

Maxincom

MUC1002/1004/2008/2016

Release Notes of Versions

20/1/12/13.1.0.29-beta06

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MUC1002/1004/2008/2016 Upgrade Firmware Notice

- We strongly recommend you to back up the configurations before you upgrade.
- You need to RESET the device to make it work properly if you want to downgrade the firmware.
- Backup files from higher firmware version cannot be restored to the device with lower firmware version.
- It's recommended that you clear the browser cache after upgrade

✧ Release Notes of Version 20/1/12/13.1.0.29-beta06

1. Introduction

- (1) Firmware Version: 20.1.0.29-beta06, 1.1.0.29-beta06, 12.1.0.29-beta06, 13.1.0.29-beta06
- (2) Applicable Model: MUC1002, MUC1004, MUC2008, MUC2016
- (3) Release Date: July 29, 2019

CHANGES SINCE FIRMWARE RELEASE 20/1/12/13.1.0.29-beta05

2. New Features

None

3. Optimization

- (1) Repackage the image to facilitate customer identification. Upgrade the database version to avoid online upgrade image, database is not updated.

4. Bug Fixes

None

5. New Features Descriptions

None

6. Optimization Description

- (1) Repackage the image to facilitate customer identification. Upgrade the database version to avoid online upgrade image, database is not updated.

7. Bug Fixes Description

None.

✧ Release Notes of Version 20/1/12/13.1.0.29-beta05

1. Introduction

- (1) Firmware Version: 20.1.0.29-beta05, 1.1.0.29-beta05, 12.1.0.29-beta05, 13.1.0.29-beta05
- (2) Applicable Model: MUC1002, MUC1004, MUC2008, MUC2016
- (3) Release Date: July 22, 2019

CHANGES SINCE FIRMWARE RELEASE 20/1/12/13.1.0.29-beta04

2. New Features

None

3. Optimization

- (2) On the basis of the original Call Group function, add, modify, or remove the Call Group will directly modify the relevant data of the queues, extension and Ring Groups that has been added with Call Group.
- (3) Modifying or deleting the extension will directly modify the data of the added Call Group and the queue and ring group corresponding to the Call Group.
- (4) Compatible with Agent Info in Call Center to display Agent status.

4. Bug Fixes

None

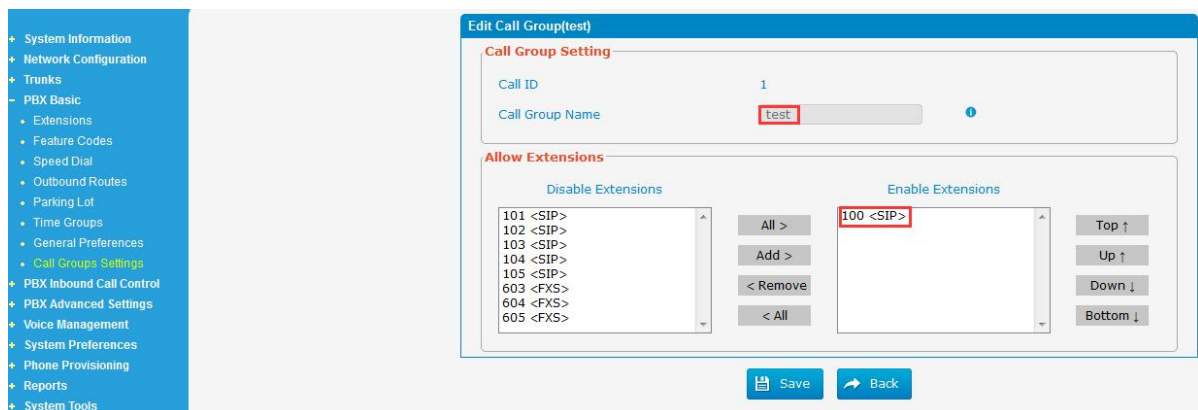
5. New Features Descriptions

None

6. Optimization Description

- (2) On the basis of the original Call Group function, add, modify, or remove the Call Group will directly modify the relevant data of the queues, extension and Ring Groups that has been added with Call Group.

Figure ->6.1 Add extension 100 to Call Group “test”



The screenshot shows the 'Edit Call Group(test)' window. Under the 'Call Group Setting' section, the 'Call ID' is 1 and the 'Call Group Name' is 'test'. The 'Allow Extensions' section has two lists: 'Disable Extensions' and 'Enable Extensions'. The 'Enable Extensions' list contains '100 <SIP>' which is highlighted with a red box. The 'Disable Extensions' list contains '101 <SIP>', '102 <SIP>', '103 <SIP>', '104 <SIP>', '105 <SIP>', '603 <FXS>', '604 <FXS>', and '605 <FXS>'. Between the lists are buttons: 'All >', 'Add >', '< Remove', and '< All'. To the right of the 'Enable Extensions' list are buttons: 'Top ↑', 'Up ↑', 'Down ↓', and 'Bottom ↓'. At the bottom of the window are 'Save' and 'Back' buttons.

Figure ->6.2 Add Call Group “test” to Queue 820

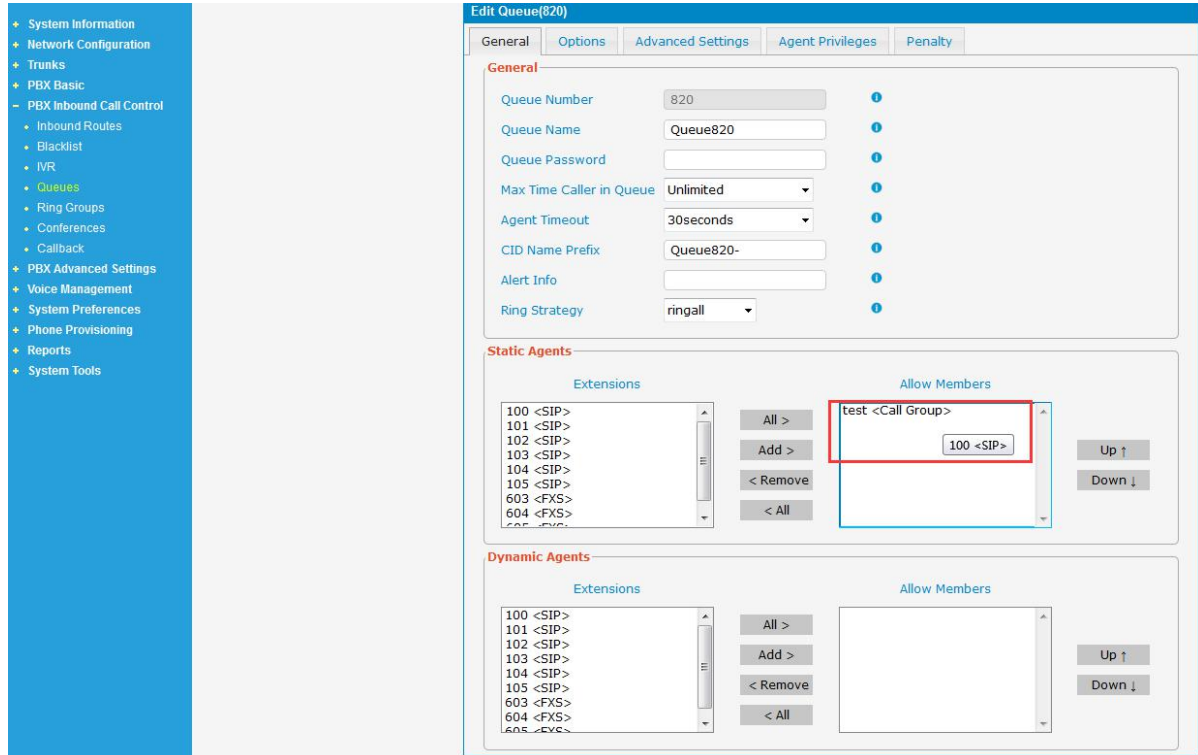


Figure ->6.3 Add Call Group “test” to Ring Group 920

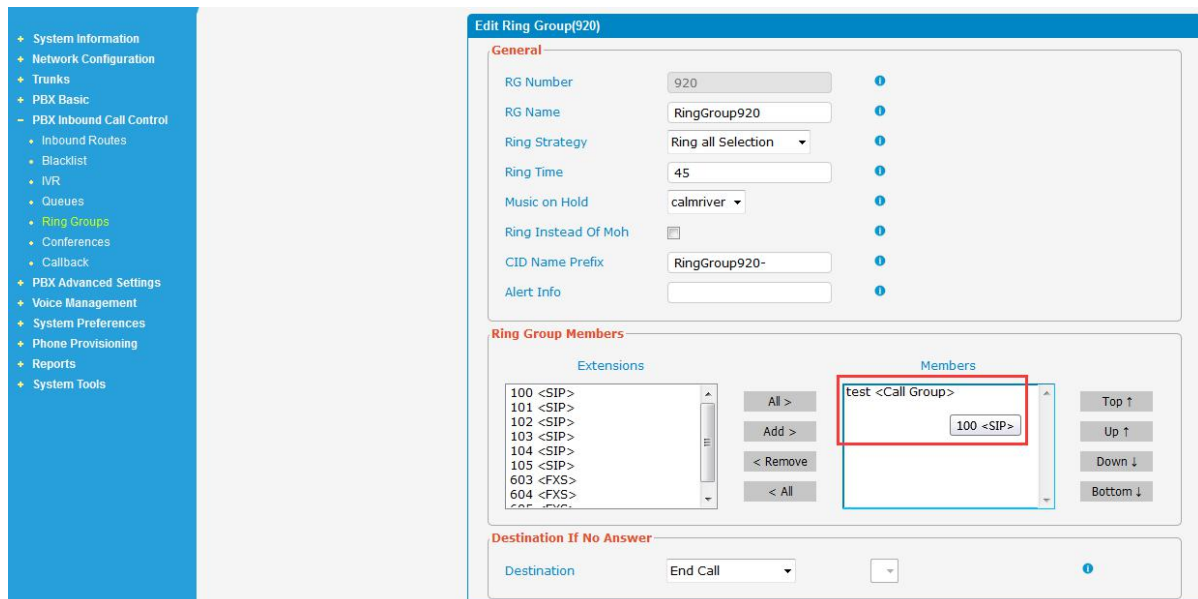


Figure->6.4 Add extension 101 to Call Group “test”

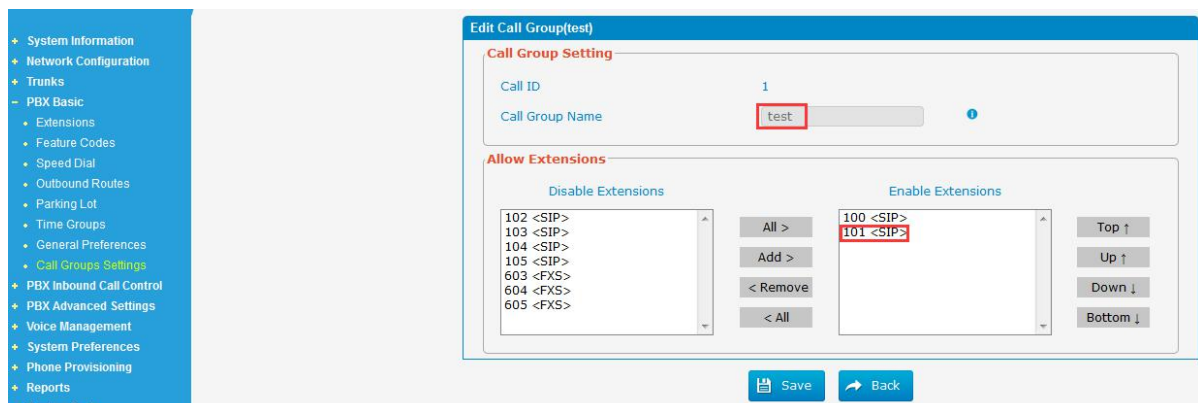
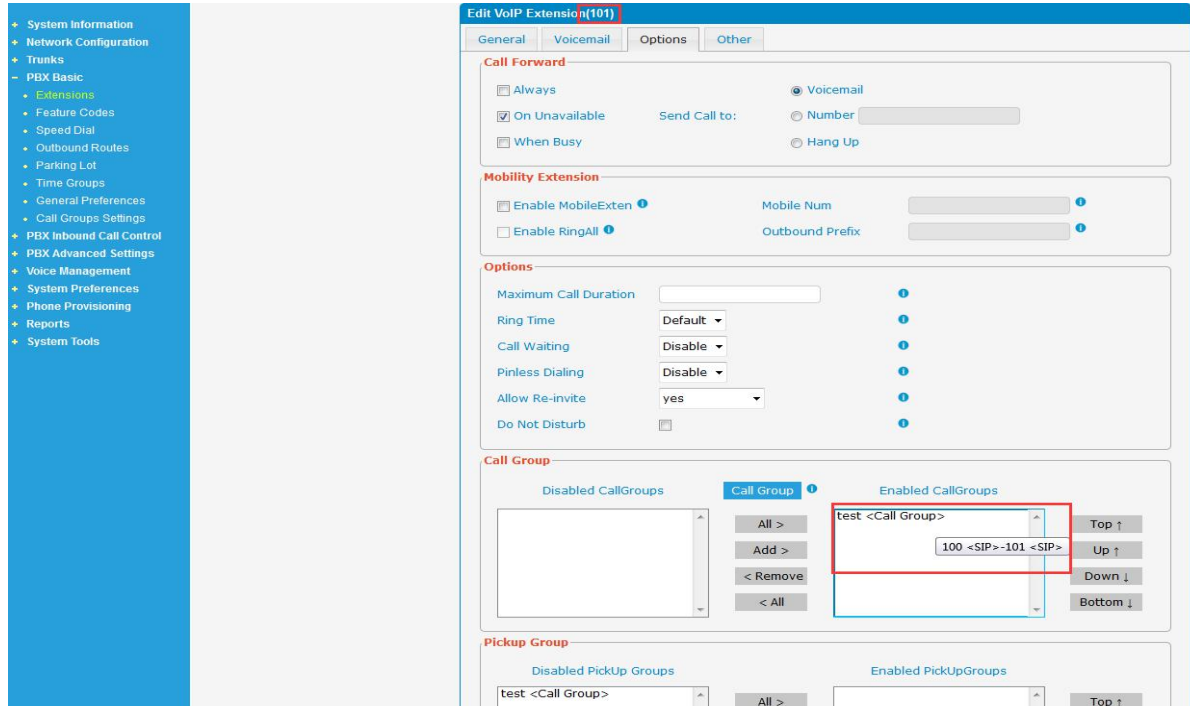


Figure->6.5 Call Group “test” in Extension 101



Edit VoIP Extension(101)

Call Forward

- ☐ Always
- ☒ On Unavailable Send Call to: ☐ Number ☐ Hang Up
- ☐ When Busy

Mobility Extension

- ☐ Enable MobileExten Mobile Num
- ☐ Enable RingAll Outbound Prefix

Options

- Maximum Call Duration
- Ring Time
- Call Waiting
- Pinless Dialing
- Allow Re-invite
- Do Not Disturb ☐

Call Group

Disabled CallGroups

Enabled CallGroups

- test <Call Group>
- 100 <SIP>-101 <SIP>

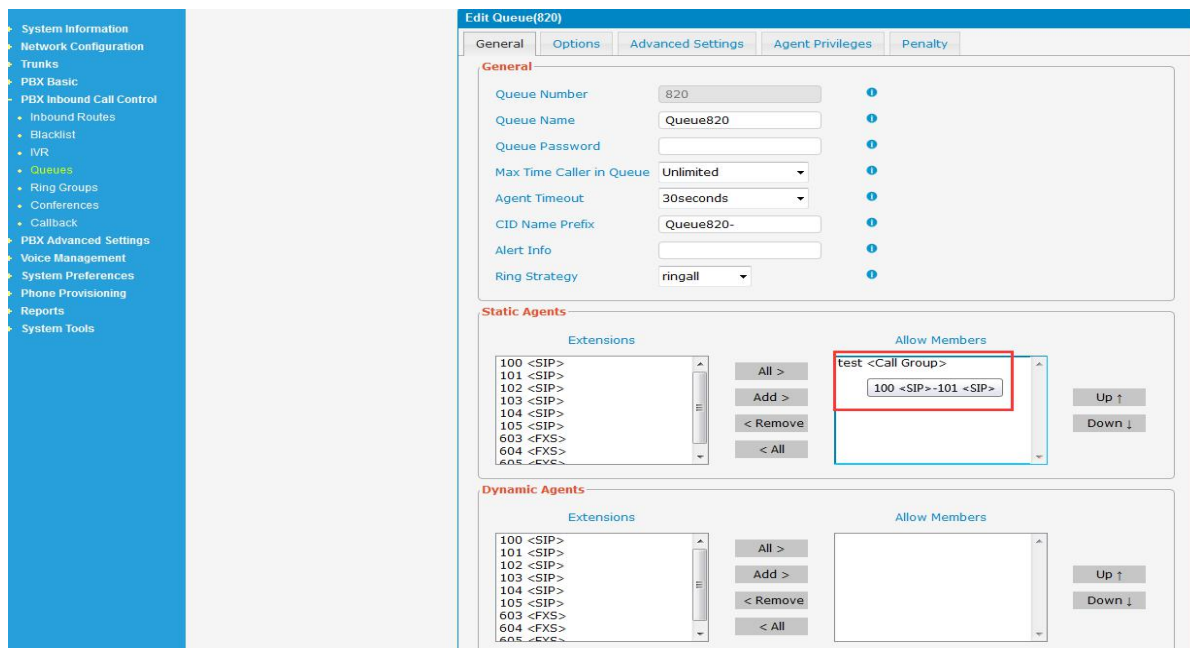
Pickup Group

Disabled PickUp Groups

Enabled PickUpGroups

- test <Call Group>

Figure->6.6 After added Extension 101 in Queue 820



Edit Queue(820)

General

- Queue Number
- Queue Name
- Queue Password
- Max Time Caller in Queue
- Agent Timeout
- CID Name Prefix
- Alert Info
- Ring Strategy

Static Agents

Extensions

- 100 <SIP>
- 101 <SIP>
- 102 <SIP>
- 103 <SIP>
- 104 <SIP>
- 105 <SIP>
- 603 <FXS>
- 604 <FXS>
- 605 <FXS>

Allow Members

- test <Call Group>
- 100 <SIP>-101 <SIP>

Dynamic Agents

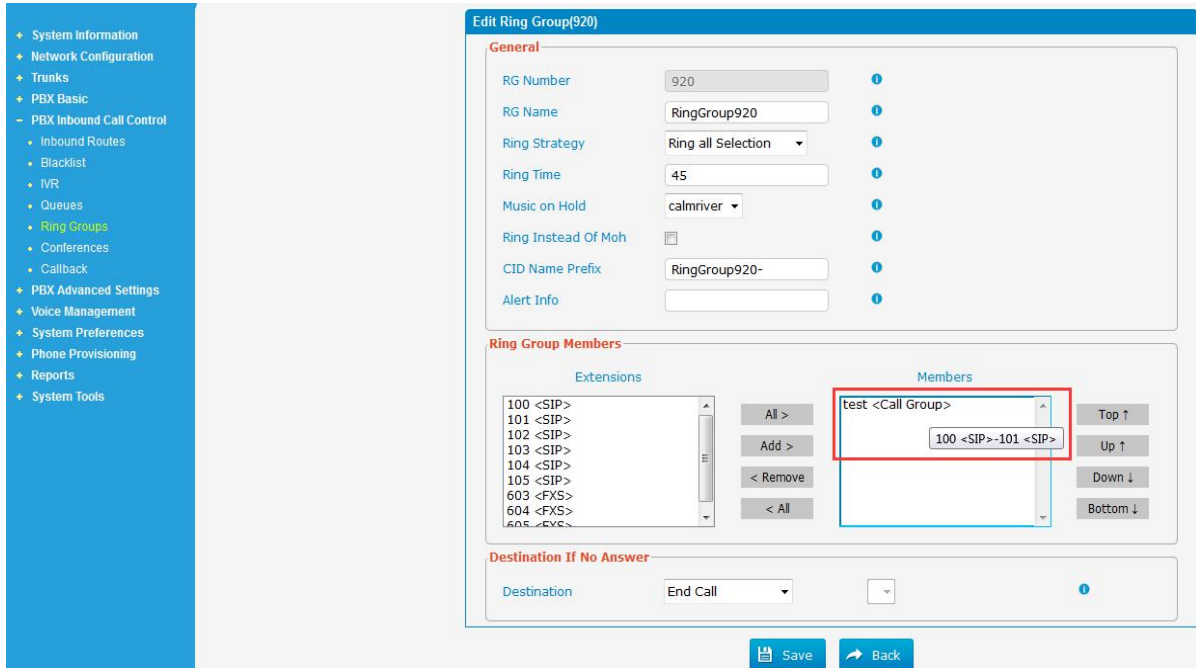
Extensions

- 100 <SIP>
- 101 <SIP>
- 102 <SIP>
- 103 <SIP>
- 104 <SIP>
- 105 <SIP>
- 603 <FXS>
- 604 <FXS>
- 605 <FXS>

Allow Members

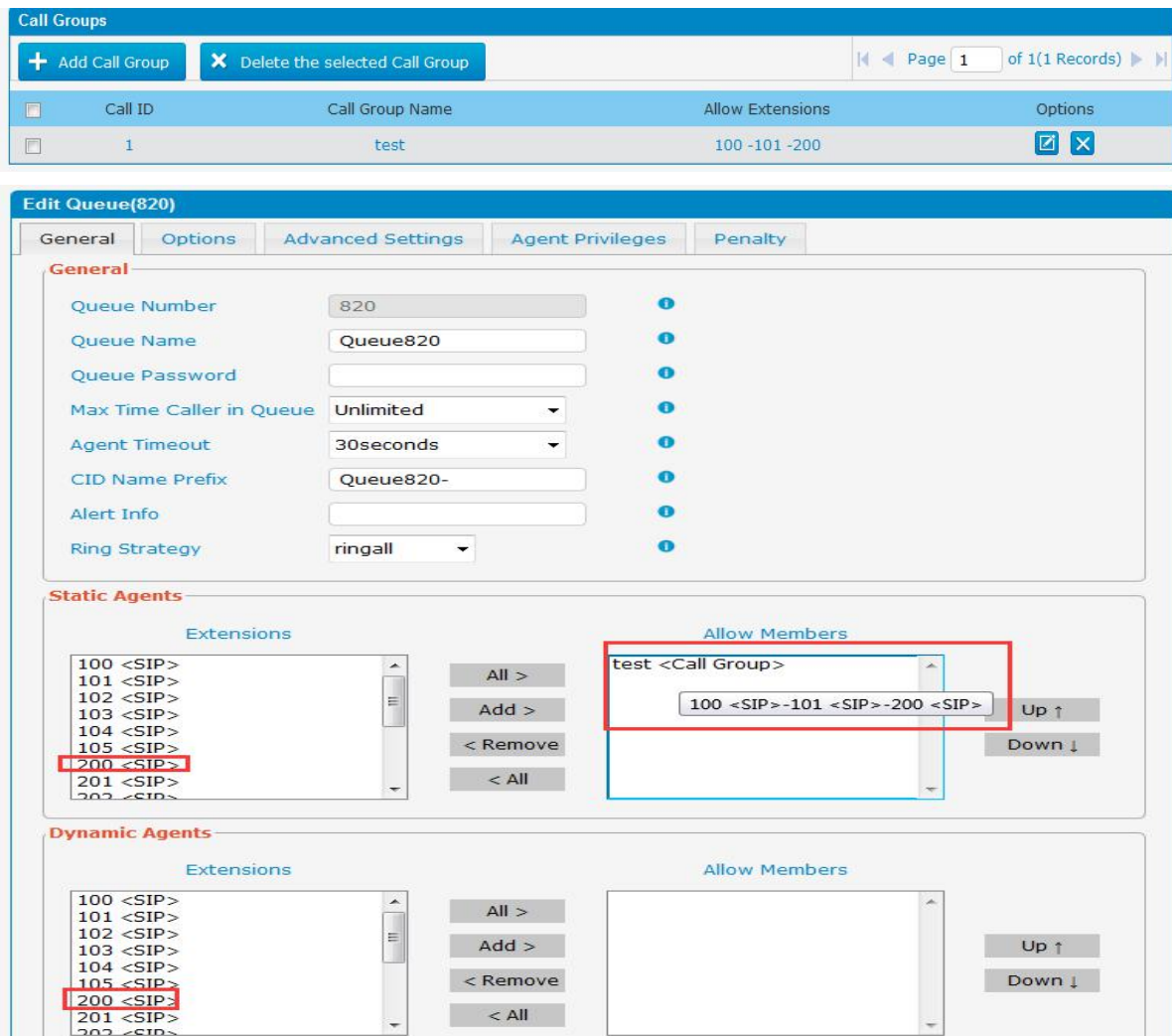
-

Figure->6.7 After added Extension 101 in Ring Group 920



- (3) Modifying or deleting the extension will directly modify the data of the added Call Group and the queue and ring group corresponding to the Call Group.

Figure->6.8 Before delete Extension 200



Edit Ring Group(920)

General

RG Number: 920
RG Name: RingGroup920
Ring Strategy: Ring all Selection
Ring Time: 45
Music on Hold: calmriver
Ring Instead Of Moh: ☐
CID Name Prefix: RingGroup920-
Alert Info:

Ring Group Members

Extensions

100 <SIP>
101 <SIP>
102 <SIP>
103 <SIP>
104 <SIP>
105 <SIP>
200 <SIP>
201 <SIP>
202 <SIP>

All >
Add >
< Remove
< All

Members

test <Call Group>
100 <SIP>-101 <SIP>-200 <SIP>

Top ↑
Up ↑
Down ↓
Bottom ↓

Destination If No Answer

Destination: End Call

Figure->6.8 After deleted Extension 200

Call Groups

+ Add Call Group X Delete the selected Call Group

Page 1 of 1(1 Records)

Call ID	Call Group Name	Allow Extensions	Options
1	test	100 -101	☒ ☐

Edit Queue(820)

General Options Advanced Settings Agent Privileges Penalty

General

Queue Number: 820
Queue Name: Queue820
Queue Password:
Max Time Caller in Queue: Unlimited
Agent Timeout: 30seconds
CID Name Prefix: Queue820-
Alert Info:
Ring Strategy: ringall

Static Agents

Extensions

100 <SIP>
101 <SIP>
102 <SIP>
103 <SIP>
104 <SIP>
105 <SIP>
201 <SIP>
202 <SIP>

All >
Add >
< Remove
< All

Allow Members

test <Call Group>
100 <SIP>-101 <SIP>

Up ↑
Down ↓

Dynamic Agents

Extensions

100 <SIP>
101 <SIP>
102 <SIP>
103 <SIP>
104 <SIP>
105 <SIP>
201 <SIP>
202 <SIP>

All >
Add >
< Remove
< All

Allow Members

Up ↑
Down ↓

Edit Ring Group(920)

General

RG Number: 920
RG Name: RingGroup920
Ring Strategy: Ring all Selection
Ring Time: 45
Music on Hold: calmriver
Ring Instead Of Moh: ☐
CID Name Prefix: RingGroup920-
Alert Info:

Ring Group Members

Extensions

100 <SIP>
101 <SIP>
102 <SIP>
103 <SIP>
104 <SIP>
105 <SIP>
201 <SIP>
202 <SIP>
203 <SIP>

All >
Add >
< Remove
< All

Members

test <Call Group>
100 <SIP>-101 <SIP>

Top ↑
Up ↑
Down ↓
Bottom ↓

Destination If No Answer

Destination: End Call

(4) Compatible with “Agent Info” in Call Center to display Agent status.

Figure->Before Optimization

Call Groups

+ Add Call Group X Delete the selected Call Group Page 1 of 0(0 Records)

	Call Group Name	Allow Extensions	Options
☐	sales	501 -511 -502	☒ ☒
☐	bbb	506 -601	☒ ☒

Change Password
Logout(admin)

Edit Queue(820)

General Options Advanced Settings Agent Privileges Penalty

General

Queue Number: 820
Queue Name: Queue820
Queue Password:
Max Time Caller in Queue: Unlimited
Agent Timeout: 30seconds
CID Name Prefix: Queue820-
Alert Info:
Ring Strategy: ringall

Static Agents

Extensions

102 <SIP>
500 <SIP>
501 <SIP>
502 <SIP>
503 <SIP>
505 <SIP>
506 <SIP>
507 <SIP>
508 <SIP>

All >
Add >
< Remove
< All

Allow Members

504 <SIP>
sales <Call Group>
bbb <Call Group>

Up ↑
Down ↓

- General
- Queue Report
- Queue Info
- Answer Unanswer
- Agent Calls
- Abandoned Calls
- Detailed Abandoned Calls
- Detailed Answered Calls
- SLA Breaches
- Detailed SLA Breaches
- CallBack Report
- Detailed Failed Callback
- Agent Info

Agent Info											
Queue	ALL	Start	2019-06-24 03:14	End	2019-07-24 03:14	Search	Download				
Type	Agent	Queue	Total Login Time	Successful Callback	TotalCall	Answered	RingNoAnswer	RingTime	AvrRingTime	TalkTime	AvrTalkTime
Static	504	820	00:00:21	0	0	0	0	00:00:00	00:00:00	00:00:00	00:00:00
Static	1	820	00:00:00	0	0	0	0	00:00:00	00:00:00	00:00:00	00:00:00
Static	2	820	00:00:00	0	0	0	0	00:00:00	00:00:00	00:00:00	00:00:00
TotalCall				0	0	0	0	00:00:00			

Change Password
Logout(callcenter)

Call Group ID

Figure->After Optimization

- General
- Queue Report
- Queue Info
- Answer Unanswer
- Agent Calls
- Abandoned Calls
- Detailed Abandoned Calls
- Detailed Answered Calls
- SLA Breaches
- Detailed SLA Breaches
- CallBack Report
- Detailed Failed Callback
- Agent Info

Agent Info											
Queue	ALL	Start	2019-06-24 03:23	End	2019-07-24 03:23	Search	Download				
Type	Agent	Queue	Total Login Time	Successful Callback	TotalCall	Answered	RingNoAnswer	RingTime	AvrRingTime	TalkTime	AvrTalkTime
Static	504	820	00:01:55	0	0	0	0	00:00:00	00:00:00	00:00:00	00:00:00
Static	501	820	00:01:55	0	0	0	0	00:00:00	00:00:00	00:00:00	00:00:00
Static	511	820	00:01:55	0	0	0	0	00:00:00	00:00:00	00:00:00	00:00:00
Static	502	820	00:01:55	0	0	0	0	00:00:00	00:00:00	00:00:00	00:00:00
Static	506	820	00:00:04	0	0	0	0	00:00:00	00:00:00	00:00:00	00:00:00
Static	601	820	00:01:55	0	0	0	0	00:00:00	00:00:00	00:00:00	00:00:00
TotalCall				0	0	0	0	00:00:00			

Change Password
Logout(callcenter)

7. Bug Fixes Description

None.

✧ Release Notes of Version 20/1/12/13.1.0.29-beta04

1. Introduction

- (1) Firmware Version: 20.1.0.29-beta04, 1.1.0.29-beta04, 12.1.0.29-beta04, 13.1.0.29-beta04
- (2) Applicable Model: MUC1002, MUC1004, MUC2008, MUC2016
- (3) Release Date: June 20, 2019

CHANGES SINCE FIRMWARE RELEASE 20/1/12/13.1.0.29-beta03

2. New Features

None

3. Optimization

- (1) In the queue, move the extension to the dynamic agent. The corresponding number in the static agent should be removed.
- (2) Delete one of extensions in the Call Group, and the corresponding Call Group, queues and ring groups containing this extension will automatically delete this extension. When the deleted extension is the last member of the Call Group, the Call Group will be deleted together.

4. Bug Fixes

- (1) In the initial state, the extension dials 820 or 920, it can't get through. You need to save it again before you can get through.
- (2) The test of "Number" in Call Forward in the extension interface failed.
- (3) In the initial state, after deleting the extension, the same extension number in the queue has not been deleted.

5. New Features Descriptions

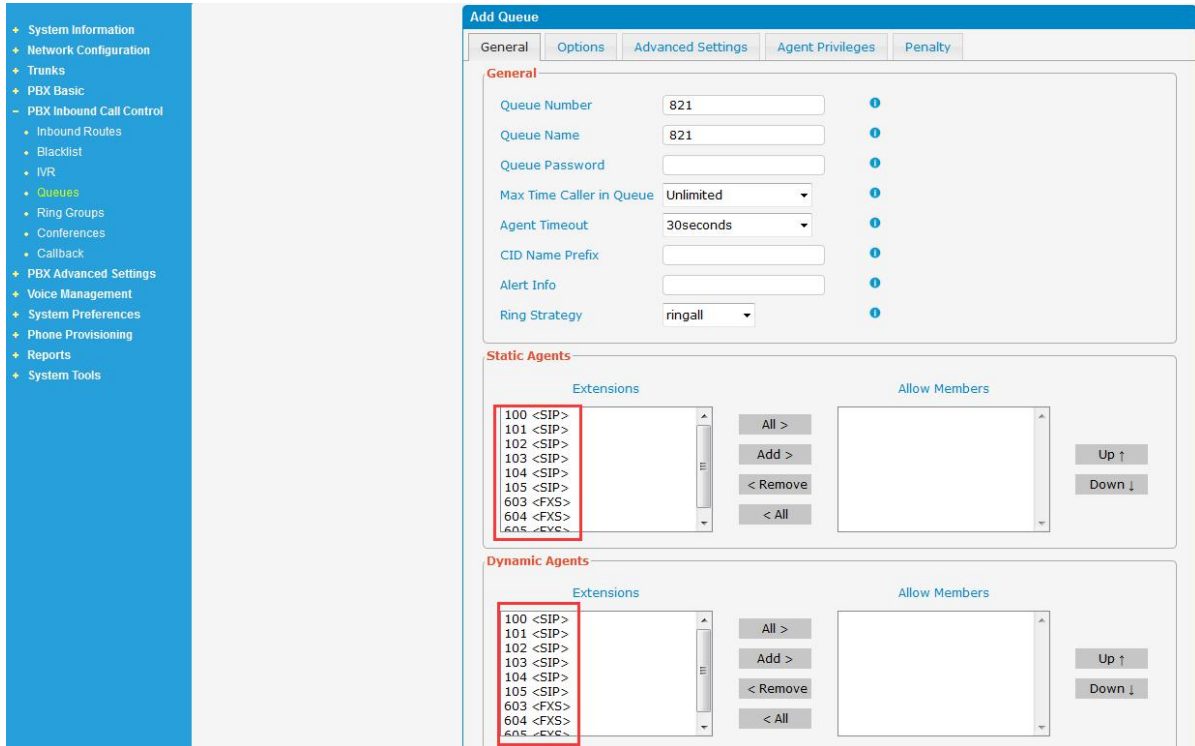
None.

6. Optimization Description

- (1) In the queue, move the extension to the dynamic seat. The corresponding number in the static seat should be removed.

