



TRBOnet Enterprise/PLUS

Phone Interconnect

User Guide Version 5.7

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

Configuration		Advanced Settings			
Redundancy		Voice Call Hang Time (male		
Database		voice call hally fille (iiis).		
😡 Reports		Group Call:	3000	÷	
Service Management		Private Call:	4000	*	
🔀 Advanced Settings		Private Call;	4000	Ŧ	
Geocoding Servers		Emergency Call:	4000	÷	
Radio Systems					
Services		TX Preamble:	120	÷	
IPSC 1		TX Timeout:	60	÷	seconds
Advanced Settings					
Privacy		Phone System:	Motorola Phone	System	*
Slot #1		TX Interrupt Mode:	MSI Proprietary		
III Slot #2		TX Interrupt Houe.	rior rioprictary		
Local Slots		Allow CSBK Data			
60 CP 1					
PTT over Cellular					
Advanced Settings					
TRBOnet.Mobile gateway					
Remote Agents					
Friendly Servers					
Telephony					
1.010	~				
< >					
Set Defaults			Apply	/	OK Cancel

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

Configuration	Telephony					
IPSC 1 CP1 Advanced Settings Privacy DDMS service Advanced Setting Advanced Setting Advanced Setting Transcript Advanced Setting Transcript Advanced Settings TRBOnet.Mobile gateway Remote Agents Friendly Servers Telephony Advanced Settings Internal PBX Server Advanced Settings SMS Notifications	✓ Use Telephony SIP Server 1 ✓ Internal PBX Server					
	Add Delete		Test	A V		
Set Defaults		Apply	ОК	Cancel		



3.1.1 Telephony Advanced Settings

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

Configuration	Advanced Settings			
Privacy Privacy DDMS service Advanced Settings MNIS data service Advanced Setting MNIS data service Advanced Setting TRBOnet.Mobile gateway Remote Agents Friendly Servers Advanced Settings Internal PBX Server Advanced Settings Data Sources SMS Notifications	Max Ring Time: Max Call Time: Public Address: Phone owner address:	30 \$ 10 \$ 84.52.107.217 Take from database	e seconds	••• • <u>Configure</u>
Set Defaults		Apply	ОК	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. Enter the Public Address if your TRBOnet Server is behind a router and an <u>External PBX Server</u> will be used. To detect the public address, click the ellipsis (...) button.



• 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	
Advanced Settings	Use Internal PBX Serve	r .10.100.99 • ¢ ³ Port: 5060 ¢
CI Local Slots	Dispatch Center	
💋 CP1	User Extension: 12	34
PTT over Cellular	User Name: 12	34
TRBOnet.Mobile gateway		
Remote Agents		
Friendly Servers		
Telephony		
Advanced Settings		
Internal PBX Server		
↓ Data Sources		
K Email		
SMS Notifications		
Push Notifications		
🗔 License		
< >		
Set Defaults		Apply OK 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

User Extension

Enter the user extension number that will be used by the Dispatch Center.

User Name

Enter the 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	
Advanced Settings 🔺	、	
	Packet time (ms):	60 ‡
DDMS service	Codecs:	SPEEX-WB,G711µ,G711a,OPUS,OPUS-WB,
MNIS data service	Registration Interval (sec):	3600 ‡
Advanced Setting	DTMF Send Mode:	RFC 2833 -
Audio Paths	First VoIP port:	Default ‡
PTT over Cellular	Use VoIP ports:	All 👻
TRBOnet.Mobile gateway		
📷 Remote Agents		
Friendly Servers		
Telephony Telephony		
X Advanced Settings		
Thernal PBX Server		
Advanced Settings		
∲ Data Sources		
🔀 Email		
SMS Notifications		
Push Notifications		
📮 License 🗸 🗸	,	
< >		
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.

• **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 VolP 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**.

• 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.



User Extension

Enter the extension number that will be associated with TRBOnet Server to make and receive calls.

User Name

Enter the user name for the login.

User Password

Enter the user 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**.

- 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.

SIF	Authorization users	×
	User Extension	Authorization User Name
Þ	010101	User 1
÷	Add XDelete	OK Cancel

• Click **Add** to add a new user authorization.

• User Extension

Enter the extension number for the new user.

Authorization User Name

Enter the name that will be used as the authorization user name.

