

Avaya Solution & Interoperability Test Lab

## 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

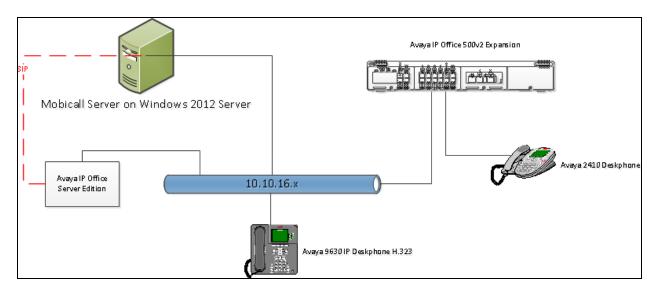


Figure 1: Connection of Mobicall with Avaya IP Office Server Edition 9.1 and 500v2 Expansion

# 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 Manger under **Configuration** select **Licenses** and check the number of SIP Trunk Channels are enough for the Mobicall SIP trunk requirements

							ĊÌ.
icense Remote Serv	/e/						
License Mode	License Normal						
Licensed Version	9.1						
Serial Number (ADI)	b3f3b028a84a7	fe5ee488c4515476b2220bd480					
PLDS Host ID	416720080541						
PLDS File Status	Valid						
Select Licensing	Valid						
select Licensing	Valid	11.55			210 IN IS 200 YO		_
Feature		Key	Instances	Status	Expiry Date	Source	
VMPro Recordings	Administrators	N/A	180	Valid	Never	PLDS Nodal	
Office Worker		N/A	199	Valid	Never	PLDSNodal	
VMPro TTS Professi	ional	N/A	100	Valid	Never	PLDSNodal	
IPSec Tunneling		N/A	100	Valid	Never	PLDS Nodal	
Power User		N/A	100	Valid	Never	PLDS Nodal	
Avays IP Endpoints		N/A	100	Valid	Never	PLDS Nodel	T
SIP Trunk Channels		N/A	100	Valid	Nevel	PLDS Nodel	
IP500 Universal PR	- 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	PLDSNodal	
3rd Party IP Endpoir	nts	N/A	100	Valid	Never	PLDS Nodal	
Server Edition R9.1		N/A	190	Valid	Never	PLDS Nodal	
IPO Branding		N/A	180	Valid	Never	PLDS Nodel	
UMS Web Services		N/A	100	Valid	Never	PLDS Nodel	
Avaya Mac Softpho	ne	N/A	190	Valid	Never	PLDS Nodal	
Server Edition for R	ussia R9.1	N/A	100	Valid	Never	PLDS Nodel	
	errana e territa i		- HI	VIN W	a bet we have		

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

System LAN1 LAN2 DNS Vo	icemail Telephony	Directory Ser	vices	System Events	SMTP	SMDR	Twinning	Codecs	VoIP Sec
LAN Settings VoIP Network Topology									
· ↓ H323 Gatekeeper Enable									
📝 Auto-create Extn	🔲 Auto-create Us	ser		🔲 H323 Rem	ote Extn I	Enable			
				Remote Call S	ignalling	Port 17	20	A V	
SIP Trunks Enable									
SIP Registrar Enable					_				
🔽 Auto-create Extn/User 🔥						SIP Rer	note Extn Er	nable	
Domain Name									
	VDP	UDP Port	5060	* *	Rem	iote UDP	Port 5060		A V
Layer 4 Protocol	📝 ТСР	TCP Port	5060		Rem	iote TCP	Port 5060		×
	🔽 TLS	TLS Port	5061	* *	Rem	iote TLS F	ort 5061		A V
Challenge Expiry Time (secs)	10								

Solution & Interoperability Test Lab Application Notes ©2015 Avaya Inc. All Rights Reserved. From **Solution** $\rightarrow$ **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** 

Short Code		
Code	*79N	
	* This Short Code is common to all systems.	
Feature	Priority Call	-
Telephone Number	N	
Line Group ID	0	•
Locale		•
Force Account Code		
Force Authorization Code		

From **Solution** $\rightarrow$ **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**.

