Yealink Hybrid-mode Feature Compatible with AudioCodes SBC

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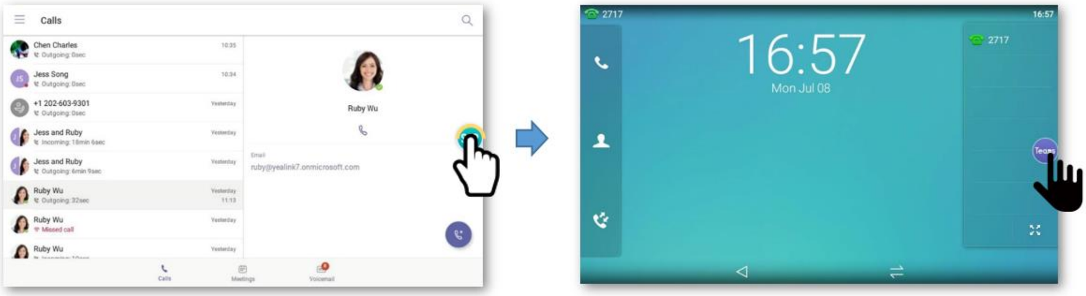
Version 15.1

Brief Introduction of Yealink Hybrid-mode Feature[1]

The Teams & SIP[2] hybrid-mode feature on Yealink Teams IP phone is that two applications (herein referred to as Teams APP and SIP APP) run on the phone simultaneously. SIP APP can be used as a back-up communication client when Teams phone system ran into trouble. Yealink hybrid-mode feature can also be used across an enterprise, where SIP user groups and Teams user groups need communicate with each other (typically for the customers migrating from traditional SIP solutions to Microsoft Teams solutions, and during the migration, the two groups will coexist.)



#1 Attach a video for a more intuitive understanding of the features: Quick View of Yealink Hybrid-mode Feature on Teams Phone.

#2 The 'SIP' in this document means traditional UC solution using separate Open SIP devices (e.g. Yealink Open SIP IP phone) and 3rd party IP PBX (e.g. BroadSoft) without relying on Teams phone system.

Applicable Deployments

Yealink hybrid-mode feature can be applied to the following deployments:

Teams Direct Routing (with/without media bypass) Deployment

Yealink Teams phone can work with AudioCodes SBC to use PSTN services provided by local telephone carriers. It is suitable for customers whose area is not covered by Microsoft Calling plan, or the organization has an existing contract with a PSTN carrier.

For more information, refer to Microsoft blog for Phone System Direct Routing.

Teams & SIP Hybrid-mode Deployment

Yealink Teams phone can work with AudioCodes SBC to support calls based on SIP clients and Teams clients in local deployment where SIP and Teams telephone systems go together. It is suitable for customers that need both SIP and Teams communication capabilities (as mentioned, such as the customers in migration).

This document focuses on hybrid-mode deployment, refer to Topology of Yealink Hybrid-mode Feature for the detailed topology.

Applicable Products and Software Versions

Yealink hybrid-mode feature applys to all Yealink Teams certified IP phone and AudioCodes Teams certified SBCs, including:

Supported AudioCodes SBC models

Vendor	Product	Software Version
AudioCodes	Mediant 500 SBC	7.20A.250 and above
	Mediant 800 SBC	7.20A.250 and above
	Mediant 2600 SBC	7.20A.250 and above
	Mediant 4000 SBC	7.20A.250 and above
	Mediant 1000B SBC	7.20A.250 and above
	Mediant 9000 SBC	7.20A.250 and above
	Virtual Edition SBC	7.20A.250 and above

For the Teams & SIP hybrid-mode deployment, you need to purchase Teams and SIP licenses from AudioCodes before deploying AudioCodes SBC. The required licenses include:

