in SAS -> SAS Extension Status. You can view the status of all the currently registered IP Phones, including their IP address, MAC address and working mode.

There are three working modes in total: **Registered (Normal)**, **Registered (Standalone)** and **Unavailable**.

The description to different mode are as follows:

Mode	Description
Registered (Normal)	The IP phone is registered to the IP PBX and the PBX is in normal status.
Registered (Standalone)	1. The PBX is offline, FTA5120 enters Emergency mode 2. FTA5120 works in Local Mode
Unavailable	The IP phone is offline,removed or broken.