



TRBOnet Enterprise/PLUS

Phone Interconnect

User Guide

Version 5.2



Notices

This document is for informational purposes only. Neocom Software offers no warranties, express or implied, in this document.

Neocom and the Neocom logo, TRBOnet and the TRBOnet logo are either registered trademarks or trademarks of Neocom Software, Ltd.

MOTOROLA, MOTO, MOTOROLA SOLUTIONS and the Stylized M logo are trademarks or registered trademarks of Motorola Trademark Holdings, LLC.

Intellectual property rights protect the voice coding technology embodied in this product including patent rights, copyrights and trade secrets of Digital Voice Systems, Inc. This voice coding technology is licensed solely for use within this communications equipment. U.S. Pat. Nos. 6,199,037, 5,870,405, 5,754,974, 5,664,051, 5,630,011, 5,517,511, 5,491,772, 5,247,579, 5,226,108, 5,226,084, 5,216,747 and 5,081,681.

Microsoft, Windows, SQL Server and the .NET logo are either registered trademarks or trademarks of Microsoft Corporation in the United States and/or other jurisdictions.

Other product or company names mentioned herein may be trademarks of their respective owners.

© 2017 by Neocom Software, Ltd. All rights reserved.

This document was last revised on December 7, 2017.



Contents

1	Introduction1			
	1.1	Abo	out This Guide	1
	1.2	Abo	out TRBOnet	1
	1.3	Cor	ntacts	1
2	Over	view.		2
3	Telephony Configuration			
	3.1	TRE	3Onet Server	3
		3.1.1	Internal PBX Server	4
		3.1.2	External PBX Server	5
	3.2 TRB		3Onet Dispatch Console	8
		3.2.1	Radio Calls Configuration	8
		3.2.2	Incoming Calls Configuration	10
		3.2.3	Ring Tones Configuration	12
		3.2.4	Adding SIP Extensions	13
	3.3	МС	OTOTRBO CPS	14
		3.3.1	Repeater	14
		3.3.2	Subscriber radio	15
		3.3.3	TRBOnet Phone System	16



1 Introduction

1.1 About This Guide

This document is intended for administrators setting up evaluation and proof-of-concept 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

Region	Phone	Email & Support	
EMEA	+44 203 608 0598	info@trbonet.com — general and commercial inquiries	
Americas	+1 872 222 8726	support@trbonet.com — technical support http://kb.trbonet.com — online knowledge base	
APAC	+61 28 607 8325		

1



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 a SIP trunk connection with an external PBX or route calls to an external PBX as if it is just another SIP endpoint.



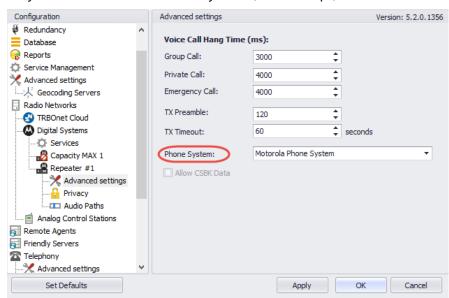
3 Telephony Configuration

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

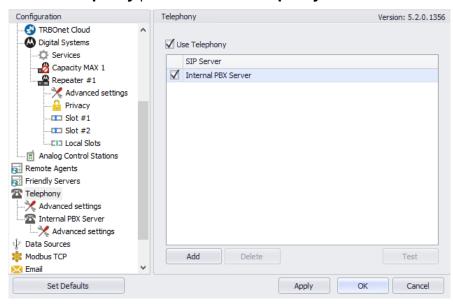
- TRBOnet Server
- TRBOnet Dispatch Console
- Motorola CPS to edit the codeplugs for the repeater and the radio stations Follow the guidance below to configure the necessary telephony-related settings.

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)'.



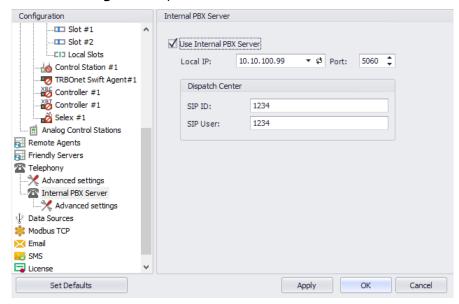
- In the Configuration pane, select Telephony
- In the **Telephony** pane, select **Use Telephony**.





3.1.1 Internal PBX Server

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



- 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).

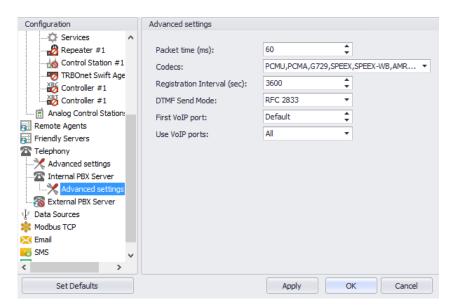
Dispatch Center

- 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.

