



Hytera DMR Conventional Series SIP Phone Gateway to Simultaneous Calls Application Notes

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Revision History

Version	Release Date	Description
R3.0	02-2015	Instructions on Dial-up Mapping are supplemented. Used with R7.0
R2.0	09-2014	IPPBX configuration is supplemented. Registration procedure is supplemented in Principle section. Radio private contact is added in repeater CPS programming. The restriction on operation slot of the radio is deleted. Add Priority Setting.
R1.2	04-2014	The restriction on operation slot of the radio is supplemented.
R1.1	03-2014	Updated based on R1.0.
R1.0	12-2013	Initial release.

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1. Overview

1.1 Definition

SIP (Session Initiation Protocol) Phone Gateway to Simultaneous Calls is a feature complied with and based on SIP protocol standard. This feature takes the repeater as a carrier to realize the real-time communications between the Radio and telephones such as PSTN phones, VoIP phones and mobile phones. Radio indicates terminals which can initiate and receive calls, including portable radio and mobile radio. Here we take portable radio for examples.

SIP is a standard protocol defined by IETF (Internet Engineering Task Force). It is mainly used to set up, terminate and modify interactive user sessions which include multimedia elements such as video, voice, instant messaging, online game and virtual reality. Due to its openness and flexibility, SIP protocol is now widely used in VoIP phones (IP phones) with the development of the Internet. SIP Phone Gateway to Simultaneous Calls feature can realize the real-time communications between radios and VoIP phones, PSTN phones and mobile phones.

SIP Phone Gateway to Simultaneous Calls feature has the following highlights:

- Reducing the cost of building and maintaining communication networks.
- Allowing users to choose appropriate contacts conveniently.

1.2 Principle

The repeaters register the private call contacts and group call contacts to the IPPBX device, and then communicate with other telephone devices via the IPPBX device to realize the real-time communications between radios and telephone devices.

1.2.1 Registering

After connected with the SIP phone network, the repeater will register the phone call contacts with the IPPBX device upon power-on. Afterwards, the radios can communicate with telephone devices.

Phone call contacts indicate private contacts and group contacts which can communicate with telephone devices.

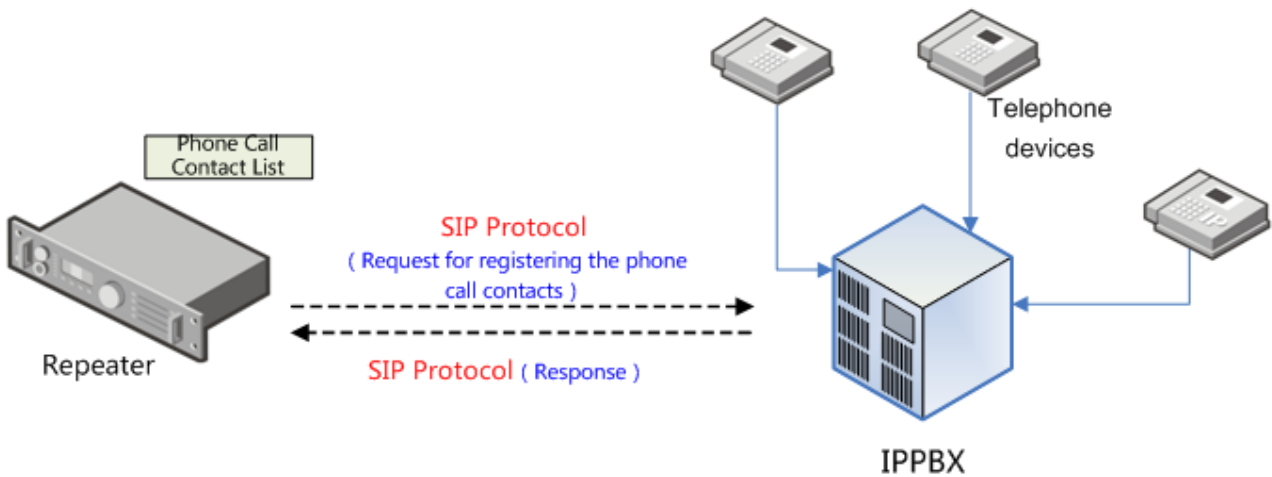


Figure 1-1 Diagram of Repeater Registering Phone Call Contacts

1.2.2 Radios Calling Telephone Devices

When making a call from the radio to the telephone device, you can input the phone number via the DTMF (Dual Tone Multi-Frequency) keypad. The called number will generate a DTMF signaling which will be sent to the repeater by the radio. The repeater will decode the received signaling to acquire the called number, and use the number to generate a SIP Protocol call request for the IPPBX device. The IPPBX device will search the address and location of the called number, and then access the telephone network via the corresponding interface according to the corresponding rules and make a call to the called number. See below:

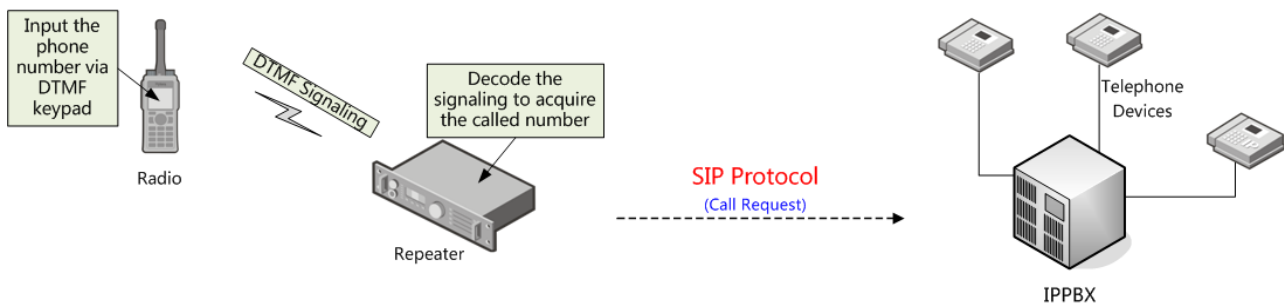


Figure 1-2 Diagram of Radios Calling Telephone Devices

1.2.3 Telephone Devices Calling Radios

When making a call from the telephone device to the radio, you can input the called number via the telephone device, and then the telephone network will send the call request to the IPPBX device. The IPPBX device will forward the request to the repeater, and the repeater will make a call to the called radio. See below:

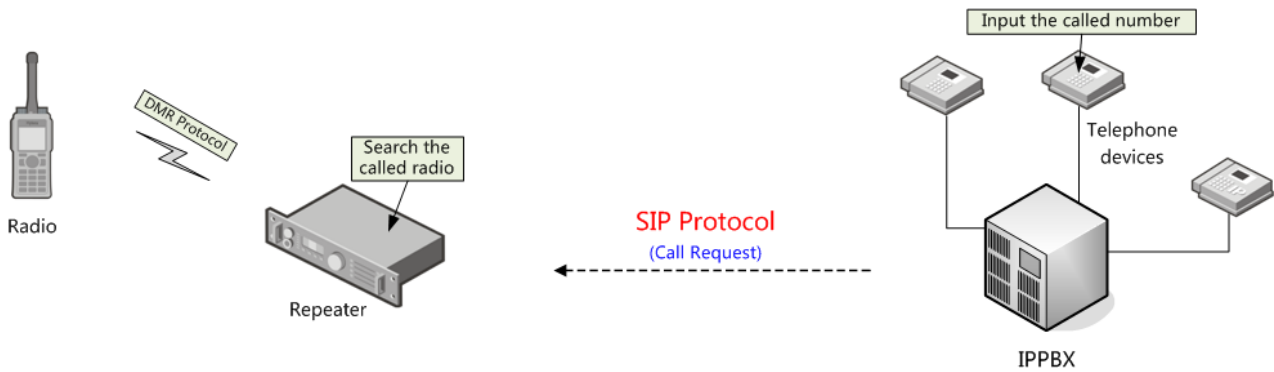


Figure 1-3 Diagram of Telephone Devices Calling Radios

1.3 Typical Network Topological Structure

The Typical Network Topological Structure of radio network and telephone network includes radios, repeaters, Ethernet exchange, IPPBX and telephone devices. The radios access the Ethernet via repeater, then the repeaters connect to IPPBX via the Ethernet, and finally IPPBX connects to the telephone devices via the telephone network. See below.

