



## **Avaya Solution & Interoperability Test Lab**

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# **Application Notes for New Voice Technologies Mobicall 8.0.3 with Avaya IP Office Server Edition 9.1 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 9.1 and 500v2 Expansion. Mobicall is an Alarm generation and distribution solution that connects to Session Manager as a SIP entity.

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 9.1 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 and distributed to Communication Manager endpoints.

## 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.
- Alarms can be set and distributed from IP Office to/from Mobicall.
- Priority calling and Whisper Paging functionality can be initiated from Mobicall
- Failover/Service – Tests the behaviour of Mobicall Server during certain failed conditions.

### 2.2. Test Results

All test cases were passed.

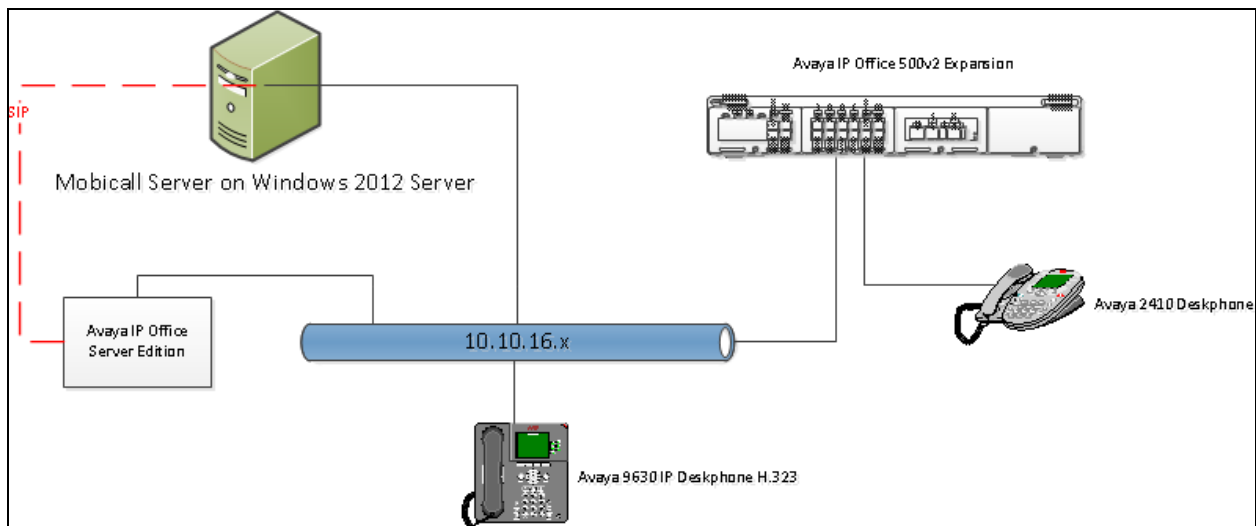
## 2.3. Support

NewVoice AG  
Militärstrasse 90, 8004 Zürich

Telephone	+41 58 750 11 11
Fax	+41 58 750 11 12
E-Mail	support@newvoice.ch
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 9.1 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	9.1 R017.00.0.441.22438
Avaya IP Office 500v2	9.1
Avaya 96xx Series IP Deskphones H.323	3.2
Avaya 2420 Series Digital Deskphones	N/A
Mobicall	8.0.3

