



# **TRBOnet Enterprise/PLUS**

## **Phone Interconnect**

User Guide Version 5.3.5

World HQ

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## **1** Introduction

## 1.1 About This Guide

This document is intended for administrators setting up evaluation and proof-ofconcept deployments of MOTOTRBO Dispatch over IP solutions. The document describes the minimum steps to configure the telephony voice dispatch workplace.

## **1.2 About TRBOnet**

TRBOnet is a suite of professional applications for MOTOTRBO digital two-way radio networks. TRBOnet manages voice and data communication paths across network endpoints. It provides a unified graphical dispatcher workbench interface for the entire range of workforce fleet management tasks.

For more information about TRBOnet products, refer to our website.

#### 1.3 Contacts

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EMEA	+44 203 608 0598	info@trbonet.com — general and commercial inquiries
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АРАС	+61 28 607 8325	<u>http://kb.trbonet.com</u> — online knowledge base



## 2 **Overview**

The Phone Interconnect feature is designed to extend the TRBOnet voice functions to SIP and PSTN endpoints. From the dispatcher's perspective, the multi-line phone connection becomes part of the voice dispatch workspace along with the radio PTT boxes.

When the feature is activated, the phone subscribers registered on the built-in or external SIP servers as well as PSTN users become first-class peers in TRBOnet: the phone calls can be joined to a conference with the radio subscribers, the phone endpoints can directly dial into and be reachable for the radios and the dispatcher, the audio messages can be sent to phones (played back when the number answers), etc.

TRBOnet includes an internal SIP telephony PBX server. You can also create SIP trunk connections with external PBX servers or route calls to an external PBX server as if it is just another SIP endpoint.



## **3 Telephony Configuration**

You will need to use the following three software tools to edit the configuration settings for the telephony integration:

- TRBOnet Server
- TRBOnet Dispatch Console
- Motorola CPS to edit the codeplugs for the repeater and the radio stations

Follow the guidance below to configure the necessary telephony-related settings.

#### 3.1 **TRBOnet Server**

 Navigate to Radio Networks > Digital Systems > Repeater > Advanced settings and from the Phone System list, select either 'Motorola Phone System' or 'TRBOnet Phone System (TX Interrupt)'.

Configuration		Advanced settings			Version: 5.2.0.1356
🖗 Redundancy	^				
Database		Voice Call Hang Time (	(ms):		
😪 Reports		Group Call:	3000	\$	
Service Management		Private Call:	4000		
🔀 Advanced settings		Private Call;	4000	-	
Geocoding Servers		Emergency Call:	4000	÷	
Radio Networks		TV Descelles		*	
		TX Preamble:	120	÷	
		TX Timeout:	60	÷	seconds
Services					
Capacity MAX 1		Phone System:	Motorola Phone S	ystem	•
Repeater #1		Allow CSBK Data			
		Allow Cobic Data			
🔒 Privacy					
Audio Paths					
Analog Control Stations					
Remote Agents					
Friendly Servers					
Telephony Telephony					
	<b>Y</b>				
Set Defaults			Apply		OK Cancel

- In the **Configuration** pane, select **Telephony**
- In the **Telephony** pane, select **Use Telephony**.

Configuration		Telephony	Version: 5.2.0.1356
	~		
Digital Systems		✓ Use Telephony	
Services		SIP Server	
Repeater #1		Internal PBX Server	
X Advanced settings			
<b>III</b> Slot #1			
<b>III</b> Slot #2			
Local Slots			
Analog Control Stations			
Remote Agents			
Friendly Servers			
Telephony			
Internal PBX Server			
Advanced settings			
🜵 Data Sources			
🍀 Modbus TCP		Add Delete	Test
🔀 Email	×		
Set Defaults		Apply OK	Cancel



## 3.1.1 Telephony Advanced Settings

• In the Configuration pane, under Telephony, select Advanced Settings.

Configuration	Advanced settings		Version: 5.3.0.1670
Advanced setti A	Max Ring Time:	30 second 10 minute	
Slot #2	Max Call Time: Public Address:	10  minute 77.232.61.122	·S
Control Station #1  TRBOnet Swift A1C  Repeater #1  Analog Control Station:	Phone owner address:	Take from database	▼ <u>Configure</u>
Remote Agents Friendly Servers Advanced settings Telephony Advanced settings External PBX Server Advanced settings External PBX Server Advanced settings			
Set Defaults		Apply OK	Cancel

• In the Advanced Settings pane, specify the following parameters:

#### Max Ring Time

Specify the maximum ringing time, in seconds.

