



# SmartPTT Tutorial

## Telephone Interconnect

# Telephone Interconnect Overview

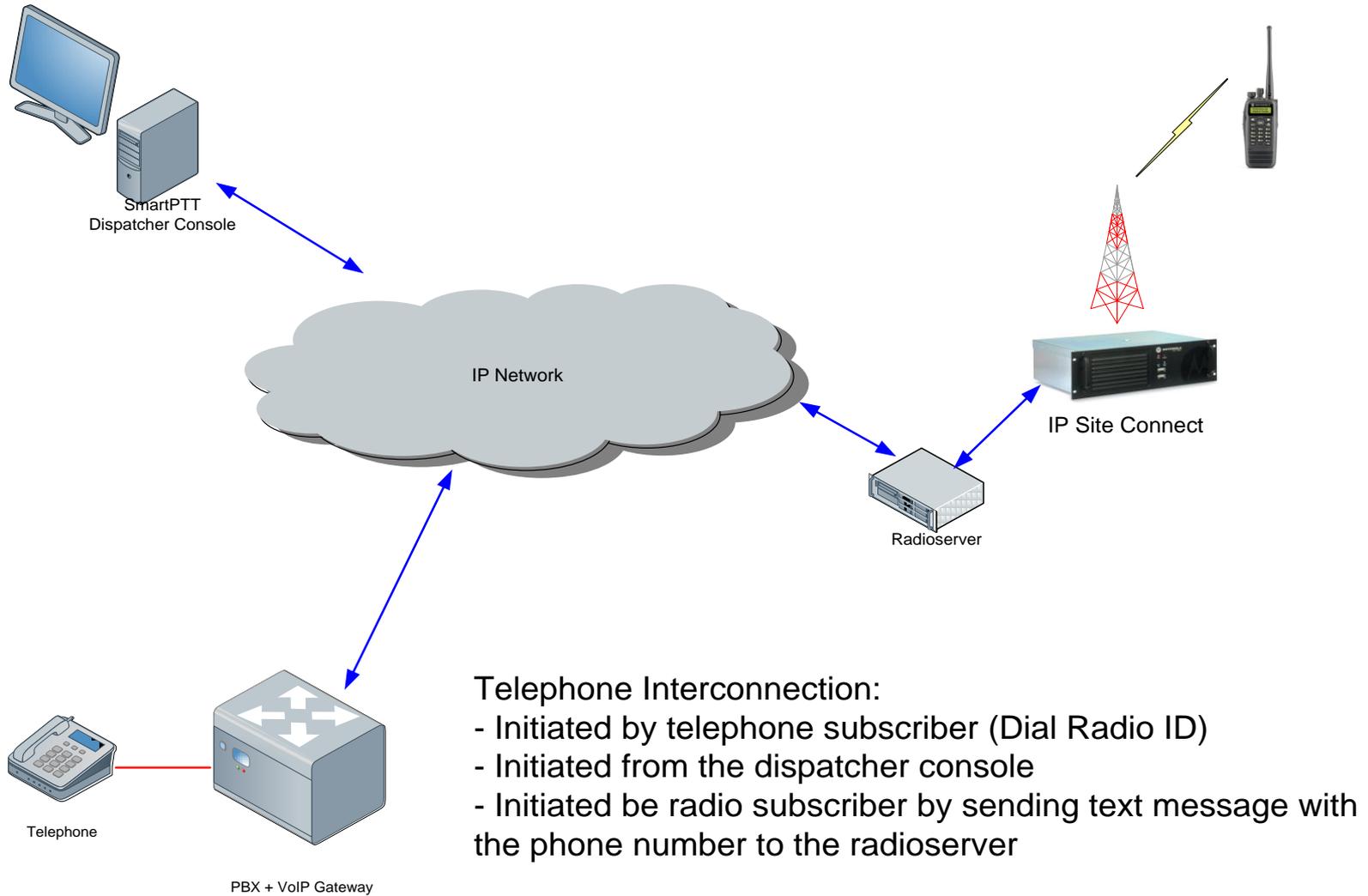
Telephone Interconnect is the service included into SmartPTT Radioserver to establish interconnection between Radio and Telephone Subscribers.

For communication with telephone subscribers Telephone Interconnect Service connects to VoIP Gateway having SIP trunk service support by means of SIP protocol. VoIP Gateway must be either embedded into PABX system or connected by means of digital or analog channels.

For communication with radio subscribers Telephone Interconnect Service uses control stations or repeaters connected to Radioserver.

To enable Telephone Interconnect functionality it is required to get telephone interconnect license per each Radioserver.

# Telephone Interconnect Architecture



# VoIP Gateway Requirements

VoIP gateway provides the conversion interface between the public switched telephone network and an IP network for voice and fax calls. The best choice is native VoIP gateways integrated into PBX as they provide maximum flexibility and performance.