- SBC Session License
- SBC Registered Users License
- Microsoft Teams License

Supported Yealink Teams phone models

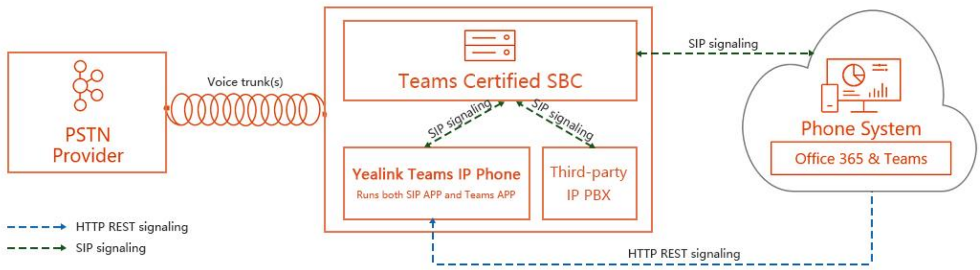
Vendor	Product	Software Version
Yealink	SIP-T58A	58.15.0.38 and above
	SIP-T56A	58.15.0.38 and above
	SIP-T55A	Available by CY19Q4
	CP960	73.15.0.38 and above
	VP59	Available by CY20Q2

The software can be downloaded from Yealink Support.

Introduction of Yealink Hybrid-mode Feature

Topology of Yealink Hybrid-mode Feature

In Teams & SIP hybrid-mode deployment, Teams calls will be carried by Teams phone system, but the PSTN calls from SIP client or Teams client (i.e. the Teams APP and SIP APP on Yealink Teams phone) will be mainly routed by the SBC. You can also deploy additional voice devices, such as IP PBX. The topology is as follows:



SBC can also be installed on the local virtual machine or Microsoft Azure. This document focuses on the configuration steps of installing VE SBC on the local VMware virtual machine, see Steps for AudioCodes SBC

Configuration for details.

When WAN outage occurs, the SIP signaling path provided by SBC will serve as a backup communication method, the topology is as follows:

SIP-related Function List of Yealink Hybrid-mode Feature

The SIP functions supported by Yealink Teams phone are as follows

Application	Module	Features
SIP App (For the Teams-related functions, refer to Phones for Microsoft Teams)	Apps Switch	One-click quick switch between Teams&SIP Apps via Float Ball
		Received calls from Teams or SIP client
	SIP Registration	16 SIP account registrations
		Line keys
	SIP Call	P2P call
		PSTN call
		Call control: e.g. call hold, call mute, call transfer, call forward etc.
		Conference call
		Conference call management: split call, invite new participants etc.
		Call history
	Contacts	Local Contacts/ Remote Contacts
		Search Contacts
		Contacts management: add contacts, create groups, set delegated ringtone etc.

Configuration Steps to Enable Yealink Hybrid-mode Feature on Yealink Teams Phone and AudioCodes SBC

Enabling Yealink Hybrid-mode Feature on Yealink Teams Phone

1. Precondition

If you want to configure the phones in batches, you need to build an auto provisioning (herein referred to as AutoP) server or connect the phones to Yealink device management platform (herein referred to as YDMP), and push resource files to the phones via AutoP / YDMP. To learn how to perform AutoP, refer to

AutoP Guide on Yealink Support.

If you want to configure a single phone, you need to import the resource file to the phone via the web user interface of the phone (AutoP/YDMP also work but not preferred). For detailed configuration steps, refer to Steps for Yealink Teams Phone Configuraion.

2. Configuration steps

a) Configure CFG resource file

b) Import the CFG resource file into a single phone

c) Check whether all configurations take effect successfully

Steps for Yealink Teams Phone Configuration

1. Configure CFG resource file

a) You can write a CFG file by yourself, or you can access CFG Resource Files for Yealink Hybrid-mode Feature to download sample CFG files directly. The CFG file corresponding to the phone model is as follows. Select the corresponding CFG file and import it into the phone.