• 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							
	Redundancy						
<u>^ </u>	Redun	dant	PBX Servers:				
			PRY Server Address	PRY Server Port	+	TRBOnetio	cal Port
							carrore
	1	\checkmark	123.21.102.201	5060		5061	
	Z Re	reai	ster users when changi	ng PBX server			
		-					
·	4	٩dd	Delete			Test	
					_		
				Apply		ОК	Cancel
		Redun	Redundani 1 V	Redundant PBX Servers: PBX Server Address 1 23.21.102.201	Redundant PBX Servers: PBX Server Address PBX Server Address PBX Server Address PBX Server Por 1 I <	Redundant PBX Servers: PBX Server Address PBX Server Add Delete	Redundant PBX Servers: PBX Server Address PBX Server Port TRBOnet Lo 1 123.21.102.201 5060 5061 V Reregister users when changing PBX server Add Delete Test

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

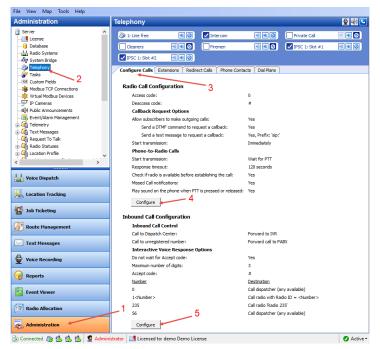
PBX Server Address

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

- Go to Administration (1), Telephony (2).
- In the **Telephony** pane, click the **Configure Calls** tab (3).





3.2.1 Radio Call Configuration

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

adio Call Configuration	
Access code:	0
Deaccess code:	#
Callback Request Options	
✓ Allow subscribers to make outgoing call	s
✓ Send a DTMF command to request	a callback
☑ Send a text message to request a	callback
Use this prefix in text messages:	sip:
Start transmission:	Immediately 🚽
Phone-to-Radio Calls	
Start transmission:	Wait for PTT 🚽
Response timeout:	120 seconds
✓ Check if radio is available before estab	lishing the call
Missed Call notifications	
Play sound on the phone when PTT is p	pressed or released
Sound level:	+
	_
	OK Cancel

Access Code

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

• Deaccess Code

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

Callback Request Options

- Allow radio users to make outgoing calls Select this option to enable outgoing phone calls from the radios.
- Send a DTMF command to request a callback Select this option to allow radio users to dial the phone number as a sequence of DTMF tones sent over the radio channel.
- Send a text message to request a callback Select this option to allow radio users to initiate phone calls via sending TMS messages with a specified prefix to the dispatcher.
 - Use this prefix in text messages

Enter the specific prefix for a text message.

• Start transmission

Select the mode for starting transmission on the radio that initiates a call.

Wait for answer

Transmission will be started as soon as the called party answers the call.

Immediately

Transmission will be started immediately that is without waiting for the called party to answer the call.



Phone-to-Radio Calls

• Start transmission

Select the option how to start transmission on the radio receiving a phone call.

- Immediately Select to start the call immediately.
- Wait for PTT Select to play a ringtone until the radio user presses the PTT.
- Response timeout

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

- Check if radio is available before establishing the call Select this option to execute a Check Radio command before placing a call.
- Missed Call notifications

Select this option to send a text message to the radio when the channel is busy and a phone call cannot be established.

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

3.2.2 Inbound Call Configuration

Click the **Configure** button (5) to set incoming call configuration parameters:

Call to Dispatch C	Center:	Forward to IVR
Call to unregister	ed number:	Reject
Interactive Vo	ice Respon	se Options
Do not wait fo	r Accept code	e if one of the fixed numbers below is dialed
Maximum number	of digits:	3
Accept code:		#
Number	Destina	tion
0	Call disp	atcher (any available)
1 <number></number>	Call radi	io with Radio ID = <number></number>
235	Call radi	io 'Radio 235'
56	Call disp	atcher (any available)

Inbound Call Control

• Call to Dispatch Center

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

Reject

All incoming phone calls will be declined.

Forward to IVR

When an incoming call arrives, the phone user will hear Voice Menu commands.



Redirect to online 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 unregistered number

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

Reject

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

Use this 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 this number as Internal Phone Number

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

Forward call to PABX

Select this option to forward all calls from unregistered numbers to the External PBX.