In case you have PBX without integrated VoIP gateway, it is possible to use separate gateway, connected to PBX by standard interfaces: digital (e.g. ISDN PRI, ISDN BRI) or analog (FXS, FXO, E&M). Digital interfaces are preferred to analog ones, as they provide more full-functional signaling features (Caller ID, Dialed Number Identification Service (DNIS), Hang-up signals). FXS and FXO interfaces support DNIS and reliable hang-up signaling just in one way, so it's recommended to combine them for transferring calls in opposite directions (FXS/FXO gateway) or use the Direct Inward Dialing (DID).

# Types of calls

- **Telephone Interconnect initiated by telephone subscriber.** Telephone subscriber dials Prefix + Radio ID to do the call. PABX and VoIP Gateway must be programmed correspondingly.
- **Telephone Interconnect initiated by radio subscriber.** Radio subscriber sends text message with the phone number to Radioserver to process. Phone number can have prefix to identify that text message is the phone number, but not an ordinary text.
- **Telephone Interconnect initiated by Dispatcher Console.** Radio and Telephone Subscribers can be patched from Radio Subscriber Call Window in SmartPTT Dispatcher Console.

# Telephone call from SmartPTT Dispatcher Console

**SmartPTT Dispatcher has a built-in functionality to make telephone connections to telephone subscribers and vice versa.**

The following **connection options** are possible:

- SmartPTT dispatcher calls office PBX subscriber
- Office PBX subscriber calls SmartPTT dispatcher

In both cases connections and addressing are implemented via SIP protocol. Voice is transmitted in full-duplex mode via RTP protocol.

A valid unique SIP address for each telephone subscriber must be assigned in SmartPTT Dispatcher in accordance with the telephone subscriber's phone number.

# Phone calls from SmartPTT Dispatcher – SIP addressing system

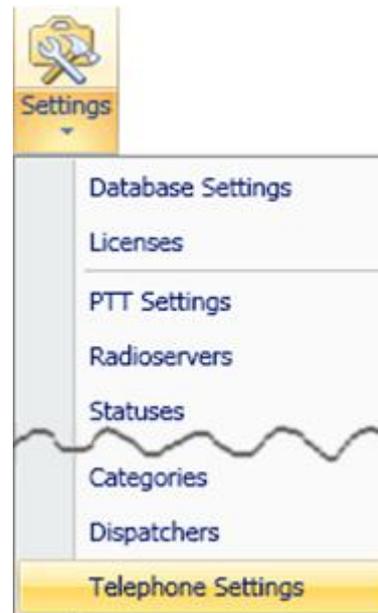
SIP determines the following addressing system for every subscriber:

***name@address:port***

- ***address*** is the IP address of the device (VoIP gateway for telephone subscribers, IP address of SmartPTT dispatcher PC for dispatcher).
- ***port*** is a UDP port that is used by device to receive and send commands, port 5060 is used by default.
- ***name*** is a subscriber ID which can be interpreted in various ways:
  - for *telephone subscribers* names consist of an optional numerical prefix and the subscriber's phone number.
  - for *SmartPTT dispatcher* the name is not important. While the VoIP gateway is being configured, it must be set to 1 (1@dispatcher\_address:port).

# Phone calls from SmartPTT Dispatcher – Configuring Telephone Settings

To configure connection parameters with telephone subscribers, expand the **Settings** list in the main menu of the SmartPTT Dispatcher window and click **Telephone Settings**.



The Configuration window will be displayed for configuring connection parameters and a list of telephone subscribers (presented on next slide).

# Phone calls from SmartPTT Dispatcher – Telephone Settings Configuration dialog

**Configuration** [X]

**Telephone Settings**  
Settings and Subscriber List for Telephone Interconnect



**Dispatcher Settings**

Allow Calls to Telephone Subscribers  Interface 192.168.0.54

Incoming Sound Port 18600 - 18650 SIP Port 5060

**VoIP Gateway Default Settings**

Dial Prefix for Outgoing Calls 12

Dial Prefix for Incoming Calls

Default VoIP Gateway Address 192.168.0.243 SIP Port 5060

Telephone Subscriber	Address
Andrew	1179
Denis	1180
John	1185
Nick	1184
Stephen	1170

Play Sound Alert for Incoming Call

Play Sound Alert for Outgoing Call

Buttons: Add, Edit, Delete, Help, Finish

# Phone calls from SmartPTT Dispatcher – Configuring Telephone Settings

## Dispatcher Settings section

This section specifies the dispatcher's local computer settings.

**Allow Calls to Telephone Subscribers** – enables the feature for this dispatcher.

**Incoming Sound Port** – the range of ports to use for voice data transferring.

Each connection with the telephone subscriber requires two ports (e.g. the first connection will use UDP port 18600 and TCP port 18601, the second will use UDP port 18602 and TCP port 18603 and so on). Thus, the specified ports must be available both for incoming and outgoing traffic.

**Interface** – IP address at the local computer, used to receive and transmit commands and voice data. If dispatcher's computer has only one IP address, the system may select it automatically; if it has several IP addresses, the IP address (interface) should be specified expressly. The specified IP must be available from the VoIP gateway end.