*80N	
* This Short Code is common to all systems.	
Whisper Page	-
N	
0	•
1	•]
10	
	* This Short Code is common to all systems. Whisper Page N 0

From **Solution** $\rightarrow$ **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**.

Short Code		
Code	*BIN	
	* This Short Code is common to all systems.	
Feature	Call Intrude	
Telephone Number	N	1
Line Group ID	0	-
Locale		•
Force Account Code	(C)	
Force Authorization Code	12	

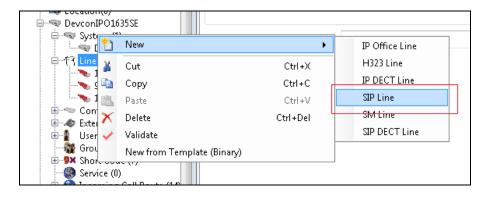
From **Solution** $\rightarrow$ **Short Code** (not shown) right click and select **New** (not shown). Enter the code you wish to use for Dial Paging followed by **N**, Select Dial Paging from the **Feature** drop down and enter N as the **Telephone Number**.

Shart Code		
Code	*82N	)
	* This Short Code is common to all systems.	
Feature	Dial Paging	•
Telephone Number	N	Ĩ
Line Group ID	0.	· · · · · · · · · · · · · · · · · · ·
Locale		•
Force Account Code	20	
Force Authorization Code	E	

From **Solution** $\rightarrow$ **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		٦
_	* This Short Co	de is common to all systems.	
Feature	Dial Direct		•
Telephone Number	81024		
Line Group ID	9	•	•
Locale	Ireland (UK Eng	lish)	•
Force Account Code			
Force Authorization Code			

From the **Server Edition→Line** right cick 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

Line Number	9.		In Service		
ITSP Domain Name	10.10.16.95		Check 00S	1921	
URIType	SP.		Session Timers		
Location	Cloud		Refresh Method	Auto	
	00	6	Timer (seconds)	On Demand	
Prefix			Forwarding and Twinning		
National Prefix	0		Originator number		
International Prefix	00	1	Send Caller ID	None	
Country Code		-	Redirect and Transfer		
Name Priority	System Default		Incoming Supervised REFER	Auto	( <b>•</b> )
Description			Outgoing Supervised REFER	Auto	2 <b>4</b> )
			Send 382 Moved Temporarily	13	
			Outgoing Blind REFER	10	

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	0.16.95
Network Configuration -	
Layer 4 Protocol	UDP   Send Port 5060
Use Network Topology Ir	nfo None 🔻 Listen Port 5060 👻
Explicit DNS Server(s)	
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 SIF	Credentials SIP Advanced Engineer	ing	
Channel Groups Via Loca	URI Contact Display Name PAI	Credential Max Calls	Add
1 99 <*	* * N	0: <non 10<="" td=""><td>Remove</td></non>	Remove
			Edit
- Edit Channel			ок
Via	<none></none>		
Local URI	*	-	Cancel
Contact	<b>*</b>	-	
Display Name	<b>*</b>	-	
PAI	None	•	
Registration	0: <none></none>	•	
Incoming Group	9		
Outgoing Group	9		
Max Calls per Channel	10		

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Addressing		Media		
Association Method	By Source IP address	Allow Empty INVITE	10	
Call Routing Method	Request URI *	Send Empty re-INVITE Allow To Tag Change	5	
Suppress DNS SRV Lookups	E	P-Early-Media Support	None	
Identity		Send SilenceSupp=Off Force Early Direct Media	0	
Use Phone Context Add user=phone	5	Media Connection Preservation	Disabled	
Use + for International	10			
Use PAL for Privacy	13	Call Control		
Use Domain for PAI Swap From and PAI	10	Call Initiation Timeout (3)	4 重	
Caller ID from From header	190	Call Queuing Timeout (m)	5 15	
Send From In Clear	121	Service Bury Response	416 - Busy Here	
Cache Auth Credentials User-Agent and Server Headers	191	on No User Responding Send	408-Request Timeout	
		Action on CAC Location Limit	Allow Voicemail	
		Suppress Q.850 Reason Header	8	
		Emulate NOTIFY for REFER	10	
		No REFER if using Diversion	E)	