Phone Model	Common CFG file
T58A	y000000000058.cfg
T56A	y000000000056.cfg
T55A	y000000000099.cfg
CP960	y000000000073.cfg
VP59	y000000000091.cfg

b) In the CFG file, the required parameters are as follows (The configuration shall be split into two steps)

Step 1

```
#!version:1.0.0.1
features.hybrid_mode.enable=1
```

Step 2

```
#!version:1.0.0.1
account.X.enable = 1          [X ranges from 1 to 16]
account.X.sip_server.1.address = <SBC IP Address>
account.X.sip_server.1.port = <SBC Port >
account.X.user_name = <Username>
account.X.password = <Password>
account.X.srtp_encryption= 2          [This item is optional]
features.hybrid_mode.quick_ball.enable=1          [This item is optional]
```

The parameters are described in detail as below

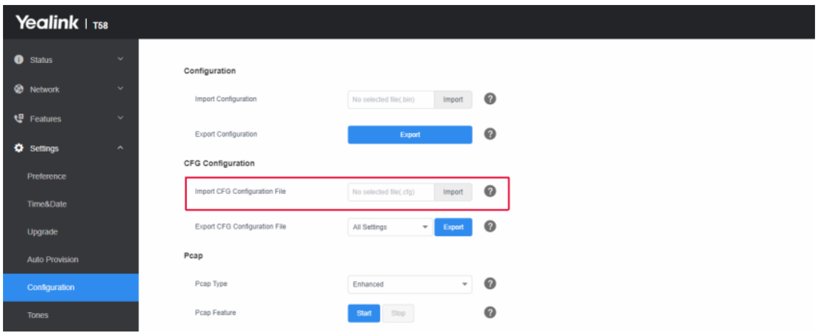
Parameters	Permitted Values	Default
features.hybrid_mode.enable	0 or 1	0
It enables or disables the hybrid mode feature. 0-Disabled, the Hybrid Mode configuration is not display on the phone user interface and the		

Account and Directory configurations are not display on the webuser interface. 1-Enabled		
account.X.enable	0 or 1	0
It enables or disables a specific account. 0-Disabled 1-Enabled		
account.X.sip_server.1.address	String within 256 characters	Blank
It configures the IP address (or domain name) of the SIP server Y(i.e. the SBC).		
account.X.sip_server.1.port	Integer from 0 to 65535	5060
It configures the port of the SIP server Y(i.e. the SBC).		
account.X.user_name	String within 99 characters	Blank
It configures the user name of a specific account for registration.		
account.X.password	String within 99 characters	Blank
It configures the password of a special account for registration and authentication.		
account.X.srtp_encryption	0, 1 or 2	0
Configures whether to use audio/video encryption service for account X. 0-Disabled, the IP phone will not use audio/voice encryption service. 1-Optional, the IP phone will negotiate with the other IP phone that which type of encryption service is to be used for the session. 2-Compulsory, the SRTP is required during a call.		
features.hybrid_mode.quick_ball.enable	0 or 1	0
It enables or disables the quick ball for quickly switching between Teams APP andSurvivability APP. 0-Disabled 1-Enabled		

2. Import the CFG resource file into a single phone

    a) Use the Teams phone's IP to access the web user interface (the default credential: admin/admin).

    b) On web user interface, click **Settings** -> **Configurations**, find **Import CFG Configuration File** and import the first CFG file from Step 1 to enable hybrid-mode feature. If the file is imported successfully, the phone will restart automatically with hybrid-mode feature enabled.

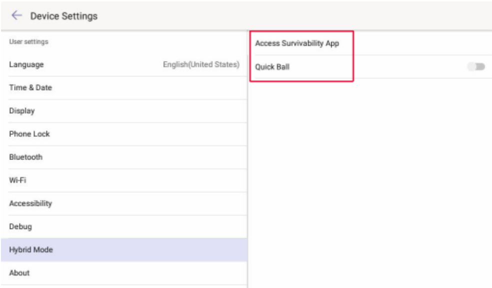


    c) Repeat the last step to import the second CFG file from Step 2 for account configuration. Or, you can configure the account via web user interface directly instead of auto provisioning.

    # To configure phones in batches, refer to Precondition.

3. Check whether all configurations take effect successfully

   a) On phone user interface, click **Settings** -> **Device Settings**, you can find that **Hybrid Mode** option appears, and you can open SIP APP by clicking **Access Survivability App**. You can also enable **Quick Ball** for a quick switch between Teams APP and SIP APP.