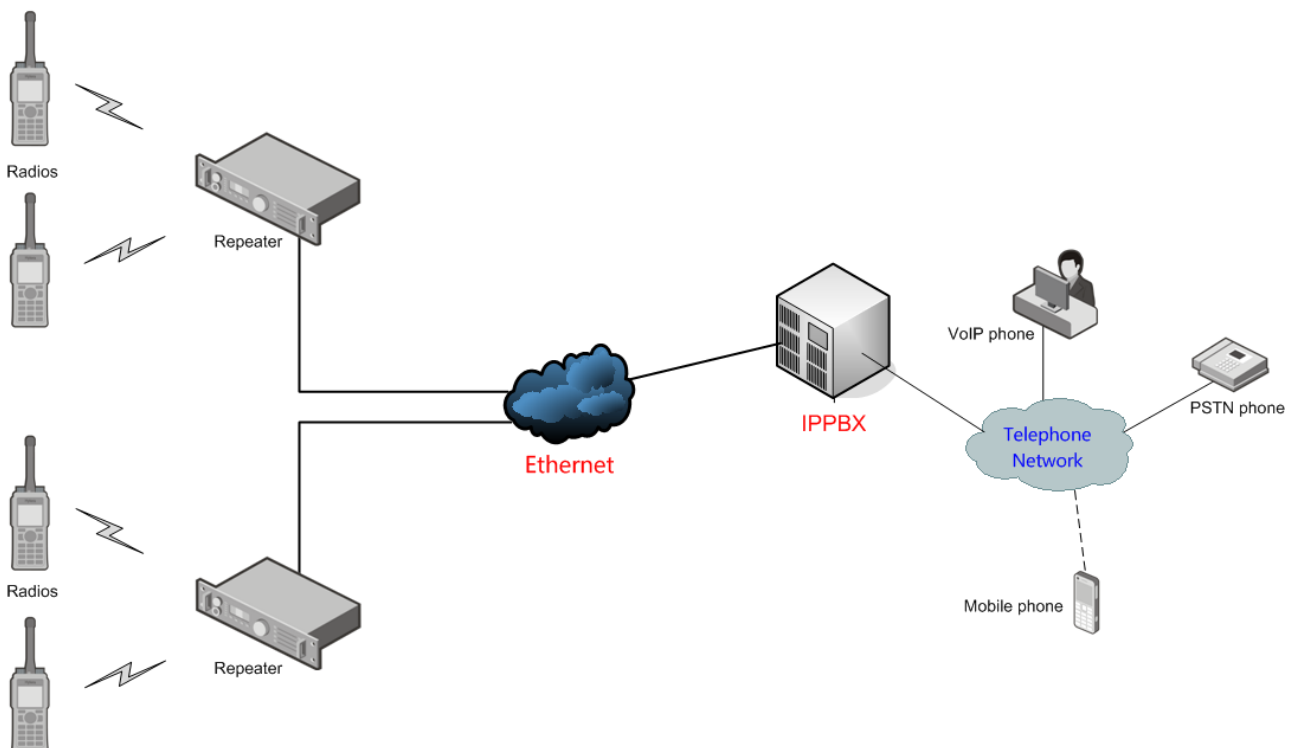


Figure 1-4 Typical Network Topological Structure

In one SIP phone network, multiple repeaters can be connected to the telephone network simultaneously. The communication volume depends on the IPPBX and network configurations. Moreover, multiple IPPBX can form a larger SIP phone network via cascade. IPPBX can also connect to

the PSTN phone via the telephone interface and connect to the VoIP phone via the Ethernet, thus you can add the VoIP phone in the SIP phone network to build a larger telephone network. In such case, the radios, PSTN phones, VoIP phones and mobile phones can communicate with each other via the telephone network.

1.4 Restriction

- The repeater complies with the protocol “SIP/2.0(RFC 3261)”.
- In the IP Multi-site Connect system, it is not advised to enable SIP feature in the dispatcher or any repeater connecting with a third-party device, so as to avoid heavy load of the repeater.
- This feature is available for conventional series radios and repeaters on digital channel only.
- This feature is available for portable radios and mobile radios with display only.
- The parameter **Telephone Interconnection Enable** must be checked via CPS for the repeater to support phone call service. Please refer to [4.3.1 Repeater Setting](#) for detailed configuration.
- Only after the channel of the radio is related to **Phone System**, the radio will be able to initiate phone calls, that is, the parameter **Phone System** of the channel cannot be set to **None**. If the firmware version of the radio is lower than R6.5, the channel parameter **TX Admit** must be set to **Always Allow**, otherwise, the radio will not be able to respond to the phone request. Please refer to [4.3.2 Radio Setting](#) for detailed configuration.
- In the IP Multi-site Connect system, if the parameter **Third Party Connect Mode** of a repeater is set to **Normal** (CPS configuration path: Conventional -> General Setting -> Network -> Application Programming Interface), then only one repeater in the system can have the Telephone Interconnection Enable feature enabled, otherwise the communications may be terminated abnormally. If the parameter **Third Party Connect Mode** is set to **Selective**, all repeaters in the system must enable this feature. If the firmware version of the repeater is R6.0 or R6.5, it is recommended to set **Third Party Connect Mode** to **Normal**. From firmware version R7.0 and above, when using the SIP phone feature, the Phone feature is used by the whole network by default and it is not associated to **Third Party Connect Mode**. That is to say, only one repeater in IP interconnect system is enabled with Phone feature, all the radios in the system will communicate with telephone system via this repeater.
- In the SIP phone network, the phone gate ID, radio ID and repeater ID must be unique from each other.

1.5 Version

- The SIP Phone Gateway to Simultaneous Calls feature is supported in repeaters and radios with firmware version of R6.0 and above. Please upgrade the firmware version of repeaters and radios to R6.0 and above for proper operation.
- From R6.5 and above, the registration method of SIP Phone extension number is changed from RRS service registration of R6.0 to registration via repeater.
- The Dial-up Mapping feature is supported in repeaters and radios with firmware version of R7.0 and above. Please upgrade the firmware version of repeaters and radios to R7.0 and above for proper operation.
- For more upgrade instructions on SIP Phone Gateway to Simultaneous Calls feature, please refer to the corresponding *Release Notes*.

2. Application Requirements

2.1 Device Requirements

2.1.1 Radio and Telephone Device

Radio

Generally, DMR/PDT conventional series radios operate as the calling party in the phone network, including PD78X, MD78X and X1p series. Here takes DMR PD78X for example.

Repeater

In SIP phone system, the repeaters serve as the gateway between the radio network and Ethernet.

Generally, DMR repeaters (such as RD98X) are employed. Here we take RD98X for example.

Telephone Device

Subject to actual conditions.

2.1.2 Network Device

Exchange

Include Ethernet exchange, optical fiber exchange, and phone call exchange. Please consult the supplier for detailed information.

Router Device

Include firewall, NAT and router (such as CISCO 1841). Please consult the supplier for detailed information.

It is recommended to use Grandstream UCM6102.

IPPBX Device

IPPBX Device is a private IP exchange with built-in router and firewall. It provides data exchange services for different physical interfaces to realize the logical abstraction of the physical interfaces.

2.2 Network Requirements

- A telephone network provided by Telecommunication Operator or a private telephone network provided by users is needed.
- The repeaters access the telephone network via IP network.

3. Reference

N/A

4. Connection and Configuration

4.1 Configuration Tools

Please choose the SIP Phone Gateway to Simultaneous Calls configuration according to network topological structure and actual requirements. Configuration parameters and tools are listed below:

CPS

The dealer can configure the radio and repeater via Customer Programming Software (CPS).

The CPS version must be R7.0 or above. You can consult your local dealer for more information on CPS.

For better configuration, please refer to the help file of CPS for details.

4.2 IPPBX Device Configuration

To make sure the radio and telephone device can access the telephone network properly, please configure the following parameters according to the reference provided the device supplier.

- SIP extension: Sets the SIP phone extension number of the call contacts such as PSTN phone contacts, mobile phone contacts, VoIP phone contacts, radio private call contact, radio group call contact.

Caution

For dialing rules of radio private contacts/ group contacts configured in SIP extension, please see [Phone Call Configuration](#).

The radio private contacts/ group contacts in the Phone Call List (See [Phone Call Configuration](#)) of the repeater must be added to the SIP extension; otherwise, the phone call cannot be established successfully.

- SIP extension password: Sets the code for SIP extension to access IPPBX. It corresponds to the "Phone Gateway ID" under the phone configuration interface of repeater (CPS Configuration Path: Conventional -> Phone -> Phone System -> Phone System N -> Phone Gateway ID). Each radio is considered to be an extension. The repeater serves as the agent of the radio and registers with the IPPBX.

 **Caution**

In one telephone system, the SIP passwords of all SIP extensions registered with IPPBX via the repeater must be the same, otherwise the contacts cannot be registered properly.