Interactive Voice Response (IVR) Options

• **Do not wait for Accept code if one of the fixed numbers below is dialed** Select this option to search for fixed the number in the Extensions table automatically.

When this option is disabled, the subscriber must dial the number according to the following example: **(phone number)#**. The character **#** (or *****, if selected as the Accept code) is used to search for the phone number in the table.

- **Maximum number of digits** Specify the maximum number of characters allowed in a phone number.
- Accept Code

Specify the character that will be used to finish dialing the extension 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 (fixed) number

• Choose **Static number**.

Extension numbe	r		×
Static number Dynamic num			
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
 Calest this type to make a phone call to the calest

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 numbe	er X
C Static numb	-
Prefix:	123
Call Type:	Call Radio
Channel:	Auto Detect
Radio:	Detected by Radio ID
	OK 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.

eue all ir his num	i ty	one calls	Ac	10 tion Play Play Play Play	<u>A</u>	
his num one col ng Call Il Priori ormal: nergenc: gh:	ber of Recent I nfiguration Call Waiting ty y:	Phone Calls: Held Call Ringtone Default Default Default		tion Play Play Play	4	
ng Call Il Priori ormal: hergenco	call Waiting ty	Held Call Ringtone Default Default Default		tion Play Play Play	Ī	
ng Call Il Priori ormal: nergenc gh:	Call Waiting	Ringtone Default Default Default		<u>Play</u> Play Play		
Il Priori ormal: nergenc gh:	i ty	Ringtone Default Default Default		<u>Play</u> Play Play		
ormal: nergenc gh:	y:	<u>Default</u> <u>Default</u> <u>Default</u>		<u>Play</u> Play Play		
nergenc gh:	y:	<u>Default</u> <u>Default</u>	♪ ♪	<u>Play</u> Play		
gh:		Default	▶	Play		
w:		<u>Default</u>	Þ	<u>Play</u>		
					٥	OK

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

- Go to Administration (1), Telephony (2).
- In the **Telephony** pane, click the **Extensions** tab (3), and then **Add** (4).

Administration	Telephony						🔮 🚳 🔽
Server Server Dotabase Rado Systems System Bridge	 2: Line free Cleaners IPSC 1: Slot # 		Intercon		 <	Private Call	1 0 E 0
Tasks 2		Extensions Redre	rnal Numbers				ettings Logical Groups
Voice Dispatch 4	Dispatch Cer	Phone User			×	Internal PB sipnet.ru	
Location Tracking	X SIP Phone X SIP Phone V SIP Phone	General Logical Gro User Extension: User Name:	2408 John	Fields		Walt Eugene Prune	
Route Management		User Password: Password (repeat):	•••••				
Voice Recording		First Name: Last Name:	John Bingham				
Reports		Display Name:	John B.				
Badio Allocation				ОК	Cancel		
administration	HI 41 4 Record 1 0	5 + ++ ++ 4					

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

User Extension

Enter the extension number that will be used by the phone user.

User Name

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

User Password

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

Password (repeat)

Enter the password again.

Fist Name/Last Name/Display Name

Enter the first, last and display names of the phone user.

- In addition, on the **Logical Groups** tab, specify logical groups for the phone user.
- On the **Custom Fields** tab, specify the desired values for the custom fields.



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			🔮 🐠 🕒
- Hardio Systems - Harding - W Telephony - Taks - 2 - Modus TCP Connections	i: Line fi disp 15		Ø ✓ Slot 1 •)		
< >	Configure (Redirect Calls Aliases Profiles		
Voice Dispatch			🖁 Grouping 🍸 Auto Filter 🐡 De	fault Settings	
Location Tracking	Drag a column	Redirect Call	3	×	
Job Ticketing	4	Call to: Target:	Sispatcher 1	•	
😥 Route Management		Timeout:	30 🗘 seconds		
		Redirect to:			
Text Messages		Type:	🔞 Radio Group	•	
🔮 Voice Recording		Radio System:	Slot 1	•	
Event Viewer		Target:	👘 Firemen	•	
		Priority:	Normal	•	
Radio Allocation	1		ОК	Cancel	
Administration	HI 41 4 Reco	urd0of0 ► 🕨 ₩ 🖣			Þ
🐻 127.0.0.1 🔉 🛸 🛸 🛸 🖉 Administ	rator 🔥 21 da	vs before your Suppo	ort expires		👩 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

Type

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 Phone Contacts

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

- Go to Administration (1), Telephony (2).
- In the Telephony pane, click the Phone Contacts tab (3), and then Add (4).