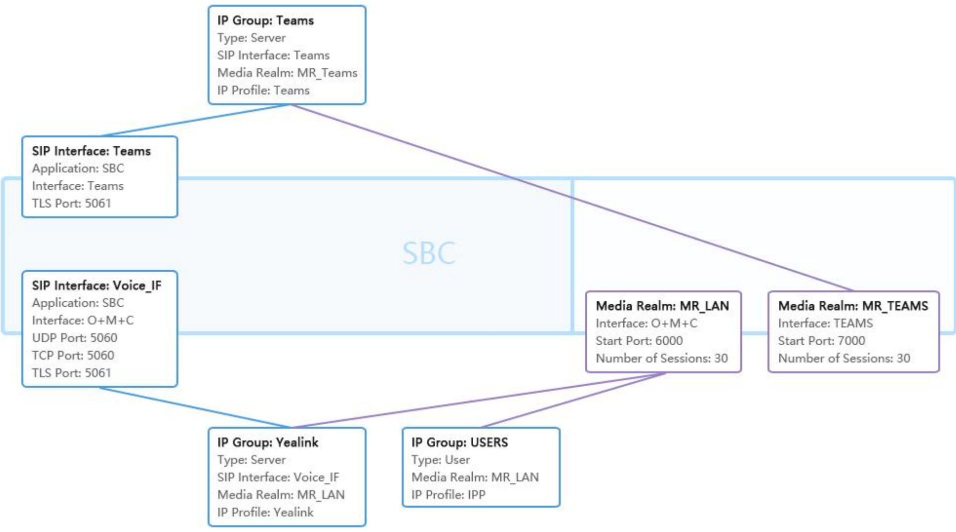
   b) With SIP APP open, you can find that the registered account is displayed in the account list on the right.

 indicates the account is available;

 indicates the account is abnormal.

Signaling and Media Topology of AudioCodes SBC

The signaling and media topology of AudioCodes SBC is as follows,



If you have trouble in setting up AuidoCodes SBC, please contact Audiocodes support team in your area.

Configuring AudioCodes SBC for Yealink Hybrid-mode Feature

1. Precondition

The local physical environment for SIP communicaiton needs to be set up in advance, and the parameters for network connection on AudioCodes SBC should be configured. There should be a network access for the local SIP calls and outbound PSTN calls:

- Separate **IP Interface**
- Separate **Ethernet Devices** and **Ethernet Groups**
- Separate **Physical Ports**
- Separate **TLS Contexts**
- Separate **Internal SRV Entry**

The Team Direct Routing deployment needs be set up and the SBC configuration needs be completed in advance.

2. Configuration steps

a) Create IP Profiles

b) Create Default Routing Policies

c) Create Default SRDs

d) Create Media Realms

e) Create SIP Interfaces

f) Create Proxy Policies

g) Create IP Groups

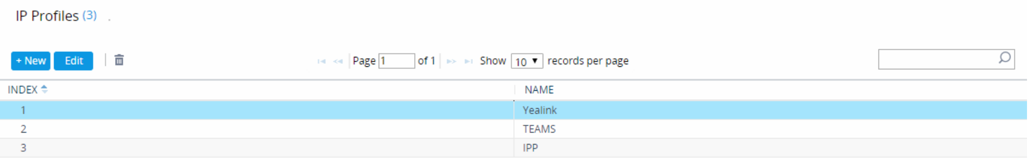
h) Create Classifications

i) Create Call Routing Policies

Steps for AudioCodes SBC Configuration

1. Create IP Profiles

Create IP Profiles to configure the media mode used by Teams phones during registrations and calls. Enable SRTP if encryption media is required. Click **CODERS & PROFILES** -> **IP Profiles**, and create an IP Profile for outbound PSTN calls, e.g. named **Yealink**. Attach the parameters for your reference.



Field	Recommended Parameter
Name	e.g. Yealink
SBC Media Security Mode	RTP
Extension Coders Group	e.g. AudioCodersGroups_0
Remote REFER Mode/Remote Replaces Mode/Remote 3xx Mode	Handle Locally

Create another IP Profile for local SIP calls, e.g. named **IPP**

Field	Recommended Parameter
Name	e.g. IPP
SBC Media Security Mode	RTP

2. Create Default Routing Policies

Create a default routing policy that applies to all call routings. Click **SBC** -> **Routing** -> **Routing Polices** and create a new item, e.g. named **Default_SBCRoutingPolicy**. Keep the original parameters for each configuration.