- IP address of Local Area Network (LAN): The repeater connects to the IPPBX device via this IP address. This parameter is corresponding to the Telephone Gateway IP parameter of the repeater (CPS Configuration Path: Conventional -> Phone -> Phone System -> Phone System N -> Telephone Gateway IP). When connecting to the IPPBX via LAN port, the IP address of the IPPBX must be set properly (UCM Configuration Path: Settings -> Network Settings -> Basic Settings -> LAN); When connecting to the IPPBX via WAN port, you can acquire the IP address of the IPPBX from the LCD display of the IPPBX.
- Keep-alive: It is recommended that the Keep-alive parameter of IPPBX devices should be disabled to improve the stability of the link.

Here we take Grandstream UCM 6102 for example. In the following example, only the parameters which must be configured will be described. For more parameter details, please refer to the corresponding configuration guide of Grandstream.

Step 1 Access WEB configuration screen of UCM via browser.

The web address of WEB screen of UCM is <https://IP:8089>.

In the above address, IP denotes the IP address of UCM, which is IP address of WAN/LAN; 8089 is the default port value of UCM. The default administrator account and password are both admin.

Step 2 In the UCM screen, go to “PBX->Basic/Call Routes->Extensions->Create New User” to add a new extension.

Set Extension and SIP/IAX Password properly in Create New User screen, select SIP and deselect Enable Keep-alive.

The screenshot shows the 'Create New User' configuration window with the following settings:

- General:**
 - Extension: 5003
 - CallerID Number: (empty)
 - SIP/IAX Password: 123456s
 - Voicemail Password: 110000
 - Call Forward Unconditional: (empty)
 - Call Forward Busy: (empty)
 - Language: Default
 - Skip Voicemail:
 - CallerID Name: (empty)
 - Permission: Internal
 - Enable Voicemail:
 - Email Address: (empty)
 - Call Forward No Answer: (empty)
 - Ring Timeout: (empty)
 - Auto Record: No
 - Support Hot-Desking Mode:
- Technology:**
 - SIP:
 - IAX:
 - Analog Station: None
- SIP Settings:**
 - NAT:
 - DTMF Mode: RFC2833
 - Enable Keep-alive:
 - AuthID: (empty)
 - Can Reinvite: No
 - Insecure: Port
 - Keep-alive Frequency: 60
- Other Settings:**
 - SRTTP:
 - Strategy: Allow All
 - Codec Preference: (empty)
 - Fax Detection:
 - Skip Trunk Auth:

Buttons: Cancel, Save

Figure 4-1 Adding a New Extension

The value of Extension must be consistent with the Called Number in 6.1 **Radio Called Number**; otherwise, the call will fail to be established. Moreover, the value of Extension cannot be out of the extension number range. The extension number range can be viewed and modified in General screen (path: PBX->Internal Options-> General -> Extension Preference -> User Extension).

Step 3 In the UCM screen, go to “PBX->Basic/Call Routes ->Analog Trunks->Create New Analog Trunk” to create an analog trunk.

⚠ Caution: If you need to connect to PBX via telephone lines, you must create an analog trunk via this procedure. If not, you can skip this procedure.

In this screen below, select the Channels per actual needs and enter the analog trunk name in Trunk Name.

Edit Analog Trunk: trunk_1

Channels: 1 2

i Trunk Name:

Advanced Options

i Enable Polarity Reversal: ▼

i Current Disconnect Threshold (ms): ▼

i RX Gain:

i Use CallerID: ▼

i Caller ID Scheme: ▼

i Auto Record: ▼

i Ring Timeout:

i TX Gain:

i Fax Detection: ▼

Tone Settings

i Busy Detection: ▼

i Congestion Detection: ▼

i Tone Country: ▼

i Busy Tone:

i Congestion Tone:

i PSTN Detection:

Figure 4-2 Creating an Analog Trunk

Step 4 In the UCM screen, go to “PBX->Basic/Call Routes ->VoIP Trunks->Create SIP/IAX Trunks” to create a VoIP trunk.

⚠ Caution: If you need to connect to PBX via network cable, you must create a VoIP trunk via this procedure. If not, you can skip this procedure.

Enter the IP address of upper level PBX in Host Name and enter a valid extension number assigned by this PBX (for example, 2126). Leave the Password blank.

Figure 4-3 Creating a VoIP Trunk

Step 5 In the UCM screen, go to “PBX->Basic/Call Routes ->Outbound Routes->Create New Outbound Rule” to create the outbound rule.

Please configure the Pattern according to the digits of the extension number. Generally, the extension number has four digits, thus, the Pattern can be set to “_XXXX”. If you need to make a call to the mobile phone, you need to supplement the Pattern of mobile phone number (for example, _XXXXXXXXXX).

Select the created analog trunk or VoIP trunk in the Use Trunk.

Enter 0 in Strip, which allows you to make a phone call directly.

⚠ Caution: If you need to connect to PBX via telephone lines as well as network cable, you must create two outbound rules, which are used for analog trunk and VoIP trunk respectively.

Create New Outbound Rule [x]

Calling Rule Name:

Pattern:

Privilege Level:

Password:

Send this call through trunk

Use Trunk:

Strip:

Prepend:

Use Failover Trunk:

Trunks	Strip	Prepend	Options
Click to add failover trunk			

Figure 4-4 Creating a Outbound Rule

Step 6 In the UCM screen, go to “PBX->Call Features->IVR->Create New IVR” to create an IVR.

Figure 4-5 Creating an IVR

Step 7 In the UCM screen, go to “PBX->Basic/Call Routes ->Inbound Routes->Create New Inbound Rule” to create the inbound rule.

Select the created analog trunk or VoIP trunk in the Trunks and select the created IVR in Default Destination.

⚠️ Caution: If you need to connect to PBX via telephone lines as well as network cable, you must create two inbound rules, which are used for analog trunk and VoIP trunk respectively.

Trunks: AnalogTrunks -- Analog_Phone

DID Pattern: s /

Default Destination: IVR 7001

Time Condition:

Time	Destination	Options
Click to add Time Condition		

Cancel Save

Figure 4-6 Creating a Inbound Rule

4.3 CPS Configuration

To use the Phone feature, the following information should be configured:

4.3.1 Repeater Setting

Phone System Parameter Configuration

Path: Conventional -> Phone -> Phone System -> Phone System N. See [Figure 4-1](#).

Parameters: See the parameters in [Figure 4-1](#).

Description: See [Table 4-1](#).

⚠ Caution: The Connect Code and Disconnect Code must be different; otherwise, the repeater cannot access the phone system properly.