Path : PBX Inbound Call Control -> Queues -> Add Queue (Edit Queue)

Figure ->6.1 Before move 101



Add Queue

General | Options | Advanced Settings | Agent Privileges | Penalty

General

Queue Number: 821

Queue Name: 821

Queue Password:

Max Time Caller in Queue: Unlimited

Agent Timeout: 30seconds

CID Name Prefix:

Alert Info:

Ring Strategy: ringall

Static Agents

Extensions: 100 <SIP>, 101 <SIP>, 102 <SIP>, 103 <SIP>, 104 <SIP>, 105 <SIP>, 603 <FXS>, 604 <FXS>, 605 <FXS>

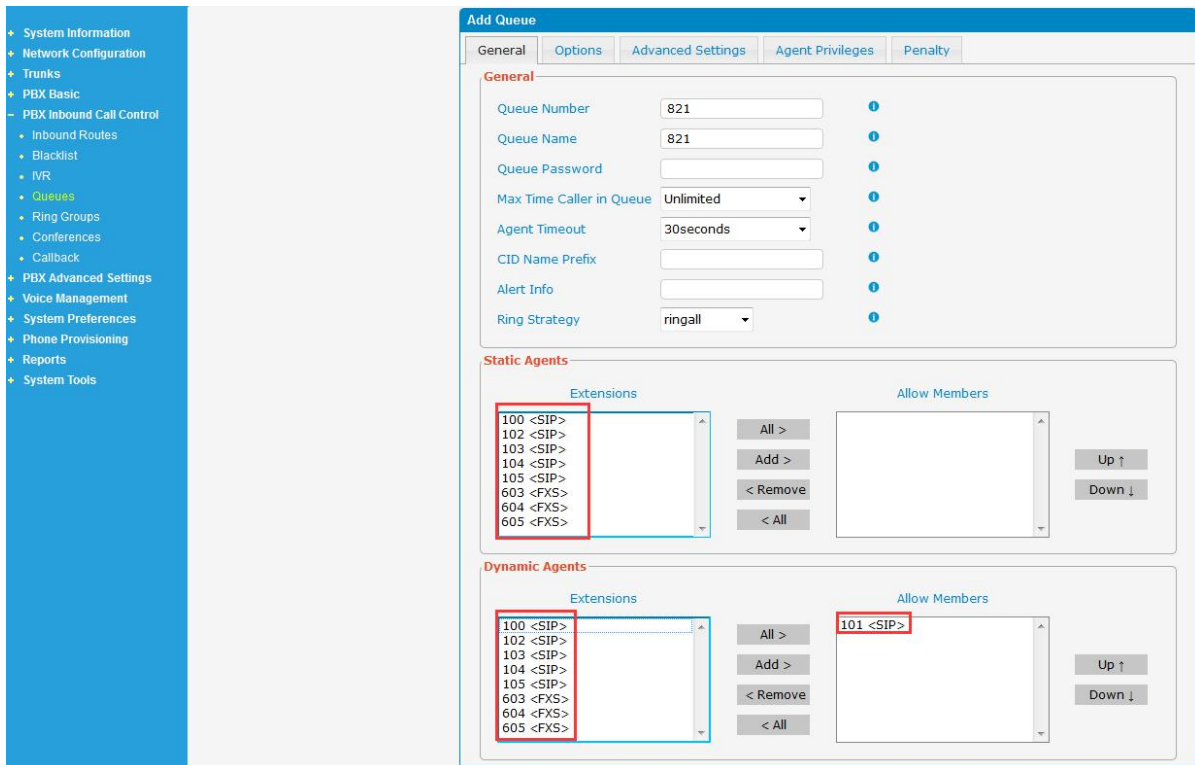
Allow Members:

Dynamic Agents

Extensions: 100 <SIP>, 101 <SIP>, 102 <SIP>, 103 <SIP>, 104 <SIP>, 105 <SIP>, 603 <FXS>, 604 <FXS>, 605 <FXS>

Allow Members:

Figure->6.2 After move 101



Add Queue

General | Options | Advanced Settings | Agent Privileges | Penalty

General

Queue Number: 821

Queue Name: 821

Queue Password:

Max Time Caller in Queue: Unlimited

Agent Timeout: 30seconds

CID Name Prefix:

Alert Info:

Ring Strategy: ringall

Static Agents

Extensions: 100 <SIP>, 102 <SIP>, 103 <SIP>, 104 <SIP>, 105 <SIP>, 603 <FXS>, 604 <FXS>, 605 <FXS>

Allow Members:

Dynamic Agents

Extensions: 100 <SIP>, 102 <SIP>, 103 <SIP>, 104 <SIP>, 105 <SIP>, 603 <FXS>, 604 <FXS>, 605 <FXS>

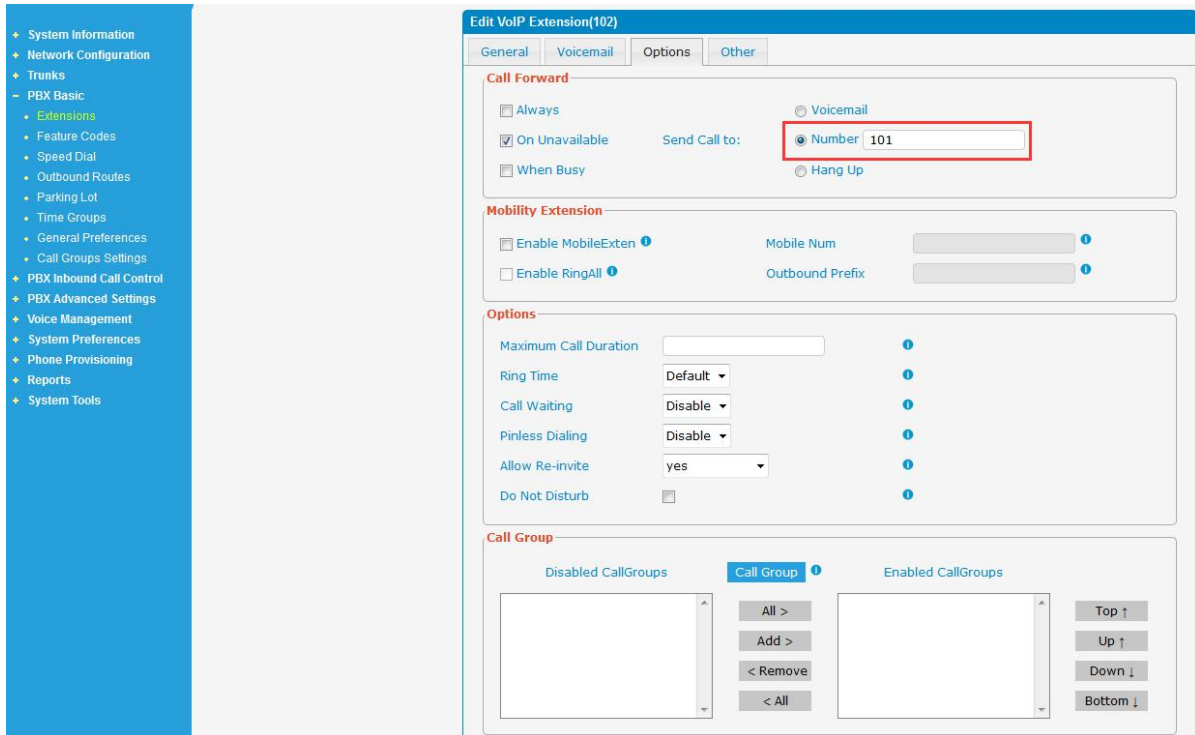
Allow Members: 101 <SIP>

- (2) Delete one of extensions in the Call Group, and the corresponding Call Group, queues and ring groups containing this extension will automatically delete this extension. When the deleted extension is the last member of the Call Group, the Call Group will be deleted together.

7. Bug Fixes Description

- (1) In the initial state, the extension dials 820 or 920, it can't get through. You need to save it again before you can get through.
- (2) The test of "Number" in Call Forward in the extension interface failed.

Figure->7.1 Use "Number" in extension



The screenshot shows the 'Edit VoIP Extension(102)' configuration page. The 'Call Forward' section is expanded, and the 'Send Call to:' field is set to 'Number 101', which is highlighted with a red rectangle. The 'Options' section shows various settings like 'Maximum Call Duration', 'Ring Time', 'Call Waiting', 'Pinless Dialing', 'Allow Re-invite', and 'Do Not Disturb'. The 'Call Group' section at the bottom shows 'Disabled CallGroups' and 'Enabled CallGroups' lists with associated buttons.

There are two registered extensions 101 and 104, as shown in the figure above, set the value of "Number" to 101 in extension 102 (extension 102 is not available), then extension 104 dials extension 102 and waits for the dial result.

Before Fixed:

There's no ring on extension 101.

After Fixed:

After hearing a voice prompt, extension 101 started ringing.

- (3) In the initial state, after deleting the extension, the same extension number in the queue has not been deleted.

✧ Release Notes of Version 20/1/12/13.1.0.29-beta03

1. Introduction

- (1) Firmware Version: 20.1.0.29-beta03, 1.1.0.29-beta03, 12.1.0.29-beta03, 13.1.0.29-beta03
- (2) Applicable Model: MUC1002, MUC1004, MUC2008, MUC2016
- (3) Release Date: June 5, 2019

CHANGES SINCE FIRMWARE RELEASE 20/1/12/13.1.0.29-beta02

2. New Features

None

3. Optimization

None

4. Bug Fixes

- (1) Fixed outside call problems

5. New Features Descriptions

None

6. Bug Fixes Description

- (1) Fixed an issue with an outside line failing to call directly into an extension via the SIP Trunk

Preparation for the test:

Register extensions 206 and 101 on two different IPPBX (such as 192.168.6.97 and 192.168.6.48), register the SIP Trunk(48pbx) on 97, and remove one bit from the dial rule configured in 97.

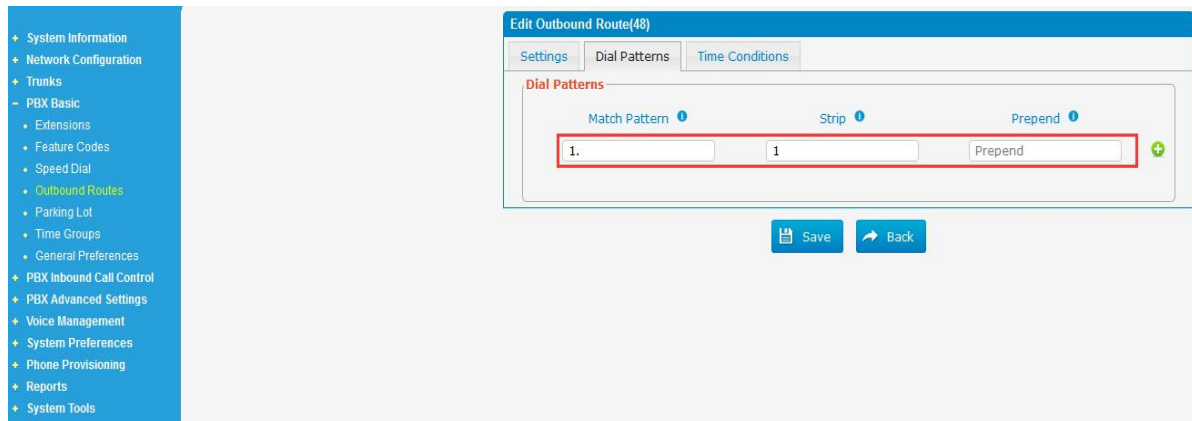
Figure 1-1 SIP Trunk of 48pbx in 192.168.6.97

Trunk Status						
Page 1 of 1(7 Records)						
Status	Trunk Type	Trunk Name	SIP/IAX	Transport	User Name/Signal	Hostname/Port
Registered	Trunk	GW_1003	SIP	udp	1003	192.168.6.55
Registered	Trunk	elastix3005	SIP	udp	3005	192.168.6.252
OK (1 ms)	Service Provider	95pbx	SIP	udp	--	192.168.6.95
OK (2 ms)	Service Provider	48pbx	SIP	udp	--	192.168.6.48
OK (2 ms)	Service Provider	96pbx_vpn	SIP	udp	--	10.8.0.1
Unavailable	FXO	pstn6	--	--	--	Port 6
Unavailable	GSM	GSM3	--	--	--	Port 3

Figure 1-2 SIP Trunk of 97 in 192.168.6.48

Trunk Status						
Page 1 of 1(1 Records)						
Status	Trunk Type	Trunk Name	SIP/IAX	Transport	User Name/Signal	Hostname/Port
OK (1 ms)	Service Provider	97	SIP	udp	--	192.168.6.97

Figure 1-3 Outbound Routes of 48 in 192.168.6.97



Dial Test:

<1> Before Fixed:

When the SIP Trunk (97) is registered on 48 but the inbound route with the 97 relay is not registered, dial 1101 on 206 and you will not be able to connect to extension 101.

<2>After Fixed:

When the SIP Trunk (97) is registered on 48 but the inbound route with the 97 relay is not registered, dial 1101 on 206 and you can connect to extension 101.

✧ Release Notes of Version 20/1/12/13.1.0.29-beta02

1. Introduction

- (1) Firmware Version: 20.1.0.29-beta02, 1.1.0.29-beta02, 12.1.0.29-beta02, 13.1.0.29-beta02
- (2) Applicable Model: MUC1002, MUC1004, MUC2008, MUC2016
- (3) Release Date: May 6, 2019

CHANGES SINCE FIRMWARE RELEASE 20/1/12/13.1.0.29-beta01

2. New Features

- (1) Call Groups Settings in PBX Basic

3. Optimization

None

4. Bug Fixes

None

5. New Features Descriptions

- (1) Call Groups Settings in PBX Basic

Path: PBX Basic->Call Groups Settings

Description:

Add "Call Groups Settings" function. Multiple extensions can be added quickly in the "Queues" and "Ring Groups" function, and can also be used for the "Pick Up" function.

Test Progress:

<1> Call Groups and Extensions

Figure->Add Call Group

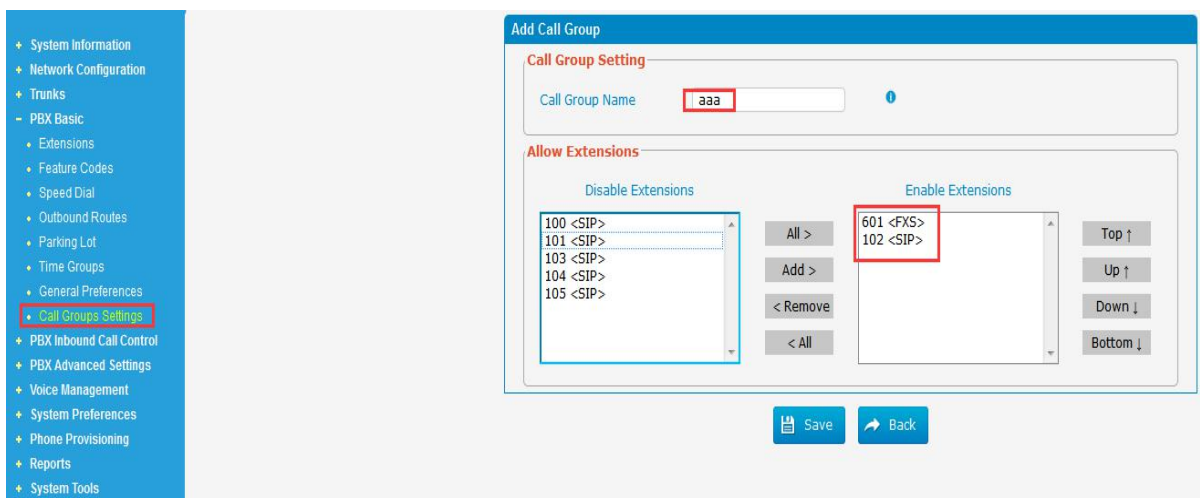


Figure->After Added for 601

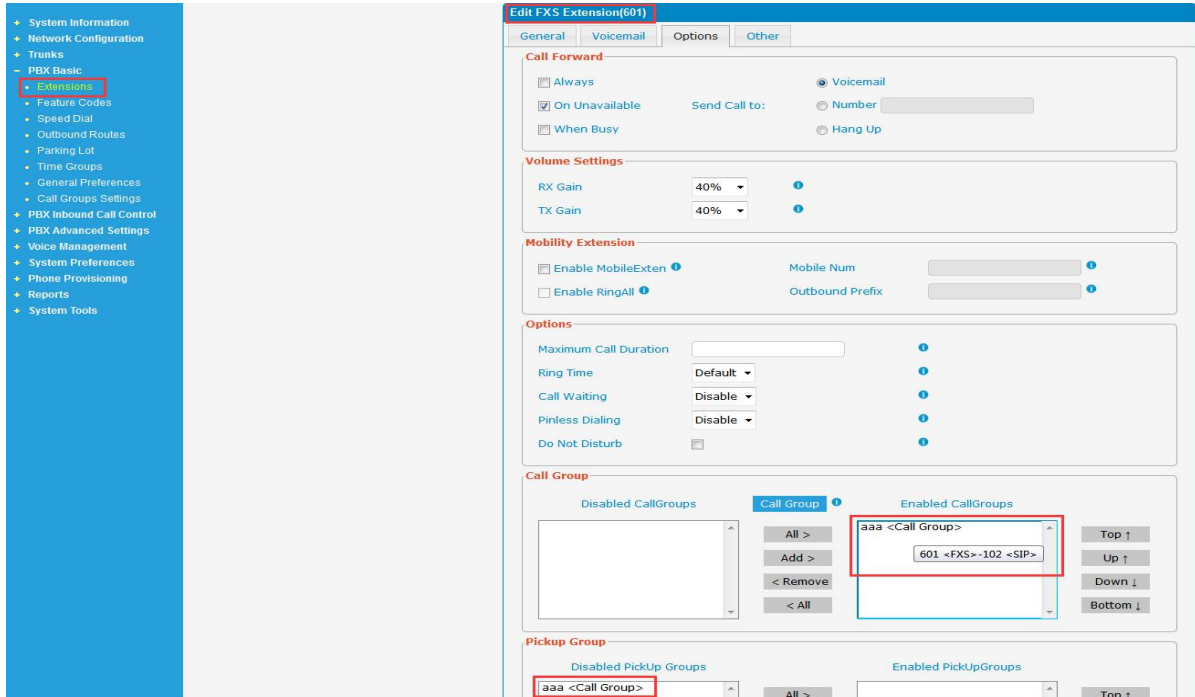
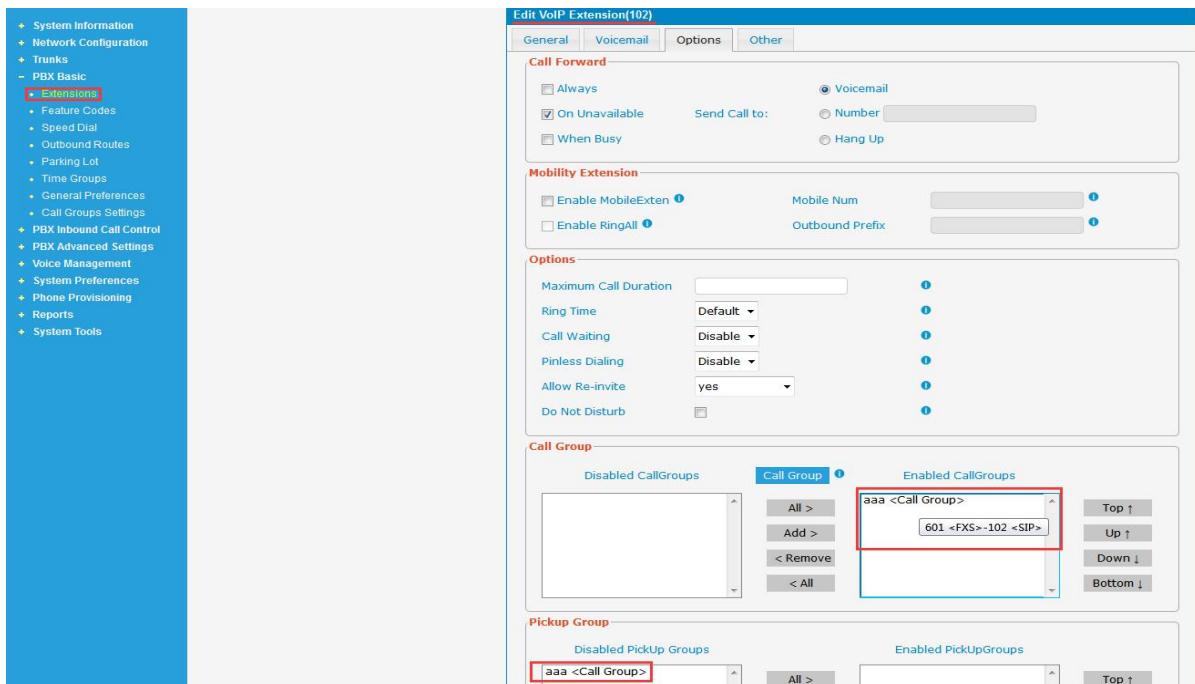


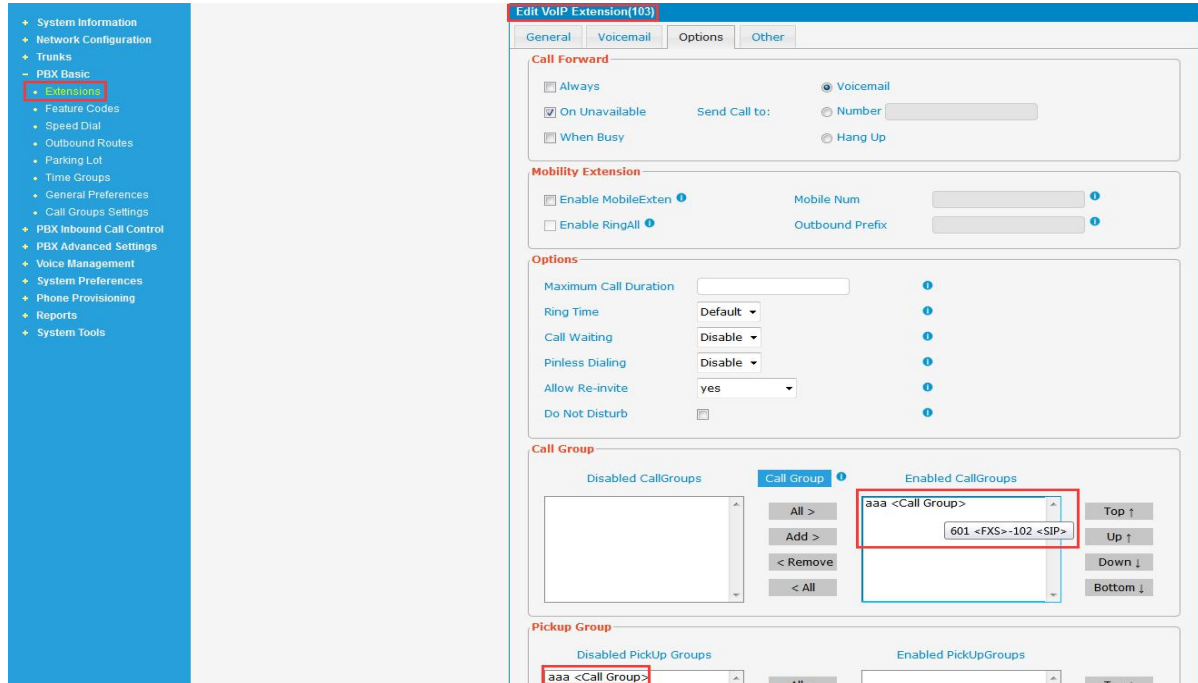
Figure->After Added for 102



Description:

Add the Call Group (aaa) as shown above. "aaa" should be added to "Enabled Call Groups" in the respective extensions of 601 and 102. At the same time, when the mouse hovers over "aaa", you can see the members of "aaa".

Figure->Before Add Call Group in Extension



Edit VoIP Extension(103)

General Voicemail Options Other

Call Forward

☐ Always ☒ Voicemail

☒ On Unavailable Send Call to: ☐ Number ☐ Hang Up

☐ When Busy

Mobility Extension

☐ Enable MobileExten Mobile Num

☐ Enable RingAll Outbound Prefix

Options

Maximum Call Duration

Ring Time Default

Call Waiting Disable

Pinless Dialing Disable

Allow Re-invite yes

Do Not Disturb ☐

Call Group

Disabled CallGroups Call Group Enabled CallGroups

aaa <Call Group> 601 <FXS>-102 <SIP>

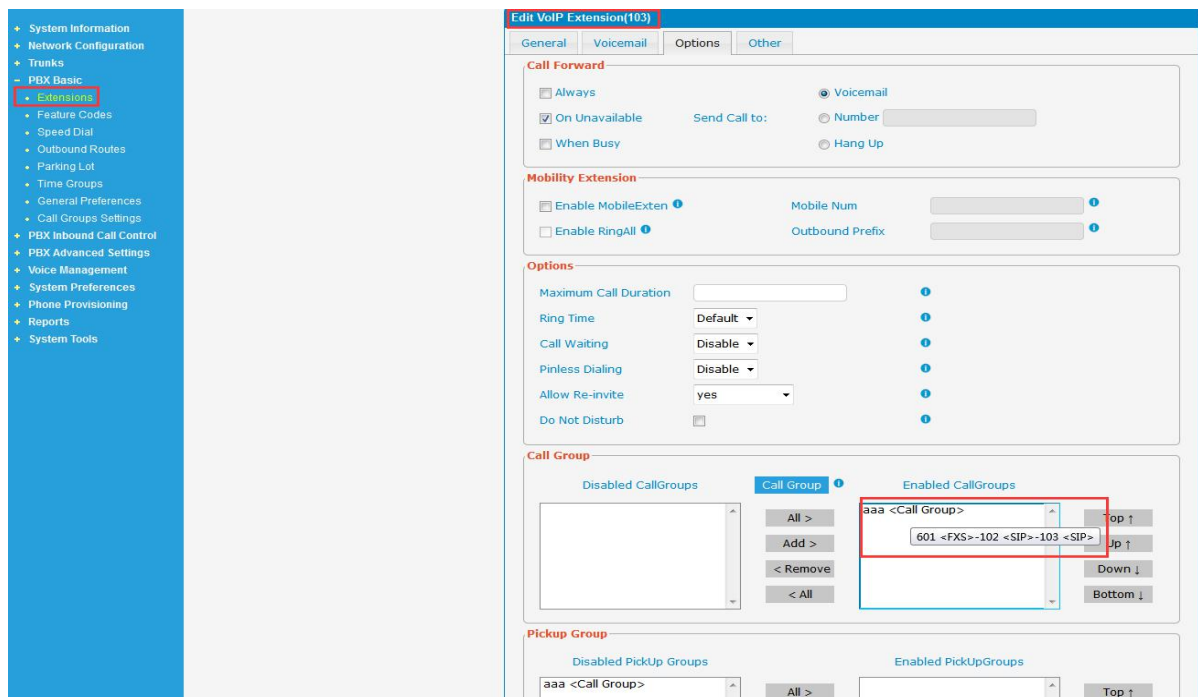
Top ↑ Up ↑ Down ↓ Bottom ↓

Pickup Group

Disabled Pickup Groups Enabled PickupGroups

aaa <Call Group>

Figure->After Add Call Group in Extension



Edit VoIP Extension(103)

General Voicemail Options Other

Call Forward

☐ Always ☒ Voicemail

☒ On Unavailable Send Call to: ☐ Number ☐ Hang Up

☐ When Busy

Mobility Extension

☐ Enable MobileExten Mobile Num

☐ Enable RingAll Outbound Prefix

Options

Maximum Call Duration

Ring Time Default

Call Waiting Disable

Pinless Dialing Disable

Allow Re-invite yes

Do Not Disturb ☐

Call Group

Disabled CallGroups Call Group Enabled CallGroups

aaa <Call Group> 601 <FXS>-102 <SIP>-103 <SIP>

Top ↑ Up ↑ Down ↓ Bottom ↓

Pickup Group

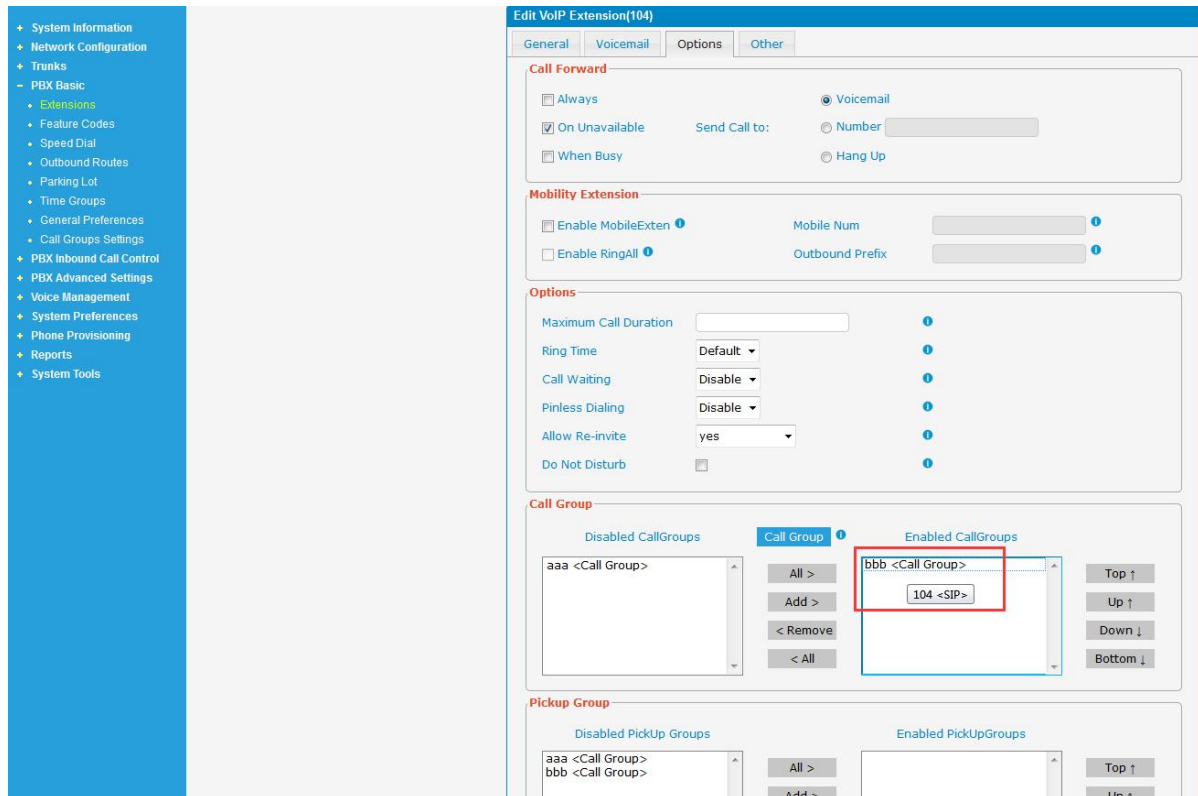
Disabled Pickup Groups Enabled PickupGroups

aaa <Call Group>

Description:

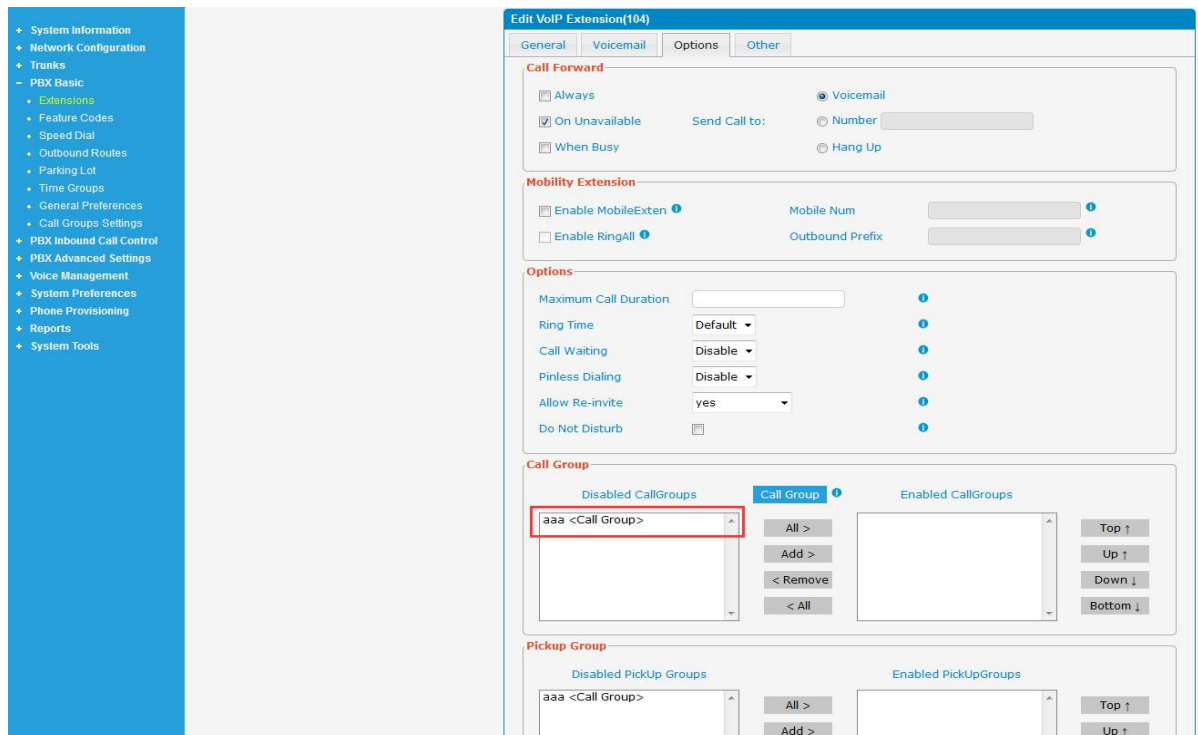
As shown in the figure above, after adding aaa to Enabled Call groups, the extension of 103 can see that the members of aaa have changed when hovering over aaa.

Figure->Before Remove members from Enabled Call Groups



The screenshot shows the 'Edit VoIP Extension(104)' interface. The 'Call Group' section is active, showing two lists: 'Disabled CallGroups' and 'Enabled CallGroups'. The 'Enabled CallGroups' list contains 'bbb <Call Group>' and '104 <SIP>'. The 'Disabled CallGroups' list contains 'aaa <Call Group>'. The 'Call Group' button is highlighted. The 'Options' section shows various settings like 'Maximum Call Duration', 'Ring Time', 'Call Waiting', 'Pinless Dialing', 'Allow Re-invite', and 'Do Not Disturb'.

Figure->After Remove members from Enabled Call Groups

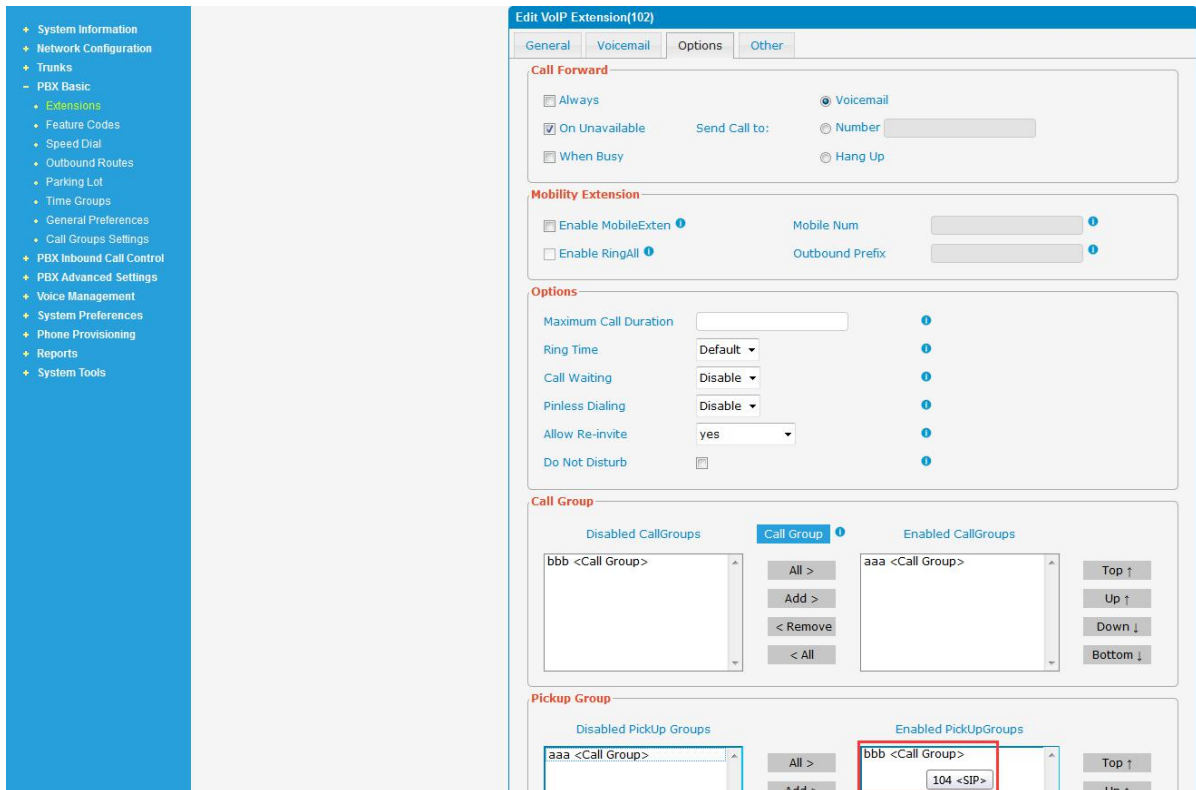


The screenshot shows the 'Edit VoIP Extension(104)' interface after the removal of members from the 'Enabled CallGroups' list. The 'Enabled CallGroups' list is now empty. The 'Disabled CallGroups' list still contains 'aaa <Call Group>'. The 'Call Group' button is highlighted. The 'Options' section remains the same as in the previous figure.

Description:

If a Call Group has only one member, when the extension interface removes the call group, the Call Groups are deleted directly. For example, the member of bbb in the above figure has only one 104 extension.

Figure->Set Pick Up Groups

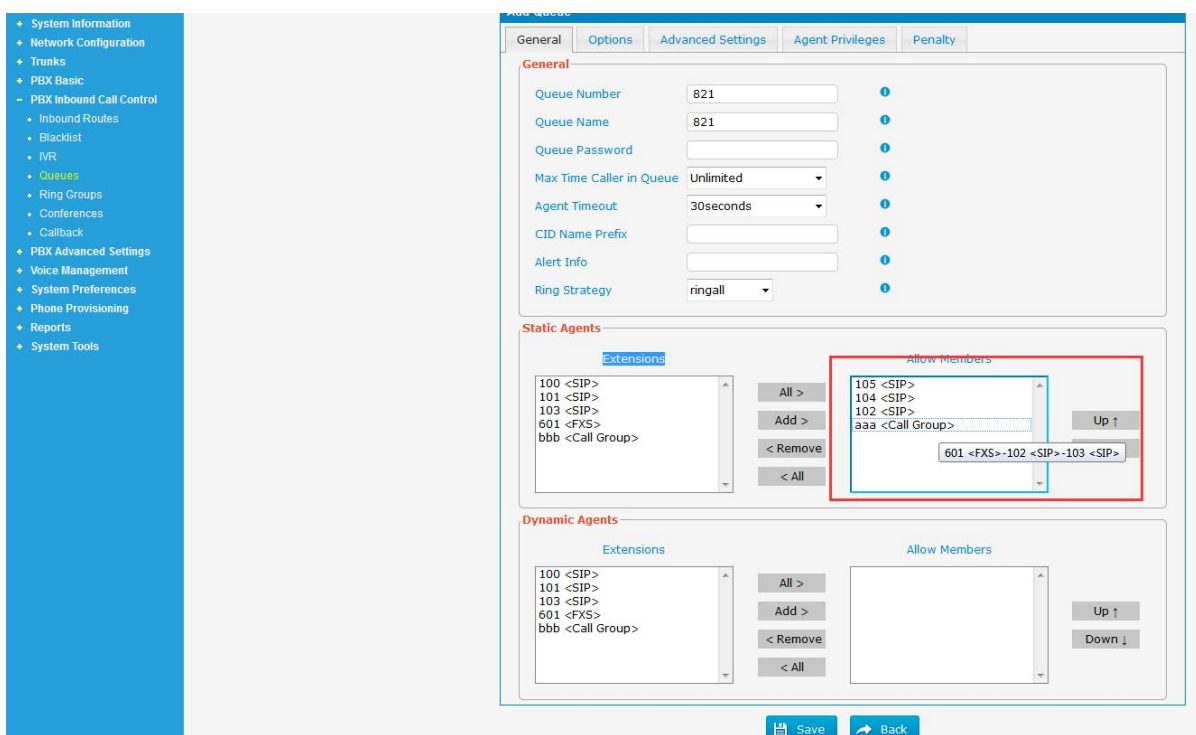


Description:

As shown in the figure above, Enabled Pick Up Groups of extension 102 selects call group(bbb) with extension 104. When extension 101 dials extension 104 and extension 104 rings, extension 102 can intercept by press *8. If the extension 104 dials the extension 101 and the extension 101 rings, the extension 102 cannot intercept the extension 104. In short, the intercept function only intercepts the called party (when using *8).