## 5. Configure Avaya IP Office

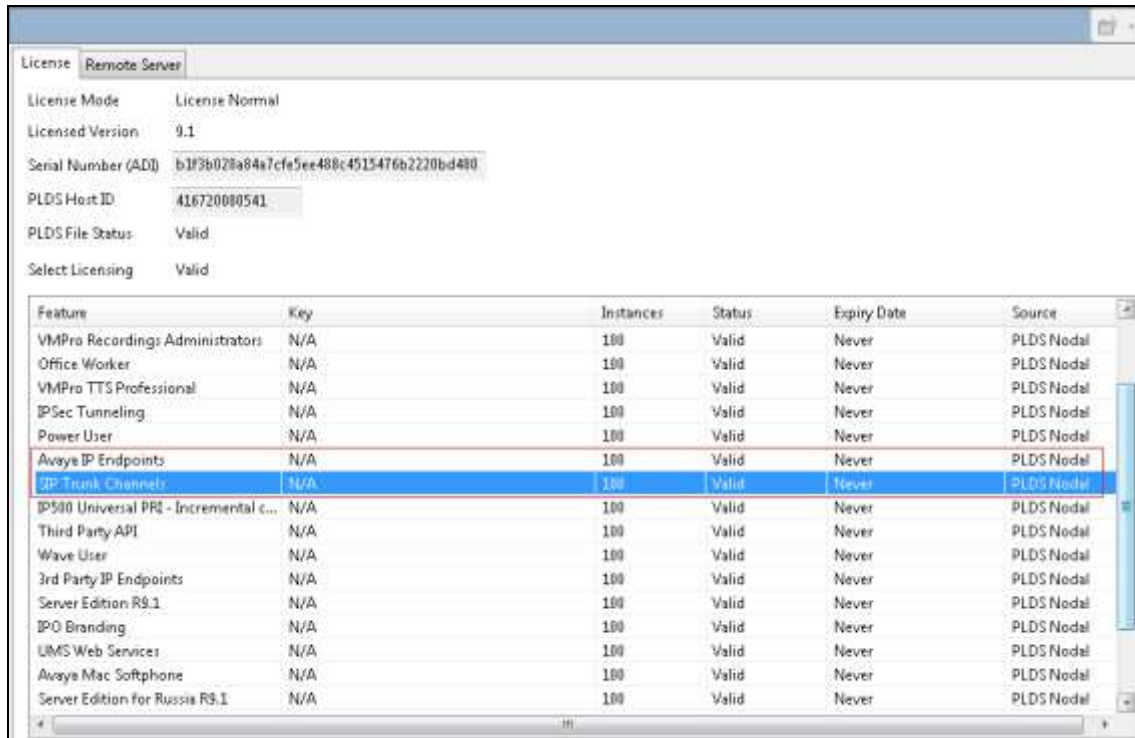
This section describes the steps required to allow IP Office to communicate with Mobicall. It is assumed that IP Office is installed and configured before implementing the configuration step below. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 10**.

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

Configuration steps include:

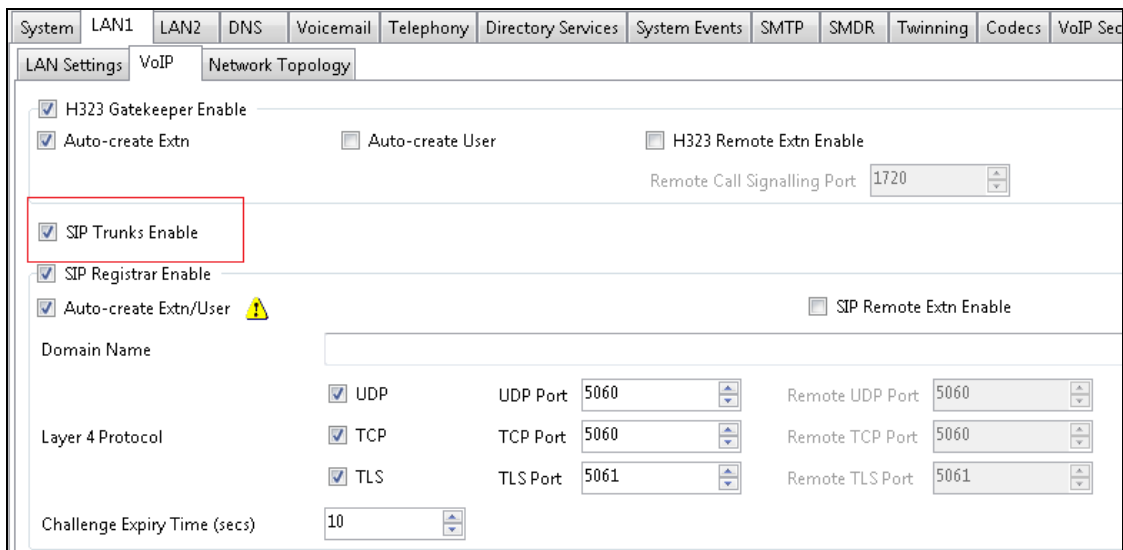
- Check SIP Trunk Licensing.
- Administer System Lan settings.
- Administer Short Codes for routing and feature.
- Administer SIP Line.

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	Key	Instances	Status	Expiry Date	Source
VMPro Recordings Administrators	N/A	100	Valid	Never	PLDS Nodal
Office Worker	N/A	100	Valid	Never	PLDS Nodal
VMPro TTS Professional	N/A	100	Valid	Never	PLDS Nodal
IPSec Tunneling	N/A	100	Valid	Never	PLDS Nodal
Power User	N/A	100	Valid	Never	PLDS Nodal
Avaya IP Endpoints	N/A	100	Valid	Never	PLDS Nodal
<b>SIP Trunk Channels</b>	N/A	100	Valid	Never	PLDS Nodal
IP500 Universal PRS - Incremental c...	N/A	100	Valid	Never	PLDS Nodal
Third Party API	N/A	100	Valid	Never	PLDS Nodal
Wave User	N/A	100	Valid	Never	PLDS Nodal
3rd Party IP Endpoints	N/A	100	Valid	Never	PLDS Nodal
Server Edition R9.1	N/A	100	Valid	Never	PLDS Nodal
IP0 Branding	N/A	100	Valid	Never	PLDS Nodal
UMS Web Services	N/A	100	Valid	Never	PLDS Nodal
Avaya Mac Softphone	N/A	100	Valid	Never	PLDS Nodal
Server Edition for Russia R9.1	N/A	100	Valid	Never	PLDS Nodal

Select the **Server edition** → **System** (not shown) and check that **SIP Trunks Enable** is selected.