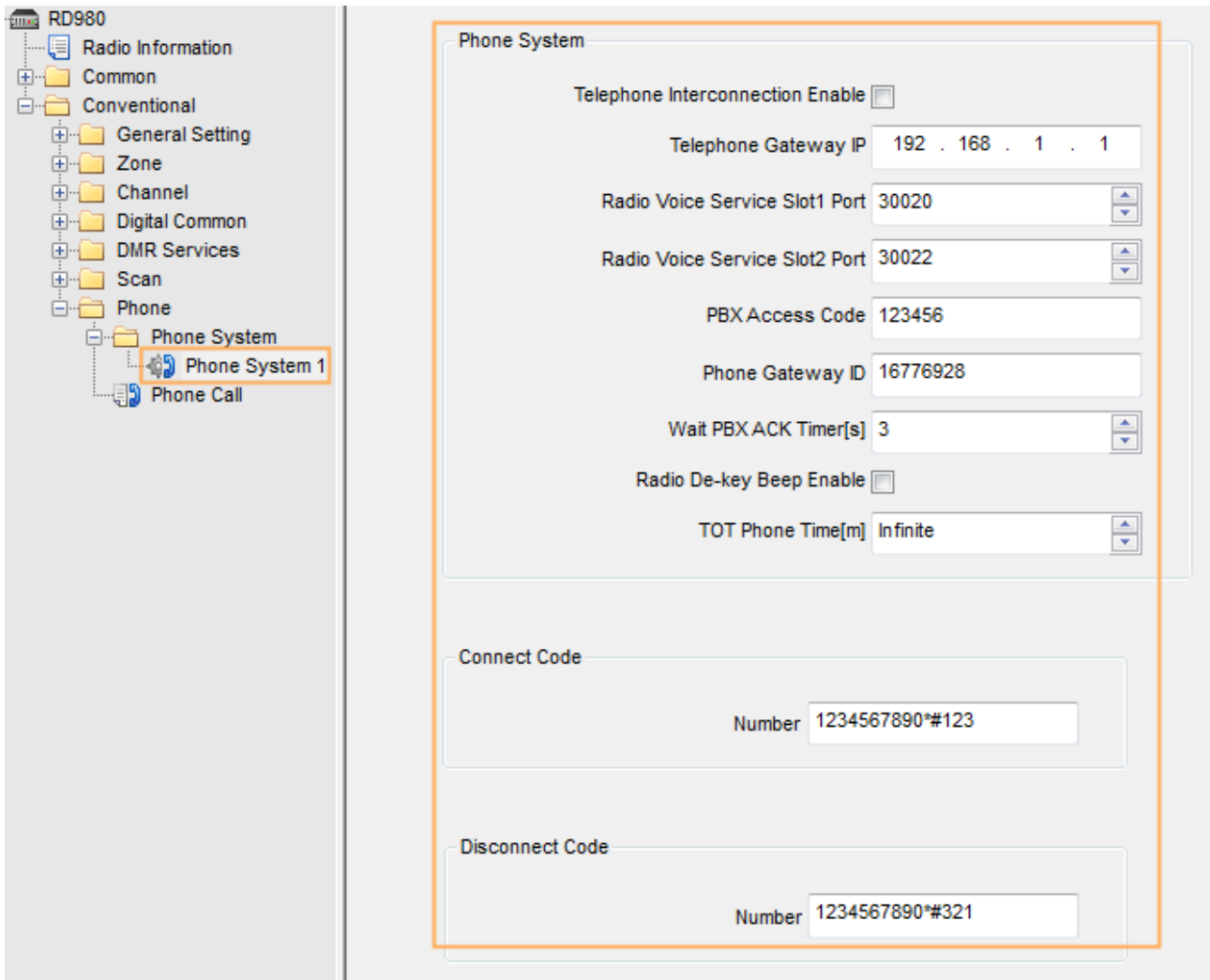


Figure 4-7 Phone System Configuration Interface for Repeater

Parameters	Description	Setting
Telephone Interconnection Enable	Sets whether to enable or disable the Telephone Interconnection Enable feature.	Method: Check
Telephone Gateway IP	Sets the IP address of the IPPBX device. This parameter must be consistent with the parameter IP address of LAN of the IPPBX device; otherwise the repeater cannot be connected to IPPBX.	Method: Manual input Range: 0.0.0.0–255.255.255.255

Parameters	Description	Setting
Radio Voice Service Slot1 Port	Sets the port which the repeater uses when transferring telephone voice services in Slot 1 or 2. Make sure the port number is unique and even.	Method: Manual input Range: 1024–65535
Radio Voice Service Slot2 Port		
PBX Access Code	Sets authentication code which the repeater uses to register the called contact information with the IPPBX device. This parameter must be consistent with the parameter SIP extension password of the IPPBX device; otherwise the called contact cannot be registered properly. If there is no IPPBX device extension password, leave blank here.	Method: Manual input Range: 32-digits ASCII characters
Phone Gateway ID	Sets the ID which the repeater uses to identify the current call as phone call. In the SIP phone network, the phone gate ID, radio ID and repeater ID must be unique from each other.	Method: Manual input Range: 1–16776943
Wait PBX ACK Timer	Sets the time in which the repeater will wait for ACK from the telephone device after the radio user initiates a SIP phone call via the repeater. If the repeater cannot receive ACK from the telephone device before the time expires, the call establishment fails. Please set this parameter according to actual situation. For example, it takes a long time for the telephone device to connect with the external phone network via IPPBX; in this case, the timer should be set long enough for successful call establishment.	Method: Manual input Range: 1–10 seconds

Parameters	Description	Setting
Radio De-key Beep Enable	With this feature enabled, the phone user will hear a beep when the radio user release the PTT during a call.	Method: Check
TOT Phone Time	With this feature enabled, the phone user will hear a beep when the radio user release the PTT during a call.	Method: Manual input Range: Infinite, 1-255 minutes
Number (Connect Code)	Sets the number which the repeater uses to identify phone call answering status of the radio. The radio user sends the Connect Code by holding down the PTT key. The repeater identifies phone call answering status of the radio according to the Connect Code.	Method: Manual input Range: 0-9 (whole number), A, B, C, D, *, #
Number (Disconnect Code)	Sets the number which the repeater uses to identify phone call rejection status of the radio. The radio user sends the Disconnect Code by holding down the PTT key. The repeater identifies phone call rejection status of the radio according to the Disconnect Code.	

Table 4-1 Phone System Description for Repeater

Phone Call Configuration

Path: Conventional -> Phone -> Phone Call -> Phone Call List. See [Figure 4-2](#).

Parameters: See the parameters in the orange circles in [Figure 4-2](#).

Description: Sets the Phone Call ID and the Slot ID which the repeater uses to forward the Phone Call. Phone call contacts indicate private contacts and group contacts which can communicate with telephone devices.

Please refer to the CPS help file for parameter description.

When configuring the phone call contacts, the phone ID must be unique; otherwise the phone ID cannot be registered properly.

 **Note:**

- When “Keep-alive” option is enabled for IPPBX, it is recommended to keep the number of phone call contacts capped at 32; when this option is disabled, the maximum number of phone call contacts is recommended to be capped at 64.
- After the SIP phone network is connected, the repeater will register the set Phone Call information with the IPPBX device upon power-on. When the phone makes a call to the Phone Call ID, please refer to [6.1 Radio Called Number](#) for proper dial scheme; otherwise, the call cannot reach the called radio.
- The slot set here must be consistent with the current slot used by the radio. For example, if the slot here is set to Slot 1 while the radio used Slot 2, the radio will not be able to receive the call. If the slot of the radio is set to Pseudo Trunk, the slot here can be set to Slot 1 or Slot 2 and the radio still can receive calls.

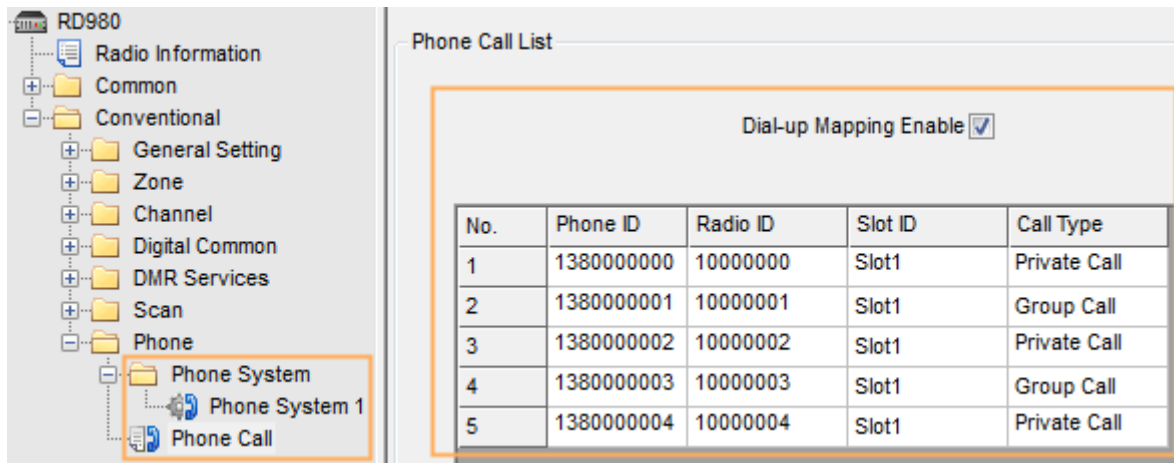


Figure 4-8 Phone Call Configuration Interface

Priority Setting

Path: Conventional -> General Setting -> Accessories -> Priority Control. See [Figure 4-3](#) and [Figure 4-4](#).

Parameters: See the parameters in the orange circles in [Figure 4-3](#) and [Figure 4-4](#).

Description:

- To ensure normal phone call service, it is recommended to enable Phone Priority option.



Figure 4-9 Priority Control for Repeater enabling Phone Call feature

- If a repeater not enabling the Phone Call feature need phone call service, you must set Path Priority to “Repeat Request” and set Repeat Request Priority to “Local Repeating”.

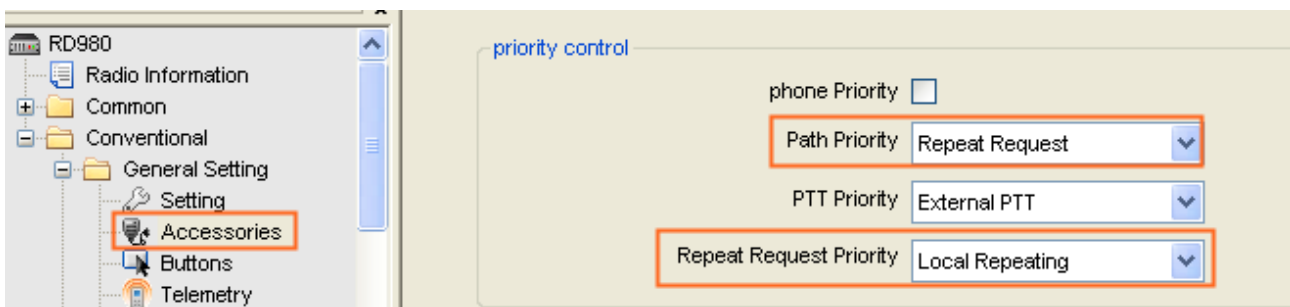


Figure 4-10 Priority Control for Repeater not enabling Phone Call feature

4.3.2 Radio Setting

Phone System Parameter Configuration

Path: Conventional -> Phone -> Phone System -> Phone System N. See [Figure 4-5](#).

