



Avaya Solution & Interoperability Test Lab

Application Notes for New Voice Technologies Mobicall 8.2 with Avaya IP Office Server Edition 10 and 500v2 Expansion - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate New Voice Technologies Mobicall with Avaya IP Office Server Edition 10 and 500v2 Expansion. Mobicall is an Alarm generation and distribution solution that connects to IP Office as a SIP line.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the configuration steps required to successfully integrate New Voice Technologies Mobicall with Avaya IP Office Server Edition 10 and 500v2 Expansion. Mobicall is an Alarm generation and distribution solution that connects to IP Office as a SIP Line. System alarms are recorded on the Mobicall server by calling Mobicall and recording an alarm which is automatically or manually distributed to IP Office endpoints by the Mobicall server.

2. General Test Approach and Test Results

The general test approach was to configure the Mobicall Server to communicate with the IP Office via a SIP Trunk. Stations present on the IP Office were configured on the Mobicall server and a number was configured to dial Mobicall and create and initiate alarms.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

The interoperability compliance test included both feature functionality and serviceability testing. The feature functionality testing focused on setting and distributing alarms in different call scenarios with good quality audio. The tests included:

- Mobicall SIP trunk is connected and in Service.
- Mobicall can route alarms to SIP, Digital and H.323 endpoints.
- Mobicall can use the Dial Paging feature in IP Office to alert extensions and groups to alarms created in Mobicall.
- Alarms can be set and distributed from IP Office to/from Mobicall.
- Failover/Service – Tests the behaviour of Mobicall Server during certain failed conditions.

2.2. Test Results

All test cases were passed.

2.3. Support

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Internet mobilisierung.com

3. Reference Configuration

The configuration shown in Figure 1 was used during the compliance test of New Voice Technologies Mobicall with IP Office. Mobicall utilizes a SIP trunk to communicate with IP Office handsets.

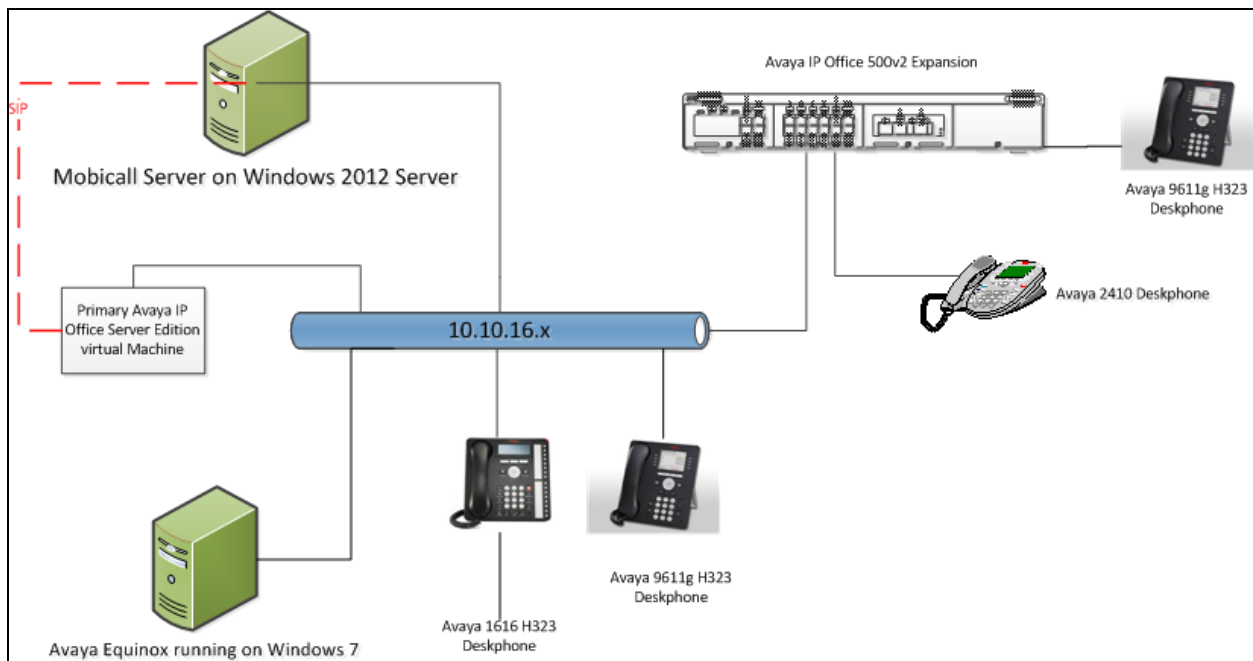


Figure 1: Connection of Mobicall with Avaya IP Office Server Edition 10 and 500v2 Expansion

4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Equipment/Software	Release/Version
Avaya IP Office Server Edition	10.0.0.0.3.0 Build 5
Avaya IP Office 500v2	10.0.0.0.3.0 Build 5
Avaya 1616 IP Deskphone H.323	1.390A
Avaya 9611g IP Deskphones H.323	6.6401
Avaya 2420 Series Digital Deskphones	N/A
Avaya Equinox for Windows SIP	3.0.2.11
Mobicall	8.2

Note: Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and also when deployed with IP Office Server Edition in all configurations

5. Configure Avaya IP Office

This section describes the steps required to allow IP Office to communicate with Mobicall. It is assumed that IP Office is installed and configured before implementing the configuration step below. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 9**.