Routing Policies (1)

+ New | Edit | Page 1 of 1 | Show 10 records per page

INDEX	NAME	LDAP SERVERS GROUP NAME	LCR CALL DURATION [MIN]	DEFAULT CALL COST	LCR FEATURE
0	Default_SBCRoutingPolicy	--	1	Lowest Cost	Disable

3. Create Default SRDs

Create a single SRD and apply to all SIP Interfaces and IP Groups, and then invoke the routing policy you created earlier. Click **CORE ENTITIES** -> **SRDs** and create new item, e.g. named **DefaultSRD**.

SRDs (1)

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INDEX	NAME	SHARING POLICY	SBC OPERATION MODE	SBC ROUTING POLICY	MAX. NUMBER OF REGISTERED USERS	USER SECURITY MODE
0	DefaultSRD (#0)	Shared	B2BUA	Default_SBCRoutingPolicy	-1	Accept All

Field	Recommended Parameter
Name	e.g. DefaultSRD
SBC Routing Policy	e.g. Default_SBCRoutingPolicy

4. Create Media Realms

To create Media Realms, click **CORE ENTITIES** -> **Media Realms** and create a new item, e.g. named **MR_LAN**. You can randomly assign some ports to Media Realms, depending on the number of concurrent SIP calls across your organization.

Once configured, to apply Media Realms to specific calls, you need to assign them to SIP Interfaces and IP Groups.

Media Realms (3)

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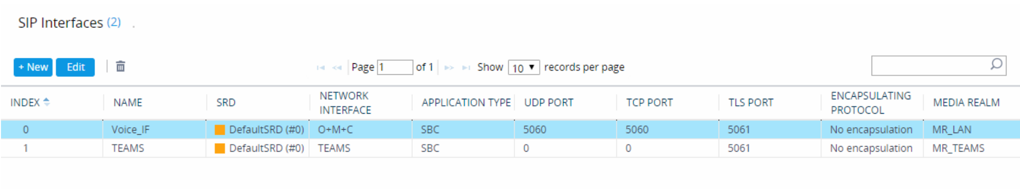
INDEX	NAME	IPV4 INTERFACE NAME	PORT RANGE START	NUMBER OF MEDIA SESSION LEGS	PORT RANGE END	DEFAULT MEDIA REALM
1	MR_LAN	O+M+C	6000	30	6119	No
2	MR_IPP	O+M+C	6500	30	6619	No
3	MR_TEAMS	TEAMS	7000	30	7119	No

Field	Recommended Parameter
Name	e.g. MR_LAN
IPv4 Interface Name	e.g. O+M+C
Port Range Start	6000
Number Of Media Session Legs	30

5. Create SIP Interfaces

To create the interface to be called when registering a phone, you need to configure IP interface, port, protocol, and media template for phone registration. Click **CORE ENTITIES** -> **SIP Interfaces** and create a new item, e.g. named **Voice_IF**.

(Optional) You can enable TLS authentication, but you need to configure TLS contexts in advance.



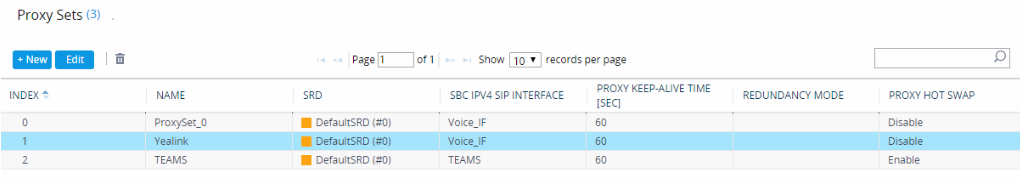
SIP Interfaces (2)