Parameters: See the parameters in [Figure 4-5](#).

Description: Please refer to [Table 4-2](#) for key parameters of Phone System and refer to the CPS help file for other parameter descriptions.

⚠ Caution: The Connect Code and Disconnect Code must be different; otherwise, the radio cannot access the phone system properly.

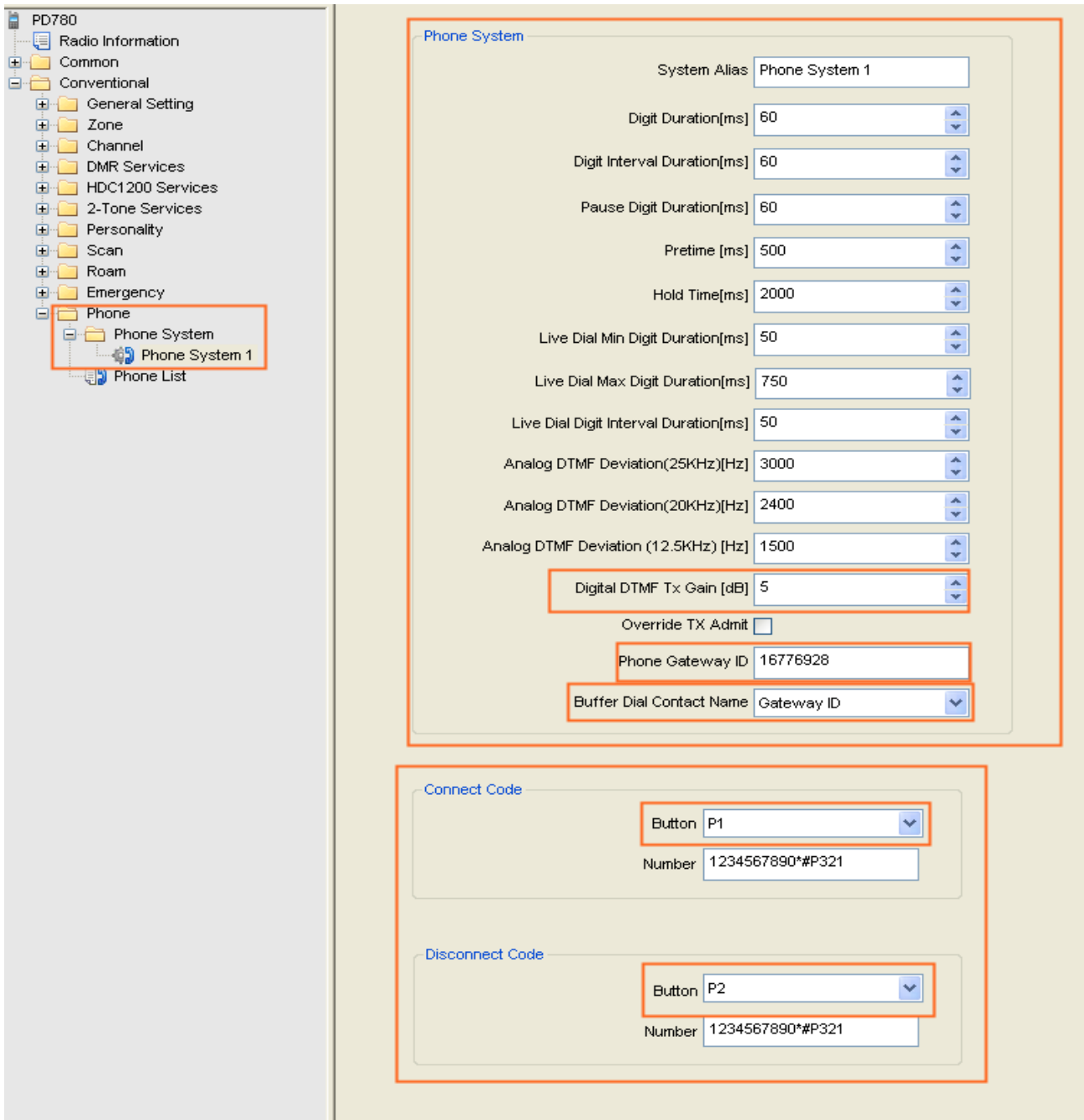


Figure 4-11 Phone System Configuration Interface for Radio

Parameters	Description	Setting
Digital DTMF Tx Gain	<p>Sets the DTMF Tx Gain in digital channel. The larger the gain value is, the stronger the transmitted DTMF signal is.</p> <p>When the radio employs AMBE+2™ Audio Codec Technology, the value of this parameter must be larger than 4 for the radio to support extension dialing or making phone call with password properly.</p>	<p>Method: Manual input</p> <p>Range: -8–8 dB</p>
Phone Gateway ID	<p>Sets the ID which the radio uses to identify the current call as phone call.</p> <p>This parameter must be consistent with the Phone Gateway ID of the repeater (See 4.3.1 Repeater Setting) for the repeater to identify phone calls properly.</p>	<p>Method: Manual input</p> <p>Range: 1–16776943</p>
Buffer Dial Contact Name	<p>Sets the type of Buffer Dial Contact Name.</p> <ul style="list-style-type: none"> ● Follow Tx Contact Name: Save the Tx Contact Name preset for the channel. ● Gateway ID: Save the Phone Gateway ID. 	<p>Method: Selects from the dropdown list.</p> <p>Note: To make phone calls, this parameter must be set to Gateway ID</p>
Button (Connect Code)	<p>Sets the button to quickly view the Connect Code of the radio.</p> <p>When the radio receives a phone call, the radio user can press this button to access the DTMF dial box. After inputting the preset Connect Code in the box, hold down the PTT key to answer the call.</p> <ul style="list-style-type: none"> ● None (Radio user must input the Connect Code via the keypad manually) ● P1 (for portable radio with display only) ● P5 (for mobile radio only) 	<p>Method: Selects from the dropdown list.</p> <p>Note: The button is valid only in DTMF keypad mode.</p>

Parameters	Description	Setting
Number (Connect Code)	<p>Sets the code for the radio to access the phone system.</p> <p>The radio user sends the Connect Code by holding down the PTT key. The repeater identifies phone call answering status of the radio according to the Connect Code.</p> <p>This parameter must be consistent with the Number (Connect Code) of the repeater (See 4.3.1 Repeater Setting) for the repeater to identify phone call status properly.</p>	<p>Method: Manual input</p> <p>Range: 0–9 (whole number), A, B, C, D, *, #</p> <p>Note: A, B, C, D are not available for radio keypad inputting now.</p>
Button (Disconnect Code)	<p>Sets the button to quickly view the Disconnect Code of the radio.</p> <p>In case of an incoming call or during an ongoing call, the radio user can press this button to access the DTMF dial box. After inputting the preset Disconnect Code in the box, hold down the PTT key to reject the incoming call or hang up the ongoing call.</p> <ul style="list-style-type: none"> ● None (Radio user must input the Disconnect Code via the keypad manually) ● P2 (for portable radio with display only) ● P6 (for mobile radio only) 	<p>Method: Selects from the dropdown list.</p> <p>Note: The button is valid only in DTMF keypad mode.</p>

Parameters	Description	Setting
Number (Disconnect Code)	<p>Sets the code for the radio reject to access or exit the phone system.</p> <p>The radio user sends the Disconnect Code by holding down the PTT key. The repeater identifies phone call rejection or termination status of the radio according to the Disconnect Code.</p> <p>This parameter must be consistent with the Number (Connect Code) of the repeater (See 4.3.1 Repeater Setting) for the repeater to identify phone call status properly.</p>	<p>Method: Manual input</p> <p>Range: 0–9 (whole number), A, B, C, D, *, #</p> <p>Note: A, B, C, D are not available for radio keypad inputting now.</p>

Table 4-2 Phone System Key Description for Radio

Phone List Configuration

Path: Conventional -> Phone -> Phone List. See [Figure 4-6](#).

Parameters: See the parameters in the orange circles in [Figure 4-6](#).

Description: Radio user can view and make phone calls to the preset Phone Call Alias and corresponding numbers via the radio menu. Please refer to the CPS help file for parameter description.