**SIP Port** is a UDP port to receive and send commands to connect to telephone subscribers. The default value is 5060. Accordingly, the specified port must be available for incoming and outgoing traffic.

**Please note:** If several network interfaces are presented on the PC you may have to configure the routing table using the command "route add ...".

# Phone calls from SmartPTT Dispatcher – Configuring Telephone Settings

## VoIP Gateway Default Settings section

If most of the telephone subscribers the SmartPTT dispatcher is connected via the same VoIP gateway, it is more convenient to specify default values in this section. These settings allow to register telephone subscribers by specifying only the internal phone number.

**Default VoIP Gateway Address** – the gateway's IP address.

**SIP Port** – a VoIP gateway port to receive and send SIP commands. By default VoIP gateways use the UDP port 5060.

**Dial Prefix for Outgoing/Incoming Calls** – default prefix number for outgoing/incoming calls in accordance with VoIP Gateway settings.

### **Notes:**

1. If the default VoIP gateway is used, the telephone subscriber's SIP address for outgoing calls is automatically composed with the following rule:

SIP address = *Dial Prefix for Outgoing Calls* + Phone Number + @ + *Default VoIP Gateway Address* + “:” + *SIP Port*

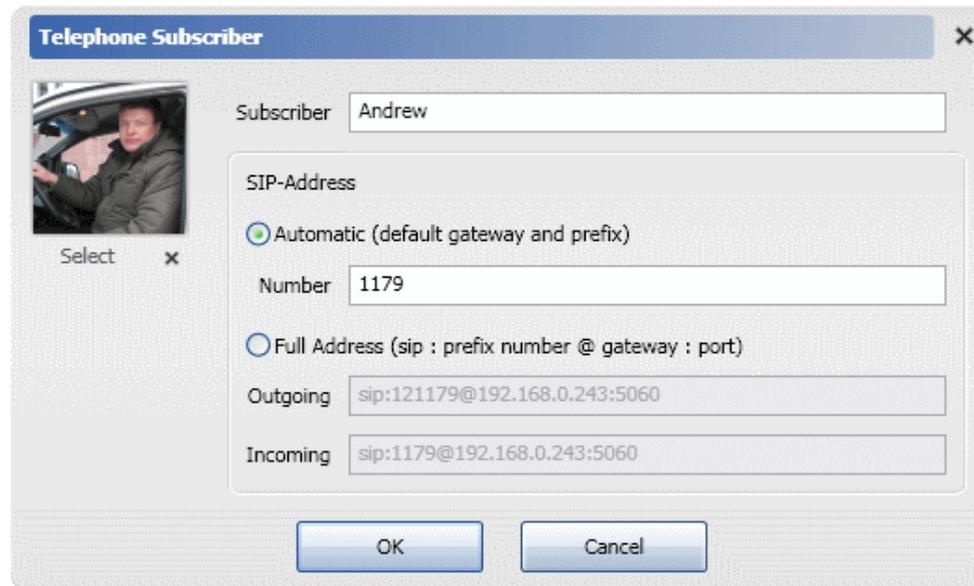
For example, with Default VoIP Gateway Address = 192.168.0.243, SIP Port = 5060, Dial Prefix for Outgoing Calls = 12, system will assume SIP address *121140@192.168.0.243:5060* for telephone subscriber with the phone number 1140.

2. When the SmartPTT dispatcher is called by the telephone subscriber, the subscriber is identified by the incoming SIP address, which can be different from the outgoing one depending on the VoIP gateway settings. That is why a separate prefix for incoming addresses is set in the system.

# Phone calls from SmartPTT Dispatcher – Configuring Telephone Settings

## Telephone Subscribers list

The SmartPTT dispatcher stores a list of telephone subscribers. You can manage the list using **Add**, **Edit** or **Delete** buttons. Telephone subscriber properties are managed in separate window.



The screenshot shows a window titled "Telephone Subscriber" with a close button (X) in the top right corner. On the left side, there is a small image of a man in a car, with a "Select" button and a close button (X) below it. The main area contains the following fields and options:

- Subscriber:** A text box containing the name "Andrew".
- SIP-Address:** A section with two radio button options:
  - Automatic (default gateway and prefix)
  - Full Address (sip : prefix number @ gateway : port)
- Number:** A text box containing the number "1179".
- Outgoing:** A text box containing the SIP address "sip:121179@192.168.0.243:5060".
- Incoming:** A text box containing the SIP address "sip:1179@192.168.0.243:5060".

At the bottom of the window, there are two buttons: "OK" and "Cancel".

In this window you can enter the subscriber name and choose a picture to display in call window. To specify the SIP address you can select among two options:

- Specify just the phone number (default VoIP gateway settings will be used).
- Enter both outgoing and incoming SIP address manually in full form.