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

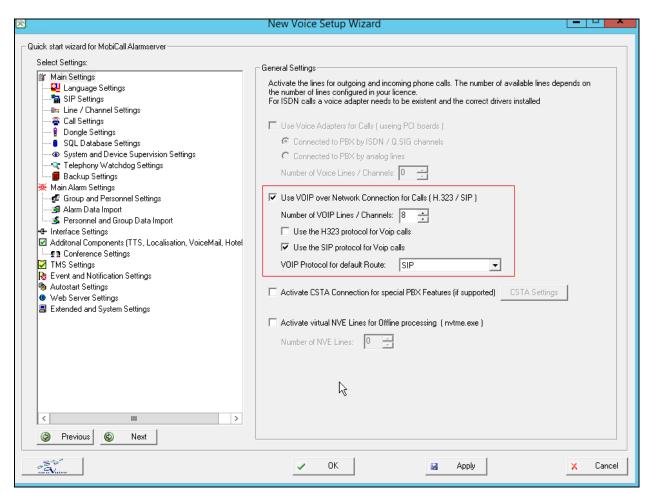
Open New Voice Setup Wizard – Select **Main Settings**  $\rightarrow$  **Dongle Settings** from the left hand menu

次	New Voice Setup Wizard	
<ul> <li>Quick start wizard for MobiCall Alarmserver</li> <li>Select Settings:</li> <li>Main Settings</li> <li>Dongle Settings</li> <li>SQL Database Settings</li> <li>SQL Database Settings</li> <li>SQL Database Settings</li> <li>Subsective Supervision Settings</li> <li>Telephony Watchdog Settings</li> <li>Backup Settings</li> <li>Main Alarm Settings</li> <li>Group and Personnel Settings</li> <li>Alarm Data Import</li> <li>Interface Settings</li> <li>Additonal Components (TTS, Localisation, VoiceMail, Hote</li> <li>Conference Settings</li> <li>TMS Settings</li> <li>Autostart Settings</li> <li>Web Server Settings</li> <li>Extended and System Settings</li> </ul>	Dongle Settings         A dongle is a USB device used to protect our software. To run a MobiCall a connect the dongle to a USB port on the alarmserver.         To activate the dongle, the licence has to be updated with the code receive Contact New Voice to receive your dongel update code.         ✓ Activate the dongle supervision. If no dongle exists (demo version), dea If the dongle is removed, following alarm is started:         9860 Dongle N         □ Deactivate the dongle supervision alarm until the dongle is connect         Dongle Information:       ② Refresh         (1)       New Voice Tool Version:       8.0 Professional I         Registered for Region:       Demo Version       [3]         Registered for Types:       Dialogic (nvtmt) & Diva Server CAPI (nvtaf) & V         (2)       Number of Lines:       60         Number of Interfaces:       128       Number of Relais:       8         Number of Interfaces:       4       Text to Speech:       1	ved from New Voice. activate this supervison watchDog ved for the first time OIP (nvtvoip) & AI-Logix (nv umber of Clients: 10
Previous     C     Next		
S S	🖌 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

Solution & Interoperability Test Lab Application Notes ©2015 Avaya Inc. All Rights Reserved. New Voice Setup Wizard - Main Settings



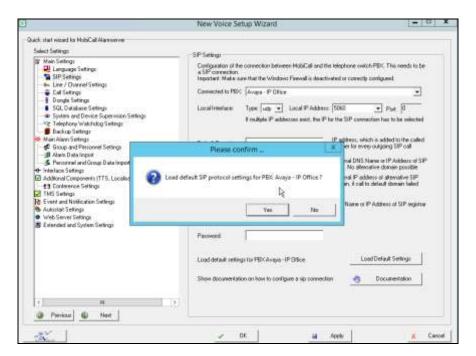
Activate the SIP Lines licensed by setting Number of VOIP Lines / Channels to no more than the licensed number.