The configuration illustrated in this section was performed using IP Office Manager

Configuration steps include:

- Check SIP Trunk Licensing
- Administer System Lan settings
- Administer Short Codes for routing and feature
- Administer Short Code for Paging
- Administer Incoming Call route for activating alarms
- Administer SIP Line


5.1. Check IP Office Licenses

In IP Office Manager under **Configuration** select **Licenses** and check the number of **SIP Trunk Channels** are enough for the Mobicall SIP trunk requirements.

Feature	Instances	Status	Expiry Date	Source
Receptionist	10	Valid	Never	PLDS Nodal
Additional Voicemail Pro Ports	252	Valid	Never	PLDS Nodal
VMPPro Recordings Administrators	10	Valid	Never	PLDS Nodal
Office Worker	1000	Valid	Never	PLDS Nodal
VMPPro TTS Professional	40	Valid	Never	PLDS Nodal
IPSec Tunnelling	10	Obsolete	Never	PLDS Nodal
Power User	1000	Valid	Never	PLDS Nodal
Customer Service Agent	10	Obsolete	Never	PLDS Nodal
CCR SUP	5	Obsolete	Never	PLDS Nodal
Avaya IP endpoints	1000	Valid	Never	PLDS Nodal
SIP Trunk Channels	256	Valid	Never	PLDS Nodal
IP500 Universal PRI (Additional cha...	120	Obsolete	Never	PLDS Nodal
CTI Link Pro	1	Valid	Never	PLDS Nodal

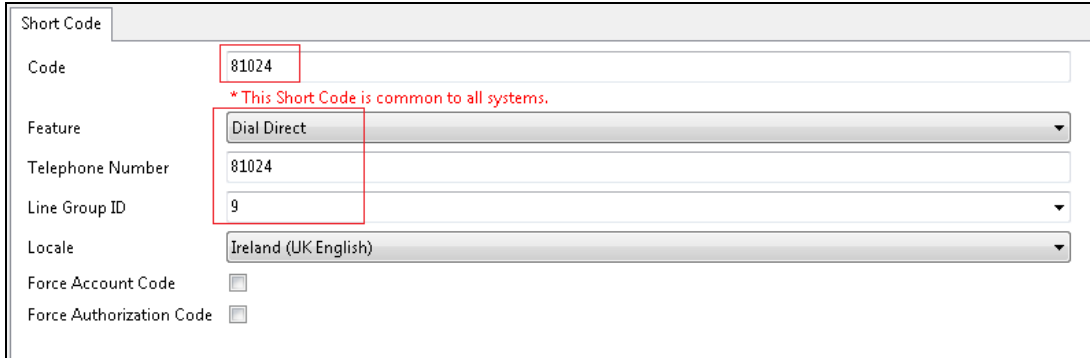
5.2. Check System LAN Settings

Select the **Server edition** → **System** (not shown) and Go to the **LAN1** tab. Check that **SIP Trunks Enable** is selected on the **VoIP** form.

System	LAN1	LAN2	DNS	Voicemail	Telephony	Directory Services	System Events	SMTP	SMDR	Twinning	Codecs	VoIP Sec
LAN Settings												
VoIP												
Network Topology												
<input checked="" type="checkbox"/> H323 Gatekeeper Enable												
<input checked="" type="checkbox"/> Auto-create Extn <input type="checkbox"/> Auto-create User <input type="checkbox"/> H323 Remote Extn Enable												
Remote Call Signalling Port: 1720												
<input checked="" type="checkbox"/> SIP Trunks Enable												
<input checked="" type="checkbox"/> SIP Registrar Enable <input type="checkbox"/> SIP Remote Extn Enable												
<input checked="" type="checkbox"/> Auto-create Extn/User 												
Domain Name: <input type="text"/>												
<input checked="" type="checkbox"/> UDP UDP Port: 5060 Remote UDP Port: 5060												
Layer 4 Protocol <input checked="" type="checkbox"/> TCP TCP Port: 5060 Remote TCP Port: 5060												
<input checked="" type="checkbox"/> TLS TLS Port: 5061 Remote TLS Port: 5061												
Challenge Expiry Time (secs): 10												

5.3. Add a Short code for routing calls to Mobicall

A short code is required to allow calls to be made to Mobicall and record and distribute alarms. From **Solution**→**Short Code** (not shown) right click and select **New** (not shown). Enter the number you wish to dial to access Mobicall as the **Code**. Select **Dial Direct** from the **Feature** drop down, enter the number again as the **Telephone Number** and select the **SIP Line Group ID** used to dial Mobicall.