File View Map Tools Help	
Administration	Telephony 🔮 🚳 😉
Server	③ 1: Line free ■ (④) ✓ Intercom ■ (■ (●) ● (■) ● (■)
Hadio Systems 생활 System Bridge 	
Tasks 2 v	🖪 Add 📑 Edit 🗙 Delete 🐺 Grouping 🍸 Auto Filter 🗞 Default Settings
Voice Dispatch	User Extension User Name First Name Last Name Derniau Name Logical Groups
Location Tracking	5547 Eu General Logical Groups Custom Fields 3
Job Ticketing	4 User Extension: 5548 User Name: Trun
Route Management	First Name:
Text Messages	Last Name: Wilson Display Name: Trun W
Voice Recording	
Reports	
Event Viewer	
Radio Allocation	OK Cancel
Administration	!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!!
🐻 Connected 🛞 🕵 🥵 🕵 Adm	inistrator 📃 Licensed to: demo Demo License 🥑 Active 🕶

In the Phone Contact dialog box, specify the following parameters:

User Extension

Enter the extension number for the phone contact.

User Name

Enter the user name for the phone contact.

- Fist Name/Last Name/Display Name
 Enter the first, last and display names of the phone contact.
- In addition, on the Logical Groups tab, specify logical groups for the phone contact.
- On the **Custom Fields** tab, specify the desired values for the custom fields.



3.2.7 Dial Plans

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

- Go to Administration (1), Telephony (2).
- In the **Telephony** pane, click the **Dial Plans** tab (3), and then **Add** (4).

Administration	Telephony	
Server Server Database Addo Systems System Bridge System Bridge System Sridge System Sr	I: Line free Intercom Intercom	
< Tasks 2 ~	Configure Calls Extensions Redirect Calls Phone Contacts, Dial Plans	
Voice Dispatch	Add B Edit X Polate Torumine Auto Eller & Default China 3 X	
Location Tracking	Fedora General Patterns Inheritance	
🚰 Job Ticketing	4 Description: Restrict dialing to Jeff	
💓 Route Management		
C Text Messages		
🔮 Voice Recording	Apply to Incoming Calls	
Reports	Apply to Outgoing Calls	
📔 Event Viewer		
8 Radio Allocation		
administration	In the second l of 2 P	

In the **Dial Plan** dialog box, specify the following parameters:

Name

Enter a name for the dial plan.

Description

Enter a description for the dial plan.

• Apply to Incoming Calls

Select this checkbox to apply the dial plan to incoming calls.

Apply to Outgoing Calls

Select this checkbox to apply the dial plan to outgoing calls.

• Click the **Patterns** tab.

Dial Plan				×
General	Patterns	Inheritance		
* - any nu ? - one cha	mber of cha aracter	hars in patterr racters mbers from xx:		
1				
*411???				×
Call Priority	y:	Normal		•
			ОК	Cancel

• 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.

Dial Pl	an		\times
Gen	eral Patterns	Inheritance	_
	Call Priority	Profile Name	
\checkmark	Emergency	 Fedora 	
	Inherit	Mals	
	Emergency High		
	Normal		
	Low		
		OK Cancel	

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

Call Priority

File Device Licenses Tools Help

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

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 Set Categories pane, select Zone/Channel Assignment> Zone>Zone Items>Channel.
- In the right pane, select the Network Application Interface Phone check box.

Ipen Save Read Wr	
000 ► Zone ► Zone1 ► Zone Items ► Ch Categories	General Enhanced GNSS RX/TX
Configuration*	⊘ General
General General Settings Accessories	Channel Type Digital Channel Name Channel1
Security	Color Code 1
Link Establishment	System Controller Mode No
 Talkgroups Zone/Channel Assignment 	IP Site Connect (Repeater) None Messaging Delay (ms) 60
Zone	Repeater RSSI Threshold (dBm) -100 IF Filter Type Wide
	BSI Mode Analog
dation Results Warning Messages Search Resul	



3.3.2 Subscriber Radio

3.3.2.1 Motorola Phone System

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- In the Set Categories pane, select Systems > Phone Systems.
- In the right pane, click the plus sign button and add the corresponding phone system.
- In the left pane, select the phone system you have added.