<2> Queue and Call Groups

Figure->Set Call Groups

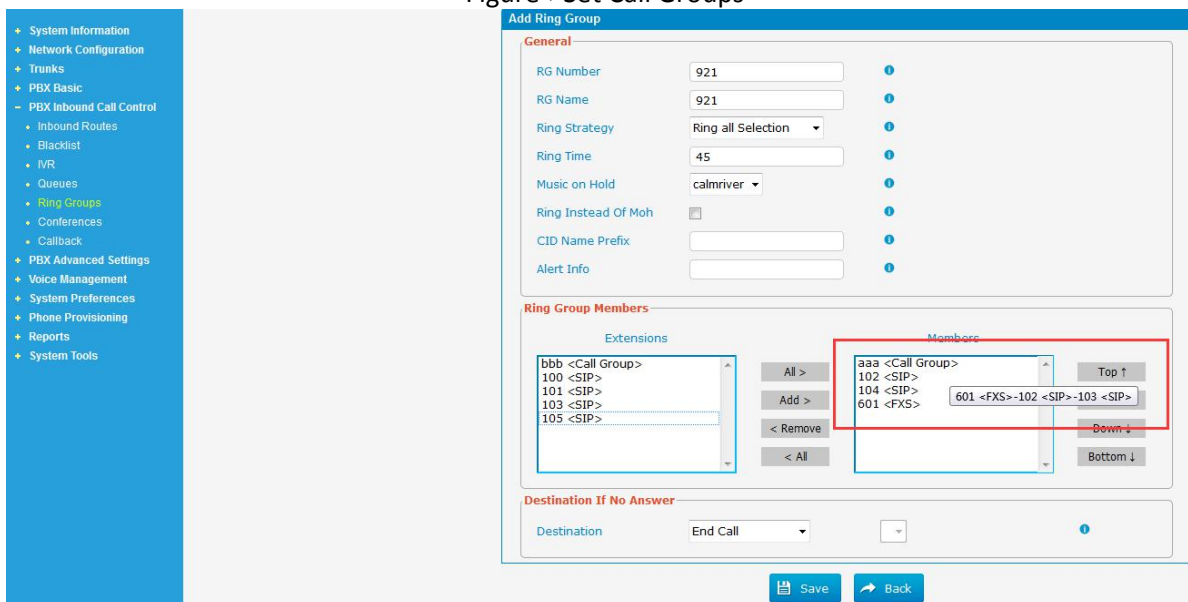


Description:

As shown in the figure above, the actual members of queue 820 are 105, 104, 103, 601, and 102. The members in call group(aaa) have been compared with other extensions (105, 104, and 102), and duplicate extensions are removed.

<3> Ring Groups and Call Groups

Figure->Set Call Groups



Description:

As shown in the figure above, the actual members of ringgroups920 are 102 104, 601 and 103.

The members in call group(aaa) have been compared with other extensions (102, 104 and 601)and duplicate extensions are removed.

6. Bug Fixes Description

none

✧ Release Notes of Version 20/1/12/13.1.0.29-beta01

1. Introduction

- (1) Firmware Version:
20.1.0.29-beta01, 1.1.0.29-beta01, 12.1.0.29-beta01, 13.1.0.29-beta01
- (2) Applicable Model: MUC1002, MUC1004, MUC2008, MUC2016
- (3) Release Date: April 15, 2019

CHANGES SINCE FIRMWARE RELEASE 20/1/12/13.1.0.29

2. New Features

- (1) Concurrent Registration of Extensions
- (1) Encrypt the parameters displayed by the url path.

3. Optimization

None

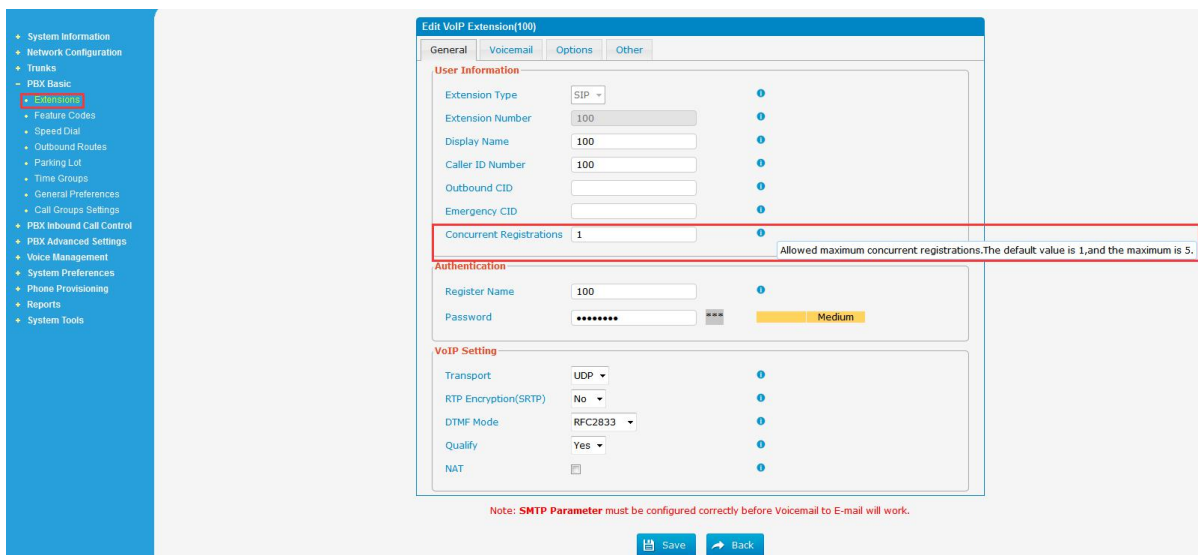
4. Bug Fixes

- (1) The user registered with User permission in the monitor account cannot log in
- (2) The Fanvil phone failed to manually add a MAC address.

5. New Features Descriptions

- (1) Concurrent Registration of Extensions

Figure->Call Groups



Path: PBX Basic->Extensions

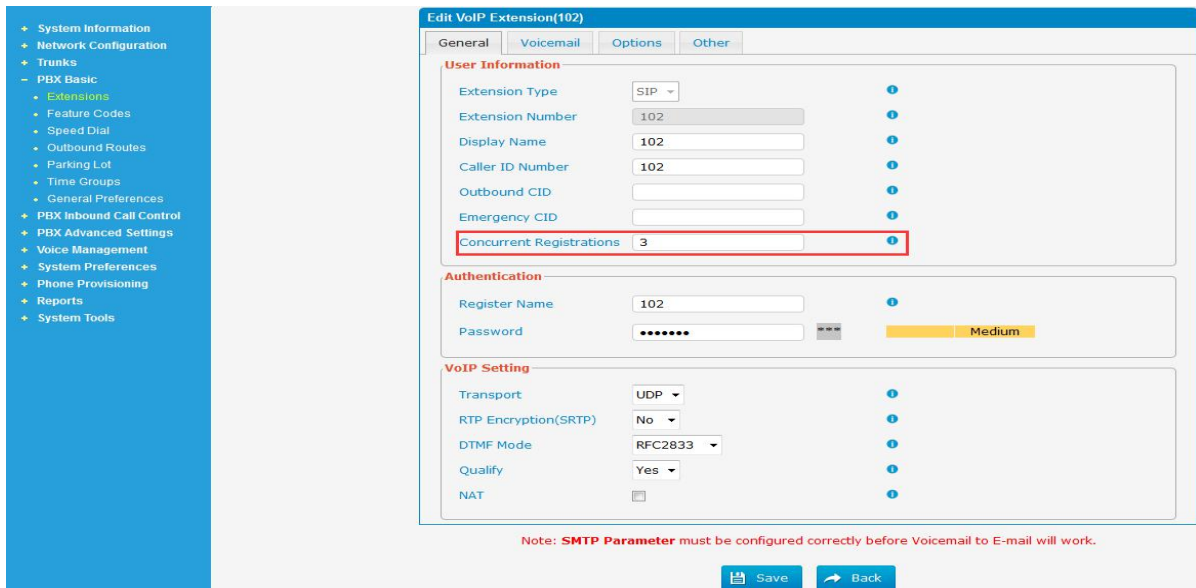
Description:

Add the "Concurrent Registrations" function, the same extension number can be registered to multiple phones, at least one and up to five. When other extensions dial in, all phones registered with this extension number will ring at the same time. Compatible with "Queues" and "Ring Groups" function.

Test Progress:

<1>Configure the "Concurrent Registrations" option in the extension page, if set to 3.

Figure->Set Concurrent Registrations of 102



Edit VoIP Extension(102)

General | Voicemail | Options | Other

User Information

Extension Type: SIP

Extension Number: 102

Display Name: 102

Caller ID Number: 102

Outbound CID:

Emergency CID:

Concurrent Registrations: 3

Authentication

Register Name: 102

Password: *****

VoIP Setting

Transport: UDP

RTP Encryption(SRTP): No

DTMF Mode: RFC2833

Qualify: Yes

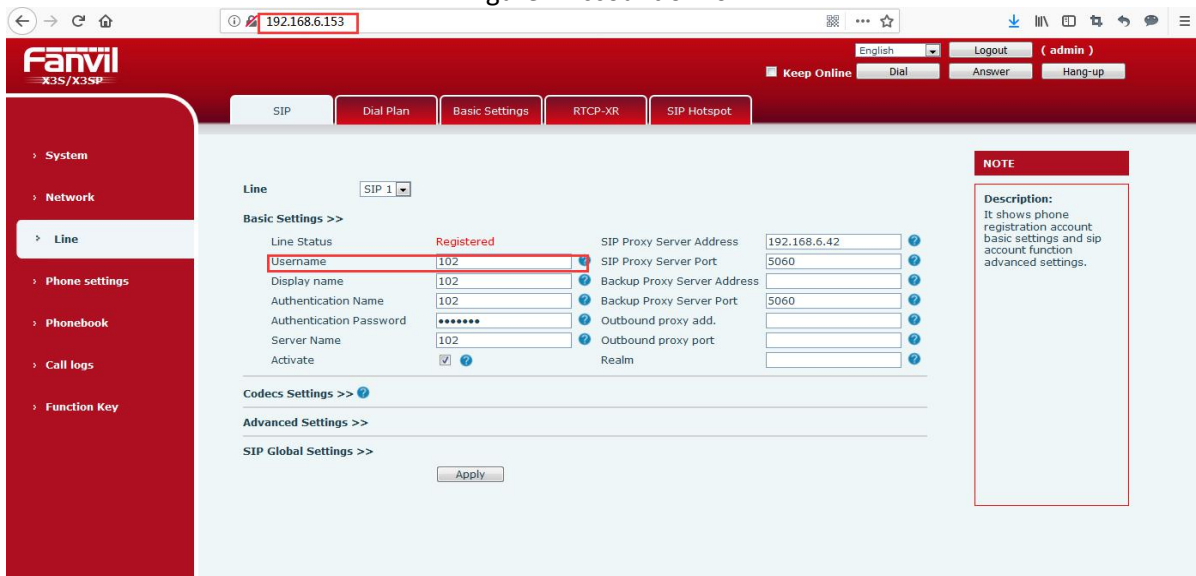
NAT: ☐

Note: SMTP Parameter must be configured correctly before Voicemail to E-mail will work.

Save Back

<2>The extension 102 is separately registered on the three telephones. The “Username” or “SIP UserID” options on the phone should be filled in the order of 102, 102sip1 and 102sip2. By analogy, if the number of concurrent registrations is 5, the registered accounts are 102, 102sip1, 102sip2, 102sip3 and 102sip4. The passwords are the same.

Figure->Account of 102



Fanvil X3S/X3SP

English | Logout | (admin)

Keep Online | Dial | Answer | Hang-up

SIP | Dial Plan | Basic Settings | RTPCP-XR | SIP Hotspot

Line: SIP 1

Basic Settings >>

Line Status: Registered

Username: 102

Display name: 102

Authentication Name: 102

Authentication Password: *****

Server Name: 102

Activate: ☒

SIP Proxy Server Address: 192.168.6.42

SIP Proxy Server Port: 5060

Backup Proxy Server Address:

Backup Proxy Server Port: 5060

Outbound proxy add.:

Outbound proxy port:

Realm:

Codecs Settings >>

Advanced Settings >>

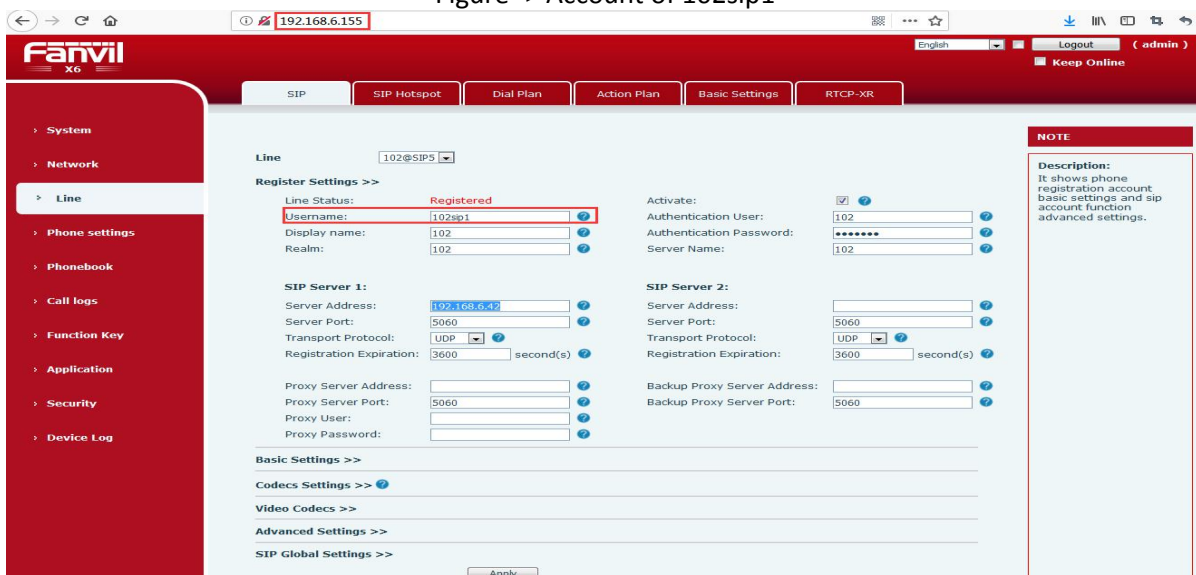
SIP Global Settings >>

Apply

NOTE

Description: It shows phone registration account basic settings and sip account function advanced settings.

Figure -> Account of 102sip1



Fanvil X6

English | Logout | (admin)

Keep Online

SIP | SIP Hotspot | Dial Plan | Action Plan | Basic Settings | RTPCP-XR

Line: 102@SIP5

Register Settings >>

Line Status: Registered

Username: 102sip1

Display name: 102

Realm: 102

Activate: ☒

Authentication User: 102

Authentication Password: *****

Server Name: 102

SIP Server 1:

Server Address: 192.168.6.42

Server Port: 5060

Transport Protocol: UDP

Registration Expiration: 3600 second(s)

Proxy Server Address:

Proxy Server Port: 5060

Proxy User:

Proxy Password:

SIP Server 2:

Server Address:

Server Port: 5060

Transport Protocol: UDP

Registration Expiration: 3600 second(s)

Backup Proxy Server Address:

Backup Proxy Server Port: 5060

Basic Settings >>

Codecs Settings >>

Video Codecs >>

Advanced Settings >>

SIP Global Settings >>

Apply

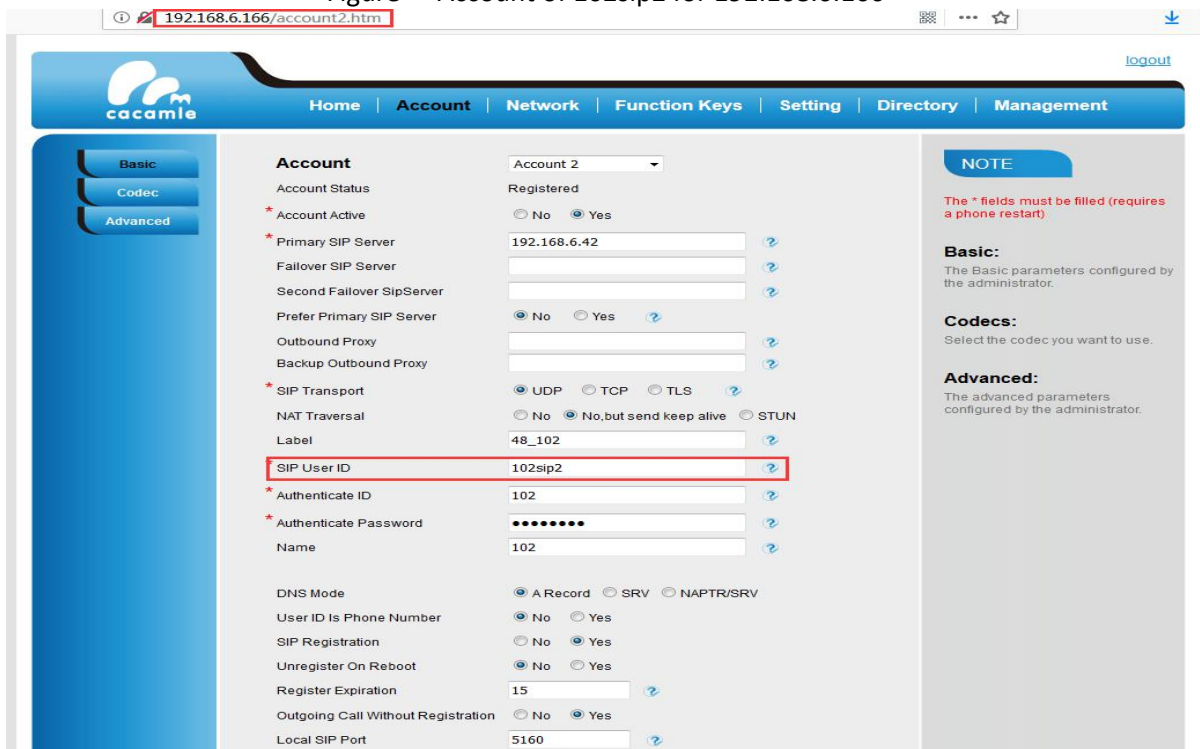
NOTE

Description: It shows phone registration account basic settings and sip account function advanced settings.

Notes: When the same extension number is registered on a different extension, the newly

registered phone ip address will replace the previous ip address.

Figure -> Account of 102sip2 for 192.168.6.166



192.168.6.166/account2.htm

Home | Account | Network | Function Keys | Setting | Directory | Management

Account Account 2

Account Status: Registered

* Account Active: ☐ No ☒ Yes

* Primary SIP Server: 192.168.6.42

Failover SIP Server:

Second Failover SIP Server:

Prefer Primary SIP Server: ☒ No ☐ Yes

Outbound Proxy:

Backup Outbound Proxy:

* SIP Transport: ☒ UDP ☐ TCP ☐ TLS

NAT Traversal: ☐ No ☒ No, but send keep alive ☐ STUN

Label: 48_102

SIP User ID: 102sip2

* Authenticate ID: 102

* Authenticate Password:

Name: 102

DNS Mode: ☒ A Record ☐ SRV ☐ NAPTR/SRV

User ID Is Phone Number: ☒ No ☐ Yes

SIP Registration: ☐ No ☒ Yes

Unregister On Reboot: ☒ No ☐ Yes

Register Expiration: 15

Outgoing Call Without Registration: ☐ No ☒ Yes

Local SIP Port: 5160

NOTE

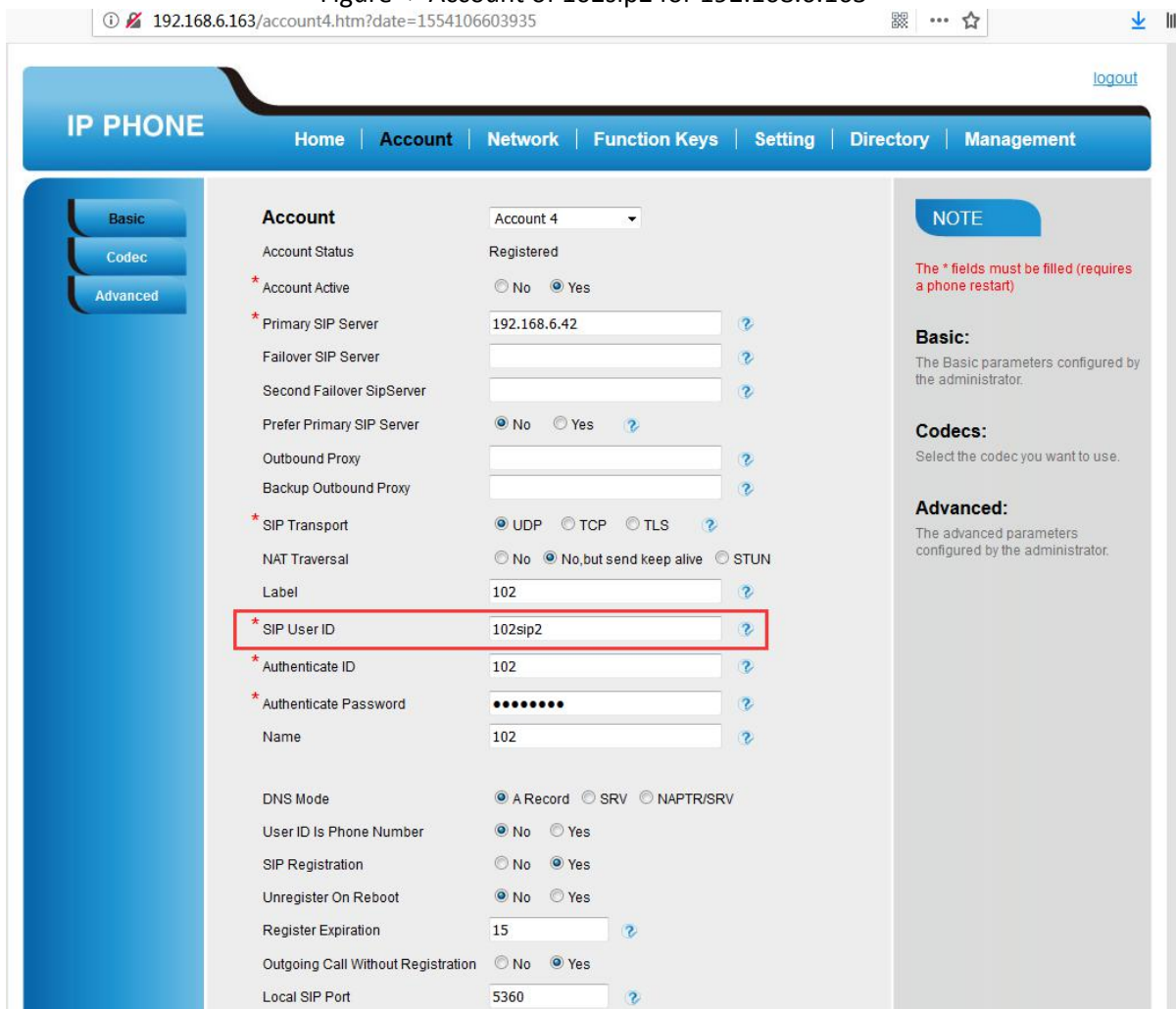
The * fields must be filled (requires a phone restart)

Basic:
The Basic parameters configured by the administrator.

Codecs:
Select the codec you want to use.

Advanced:
The advanced parameters configured by the administrator.

Figure -> Account of 102sip2 for 192.168.6.163



192.168.6.163/account4.htm?date=1554106603935

Home | Account | Network | Function Keys | Setting | Directory | Management

IP PHONE

Account Account 4

Account Status: Registered

* Account Active: ☐ No ☒ Yes

* Primary SIP Server: 192.168.6.42

Failover SIP Server:

Second Failover SIP Server:

Prefer Primary SIP Server: ☒ No ☐ Yes

Outbound Proxy:

Backup Outbound Proxy:

* SIP Transport: ☒ UDP ☐ TCP ☐ TLS

NAT Traversal: ☐ No ☒ No, but send keep alive ☐ STUN

Label: 102

SIP User ID: 102sip2

* Authenticate ID: 102

* Authenticate Password:

Name: 102

DNS Mode: ☒ A Record ☐ SRV ☐ NAPTR/SRV

User ID Is Phone Number: ☒ No ☐ Yes

SIP Registration: ☐ No ☒ Yes

Unregister On Reboot: ☒ No ☐ Yes

Register Expiration: 15

Outgoing Call Without Registration: ☐ No ☒ Yes

Local SIP Port: 5360

NOTE

The * fields must be filled (requires a phone restart)

Basic:
The Basic parameters configured by the administrator.

Codecs:
Select the codec you want to use.

Advanced:
The advanced parameters configured by the administrator.

The displayed of extension status is change from:

Free Busy Hold Unavailable Ringing Extension Status Show Filter				
Extension	Extension	Extension	Extension	Extension
100(SIP)	101(SIP 192.168.6.164)	102(SIP 192.168.6.153 / 192.168.6.155 / 192.168.6.166)	103(SIP)	104(SIP)
105(SIP)	106(SIP)	--	--	--

to:

Free Busy Hold Unavailable Ringing Extension Status Show Filter				
Extension	Extension	Extension	Extension	Extension
100(SIP)	101(SIP 192.168.6.164)	102(SIP 192.168.6.153 / 192.168.6.155 / 192.168.6.166)	103(SIP)	104(SIP)
105(SIP)	106(SIP)	--	--	--

<3> Registration completion diagram and dialing process.

Figure->Register 101 extension and 102 extension for dial test.

Free Busy Hold Unavailable Ringing Extension Status Show Filter				
Extension	Extension	Extension	Extension	Extension
100(SIP)	101(SIP 192.168.6.164)	102(SIP 192.168.6.153 / 192.168.6.155 / 192.168.6.166)	103(SIP)	104(SIP)
105(SIP)	106(SIP)	--	--	--

101 extension dials 102 extension

Figure->Ringing

Free Busy Hold Unavailable Ringing Extension Status Show Filter				
Extension	Extension	Extension	Extension	Extension
100(SIP)	101(SIP 192.168.6.164)	102(SIP 192.168.6.153 / 192.168.6.155 / 192.168.6.166)	103(SIP)	104(SIP)
105(SIP)	106(SIP)	--	--	--

Figure->Answered

Free Busy Hold Unavailable Ringing Extension Status Show Filter				
Extension	Extension	Extension	Extension	Extension
100(SIP)	101(SIP 192.168.6.164)	102(SIP 192.168.6.153 / 192.168.6.155 / 192.168.6.166)	103(SIP)	104(SIP)
105(SIP)	106(SIP)	--	--	--

<4> Queues and Ring Groups Testing

Adding extensions 102 to the Queues(820) and Ring Groups(920), and then dialing 820 and 920 through extension 101, it can be seen that the plurality of telephones registered by the extension 102 are ringing at the same time.

<5> CDR Display Situation

When the extension 101 dials 820 and 920, if any one of the plurality of telephones registered at the extension 102 is answered, the CDR Report will display the account name of the telephone.

- System Information
- Network Configuration
- Trunks
- PBX Basic
- PBX Inbound Call Control
- PBX Advanced Settings
- Voice Management
- System Preferences
- Phone Provisioning
- Reports
 - CDR Report
 - Voicemail List
 - System Logs
 - Firewall Logs
 - Trace Log
- System Tools

Date	Source	Destination	Src. Trunk	Account Code	Dst. Trunk	Call Direction	Status	Duration	Billing Duration
2019-04-01 16:29:50	101	920(102sip2)				Internal	ANSWERED	11s	4s
2019-04-01 16:29:23	101	820(102sip1)				Internal	ANSWERED	9s	9s
2019-04-01 16:26:15	101	820(102)				Internal	ANSWERED	12s	12s

(2) Encrypt the parameters displayed by the url path to avoid being directly modified by others.

Figure->Add User in Admin

- System Information
- Network Configuration
- Trunks
- PBX Basic
- PBX Inbound Call Control
- PBX Advanced Settings
- Voice Management
- System Preferences
- Firewall Rules
- Security Info
- Firmware Update
- Data Backup
- Data Restore
- Password
- User Permission
- Time & Date
- Reset
- Reboot
- Phone Provisioning
- Reports
- System Tools

Add User

User Name: test

Enable User: Yes

Enter New Password: 123456

Retype New Password: *****

User Permissions:

- System Information ☒
- Extension Status ☐
- Trunk Status ☐
- Outbound Routes ☐
- Inbound Routes ☐
- IVR ☐
- Reminder Call Settings ☐
- System Recordings ☐
- CDR Report ☒
- Firewall Logs ☐
- Trace Log ☐

Save Back

Change Password

Logout(admin)

Figure->Add User in Monitor

- General Preferences
- Advanced Settings
- USB Sharing Settings
- Windows SMB
- View Records Management
- User Permission
- Password

Add User

User Name: monitor1

Enable User: Yes

Enter New Password: 123456

Retype New Password: *****

User Permissions:

- USB Devices ☐
- Recording Settings ☒
- Call Detail Records ☒

Save Back

Change Password

Logout(monitor)

Figure->Before Encrypt

192.168.6.95/index-user.html?FMDate=1533721701&UserPermission=on,off,off,off,off,off,off,off,off,off

MAXINCOM Web Management System

System Information

LAN Status

MAC Address: 58:e8:76:d5:00:16

Hostname: 100U8TR

Network: 192.168.6.95 255.255.255.0 192.168.6.1

DNS Server: 192.168.6.1

Traffic Statistics: RX bytes: 873419920 (832.9 MiB) TX bytes: 397298915 (378.8 MiB)

System Time

Current Server Time: May 6, 2019 16:46:09

System Up Time: 13 days 6 hours 37 minutes 0 seconds

System Resources

Disk Usage(1K-blocks): Used: 18160 Total: 389120 Use%: 5%

Memory Usage(1K-blocks): Used: 130248 Total: 222896 Use%: 58%

Version Information

Product: MUC2008

Hardware Version: V1.0.00

Firmware Version: 12.1.0.27

Change Password

Logout(user01)

Figure->After Encrypt

192.168.6.42/index-user.html?FMDate=1557189494&UserPermission=bd38ec61cd9c7b74c3990d4ac857c38461d9eba6b27a8f2c02054e54defbc3

MAXINCOM Web Management System

System Information

LAN Status

MAC Address: 50:8c-b1-e7:c0:09

Hostname: MUC1004

Network: 192.168.6.42 255.255.255.0 192.168.6.1

DNS Server: 192.168.6.1

Traffic Statistics: RX bytes: 423473 (413.5 KiB) TX bytes: 655141 (639.7 KiB)

System Time

Current Server Time: May 6, 2019 00:48:02

System Up Time: 6 minutes 49 seconds

System Resources

Disk Usage(1K-blocks): Used: 8800 Total: 425984 Use%: 2%

Memory Usage(1K-blocks): Used: 109184 Total: 254376 Use%: 42%

Version Information

Product: MUC1004

Hardware Version: V1.1.00

Firmware Version: 1.1.01.29-beta02

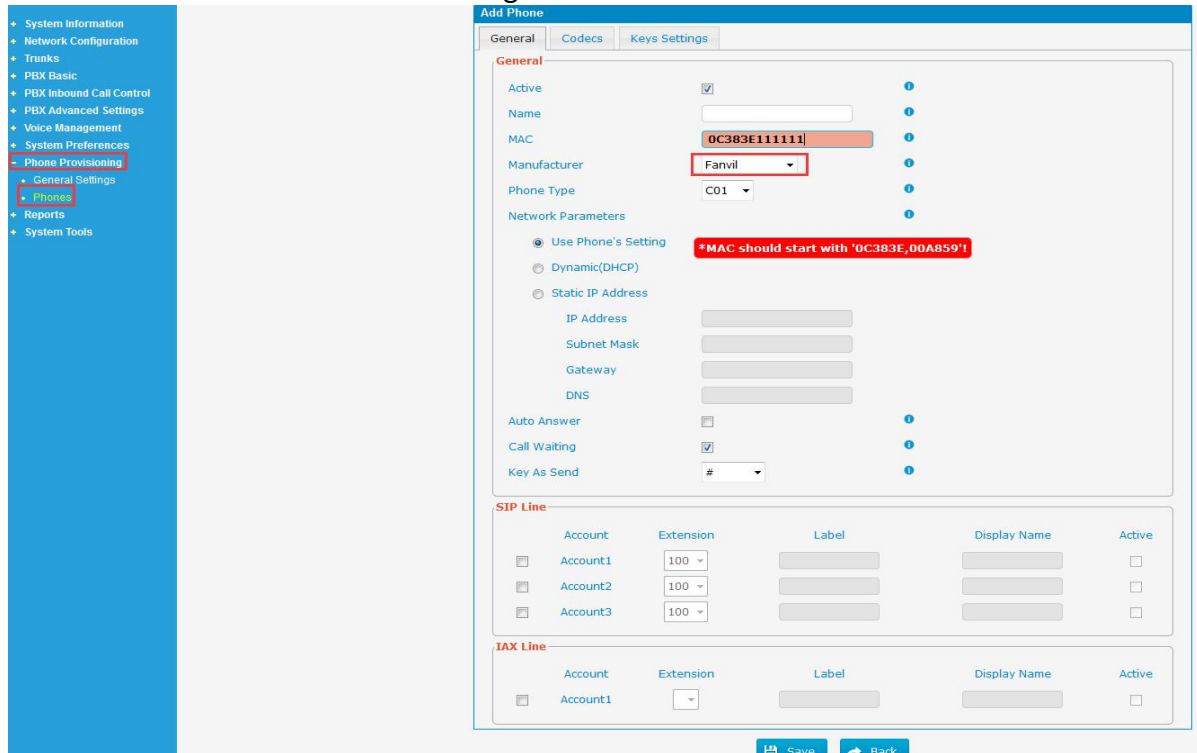
Change Password

Logout(test)

6. Bug Fixes Description

- (1) The user registered with User permission in the monitor account cannot log in
- (2) Fixed the problem that the Fanvil phone failed to manually add a MAC address.