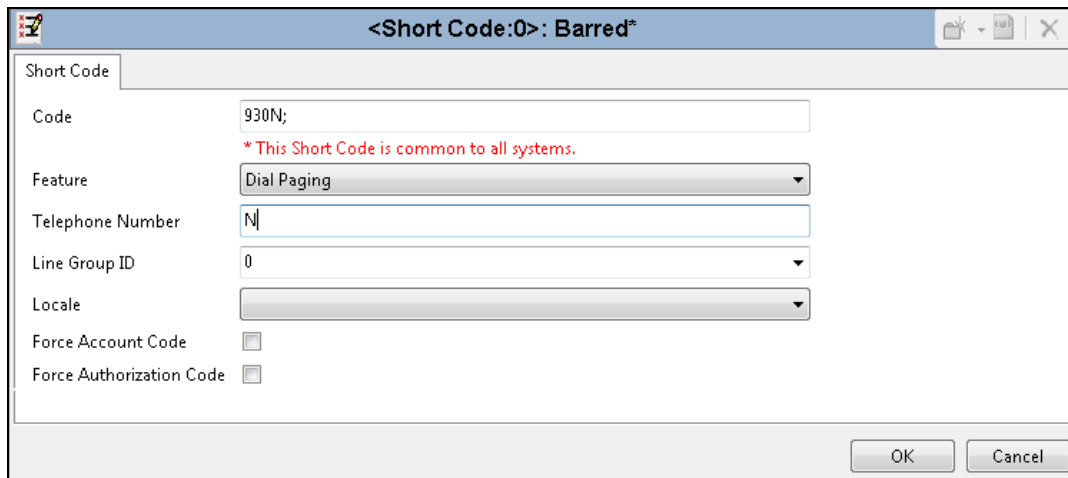
The screenshot shows a configuration window titled "Short Code". The fields are as follows:

Code	81024
Feature	Dial Direct
Telephone Number	81024
Line Group ID	9
Locale	Ireland (UK English)
Force Account Code	<input type="checkbox"/>
Force Authorization Code	<input type="checkbox"/>

A red box highlights the Code, Feature, and Telephone Number fields. A red note below the Code field states: "* This Short Code is common to all systems."

5.4. Add a Short code for Paging

Alarms from Mobicall can be distributed to Extensions or groups dialing direct or by paging the IP Office users. A Dial Paging short code needs to be administered to access this feature. From **Solution**→**Short Code** (not shown) right click and select **New** (not shown). Enter the number you wish to dial to activate the Dial Paging feature as the **Code**. Select **Dial Paging** from the **Feature** drop down, enter **N** as the **Telephone Number** and select the **Line Group ID 0** so that it can be used globally.



The screenshot shows a configuration window titled "<Short Code:0>: Barred*". The fields are as follows:

Code	930N;
Feature	Dial Paging
Telephone Number	N
Line Group ID	0
Locale	
Force Account Code	<input type="checkbox"/>
Force Authorization Code	<input type="checkbox"/>

Buttons for "OK" and "Cancel" are visible at the bottom right.

5.5. Add an Incoming Call Route for activating alarms

An Incoming Call Route must be added to allow any calls from Mobicall to be routes correctly. From the **Solution** → **Incoming Call Route** right click and select **New** (not shown). Select the **Line Group ID** used for the Mobicall SIP Line. Enter the **Incoming Number** as the number of **X**'s to cover the length of any number dialed form Mobicall (10 were used during testing to cover the Paging Short Code and the Extension Lengths).

The screenshot shows the 'Standard' tab of the Incoming Call Route configuration dialog. The fields are as follows:

Bearer Capability	Any Voice
Line Group ID	9
Incoming Number	XXXXXXXXXX
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source
Ring Tone Override	None

Buttons: OK, Cancel

Select the **Destinations** tab and enter a “.” under **Destination**. This this means that calls will be routed to the number dialed.

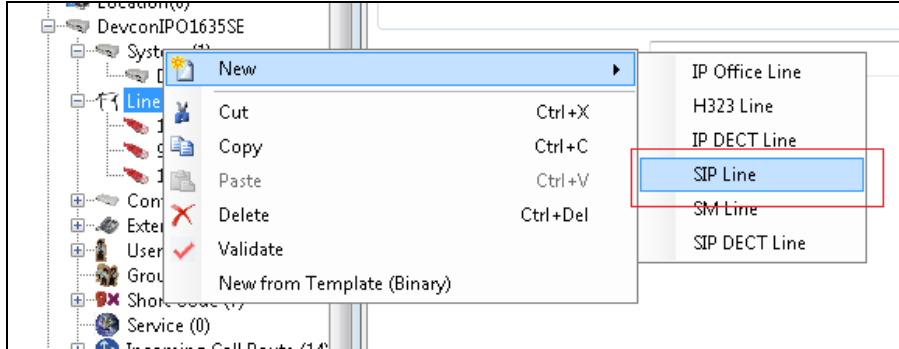
The screenshot shows the 'Destinations' tab of the Incoming Call Route configuration dialog. It contains a table with the following data:

TimeProfile	Destination	Fallback Extension
Default Value	.	

Buttons: OK, Cancel

5.6. Administer a SIP Line

A SIP Line is required for call routing between the IP Office and Mobicall. From the **Server Edition** → **Line** right click and select **New** → **SIP Line**.



On the **SIP Line** tab enter an unused **Line Number** and an **ITSP Domain Name**. The Mobicall Server IP Address is used here.