#### 6.2. SIP Settings

xaya - Alais Connunication Manager xaya - Delauti xaya - ACH 6.2 Quick start espect for HobiCal Alarmoniver Select Settings IP Seting: Configuration of the Cloco - Call Manager H 323 Configuration of the Cloco - Call Manager Signmun H 323 SIP connection Important Make turn DeTelwa - Opercon SIPSetingi 1 Main Setting Language Setings Elit Setting:
 En Unix / Channel Setting:
 Cal Setting:
 Congle Setting:
 Output: Setting:
 StU. Dissbare Setting:
 Sub. Dissbare Setting:
 Sub. Setting:
 Sub. Setting:
 Setting: Supervision Setting:
 Setting: Setting:
 Setting: Setting:
 Setting: Setting: Connected to PBX Innovaphone - VolP Gateway IP6010 Microsoft - Lyne Local Interface: Mitel - MiVoice Business SIP Mitel - MC One SIP Mitel - Austra 5000 Backup Settingi Default Deniery NEC - 53000 SPNgHier DHLS NEC - UNIVERSE SV8500 + DHLS NEC - UNIVERSE SV8500 Alternative Domain Alternative Domain NEC - UNIVERSE SV8500 + DHLS Main Alam Setingi Group and Personnel Setingi Alem Data Import Additional Conception
 Additional Components
 Additional Components (TTS: Localisation, VoiceMail, Hotel ## Conference Settings TMS Settings Settings Autostat Settings Nortel - CS1000 **SPRepter** Panasonic - TDE / NCP Web Server Setting:
 Entended and System Settings Lises Name: Siemens HPath 9000 Parmond Unity - HP4th 3000 Unity - Developee Business F565 Unity - HP4th 4000 Unity - HP4th 4000 e Ion - SIP Casie Showdame SWW-SWW 1.1 н.

Select Main Setting  $\rightarrow$  SIP Settings from the left hand menu

Load the necessary PBX profile



Solution & Interoperability Test Lab Application Notes ©2015 Avaya Inc. All Rights Reserved. NewVoice Setup Wizard - SIP Settings

lect Settings	SIP Settings			
	Configuration of the a SIP connection. Important: Nalle sou Connected to PBX: Local Interface: Default Domein C SIP Progr. Alternative Domain SIP Registrat User Name: Persecet Lood default setting	Type: Judo - Local IP Addien: U multiple IP addiences exeit, the IP	Valed of correctly confi (10,10,16.95) Its the SIP connection IP address, which is manufact for SIP connection Diptor ID MS Name procy. No adversalism Optional IP address of domain, if call to defin DNIS Name or IP Add Load De	pued Part: 5060 has to be selected added to the called going SIP call or IP Address of SIP domain parable if atomistive SIP aff domain failed
8 3				

Fill in the required settingsLocal Interface Type:UDP (!)Local Interface IP Address:IP Address of MobiCallLocal Interface Port:5060Default Domain:IP Address of your Avaya IP Office

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

General Settings	X
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)	
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	
$\Box$ 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:	
0K Cancel	