Figure->Before Fixed



Add Phone

General | Codecs | Keys Settings

General

Active ☒ ⓘ

Name ⓘ

MAC ⓘ

Manufacturer ⓘ

Phone Type ⓘ

Network Parameters ⓘ

☒ Use Phone's Setting ***MAC should start with '0C383E,00A859!'**

☐ Dynamic(DHCP)

☐ Static IP Address

IP Address

Subnet Mask

Gateway

DNS

Auto Answer ☐ ⓘ

Call Waiting ☒ ⓘ

Key As Send ⓘ

SIP Line

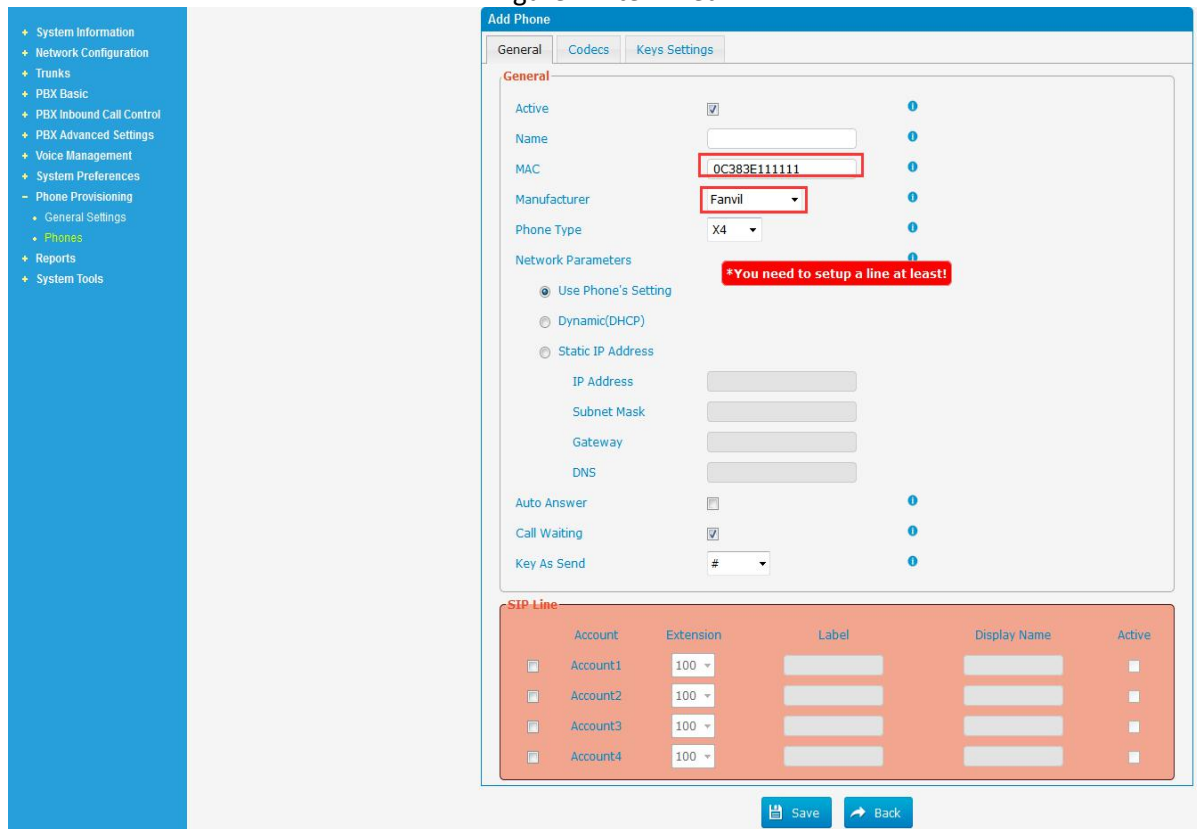
Account	Extension	Label	Display Name	Active
☐ Account1	100			☐
☐ Account2	100			☐
☐ Account3	100			☐

IAX Line

Account	Extension	Label	Display Name	Active
☐ Account1				☐

Save Back

Figure->After Fixed



Add Phone

General | Codecs | Keys Settings

General

Active ☒ ⓘ

Name ⓘ

MAC ⓘ

Manufacturer ⓘ

Phone Type ⓘ

Network Parameters ⓘ

☒ Use Phone's Setting ***You need to setup a line at least!**

☐ Dynamic(DHCP)

☐ Static IP Address

IP Address

Subnet Mask

Gateway

DNS

Auto Answer ☐ ⓘ

Call Waiting ☒ ⓘ

Key As Send ⓘ

SIP Line

Account	Extension	Label	Display Name	Active
☐ Account1	100			☐
☐ Account2	100			☐
☐ Account3	100			☐
☐ Account4	100			☐

Save Back

✧ Release Notes of Version 20/1/12/13.1.0.29

1. Introduction

- (1) Firmware Version: 20.1.0.29,1.1.0.29,12.1.0.29,13.1.0.29
- (2) Applicable Model:MUC1002,MUC1004,MUC2008,MUC2016
- (3) Release Date: March 5, 2019

CHANGES SINCE FIRMWARE RELEASE 20/1/12/13.1.0.28-beta03

2. New Features

- (1) Announcement in Paging and Intercom

3. Optimization

None

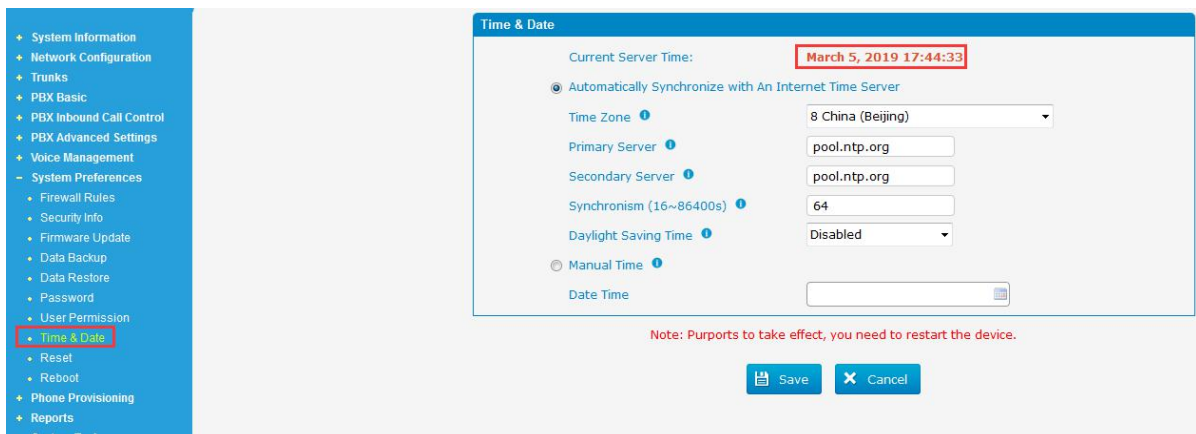
4. Bug Fixes

None

5. New Features Descriptions

- (1) Announcement in Paging and Intercom

Figure->view system date

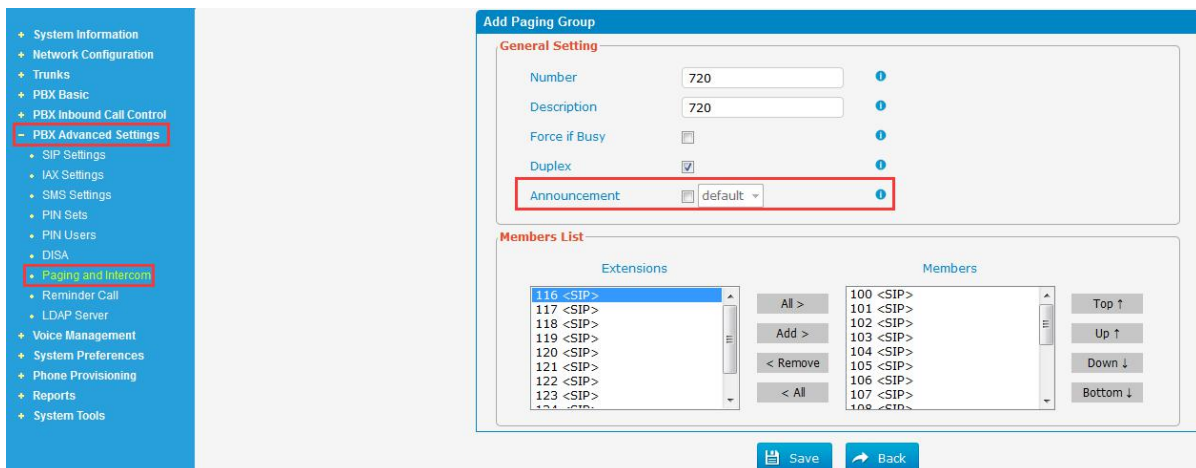


The screenshot shows the 'Time & Date' configuration page. The left sidebar has a menu with 'Time & Date' highlighted. The main content area shows the following settings:

- Current Server Time: March 5, 2019 17:44:33
- Automatically Synchronize with An Internet Time Server: ☒ (selected)
- Time Zone: 8 China (Beijing)
- Primary Server: pool.ntp.org
- Secondary Server: pool.ntp.org
- Synchronism (16~86400s): 64
- Daylight Saving Time: Disabled
- Manual Time: ☐ (not selected)
- Date Time:

At the bottom, there is a note: 'Note: Purports to take effect, you need to restart the device.' and buttons for 'Save' and 'Cancel'.

Figure->Do not enable play announcement function

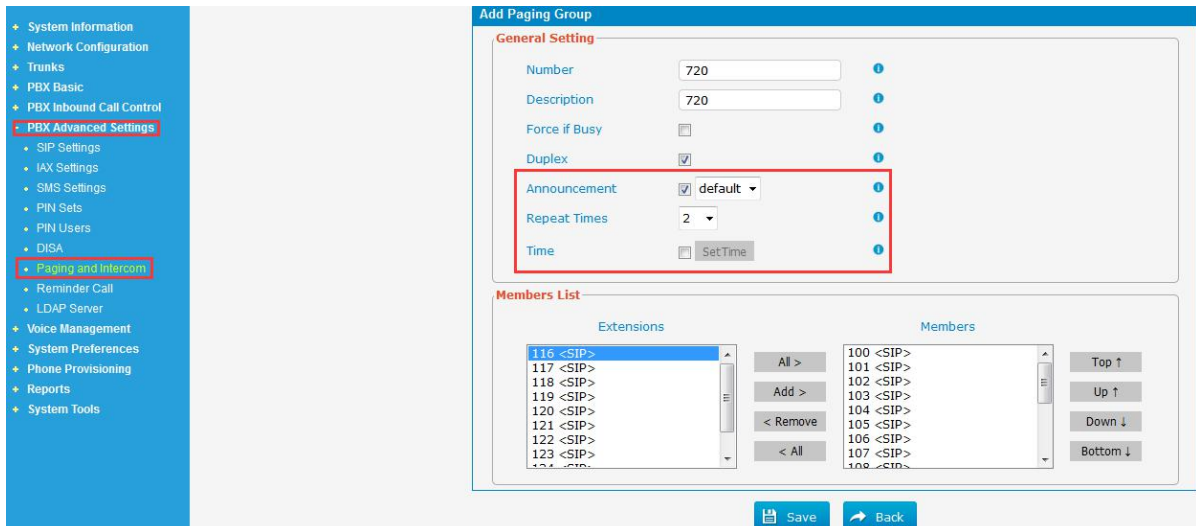


The screenshot shows the 'Add Paging Group' page. The left sidebar has a menu with 'PBX Advanced Settings' and 'Paging and Intercom' highlighted. The main content area shows the following settings:

- Number: 720
- Description: 720
- Force if Busy: ☐
- Duplex: ☒
- Announcement: default

Below the settings is a 'Members List' section with two columns: 'Extensions' and 'Members'. The 'Extensions' column lists SIP addresses (116 <SIP> to 123 <SIP>). The 'Members' column lists SIP addresses (100 <SIP> to 108 <SIP>). There are buttons for 'All >', 'Add >', '< Remove', and '< All' between the columns. At the bottom, there are buttons for 'Save' and 'Back'.

Figure-> enable play announcement function



Add Paging Group

General Setting

Number: 720

Description: 720

Force if Busy: ☐

Duplex: ☒

Announcement: ☒ default

Repeat Times: 2

Time: ☐ SetTime

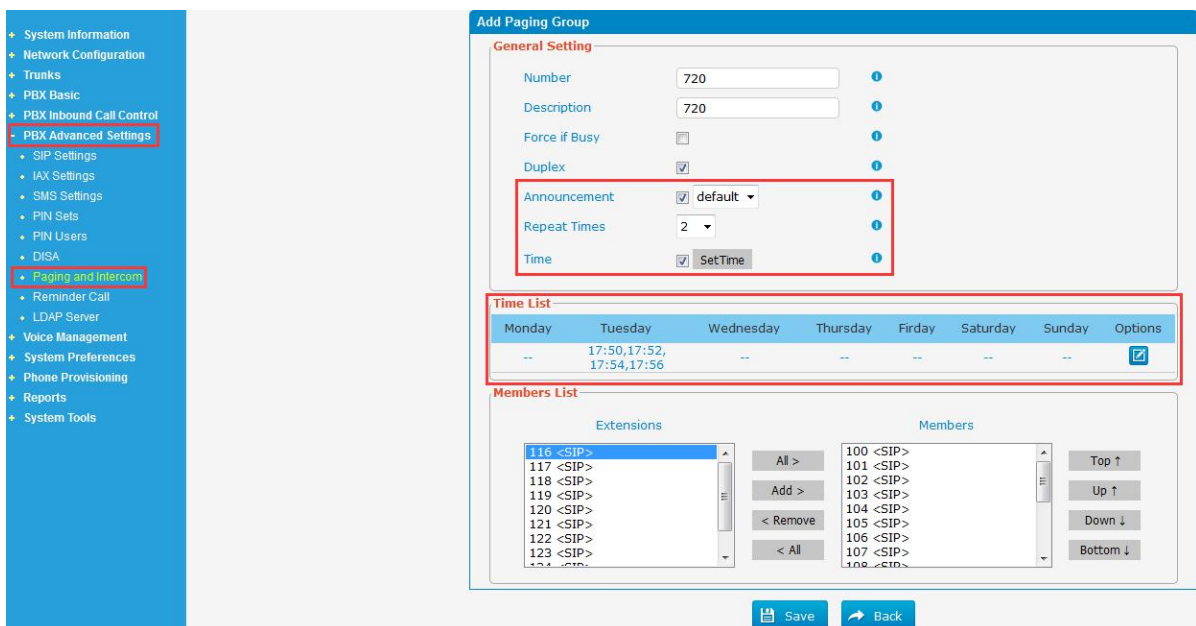
Members List

Extensions: 116 <SIP>, 117 <SIP>, 118 <SIP>, 119 <SIP>, 120 <SIP>, 121 <SIP>, 122 <SIP>, 123 <SIP>

Members: 100 <SIP>, 101 <SIP>, 102 <SIP>, 103 <SIP>, 104 <SIP>, 105 <SIP>, 106 <SIP>, 107 <SIP>, 108 <SIP>

Buttons: Save, Back

Figure->Enable auto-play announcements



Add Paging Group

General Setting

Number: 720

Description: 720

Force if Busy: ☐

Duplex: ☒

Announcement: ☒ default

Repeat Times: 2

Time: ☒ SetTime

Time List

Monday	Tuesday	Wednesday	Thursday	Friday	Saturday	Sunday	Options
--	17:50,17:52, 17:54,17:56	--	--	--	--	--	

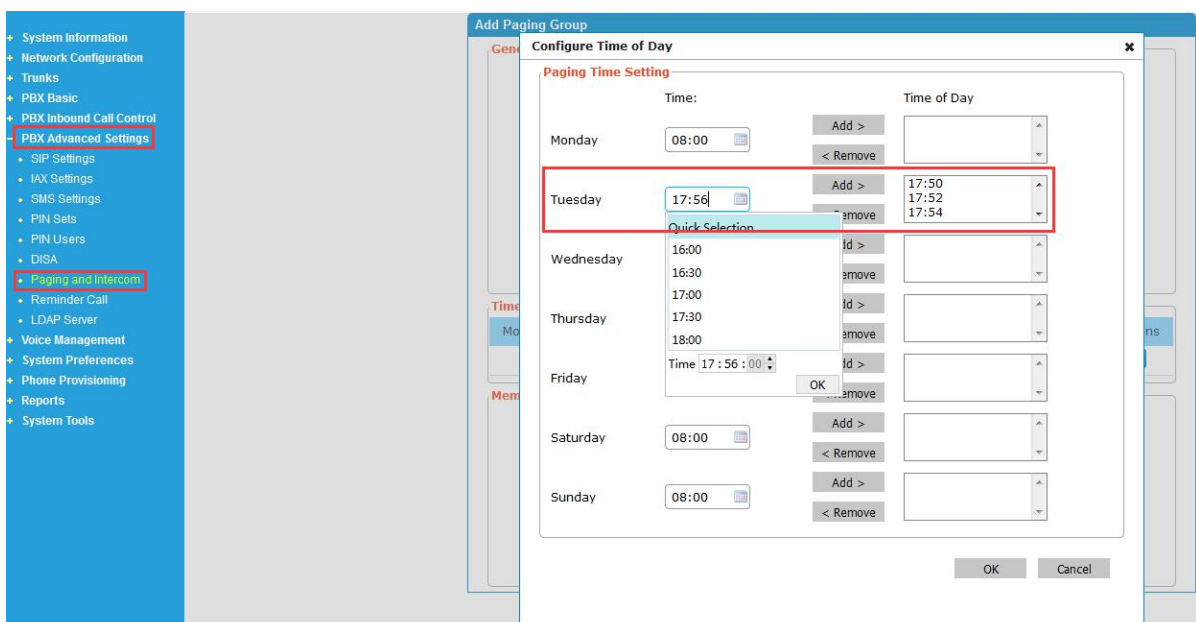
Members List

Extensions: 116 <SIP>, 117 <SIP>, 118 <SIP>, 119 <SIP>, 120 <SIP>, 121 <SIP>, 122 <SIP>, 123 <SIP>

Members: 100 <SIP>, 101 <SIP>, 102 <SIP>, 103 <SIP>, 104 <SIP>, 105 <SIP>, 106 <SIP>, 107 <SIP>, 108 <SIP>

Buttons: Save, Back

Figure->Set the specific time of the auto-play announcement



Add Paging Group

Configure Time of Day

Paging Time Setting

Time	Time of Day
Monday: 08:00	Add > < Remove
Tuesday: 17:56	Add > 17:50 < Remove 17:52 17:54
Wednesday: 16:00, 16:30, 17:00	Add > < Remove
Thursday: 17:30, 18:00	Add > < Remove
Friday: 17:56:00	Add > < Remove
Saturday: 08:00	Add > < Remove
Sunday: 08:00	Add > < Remove

Buttons: OK, Cancel

Path: PBX Advanced Settings->Paging and Intercom

Description:

Add the "Announcement" and "Repeat Times" options and update the "Time" setting. The Paging Groups feature is divided into the Enable Broadcast Announcement feature and is not enabled. Among them, the broadcast announcement function is divided into manual and automatic timing. The manual broadcast function broadcasts an announcement to the called party through the system after the calling party and the called party are connected. After the announcement is over, the calling party and the called party can make a call. If the "Duplex" option is enabled, all phones in the paging group are allowed to talk and be heard by all; if "Duplex" is not enabled, the caller can only speak, while other members can only hear and not participate in the call. The Auto play function is done by setting the time in "Time List". When system time reaches a specified point in time, all selected available extensions automatically answer and play the announcement, and then hang up automatically when the playback is complete.

NOTICE: When the timing broadcast function is enabled, the set time interval cannot be lower than the time length of the announcement. If the announcement time is 2 minutes, the timing is not allowed at 11:30 and 11:31.

Figure->MUC1004 PBX Maximum number of members in manual mode

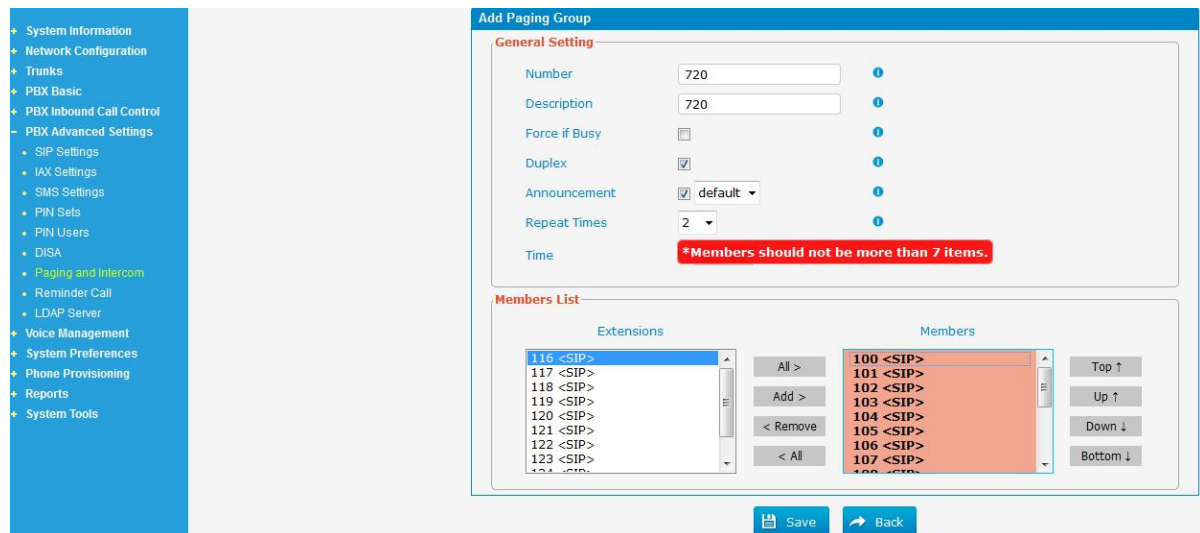


Figure->MUC1004 PBX Maximum number of members in automatic mode

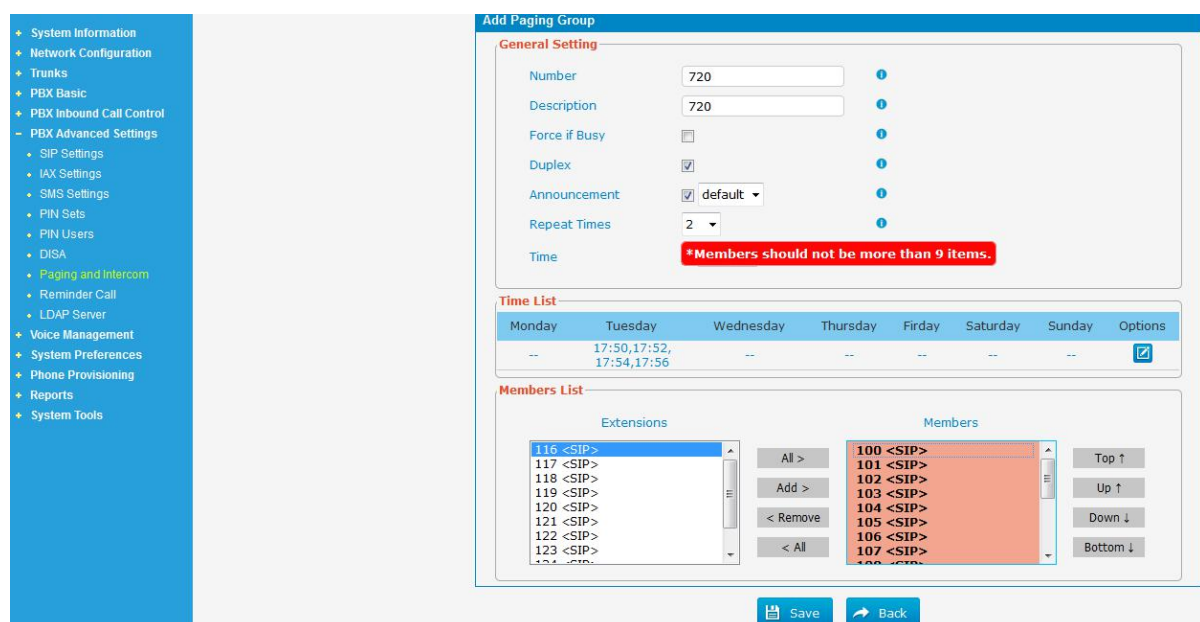
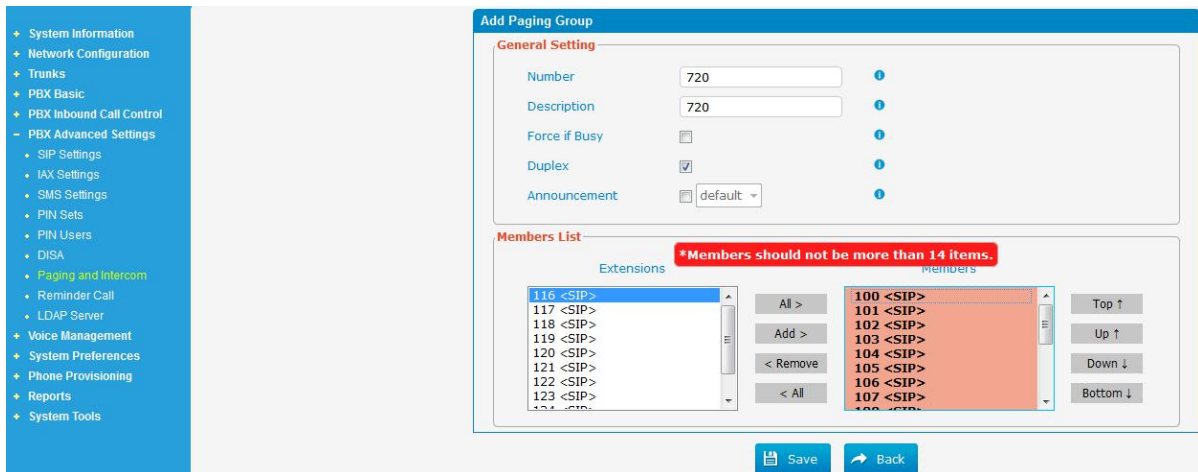


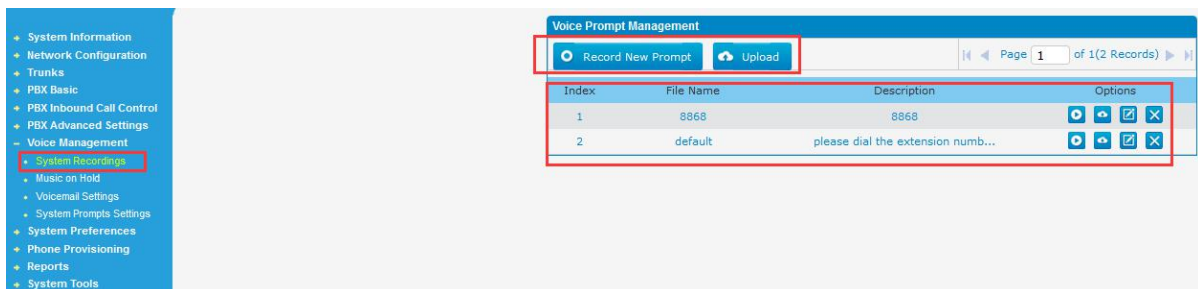
Figure->MUC1004 PBX Maximum number of members for announcement mode is not enabled



Description:

Due to IPPBX performance constraints, the maximum number of "Members" in the Paging Groups of the four types of PBX of MUC1002 / MUC1004 / MUC2008 / MUC2016 is different. As shown in the figure above, the maximum number of members for MUC1004 automatic mode is 9, the maximum number for manual mode is 7, and the maximum number for unenabled announcement mode is 14. At the same time, the maximum number of members of MUC1002 automatic mode is 5, that of manual mode is 4, and the maximum number of members of unenabled announcement mode is 9. The maximum number of members in MUC2008 auto mode is 20, the maximum number of members in manual mode is 15, the maximum number of members unenabled announcement mode is 30. The maximum number of members in Muc2016 auto mode is 30, and the maximum number of members in manual mode is 24. the maximum number for unenabled announcement mode is 49. When a member exceeds the maximum limit, one or more extensions fail to hang up. To avoid this, it is recommended that the number of registered members should not exceed the smaller value in manual and automatic mode, or disable "Set Time" before using manual mode. If there is a failure to hang up, you need to restart PBX.

Figure->Announcement content recording and uploading



Path: Voice Management->System Recordings

Description: Announcements can be recorded via the Record new Prompt button or uploaded via the Upload button. The announcement file format must be a gsm file.

6.Bug Fixes Description

none

✧ Release Notes of Version 20/1/12/13.1.0.28-beta03

1. Introduction

- (1) Firmware Version:
20.1.0.28-beta03, 1.1.0.28-beta03, 12.1.0.28-beta03, 13.1.0.28-beta03
- (2) Applicable Model: MUC1002, MUC1004, MUC2008, MUC2016
- (3) Release Date: January 18, 2019

CHANGES SINCE FIRMWARE RELEASE 20/1/12/13.1.0.28-beta02

2. New Features

None

3. Optimization

None

4. Bug Fixes

- (1) PBX Advanced Settings->SIP Settings: After the Local Network is saved, some data is lost.

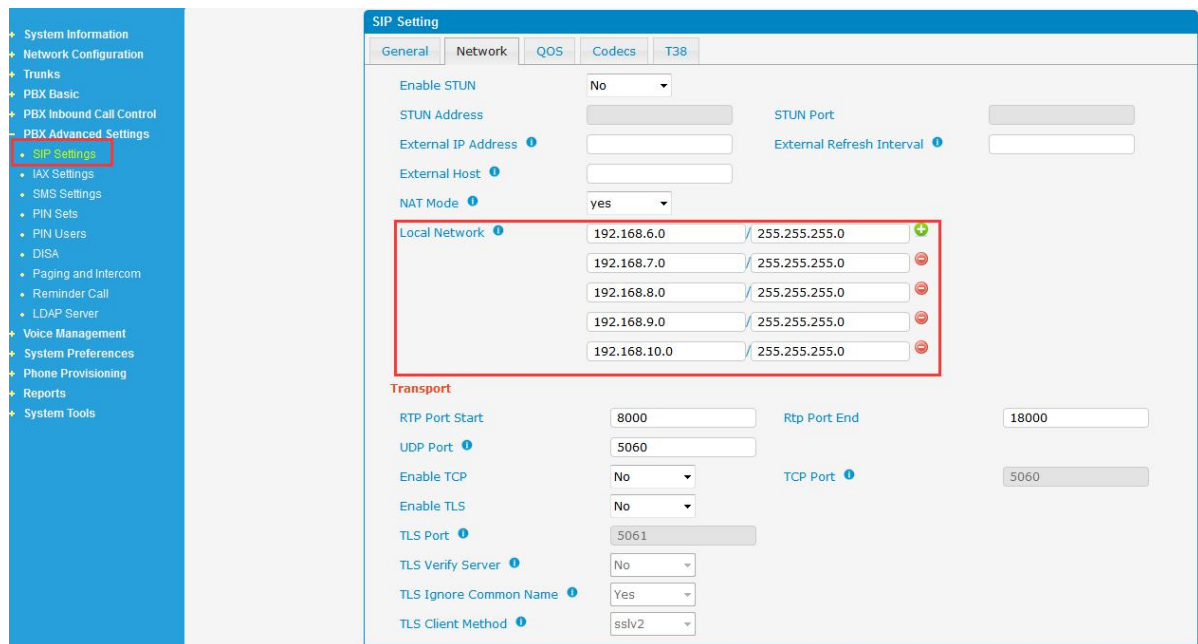
5. New Features Descriptions

None

6. Bug Fixes Description

- (1) PBX Advanced Settings->SIP Settings: After the Local Network is saved, some data is lost.

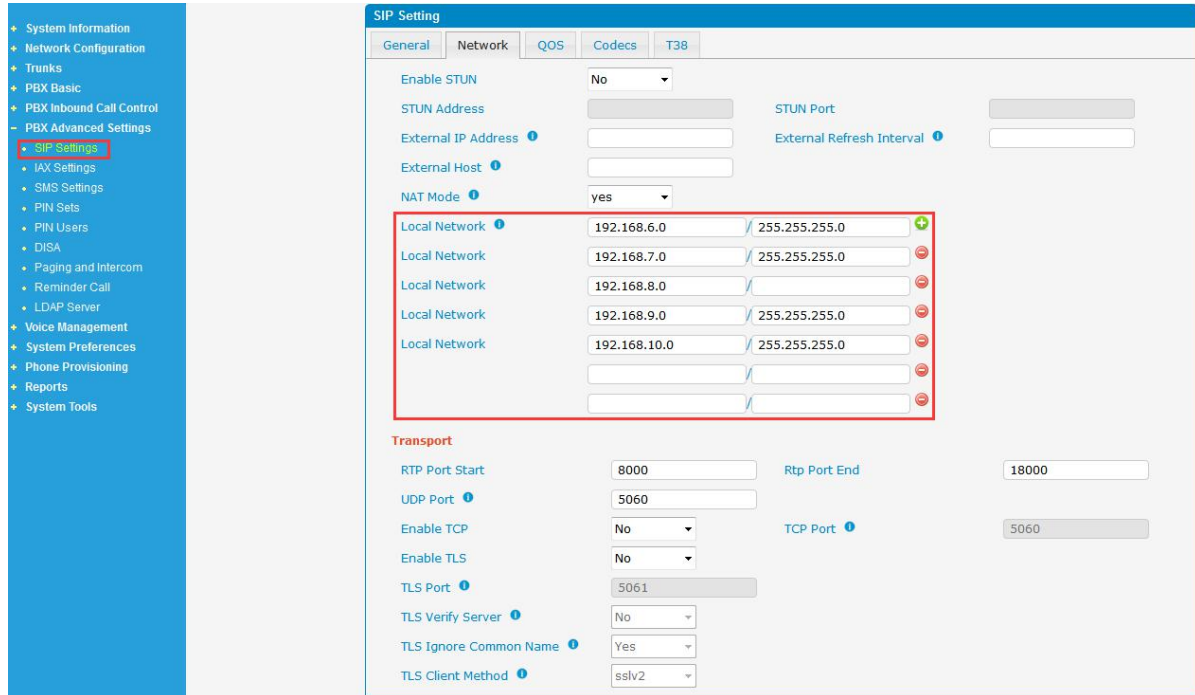
Figure->Before fixed the bug



The screenshot shows the 'SIP Setting' configuration page. On the left is a sidebar menu with 'SIP Settings' highlighted. The main panel has tabs for 'General', 'Network', 'QOS', 'Codecs', and 'T38'. The 'Network' tab is active, showing fields for 'Enable STUN', 'STUN Address', 'STUN Port', 'External IP Address', 'External Host', 'External Refresh Interval', and 'NAT Mode'. Below these is a 'Local Network' table with 5 rows, each containing an IP address and a port range. A red box highlights this table. Below the table is the 'Transport' section with fields for 'RTP Port Start', 'RTP Port End', 'UDP Port', 'Enable TCP', 'Enable TLS', 'TLS Port', 'TLS Verify Server', 'TLS Ignore Common Name', and 'TLS Client Method'.

Local Network	IP Address	Port Range
192.168.6.0	255.255.255.0	
192.168.7.0	255.255.255.0	
192.168.8.0	255.255.255.0	
192.168.9.0	255.255.255.0	
192.168.10.0	255.255.255.0	

Figure-> Saved before fixed the bug



SIP Setting

General Network QOS Codecs T38

Enable STUN: No

STUN Address: [Empty] STUN Port: [Empty]

External IP Address: [Empty] External Refresh Interval: [Empty]

External Host: [Empty]

NAT Mode: yes

Local Network List:

Local Network	192.168.6.0	255.255.255.0	+
Local Network	192.168.7.0	255.255.255.0	-
Local Network	192.168.8.0	255.255.255.0	-
Local Network	192.168.9.0	255.255.255.0	-
Local Network	192.168.10.0	255.255.255.0	-
Local Network	[Empty]	[Empty]	-
Local Network	[Empty]	[Empty]	-

Transport

RTP Port Start: 8000 Rtp Port End: 18000

UDP Port: 5060

Enable TCP: No TCP Port: 5060

Enable TLS: No

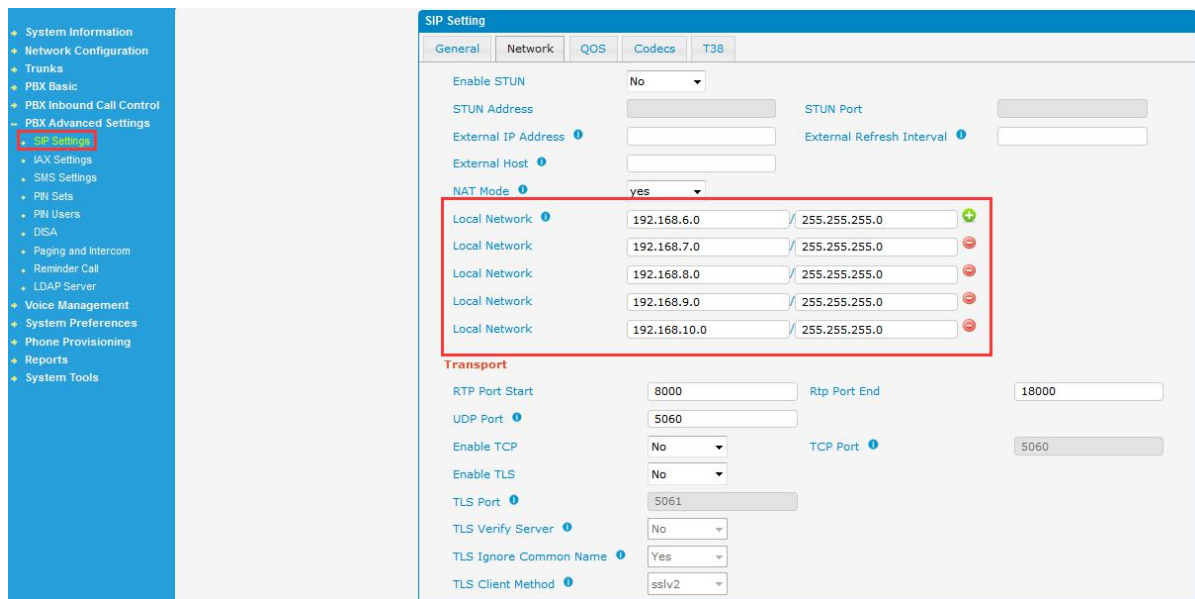
TLS Port: 5061

TLS Verify Server: No

TLS Ignore Common Name: Yes

TLS Client Method: sslv2

Figure-> Saved after fixed the bug



SIP Setting

General Network QOS Codecs T38

Enable STUN: No

STUN Address: [Empty] STUN Port: [Empty]

External IP Address: [Empty] External Refresh Interval: [Empty]

External Host: [Empty]

NAT Mode: yes

Local Network List:

Local Network	192.168.6.0	255.255.255.0	+
Local Network	192.168.7.0	255.255.255.0	-
Local Network	192.168.8.0	255.255.255.0	-
Local Network	192.168.9.0	255.255.255.0	-
Local Network	192.168.10.0	255.255.255.0	-
Local Network	[Empty]	[Empty]	-
Local Network	[Empty]	[Empty]	-

Transport

RTP Port Start: 8000 Rtp Port End: 18000

UDP Port: 5060

Enable TCP: No TCP Port: 5060

Enable TLS: No

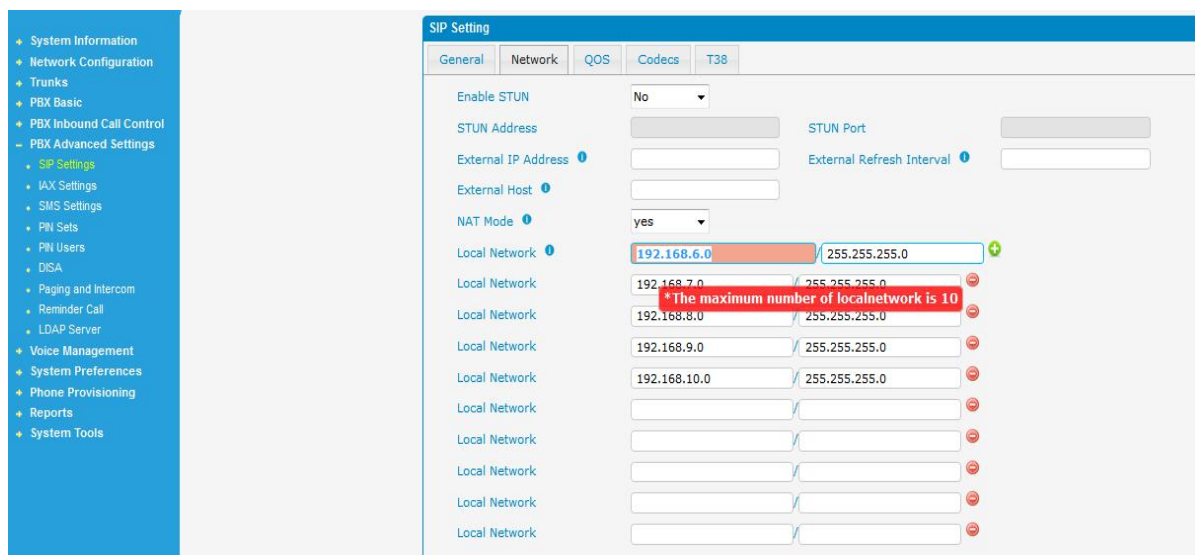
TLS Port: 5061

TLS Verify Server: No

TLS Ignore Common Name: Yes

TLS Client Method: sslv2

Figure->The maximum number of additions is limited to 10



SIP Setting

General Network QOS Codecs T38

Enable STUN: No

STUN Address: [Empty] STUN Port: [Empty]

External IP Address: [Empty] External Refresh Interval: [Empty]

External Host: [Empty]

NAT Mode: yes

Local Network List:

Local Network	192.168.6.0	255.255.255.0	+
Local Network	192.168.7.0	255.255.255.0	-
Local Network	192.168.8.0	255.255.255.0	-
Local Network	192.168.9.0	255.255.255.0	-
Local Network	192.168.10.0	255.255.255.0	-
Local Network	[Empty]	[Empty]	-
Local Network	[Empty]	[Empty]	-
Local Network	[Empty]	[Empty]	-
Local Network	[Empty]	[Empty]	-
Local Network	[Empty]	[Empty]	-
Local Network	[Empty]	[Empty]	-

Transport

RTP Port Start: 8000 Rtp Port End: 18000

UDP Port: 5060

Enable TCP: No TCP Port: 5060

Enable TLS: No

TLS Port: 5061

TLS Verify Server: No

TLS Ignore Common Name: Yes

TLS Client Method: sslv2

*The maximum number of localnetwork is 10

Path: PBX Advanced Settings->SIP Settings

Description: Before the bug is fixed, when the Local Network is saved, some data is lost. After the bug is fixed, the data will not be deleted, and the maximum number of additions is limited to 10.