System LAN1 LAN2 DNS Voicemail Telephony Directory Services System Events SMTP SMDR Twinning Codecs VoIP Sec

LAN Settings VoIP Network Topology

☒ H323 Gatekeeper Enable

☒ Auto-create Extn ☐ Auto-create User ☐ H323 Remote Extn Enable

Remote Call Signalling Port 1720

☒ **SIP Trunks Enable**

☒ SIP Registrar Enable

☒ Auto-create Extn/User ☐ SIP Remote Extn Enable

Domain Name

Layer 4 Protocol

☒ UDP UDP Port 5060 Remote UDP Port 5060

☒ TCP TCP Port 5060 Remote TCP Port 5060

☒ TLS TLS Port 5061 Remote TLS Port 5061

Challenge Expiry Time (secs) 10

From **Solution→Short Code** (not shown) right click and select **New**(not shown). Enter the code you wish to use for Priority Call followed by **N**, Select **Priority Call** from the **Feature** drop down and enter **N** as the **Telephone Number**

A screenshot of a web-based configuration form titled "Short Code". The form contains several fields: "Code" with the value "\*79N", "Feature" with a dropdown menu showing "Priority Call", "Telephone Number" with the value "N", "Line Group ID" with the value "0", "Locale" with a dropdown menu, "Force Account Code" with an unchecked checkbox, and "Force Authorization Code" with an unchecked checkbox. A red box highlights the "Code" field, and another red box highlights the "Feature" dropdown menu. A red asterisk note below the "Code" field states: "\* This Short Code is common to all systems."

From **Solution→Short Code** (not shown) right click and select **New** (not shown). Enter the code you wish to use for Whisper Page followed by **N**, Select **Whisper Page** from the **Feature** drop down and enter **N** as the **Telephone Number**.

A screenshot of a web-based configuration form titled "Short Code". The form contains several fields: "Code" with the value "\*80N", "Feature" with a dropdown menu showing "Whisper Page", "Telephone Number" with the value "N", "Line Group ID" with the value "0", "Locale" with a dropdown menu, "Force Account Code" with an unchecked checkbox, and "Force Authorization Code" with an unchecked checkbox. A red box highlights the "Code" field, and another red box highlights the "Feature" dropdown menu. A red asterisk note below the "Code" field states: "\* This Short Code is common to all systems."

From **Solution→Short Code** (not shown) right click and select **New**(not shown). Enter the code you wish to use for Call Intrude followed by **N**, Select **Call Intrude** from the **Feature** drop down and enter **N** as the **Telephone Number**.

A screenshot of a web-based configuration form titled "Short Code". The form contains several fields: "Code" with the value "\*81N", "Feature" with a dropdown menu showing "Call Intrude", "Telephone Number" with the value "N", "Line Group ID" with the value "0", "Locale" with a dropdown menu, "Force Account Code" with an unchecked checkbox, and "Force Authorization Code" with an unchecked checkbox. A red box highlights the "Code" field, and another red box highlights the "Feature" dropdown menu. A red asterisk note below the "Code" field states: "\* This Short Code is common to all systems."

From **Solution→Short Code** (not shown) right click and select **New** (not shown). Enter the code you wish to use for Dial Paging followed by **N**, Select **Dial Paging** from the **Feature** drop down and enter **N** as the **Telephone Number**.

Short Code

Code: \*82N  
\* This Short Code is common to all systems.

Feature: Dial Paging

Telephone Number: N

Line Group ID: 0

Locale:

Force Account Code: ☐

Force Authorization Code: ☐

From **Solution**→**Short Code** (not shown) right click and select **New** (not shown). Enter the number you wish to dial to access Mobicall, 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.

Short Code

Code: 81024  
\* This Short Code is common to all systems.