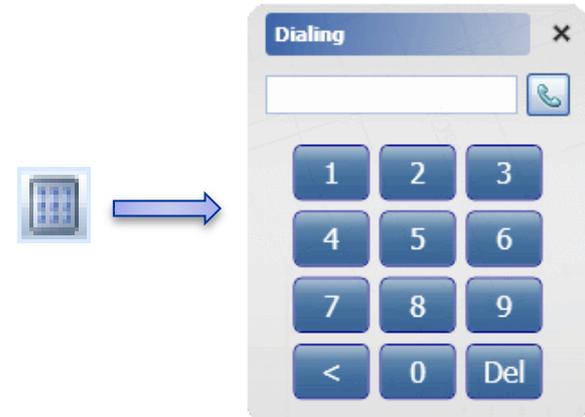
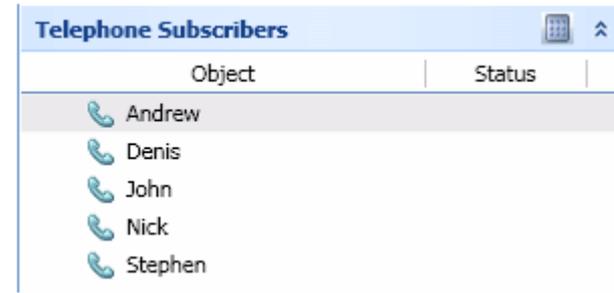
# Phone calls from SmartPTT Dispatcher – Telephone Subscribers access pane

If the settings allow connections to telephone subscribers, a telephone subscribers access pane is available in the SmartPTT Dispatcher main window.

Double click in the Telephone subscribers pane to open the Telephone Subscriber call window.

To call the subscriber click **Connect**. To end call click **Disconnect**.

The **Telephone Subscribers** pane has a speed-dial button (at top-right corner) to call the subscriber by phone number. In this case default VoIP gateway settings are used to generate the full SIP address.



# Phone calls from SmartPTT Dispatcher – Telephone Subscribers access pane

To display the telephone subscriber settings window, right-click the desired telephone subscriber

To set a sound for an incoming call alert, click the button “...” under corresponding section.

To display the telephone subscriber call window, click the button **Call Window**.



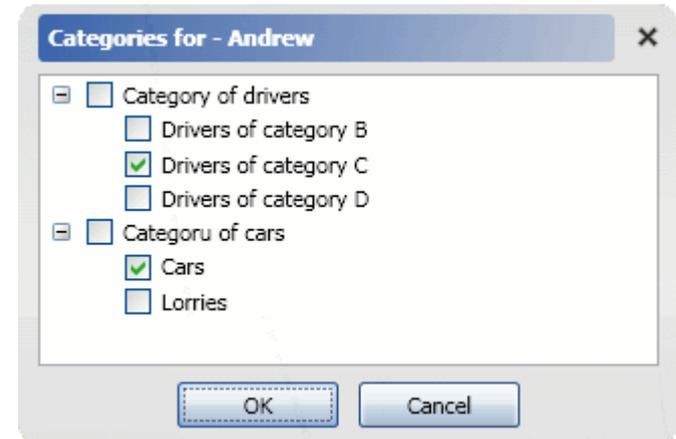
To assign the selected telephone subscriber to specific categories, click the button **Categories**.



Select desired categories in the displayed window and click **OK**.



The image shows a software window titled "Andrew" with a close button (X) in the top right corner. On the left, there is a small photo of a man in a car. To the right of the photo, there are two text input fields: "Name" containing "Andrew" and "Address" containing "1179". Below these fields is a "Select" button with a small X icon. Further down, there is a section for "Sound for Incoming Call Alert" with a list of sound icons and a "..." button to the right. At the bottom of the window are "Save" and "Cancel" buttons.



The image shows a software window titled "Categories for - Andrew" with a close button (X) in the top right corner. The window contains a list of categories with checkboxes. Under "Category of drivers", there are three sub-items: "Drivers of category B" (unchecked), "Drivers of category C" (checked), and "Drivers of category D" (unchecked). Under "Categori of cars", there are two sub-items: "Cars" (checked) and "Lorries" (unchecked). At the bottom of the window are "OK" and "Cancel" buttons.

# Telephone Interconnect – Radioserver Settings

**SmartPTT Radioserver has integrated Telephone Interconnect service which allows establishing of communication between radio and telephone subscribers.**

Telephone Interconnect service uses the following protocols:

- SIP – for signaling
- RTP – for voice data transmission

Voice data transmission is done in *half-duplex mode*.

There are 3 ways to initiate the interconnection:

- Telephone subscriber initiates a call.
- Radio subscriber initiates a call.
- Call is initiated from dispatcher console.

Select **Telephone Interconnect** item in Radioserver Configuration Window to set up Telephone Interconnect Service.

**Note!** Telephone Interconnect Service requires corresponding license to be installed at Radioserver. Licenses are installed under Licenses section of Radioserver Configuration tool.