✧ Release Notes of Version 20/1/12/13.1.0.28-beta02

1. Introduction

- (1) Firmware Version:
20.1.0.28-beta02, 1.1.0.28-beta02, 12.1.0.28-beta02, 13.1.0.28-beta02
- (2) Applicable Model: MUC1002, MUC1004, MUC2008, MUC2016
- (3) Release Date: December 12, 2018

CHANGES SINCE FIRMWARE RELEASE 20/1/12/13.1.0.28-beta01

2. New Features

- (1) Added "Voicemail List" on "Reports"

3. Optimization

- (1) Optimized the save of special destination in Inbound Route.

4. Bug Fixes

5. New Features Descriptions

- (1) Added "Voicemail List" on "Reports"

Figure->Voicemail List in Admin

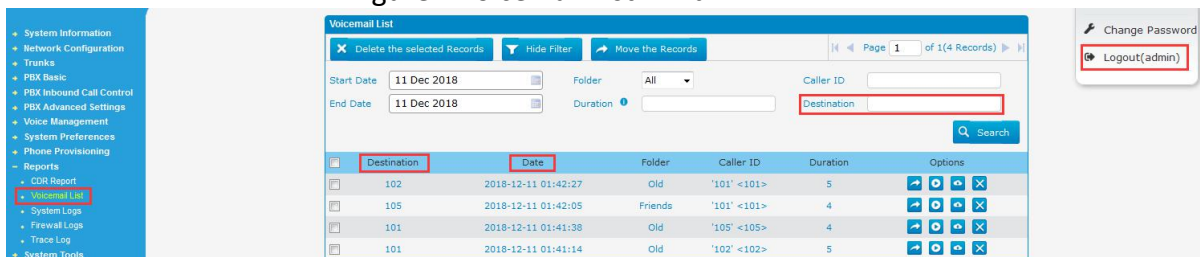
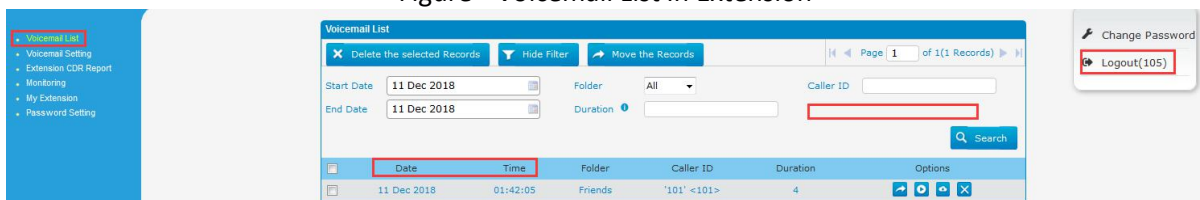


Figure->Voicemail List In Extension



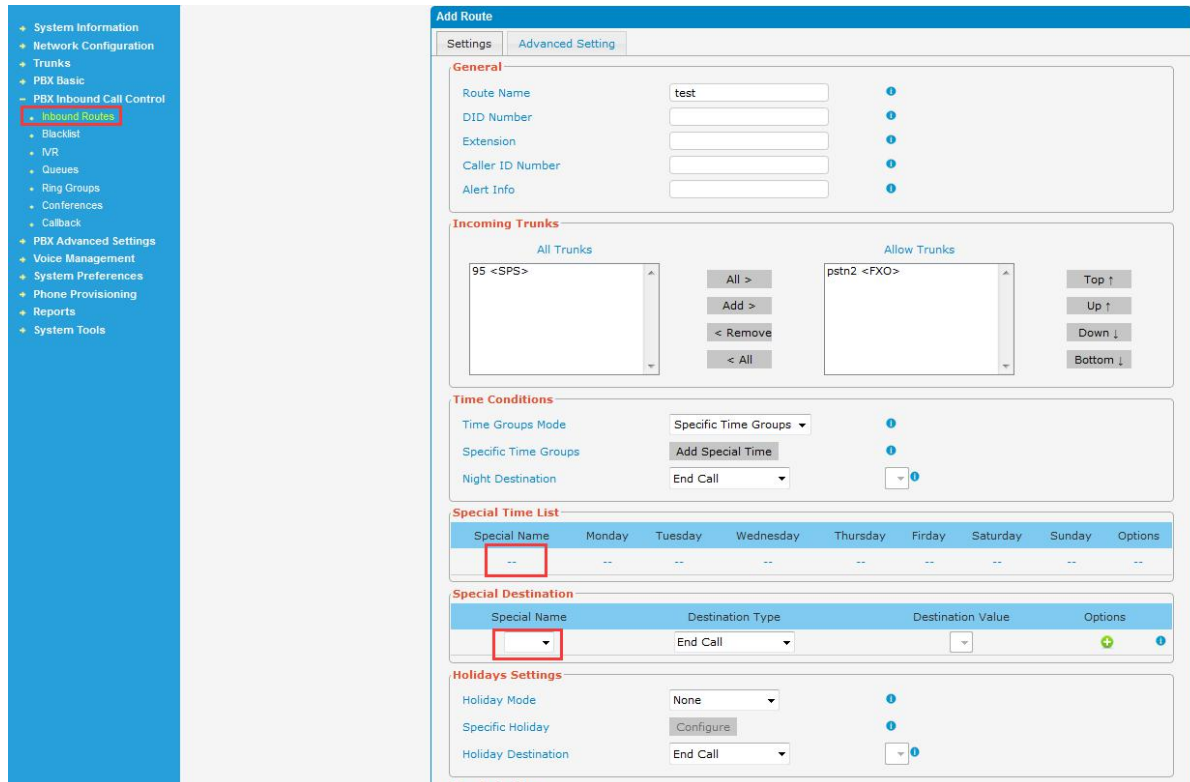
Path: Reports->Voicemail List

Description: Integrate the voicemail records of each extension into the Admin interface for unified processing. In the Admin page,

- (1) Add Destination to distinguish each extension.
- (2) To integrate date and time together, easy to sort by time.
- (3) Add destination in the Filter.

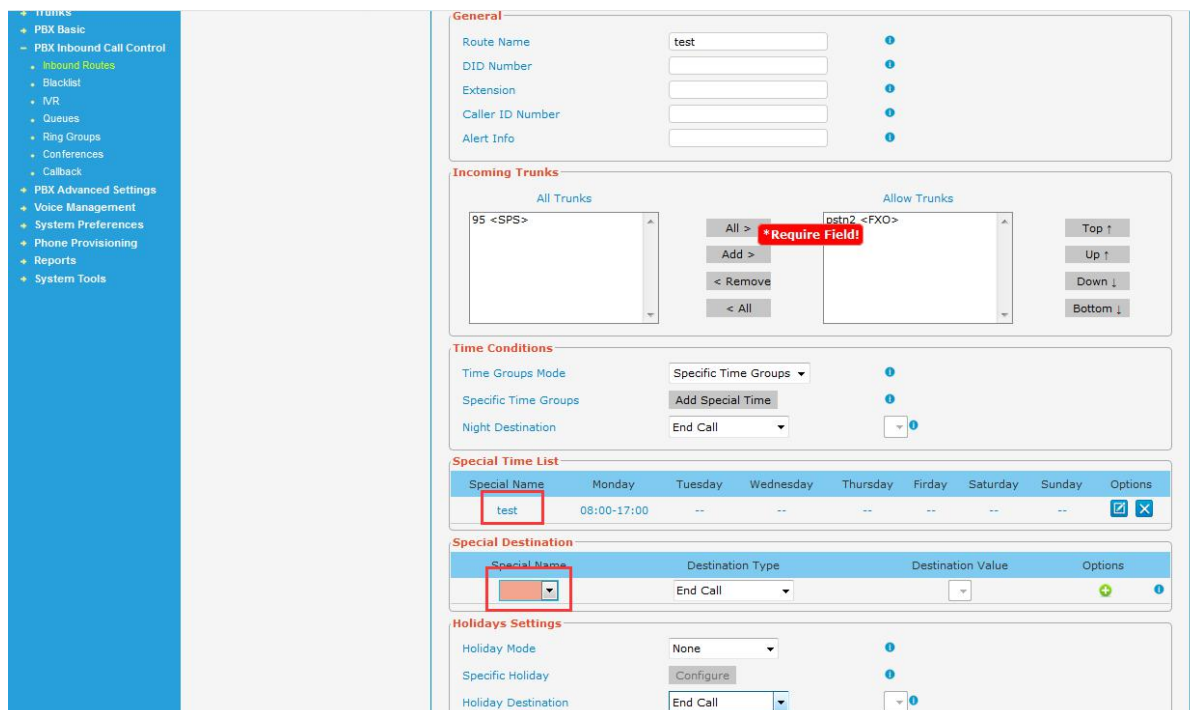
6. Optimization Description

Figure->Special Time List is empty



The screenshot shows the 'Add Route' configuration page. The 'Special Time List' table is empty, with columns for Special Name, Day of the week, and Options. The 'Special Destination' table is also empty, with columns for Special Name, Destination Type, Destination Value, and Options. The 'Holidays Settings' section shows 'Holiday Mode' set to 'None' and 'Holiday Destination' set to 'End Call'.

Figure->Special Time List isn't empty



The screenshot shows the 'Add Route' configuration page with a non-empty 'Special Time List'. The table has one row with 'test' as the Special Name, '08:00-17:00' as the time range, and 'test' as the destination. The 'Special Destination' table is empty. A red box highlights the 'Special Name' field in the 'Special Destination' table, and a red arrow points to it with the text '*Require Field!'.

Path: PBX Inbound Call Control->Inbound Routes

Description: When the Special Time List and Special Destination are both empty, clicking Save will not prompt an error. When the Special Time List is not empty and the Special Destination is empty, clicking Save will prompt an error.

✧ Release Notes of Version 20/1/12/13.1.0.28-beta01

1. Introduction

- (1) Firmware Version:
20.1.0.28-beta01, 1.1.0.28-beta01, 12.1.0.28-beta01, 13.1.0.28-beta01
- (2) Applicable Model: MUC1002, MUC1004, MUC2008, MUC2016
- (3) Release Date: November 9, 2018

CHANGES SINCE FIRMWARE RELEASE 20/1/12/13.1.0.28

2. New Features

- (1) Added "Special Time Groups" on "Inbound Routes"

3. Optimization

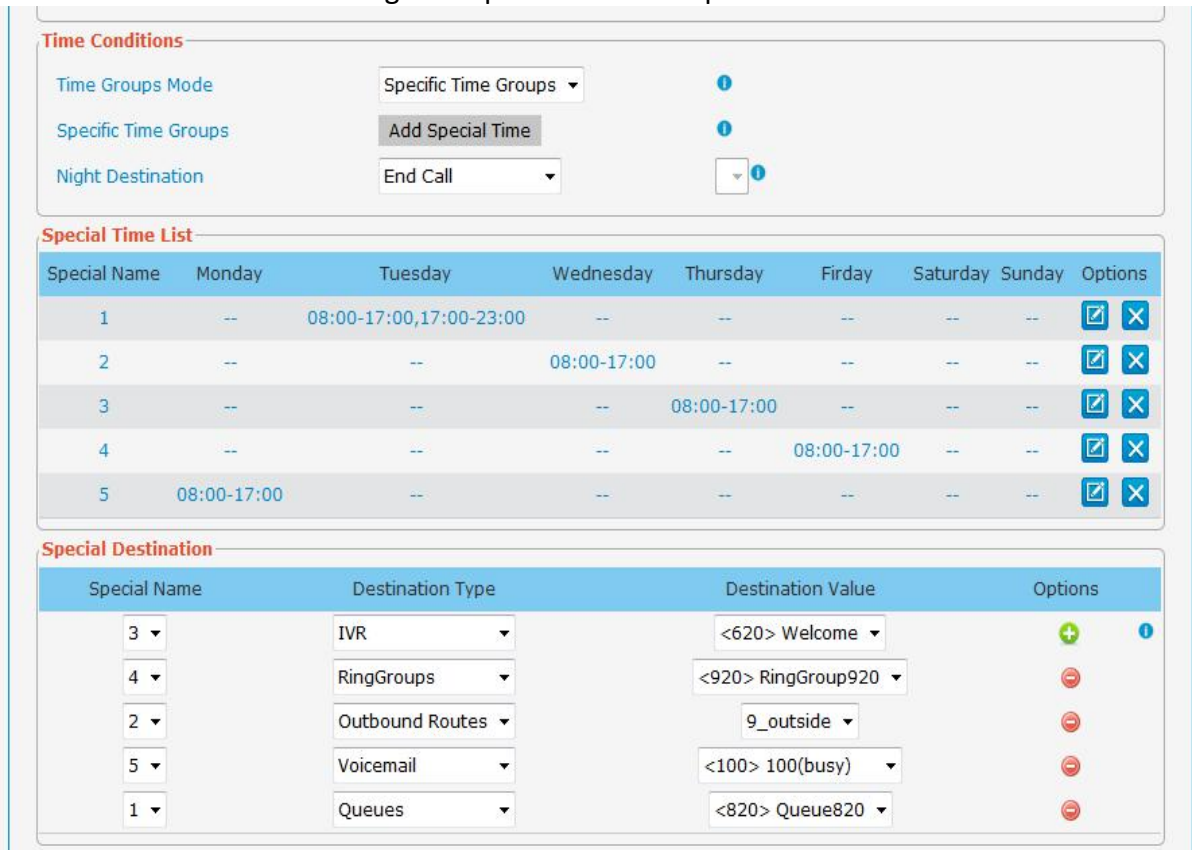
- (1) Optimized the display of outbound CID in cdr report source

4. Bug Fixes

5. New Features Descriptions

- (1) Added "Special Time Groups" on "Inbound Routes"

Figure->Special Time Groups



Time Conditions

Time Groups Mode: Specific Time Groups

Specific Time Groups: Add Special Time

Night Destination: End Call

Special Time List

Special Name	Monday	Tuesday	Wednesday	Thursday	Friday	Saturday	Sunday	Options
1	--	08:00-17:00, 17:00-23:00	--	--	--	--	--	[Edit] [Delete]
2	--	--	08:00-17:00	--	--	--	--	[Edit] [Delete]
3	--	--	--	08:00-17:00	--	--	--	[Edit] [Delete]
4	--	--	--	--	08:00-17:00	--	--	[Edit] [Delete]
5	08:00-17:00	--	--	--	--	--	--	[Edit] [Delete]

Special Destination

Special Name	Destination Type	Destination Value	Options
3	IVR	<620> Welcome	[Add] [Info]
4	RingGroups	<920> RingGroup920	[Remove]
2	Outbound Routes	9_outside	[Remove]
5	Voicemail	<100> 100(busy)	[Remove]
1	Queues	<820> Queue820	[Remove]

Path: PBX Inbound Call Control->Inbound Routes-->Add Route(Edit Route)

Description: You can route calls to different destination at different times. Calls that do not match the time periods will be routed to Other Time destination. If all the set time periods do not match, they will be transferred to Night Destination.

6. Optimization Description

(1) Optimized the display of outbound CID in cdr report source

Figure->Before optimization

CDR Report									
X Delete the records		Show Filter		Download the records		Page 1 of 1 (13 Records)			
Date	Source	Destination	Src. Trunk	Account Code	Dst. Trunk	Call Direction	Status	Duration	Billing Duration
2018-10-31 10:28:41	568932	505	to48			Inbound	ANSWERED	12s	12s
2018-10-31 10:27:26	568932	620(505)	to48			Inbound	ANSWERED	7s	2s
2018-10-31 10:24:21	568932	500	to48			Inbound	ANSWERED	14s	14s
2018-10-31 10:22:13	568932	620(500)	to48			Inbound	ANSWERED	7s	1s

Figure->After optimization

2018-10-30 18:25:54	568932(202)	5123			96	Outbound	ANSWERED	13s	13s
2018-10-30 18:24:39	568932(201)	5123			96	Outbound	ANSWERED	8s	8s
2018-10-30 18:21:34	568932(201)	501			96	Outbound	ANSWERED	14s	14s
2018-10-30 18:20:20	201	103				Internal	NO ANSWER	2s	0s
2018-10-30 18:20:13	201	103				Internal	NO ANSWER	4s	0s
2018-10-30 18:19:26	568932(103)	52			96	Outbound	ANSWERED	8s	8s

Path: Reports->CDR Reports

Description: The number(568932) is CID number of Outbound Route. The number(202 and 201) is Calling number. The optimized CID number will be accompanied by the calling number.

✧ Release Notes of Version 20/1/12/13.1.0.28

1. Introduction

- (1) Firmware Version: 20.1.0.28, 1.1.0.28, 12.1.0.28, 13.1.0.28
- (2) Applicable Model: MUC1002, MUC1004, MUC2008, MUC2016
- (3) Release Date: October 29, 2018

CHANGES SINCE FIRMWARE RELEASE 1/12/13.1.0.27

2. New Features

- (1) Add X series phone of Fanvil.

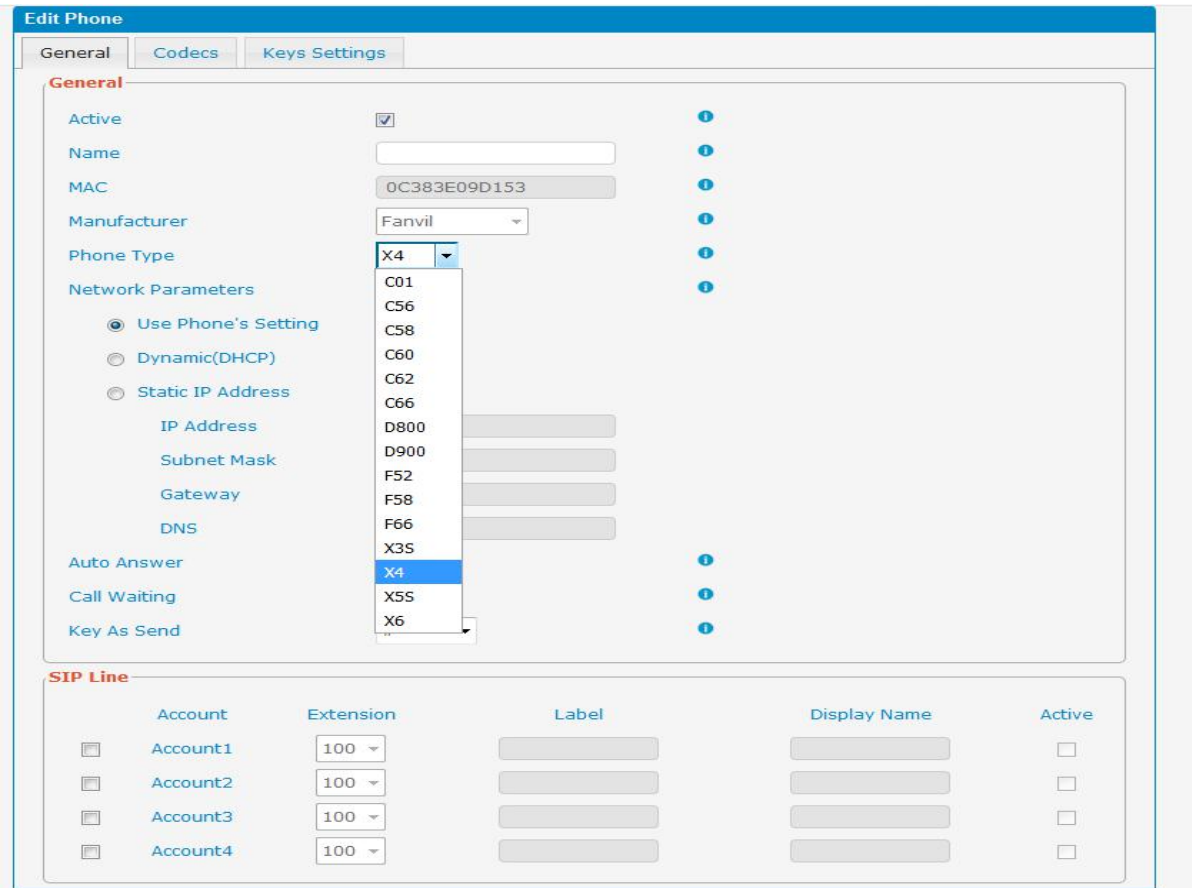
3. Optimization

4. Bug Fixes

5. New Features Descriptions

- (1) Add X series phone of Fanvil.

Figure1->Web edit phone



Edit Phone

General | Codecs | Keys Settings

General

Active ☒ ⓘ

Name ⓘ

MAC 0C383E09D153 ⓘ

Manufacturer Fanvil ⓘ

Phone Type **X4** ⓘ

Network Parameters

- ☒ Use Phone's Setting
- ☐ Dynamic(DHCP)
- ☐ Static IP Address

IP Address

Subnet Mask

Gateway

DNS

Auto Answer ⓘ

Call Waiting ⓘ

Key As Send ⓘ

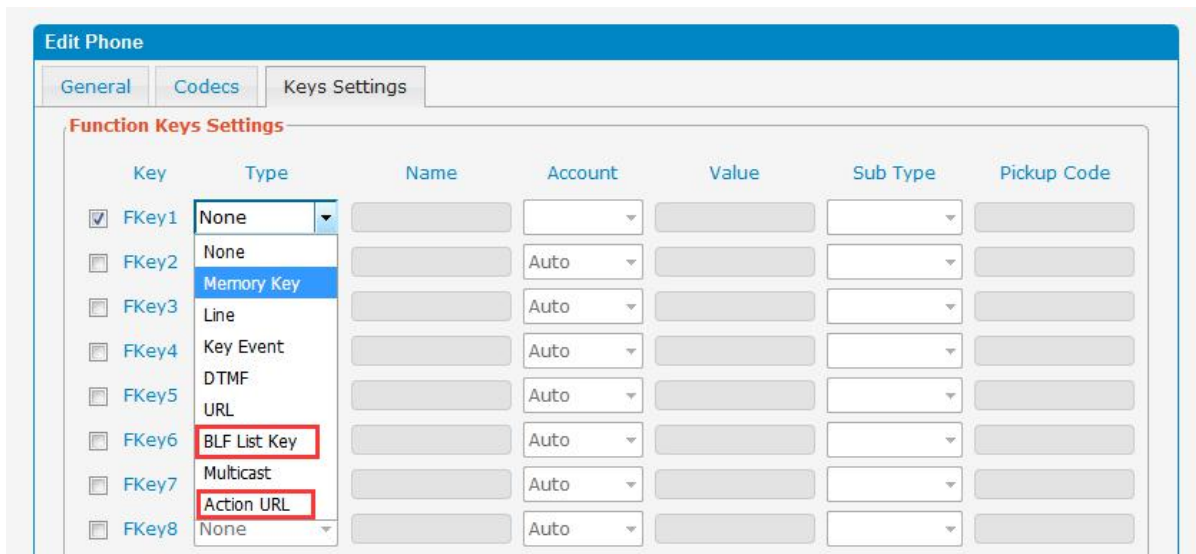
SIP Line

	Account	Extension	Label	Display Name	Active
☐	Account1	100			☐
☐	Account2	100			☐
☐	Account3	100			☐
☐	Account4	100			☐

Path: Phone Provisioning --> Phones

Description: The new X-series fanvil phones are modeled as 'X3S', 'X4', 'X5S' and 'X6'. X3S has 2 SIP Lines, 2 DSS Keys; X4 has 4 SIP Lines, 30 DSS Keys; X5 has 6 SIP Lines, 40 DSS Keys; X6 has 6 SIP Lines and 60 DSS Keys.

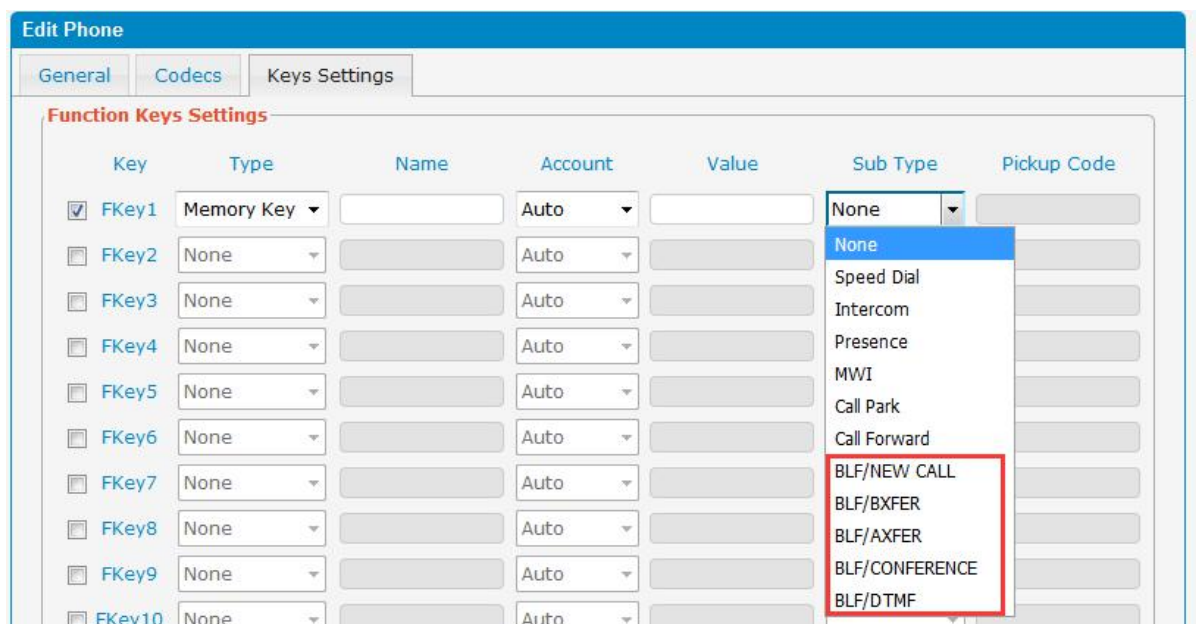
Figure2->New items about Type



Key	Type	Name	Account	Value	Sub Type	Pickup Code
☒ FKey1	None					
☐ FKey2	None		Auto			
☐ FKey3	Memory Key		Auto			
☐ FKey4	Line		Auto			
☐ FKey5	Key Event		Auto			
☐ FKey6	DTMF		Auto			
☐ FKey7	URL		Auto			
☐ FKey8	BLF List Key		Auto			
☐ FKey9	Multicast		Auto			
☐ FKey10	Action URL		Auto			
☐ FKey11	None		Auto			

Figure2:Added BLF List Key and Action URL

Figure3->New items about Sub Type of Memory Key



Key	Type	Name	Account	Value	Sub Type	Pickup Code
☒ FKey1	Memory Key		Auto		None	
☐ FKey2	None		Auto		None	
☐ FKey3	None		Auto		Speed Dial	
☐ FKey4	None		Auto		Intercom	
☐ FKey5	None		Auto		Presence	
☐ FKey6	None		Auto		MWI	
☐ FKey7	None		Auto		Call Park	
☐ FKey8	None		Auto		Call Forward	
☐ FKey9	None		Auto		BLF/NEW CALL	
☐ FKey10	None		Auto		BLF/BXFER	
☐ FKey11	None		Auto		BLF/AXFER	
☐ FKey12	None		Auto		BLF/CONFERENCE	
☐ FKey13	None		Auto		BLF/DTMF	

Figure3:Replace the original BLF with BLF/NEW CALL, BLF/BXFER, BLF/AXFER, BLF/CONFERENCE and BLF/DTMF in Memory Key

Figure4->New items about Sub Type Key Event

Edit Phone

General | Codecs | **Keys Settings**

Function Keys Settings

Key	Type	Name	Account	Value	Sub Type	Pickup Code
☒ FKey1	Key Event				Phone Book	
☐ FKey2	None		Auto			
☐ FKey3	None		Auto			
☐ FKey4	None		Auto			
☐ FKey5	None		Auto			
☐ FKey6	None		Auto			
☐ FKey7	None		Auto			
☐ FKey8	None		Auto			
☐ FKey9	None		Auto			
☐ FKey10	None		Auto			
☐ FKey11	None		Auto			
☐ FKey12	None		Auto			
☐ FKey13	None		Auto			
☐ FKey14	None		Auto			
☐ FKey15	None		Auto			

Sub Type Dropdown Menu:

- Phone Book
- DND
- Release
- Lock
- SMS
- Prefix
- Hot Desking
- Power Light
- Hide Dtmf
- Agent
- Private Hold
- Disposition
- Escalate
- Auto Headset
- Record
- Reset to factory default
- Local Contacts
- Group Listen
- LDAP
- Cloud PhoneBook
- Broad Soft

Figure4:New options such as Reset to factory default in Key Event

✧ Release Notes of Version 1/12/13.1.0.27

1. Introduction

- (1) Firmware Version: 1.1.0.27, 12.1.0.27, 13.1.0.27
- (2) Applicable Model: MUC1004, MUC2008, MUC2016
- (3) Release Date: August 9, 2018

CHANGES SINCE FIRMWARE RELEASE 1/12/13.1.0.22

2. New Features

- (1) Added Portuguese (Brazil customer).
- (2) Added "VPN Server" on "Network Configuration"
- (3) Added "VPN Client" on "Network Configuration"
- (4) Increase the Time Zone configuration of the Htek phone Phone Prov
- (5) Added "LDAP Server" on "PBX Advanced Settings"
- (6) Added "operator" on "Voice Management->Voicemail Settings"
- (7) Added "User Permission" on "monitor->Advanced Settings"
- (8) Added "Callback When Busy"
- (9) Added "Three-way Conferences"
- (10) Added "Import and Export" on "admin -> PBX Basic->Extensions"

3. Optimization

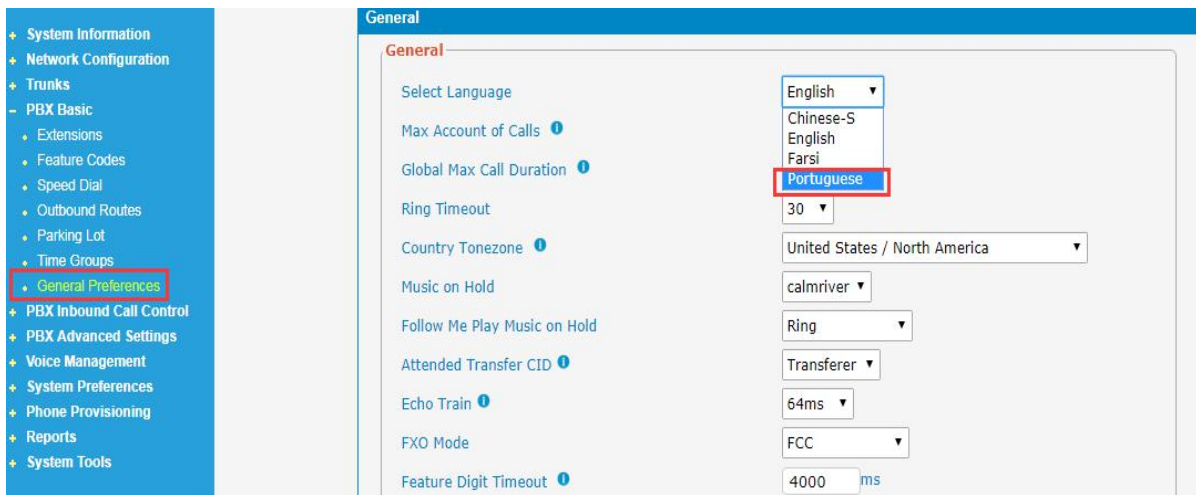
4. Bug Fixes

- (1) Fix the first GSM trunk in the outgoing route is in the call, the second call out cannot pass the next trunk error
- (2) Queue password limit adjustment

5. New Features Descriptions

- (1) Added Portuguese (Brazil customer).
Path: PBX Basic --> General Settings

Figure-Portuguese



General

Select Language: English

Max Account of Calls: 1

Global Max Call Duration: 1

Ring Timeout: 30

Country Tonezone: United States / North America

Music on Hold: calmriver

Follow Me Play Music on Hold: Ring

Attended Transfer CID: Transferer

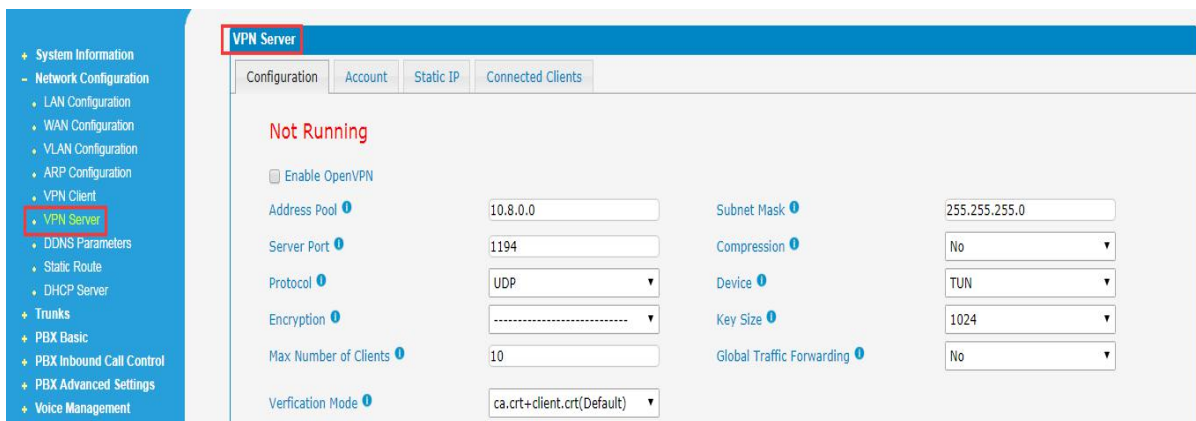
Echo Train: 64ms

FXO Mode: FCC

Feature Digit Timeout: 4000 ms

(2) Added “VPN Server” on “Network Configuration”

Figure->VPN Server



VPN Server

Configuration Account Static IP Connected Clients

Not Running

Enable OpenVPN: ☐

Address Pool: 10.8.0.0

Subnet Mask: 255.255.255.0

Server Port: 1194

Compression: No

Protocol: UDP

Device: TUN

Encryption: -----

Key Size: 1024

Max Number of Clients: 10

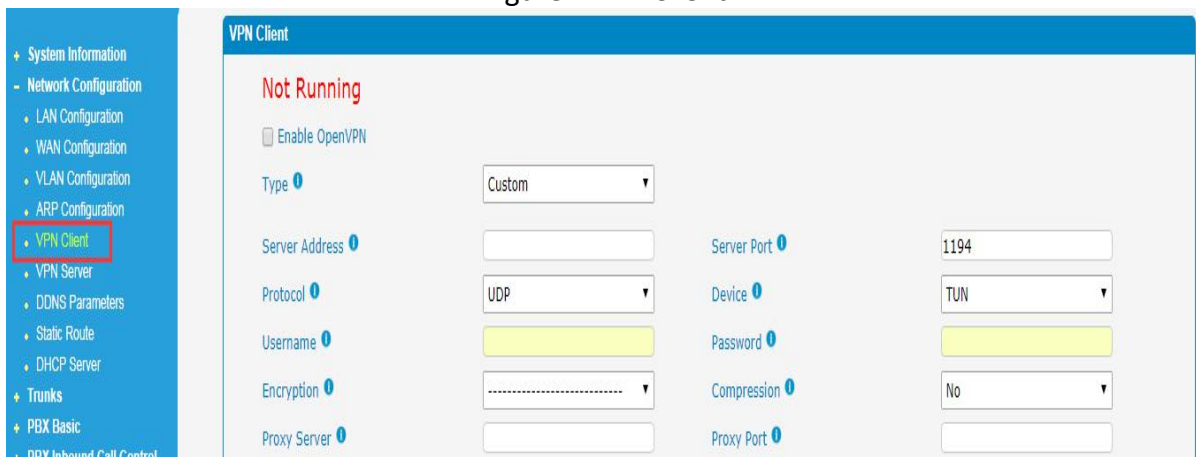
Global Traffic Forwarding: No

Verification Mode: ca.crt+client.crt(Default)