#### Max Call Time

Specify the maximum call time, in minutes.

#### Public Address

This is the public IP address of your PC. To detect the public address, click the ellipsis (...) button.

Public IP Address				>	×
Detect pu	ıblic IP Address	ove	er STUN ser	ver	
STUN Server:	stun.ekiga.net			•	
Port:	3478	¢			
IP Address detected:	77.232.61.122				
Detect			ОК	Cancel	

#### • STUN Server

From the drop-down list, select the STUN Server.

• Detect

Click this button to detect your public IP address.

#### Phone owner address

From the drop-down list, select 'Take from database'. This will enable determining the street address of a caller. Click the **Configure** link.



ddress format	
Address format:	
, %HOUSE_NUMBER%, %STREET_NAME%	•
Availablefields: %HOUSE NUMBER% %FLAT NUMBER% %STREET NAME% %COMMUNITY NAM %STATE NAME% %POSTCODE% %COUNTRY NAME% %EXTRA INFO%	<u>E%</u>

• Add the required fields to the address information.

#### 3.1.2 Internal PBX Server

- Make sure the Internal PBX Server option is selected in the Telephony pane.
- In the Configuration pane, select Internal PBX Server.

Configuration	Internal PBX	Server			
Slot #1		ernal PBX Server	9 <b>국</b> 약 Por	t: 5060 🛟	]
TRBOnet Swift Agent#1	Dispa	tch Center			
Controller #1	SIP ID	): 1234			
Selex #1	SIP U	ser: 1234			
Remote Agents					
Telephony					
Advanced settings Thternal PBX Server					
1 Data Sources					
🍀 Modbus TCP					
🔀 Email					
SMS					
📮 License	<b>*</b>				
Set Defaults			Apply	ОК	Cancel

- In the Internal PBX Server pane, specify the following parameters:
  - Local IP

Enter the IP address of the PC with TRBOnet Server.

Port

Enter the local UDP port number for the SIP service (5060, by default).

#### **Dispatch Center**

SIP ID

Enter the SIP ID that will be used by the Dispatch Center.

SIP user

Enter the SIP user name that will be used by the Dispatch Center.

#### 3.1.2.1 Advanced Settings

• In the **Configuration** pane, under **Internal PBX Server**, select **Advanced Settings**.



Configuration		Advanced settings		
Services	<b>^</b>			
		Packet time (ms):	60	* *
		Codecs:	PCMU,PCMA,G729,S	PEEX,SPEEX-WB,AMR 🔻
TRBOnet Swift Age		Registration Interval (sec):	3600	\$
Controller #1				
Controller #1		DTMF Send Mode:	RFC 2833	•
Analog Control Station:		First VoIP port:	Default	*
Remote Agents		Use VoIP ports:	All	•
Friendly Servers				
Telephony				
Advanced settings				
Internal PBX Server				
Advanced settings				
↓ Data Sources				
🍀 Modbus TCP				
🔀 Email				
SMS	~			
< >				
Set Defaults			Apply	OK Cancel

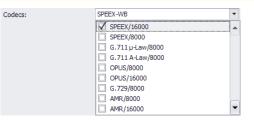
- In the **Advanced Settings** pane, specify the following Internal PBX Serverrelated advanced settings:
  - Packet time

Enter the packet length, in milliseconds.

Codecs

In the drop-down list, select/unselect the codecs to be used for audio compression.

Note: For the best user experience, we strongly recommend that **SPEEX-WB** codec be used.



#### Registration Interval (sec)

Enter the time interval, in seconds, to check the SIP registration status of subscribers.

#### DTMF Send Mode

Enter mode for sending DTMF tones. The available modes are RFC 2833, SIP INFO (DTMF relay), and SIP INFO (DTMF).

#### First VoIP port

Enter the number of the first VoIP port for audio communications

#### Use VoIP ports

From the drop-down list, select which VoIP ports will be used (all, even, or odd).



#### 3.1.3 External PBX Server

You can enable multiple external PBX servers to use the SIP Interconnect feature. This feature enables calls from the radio to the phone and vice versa. The dispatcher can make a call from the Dispatch Console to a phone as well as redirect a phone call to a subscriber radio.