3.1.1.1 Advanced Settings

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





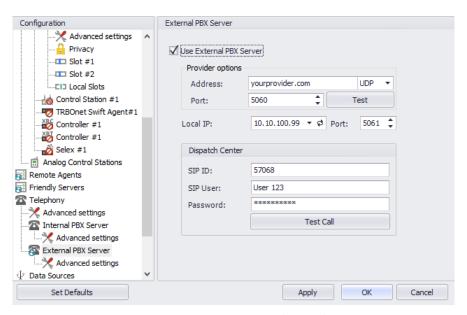
- In the **Advanced Settings** pane, specify the following Internal PBX Server-related advanced settings:
 - Packet time
 Enter the packet length, in milliseconds.
 - Codecs
 In the drop-down list, select/deselect the codecs to be used.
 - Registration Interval (sec)
 Enter the time interval, in seconds, 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 VolP port
 Enter the number of the first VolP port for audio communications
 - Use VoIP ports
 From the drop-down list, select which VoIP ports will be used (all, even, or odd).

3.1.2 External PBX Server

You can enable an external PBX server 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 drop-down 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

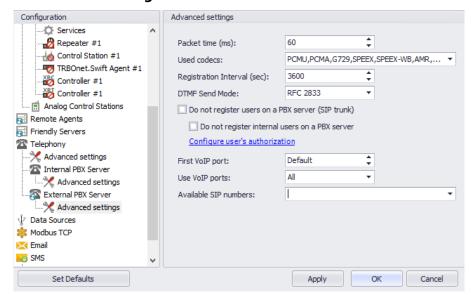
Click this button to make a test call.



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

3.1.2.1 Advanced Settings

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



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

Enter the same value as specified in the phone system;

Codecs

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

Registration Interval (sec)

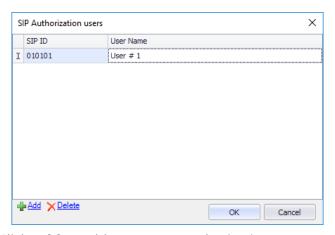
Enter the time interval, in seconds, to check the SIP registration status of subscribers.

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.

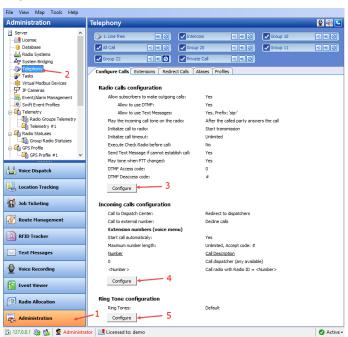




- Click **Add** to add a new user authorization.
- **SIP ID**Enter the SIP ID for the new user.
- User Name Enter the User Name for the new user;
- Click **OK** to save the new user authorization.

3.2 TRBOnet Dispatch Console

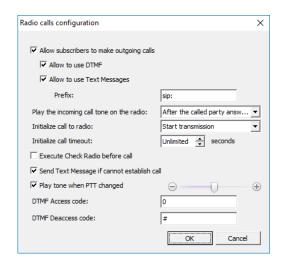
• Click **Administration** (1), and then **Telephony** (2) to configure incoming and outgoing SIP calls:



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

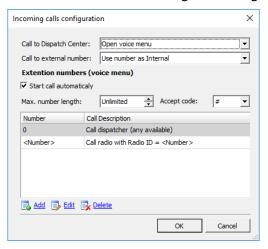
Set the value to that configured for the radios as **Access Code** in MOTOTRBO CPS.

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 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

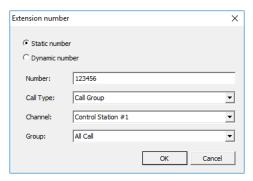
Specify the character that will be used to finish dialing the number.

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.



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

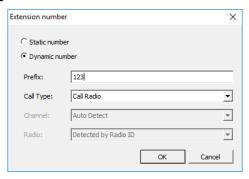
Select the channel to make a group phone call through (available for Group Calls only).

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.



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.