Feature: Dial Direct

Telephone Number: 81024

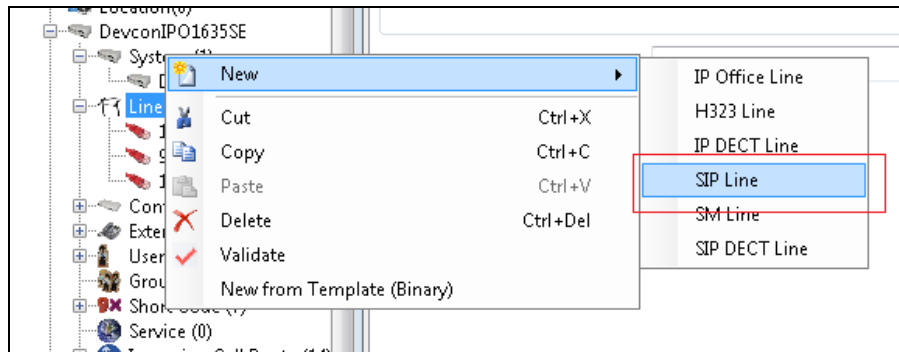
Line Group ID: 9

Locale: Ireland (UK English)

Force Account Code: ☐

Force Authorization Code: ☐

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				
ITSP Domain Name		10.10.16.95				
URI Type		SIP				
Location		Cloud				
Prefix						
National Prefix		0				
International Prefix		00				
Country Code						
Name Priority		System Default				
Description						
In Service		<input checked="" type="checkbox"/>				
Check OOS		<input checked="" type="checkbox"/>				
Session Timers						
Refresh Method		Auto				
Timer (seconds)		On Demand				
Forwarding and Twinning						
Originator number						
Send Caller ID		None				
Redirect and Transfer						
Incoming Supervised REFER		Auto				
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 enter \* for **Local URI**, **Contact** and **Display Name**. 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

Channel	Groups	Via	Local URI	Contact	Display Name	PAI	Credential	Max Calls
1	9 9	<...>	*	*	*	N...	0: <Non...	10

Add... Remove Edit...

Edit Channel

Via <None>

Local URI \*

Contact \*

Display Name \*

PAI None

Registration 0: <None>

Incoming Group 9

Outgoing Group 9

Max Calls per Channel 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 tab. The 'Identity' section is expanded, displaying the following options:

- Use Phone Context: ☐
- Add userphone: ☐
- 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 'Addressing' section shows:

- Association Method: By Source IP address
- Call Routing Method: Request URI
- Suppress DNS SRV Lookups: ☐

The 'Media' section shows:

- Allow Empty INVITE: ☐
- Send Empty re-INVITE: ☐
- Allow To Tag Change: ☐
- P-Early-Media Support: None
- Send SilenceSupp=Off: ☐
- Force Early Direct Media: ☐
- Media Connection Preservation: Disabled

The 'Call Control' section shows:

- Call Initiation Timeout (s): 4
- Call Queuing Timeout (ms): 5
- Service Busy Response: 486 - Busy Here
- on No User Responding Send: 486-Request Timeout
- Action on CAC Location Limit: Allow Voicemail
- 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 documents provided by NewVoice. (see **Section 10**)

### 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 (nvtmt) & Diva Server CAPI (nvtaf) & VOIP (nvtoip) & AI-Logix (nv)