• In the **Telephony** pane, click **Add**.

Or, in the **Configuration** pane, right-click **Telephony** and choose **Add PBX Server**.

Configuration	External PBX Server
Advanced settings Privacy Slot #1 Slot #2 Control Station #1 TRBOnet Swift Agent#1	✓ Use External PBX Server       Provider options       Address:     yourprovider.com       Port:     5060
Controller #1	Local IP: 10.10.100.99 ▼ ♥ Port: 5061 ♀
Analog Control Stations Remote Agents Friendly Servers	SIP ID:         57068           SIP User:         User 123
Telephony Advanced settings	Password: Test Call
Advanced settings	
Set Defaults	Apply OK Cancel

• In the External PBX Server pane, specify the following parameters:

#### Use External PBX Server

Select this option to enable an external PBX server.

#### **Provider options**

Address

Enter your SIP provider address, and select the protocol from the dropdown list on the right of the address (for more details, contact your SIP provider).

Port

Enter the port number of the SIP provider (5060, by default).

- Click **Test** to check the connection to the provider.
- Local IP

Enter the IP address of the PC with TRBOnet Server.

Port

Enter the local port number to make connections from.

#### **Dispatcher Center**

Note: This information is provided by the SIP provider.



#### SIP ID

Enter the SIP ID that will be associated with TRBOnet Server to make and receive calls.

SIP user

Enter the SIP user name for the login.

Password

Enter the password for the login.

Test Call

Click this button to make a test call.

Note: To make a test call, make sure that the TRBOnet Server service is not running.

#### 3.1.3.1 Advanced Settings

• In the **Configuration** pane, under the corresponding **External PBX Server**, select **Advanced Settings**.

Configuration		Advanced settings		
Services	^			
		Packet time (ms):	60	÷
		Used codecs:	PCMU,PCMA,G729,S	SPEEX, SPEEX-WB, AMR,
TRBOnet.Swift Agent #1		Registration Interval (sec):	3600	*
Controller #1				•
Controller #1		DTMF Send Mode:	RFC 2833	*
Analog Control Stations		Do not register users on a P	BX server (SIP trunk)	
Remote Agents		Do not register internal u	isers on a PBX server	
Friendly Servers		Configure user's authorizat		
Telephony		configure user's authorizat		
Advanced settings		First VoIP port:	Default	* *
Monternal PBX Server		Use VoIP ports:	All	•
The second secon		Available SIP numbers:	1	•
Advanced settings		Available SIF Humbers:	1	
↓ Data Sources				
Modbus TCP				
Email				
SMS	~			
Set Defaults	-		Apply	OK Cancel

- In the **Advanced Settings** pane, specify the following External PBX Serverrelated advanced settings:
  - Packet time

Enter the same value as specified in the phone system;

Codecs

In the drop-down list, select/unselect the codecs specified in the phone system.

Registration Interval (sec)

Enter the time interval, in seconds, to check the SIP registration status of subscribers.

Do not register users on a PBX server (SIP trunk)

Select this option so that radios will use the SIP trunk system to get extensions.



#### Configure user's authorization

Click this link to set up user authorization for the systems with enhanced authorization parameters. It is recommended to be used when Radio ID is equal to SIP ID. In case when Voice is transmitted via Radio Channel, Radio ID is used. When voice is transmitted via GSM channel, SIP ID is used.

SI	P Authorization use	rs	×
	SIP ID	User Name	
I	010101	User # 1	
	Add Sc Dalata		
ł	Add X Delete		OK Cancel

- Click **Add** to add a new user authorization.
- SIP ID

Enter the SIP ID for the new user.

• User Name

Enter the User Name for the new user;

• Click **OK** to save the new user authorization.

#### First VolP port

Enter the number of the first VoIP port for audio communications. The default value is specified in Network Parameters.

#### • Use VoIP ports

From the drop-down list, select which VoIP ports will be used (all, even, or odd).

#### Available SIP numbers

Enter the range of SIP phone numbers that will be available for outgoing calls.

#### 3.1.3.2 Redundancy

A Redundant PBX Server be used when a connection to the main PBX Server fails.

• In the **Configuration** pane, under the corresponding **External PBX Server**, select **Redundancy**.