INDEX	NAME	SRD	NETWORK INTERFACE	APPLICATION TYPE	UDP PORT	TCP PORT	TLS PORT	ENCAPSULATING PROTOCOL	MEDIA REALM
0	Voice_IF	DefaultSRD (#0)	O+M+C	SBC	5060	5060	5061	No encapsulation	MR_LAN
1	TEAMS	DefaultSRD (#0)	TEAMS	SBC	0	0	5061	No encapsulation	MR_TEAMS

Field	**Recommended Parameter**
Name	e.g. Voice_IF
Network Interface	e.g. O+M+C
Media Realm	e.g. MR_LAN
TLS Context Name	e.g. LAN-TLS
TLS Mutual Authentication	Disable

6. Create Proxy Policies

Create a proxy policy for outbound PSTN calls and then invoke the SIP Interface created earlier. Click **CORE ENTITIES** -> **Proxy Set** and create a new item, e.g. named **Yealink**.



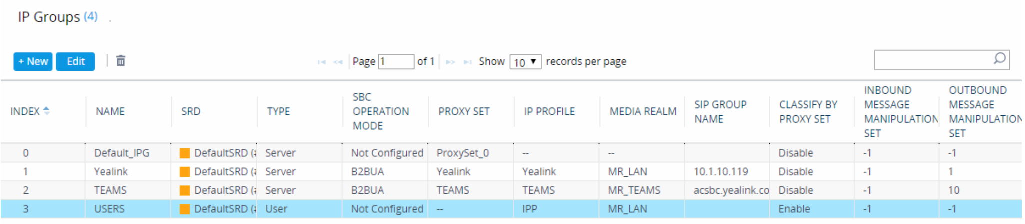
Proxy Sets (3)

INDEX	NAME	SRD	SBC IPV4 SIP INTERFACE	PROXY KEEP-ALIVE TIME [SEC]	REDUNDANCY MODE	PROXY HOT SWAP
0	ProxySet_0	DefaultSRD (#0)	Voice_IF	60		Disable
1	Yealink	DefaultSRD (#0)	Voice_IF	60		Disable
2	TEAMS	DefaultSRD (#0)	TEAMS	60		Enable

Field	**Recommended Parameter**
Name	e.g. Yealink
SBC IPv4 SIP Interface	e.g. Voice_IF
Proxy Keep-Alive	Using OPTIONS

7. Create IP Groups

To create an IP Group for local SIP calls, Click **CORE ENTITIES** -> **IP Group** and create a new item, e.g. named **USERS**.



IP Groups (4)

INDEX	NAME	SRD	TYPE	SBC OPERATION MODE	PROXY SET	IP PROFILE	MEDIA REALM	SIP GROUP NAME	CLASSIFY BY PROXY SET	INBOUND MESSAGE MANIPULATION SET	OUTBOUND MESSAGE MANIPULATION SET
0	Default_IPG	DefaultSRD (	Server	Not Configured	ProxySet_0	--	--		Disable	-1	-1
1	Yealink	DefaultSRD (	Server	B2BUA	Yealink	Yealink	MR_LAN	10.1.10.119	Disable	-1	1
2	TEAMS	DefaultSRD (	Server	B2BUA	TEAMS	TEAMS	MR_TEAMS	acsbc.yealink.co	Disable	-1	10
3	USERS	DefaultSRD (	User	Not Configured	--	IPP	MR_LAN		Enable	-1	-1

Field	**Recommended Parameter**
Name	e.g. USERS
Type	User
IP Profile	e.g. IPP
Media Realm	e.g. MR_LAN
DTLS Context	default
Username/Password	e.g. Admin/*

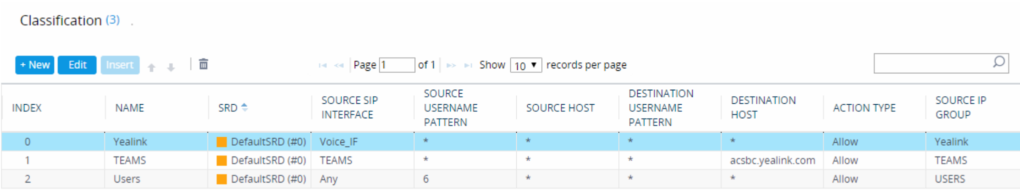
Create another IP Group for outbound PSTN calls, e.g. named **Yealink**.