SIP Line	Transport	SIP URI	VoIP	SIP Credentials	SIP Advanced	Engineering
Line Number		9				In Service <input checked="" type="checkbox"/>
ITSP Domain Name		10.10.16.95				Check OOS <input checked="" type="checkbox"/>
URI Type		SIP				Session Timers
Location		Cloud				Refresh Method: Auto
						Timer (seconds): On Demand
Prefix						Forwarding and Twinning
National Prefix		0				Originator number: []
International Prefix		00				Send Caller ID: None
Country Code						Redirect and Transfer
Name Priority		System Default				Incoming Supervised REFER: Auto
Description						Outgoing Supervised REFER: Auto
						Send 302 Moved Temporarily: <input type="checkbox"/>
						Outgoing Blind REFER: <input type="checkbox"/>

On the **Transport** tab enter the Mobicall Server IP Address as the **ITSP Proxy Address** and set the **Layer 4 Protocol** and **Send Port**.

SIP Line Transport SIP URI VoIP SIP Credentials SIP Advanced Engineering

ITSP Proxy Address 10.10.16.95

Network Configuration

Layer 4 Protocol UDP Send Port 5060

Use Network Topology Info None Listen Port 5060

Explicit DNS Server(s) 0 . 0 . 0 . 0 0 . 0 . 0 . 0

Calls Route via Registrar

Separate Registrar

On the **SIP URI** tab click on **Add** and select **Auto** for **Local URI**, **Contact**, **Display Name** and **Diversion Header**. Set the **Incoming** and **Outgoing Group** to the SIP Line number. Click on **OK** to save changes.

SIP Line Transport SIP URI VoIP SIP Credentials SIP Advanced Engineering

URI	Groups	Local URI	Contact	Display Name	Identity	Header	Originator Number	Send Caller ID	Diversion Header
1	2	Auto	Auto	Auto	Auto	PAI		None	Auto

Edit URI

Local URI Auto

Contact Auto

Display Name Auto

Identity

Identity Auto

Header P Asserted ID

Forwarding And Twinning

Originator Number

Send Caller Id None

Diversion Header Auto

Registration 0: <None>

Incoming Group 9

Outgoing Group :9

Max Sessions 10

OK

Cancel

On the **SIP Advanced** tab select **Caller ID from From header** and **Send From In Clear**.

The screenshot shows the 'SIP Advanced' configuration page for an 'Engineering' SIP Line. The 'Identity' section contains the following options:

- Use Phone Context
- Add user=phone
- Use + for International
- Use PAI for Privacy
- Use Domain for PAI
- Swap From and PAI
- Caller ID from From header
- Send From In Clear
- Cache Auth Credentials
- User-Agent and Server Headers

The 'Media' section contains the following options:

- Allow Empty INVITE
- Send Empty re-INVITE
- Allow To Tag Change
- P-Early-Media Support
- Send SilenceSupp=Off
- Force Early Direct Media
- Media Connection Preservation

The 'Call Control' section contains the following options:

- Call Initiation Timeout (s)
- Call Queuing Timeout (m)
- Service Busy Response
- on No User Responding Send
- Action on CAC Location Limit
- Suppress Q.850 Reason Header
- Emulate NOTIFY for REFER
- No REFER if using Diversion

After all IP Office configuration has been completed, the Configuration needs to be saved and the IP Office must be rebooted.

6. Configure New Voice Technology Mobicall

Setting up the MobiCall installation is not described here. Please take a look into the link for documents provided by NewVoice. (see **Section 9**)

6.1. License Settings

Open New Voice Setup Wizard – Select **Main Settings** → **Dongle Settings** from the left hand menu

New Voice Setup Wizard

Quick start wizard for Mobicall Alarmserver

Select Settings:

- Main Settings
- Language Settings
- Dongle Settings
- SQL Database Settings
- System and Device Supervision Settings
- Telephony Watchdog Settings
- Backup Settings
- Main Alarm Settings
- Group and Personnel Settings
- Alarm Data Import
- Personnel and Group Data Import
- Interface Settings
- Additional Components (TTS, Localisation, VoiceMail, Hotel)
- Conference Settings
- TMS Settings
- Event and Notification Settings
- Autostart Settings
- Web Server Settings
- Extended and System Settings

Dongle Settings

A dongle is a USB device used to protect our software. To run a Mobicall alarm server you need to connect the dongle to a USB port on the alarmserver.
To activate the dongle, the licence has to be updated with the code received from New Voice. Contact New Voice to receive your dongle update code.