(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

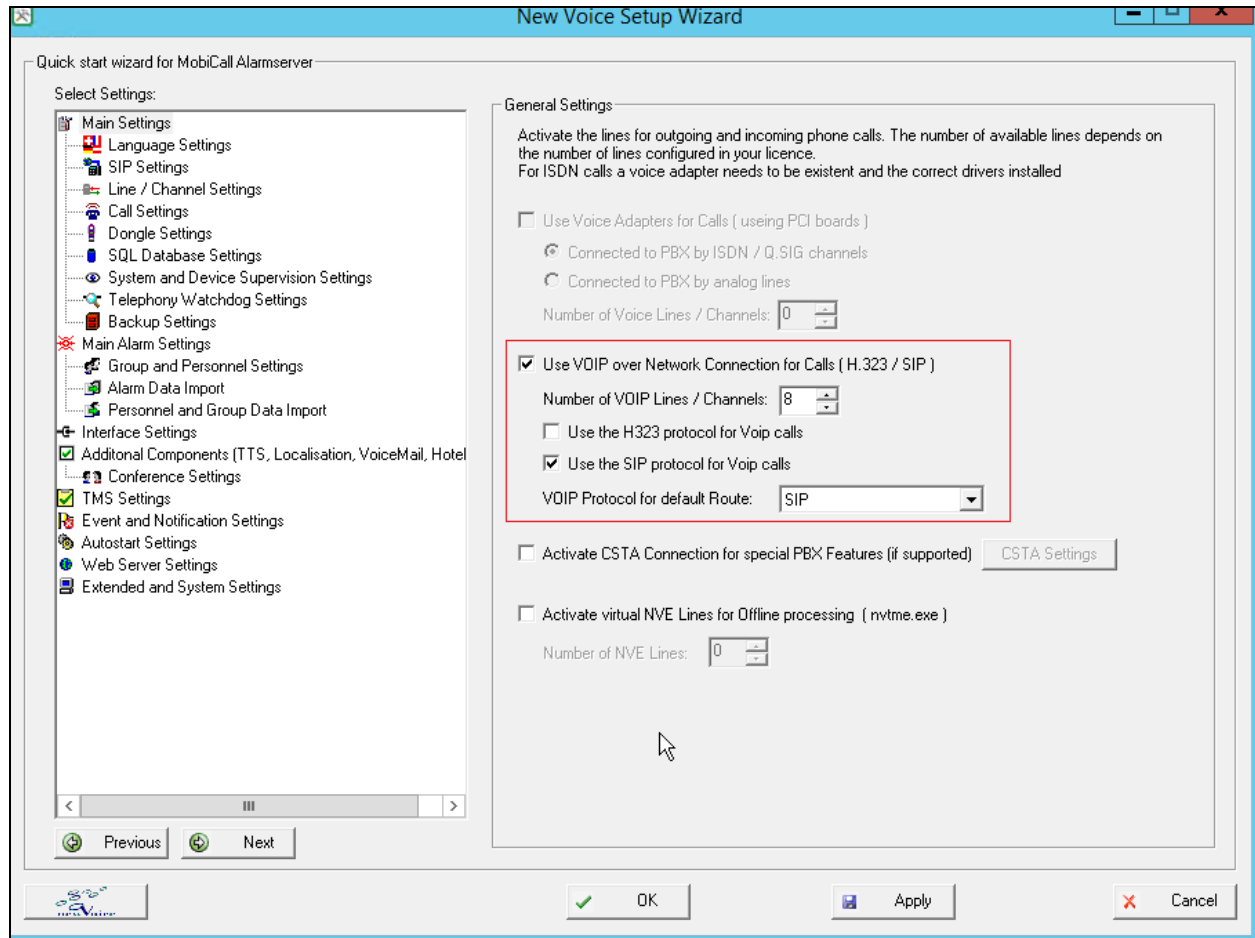
Update License

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) **invtoip** as registered type

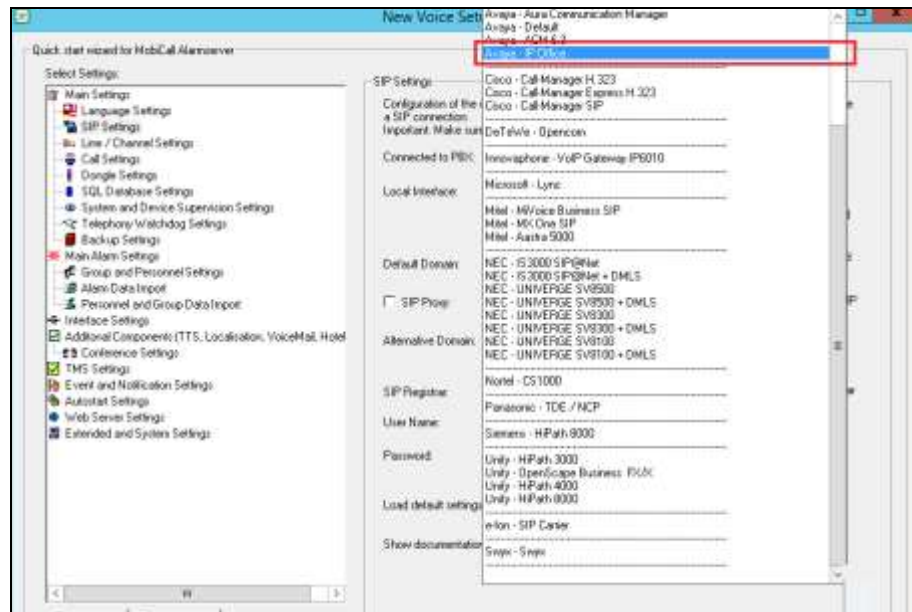
## New Voice Setup Wizard – Main Settings