Open Save P Read Write DP4801e Phone Systems Phone 100*	법법 득법 ↑립 ☆ ★법 Clone Clone Express Update Register Activate	×
Set Categories	General DTMF	
 Dob Tickets Options Dob Tickets Templates C Systems C Signaling Systems C User Defined 5 Tone C Tone Systems 	General System Name Phone_100 Gateway ID 100 Caccess Code Deaccess Code #	
MDC Systems Quik-Call II Systems	⊙ DTMF	
 Digital Emergency Systems Capacity Plus Emergency Syste Phone Systems 	Pretime (ms) 500 CTX Tone Duration (ms) 120 TX Tone Interval (ms) 80	
Phone_100 \$	Pause Duration (ms) 4000	
/alidation Results(1*) Warning Messages Search Results	Help	

• 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
TRBOnet Peer ID:	100 🗘
TRBOnet Radio ID:	64250
TRBOnet Local Port:	50000
Master Repeater Con	nection 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**.



• In the Set Categories pane, select Zone/Channel Assignment> Zone>Zone Items>Channel.

File Device Licenses Tools Help	답답 -=□□□□ ↑□□ ☆ ★□□ Clone Clone Express Update Register Activate	
DP4801e F Zone F Zone1 F Zone Items F Channe Set Categories	General RX/TX	×
♥	⊙ General	_î
Job Tickets Job Tickets Systems Encoder	Channel Type Digital Channel Name Channel 1 Voice Announcement File None	
Decoder Contacts RX Group Lists	Dual Capacity Direct Mode	
 Tone/Channel Assignment Zone Zone Zone1 	Scan/Roam List None Auto Scan No Color Code 1	
Channel Pool Scan Lists Capacity Plus Lists	Repeater/Time Slot 1 Phone_100	
		•
Validation Results(1*) Warning Messages Search Results	Help Sarial Number 87	

- In the right 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 Set Categories pane, select General > GeneralSettings

Dpen Save Read) ⊟ Write	Id Id →Id ☆ ★Id Clone Clone Express Update Register Activate
1801e 🕨 General Settings" : Categories	4	General CWID Audio Profile Microphone Backlight Battery Saver Allerts Persistent LRRP Regu
 Configuration* 		Lone Worker Power Up Password and Lock Front Programming Password Delete All 5 Tone ID
Device Information		
🔻 📋 General	- 11	() General
Welcome Bitmap	- 11	
Language Packs		Radio Alias Motorola
🗋 General Settings 🔅		Radio ID 1
Accessories		GNSS 🗌
Control Buttons		GNSS GPS/GLONASS/QZSS
Text Messages		Private Calls
Telemetry		Site Search Timer (sec) 15
Menu		ARS Initialization Delay (min) 0
Security		TX Preamble Duration (ms) 960
Network	w	
		TX Inhibit Quick Key Override
		Monitor Type Open Sauelch

- In the right pane, make sure the **Private Calls** check box is selected.
- In the Set Categories pane, select Zone/Channel Assignment> Zone>Zone Items>Channel.

pen Save Read) डि. डि. डि. चि. चि. चि. चि. चि. चि. चि. चि. चि. च	★ ★量 Register Activate	
Categories	Channel I*	General RX/TX	
 □ Configuration* ■ Device Information ▶ □ General ▶ □ Job Tickets 	⊙ RX/TX RX	<u>Tx</u>	
Systems Encoder Decoder Contacts	Frequency (MHz) 300.025000	Offset (MHz) 0.000000 Copy	
RX Group Lists Dire/Channel Assignment D Zone Zone Zone	Group List DigitalRXGroupList/List Emergency Alarm Indication Emergency Alarm Ack No		•
 Channel Pool Can Lists Capacity Plus Lists 	Emergency Call Indication <i>Emergency Call Decode Tone No</i>	TOT (sec) 60	4) 4) 4) 4
,		X Interruptible Frequencies V Admit Criteria Color Code Free	•

- In the right pane, navigate go to **RX/RTX** section.
- 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'.