Activate the dongle supervision. If no dongle exists (demo version), deactivate this supervision

If the dongle is removed, following alarm is started: 9860 Dongle WatchDog

Deactivate the dongle supervision alarm until the dongle is connected for the first time

Dongle Information: Refresh

(1) New Voice Tool Version: 8.0 Professional

Registered for Region: Demo Version

(3) Registered for Types: Dialogic (invtm) & Diva Server CAPI (invtaf) & VOIP (invtvoip) & AI-Logix (inv)

(2) Number of Lines: 60

Number of Fax-lines: 0

Number of Contacts: 128

Number of Relais: 8

Number of Interfaces: 4

Text to Speech: 1

3G-4G Server for OS: iOS & Android & Blackberry & W-Mobile

Number of Clients: 10

Web Interface Packages: Mobilization & Conference & Group Organizer & Messenger & IVR & Playback & Visualizer

Web Interface Features: Extended Rights Management

Dongle Serial Number: Hex: A96E Dec: 43374 Update License

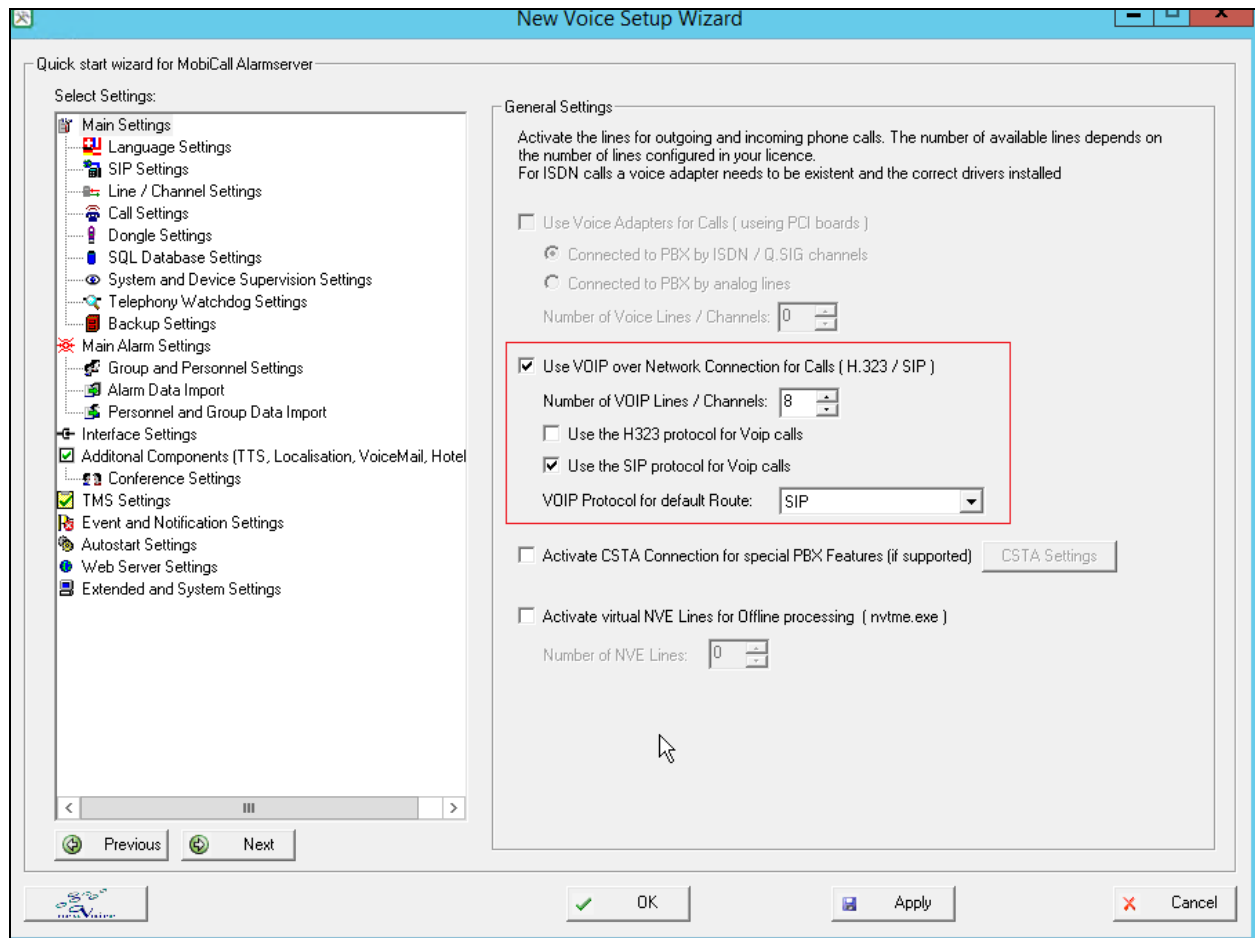
Previous Next

OK Apply Cancel

Check the licenses on your USB Dongle:

Required is at least (1) **NewVoice Tool Version - 8.x** with (2) **2 lines** and (3) **invtvoip** as registered type.