Path: Network Configuration-->VPN Server

(3) Added “VPN Client” on “Network Configuration”

Figure->VPN Client



VPN Client

Not Running

Enable OpenVPN: ☐

Type: Custom

Server Address:

Server Port: 1194

Protocol: UDP

Device: TUN

Username:

Password:

Encryption: -----

Compression: No

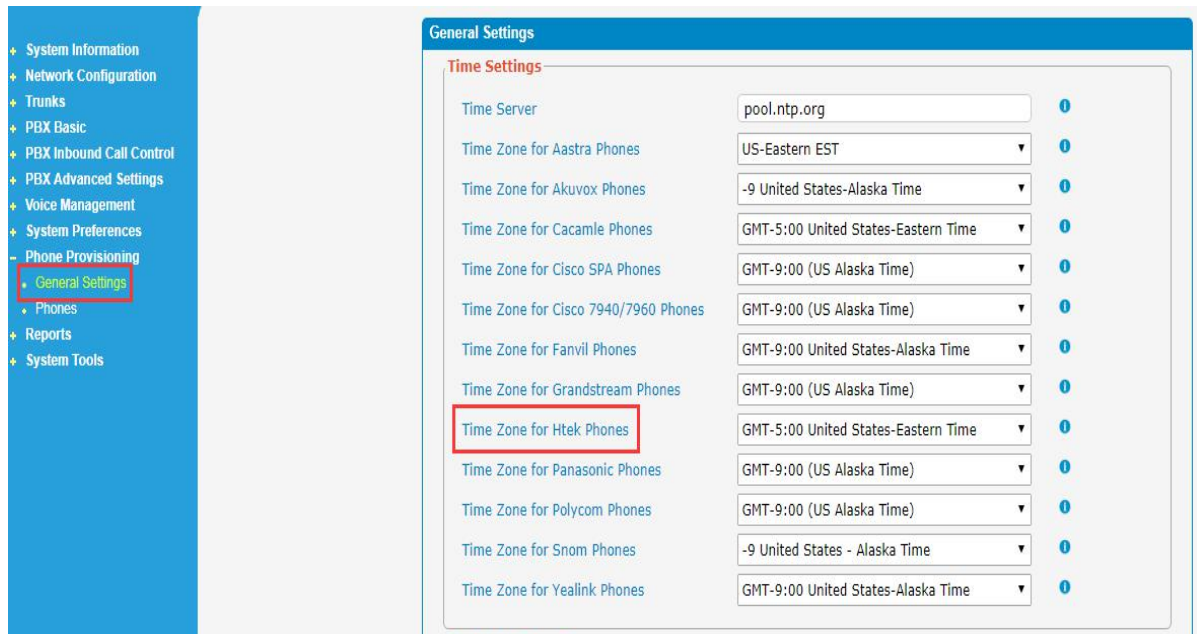
Proxy Server:

Proxy Port:

Path: Network Configuration -->VPN Client

(4) Increase the Time Zone configuration of the Htek phone PhoneProv

Figure-->Time zone



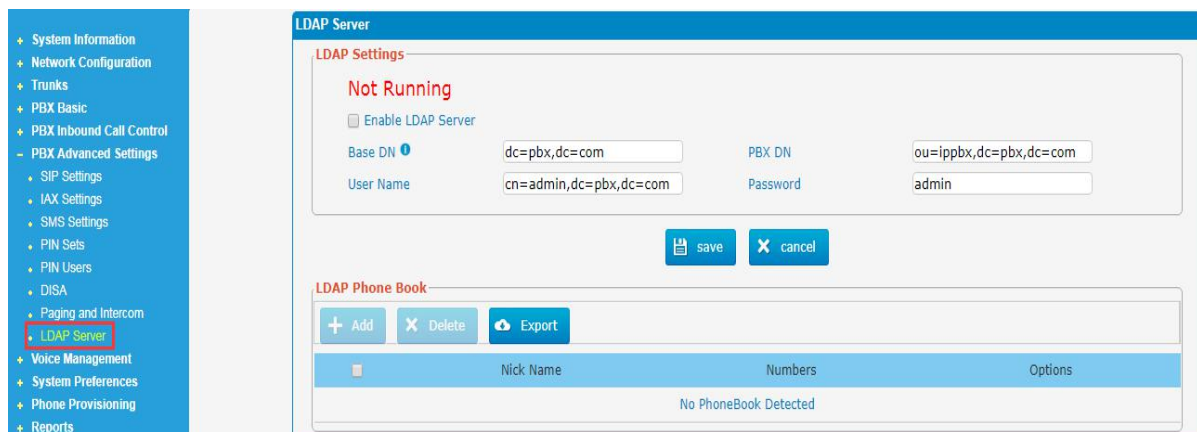
General Settings

Time Settings

Phone Model	Time Zone	Icon
Time Server	pool.ntp.org	0
Time Zone for Aastra Phones	US-Eastern EST	0
Time Zone for Akuvox Phones	-9 United States-Alaska Time	0
Time Zone for Cacomle Phones	GMT-5:00 United States-Eastern Time	0
Time Zone for Cisco SPA Phones	GMT-9:00 (US Alaska Time)	0
Time Zone for Cisco 7940/7960 Phones	GMT-9:00 (US Alaska Time)	0
Time Zone for Fanvil Phones	GMT-9:00 United States-Alaska Time	0
Time Zone for Grandstream Phones	GMT-9:00 (US Alaska Time)	0
Time Zone for Htek Phones	GMT-5:00 United States-Eastern Time	0
Time Zone for Panasonic Phones	GMT-9:00 (US Alaska Time)	0
Time Zone for Polycom Phones	GMT-9:00 (US Alaska Time)	0
Time Zone for Snom Phones	-9 United States - Alaska Time	0
Time Zone for Yealink Phones	GMT-9:00 United States-Alaska Time	0

(5) Added "LDAP Server" on "PBX Advanced Settings"

Figure->LDAP Server



LDAP Server

LDAP Settings

Not Running

☐ Enable LDAP Server

Base DN: dc=pbx,dc=com PBX DN: ou=ippbx,dc=pbx,dc=com

User Name: cn=admin,dc=pbx,dc=com Password: admin

LDAP Phone Book

Nick Name	Numbers	Options
No PhoneBook Detected		

Path: PBX Advanced Settings->LDAP Server

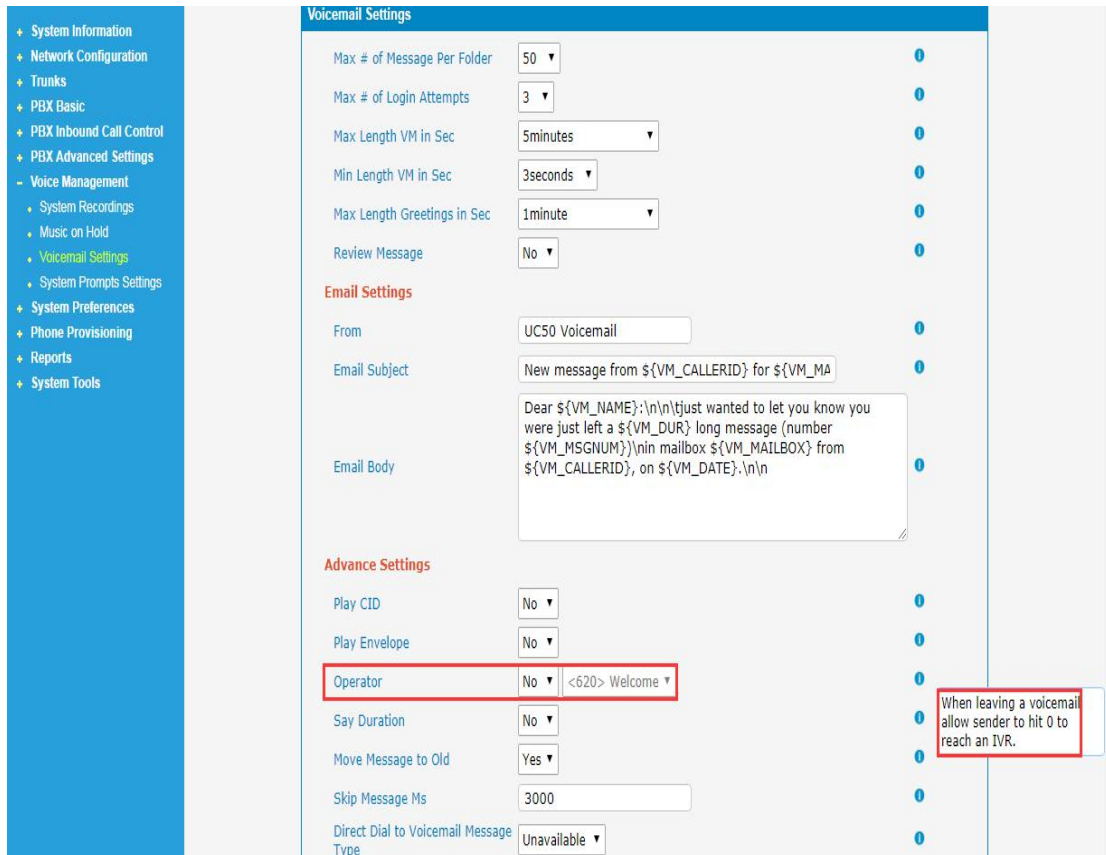
Description: The main idea of LDAP is to keep in one place all the information of a user (contact details, login, password), so that it is easier to maintain by network administrators. For example you can:

- use the same login/password to login on an Intranet.
- get all the contact details of the people in a company on Outlook for example.

LDAP is used as a phone book on PBX so that you can search a key word from your IP phone.

(6) Added "operator" on "Voice Management->Voicemail Settings"

Figure->operator



Path: Voice Management->Voicemail Settings

Description: when leaving a voicemail allow sender to hit 0 to reach an IVR.

(7) Added "User Permission" on "monitor->Advanced Settings"

Login with username: monitor

Path: Advanced Settings --> User Permission

Descriptions: Add user permission functionality to Monitor, create new users and allow users to control what is displayed in the web page. Options are USB devices, Recording Settings and Call Detail Records

Figure-> User Permission

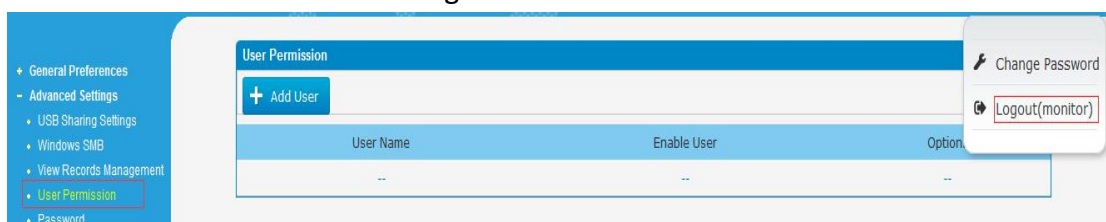


Figure- Add User Permission

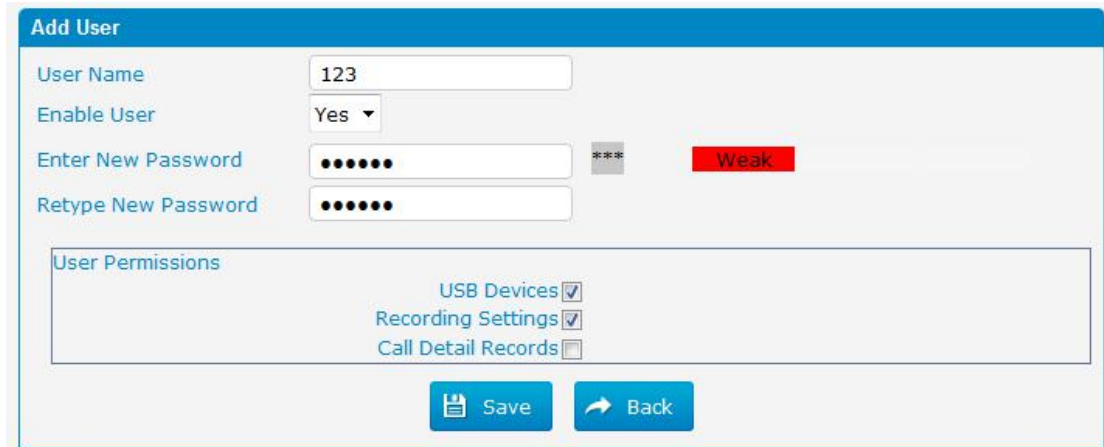


Figure-Edit User Permission

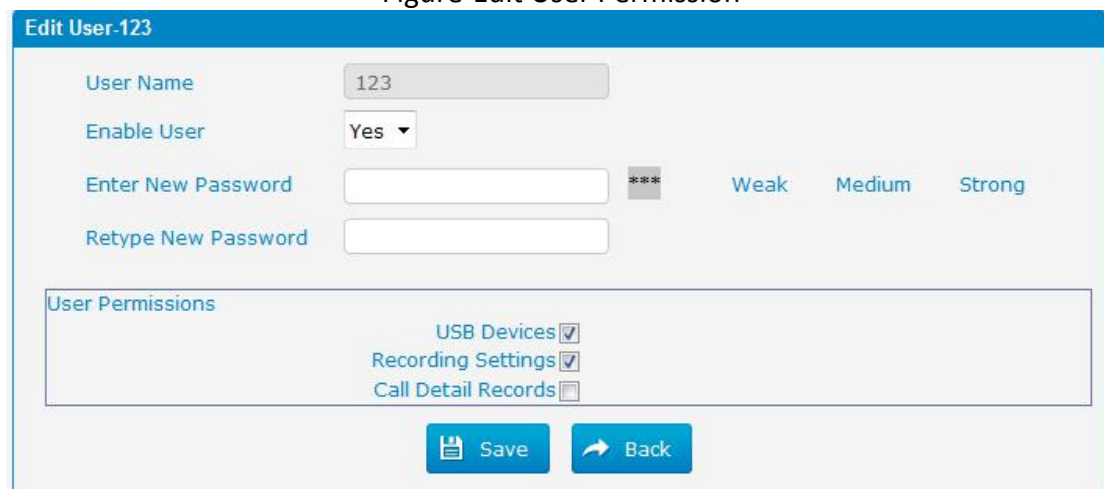
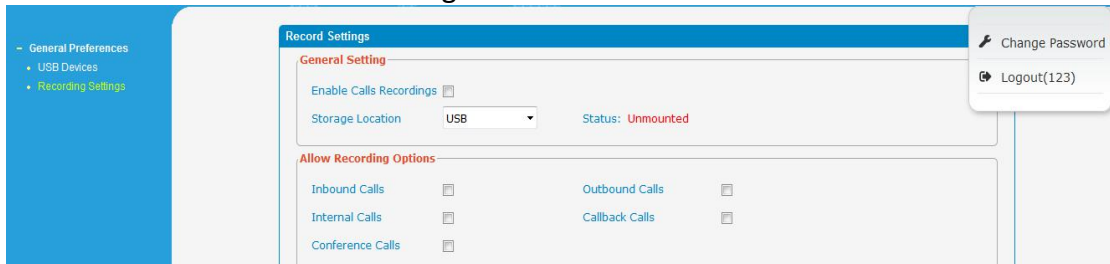


Figure- User Interface



After you have created a new user, you need to go back to the login page, enter the user name and password of the new user, and enter the user interface.

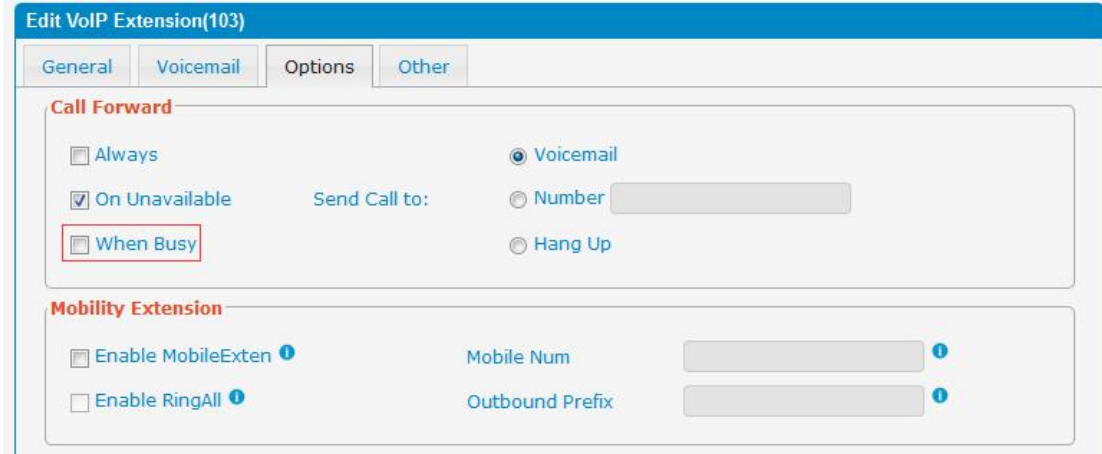
(8) Added “Callback When Busy”

Descriptions: When the callee user is on the phone, after hearing the prompts, then dial 5 to enable this function, hang up. Within 10 seconds after the callee user hangs up, the caller user will ring the bell first, and when connected, the callee user will ring the bell. The maximum waiting time for the callee user hangs up is 20 minutes. For timeout, repeat the operation.

The conditions for the callback to be effective are:

Path: PBX Basic-> extensions-> Edit VoIP (FXS) Extensions-> Options-> Call Forward-> When Busy .The ‘When Busy’ option not enabled.

Figure-Callback effective condition 1



Edit VoIP Extension(103)

General Voicemail Options Other

Call Forward

☐ Always ☒ Voicemail

☒ On Unavailable Send Call to: ☐ Number

☒ When Busy ☐ Hang Up

Mobility Extension

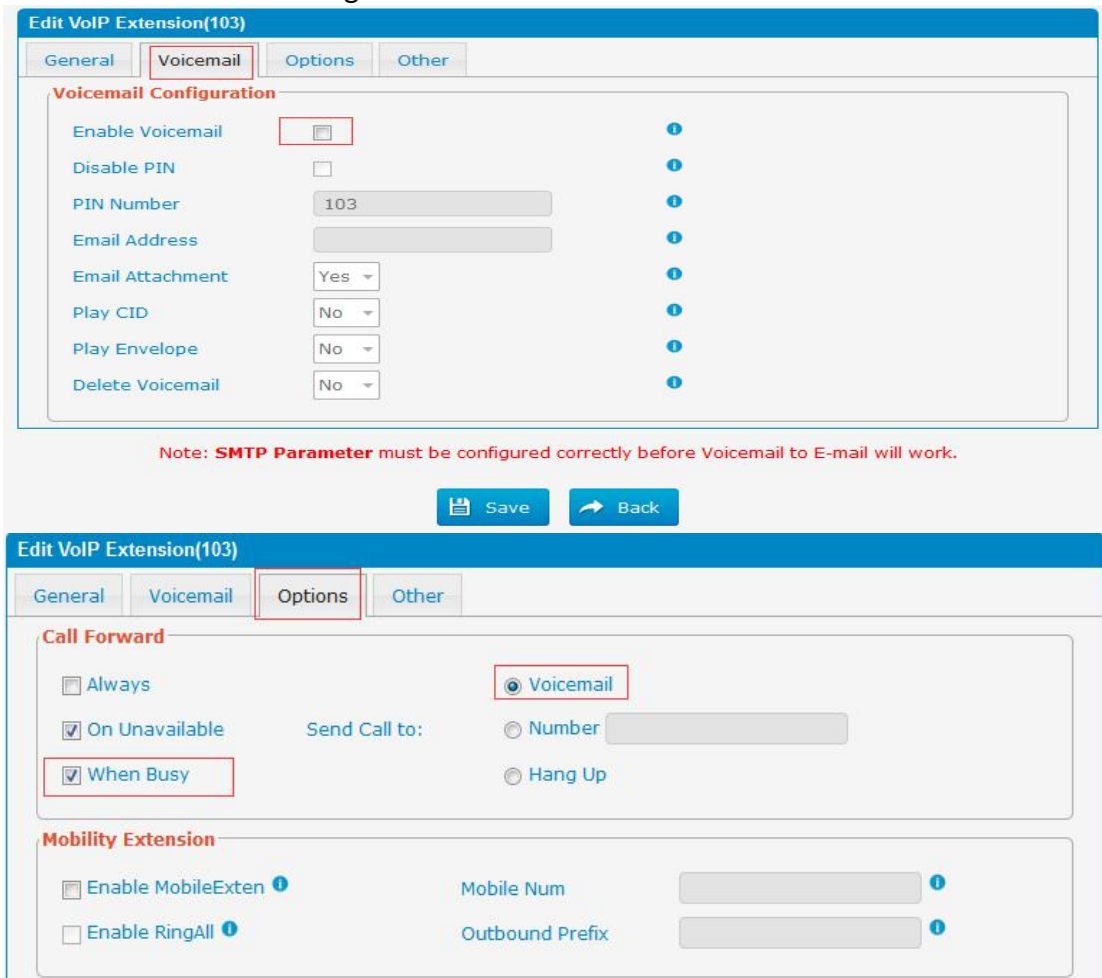
☐ Enable MobileExten ⁱ Mobile Num ⁱ

☐ Enable RingAll ⁱ Outbound Prefix ⁱ

Path: PBX Basic-> extensions-> Edit VOIP (FXS) Extensions-> Options-> Voicemail->Enable Voicemail. The 'Enable Voicemail' option not enabled.

Path: PBX Basic-> extensions-> Edit VOIP (FXS) Extensions-> Options-> Call Forward-> When Busy and Voicemail. The 'When Busy' option and 'Voicemail' option enabled.

Figure-Callback effective condition 2



Edit VoIP Extension(103)

General Voicemail Options Other

Voicemail Configuration

Enable Voicemail ☐ ⁱ

Disable PIN ☐ ⁱ

PIN Number ⁱ

Email Address ⁱ

Email Attachment Yes ⁱ

Play CID No ⁱ

Play Envelope No ⁱ

Delete Voicemail No ⁱ

Note: **SMTP Parameter** must be configured correctly before Voicemail to E-mail will work.

Save Back

Edit VoIP Extension(103)

General Voicemail Options Other

Call Forward

☐ Always ☒ Voicemail

☒ On Unavailable Send Call to: ☐ Number

☒ When Busy ☐ Hang Up

Mobility Extension

☐ Enable MobileExten ⁱ Mobile Num ⁱ

☐ Enable RingAll ⁱ Outbound Prefix ⁱ

Notes:

- (1) Modify: The 'When Busy' is not enabled by default.
- (2) If the option 'Always' enabled in Call Forward, the Callback When Busy will not effective.
- (3) If you want to enable other functions when busy, such as Voicemail, Number or Hang up. You need to enabled the option 'when Busy'.

(9) Added “Three-way Conferences”

Descriptions: To invite others into the conference room during the call.

Operating Process, for example:

Step 1: Prepare three extensions, 101, 102, 103.

Step 2: Extension 101 dial 102, extension 102 answered. At this time, extension 101 dial *00, extension 101 to hear the dial tone, extension 102 enter the conference room 860 and hear the waiting music.

Step 3: During the dial tone, extension 101 dial 103#, then extension 103 ringing and answered it. After that, extension 101 dial *11, extensions 101, 102, and 103 enter conference room 860 together.

Notes:

- (1) The callee user dial *00 will not make a correct Three-way conferences.
- (2) Three-way conferences with external calls is not be effective.
- (3) The dial tone lasts about 10 seconds. If you do not dial the extension number you want to invite, the caller (such as 101) and the callee (such as 102) will enter the conference room automatically. In this moment, the callee(102) can invite a new user(such as 103) into the conference room by dialing 0 and then following **Step 3**.
- (4) At present, the three-party calling function is only implemented when an internal call is made, and the incoming call through the external relay cannot be realized.

(10) Added “Import and Export” on “admin -> PBX Basic->Extensions”

Login with username: admin

Path: PBX Basic->Extensions->Import(Export)

Descriptions: Add Import and Export functionality to Extensions, then user can import the new extensions by filling a fixed format CSV form and using the import function. You can also export the total extensions via the export function.

Figure-Import and Export

- System Information - Network Configuration - Trunks - PBX Basic - Extensions - Feature Codes - Speed Dial - Outbound Routes - Parking Lot - Time Groups - General Preferences - PBX Inbound Call Control - PBX Advanced Settings - Voice Management - System Preferences - Phone Provisioning - Reports - System Tools	FXS Extensions						
	Port	Extension Number	Display Name	Caller ID Number	RX Gain	TX Gain	Options
	1	601	601	601	40%	40%	☒ ☒
	2	602	602	602	40%	40%	☒ ☒
	3	603	603	603	40%	40%	☒ ☒
	4	604	604	604	40%	40%	☒ ☒
	5	605	605	605	40%	40%	☒ ☒

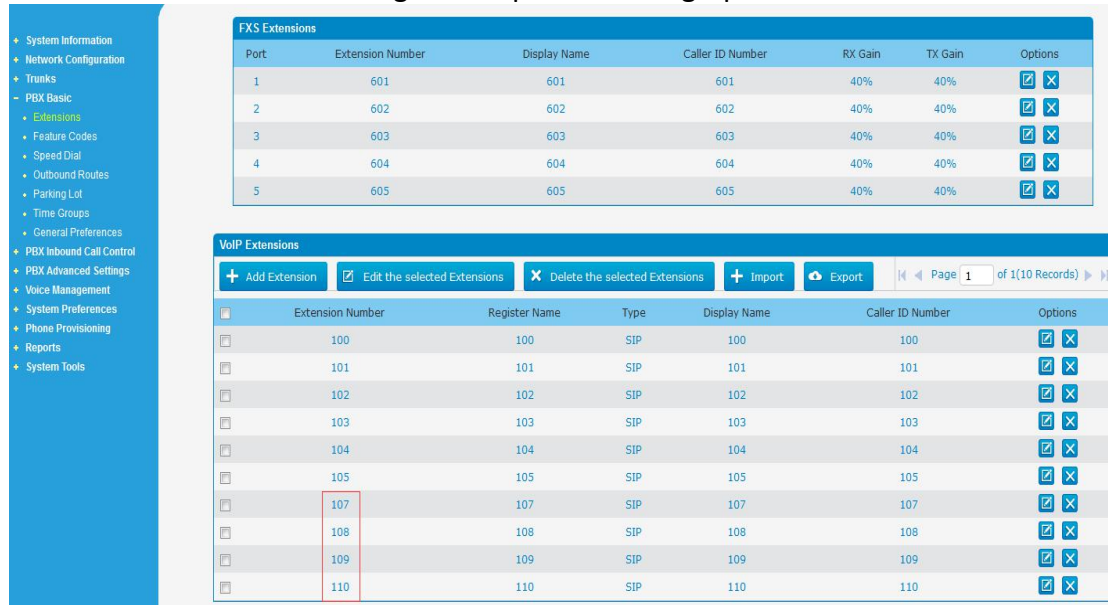
- System Information - Network Configuration - Trunks - PBX Basic - Extensions - Feature Codes - Speed Dial - Outbound Routes - Parking Lot - Time Groups - General Preferences - PBX Inbound Call Control - PBX Advanced Settings - Voice Management - System Preferences - Phone Provisioning - Reports - System Tools	VoIP Extensions						
	☒ Add Extension ☒ Edit the selected Extensions ☒ Delete the selected Extensions ☒ Import ☒ Export	Page 1 of 1(6 Records)					
	☐	Extension Number	Register Name	Type	Display Name	Caller ID Number	Options
	☐	100	100	SIP	100	100	☒ ☒
	☐	101	101	SIP	101	101	☒ ☒
	☐	102	102	SIP	102	102	☒ ☒
	☐	103	103	SIP	103	103	☒ ☒
	☐	104	104	SIP	104	104	☒ ☒
	☐	105	105	SIP	105	105	☒ ☒

Figure->CSV File Content

extensionimporttemplate.csv																													
R28C15																													
	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19										
1	Type	Number	Display	NCID	NumA1	CID	Emergency	Outbound	Register	Password	Transport	PIN	Less	Di	Ring	Time	Call	Wait	DTMF	Mode	Nat	Qualify	Can	Reinvi	Call	Group	Pickup	Gr	DND
2	SIP	107	107	107					107	FUGFF107	udp	no		0	no		rfc2833	off	yes	yes		yes	yes					no	
3	SIP	108	108	108					108	FUGFF108	udp	no		0	no		rfc2833	off	yes	yes		yes	yes					no	
4	SIP	109	109	109					109	FUGFF109	udp	no		0	no		rfc2833	off	yes	yes		yes	yes					no	
5	SIP	110	110	110					110	FUGFF110	udp	no		0	no		rfc2833	off	yes	yes		yes	yes					no	
6																													

We will provide you with a template for uploading files, you only need to modify the content of the green area

Figure->Import Success graph



Port	Extension Number	Display Name	Caller ID Number	RX Gain	TX Gain	Options
1	601	601	601	40%	40%	☒ ☒
2	602	602	602	40%	40%	☒ ☒
3	603	603	603	40%	40%	☒ ☒
4	604	604	604	40%	40%	☒ ☒
5	605	605	605	40%	40%	☒ ☒

Extension Number	Register Name	Type	Display Name	Caller ID Number	Options
100	100	SIP	100	100	☒ ☒
101	101	SIP	101	101	☒ ☒
102	102	SIP	102	102	☒ ☒
103	103	SIP	103	103	☒ ☒
104	104	SIP	104	104	☒ ☒
105	105	SIP	105	105	☒ ☒
107	107	SIP	107	107	☒ ☒
108	108	SIP	108	108	☒ ☒
109	109	SIP	109	109	☒ ☒
110	110	SIP	110	110	☒ ☒

Figure->Export Success graph

Extensions-2018Jun27233812.csv (只读)																													
R23C2																													
fx																													
	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19										
1	Type	Number	Display	NCID	NumA1	CID	Emergency	Outbound	Register	Password	Transport	PIN	Less	Di	Ring	Time	Call	Wait	DTMF	Mode	Nat	Qualify	Can	Reinvi	Call	Group	Pickup	Gr	DND
2	FXS	601	601	601								no		0	no														no
3	FXS	602	602	602								no		0	no														no
4	FXS	603	603	603								no		0	no														no
5	FXS	604	604	604								no		0	no														no
6	FXS	605	605	605								no		0	no														no
7	SIP	100	100	100					100	P7SKS100	udp	no		0	no		rfc2833	off	yes	yes		yes	yes						no
8	SIP	101	101	101					101	P7SKS101	udp	no		0	no		rfc2833	off	yes	yes		yes	yes						no
9	SIP	102	102	102					102	P7SKS102	udp	no		0	no		rfc2833	off	yes	yes		yes	yes						no
10	SIP	103	103	103					103	P7SKS103	udp	no		0	no		rfc2833	off	yes	yes		yes	yes						no
11	SIP	104	104	104					104	P7SKS104	udp	no		0	no		rfc2833	off	yes	yes		yes	yes						no
12	SIP	105	105	105					105	P7SKS105	udp	no		0	no		rfc2833	off	yes	yes		yes	yes						no
13	SIP	107	107	107					107	FUGFF107	udp	no		0	no		rfc2833	off	yes	yes		yes	yes						no
14	SIP	108	108	108					108	FUGFF108	udp	no		0	no		rfc2833	off	yes	yes		yes	yes						no
15	SIP	109	109	109					109	FUGFF109	udp	no		0	no		rfc2833	off	yes	yes		yes	yes						no
16	SIP	110	110	110					110	FUGFF110	udp	no		0	no		rfc2833	off	yes	yes		yes	yes						no

✧ Release Notes of Version 1/12/13.1.0.22

1. Introduction

- (1) Firmware Version: 1.1.0.22, 12.1.0.22, 13.1.0.22,
- (2) Applicable Model: MUC1004, MUC2008, MUC2016
- (3) Release Date: Aug 8, 2018

CHANGES SINCE FIRMWARE RELEASE 1/12/13.1.0.18

2. New Features

- (1) Added fax t38 and t38 configuration to the outbound route.
- (2) Added Myanmar time option on device Date function
- (3) Added "Morning Call" on "PBX Advanced Settings"
- (4) Added "SMS Settings" on "PBX Advanced Settings"
- (5) Added "User Permission" on "System Preferences"
- (6) Added outbound route memory trunk configuration
- (7) Added "modelGXP1610/1615-GXP1620/1625-GXP1628-GXP1630" on "Phone Provisioning->Phones->Grand stream"
- (8) Added the Txgain option of the FXO line
- (9) Added Echo Train in General and select the set echo train time value.
- (10) Added CUSTOM option (at the end of the list of options) on FXO Mode.
- (11) Added Htek UC902/UC903/UC923/UC924/UC926 model support to Phone Provisioning

3. Optimization

- (1) The time zone option values of grandstream and yealink are synchronized to the latest version of the phone.
- (2) Modify the reset time of the MUC series device 8s --> 4s

4. Bug Fixes

- (1) Fixed an error that did not hear the MOH sound when the incoming route was set to the queue.
- (2) Fixed click on the fxs extension on the extension status page to enter the edit page to return data error.
- (3) Fixed the error that the extension range value that fxs can set is different from the set extension range.
- (4) Fixed MUC1004 multi-way call, reset does not take effect.
- (5) Fixed deleting the fxs port causes the optional extension box to be empty when adding the outbound route
- (6) Fixed an error that the extension number cannot be saved when the extension fxs extension exceeded the extension range

(7) Fixed fxs extension web page transfer number not displayed error.

(8) Fixed the value of dst in cdr of reminder Call is s error.

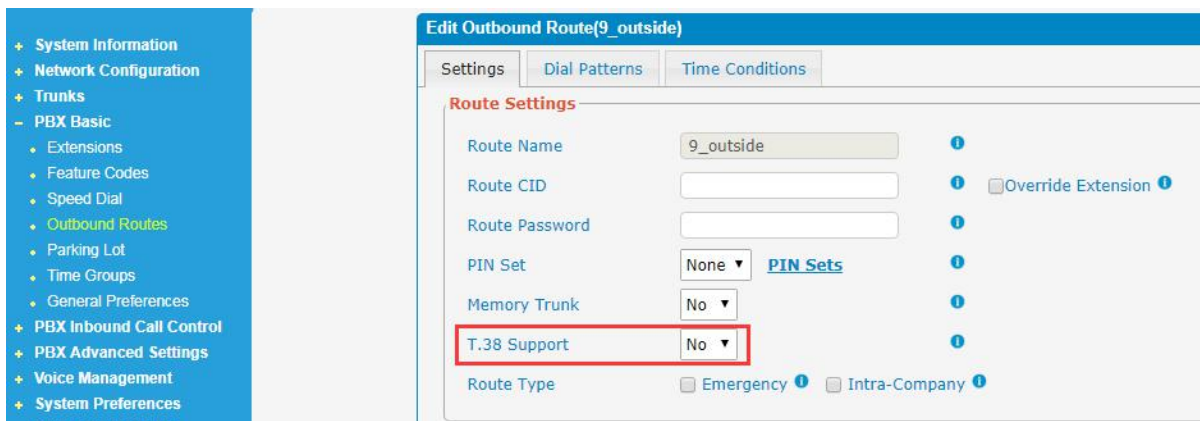
5. New Features Descriptions

(1) Added fax t38 and t38 configuration to the outbound route.

Path: PBX Basic --> Outbound Routes

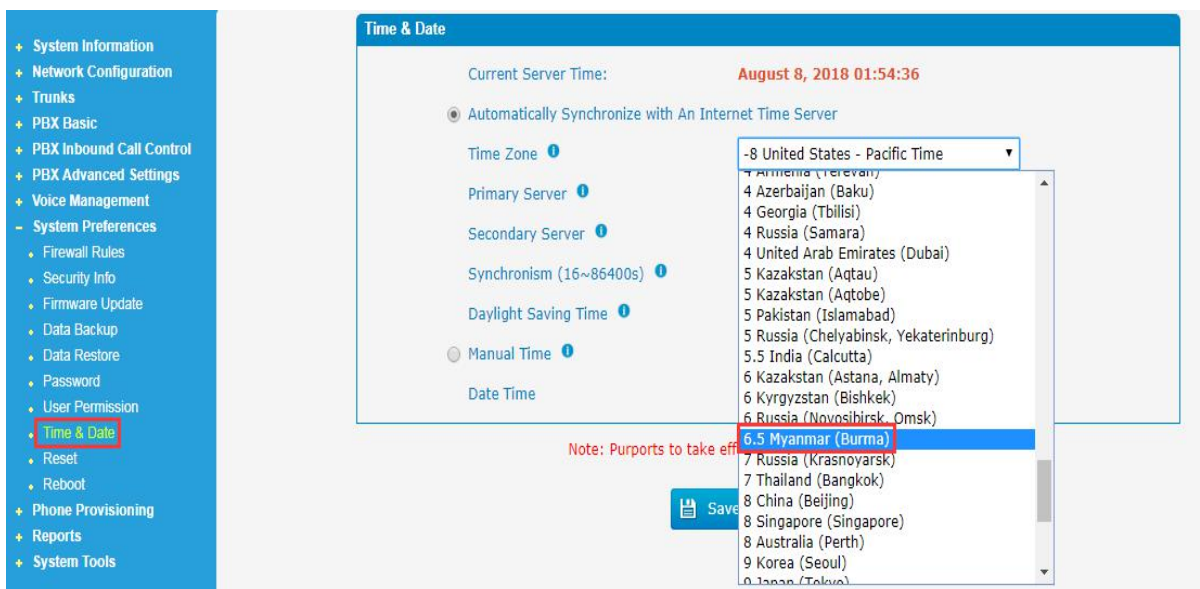
Descriptions: T38 is a protocol on how to send and receive faxes over a computer network.