Set the Calling party number for outgoing calls

Lite Photocol as default Route:	SIP				
T. Use the H323 Plotocol No VOIP Ca	h				
F Instruction		B	. 0	8	Cost recent 4 month from you IP addresses are defend for the proce-
F Gar Balaterer will Address	127	B.	. 8	1	If the gate/separate straining on the local server, check that graph ass is enabled in autostat. For multiple IP addresses specify the correct IP and not the locathost IP (127.0.0.)
Liver Kinger for Statestanger	Nutriet	4	_	_	
Farmed dive gardeness.	-	_	_	-	·
Particle to recently calls					Specify of professional activation of activated As the second activation of the activation metal of the profession of the activation of the second of the profession of the second of th
Vice the SIP Protocol for VOIP Calls					
LocalInteface	udp\$10.10	0.16.95	5060		Format: (Protocol Types \$4P Addesso: (Port Number) Exercise: udb\$10.97 16.37 5060
Default SIP Domain	10.10.16.1 PorDNS.	12/22	added to	Bec	alled number for every outgoing SIP call
Coptional SIP Picogi	1	1621010			Optional Proc DNS many of the SIP programmed to study, many simply
Attendive SIP Domain	-			_	Optional IP is DNS of alternative SIP domain, if call to default domain failed
Multiple Connections to SIP Hosts:	-				Setro
SPReprint (DNS Name or IP)	-		_	-	
	-		_	_	Panoword
User Nane			100210	140	
Activate alarm option for forced refe	ase I the cal	lednund	ber it bully		is pointed by the PBX
Autivate alarm option for forced rele     If the internal phone number is buse     Activate alarm option for intrusion if	the connect the called nu	tion is di mber is t	convecte	ed and	(the alam call is reported orted by the PEX. Interview Cathorn
Autivate alarm option for forced rele     If the internal phone number is buse     Activate alarm option for intrusion if	the connect the called nu , the intrusion call, it is sup	tion is di unber is t n mercad oported t	aconvecte bully, if it is se is playe by the PED	ed and s supply sd into t x	I de alam cali in recented. orted by the PSX the connection and the alam cali is recealed Provident California
Activate alarn option for locaed ele chivate alarn option for invester in book Activate alarn option for intrusion if If the internet phone number in book Activate alarn option for broadcast	The called nu the called nu the influsion call, if it is sup renal call and	tion is di unber is t n mercad oported t	acconnecte buny, il il in pe in plave by the PBD minimulage	ed and t suppo d into 1 X. e is pla	I de alam cali in recented. orted by the PSX the connection and the alam cali is recealed Provident California

NewVoice Alarm Central – Settings – General Settings – VOIP Settings

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

NewVoice Alarm Central - Settings - General Settings - VOIP Settings - Intrusion Settings

		0.07	ion seconds	1
etings for H323 / SIP				
Use Protocol as default floute:	SIP		1	
Use the HGZI Protocol for VOIP Ca	ste:			
C Sanda Sand Frances	0 . 0	0 0	Engl conducts some two and IP addresses an defend for the preser	
P the Casterne of P Assess	127 0 .	a 1	The gatabases is running on the local server, check that grupp, exe is enable subsists. For multiple IP addresses specify the correct IP and not the localitor.	ed in IP (127.0.0.1)
Districtance his March scoper.	HolluCal			
Partners for publication	Î			
Portes to watergrafts	ſ		Apartic at previous sectors	
🖗 Use the SP Protocol ha VD		Intr	xuion Settings	
	Settings for Call Intrusio			
Local system	er ver der ver sind der			
Default SIP Domain:	Prefix to dial to active	te Dritniser/	281	
C Optional SIP Prove	and the second second			
C Optional SIP Procy	Set specified CLI for In	ntrausen Calls	SSSS Annu Mark	
F* Optional SIP Prosy Alternative SIP Domain	Set specified CLI for In	ntrausen Call:	ssssoj vili doran faled	
	Set specified Q.I for In	ntrauen Cali:	10000	na
Alternative SIP Domain.	Set specified Q.1 for (	ntrausen Calit	all doman failed	w]
Alternative SIP Domain. Multiple Connections to SIP F	Set specified GLI for 1	ntrausen Calit	pail donan taked	ng
Alternative SIP Donaies Multiple Convections to SIP1 SIP Registria ( DNS None o User Name:	1	[	COX Careal	ne j
Attendive SP Dosen. Multiple Convections to SPH SP Registra (DNS Note o User Name: The retend phone marker in buy	nos i the called runder	r is busy, if it is provided and		_
Alternative SIP Domain. Multiple Convections to SIP is SIP Registrat (DNS None o User Name: Class Name: Activate altern option fair force-ordered If the reternal phone number is box? Activate activity and phone number in two?	non if the called number for connection is disc	t is busy. ## is presented and sy, ## is support	COX     Cannol     Proteined by the PRX     the adverse of a repeated	_
Alternative SIP Domain. Multiple Convections to SIP is SIP Registrat (DNS None o User Name: Class Name: Activate altern option fair force-ordered If the reternal phone number is box? Activate activity and phone number in two?	The called number in but Net convection in data Free called number in but I free influsion message	r In busy, if it is connected and sy, if it is support to played into 1 the PBOL	Comparing and the second	tingi I
Attensive SP Donain. Multiple Convections to SPFI SP Registra (DNS None o User Name: Activate adars option for force or relation i The interest phone results is back The interest phone results is back The interview option to back	neon if the called number . The convectors is disc . The called number is too . The information message call, if is supported by prior call and the alarma	r is busy. If it is prescribed and sy, if it is support in placed rate it is placed rate it is placed rate in the PEX. recoger is place	Comparing and the second	tingi I