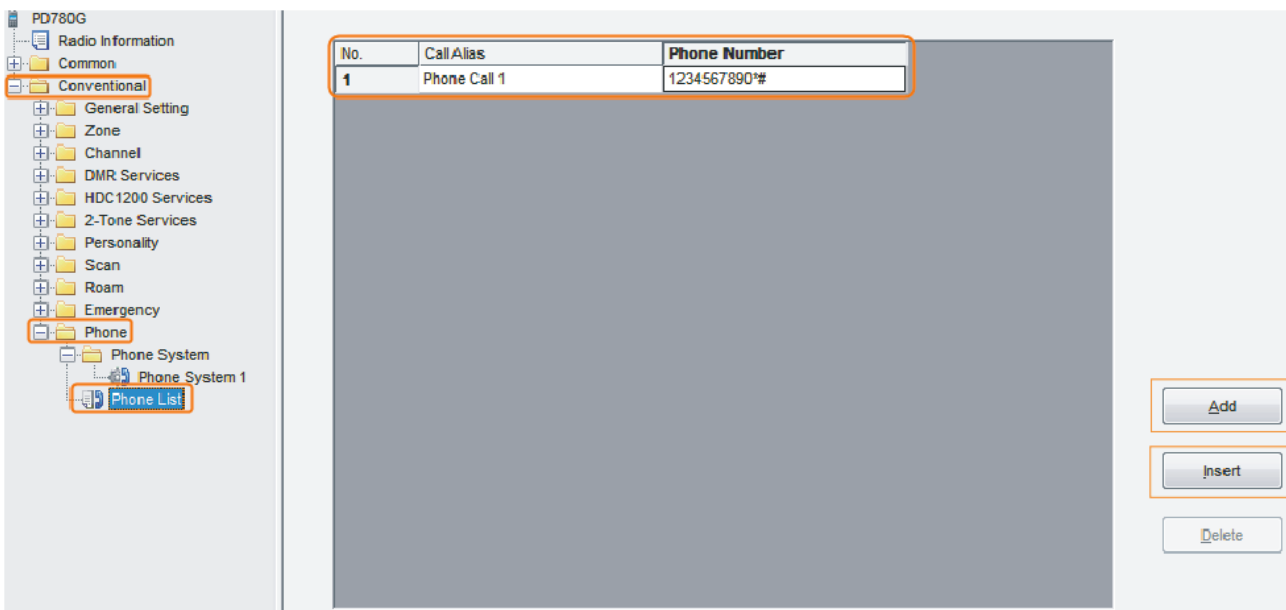


Figure 4-12 Phone List Configuration Interface

Channel Configuration

Path: Conventional -> Channel -> Digital Channel -> CH Dn. See Figure 4-7.

Parameters: See the parameters in the orange circles in Figure 4-7.

Description: See Table 4-3.

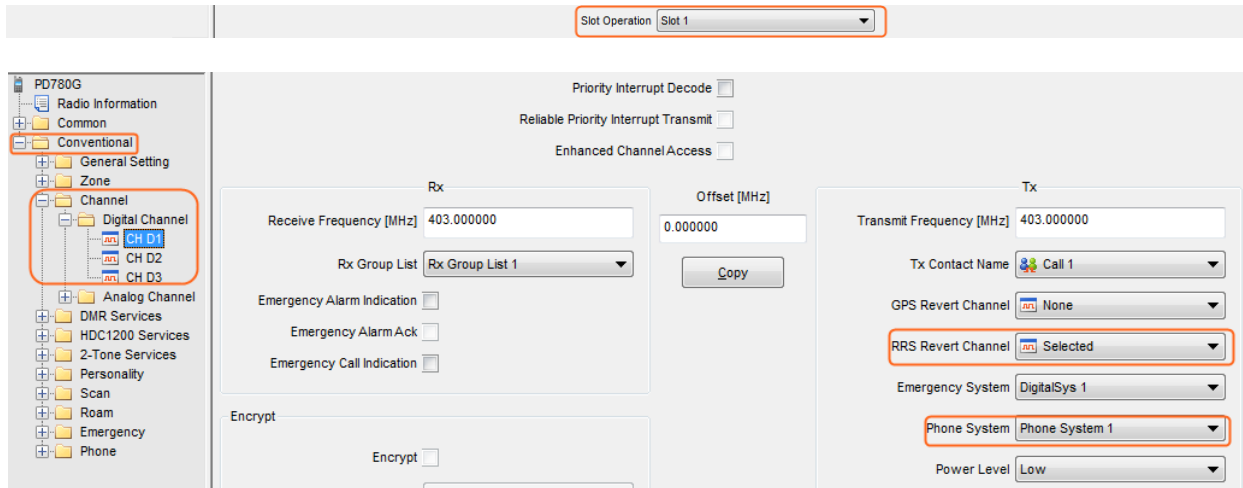


Figure 4-13 Channel Configuration Interface for Radio

Parameters	Description	Setting
Slot Operation	Sets the slot for communication or data transferring.	Method: Selects Slot 1 or Slot 2 in the dropdown list. Note: Pseudo Trunk is not supported now. If it is selected, the radio may fail to call the telephone devices.
RRS Revert Channel	Sets the channel for Radio Register Service (RRS). When the radio registers with the RRS server, it will revert to this channel for registration. After the registration, the radio will go back to work on the previous channel.	Method: Selects Selected from the dropdown list.

Parameters	Description	Setting
Phone System	Sets a preset Phone System for the digital channel. When the radio operates in that channel, radio user can use the Phone System set for the channel.	<p>Method: Selects from the dropdown list.</p> <p>Note: You can configure the Phone System in the dropdown list via “Phone -> Phone System”. Please refer to Phone System Parameter Configuration.</p>

Table 4-3 Phone System on Channel Description

Phone Menu Configuration

Path: Conventional -> General Setting -> Menu -> Common Menu -> Phone. See [Figure 4-8](#).

Parameters: Phone

Description: Sets whether to include Phone on the radio menu. Radio user can access the Phone menu via the menu.

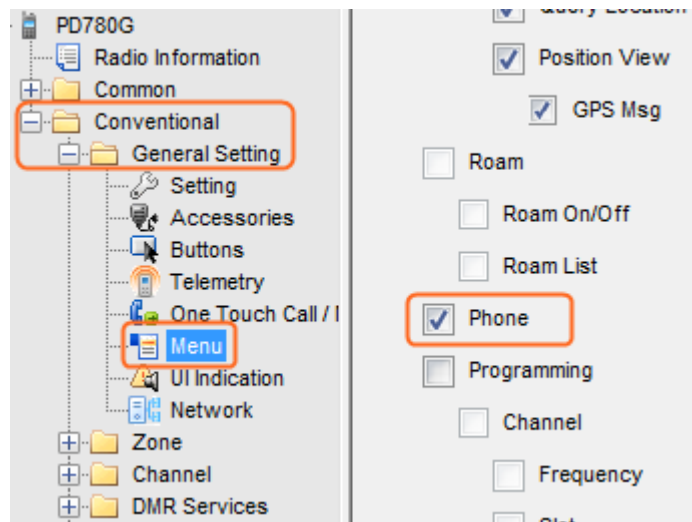
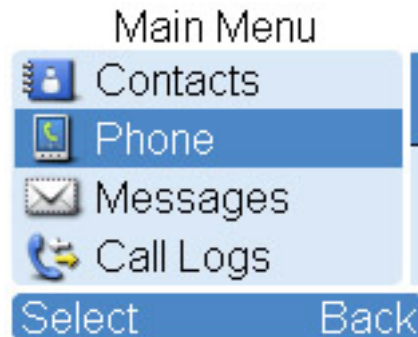


Figure 4-14 Phone Menu Configuration Interface

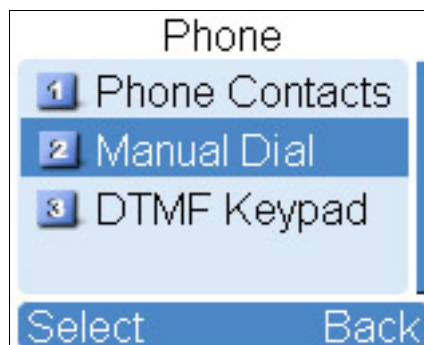
Accessing the Phone Menu: After configuration, the radio user can access the Phone menu via the following methods: (here takes the digital channels of PD78X for example)

⚠ Caution: The radio user can access the Phone menu only when the Phone menu is checked and Channel Phone System is configured.

1. In the home screen, go to “Menu -> Phone”.



2. In the Phone menu, you can view the Phone List or input a phone number via Manual Dial.



📌 Note: Phone List saves private contacts preset via CPS. Please refer to [Phone Call Configuration](#).

DTMF Keypad Programmable Keys (Optional)

Path: Conventional -> General Setting -> Buttons. See [Figure 4-9](#).