Configuration	Redundancy	Version: 5.3.0.1670
Privacy	Redundant PBX Servers:	
	PBX Server Address PBX S	Server Port TRBOnet Local Port
Local Slots	1 123.321.102.201 5060	5061
Control Station #1		
TRBOnet Swift A 10		
Repeater #1		
Analog Control Station:		
Remote Agents		
Friendly Servers		
Telephony		
Internal PBX Server		
Advanced settings		
External PBX Server		
Advanced settings		
🧰 🛱 Redundancy	Reregister users when changing PB	(server
∲ Data Sources 🗸	Add Delete	Test 🔺 🔻
< >		
Set Defaults	4	Apply OK Cancel

• In the **Redundancy** pane, click **Add**.

#### PBX Server Address

Enter the IP address of the redundant server.

- Check the corresponding port numbers (PBX Server Port and TRBOnet Local Port).
- Select the check box beside the server address.
- Register users when changing PBX server
   Select this check box so that phone users will be registered when the PBX server is switched to the redundant server.

## 3.2 TRBOnet Dispatch Console

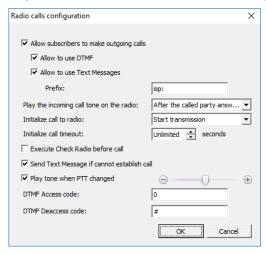
• Click **Administration** (1), and then **Telephony** (2) to configure incoming and outgoing SIP calls:





## 3.2.1 Radio Calls Configuration

• Click the **Configure** button (3) to set radio call configuration parameters:



#### Allow subscribers to make outgoing calls

Select this option to enable outgoing phone calls from the radio subscribers.

#### Allow to use DTMF

Select this option to allow radio subscribers to dial the phone number as a sequence of DTMF tones.

#### Allow to use Text Messages

Select this option to allow radio subscribers to initialize phone calls via sending TMS messages with a specified prefix to the dispatcher.

• Prefix

Enter the standard prefix for a text message.

#### Play the incoming call tone on the radio

Select the mode for playing the incoming call tone on the radio that initiates a call.

• After the called party answers the call

When a radio initiates a phone call to a subscriber via DTMF tones or a TMS message, the incoming call tone will be played on the radio after the called party answers the call.

• Immediately

When a radio initiates a phone call to a subscriber via DTMF tones or a TMS message, the incoming call tone will be played on the radio immediately that is without waiting for the called party to answer the call.

#### Initialize call to radio

Select the option how to start a call on a radio.

#### • Start transmission

Select to start a call to a radio automatically.

• Send ring tone

Select to play a ring tone until the radio user presses the PTT.



#### Initialize call timeout

Specify a timeout that defines how long to attempt to connect to the called party.

#### Execute Check Radio before call

Select this option to execute a Check Radio command before placing a call.

- Send text message if cannot establish call Select this option to send a text message to the radio when the channel is
  - busy and a phone call cannot be established.
- Play tone when PTT changed

Select this option so that the phone will sound a tone when the remote radio's PTT is pressed or released.

#### DTMF Access Code

Set the value to that configured for the radios as **Access Code** in MOTOTRBO CPS.

#### DTMF Deaccess Code

Set the value to that configured for the radios as **Deaccess Code** in MOTOTRBO CPS.

#### 3.2.2 Incoming Calls Configuration

Click the **Configure** button (4) to set incoming call configuration parameters:

Call to external num	iber: Use number as Internal
Extention numbe	ers (voice menu)
<ul> <li>Start call automa</li> </ul>	ticaly
Max. number length	n: Unlimited 🜩 Accept code: #
Number	Call Description
0	Call dispatcher (any available)
<number></number>	Call radio with Radio ID = <number></number>
🛃 Add 🗐 Edit	

#### • Call to Dispatch Center

Select the mode for handling incoming calls made to the dispatcher.

Decline calls

All incoming phone calls will be declined.

Open voice menu

When an incoming call arrives, the subscriber will hear Voice menu commands.

#### Redirect to dispatchers

All incoming voice calls will be redirected to all dispatchers of the Dispatch center and any free dispatcher will answer the phone call.



#### • Call to external number

Select the mode for handling incoming calls made from unregistered subscribers.

#### Decline Calls

Select this option to decline all phone calls from unregistered subscribers.