# Telephone Interconnect – Radioserver Settings

The screenshot shows the 'SmartPTT Server Configuration' window with the 'Telephone Interconnect' settings panel selected in the left-hand tree view. The tree view includes sections for Radio Server, Radio Network Services, Add-on Modules, Control Stations, IP Site Connect Networks, and Profiles. The 'Telephone Interconnect' section is expanded, showing sub-items like Settings, Control Station 1, Channels, Groups, Audio, Slot 1, Slot 2, and Groups.

**Telephone Interconnect**

- Active

**Server Settings**

- Interface: 192.168.0.243
- SIP Port: 5060
- RTP Ports: 18600 - 18610
- Private Call Prefix: 22
- Group Call Prefix: 23

Allow Calls to Default VoIP Gateway

**Default Gateway Settings**

- Address: 28.200.144.46
- SIP Port: 5060
- Source Prefix: 11
- Destination Prefix: 12

**Dialing Rules**

- Prefix: A

Service: Running

# Telephone Interconnect – Radioserver Settings

**Active** – Enable the Telephone interconnect feature for radioserver.

## Server Settings group

**Interface** – Local IP address which will be used for communication with VoIP Gateway.

*Note!* Radioserver IP used for communication with VoIP Gateway must be real one. NAT is not supported.

**SIP Port** – Local UDP port to receive and send SIP commands (default value: 5060).

*Note!* In case when Radioserver and Dispatcher Console are installed on the same PC and both of them are set up for telephone interconnection, SIP ports must be different for Radioserver and Dispatcher Console.

**RTP Ports** – Range of local ports used for voice data transmission to/from VoIP-Gateway (default values: 18600 – 18610).

**Private/Group Call Prefix** – If the name part in SIP address, received by radioserver from VoIP gateway, starts with this prefix, radioserver treats the rest of name as Radio ID/Group ID and performs private/group radio call to it. In case they are not set, all calls are treated as private.

# Telephone Interconnect – Radioserver Settings

**Allow Calls to Default VoIP Gateway** – enable the possibility to use default VoIP gateway settings to build the full SIP addresses in case just phone number was specified, when radio subscriber initiates call.

## Default Gateway Settings group

**Address** – network IP address of default VoIP Gateway.

**SIP Port** – SIP port which of VoIP gateway (default value: 5060).

**Source/Destination Prefix** – default prefix number, which will be added in front of radio ID/calling phone number for outgoing calls.

## Dialing Rules group

**Prefix** – This prefix is used to determine telephone call request. If text message sent to Radioserver starts with this prefix, Telephone Interconnect Service treats the rest of message as phone number and compose the following SIP addresses for outgoing call:

**From:** *[Default Gateway Settings-Source Prefix][Radio ID]@[Server Settings-Interface]:[Server Settings-SIP Port]*

**To:** *[Default Gateway Settings-Destination Prefix][Phone Number]@[Default Gateway Settings-Address]: [Default Gateway Settings-SIP Port]*

# Telephone Interconnect – Radioserver Settings

The screenshot displays the 'SmartPTT Server Configuration' application window. The title bar reads 'SmartPTT Server Configuration - C:\Program Files\SmartPTT\Server\RadioSe...'. The interface is split into two main panes. The left pane is a tree view showing the server configuration hierarchy: Radio Server, Radio Network Services (ARS, Location, TMS, Telemetry), Add-on Modules (Licenses, Event Log, Settings), Telephone Interconnect (Settings), Control Stations (Control Station 1), Channels, Groups, Audio, IP Site Connect Networks (IP Site Connect 1, Slot 1, Slot 2, Groups), and Profiles (Group 1, Group 2, All Call). The 'Settings' option under 'Telephone Interconnect' is selected. The right pane is titled 'Telephone Interconnect settings' and contains the following configuration options:

- Allowed Subscriber List
- Call Request Timeout: 300 s
- Telephone Ring Timeout: 20 s
- Radio Subscriber Call Timeout: 20 s
- Play Sound to Telephone on Connection
- Play Warning Beeps on Busy Channel
- Call Start Notification for Subscriber Radio
- Call End Notification for Subscriber Radio

At the bottom of the window, there is a status bar with a question mark icon, a play button, a blue square icon, a blue square icon with a right arrow, the text 'Service: Running', and three icons: a green circular arrow, a document with a right arrow, and a blue square with a red notification icon.

# Telephone Interconnect – Radioserver Settings

Additional Telephone Interconnect Settings:

- To limit the radios authorized to use telephone interconnect service, set **Allowed Subscriber List** checkbox and add *permitted* radios to the list.
- **Call Request Timeout** – Period of time, during which request from a radio (text message with a phone number sent to radioserver) will stay in queue waiting for pending calls.
- **Telephone Ring Timeout** – Time to wait for telephone subscriber to answer the call.
- **Radio Subscriber Call Timeout** – Time to wait for connection from telephone subscriber to radio subscriber when all available radio channels are busy.
- Also you can enable or disable sound notifications about events for phone and radio subscribers.