Parameter: See the any parameter in the orange circles in [Figure 4-9](#).

Description: After any of the following keys is programmed with **DTMF Keypad** feature, you can press this key to enable or disable the DTMF Keypad feature. With the DTMF Keypad feature enabled, radio users can input the phone number via the keypad in the home screen. Under such condition, the Connect Code button and Disconnect Code button are valid.

📌 Note: Radio users can also enable the DTMF Keypad feature by going to “Menu -> Phone -> DTMF Keypad”.

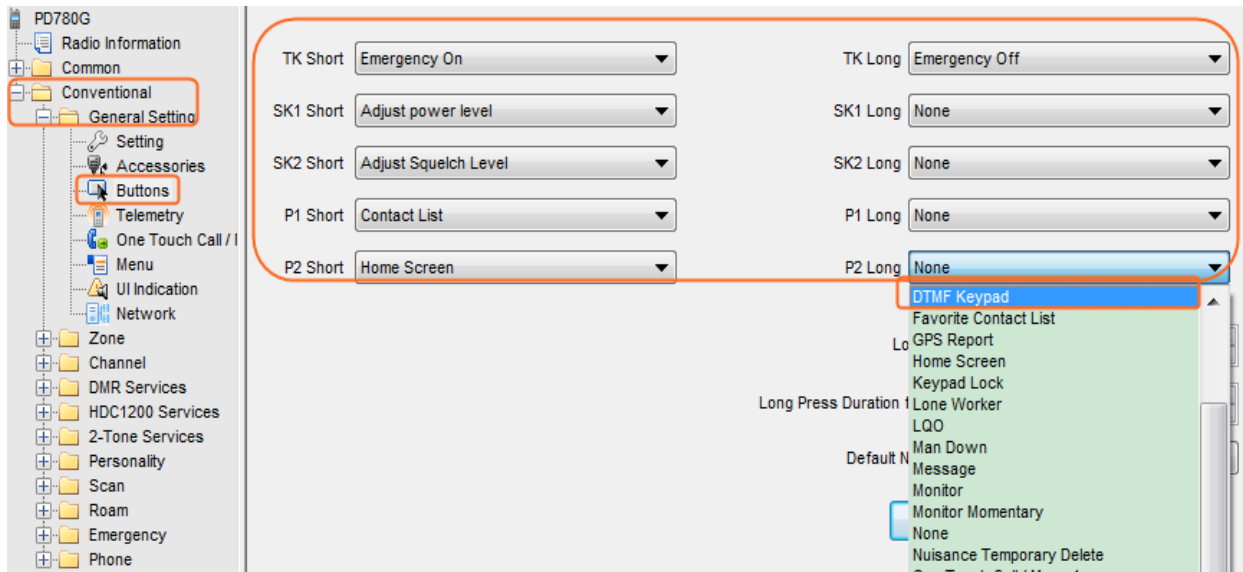
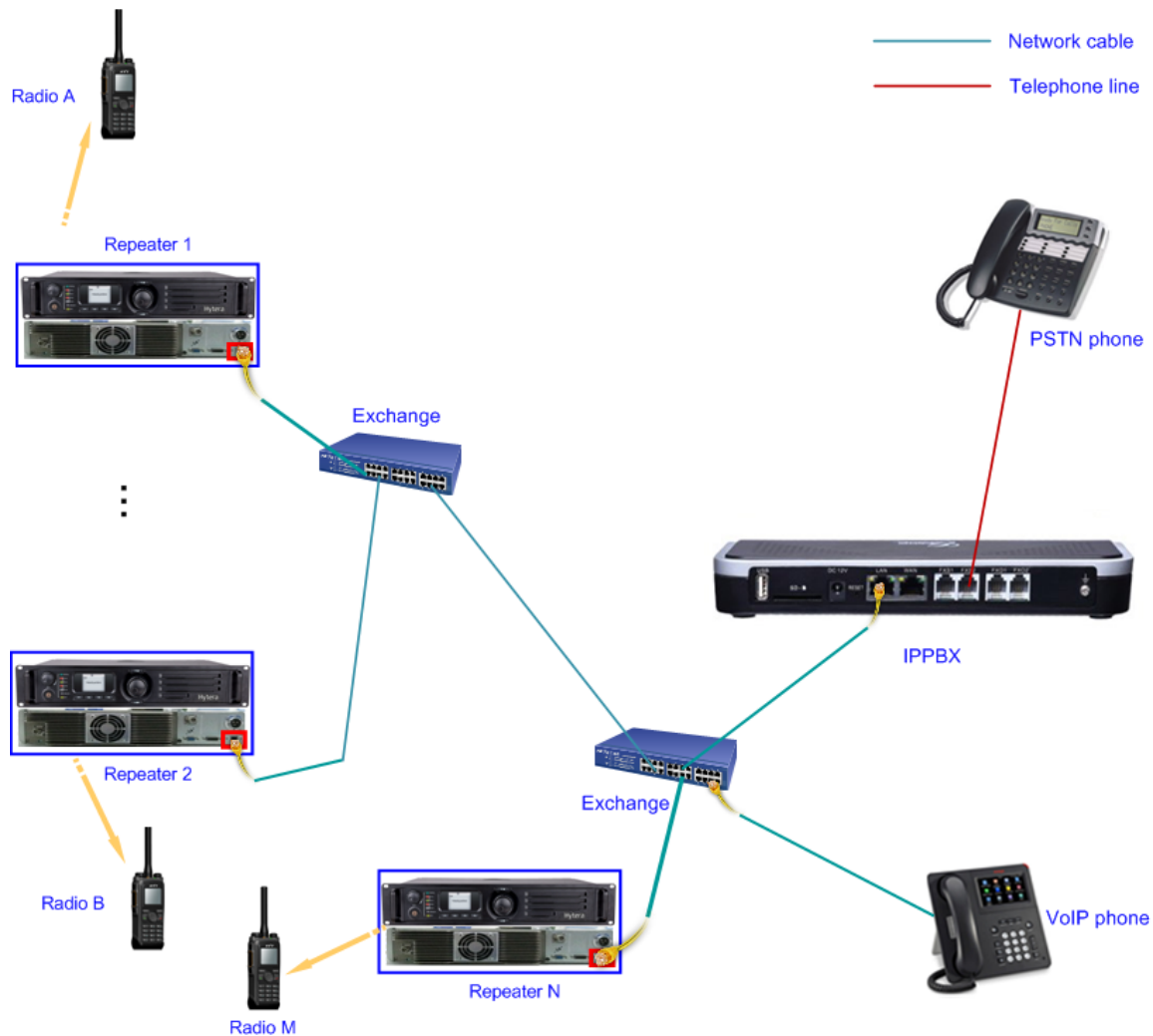


Figure 4-15 DTMF Keypad Programmable Keys Configuration Interface

4.4 Network Connection

4.4.1 Diagram of Device Connection

The repeaters are connected to the IPPBX device via one or more exchanges first, and then they are connected to the telephone devices via the IPPBX device. In this way, multiple communication devices from different locations are connected together to build a complete and sound SIP phone system. In this SIP phone system, communication devices such as radios, PSTN phones and VoIP phones can communicate with each other.



4.4.2 Device Connection Instructions

- The radio and repeater transfer data to each other via air interface protocol.
- IPPBX and PSTN phone are connected to each other via the telephone interfaces (FXO interface for external public telephone network connection; FXS interface for internal telephone devices connection) and transfer voice data via the telephone lines.
- IPPBX and VoIP phone are connected to each other via the Ethernet interface.
- Other devices are connected via Ethernet interface and transfer voice data via network cable.
- Please refer to corresponding references or consult the device operators for detailed information of different devices.

5. Communication Procedure

5.1 Dialing Example

The radios make calls to the telephone devices by accessing the telephone network and use the dialing rules specified by the telephone network. Thus, the call procedure of radios calling telephone devices is the same as that of telephone devices calling each other. In such case, the radios can be considered as telephone devices. When the radios and called telephone devices are within a same LAN, the calls between them are internal calls. When the radios and called telephone devices are not within a same LAN, the calls between them are trans-regional calls.

For example, the dialing rule of Company A is as follows:


- Internal calls: Internal phones can call each other directly.
- Trans-regional calls: Add “0” affront of the external called number.


In such case,

- If the radio user needs to call the internal telephone devices, he can input the extension number of the called party and hold down the **PTT** key to initiate a call.
- If the radio user needs to call the external telephone devices, he can add “0” affront of the external called number and hold down the **PTT** key to initiate a call.
- If the internal telephone user needs to call the internal radios, he can input the radio called number to make a call directly (for Dial Scheme, please refer to [6.1 Radio Called Number](#)).
- If the external telephone user needs to call the internal radios, he can dial the phone number of Company A first, and then input the called number following the instructions to make a call.