#### Use number as Radio ID

Select this option so that the system will use unregistered numbers as a Radio ID and start a Private Call.

Use number as Internal

Select this option to allow the system to read unregistered numbers according to Voice Menu rules.

#### **Extension numbers (Voice menu)**

#### • Start call automatically

Select this option to search for the number in the Extensions table automatically. When this option is disabled, the subscriber must dial the number according to the following example: **0(phone number)#**. The character **#** is used to search for the phone number in the table.

#### • Max. number length

Specify the maximum number of characters allowed in a phone number.

#### • Accept Code

Specify the character that will be used to finish dialing the number.

All available numbers are listed in the table below.

• Click the **Add** link to add a number to the table.

#### To add a static number

Choose Static number.

xtension number			×
Static number			
C Dynamic numb	ber		
Number:	123456		
Call Type:	Call Group		•
Channel:	Control Station #1		▼
Group:	All Call		-
		ОК	Cancel

#### Number

Enter a phone number to add to the table (contact list).

#### Call Type

Select the call type from the drop-down list.

#### • Call Dispatcher

Select this type to make a phone call to the dispatcher.

• Call Radio

Select this type to make a phone call to the selected radio.



#### Call Group

Select this type to make a phone call to the selected group.

#### Channel

Select the channel to make a group phone call through (available for Group Calls only).

#### Dispatcher/Radio/Group

Select the dispatcher, radio, or group depending on what you have selected in the **Call Type** box.

#### To add a dynamic number

Choose Dynamic number.

Extension number	r	×
C Static numbe	-	
Prefix:	123	
Call Type:	Call Radio	•
Channel:	Auto Detect	-
Radio:	Detected by Radio ID	-
	ОК	Cancel

#### Prefix

Specify a prefix to type in on the keyboard.

#### Call Type

Select the call type from the drop-down list.

#### • Call Radio

Select this type to make a phone call to a radio.

#### Call Group

Select this type to make a phone call to a group.

- **Call Phone** Select this type to make a phone call to a telephone.
- Channel

Select the channel to make a group phone call through (available for Group Calls only).



#### **3.2.3 Telephony Options**

- On the main menu, click **Tools > Options**.
- In the **Options** dialog box, click the **Telephony** tab.

Show this number of	of Recent Phone Calls:	10	÷
Ringtone config	aration		
Incoming Call	Call Waiting Held Cal	1	
Call Priority	Ringtone	Action	
Normal:	Default	Play	
Emergency:	Default	Play	
High:	Default	Play	
Low:	Default	Play	

Queue all incoming telephone calls

Select this option so that when the telephone line is busy, incoming calls will be queued rather than rejected.

Show this number of Recent Phone Calls

Enter the number of items to display in the list of recent calls. The list of recent phone calls appears when you click the arrow on the right of the dial string in the Telephony box.

#### **Ringtone configuration**

On the Incoming Call (Call Waiting) tab:

- Click a corresponding link in the **Ringtone** column, and from the pulldown menu, select either **Set Default**, **Set Custom**, or **Disable**.
   If you select **Custom**, in the dialog box that opens, browse for the audio file on the local PC and click **Open**.
- Click a **Play** link in the **Action** column to play back the corresponding ring tone.

On the Held Call tab:

• Remind after

Enter the timeout, in seconds, that will be used for playing the reminder tone when a call is on Hold.

• Ringtone

Specify the reminder tone to be played when a call is on Hold.



#### 3.2.4 Adding SIP Extensions

• Click Administration (1), Telephony (2), Extensions (3), Add > SIP Phone (4)

File View Map Tools Help	
Administration	Telephony 🔮 🐠 🕒
Server	S 1: Line free 📢 🖉 Intercom 🕥 📢 🖉
Database	Configure Calls Extensions Redirect Calls Aliases Profiles
남: Radio Systems - 친암 System Bridging	🛃 Add 🗸 🦫 Edit 🔀 Delete 🔚 Grouping 🍸 Auto Filter 🝥 Default Settings
Telephony	SIP Phone SIP User Caption
Tasks	TRBOnet Mobile Client 1234 Internal PBX Server
I have been as	Radio 125 125 125
Voice Dispatch	🗑 Radio 235 235 235
Location Tracking	
Route Management	
C Text Messages	
Radio Allocation	
Administration	HI 4I 4 Record 1 of 3 > >> >> + + + + + + + + + + + + + + +
🚯 127.0.0.1 🔉 🕱 Administrator 📑 L	icensed to: demo Demo License 📀 Active 🗸

In the **Phone User** dialog box, specify the following parameters:

Phone User		×
SIP ID:	2409	
SIP Name:	Walt	
Caption:	Walt	
Password:	•••••	
Password (repeat):	•••••	
	OK Cancel	

#### SIP ID

Enter the SIP ID that will be used by the phone user.