# Telephone Interconnect – General VoIP Gateway settings

SmartPTT Telephone Interconnect Service connects to VoIP Gateway by means of Session Initiation Protocol (SIP) in SIP Trunk connection mode. To transfer the voice data, G.711 codec is fully supported. Also it is possible to implement G.723 and G.729 support, but additional licensing is needed.

Here are some examples how to configure your VoIP gateway.

## Cisco, AddPac:

```
dial-peer voice 10 voip
 destination-pattern 22.      // all numbers beginning with 22
 session protocol sipv2
 session target ipv4:192.168.100.3 // Radioserver IP address
 codec g711alaw
!
```

# Telephone Interconnect – General VoIP Gateway settings

Using GUI of IPGear VoIP Gateway:

Radioserver IP

Mask number

SIP

Third-party Gateways Table

Change Gateway

IP: 192 168 0 164

H.323

Session Initiation Protocol (SIP)

Enter the IP address of the third-party Gateway.

OK Cancel

Mask	Description
14nn	

Prioritize...

Add... Change... Remove...

OK Cancel Help

To configure Third-party Gateways:  
1. Click the left Add button or select a Gateway and click Change.  
2. Enter its IP address.  
3. Select the IP address and click the right Add button.

# Voice call from telephone to radio – General overview

To successfully make a voice call from telephone set to radio, the following requirements must be satisfied:

- The number, which is typed by telephone subscriber, is routed from PBX to VoIP gateway.
- Then the correct SIP address is generated by VoIP gateway and routed to SmartPTT Radioserver.
- Radio subscriber is on-line and available.

On every stage the dialed number can be modified along with settled rules:

- PBX can add or remove prefix in dialed by phone subscriber number.
- VoIP gateway can choose one of different radioservers based on prefix.
- Radioserver can be configured to removed unnecessary prefixes.

Also on every stage it's possible to modify information about caller number (Automatic Number Identification/Caller ID (ANI/CID)). In case of using analog channels, all ANI/CID information will be lost.

Radio subscribers are addressed by Radio ID, so PBX must be programmed to allocate number pool corresponding to Radio IDs.

# Voice call from telephone to radio

To make a call from Telephone to Radio subscribers, it is necessary to do preliminary settings on both PBX, VoIP-Gateway and Radioserver. PBX must be programmed to allocate number pool corresponding to radio IDs (group IDs in case of group calls). For example, all radio subscribers (Radio ID= XXX) are assigned with telephone numbers = 51XXX and all radio groups (Group ID= NNN) are assigned with telephone numbers = 52NNN. PBX is programmed in the way when all calls to numbers beginning with 5 are forwarded to VoIP-Gateway. Then you can configure your systems in two different ways, which differs by the stages, where different prefixes are removed.

1. This way is suitable when your VoIP Gateway supports number modification. VoIP Gateway should be programmed to extract destination number from initial telephone numbers (remove prefix 5 in our case) and compose destination SIP-address to make the call to Radioserver:

```
[Destination number]@[IP of Radioserver]:[SIP port of Radioserver]
```

Radioserver is configured to treat prefixes in destination number to differ individual and group calls (1 and 2 respectively in our case).

2. This way is suitable when your VoIP Gateway does not support number modification. So your program VoIP Gateway to simply route all calls to Radioserver with following SIP-address:

```
[Initial dest. number]@[IP of Radioserver]:[SIP port of Radioserver]
```

Radioserver is configured to treat prefixes in SIP-name to differ individual and group calls.

# Voice call from telephone to radio – Example configuration

## ***Example 1***

PBX is programmed to route to VoIP gateway all calls to numbers beginning with 5. VoIP Gateway is programmed to route all these calls from PBX to Radioserver and remove 5 in front of number.

Telephone Interconnect Radioserver settings: Interface = 192.168.0.54, SIP Port = 5060, Private Call Prefix = 1, Group Call Prefix = 2.

To call the radio with Radio ID = 123, telephone subscriber dials 51123, PBX route this call to VoIP gateway, VoIP gateway composes SIP address = 1123@192.168.0.54:5060 and initiates session with Radioserver. Radioserver recognize the private call prefix (1) in front of SIP name and make private call to radio with Radio ID = 123.

The same for group call. To call all radios within group with Group ID = 33, telephone subscriber dials 5233, PBX route this call to VoIP gateway, which composes SIP address = 233@192.168.0.54:5060 and initiates session with Radioserver. Radioserver recognize the group call prefix (2) in front of SIP name and make group call to group with Group ID = 33.

## ***Example 2***

PBX is programmed to route to VoIP gateway all calls to numbers beginning with 5. VoIP Gateway is programmed to route all these calls from PBX to Radioserver.

Telephone Interconnect Radioserver settings: Interface = 192.168.0.54, SIP Port = 5060, Private Call Prefix = 51, Group Call Prefix = 52.

To call the radio with Radio ID = 123, telephone subscriber dials 51123, PBX route this call to VoIP gateway, VoIP gateway composes SIP address = 51123@192.168.0.54:5060 and initiates session with Radioserver. Radioserver recognize the private call prefix (51) in front of SIP name and make private call to radio with Radio ID = 123.