5.2 Radios Calling Telephone Devices

The procedure of a radio making phone calls to a telephone device is as follows:

 **Caution:** Radios can only make private call instead of group call to telephone devices. But the telephone devices can make both private call and group call to the radios. Please keep the DTMF Keypad mode enabled during calls for proper call operation.

Precondition: Press the programmed **DTMF Keypad** key or go to “Menu -> Phone -> DTMF Keypad” to enable the DTMF Keypad mode. The radio will display the DTMF Key icon  in the home screen.

Step 1 Input the phone number (for Dial Scheme, please refer to [6.2 Phone Called Number](#)) or select a preset contact. Do as follows:

- Input the phone number using the numeric keypad directly in the home screen
- Go to “Menu -> Phone -> Manual Dial” and input the phone number
- Go to “Menu -> Phone -> Phone List” and select a preset contact



Note: The radio supports two dialing methods: Buffer Dial and Live Dial. Here takes Buffer Dialing for example.

- Buffer Dial: Operates like the dialing on the mobile phone. The user inputs the complete phone number string on the keypad, and then presses the **PTT** key to call.
- Live Dial: The user inputs the complete phone number string on the keypad when holding down the **PTT** key. The radio will make a call to the input number after certain duration (configurable via CPS).

Step 2 Hold down the PTT key to initiate a phone call to the input number or preset contact.



Note: Here instructs the DTMF Buffer Dial method which is similar to mobile phone dialing. Radio user can also employ the DTMF Live Dial method which is to hold down the **PTT** key and input the phone number at the same time.

Step 3 The radio waits the called phone to answer.

The called phone will give incoming call alerts. The phone users may be required to input extension number or account and password during call establishment. Please input the numbers via the DTMF keypad following the instructions.

Step 4 When the called party answers the call, the call is established successfully.

Step 5 Radio user can hold down the PTT key to talk.

During communication, the radio is operating in simplex mode, which indicates the radio cannot transmit and receive at the same time. If a radio user holds down the **PTT** key to transmit forcibly when the radio is receiving voice, the radio cannot receive the voice from the repeater.

Step 6 When the communication is done, radio user can press the Disconnect Code button and hold down the PTT key to send the preset Disconnect Code. Then the ongoing phone call will be terminated.

**Note:**

If the Disconnect Code button is not programmed, the radio user will have to enter the Disconnect Code via the numeric keypad manually and then hold down the **PTT** key to send it.

Radio user cannot terminate the ongoing phone call, but radio user can stop receiving the phone call by switching to another channel.

5.3 Telephone Devices Calling Radios

The procedure of a telephone device making phone calls to a radio is as follows:

Step 1 Pick up the phone.

Step 2 Input the radio called number (for Dial Scheme, please refer to [6.1 Radio Called Number](#)) to make a call to the radio.



Caution: The radio cannot receive phone calls when it is on analog channel.

Step 3 The telephone device waits the called radio to answer.

The called radio will give incoming call alerts.

- When the called radio receives a phone call, the radio user can listen to the call without any operation.
- When the called radio receives a private phone call, the radio user can do as follows:
 - Press the Connect Code button and hold down the **PTT** key to send the preset Connect Code. Then the radio will answer the phone call.
 - Press the Disconnect Code button and hold down the **PTT** key to send the preset Disconnect Code. Then the radio will reject to answer the phone call.

Step 4 When the radio answers the phone call, the phone call is established successfully. If the radio rejects to answer the phone call, the phone call will not be established and it is terminated.

During communication, the telephone device operates in duplex mode which indicates the device can transmit and receive at the same time like the calls between telephones.

Step 5 When the talk is over, the call initiator can end the call by hanging up.

Radio user can also terminate the ongoing private phone call by sending the Disconnect Code.

6. Dial Scheme

6.1 Radio Called Number


When the telephone device makes a phone call to the radio, phone user inputs the Called Number using the numeric keypad directly.

- If Dial-up Mapping feature is not enabled, the dial scheme of the Called Number is as follows:

Called Number = Call Type + Slot# + Target ID

Dial scheme instructions:

Parameters	Description	Range
Call Type	Sets the type of calls made to the radio, namely the “Call Type” under Phone Call Configuration . 1 indicates private call and 2 indicates group call.	1 or 2
Slot#	Sets the slot in which the repeater transmits and receives voice signals, namely the “Slot ID” under Phone Call Configuration . 1 indicates Slot1 and 2 indicates Slot2.	1 or 2
Target ID	The private call ID or group call ID of the radio, namely the “Target ID” under Phone Call Configuration . The value of which is related to the call type.	1-16776415

 **Caution:** The Radio Called Number must be consistent with the phone number configured on the IPPBX device; otherwise the call may be made to a wrong called party.

- If Dial-up Mapping feature is enabled, the dial scheme of the Called Number is as follows:

Called Number = Phone ID

Dial scheme instructions:

Parameters	Description	Range
Phone ID	Sets the called number of the radio when the phone makes a call to the radio, which is the Phone ID in phone contact list. The Phone ID must be unique and any one parameter	1-4294967295

Parameters	Description	Range
	<p>(Radio ID, Slot# and Call Type) corresponding to each Phone ID must be unique. That is to say, at least one of the Radio ID, Slot# and Call Type corresponding to each Phone ID must be unique.</p> <p>With Dial-up Mapping feature enabled, if you need to keep the original dial scheme and extension number of IPPBX, you just need to set the Phone ID in repeater according to the original dial scheme, that is to set the Phone ID as Call Type + Slot# + Radio ID.</p>	

For example, the Radio ID, Slot# and Call Type of Radio A is 3001, Slot1 and Private Call respectively.

- If Dial-up Mapping feature is not enabled, you can only make a call to Radio A by dialing 113001.
- If Dial-up Mapping feature is enabled, you can make a call to Radio A by dialing Phone ID. For example, you can set the Phone ID of Radio A to extension number 23. Afterwards, you can dial 23 on the telephone to make a call to Radio A. If you need to keep the previous extension number 113001, you can set the Phone ID to 113001. When you use the telephone to make a call to Radio A, the number will be the same as before.

6.2 Phone Called Number

When a radio makes a phone call to the telephone device, radio user can input the phone called number using the numeric keypad directly. Please refer to 5.1 Dialing Example for detailed dial schemes.

7. FAQ

Q: When the telephone device makes a call to the radio, the radio gives no response, and the telephone device gives alert such as “the number you dialed is unreachable, please check the number and dial later”?

A: This situation may be caused by unsuccessful registration of the called radio. The inconsistency between the repeater PBX Connect Code and SIP extension password set on IPPBX will cause unsuccessful registration. Please make sure these two parameters are consistent.

Q: When the radio makes a call to the telephone device, the radio displays call end after dialing?

A: This situation may be caused by unsuccessful registration of the radio. See the answer above for solutions.

Q: When receiving a call from the telephone device, the radio still rings after inputting the Connect Code?

A: This situation may be caused by incorrect Connect Code. Make sure the Connect Code of the radio is consistent with that of the repeater.

Q: The radio cannot terminate a phone call?

A: When receiving a private phone call, make sure the Disconnect Code of the radio is consistent with that of the repeater; when receiving a phone call, the radio cannot terminate the phone call.

Q: The radio fails to make second dialing?

A: This situation may be caused by the following reasons: Network delay. When the delay is significant, the communication between the repeater and IPPBX will be affected. Make sure the network is working properly; Check if the value of the parameter Digital DTMF Tx Gain is larger than 4.

Q: When the telephone device makes a call to the radio and hangs up before the radio answers, the telephone device still rings one more time?

A: This situation may be caused by IPPBX detection. When the telephone device hangs up in a short time or too quickly, IPPBX detection may be inaccurate. Generally, hanging up the telephone device again can solve this issue.

Q: When the telephone device makes a phone call, the radio can receive voice signal from the telephone device but the telephone device cannot receive voice signal from the radio?

A: This situation may be caused by setting the Third Party Connect Mode of the repeater to Selective. In this mode, the repeater will not repeat the voice signal to the telephone device, therefore the telephone device cannot receive voice signal from the radio.

Q: In the SIP phone network, each repeater connects with each other via the IP multi-site connect. When a radio makes a phone call to the telephone device, the radio cannot terminate the phone call?

A: This situation may be caused by improper configuration of the home repeater of the radio. If the home repeater of a radio does not support the Phone feature, the radio can make a phone call using other repeaters with the Phone feature via the IP multi-site connect. In this case, the Repeat Request Priority of the home repeater must be set to "Local Repeating". CPS configuration path: Edit -> Conventional -> General Setting -> Accessories -> Priority Control -> Repeat Request Priority.