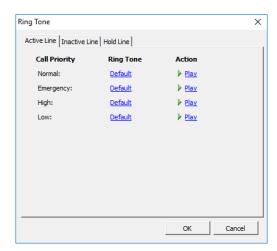
Channel

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

3.2.3 Ring Tones Configuration

• To configure ring tones, click the **Configure** button (5):

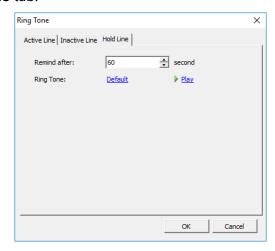




On the Active Line (Inactive Line) tab:

- Click a corresponding link in the Ring Tone column, and from the pull-down 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 Hold Line tab:



• Remind after

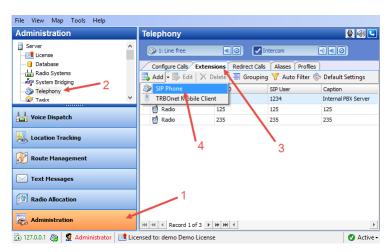
Enter the timeout, in seconds, that will be used for playing the reminder tone when a call is on Hold.

• **Ring Tone**Specify the reminder tone to be played when a call is on Hold.

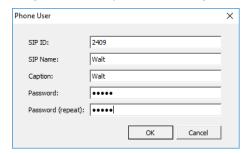
3.2.4 Adding SIP Extensions

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





In the Phone User 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.

Password

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

• Password (repeat)

Enter the password again.

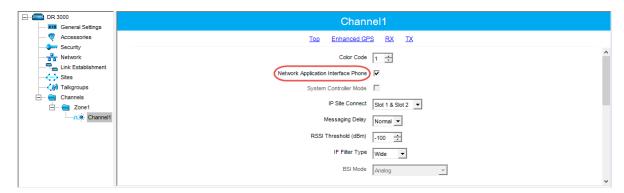
3.3 MOTOTRBO CPS

3.3.1 Repeater

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

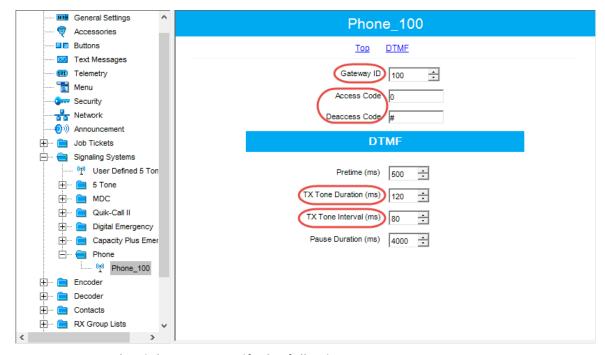
• In the left pane, select **Channels > Zone > Channel**. In the Channel pane, select the **Network Application Interface Phone** check box.



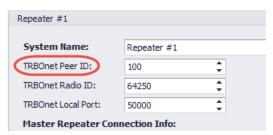


3.3.2 Subscriber radio

- 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.



- 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.





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.

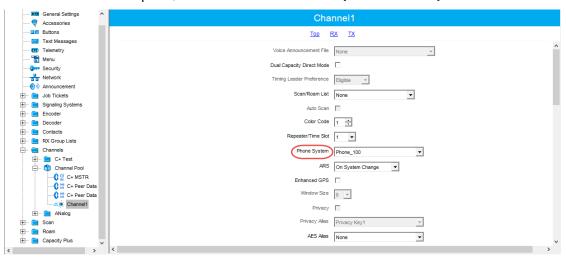
TX Tone Duration (ms)

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 left pane, select Channels > Zone (Channel Pool) > Channel.



- In the Channel pane:
 - Phone System

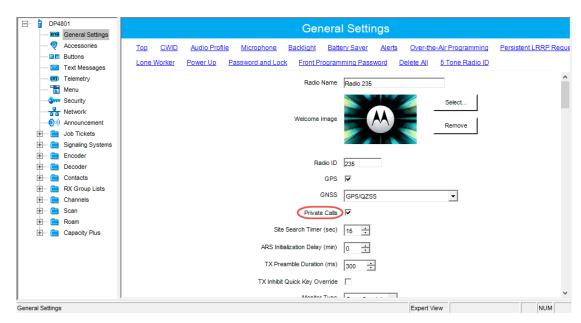
Select the phone system you have previously specified.

3.3.3 TRBOnet Phone System

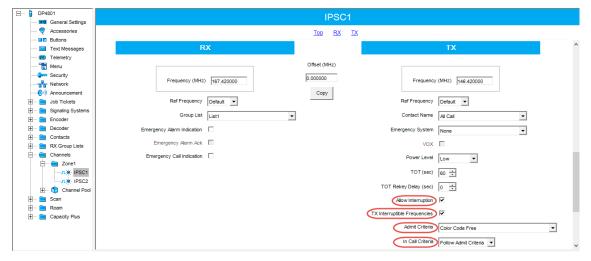
If your radio system uses TRBOnet Phone System, make sure that the following parameters are appropriately configured in the radios that are used as control stations and subscriber radios.

• In the left pane, select **General Settings**.





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



- 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 will be used as a control station. For a subscriber radio, select 'TX Interrupt'.