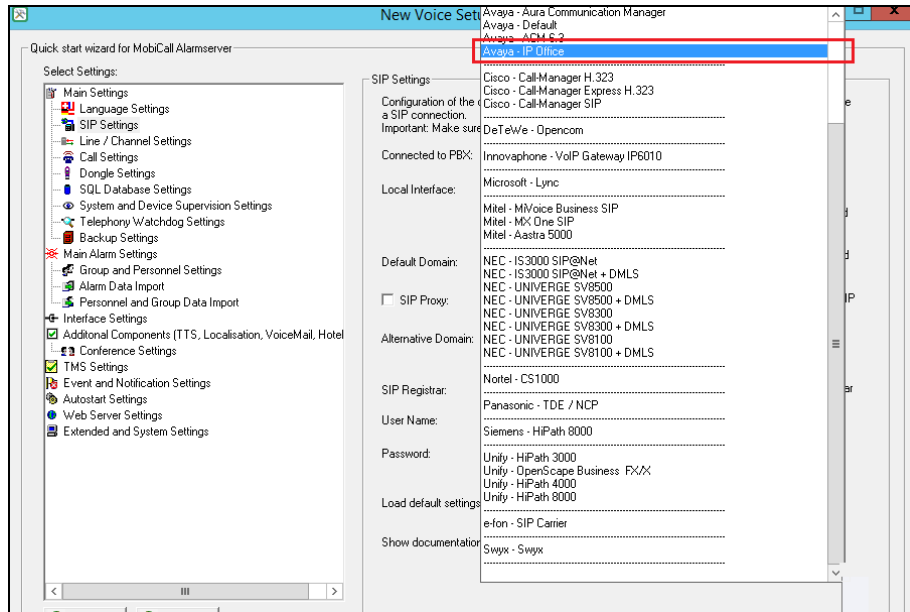
New Voice Setup Wizard – Main Settings



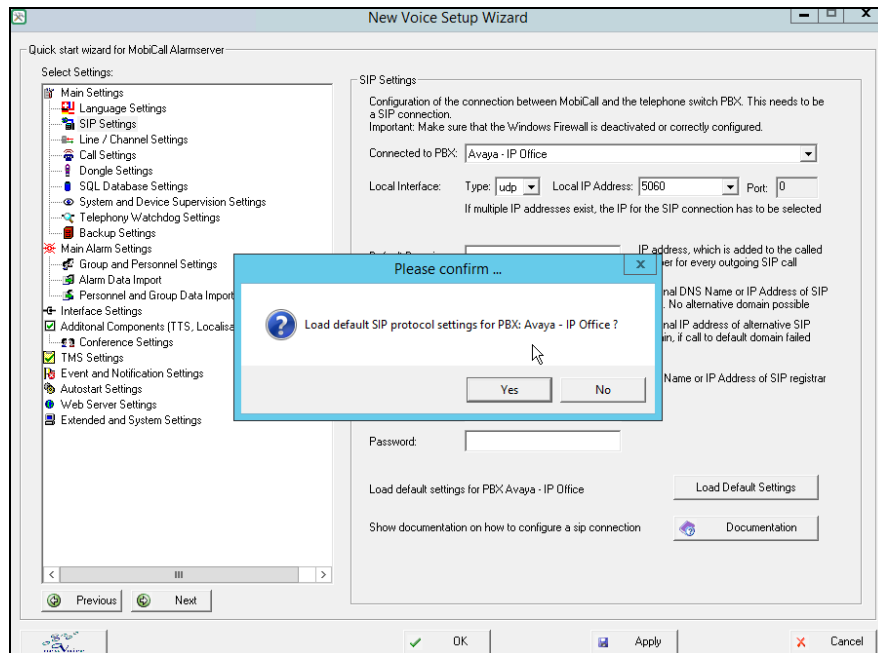
Activate the SIP Lines licensed by setting Number of VOIP Lines / Channels to no more than the licensed number.

6.2. SIP Settings

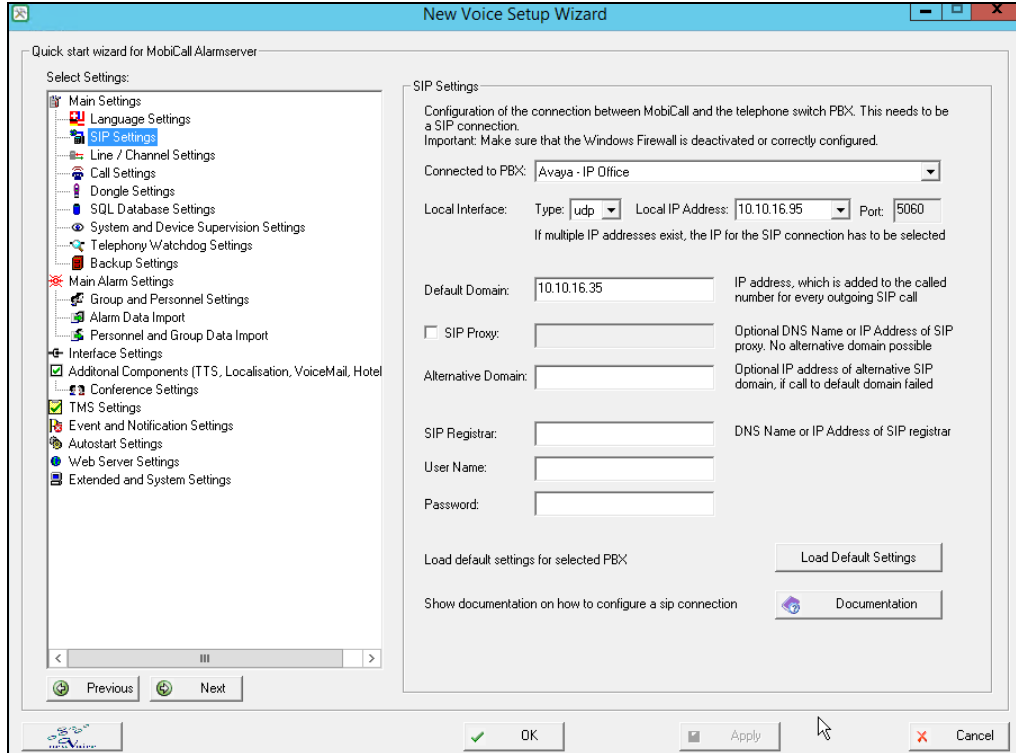
Select Main Setting → SIP Settings from the left hand menu