The same for group call. To call all radios within group with Group ID = 33, telephone subscriber dials 5233, PBX route this call to VoIP gateway, which composes SIP address = 5233@192.168.0.54:5060 and initiates session with Radioserver. Radioserver recognize the group call prefix (52) in front of SIP name and make group call to group with Group ID = 33.

# Voice call from radio to telephone

To make a call to telephone subscriber, radio subscriber needs to send text message with the phone number to base radio station. Text message is processed by Radioserver in accordance with settings done at Radioserver Configuration window and transferred to VoIP Gateway.

## In Radioserver Configurator:

1. Check *Allow Telephone Interconnect* checkbox for every Control Station which can be used for Telephone Interconnect.
2. Check *Allow Calls to Default VoIP Gateway* and set *Default Gateway Settings* to process incoming text messages in accordance with *Dialing Rules*.
3. Set *Dialing Rules – Prefix*.

## For VoIP Gateway:

Configure gateway to translate incoming calls from SIP connection to PBX. If the digital lines are used to connect VoIP Gateway and PBX, it is possible that PBX will check the incoming Caller ID and allow calls only with CID in certain range. In this case the CID modification rules should be configured on gateway. If it is not possible, you can configure the *Source Prefix* in Telephone Interconnect *Server Settings* on Radioserver.

**Note!** Radio subscriber can also make a call to telephone subscriber by sending it's full SIP-Address to the Base Radio. For example: "sip:123@100.100.100.100:5061"

# Voice call from radio to telephone – Example configuration

## **Radioserver settings:**

Server Settings: Interface = 192.168.0.54, SIP Port = 5060

Default Gateway Settings: Address = 28.200.114.46, SIP Port = 5060, Source Prefix = 1, Destination Prefix = 12

Dialing Rules: Prefix = “A”

## **VoIP Gateway Settings:**

Gateway is programmed to transfer all calls to PBX, removing prefix 12 in front of Calling ID and add prefix 5 in front of CID.

To call to Phone Number = 743, Radio Subscriber with Radio ID = 123 sends text message “A743” to base radio. Telephone Interconnect service extracts phone number from the text message, add Destination Prefix and composes following SIP Addresses for calling to VoIP Gateway:

From: 1123@192.168.0.54:5060

To: 12743@28.200.114.46:5060

VoIP Gateway processes these SIP Addresses and transfer call to PBX, in which:

Caller ID: 51123

Calling ID: 743

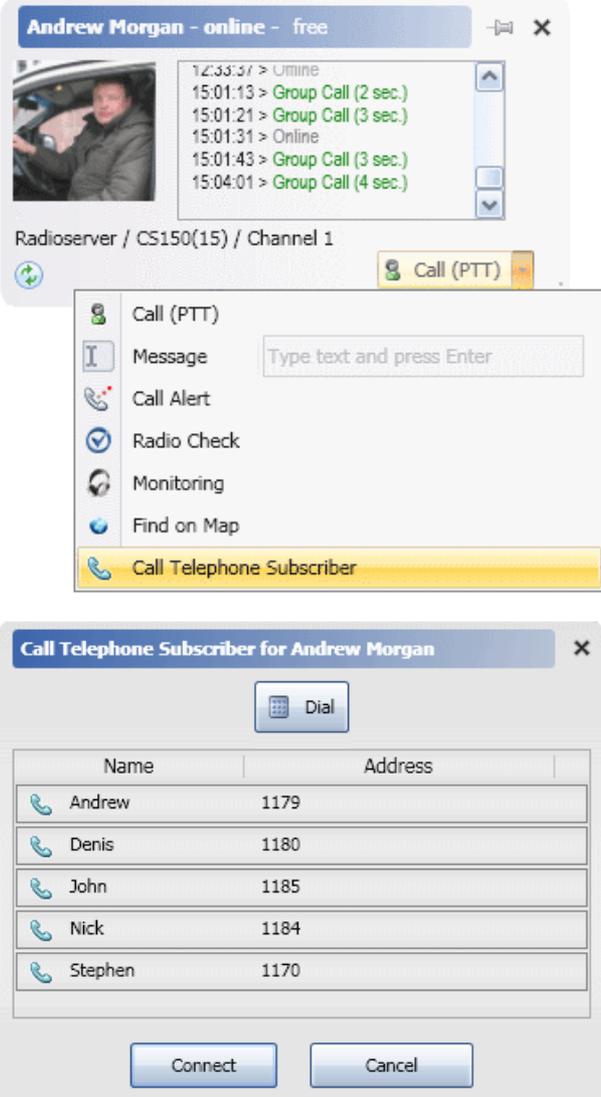
**Note!** In case your VoIP gateway does not support Caller ID/Calling ID modifications, you can obtain the same result if you modify the following Default Gateway Settings:

- Set Source Prefix to “51”.
- Empty Destination Prefix.