Figure->t38



(2) Added Myanmar time option on device Date function

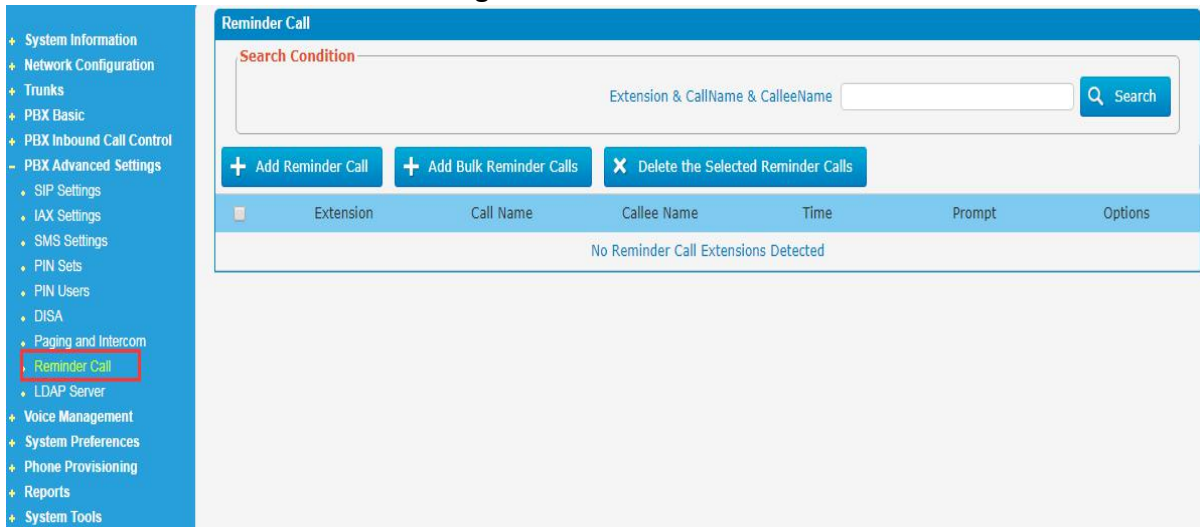
Figure->Myanmar



Path: System Preference-->Time& Date

(3) Added "Morning Call" on "PBX Advanced Settings"

Figure->Delete FXS Extension



Reminder Call

Search Condition: Extension & CallName & CalleeName

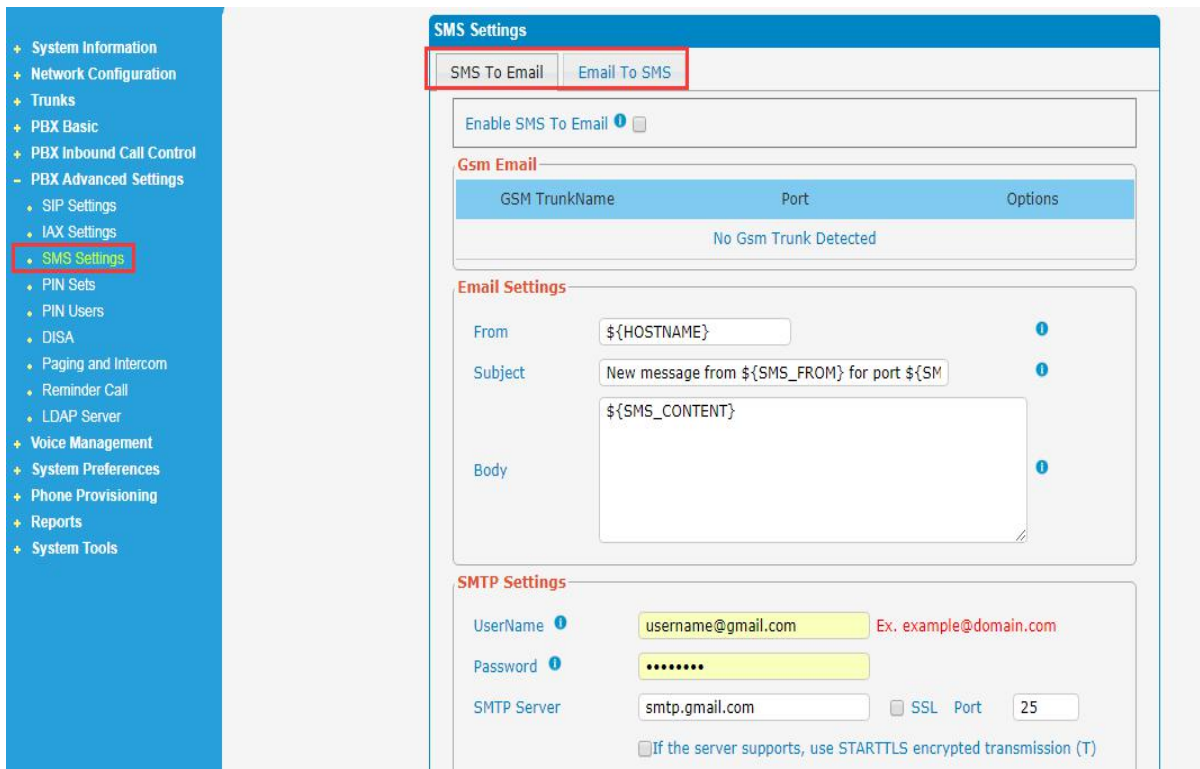
Extension	Call Name	Callee Name	Time	Prompt	Options
No Reminder Call Extensions Detected					

Path: "PBX Advanced Settings"-->"Reminder Call"

Description: Extension alarm function, you can set alarms for multiple extensions at the same time

(4) Added "SMS Settings" on "PBX Advanced Settings"

Figure->SMS Settings



SMS Settings

Enable SMS To Email ☐

Gsm Email

GSM TrunkName	Port	Options
No Gsm Trunk Detected		

Email Settings

From:

Subject:

Body:

SMTP Settings

UserName: Ex. example@domain.com

Password:

SMTP Server: ☐ SSL Port:

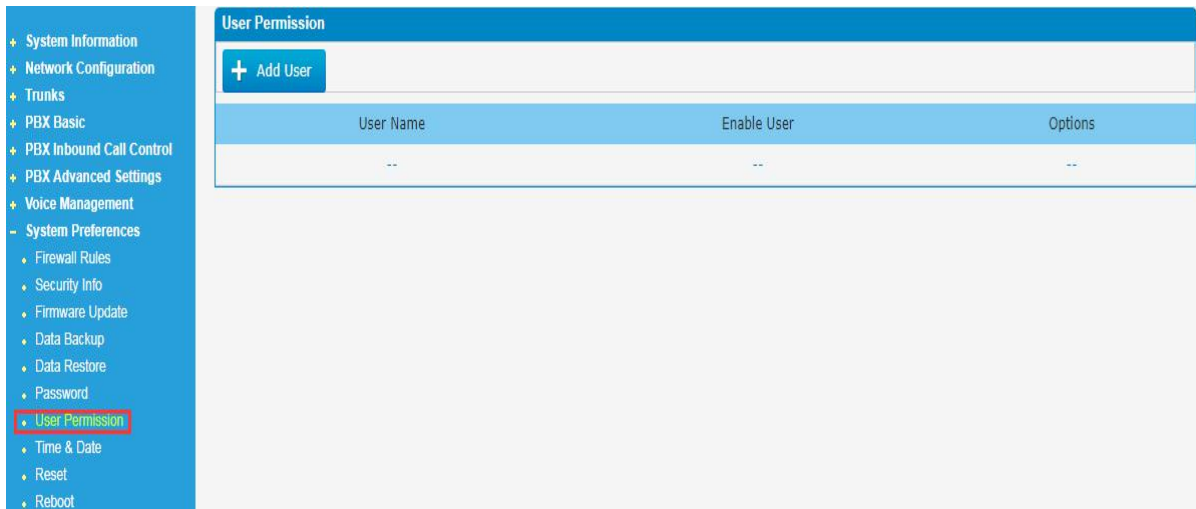
☐ If the server supports, use STARTTLS encrypted transmission (T)

Path: PBX Advanced Settings-->SMS Settings

Description: The purpose of SMS Settings is to send an email to a mailbox or send a text message to a mobile phone via email.

(5) Added "User Permission" on "System Preferences"

Figure->User Permission

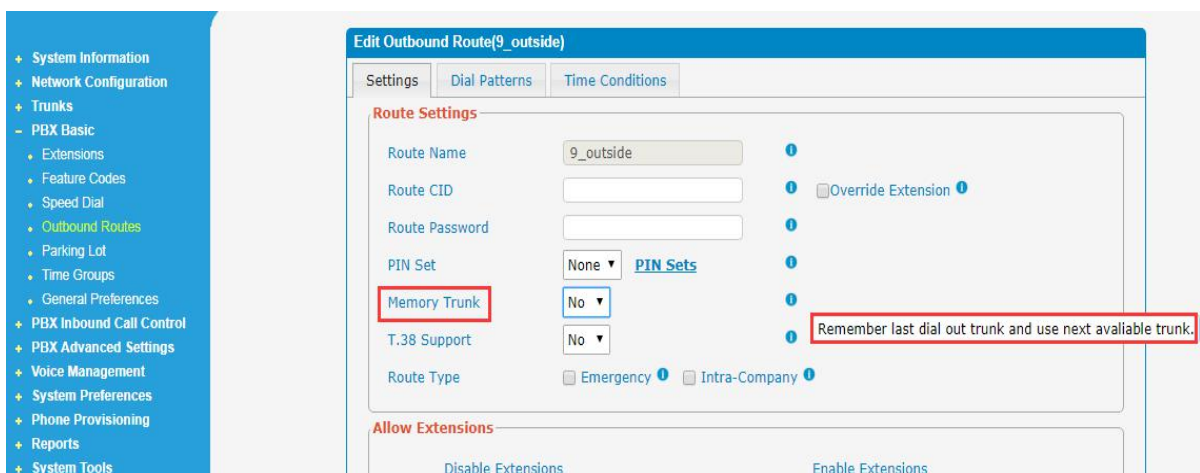


Path: System Preference-->User Permission

Description: Create new users and let users control what the page displays

(6) Added outbound route memory trunk configuration

Figure->Memory Trunk

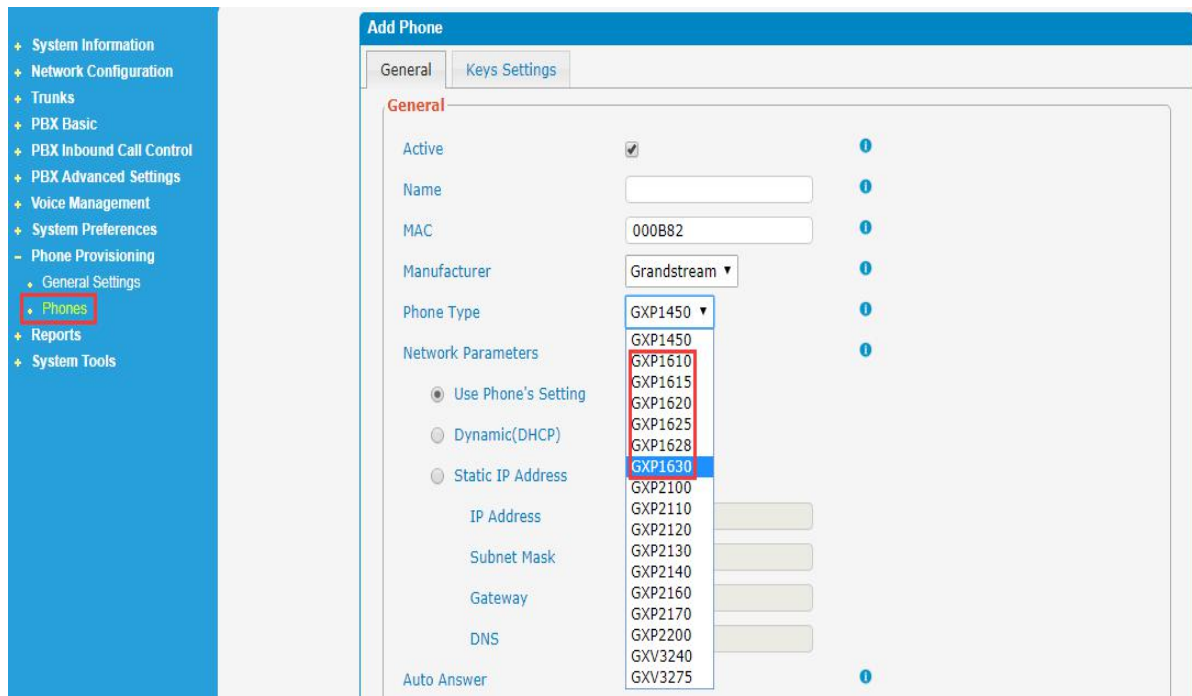


Path: PBX Basic->Outbound Routes

Descriptions: Remember last dial out trunk and use next available trunk.

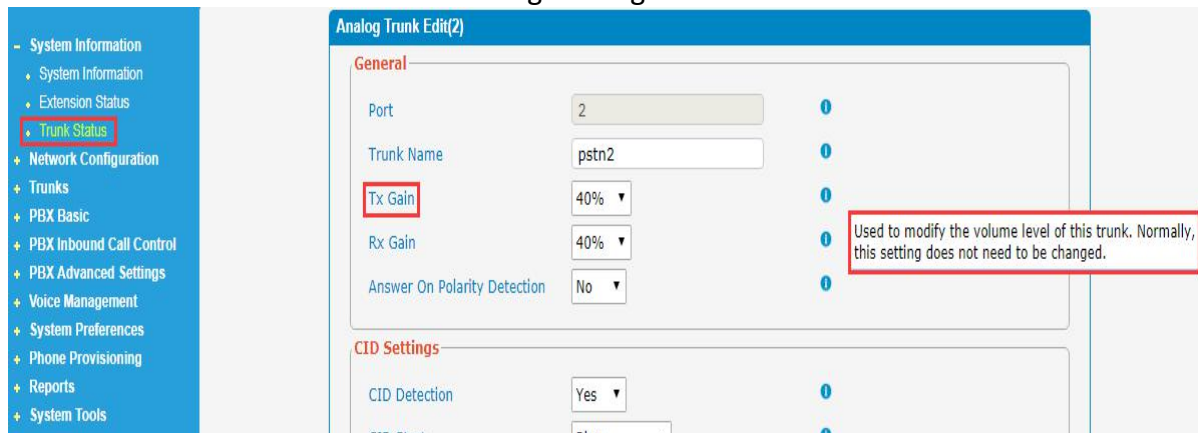
(7) Added "model GXP1610/1615-GXP1620/1625-GXP1628-GXP1630" on "Phone Provisioning->Phones->Grandstream"

Figure->Grandstream



(8) Added the Txgain option of the FXO line

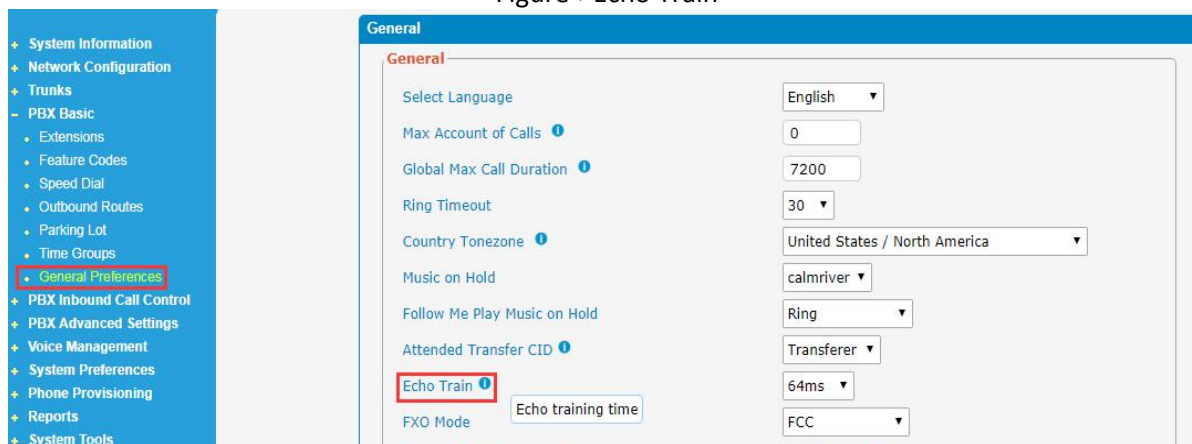
Figure->Against



Description: Used to modify the volume level of this trunk. Normally, this setting does not need to be changed.

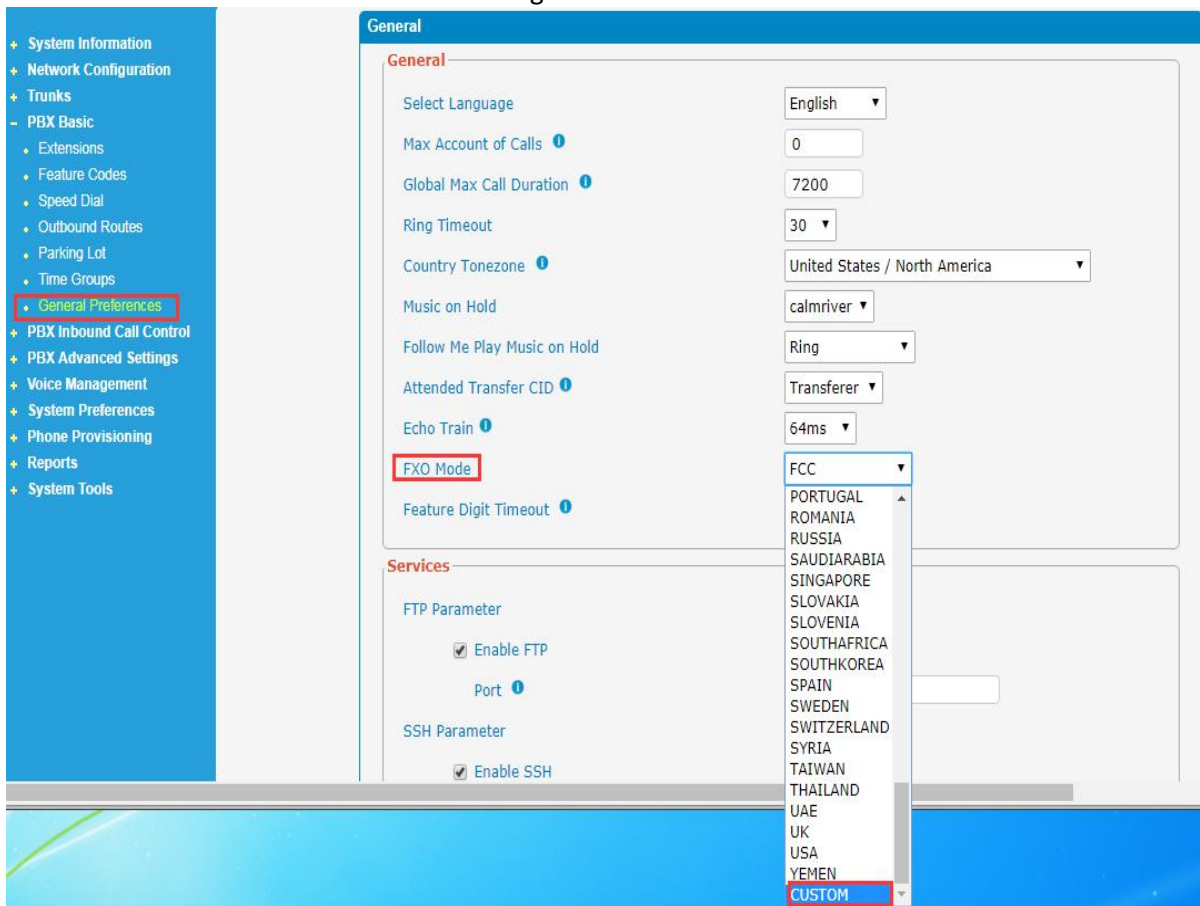
(9) Added Echo Train in General and select the set echo train time value.

Figure->Echo Train

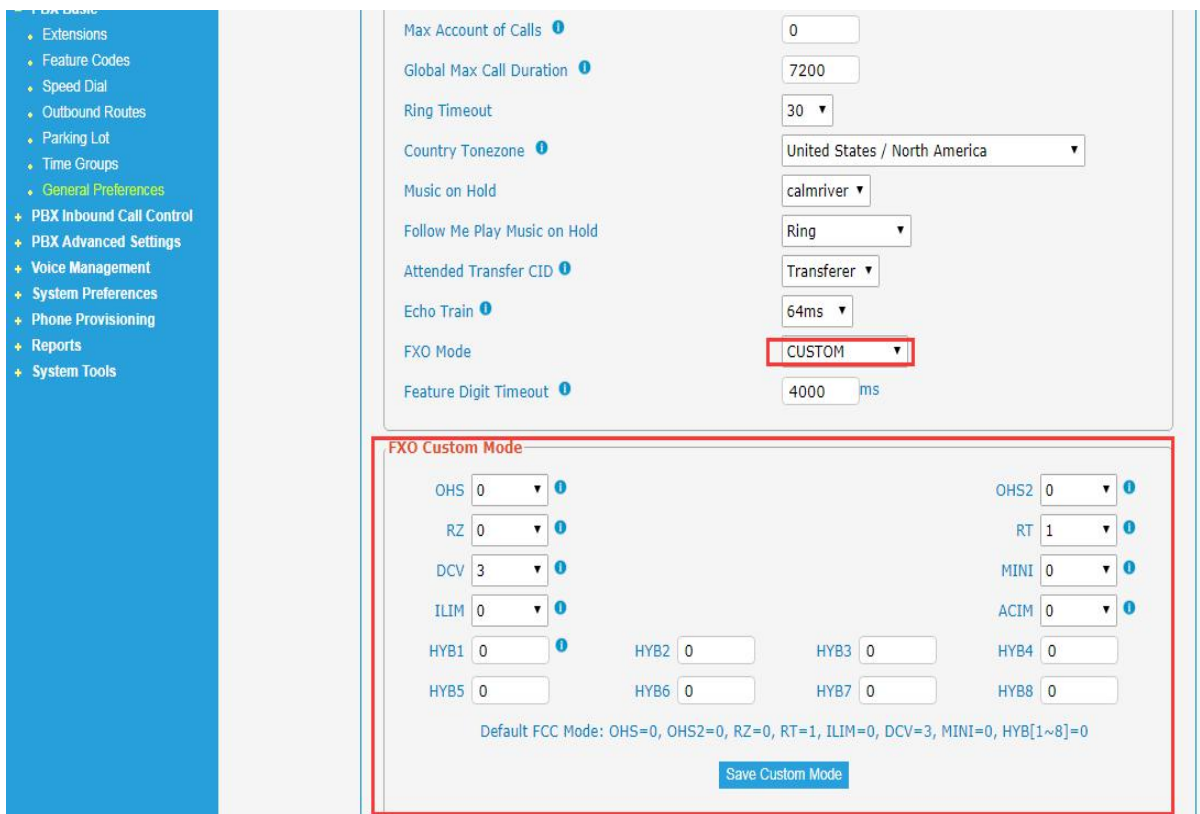


(10) Added CUSTOM option (at the end of the list of options) on FXO Mode.

Figure->Custom



The screenshot shows the 'General' configuration page. On the left sidebar, 'General Preferences' is highlighted. In the main panel, the 'FXO Mode' dropdown menu is open, showing a list of countries including PORTUGAL, ROMANIA, RUSSIA, SAUDIARABIA, SINGAPORE, SLOVAKIA, SLOVENIA, SOUTHAFRICA, SOUTHKOREA, SPAIN, SWEDEN, SWITZERLAND, SYRIA, TAIWAN, THAILAND, UAE, UK, USA, YEMEN, and 'CUSTOM' at the bottom. The 'CUSTOM' option is highlighted with a red box.

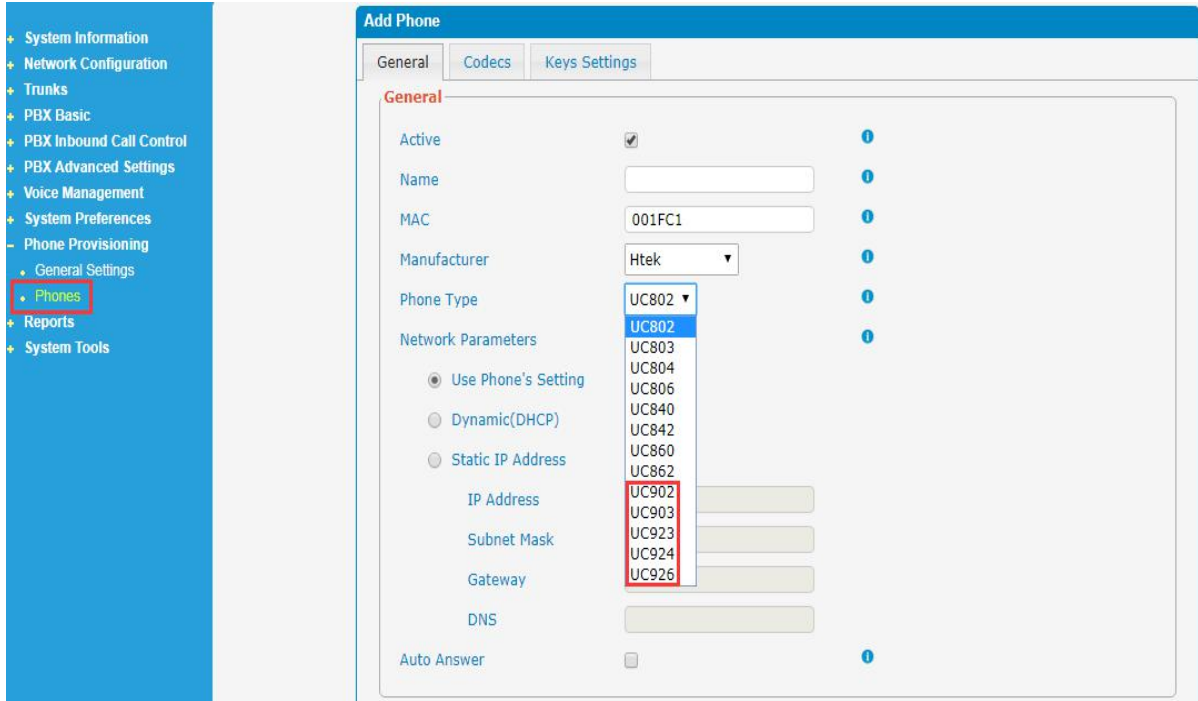


The screenshot shows the 'FXO Custom Mode' configuration page. The 'FXO Mode' dropdown is set to 'CUSTOM'. Below it, the 'FXO Custom Mode' section is expanded, showing various parameters: OHS (0), OHS2 (0), RZ (0), RT (1), DCV (3), MINI (0), ILIM (0), ACIM (0), HYB1 (0), HYB2 (0), HYB3 (0), HYB4 (0), HYB5 (0), HYB6 (0), HYB7 (0), and HYB8 (0). A 'Save Custom Mode' button is at the bottom. The default FCC Mode is listed as: Default FCC Mode: OHS=0, OHS2=0, RZ=0, RT=1, ILIM=0, DCV=3, MINI=0, HYB[1~8]=0.

(11) Added Htek UC902/UC903/UC923/UC924/UC926 model support to Phone Provisioning

Path: Phone Provisioning --> Phones --> Add Phone

Figure->Htek



Add Phone

General | Codecs | Keys Settings

General

Active ☒

Name

MAC

Manufacturer

Phone Type

Network Parameters

☒ Use Phone's Setting

☐ Dynamic(DHCP)

☐ Static IP Address

IP Address

Subnet Mask

Gateway

DNS

Auto Answer ☐

Phone Type dropdown list:

- UC802
- UC803
- UC804
- UC806
- UC840
- UC842
- UC860
- UC862
- UC902
- UC903
- UC923
- UC924
- UC926

✧ Release Notes of Version 1/12/13.1.0.18

1. Introduction

- (1) Firmware Version: 1.1.0.18, 12.1.0.18, 13.1.0.18,
- (2) Applicable Model: MUC1004, MUC2008, MUC2016
- (3) Release Date: Nov 16, 2016

CHANGES SINCE FIRMWARE RELEASE 1/12/13.1.0.14

2. New Features

- (1) Added "FXS Del Option" on "FXS Extensions"
- (2) Added "Rtp Port End" on "SIP Settings"
- (3) Added Select All and Cancel All widget on Selection box
- (4) Added "Report IP Address" on "Feature Codes"
- (5) Added pt_BR System Prompts on "System Prompts Settings"
- (6) Added "Send Privacy id" on "SIP Settings"
- (7) Added "Send Diversion" on "SIP Settings"

3. Optimization

- (1) Optimized the WAN port to the priority port
- (2) Increase the length of the SIP Trunk field input

4. Bug Fixes

- (1) Fixed the issue that SIP Trunk DOD not work properly For "From User" field error

5. New Features Descriptions

(1) Added "FXS Del Option" on "FXS Extensions"

Path: PBX Basic --> Extensions --> FXS Extensions

Descriptions: Click delete FXS extension, Web Page will retain the edit option, enter the edit page, you can modify FXS extension number

Figure->FXS Status

FXS Extensions						
Port	Extension Number	Display Name	Caller ID Number	RX Gain	TX Gain	Detail
1	601	601	601	40%	40%	 
2	602	602	602	40%	40%	 

Figure->Delete FXS Extension

FXS Extensions						
Port	Extension Number	Display Name	Caller ID Number	RX Gain	TX Gain	Options
1	--	--	--	--	--	 
2	602	602	602	40%	40%	 

Figure->Edit Deleted FXS Extension

Edit FXS Extension(602)

General

Voicemail

Options

Other

User Information

Extension Type

FXS

Port

2

Extension Number

602

Display Name

602

Caller ID Number

602

Outbound CID

Emergency CID

Note: SMTP Parameter must be configured correctly before Voicemail to E-mail will work.

Save

Back

Figure->Extension Scope

Extension Parameters

Extension Number

100

-

616

New Fxs extension number is in extension scope

In Fxs deleted edit web page, modify extension number will adjust following field at the same time

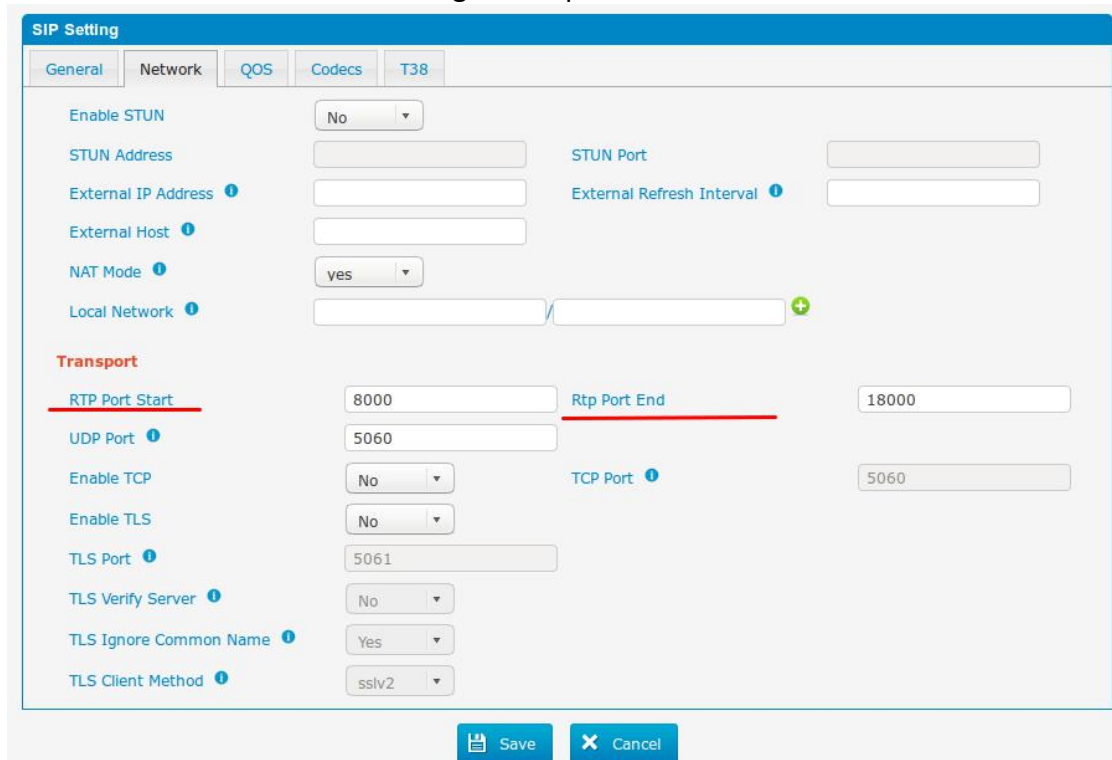
- Display Name
- Caller ID Number
- PIN Number
- Extension web Login Name and Password

(2) Added "Rtp Port End" on "SIP Settings"

Path: PBX Advanced Settings --> SIP Settings

Descriptions: Easy to fill out and view the value of rtp port range.

Figure->Rtp Port End



SIP Setting

General Network QOS Codecs T38

Enable STUN: No

STUN Address: [text box] STUN Port: [text box]

External IP Address: [text box] External Refresh Interval: [text box]

External Host: [text box]

NAT Mode: yes

Local Network: [text box] [text box] +

Transport

RTP Port Start: 8000 Rtp Port End: 18000

UDP Port: 5060

Enable TCP: No

Enable TLS: No

TLS Port: 5061

TLS Verify Server: No

TLS Ignore Common Name: Yes

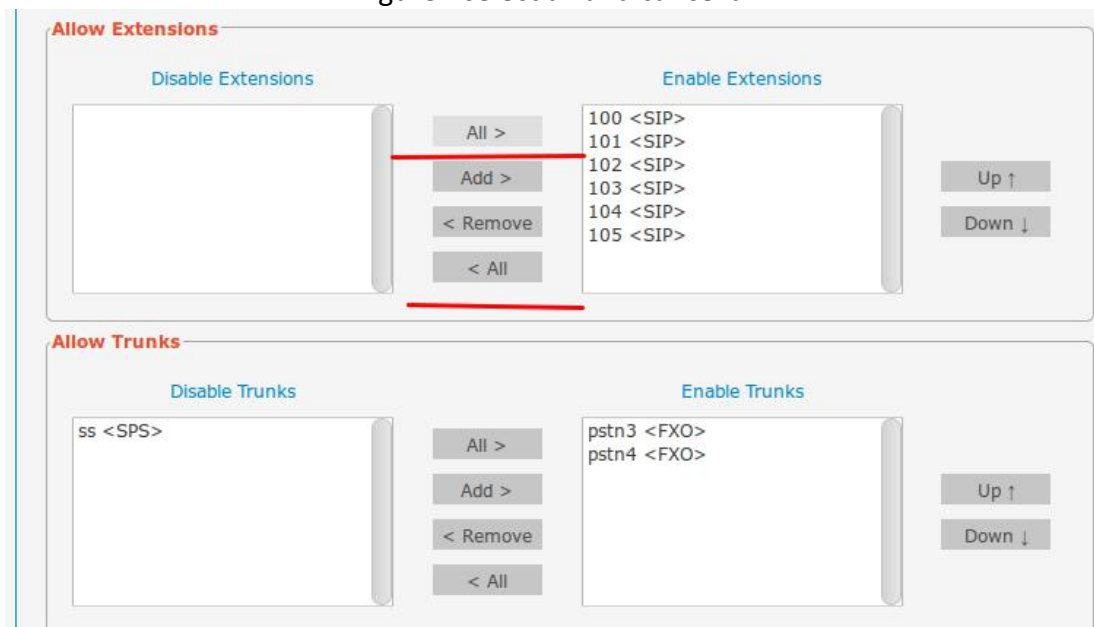
TLS Client Method: sslv2

Save Cancel

(3) Added Select All and Cancel All widget on Selection box

For example in outbound edit page

Figure->select all and cancel all



Allow Extensions

Disable Extensions: [text box]

Enable Extensions: [text box]

Buttons: All >, Add >, < Remove, < All, Up ↑, Down ↓

Allow Trunks

Disable Trunks: [text box]

Enable Trunks: [text box]

Buttons: All >, Add >, < Remove, < All, Up ↑, Down ↓

(4) Added “Report IP Address” on “Feature Codes”

Path: PBX Basic --> Feature Codes --> Report IP Address

Descriptions: Dial *** , System will play prompt of local IP.

Figure->Report IP Address



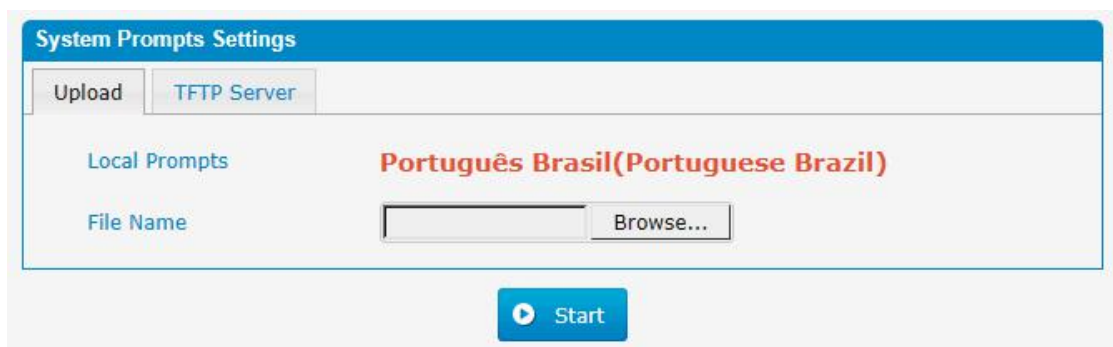
Feature Codes	Use Default?	Feature Status
General		
Call Pickup	*8	☒ Enable
Call Trace	*69	☒ Enable
Directed Call Pickup	*08	☒ Enable
Attended Transfer	*2	☒ Enable
Blind Transfer	##	☒ Enable
One Touch Record	*1	☒ Enable
Report IP Address	***	☒ Enable

(5) Added pt_BR System Prompts on “System Prompts Settings”

Path: Voice Management --> System Prompts Settings

Descriptions: Add a new System Prompts of Portuguese Brazil

Figure->System prompts of Portuguese Brazil



System Prompts Settings

Upload TFTP Server

Local Prompts **Português Brasil (Portuguese Brazil)**

File Name Browse...