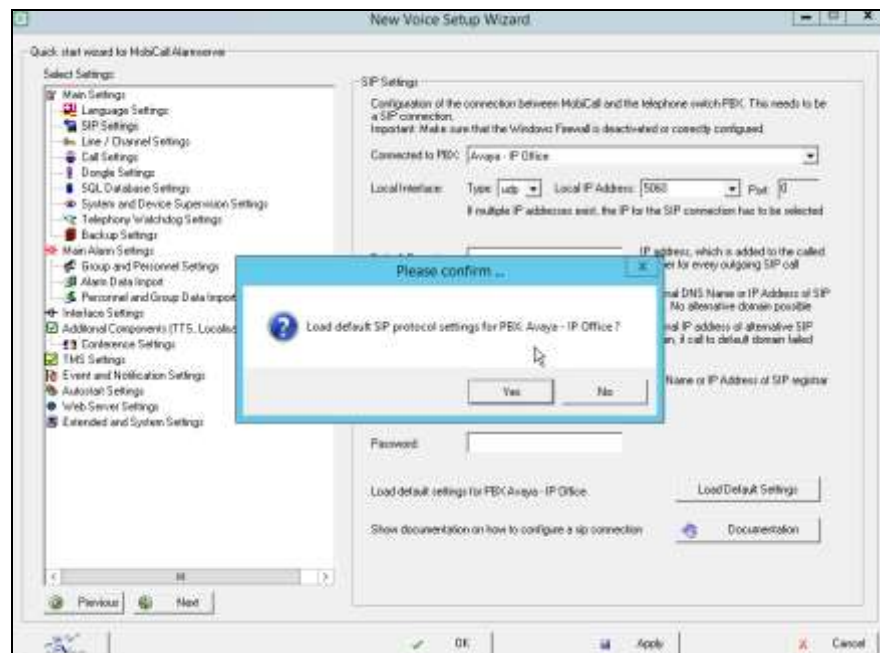
**Activate the SIP Lines** licensed by setting Number of VOIP Lines / Channels to no more than the licensed number.

## 6.2. SIP Settings

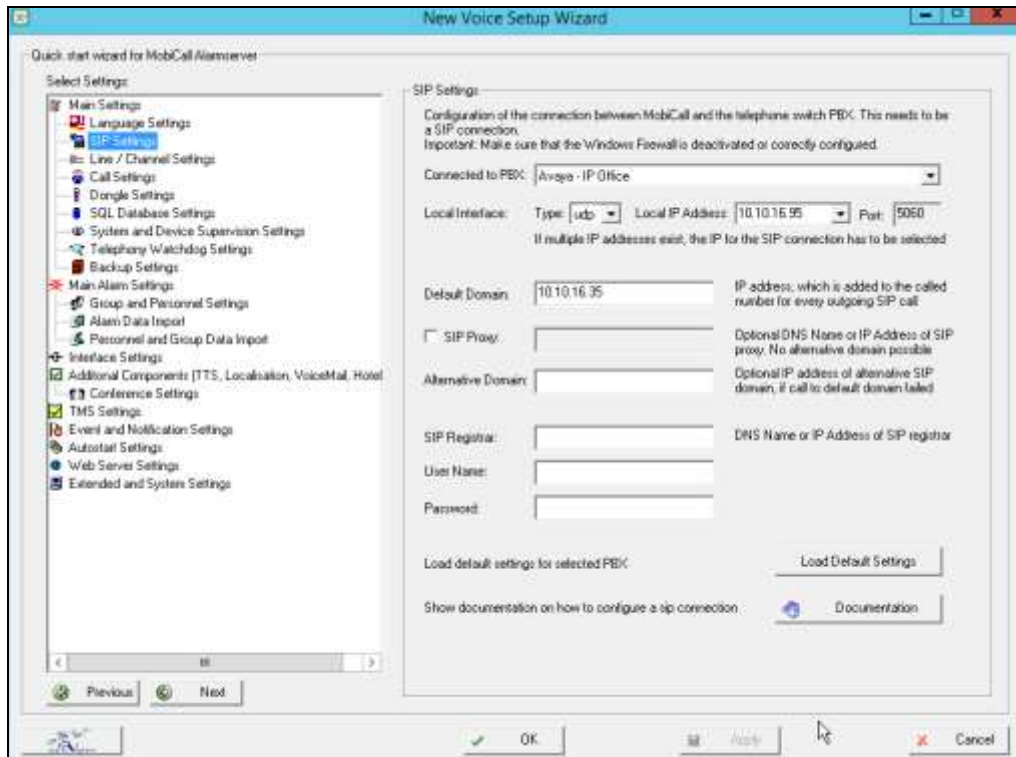
Select Main Setting → SIP Settings from the left hand menu



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: [Dropdown]

☒ 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: [Dropdown]

**OK** **Cancel**

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

## NewVoice Alarm Central – Settings – General Settings – VOIP Settings

Settings for H323 / SIP

Use Protocol as default Route: SIP

☐ Use the H323 Protocol for VOIP Calls

☐ Specify local IP Address: 0 0 0 0

☒ Use Subnet mask on IP Address: 127 0 0 1

User Name for gateway: MobileCall

Password for gateway:

Port for incoming calls:

☒ Use the SIP Protocol for VOIP Calls

Local Interface: udp\$10.10.16.95:5060

Default SIP Domain: 10.10.16.35

☐ Optional SIP Proxy:

Alternative SIP Domain:

Multiple Connections to SIP Hosts:

SIP Register (DNS Name or IP):

User Name: Password:

☐ Activate alarm option for forced release if the called number is busy, if it is supported by the PBX.  
If the internal phone number is busy, the connection is disconnected and the alarm call is repeated.