SIP Name

Enter the SIP user name that will be used by the phone user.

Caption

Enter the caption to be displayed for the phone user.

Password

Enter the password for the phone user to be authenticated by the telephone system.

Password (repeat)

Enter the password again.



#### 3.2.5 Redirect Calls

If a dispatcher doesn't take a phone call within the set period or they have changed their status to unavailable, the call is automatically forwarded to a specified dispatcher, radio, radio group, or phone account. This can be useful during night shifts when no dispatcher is available.

• Click Administration (1), Telephony (2), Redirect Calls (3), Add (4)

File View Map Tools Help					
Administration	Telephon	y			👲 🐠 🕓
Radio Systems	I: Line fi disp 15	*)) *:	<ul> <li>Slot 1</li> <li>Slot 1</li> </ul>	) <b>.</b> 0	
			Redirect Calls Aliases Profiles		
Voice Dispatch			🖁 Grouping 🍸 Auto Filter 🐵 I	Default Settings	
Location Tracking	Drag a column Call to	Redirect Call	3	×	
📸 Job Ticketing	4	Call to: Target:	😽 Dispatcher 1	•	
💓 Route Management		Timeout:	30 🗘 seconds		
🖂 Text Messages		Redirect to: Type:	Radio Group	•	
🔮 Voice Recording		Radio System:	Slot 1	•	
Event Viewer		Target:	Firemen	•	
[행] Radio Allocation	1	Priority:	Normal	▼	
Administration	141 41 4 Reco	ord 0 of 0 🕨 🕨 🖣	]		•
🔂 127.0.0.1 🔉 🔂 🔂 🔂 🖉 Administ	rator 🔥 21 da	vs before your Suppo	ort expires	0	👩 Active -

In the Redirect Calls dialog box, specify the following parameters:

#### Call to

Target

From the drop-down list, select the dispatcher or dispatcher group.

Timeout

Specify the time period, in seconds, defining how long to wait until the dispatcher answers a call.

#### **Redirect to**

• Туре

From the drop-down list, select one of the following:

• Decline

The calls will be declined.

• Dispatcher

The calls will be redirected to a specified dispatcher or dispatcher group.

• Radio unit

The calls will be redirected to a specified radio.

• Radio Group

The calls will be redirected to a specified radio group.

• Phone account

The calls will be redirected to a phone account from the phone book.



#### Radio System

If a radio group is selected as the redirection destination, select the radio system over which to make a call to the specified radio group.

Target

From the drop-down list, select a particular dispatcher/dispatcher group, radio, radio group, or phone account, depending on what you have selected in the **Type** field.

Priority

If a radio or radio group is selected as the redirection destination, select the priority with which the call will be made over the radio system.

#### 3.2.6 Aliases

In addition to SIP extensions, the SIP aliases can also be used in your Telephony system.

File View Map Tools Help		
Administration	Telephony	👲 🐠 🛂
Server	I Line free         4100         Intercom         11400         Mantenace           Sales         01400         Ø dap 15         01400         Ø Group 10	) = 0 ) = 0 ) = 0
Voice Dispatch	SP D	
Location Tracking	4 SIP ID: 2411	
Route Management	SIP Name: 2411 Caption: Eugene	
Reports	OK Cancel	
Event Viewer	-	
Administration	1  ₩  ≪ ∢ Record 1 of 1 >> >> >₩ ∢	
🐻 127.0.0.1 🛞 🥵 📩 💆 Adm	ministrator 📑 Licensed to: demo Demo License	🕑 Activ

• Click Administration (1), Telephony (2), Aliases (3), Add (4)

In the 'Phone Alias' dialog box, specify the following parameters:

SIP ID

Enter the SIP ID that will be used by the phone user.

SIP Name

Enter the SIP user name that will be used by the phone user.