Load the necessary PBX profile



NewVoice Setup Wizard – SIP Settings



Fill in the required settings

Local Interface Type: UDP
Local Interface IP Address: IP Address of MobiCall
Local Interface Port: 5060
Default Domain: IP Address of your Avaya IP Office

NewVoice Alarm Central – Settings – General Settings

General Settings

Alarm Server - General Settings

Program Language: English

First Alarm Number: 2000 First Entry Number (DDI): 10

Last Alarm Number: 9999 Last Entry Number (DDI): 1999

ISDN / Analog Connection

Connected to PBX: []

Connected to PBX by ISDN (PRI / BRI) ISDN Settings

Connected to PBX by Analog Lines

VOIP Connection by H.323 or SIP

Connected to PBX: Avaya - IP Office

Settings for VOIP Connection and Gatekeeper VOIP Settings

CSTA Connection Settings

Activate CSTA Connection for special PBX Features (if supported) CSTA Settings

Global Call Settings

Prefix for External Calls: 0 Use line specific prefixes to override default prefix Settings

Prefix for GSM Calls: 0 Use line specific prefixes to override default prefix Settings

Use calling party number for outgoing calls: 111

If alarm was launched by phone, use the caller number as calling party number for all outgoing calls

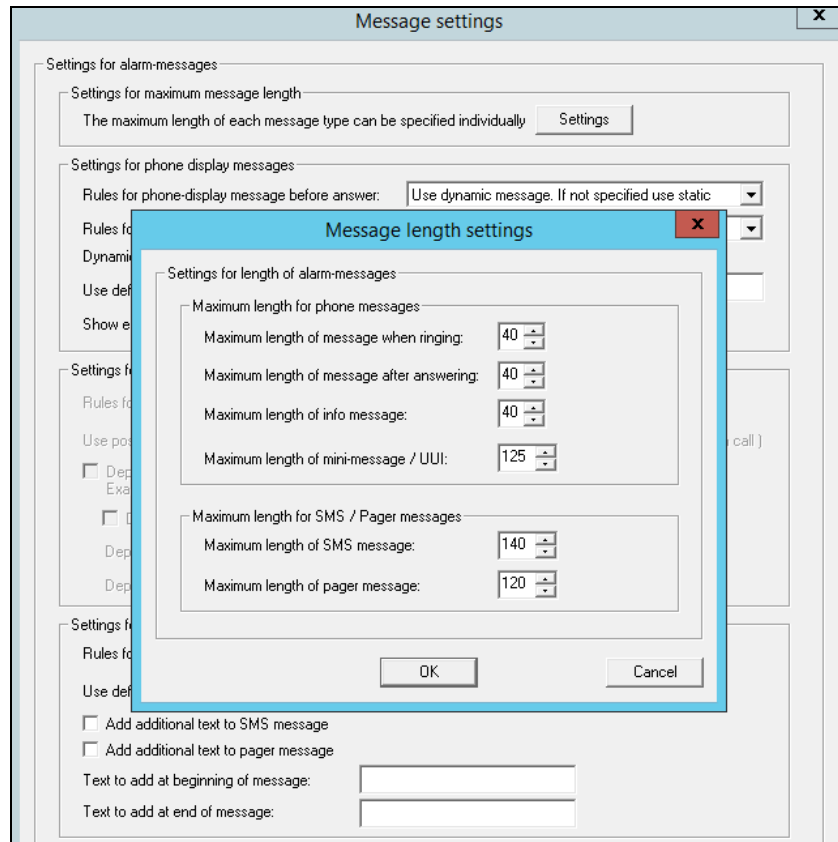
Only send calling party number for external calls (phonetype EXT and GSM)

Set number type plan for outgoing calls. Use default type: []

OK Cancel

Set the **Calling party number** for outgoing calls

NewVoice Alarm Central – Settings – Message Settings – Settings



Change the standard settings to have the maximum length supported by your phone display. Some phone types do not support that much signs, so they will cut the text at the end.

7. Verification Steps

This section describes the checks that can be carried out to verify the connection between Mobicall and IP Office

7.1. IP Office Verification

Using **IP Office System Status** select **Trunks**→**Line:x** where x is the line added above. Check that the Trunk shows **In Service** and that the trunks are Idle or Active.