☒ Activate alarm option for intrusion if the called number is busy, if it is supported by the PBX.  
If the internal phone number is busy, the intrusion message is played into the connection and the alarm call is repeated.

☒ Activate alarm option for broadcast call, if it is supported by the PBX.  
The loudspeaker is opened with internal call and the alarm message is played immediately.

Extended VOIP settings for advanced users: Advanced Settings

Load the default configuration settings for the selected PBX: Load default Settings

Intrusion Settings

Broadcast Settings

Activate the **necessary features** (only if supported by your PBX)

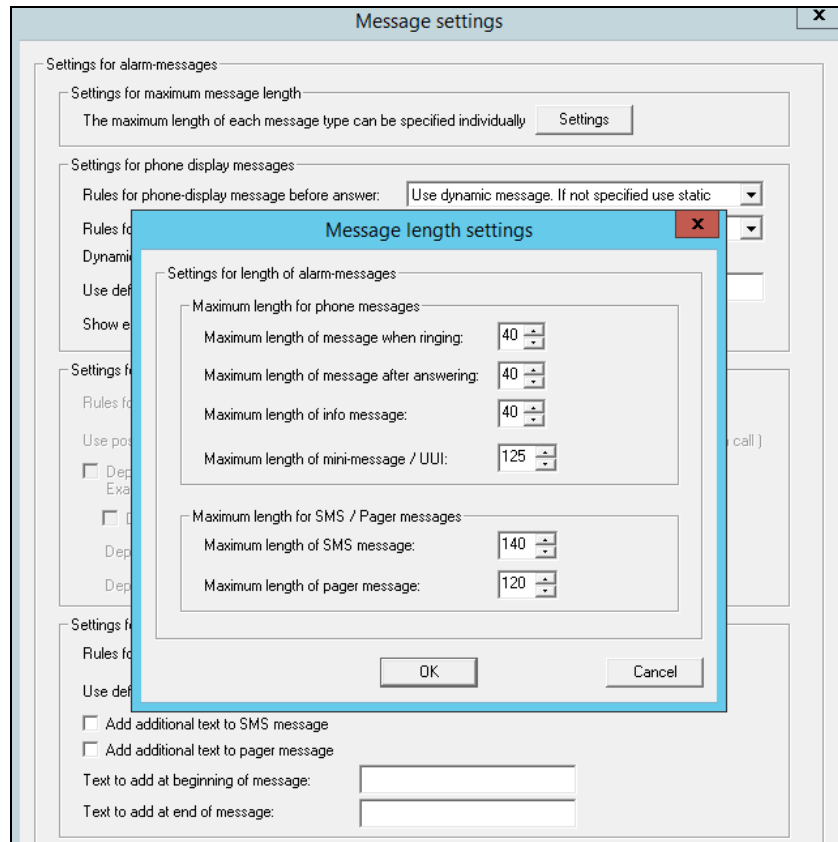


## NewVoice Alarm Central – Settings – General Settings – VOIP Settings – Intrusion Settings

The screenshot displays the 'VOIP Settings' window in the NewVoice Alarm Central application. The 'Settings for H323 / SIP' section is active, showing options for protocol selection and IP configuration. A secondary window, 'Intrusion Settings', is overlaid on top. This window contains the 'Settings for Call Intrusion' section, which has two input fields: 'Prefix to dial to activate Intrusion:' with the value '381' and 'Set specified CLI for Intrusion Call:' with the value '55555'. Both fields are highlighted with red rectangles. The 'Intrusion Settings' window also has 'OK' and 'Cancel' buttons. In the background, the 'VOIP Settings' window shows various configuration options, including 'Use Protocol as default Router' set to 'SIP', 'Use the H323 Protocol for VOIP Calls' checked, and 'Local Interface' set to 'Default SIP Domain'. There are also checkboxes for 'Optional SIP Proxy' and 'Multiple Connections to SIP'. At the bottom of the 'VOIP Settings' window, there are links for 'Advanced Settings' and 'Load default Settings'.

Prefix for Intrusion: see PBX installation section  
CLI for Intrusion Call: see PBX installation section

## 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

Avaya IP Office System Status - DevconIPO1635SE (10.10.16.35) - IP Office Linux PC 9.1.4.0 build 137

### IP Office System Status

Help Snapshot LogOff Exit About

- System
- Alarms (3)
- Extensions (0)
- Trunks (3)
  - Line:1
  - Line:9
  - Line:10
- Active Calls
- Resources
- Voicemail
- IP Networking
- Locations

#### SIP Trunk Summary

Line Service State: In Service  
Peer Domain Name: 10.10.16.95  
Resolved Address: 10.10.16.95  
Line Number: 9  
Number of Administered Channels: 10  
Number of Channels in Use: 0  
Administered Compression: G711 A, G711 Mu, G729 A  
Enable Faststart: Off  
Silence Suppression: Off  
Layer 4 Protocol: UDP  
SIP Trunk Channel Licenses: 100  
SIP Trunk Channel Licenses in Use: 0  
SIP Device Features:

Chan...	U...	Call	Current State	Time	Rem...	C...	Con...	Caller	Other	Dire...	Ro...	Rec...	Rec...	Tra...	Tra...
Ref				in S...				ID ...	Party o...						
1			Idle	4 d...											
2			Idle	4 d...											
3			Idle	4 d...											
4			Idle	4 d...											
5			Idle	4 d...											
6			Idle	4 d...											
7			Idle	4 d...											
8			Idle	4 d...											
9			Idle	4 d...											
10			Idle	4 d...											

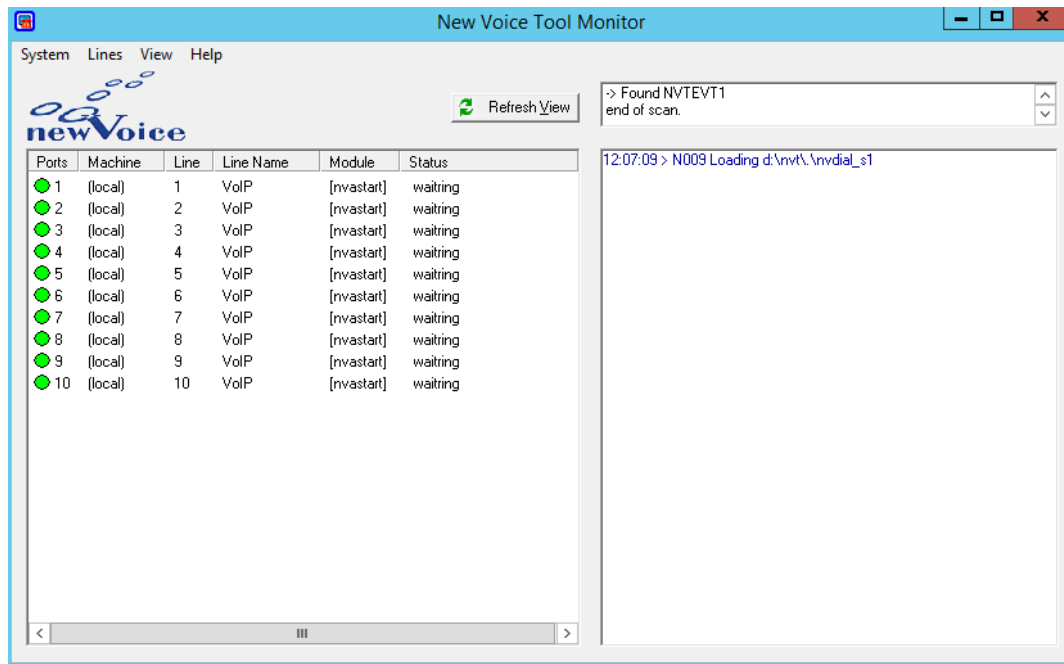
Trace Trace All Pause Ping Call Details Graceful Shutdown Force Out of Service

Print... Save As...

07:53:14 Online

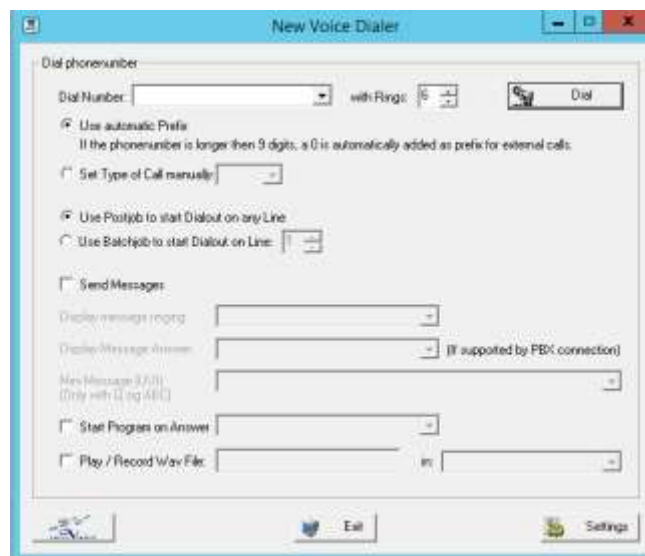
## 7.2. MobiCall Line Monitor and Dial utility

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



A successful communication between MobiCall and the Session Manager can be verified via the tools “New Voice Tool Monitor” and the “New Voice Dial Utility”

Alarmcentral – Extras – Dial Utility



Enter an internal number and press “dial” to start an outgoing call

## 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](http://www.mobilisierung.com)

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