Field	**Recommended Parameter**
Name	e.g. Yealink

Proxy Set	e.g. Yealink
IP Profile	e.g. Yealink
Media Realm	e.g. MR_LAN
SIP Group Name	e.g. 10.1.10.*
Classify By Proxy Set	Disable
SBC Operation Mode	B2BUA
DTLS Context	default
Outbound Message Manipulation Set	1
Proxy Keep-Alive using IP Group Settings	Enable
Username/Password	Admin/*

8. Create Classifications

To create Classification, you need to assign the incoming SIP dialog - initiating to a specific IP Group. Click **SBC** -> **Classification** and create a new item for outbound PSTN calls, e.g. named **Yealink**.



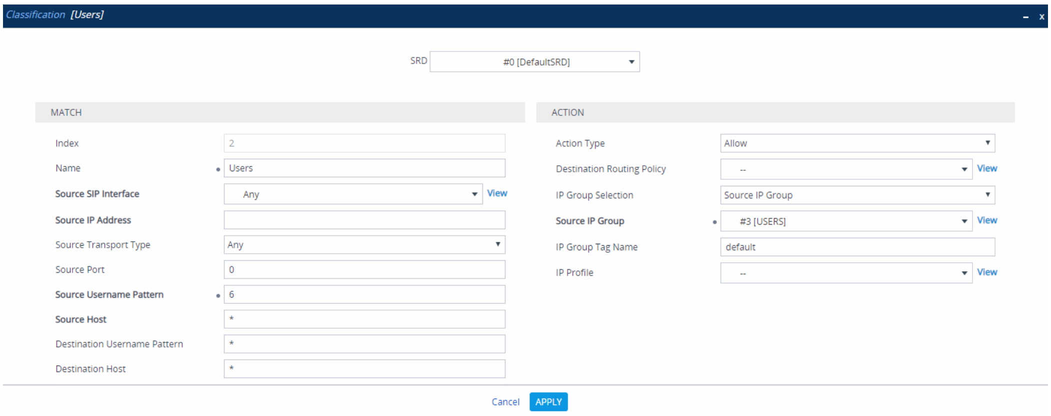
INDEX	NAME	SRD	SOURCE SIP INTERFACE	SOURCE USERNAME PATTERN	SOURCE HOST	DESTINATION USERNAME PATTERN	DESTINATION HOST	ACTION TYPE	SOURCE IP GROUP
0	Yealink	DefaultSRD (#0)	Voice_IF	*	*	*	*	Allow	Yealink
1	TEAMS	DefaultSRD (#0)	TEAMS	*	*	*	acsbc.yealink.com	Allow	TEAMS
2	Users	DefaultSRD (#0)	Any	6	*	*	*	Allow	USERS

Field	**Recommended Parameter**
Name	e.g. Yealink
Source SIP Interface	e.g. Voice_IF
Source IP Address	e.g. 10.1.10.*
Source Transport Type	UDP
Source Port	5060
Destination Routing Policy	e.g. Default_SBCRoutingPolicy
Source IP Group	e.g. Yealink

Create another item for local SIP calls, e.g. named **Users**.

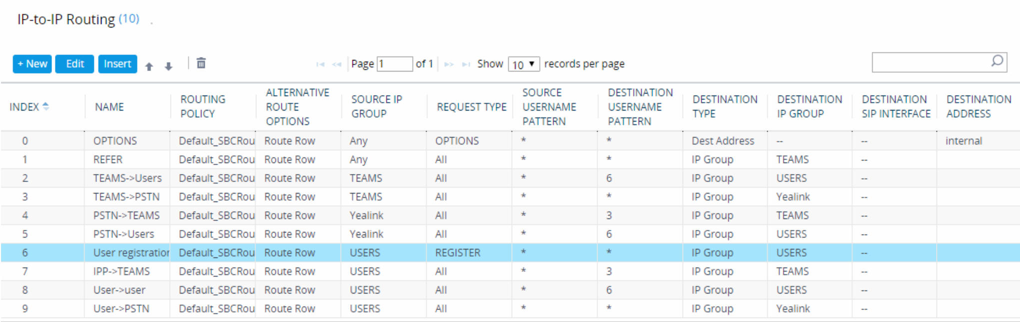
Field	**Recommended Parameter**
Name	e.g. Users
Source SIP Interface	Any
Source Username Pattern	6
Source IP Group	e.g. USERS

To Create SIP accounts, double click **SOURCE SIP INTERFACE** and create account rules. For example, you can set the user account starting with 6 but ending with any number and without any length limit. The following rules allow you to register a username starting with 6.