# Patch Radio and Telephone subscribers by Dispatcher Console

SmartPTT Dispatcher console has feature to patch radio and telephone subscribers. Voice call data is transferred by radioserver in this case, so license with enabled Telephone Interconnect Service is required to be installed on radioserver.

1. Open the **Call Window** of radio subscriber.
2. Select **Call Telephone Subscriber** menu item.
3. **Call Telephone Subscriber** window is displayed.
4. Select telephone subscriber from the stored telephone subscribers list and click **Connect** button (or double-click on the list item).
5. If necessary phone number is not in list, you can click **Dial** button to input the telephone number manually. In **Dialing** window, input the phone number and click  or press *Enter*.



The screenshot displays the 'Call Telephone Subscriber' window for Andrew Morgan. The window title is 'Andrew Morgan - online - free'. It shows a list of call events: 12:33:37 > Offline, 15:01:13 > Group Call (2 sec.), 15:01:21 > Group Call (3 sec.), 15:01:31 > Online, 15:01:43 > Group Call (3 sec.), and 15:04:01 > Group Call (4 sec.). Below the list, it indicates 'Radioserver / CS150(15) / Channel 1' and a 'Call (PTT)' button. A menu is open, showing options: Call (PTT), Message (with a text input field 'Type text and press Enter'), Call Alert, Radio Check, Monitoring, Find on Map, and Call Telephone Subscriber (highlighted in yellow). Below the menu, there is a 'Dial' button and a table of telephone subscribers.

Name	Address
Andrew	1179
Denis	1180
John	1185
Nick	1184
Stephen	1170

At the bottom of the window, there are 'Connect' and 'Cancel' buttons.

# Patch Radio an Telephone subscribers by Dispatcher Console

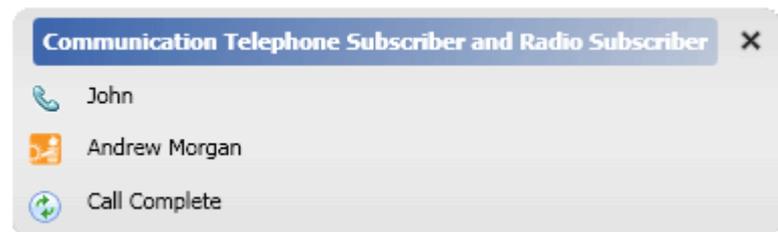
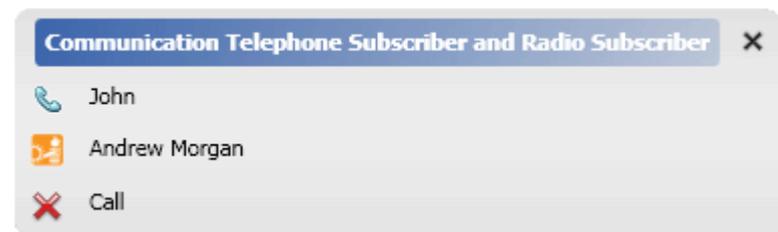
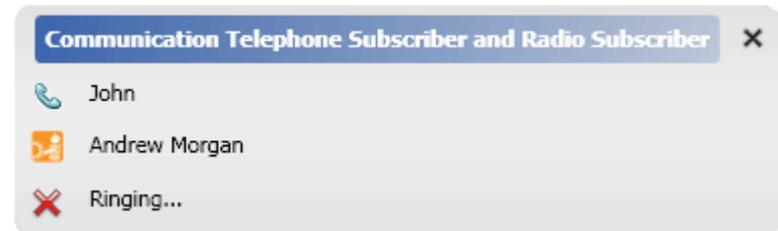
During dialing process status window will be displayed showing current state of the call.

When telephone subscriber picks up the phone, connection status is changed from Ringing to Call.

If the connection attempt is failed, status window shows corresponding error message.

To disconnect current connection click  button.

To establish the same connection again, click  button.



# Tested environments

There is no any additional system requirements for Telephone Interconnect Service except listed in *VoIP Gateway Requirements* section. But please note that the overall efficiency of the system consisting of equipment by different manufactures depends on the various factors. The main of them are the integration ability of different parts and possibility to implement unified numbering plan for PBX and radio subscribers.

Equipment, listed in this list, was tested for compatibility and worked with SmartPTT telephone interconnect.

## **VoIP Gateways:**

- ✓ Cisco integrated services routers (recommended)
- ✓ AddPac VoIP gateways
- ✓ IPGear VoIP gateways
- ✓ Audiocodes Mediant & Mediapack VoIP gateways

## **PBX:**

- ✓ Cisco CallManager
- ✓ Tadiran Coral
- ✓ LG iPECS
- ✓ Asterisk (and all Asterisk-based PBX)



## Elcomplus, LLC



Frunze 130a, 634021, Tomsk, Russia

Tel./fax: +7 3822 522-511

Mobile: +7 903 950-50-70

Skype: eugene.ermolenko

ICQ #: 77904200

[www.smartptt.com](http://www.smartptt.com)