The screenshot shows the IP Office System Status interface. The left sidebar contains a navigation menu with categories: System, Alarms (20), Extensions (4), Trunks (6), Active Calls, Resources, Voicemail, IP Networking, and Locations. Under Trunks (6), Line: 9 is selected. The main content area displays the 'SIP Trunk Summary' for Line 9. The summary includes the following details:

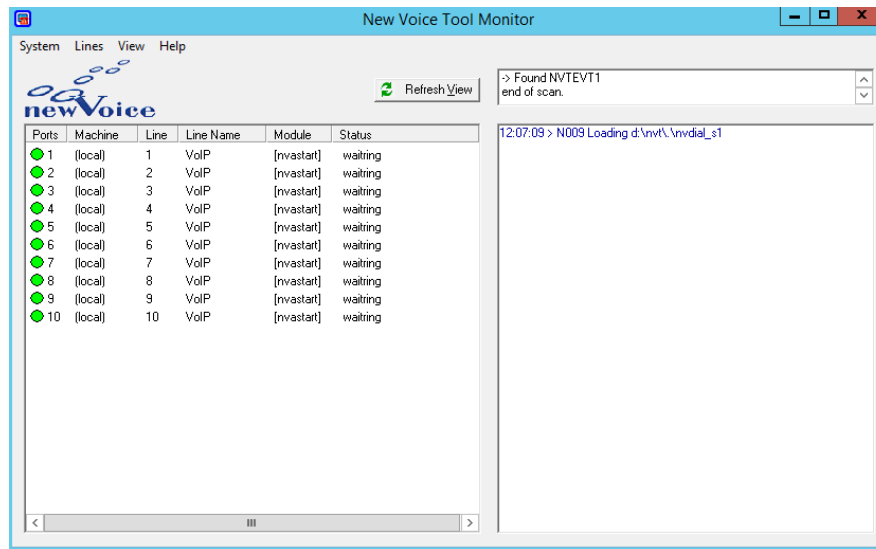
- Line Service State: In Service
- Peer Domain Name: devconnect.local
- Resolved Address: 10.10.16.95
- Line Number: 9
- Number of Administered Channels: 10
- Number of Channels in Use: 0
- Administered Compression: G729 A, G711 A
- Enable Faststart: Off
- Silence Suppression: Off
- Layer 4 Protocol: UDP
- SIP Trunk Channel Licenses: 256
- SIP Trunk Channel Licenses in Use: 0 (0%)
- SIP Device Features:

Below the summary is a table showing the status of 10 channels. All channels are currently in an 'Idle' state.

Channel Number	URI Gr...	Call Ref	Current State	Time in State	Remote Media Address	Codec	Connection Type	Caller ID or Dialed Digits	Other Party on Call
1			Idle	21:53:46					
2			Idle	1 day 20:5...					
3			Idle	1 day 20:5...					
4			Idle	1 day 20:5...					
5			Idle	1 day 20:5...					
6			Idle	1 day 20:5...					
7			Idle	1 day 20:5...					
8			Idle	1 day 20:5...					
9			Idle	1 day 20:5...					
10			Idle	1 day 20:5...					

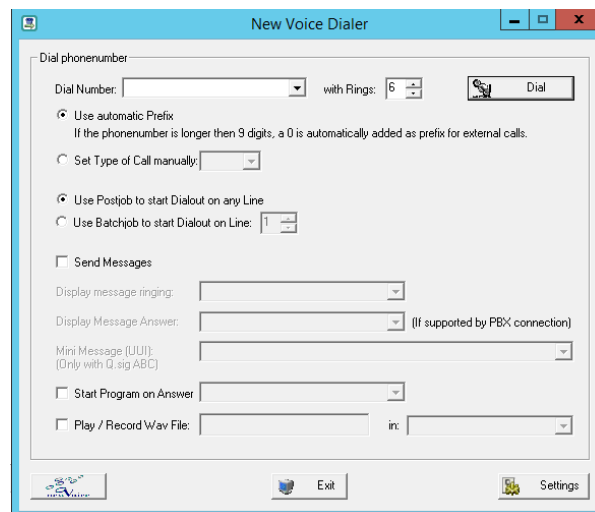
7.2. MobiCall Line Monitor and Dial utility

New Voice Tool Monitor can be found as shortcut on the desktop.



A successful communication between MobiCall and the IP Office can be verified via the tools **New Voice Tool Monitor** and the **New Voice Dial Utility**.

The **New Voice - Alarm Central** can be used to make test calls. A Shortcut to this can be found on the desktop. From the **Extras** menu select **Dial Utility** (not shown).



Enter a **Dial Number** for an extension or group on the IP Office and press **Dial** to start an outgoing call. The Paging Short Code added in **Section 5.4** must be used as a prefix for the number to use this feature.

8. Conclusion

These Application Notes describe the configuration steps required for New Voice Technologies Mobicall to interoperate with Avaya IP Office Server Edition with 500v2 Expansion. All feature functionality and serviceability test cases were completed successfully as outlined in **Section 2.2**.

9. Additional References

This section references the Avaya and New Voice Technology product documentation that are relevant to these Application Notes.

Product documentation for Avaya products may be found at <http://support.avaya.com>.

[1] *Administering Avaya IP OfficePlatform with Manager, Id: 101005673*

[2] *Using IP Office Platform System Status Id: 101005061*

Product documentation for New Voice Technologies Mobicall can be obtained by visiting the following website www.mobilisierung.com

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