9. Create Call Routing Policies

To create Call Routing policies, click **SBC** -> **Routing** -> **IP to IP Routing** and create new items.



INDEX	NAME	ROUTING POLICY	ALTERNATIVE ROUTE OPTIONS	SOURCE IP GROUP	REQUEST TYPE	SOURCE USERNAME PATTERN	DESTINATION USERNAME PATTERN	DESTINATION TYPE	DESTINATION IP GROUP	DESTINATION SIP INTERFACE	DESTINATION ADDRESS
0	OPTIONS	Default_SBCRou	Route Row	Any	OPTIONS	*	*	Dest Address	--	--	internal
1	REFER	Default_SBCRou	Route Row	Any	All	*	*	IP Group	TEAMS	--	
2	TEAMS->Users	Default_SBCRou	Route Row	TEAMS	All	*	6	IP Group	USERS	--	
3	TEAMS->PSTN	Default_SBCRou	Route Row	TEAMS	All	*	*	IP Group	Yealink	--	
4	PSTN->TEAMS	Default_SBCRou	Route Row	Yealink	All	*	3	IP Group	TEAMS	--	
5	PSTN->Users	Default_SBCRou	Route Row	Yealink	All	*	6	IP Group	USERS	--	
6	User registratior	Default_SBCRou	Route Row	USERS	REGISTER	*	*	IP Group	USERS	--	
7	IPP->TEAMS	Default_SBCRou	Route Row	USERS	All	*	3	IP Group	TEAMS	--	
8	User->user	Default_SBCRou	Route Row	USERS	All	*	6	IP Group	USERS	--	
9	User->PSTN	Default_SBCRou	Route Row	USERS	All	*	*	IP Group	Yealink	--	

a) First of all, create a rule of user registration for SIP clients, e.g. named **User registration**, without any filtering, and passing all characters by default.

Field	**Recommended Parameter**
Name	e.g. User registration
Source IP Group	e.g. USERS
Request Type	REGISTER
ReRoute IP Group	Any
Destination IP Group	e.g. USERS

b) Create a routing policy for local SIP calls, e.g. named **User→User**.

Field	**Recommended Parameter**
Name	e.g. User->User
Source IP Group	e.g. USERS
Destination Username Pattern	6
ReRoute IP Group	Any
Destination IP Group	e.g. USERS

c) Create routing policies for the calls from SIP to PSTN and PSTN to SIP, e.g. named **Users→PSTN** and **PSTN→Users**.

Users→PSTN	
Field	**Recommended Parameter**
Name	e.g. User->PSTN
Source IP Group	e.g. USERS
ReRoute IP Group	Any

Destination IP Group	e.g. Yealink
PSTN→Users	
Field	**Recommended Parameter**
Name	e.g. PSTN->Users
Source IP Group	e.g. Yealink
Destination Username Pattern	6
ReRoute IP Group	Any
Destination IP Group	e.g. USERS

d) Create routing policies for the calls from Teams to SIP and SIP to Teams, e.g. named **IPP→TEAMS** and **Teams→Users**.

IPP→TEAMS	
Field	**Recommended Parameter**
Name	e.g. IPP->TEAMS
Source IP Group	e.g. USERS
Destination Username Pattern	6
ReRoute IP Group	Any
Destination IP Group	e.g. TEAMS
TEAMS->Users	
Field	**Recommended Parameter**
Name	e.g. TEAMS->Users
Source IP Group	e.g. TEAMS
Destination Username Pattern	6
ReRoute IP Group	Any
Destination IP Group	e.g. USERS

Several routing policies applied to Teams Direct Routing deployment need to be configured in advance.