Start

(6) Added “Send Privacy id” on “SIP Settings”

(7) Added “Send Diversion” on “SIP Settings”

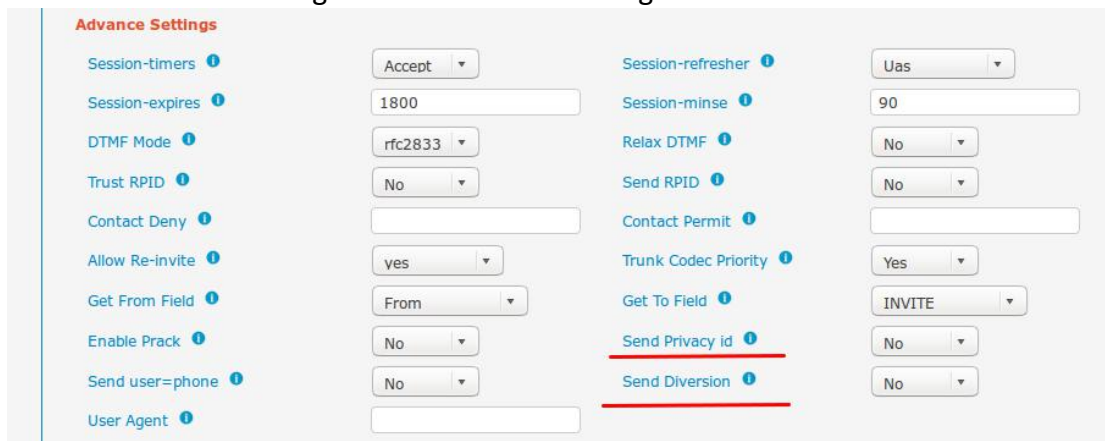
Path: PBX Advanced Settings --> SIP Settings

Descriptions:

send Privacy header to enable anonymous call feature

send Diversion header to enable caller id Pass-Through feature when forward through SIP trunk

Figure->SIP Advance Settings



The screenshot shows the 'Advance Settings' page for SIP configuration. It is divided into two columns of settings, each with a list of options on the left and input fields on the right. The left column includes settings like Session-timers, Session-expires, DTMF Mode, Trust RPID, Contact Deny, Allow Re-invite, Get From Field, Enable Prack, Send user=phone, and User Agent. The right column includes Session-refresher, Session-minse, Relax DTMF, Send RPID, Contact Permit, Trunk Codec Priority, Get To Field, Send Privacy id, and Send Diversion. Some settings have dropdown menus, while others have text input fields. The 'Send Privacy id' and 'Send Diversion' settings are highlighted with red lines.

6. Optimization Descriptions

(1) Optimized the WAN port to the priority port

Path: PBX Basic --> Network Configuration --> Static Route

Descriptions: By default, the WAN port routing priority is higher than the LAN port

Figure->WAN and LAN static route

Static Route				
Routing Table		Static Routing Rules		
Destination IP Address	Subnet Mask	Gateway	Metric ⓘ	Interface
0.0.0.0	0.0.0.0	192.168.6.1	0	WAN
192.168.5.0	255.255.255.0	0.0.0.0	0	LAN
192.168.6.0	255.255.255.0	0.0.0.0	0	WAN

If you have configure WAN port before, you may need to reset pbx

✧ Release Notes of Version 1/12/13.1.0.14

1. Introduction

- (1) Firmware Version: 1.1.0.14, 12.1.0.14, 13.1.0.14
- (2) Applicable Model: MUC1004, MUC2008, MUC2016
- (3) Release Date: July 04, 2016

CHANGES SINCE FIRMWARE RELEASE 1/12/13.1.0.5

2. New Features

- (1) Added "Security Info" on "System Preferences"
- (2) Added "AMI Settings" on "System Tools"
- (3) Added "Mobility Extension" on Extensions
- (4) Added extension Maximum Call Duration and global Maximum Call Duration
- (5) Added dod and add bulk dod on "IP Trunk" and "VOIP Trunk"
- (6) Added Extension CDR Report on Extension Web Login

3. Optimization

- (1) None

4. Bug Fixes

- (1) None

5. New Features Descriptions

- (1) Added "Security Info" on "System Preferences"

Path: System Preferences->Security Info

Alert Settings: if the device is attacked, the system will notify users the alert via call or E-mail. the attack modes include IP attack and Web Login.

IP Blacklist: if the device is attacked by IP attack. system will add IP to firewall and Disable this IP access.

Alert Settings			
Attack Type	Phone Notification	Email Notification	Option
IPATTACK	yes	yes	☒
WEBLOGIN	no	yes	☒

IP Blacklist					
<input checked="" type="button" value="Delete the selected IP Blacklist"/>					
Page 1 of 1(1 Records)					
Attacked Time	Protocol	Attacked Port	Source IP Address	MAC	Option
2016-07-03 17:57:39	udp	5060	192.168.6.33	08:00:27:90:FD:ED	☒

Edit Alert Settings(IPATTACK)

Phone Notification Settings

Phone Notification: NO
Number:
Attempts: 1
Interval: 60
Prompt: default

E-mail Notification Settings

E-mail Notification: NO
To:
Subject:

Pbx Host Name: \$(HOSTNAME)
Attack Time: \$(DATETIME)
Attack Src IP: \$(SOURCEIP)
Attack Des MAC: \$(DESTMIC)
Attack Src Port: \$(DESTPORT)
Attack Protocol: \$(PROTOCOL)

(2) Added "AMI Settings" on "System Tools"

Path: System Tools --> AMI Settings

Descriptions: The Asterisk Manager Interface(AMI) is a socket interface that you can use to get configuration and status information, request actions to be performed, and be notified about things happening to calls.

AMI Settings

Enabled AMI

☐

User Name

admin

Password

password

IP Restriction

Permit 'IP address/Subnet mask'

+

save

back

(3) Added “Mobility Extension” on Extensions

Path: Edit Extension

Descriptions:

- 1.If you set a mobile number as mobility extension, while you call in PBX with this mobile number, the mobile phone will get all permission of the associated extension. For example: dialing the extension, playing the voicemail.
- 2.When someone calls the associated extension, your mobile phone will ring together, what you need is to set outbound route and Outbound Prefix number.

Edit VoIP Extension(103)

General

Voicemail

Options

Other

Call Forward

☐ Always

☒ Voicemail

☒ On Unavailable

Send Call to:

☐ Number

☒ When Busy

☐ Hang Up

Mobility Extension

☒ Enable MobileExten

Mobile Num

15812365478

☒ Enable RingAll

Outbound Prefix

9

Options

Maximum Call Duration

12

Ring Time

Default

Call Waiting

Disable

Pinless Dialing

Disable

Allow Re-invite

yes

Call Group

Pickup Group

Do Not Disturb

☐

Note: SMTP Parameter must be configured correctly before Voicemail to E-mail will work.

Save

Back

(4) Added extension Maximum Call Duration and global Maximum Call Duration

Path: Edit Extension and PBX Basic --> General Preferences

Descriptions: The absolute maximum amount of time permitted for a call

Edit VoIP Extension(103)

General
Voicemail
Options
Other

Call Forward

☐ Always
☒ Voicemail

☒ On Unavailable
☐ Number

☒ When Busy
☐ Hang Up

Send Call to:

Mobility Extension

☒ Enable MobileExten
Mobile Num
15812365478

☒ Enable RingAll
Outbound Prefix
9

Options

Maximum Call Duration	12	
Ring Time	Default	
Call Waiting	Disable	
Pinless Dialing	Disable	
Allow Re-invite	yes	
Call Group		
Pickup Group		
Do Not Disturb	☐	

Note: **SMTP Parameter** must be configured correctly before Voicemail to E-mail will work.

Save
Back

General

General

Select Language
English

Max Account of Calls
0

Global Max Call Duration
7200

Ring Timeout
30

Country Tonezone
United States / North America

Music on Hold
calmriver

Follow Me Play Music on Hold
Ring

FXO Mode
FCC

Feature Digit Timeout
4000 ms

(5) Added dod and add bulk dod on "IP Trunk" and "VoIP Trunk"

Path: Trunks --> IP Trunk and Trunks --> VoIP Trunk

Descriptions: Add dod number and add bulk dod numbers to associated extension.

IP Trunk Edit(xx)

Trunk Name
xx
Type
SIP
Outbound Caller ID
Maximum Channels
Hostname/IP
192.168.6.32
Port
5060
Transport
UDP
DTMF Mode
rfc2833
Qualify
Yes
Allowed Audio Codecs
ulaw,alaw,gsm

DOD Settings

DOD	Associated Extension	Option
1243	100	×
1244	101	×
1245	102	×
1246	103	×
1247	104	×
1248	105	×

DOD
Associated Extension
100
+ Add DOD
+ Add Bulk DOD

Save

Back

Add Bulk DOD

Add Bulk DOD

Extensions List

100 <SIP>
101 <SIP>
102 <SIP>
103 <SIP>
104 <SIP>
105 <SIP>
106 <SIP>
107 <SIP>
108 <SIP>
109 <SIP>
110 <SIP>
111 <SIP>
112 <IAX>
603 <FXS>
604 <FXS>

Add >
< Remove

Selected Extensions

Up ↑
Down ↓

Begin

Save

Cancel

(6) Added Extension CDR Report on Extension WebLogin

Path: Extension Web Login

Descriptions: Extension cdr lo

CDR Report									
Show Filter		Download the records		Page 1 of 1 (10 Records)					
Date	Source	Destination	Src. Trunk	Account Code	Dst. Trunk	Call Direction	Status	Duration	Billing Duration
2016-06-30 23:45:29	103	603				Internal	ANSWERED	10s	9s
2016-06-30 23:36:55	603	9104			pstn2	Outbound	ANSWERED	1s	1s
2016-06-30 23:36:53	603	103				Internal	ANSWERED	9s	6s
2016-06-30 23:36:43	603	910			pstn2	Outbound	NO ANSWER	9s	0s
2016-06-30 23:35:48	603	9104			pstn2	Outbound	ANSWERED	2s	1s
2016-06-30 23:35:46	603	103				Internal	ANSWERED	28s	24s
2016-06-30 22:52:19	603	9104			pstn1	Outbound	ANSWERED	15s	7s
2016-06-30 22:50:55	603	104				Internal	ANSWERED	6s	4s
2016-06-30 22:50:17	603	9101			pstn2	Outbound	ANSWERED	31s	22s
2016-06-30 22:46:12	603	9101			pstn2	Outbound	ANSWERED	17s	8s

6.Optimization Descriptions

(1) none

✧ Release Notes of Version 1/12/13.1.0.5

1.Introduction

- (1) Firmware Version: 1.1.0.5, 12.1.0.5, 13.1.0.5
- (2) Applicable Model: MUC1004, MUC2008, MUC2016
- (3) Release Date: Dec 03, 2015

CHANGES SINCE FIRMWARE RELEASE 1/12/13.1.0.2

2.New Features

- (1) Added "System Prompt Settings" on "Voice Management".
- (2) Added support for phone provisioning of Akuvox phone.
- (3) Added the feature of "key as send" for Grandstream phones.

3.Optimization

- (1) Optimized the feature of update password, allow special characters.
- (2) Optimized the feature of special characters are allowed in "Alert-Info".

4.Bug Fixes

- (1) Fixed the issue that the Office Hours and Holiday cannot work properly.
- (2) Fixed the issue that the extension page display abnormal when extension account login first then login admin account.
- (3) Fixed the issue that the changes for "Email Attachment", "Play CID", "Play Envelope", "Delete Voicemail" settings cannot take effect (PBX Basic->Extensions. Edit).
- (4) Fixed the issue that the "Global Office Hours" (PBX Basic->General Preferences) setting cannot be saved.
- (5) Fixed the issue that the "Inbound Routes. Alert-Info" and "Queues. Alert-Info" and "Ring Groups. Alert-Info" cannot work properly.
- (6) Fixed the issue that the "Call Forward" Always send call to Voicemail setting cannot work.
- (7) Fixed the issue that the extension page update password cannot work.
- (8) Fixed the display that the GUI of Google Chromium browser press F5 to refresh, then the web page was crash.
- (9) Fixed the issue that the P2P mode of IAX trunk cannot work.
- (10) Fixed the Phone Provisioning cannot manually add the configuration of existing phone.

- (11) Fixed the Phone Provisioning cannot work properly when the PBX changes for IP or SIP port.
- (12) Fixed the Phone Provisioning cannot work properly when set the codecs empty.
- (13) Fixed the Phone Provisioning changes for "Active" setting cannot take effect.
- (14) Fixed the Grandstream phones set Key Settings can't work normally.
- (15) Fixed the Fanvil phones set time zone cannot take effect.
- (16) Fixed the Fanvil phones set Key Settings cannot take effect.
- (17) Fixed the Yealink phones set codes cannot take effect.
- (18) Fixed the Yealink phones set line of Key Settings cannot take effect.
- (19) Fixed the Htek phones set codecs empty cannot be saved successfully.
- (20) Fixed the Htek phones Line setting cannot take effect.
- (21) Fixed the Htek phones configure Key settings cannot take effect.
- (22) Fixed the Aastra phones Top Keys and Line settings cannot take effect.
- (23) Fixed the Aastra phones set IP for DHCP mode cannot take effect.
- (24) Fixed the Snom phones extension Line settings cannot take effect.
- (25) Fixed the Cisco phones set time zone part cannot take effect.
- (26) Fixed the Cisco phones set part of the spa model cannot take effect.
- (27) Fixed the Cisco phones set Key Settings cannot take effect.

5.New Features Descriptions

(1) Added upload "System Prompt Settings" on "Voice Management".

Path: Voice Management->System Prompts Settings

Descriptions:

With this feature, you can upload the system prompts.

System Prompts Settings

Upload

TFTP Server

Local Prompts

English

File Name

Browse...

Start

(2) Added support for phone provisioning of Akuvox phone.

Path: Phone Provisioning->Phones

Descriptions:

We can configuration Akuvox phones in Phone Provisioning.

Add Phone

General

Codecs

General

Active

☒

Name

MAC

001565

Manufacturer

Yealink

Phone Type

Aastra

Akuvox

Cisco

Fanvil

Grandstream

Htek

Panasonic

Polycom

Snom

Yealink

Network Parameters

Use Phone's Setting

Dynamic(DHCP)

Static IP Address

IP Address

Subnet Mask

Gateway

DNS

Auto Answer

☐

Call Waiting

☒

Key As Send

#

Account

Line

Line1

Extension

105

Label

Display Name

Active

☐

Configured Phones							
+ Add Phone		X Delete the selected Phones		Page 1 of 1(2 Records)			
	MAC	IP	Phone Model	Extension	Active	Name	Options
☐	001565565656	--	Yealink T19P	102	ON	--	☐ ☐
☐	0C1105026B54	192.168.6.61	Akuvox SP-R59 59.0.3.41	104,106	ON	--	☐ ☐

Not Configured Phones			
Refresh		Page 1 of 1(5 Records)	
MAC	IP	Phone Model	Options
000B826C7D0E	192.168.6.219	Grandstream GXP1450 1.0.6.11	☐
00A859D2919E	192.168.6.54	Fanvil C58 2.3.463.258	☐
00A859D34816	192.168.6.53	Fanvil F52 2.3.381.220	☐
0C110500388C	192.168.6.51	Akuvox SP-R53 53.0.3.41	☐
0C1105028FD8	192.168.6.71	Akuvox SP-R50 50.0.3.41	☐

(3) Added the feature of "key as send" for Grandstream phones.

Path: Phone Provisioning->Phones .Edit on the Grandstream phones and set "key as send".

Edit Phone

General

Keys Settings

General

Active

☒

Name

MAC

000B826C7D0E

Manufacturer

Grandstream

Phone Type

GXP1450

Network Parameters

Use Phone's Setting

Dynamic(DHCP)

Static IP Address

IP Address

Subnet Mask

Gateway

DNS

Auto Answer

☐

Call Waiting

☒

Key As Send

#

6.Optimization Descriptions

(1) Optimized the feature of update password, allow special characters.

Path: System Preferences->Password

Password Setting

Old Username

admin

Old Password

••••••

New Password

••••••••

Confirm Password

••••••••

Weak

(2) Optimized the feature of special characters are allowed in “Alert-Info”.

Path: PBX Inbound Call Control->Queues , Ring Groups

Edit Queue(820)

General
Options
Advanced Settings

General

Queue Number: 820 ⓘ
Queue Name: Queue820 ⓘ
Queue Password: ⓘ
Max Time Caller in Queue: Unlimited ⓘ
Agent Timeout: 30seconds ⓘ
CID Name Prefix: Queue820- ⓘ
Alert Info: <http://192.168.6.96: ⓘ
Ring Strategy: ringall ⓘ
Restrict Dynamic Agents: Yes ⓘ

Static Agents

Extensions

102 <SIP>
104 <SIP>
105 <SIP>
106 <SIP>
107 <SIP>
108 <SIP>
109 <SIP>
110 <IAX>
111 <IAX>

Add >
< Remove

Allow Members

500 <SIP>
502 <SIP>
504 <IAX>
505 <SIP>
506 <IAX>
507 <IAX>
508 <IAX>
509 <SIP>
510 <SIP>

Up ↑
Down ↓

Dynamic Agents

Extensions

102 <SIP>
109 <SIP>
110 <IAX>
111 <IAX>
112 <IAX>
114 <SIP>
115 <SIP>
116 <SIP>
117 <SIP>

Add >
< Remove

Allow Members

104 <SIP>
105 <SIP>
106 <SIP>
107 <SIP>
108 <SIP>
501 <SIP>
503 <IAX>

Up ↑
Down ↓

Edit Ring Group(920)

General

General

RG Number: 920 ⓘ
RG Name: RingGroup920 ⓘ
Ring Strategy: Ring all Selection ⓘ
Ring Time: 45 ⓘ
Music on Hold: calmriver ⓘ
Ring Instead Of Moh: ⓘ
CID Name Prefix: RingGroup920- ⓘ
Alert Info: <http://192.168.6.96>\ ⓘ

Ring Group Members

Extensions

102 <SIP>
104 <SIP>
105 <SIP>
106 <SIP>
107 <SIP>
108 <SIP>
109 <SIP>
110 <IAX>
111 <IAX>

Add >
< Remove

Members

503 <IAX>

Up ↑
Down ↓

Destination If No Answer

Destination: End Call ⓘ

✧ Release Notes of Version 1/12/13.1.0.2

1. Introduction

- (1) Firmware Version: 1.1.0.2, 12.1.0.2, 13.1.0.2
- (2) Applicable Model: MUC1004, MUC2008, MUC2016
- (3) Release Date: Nov 09, 2015

CHANGES SINCE FIRMWARE RELEASE 1/12/13.1.0.1

2. New Features

- (1) None

3. Optimization

- (2) Optimized when adding extension and changing the extension number, automatically revision Display Name, Call ID number, etc.

4. Bug Fixes

- (1) Fixed the issue that the Inbound Route to Outbound Route cannot work.
- (2) Fixed the issue that set the two Outbound Routes of the same Dial Patterns and dial fail that system will appear problems.
- (3) Fixed the issue that Record New Prompt cannot work (Voice Management - > System Recording).
- (4) Fixed the issue that the register multiple VOIP account to the same SIP server, inbound route always match to the first account.
- (5) Fixed the issue that Call Pickup, Attended Transfer, Blind Transfer, One Touch Record, Call Parking for disabled cannot take effect.
- (6) Fixed the issue that the IAX2 extension Call Pickup cannot work.
- (7) Fixed the issue that the SIP trunk P2P mode cannot work.
- (8) Fixed the issue that VOIP Trunk continue to send registration when deleted the VOIP Trunk.
- (9) Fixed the issue that changes for Tos video and Cos video setting cannot take effect (PBX Advanced Settings -> SIP Settings -> QOS)
- (10) Fixed the issue that changes for Feature Digit Timeout setting cannot take effect (PBX Basic -> General IP references)

5. New Features Descriptions

- (1) None

6. Optimization Descriptions

- (1) Optimized when adding extension and changing the extension number,

automatically revision Display Name, Call ID number ,etc.

Path: PBX Basic->Extension

Add VoIP Extension

General

Voicemail

Options

Other

User Information

Extension Type

SIP

Extension Number

200

Range

1

Display Name

200

Caller ID Number

200

Outbound CID

Emergency CID

Authentication

Register Name

200

Password

.....

Medium

VoIP Setting

Transport

UDP

RTP Encryption(SRTP)

No

DTMF Mode

RFC2833

Qualify

Yes

NAT

✧ Release Notes of Version 1/12/13.1.0.1

1. Introduction

- (1) Firmware Version: 1.1.0.1, 12.1.0.1, 13.1.0.1
- (2) Applicable Model: MUC1004, MUC2008, MUC2016
- (3) Release Date: Nov 02, 2015

CHANGES SINCE FIRMWARE RELEASE 1/12/13.1.0.0

2. New Features

- (1) Added SIP trunk and early media focus.

3. Optimization

- (1) None

4. Bug Fixes

- (1) Fixed the issue that T.38 fax for SIP trunk cannot work properly.
- (2) Fixed the issue that T.38 -> T.30 cannot work.
- (3) Fixed the issue that cannot dial to IAX2 extension.
- (4) Fixed the issue that changes for "Auto Answer" on Grandstream phones cannot take effect.
- (5) Fixed the issue that changes for "Call Waiting" on Grandstream phones cannot take effect.
- (6) Fixed the issue that changes for "Keys Settings" on Grandstream phones cannot take effect.
- (7) Fixed the issue that changes for GMT-12 on Grandstream phones cannot take effect.
- (8) Fixed the issue that unable to delete the configured phones in Phone Provisioning.
- (9) Fixed the issue that Phone Provisioning cannot scan all phones within a local area network (LAN).
- (10) Fixed the issue that Phone Provisioning for Fanvil phones cannot work.
- (11) Fixed the issue that add multiple extensions, it will not begin to increase extensions from the specified extension number.
- (12) Fixed the issue that changes for Associated Email cannot be saved (PBX Basic -> Extensions. Edit -> Other -> Fax Configuration -> Associated Email).

5. New Features Descriptions

- (1) Added SIP trunk and early media focus.

Path: PBX Advanced Settings -> SIP Settings -> Custom Settings

Descriptions: If you want use SIP prack and earlymediafocus, you can setting on SIP Settings.

Register Timers	
Max Registration Time ⓘ	3600
Default Registration Time ⓘ	120
Qualify Freq ⓘ	60
Min Registration Time ⓘ	60
Qualify Gap ⓘ	100
Outbound SIP Registrations	
Register Timeout ⓘ	20
Register Attempts ⓘ	0
RTP Timers	
RTP Timeout ⓘ	60
RTP Keepalive ⓘ	0
Status Notifications	
Notify Ringing ⓘ	Yes
Notify Hold ⓘ	Yes
Advance Settings	
Session-timers ⓘ	Accept
Session-expires ⓘ	1800
DTMF Mode ⓘ	rfc2833
Trust RPID ⓘ	No
Contact Deny ⓘ	
Allow Re-invite ⓘ	yes
Send user=phone ⓘ	No
User Agent ⓘ	
Session-refresher ⓘ	Uas
Session-minse ⓘ	90
Relax DTMF ⓘ	No
Send RPID ⓘ	No
Contact Permit ⓘ	
Trunk Codec Priority ⓘ	Yes
Custom Settings	
prack	= yes
earlymediafocus	= yes

6. Optimization Descriptions

(1)None

✧ Release Notes of Version 1/12/13.1.0.0

1. Introduction

- (1) Firmware Version: 1.1.0.0, 12.1.0.0, 13.1.0.0,
- (2) Applicable Model: MUC1004, MUC2008, MUC2016
- (3) Release Date: Oct 26, 2015

CHANGES SINCE FIRMWARE RELEASE 1/12/13.0.0.15

2. New Features

- (1) Added support for phone provisioning of Aastra, Cisco, Fanvil, Grandstream, Htek, Panasonic, Polycom, Snom, Yealink phones.
- (2) Added the feature of PIN Users.
- (3) Added the feature of Dial by Name.
- (4) Added the feature of Feature Digit Timeout.
- (5) Added the feature of Bulk add extensions.
- (6) Added the feature of FAX detection.
- (7) Added the feature of GSM module detection.

3. Optimization

- (1) None

4. Bug Fixes

- (1) Fixed the issue that Name or Number configured on Extension, IVR, Ring Groups, Conferences, Outbound Routes, Inbound Routes, Paging and Intercom starting with 0 cannot be deleted successfully.
- (2) Fixed the issue that the extension password will be covered by the browser cookie's password (PBX Basic -> Extensions. Edit).
- (3) Fixed the display that the GUI of Callback was defective when it was configured.

5. New Features Descriptions

- (1) Added support for phone provisioning of Aastra, Cisco, Fanvil, Grandstream, Htek, Panasonic, Polycom, Snom, Yealink phones.

Path: Phone Provisioning->Phones

Add Phone

General
Codecs

General

Active
☒

Name

MAC

Manufacturer

Yeastlink

Aastra
Akuvox
Cisco
Fanvil
Grandstream
Htek
Panasonic
Polycom
Snom
Yeastlink

Phone Type

Network Parameters

☒ Use Phone's Setting
☐ Dynamic(DHCP)
☐ Static IP Address

IP Address

Subnet Mask

Gateway

DNS

Auto Answer
☐

Call Waiting
☒

Key As Send

Account

Line	Extension	Label	Display Name	Active
☐ Line1	105			☐

Configured Phones

+ Add Phone
X Delete the selected Phones

Page 1 of 1(1 Records)

	MAC	IP	Phone Model	Extension	Active	Name	Options
☐	001565565656	--	Yeastlink T19P	102	ON	--	☒ ☒

Not Configured Phones

Refresh

Page 1 of 1(6 Records)

MAC	IP	Phone Model	Options
000B826C7D0E	192.168.6.219	Grandstream GXP1450 1.0.6.11	☒
00A859D2919E	192.168.6.54	Fanvil C58 2.3.463.258	☒
00A859D34816	192.168.6.53	Fanvil F52 2.3.381.220	☒
0C110500388C	192.168.6.51	Akuvox SP-R53 53.0.3.41	☒
0C1105026B54	192.168.6.61	Akuvox SP-R59 59.0.3.41	☒
0C110502BFD8	192.168.6.71	Akuvox SP-R50 50.0.3.41	☒

(2) Added the feature of PIN Users.

Path: PBX Advanced Settings->PIN Users

PIN Users

Dial 'Access Code' *99 to enter the PIN User. 'Access Code' is configured through **Feature Codes**

General

Authentication Retries: 3
Digit Timeout: 5
Join Announcement: None
Fail Announcement: None

Save Cancel

PIN Users

Add PIN User Delete the Selected PIN Users

Page 1 of 1 (3 Records)

	Name	Password	PIN Set	Options
☐	test	1234		☒ ☐
☐	test1	1245		☒ ☐
☐	test3		11	☒ ☐

(3) Added the feature of Dial by Name.

Path: PBX Inbound Call Control->IVR

Edit IVR(620)

General

IVR Number: 620
IVR Description: Welcome
Announcement: default
Enable Direct Dial: Yes
Timeout: 3
Invalid Retries: 3
Invalid Destination: End Call
Timeout Retries: 3
Timeout Destination: End Call
CID Name Prefix: IVR620-

IVR Entries

Key	Destination	Delete
1	Dial by Name	☐
2	==choose one==	☐
	Conferences	☐
	DISA	☐
	Extensions	☐
	IVR	☐
	Queues	☐
	RingGroups	☐
	Voicemail	☐
	Dial by Name	☐
	End Call	☐

Back

Path: PBX Inbound Call Control->Queues.

Edit Queue(820)
General
Options
Advanced Settings

Caller Position Announcements

Frequency 30seconds ⓘ

Announce Position Yes ⓘ

Announce Hold Time Yes ⓘ

Periodic Announcements

Prompt None ⓘ

Frequency 30seconds ⓘ

Events,Stats

Event When Called No ⓘ

Member Status Event No ⓘ

Service Level 1minute ⓘ

Fail Over Destination

Destination Dial by Name ⓘ

==choose one==

Conferences

DISA

Extensions

IVR

Queues

RingGroups

Voicemail

Dial by Name

End Call

Save Back

Path: PBX Inbound Call Control-> Ring Groups.

Edit Ring Group(920)

General

RG Number 920 ⓘ

RG Name RingGroup920 ⓘ

Ring Strategy Ring all Selection ⓘ

Ring Time 45 ⓘ

Music on Hold calmrivier ⓘ

Ring Instead Of Moh ⓘ

CID Name Prefix RingGroup920- ⓘ

Alert Info <http://192.168.6.96>\ ⓘ

Ring Group Members

Extensions

102 <SIP>

104 <SIP>

105 <SIP>

106 <SIP>

107 <SIP>

108 <SIP>

109 <SIP>

110 <IAX>

111 <IAX>

Add >

< Remove

Members

503 <IAX>

Up ↑

Down ↓

Destination If No Answer

Destination Dial by Name ⓘ

==choose one==

Conferences

DISA

Extensions

IVR

Queues

RingGroups

Voicemail

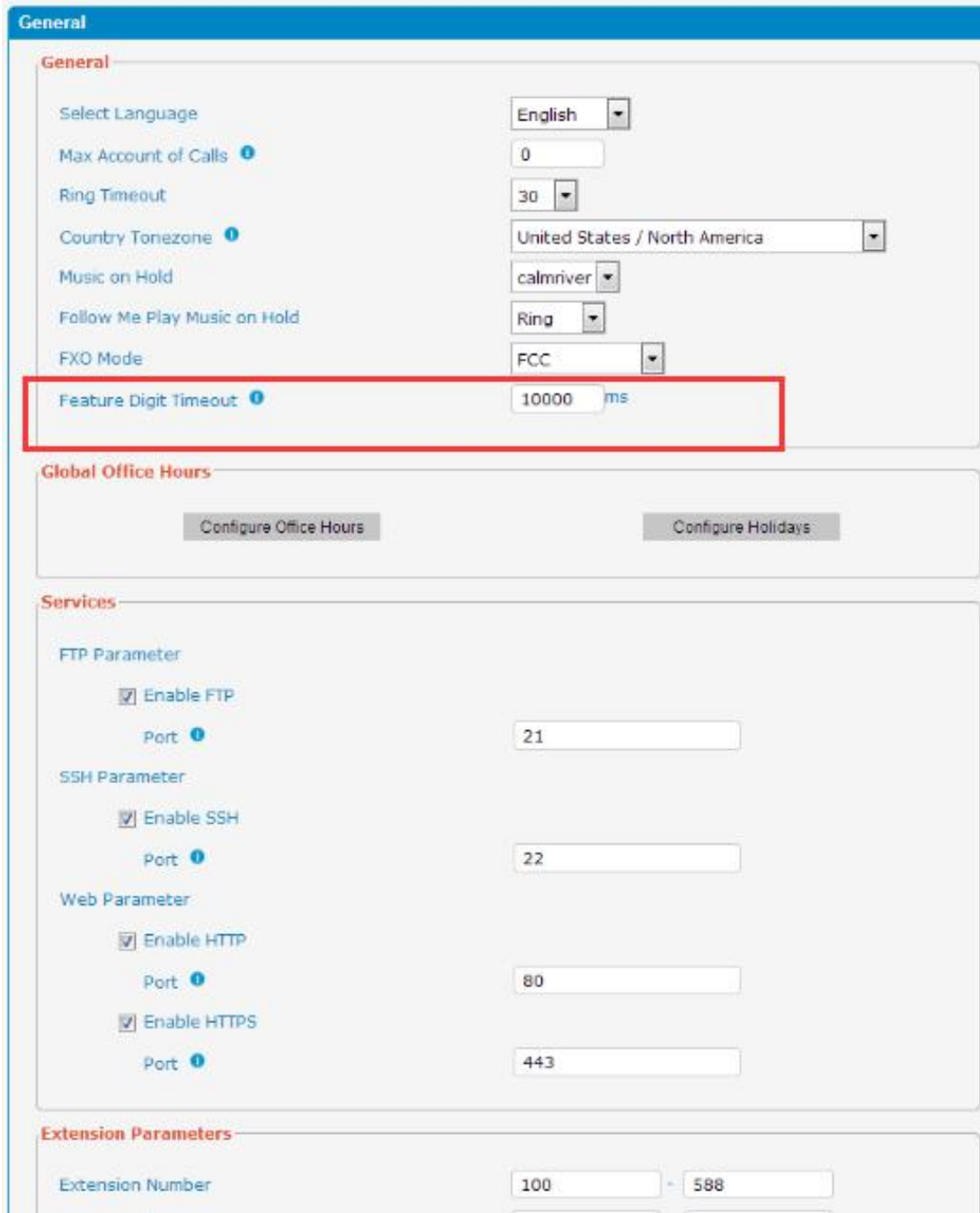
Dial by Name

End Call

Save Back

(4) Added the feature of Feature Digit Timeout.

Path: PBX Basic->General Preferences



The screenshot shows the 'General' configuration page for a PBX system. The 'Feature Digit Timeout' field is highlighted with a red box. The page is divided into several sections: General, Global Office Hours, Services, and Extension Parameters.

Section	Parameter	Value
General	Select Language	English
	Max Account of Calls	0
	Ring Timeout	30
	Country Tonezone	United States / North America
	Music on Hold	calmriver
	Follow Me Play Music on Hold	Ring
	FXO Mode	FCC
Global Office Hours	Configure Office Hours	Button
	Configure Holidays	Button
Services	FTP Parameter	
	Enable FTP	☒
	Port	21
	SSH Parameter	
	Enable SSH	☒
	Port	22
	Web Parameter	
	Enable HTTP	☒
Port	80	
Enable HTTPS	☒	
Port	443	
Extension Parameters	Extension Number	100 - 588

(5) Added the feature of Add bulk extensions.

Path: PBX Basic->Extensions

Add VoIP Extension

General
Voicemail
Options
Other

User Information

Extension Type
SIP

Extension Number
100

Range
5

Outbound CID

Emergency CID

VoIP Setting

Transport

RTP Encryption(SRTP)

DTMF Mode
33

Qualify

NAT

(6) Added the feature of FAX detection.

Path: PBX Basic->Extensions. Edit in extension 102, and Fax Associated Email address.

Edit VoIP Extension(102)

General
Voicemail
Options
Other

Spy Setting

Allow Being Spied
Disable

Spy Modes
Disable

IP Restriction

Deny

Permit

Web Login

Enable
☒

Login Name
102

Password
...
Weak

Fax Configuration

Associated Email
1425366352@qq.com

(7) Added GSM module function.

Path: Trunks->Physical Trunk

Analog Trunk				
Trunk Name	Port	Rx Gain	Ring Detect Timeout	Options
No Analog Trunk Detected				

Gsm Trunk					
				Page 1 of 1(1 Records)	
Trunk Name	Port	Type	Tx Gain	Rx Gain	Options
GSM3	3	GSM	40%	40%	

6. Optimization Descriptions

(1)None

✧ Release Notes of Version 1/12/13.0.0.15

1. Introduction

- (1) Firmware Version: 1.0.0.15, 12.0.0.15, 13.0.0.15
- (2) Applicable Model: MUC1004, MUC2008, MUC2016
- (3) Release Date: Sep 08, 2015

CHANGES SINCE FIRMWARE RELEASE 1/12/13.0.0.13

2. New Features

- (1) None

3. Optimization

- (1) Optimized of limited special characters.

4. Bug Fixes

- (1) Fixed the display that the GUI of CDR Report was defective.
- (2) Fixed the display that the GUI of Match Pattern and Strip of Outbound Routes display dislocation.
- (3) Fixed the display that the GUI of Outbound Routes Math Pattern input '-' character was defective.

5. New Features Descriptions

- (1) None

6. Optimization Descriptions

- (1) Optimized of limited special characters.

✧ Release Notes of Version 1/12/13.0.0.13

1. Introduction

- (1) Firmware Version: 1.0.0.13, 12.0.0.13, 13.0.0.13
- (2) Applicable Model: MUC1004, MUC2008, MUC2016
- (3) Release Date: Aug 18, 2015

CHANGES SINCE FIRMWARE RELEASE 1/12/13.0.0.12

2. New Features

- (1) None

3. Optimization

- (1) Optimized the feature of extension increased to 50 users from 32.
- (2) Optimized the display of Web GUI was defective after upgrade.

4. Bug Fixes

- (1) Fixed the issue that the multiple user login on different computers, the latter after the success login, the former will be logged out.
- (2) Fixed the issue that the configured the transfer number in Call Forward and configured Call Forward always to voicemail cannot take effect.
- (3) Fixed the issue that Packet Capture cannot work properly (System Tools -> Packet Capture).
- (4) Fixed the display that GUI of General Preferences Ring Timeout was defective.
- (5) Fixed the issue that dial *80 cannot work without create Paging and Intercom.

5. New Features Descriptions

- (1) None

6. Optimization Descriptions

- (1) Optimized the feature of extension increased to 50 users from 32.
- (2) Optimized the browser cache result in page fault after upgrade.