Caption

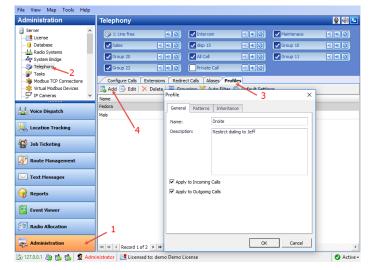
Enter the caption to be displayed for the phone user.



#### 3.2.7 Profiles

In order to restrict incoming and outgoing calls as well as set priority for calls, SIP profiles can be used in your Telephony system.

• Click Administration (1), Telephony (2), Profiles (3), Add (4)



In the 'Profile' dialog box, specify the following parameters:

Name

Enter a name for the profile.

Description

Enter a description for the profile.

Apply to Incoming Calls

Select this checkbox to apply the profile to incoming calls.

#### Apply to Outgoing Calls

Select this checkbox to apply the profile to outgoing calls.

• Click the **Patterns** tab.

ofile		
General Patte	rns Inheritance	
* - any number o ? - one character	ard chars in pattern: f characters of numbers from xxx to yyy	
*411???		×
Call Priority:	Normal	•

 Specify the pattern that will be used to match the numbers in the dial string or the incoming call numbers.



#### Call Priority

From the drop-down list, specify the priority for the calls corresponding to the specified patterns.

• Click the Inheritance tab.

Profile	×
General Patterns Inheritance	
Call Priority Profile Name	
Emergency 🔽 Fedora	
Inherit Mals	
Emergency	
High Normal	
Low	
ОК	Cancel
OK	Cancel

On this tab, you can select the profiles to include in the profile you are adding/editing.

Call Priority

From the drop-down list, select the priority for the inherited profile.

#### **3.3 MOTOTRBO CPS**

This section describes how to configure MOTOTRBO repeaters and radios using MOTOTRBO CPS.

#### 3.3.1 Repeater

Note: The Motorola's Digital Phone Patch license is not required for repeaters.

If the NAI protocol is used in your radio system, then:

• In the left pane, select **Channels** > **Zone** > **Channel**. In the Channel pane, select the **Network Application Interface Phone** check box.

DR 3000	Channel1
	Top Enhanced GPS RX IX
	Color Code 1 ÷
Link Establishment	Network Application Interface Phone
	System Controller Mode
白 💼 Channels	IP Site Connect Slot 1 & Slot 2 💌
Channel1	Messaging Delay Normal 💌
	RSSI Threshold (dBm) -100 -
	IF Filter Type Wide
	BSI Mode Analog 🗸
	v



### 3.3.2 Subscriber Radio

#### 3.3.2.1 Motorola Phone System

- In the left pane, select Signaling Systems > Phone. Right-click it, and choose Add > System.
- In the left pane, under **Phone**, select the phone system that has been just added.

General Settings	Phone_100
Accessories	
Buttons	Top DTMF
····· 🖂 Text Messages	
101 Telemetry	Gateway ID 100 ÷
🛅 Menu	
Security	Access Code 0
Network	Deaccess Code #
Announcement	
Job Tickets	DTMF
🚊 ···· 🚞 Signaling Systems	
🖤 User Defined 5 Ton	Pretime (ms) 500 +
🕂 🔁 5 Tone	
🕀 ···· 🚞 MDC	TX Tone Duration (ms) 120
🕀 🚾 📄 Quik-Call II	TX Tone Interval (ms) 80 ÷
🕂 🖳 Digital Emergency	
🕀 💼 Capacity Plus Emer	Pause Duration (ms) 4000 ÷
🖻 ···· 🔁 Phone	
🖤 Phone_100	
Encoder	
🕂 🚞 Decoder	
Contacts	
🕂 🚞 RX Group Lists 🗸 🗸	
< >	]

• In the right pane, specify the following parameters:

#### Gateway ID

Enter the same ID as **TRBOnet Peer ID** in the Repeater settings of TRBOnet Server.

Repeater #1						
System Name:	Repeater #1	Repeater #1				
TRBOnet Peer ID:	100	*				
TRBOnet Radio ID:	64250	*				
TRBOnet Local Port:	50000	*				
Master Repeater Co	nnection Info:					

#### • Access Code and Deaccess Code

These codes must be the same on all the radios and reflect the corresponding values set in the Dispatch Console.

#### TX Tone Duration (ms)

Enter the duration of the DTMF tone digits, in milliseconds, for the phone system. It is recommended to set this value to **120**.

#### TX Tone Interval (ms)

Enter the duration of the intervals between the DTMF tone digits in a transmission sequence, in milliseconds, for the phone system. It is recommended to set this value to **80**.



- General Settings TOD RX IX Voice Announcement File None ~ Dual Capacity Direct Mode Timing Leader Preference Eligible 👻 Scan/Roam List None • Auto Scan 🔲 Color Code 1 ÷ Repeater/Time Slot Phone System Phone\_100 -ARS On System Change Enhanced GPS Window Size 8 👻 Privacy 🔲 ANalog
   Scan
   Roam
   Capacity Plus Privacy Alias Privacy Key1 AES Alias None • < >
- In the left pane, select Channels > Zone (Channel Pool) > Channel.

• In the Channel pane:

#### Phone System

Select the phone system you have previously specified.

#### 3.3.2.2 TRBOnet Phone System

If your radio system uses TRBOnet Phone System, make sure that the following parameters are appropriately configured in the radios that are used as control stations and subscriber radios.

• In the left pane, select General Settings.

	ļ	DP48		General Settings									
		_	General Settings										
		× .	Accessories	Top	<u>CWID</u>	Audio Profile	Microphone	Backlight	Battery Saver	<u>Alerts</u>	Over-the-	Air Programming	Persistent LRRP Reque
			Buttons	Lone \	Norker	Power Up F	assword and Loc	<u>k Front Pr</u>	ogramming Pass	word [	Delete All	5 Tone Radio ID	
			Text Messages Telemetry									_	^
		_	Menu					Radio	Name Radio 235				
			Security									Select	
		-	Network	Welcome Image Remove							Select		
			Announcement										
	÷	-	Job Tickets								- 1	Remove	
	÷		Signaling Systems										
	÷		Encoder					_		_			
	÷		Decoder					Ra	dio ID 235				
	<b>+</b>	_	Contacts						GPS 🔽				
🕀 📄 RX Group Lists			GNSS						GPS/QZSS				
	÷	_	Channels							.5		<u> </u>	
	<b>∓</b>	_	Scan					Private	Calls 🔽				
	+ ···· + ····	_	Roam Capacity Plus				Site	e Search Timer	(sec) 15 ÷				
		-	capacity i lao					aliantina Dalau	(min)				
							ARS INI	ialization Delay	(min) 0 ÷				
							TX Pre	amble Duration	(ms) 300				
							TX Inhibit	Quick Key Ov	erride 🕅				
								Monitor	Turno	1			~
Genera	Set	ttings									Expert Vie	w	NUM

- Make sure the **Private Calls** check box is selected.
- In the left pane, select Channels > Zone (Channel Pool) > Channel.

DP4801	IPSC1								
Accessories		Top RX TX							
····· II Buttons				^					
🖂 Text Messages	RX		TX	~					
Telemetry		Offset (MHz)							
🔚 Menu		Offset (MHZ)							
Security	Frequency (MHz) 167.420000	0.00000	Frequency (MHz) 146.420000						
Announcement	·	Сору	·						
+ Job Tickets	Ref Frequency Default V		Ref Frequency Default -						
🕀 📄 Signaling Systems	Group List		Contact Name All Call						
🕀 💼 Encoder	Group List List1								
🕀 📄 Decoder	Emergency Alarm Indication		Emergency System None						
Contacts     RX Group Lists	Emergency Alarm Ack		vox E						
RX Group Lists     Channels	Emergency Call Indication								
E Chaines	Emergency Call Indication		Power Level Low -						
			TOT (sec) 60 -						
			TOT Rekey Delay (sec)						
🕀 💼 Scan			Allow Interruption						
🛨 💼 Roam			TX Interruptible Frequencies						
🗄 📄 Capacity Plus									
			Admit Criteria Color Code Free	•					
			In Call Criteria Follow Admit Criteria	~					

- Select both the **Allow Interruption** and **TX Interruptible Frequencies** check boxes.
- From the Admit Criteria drop-down list, select 'Color Code Free'.
- From the **In Call Criteria** drop-down list, select 'Follow Admit Criteria' if the radio will be used as a control station. For a subscriber radio, select 'TX Interrupt'.