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

NewVoice Alarm Central - Settings - Message Settings - Settings

ngs for alarm-r iettings for ma	iximum message length	
-	m length of each message type can be specified individually Settings	
attings for ph	one display messages	
	one display message before answer: Use dynamic message. If not specified use static	•
Rules fo	Message length settings	-
Dynami		
Use def	Settings for length of alarm-messages	
Show e	Maximum length for phone messages	
	Maximum length of message when ringing:	
ettings fi	Maximum length of message after answering:	
Rules fo	Maximum length of info message:	
Use pos		rcall )
🗖 Dep	Maximum length of mini-message / UUI:	
Exa		
	Maximum length for SMS / Pager messages	
Dep	Maximum length of SMS message:	
Dep	Maximum length of pager message: 120 🛫	
ettings fi		
Rules fo		
Use def	OK Cancel	
Add addi	tional text to SMS message	I
🗌 Add addi	tional text to pager message	

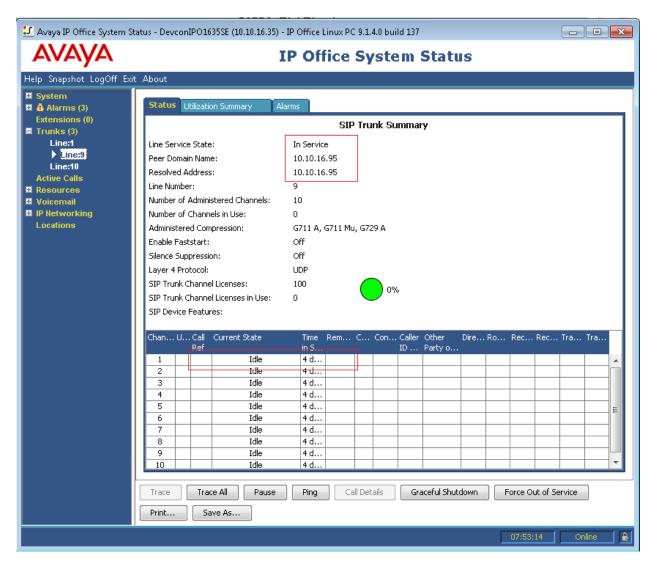
Change the standard settings to have the maximum length supported by your phone display. Some phone types do not support that much signs, so they will cut the text at the end.

# 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



#### 7.2. MobiCall Line Monitor and Dial utility

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

Ports       Machine       Line       Line       Name       Module       Status       12:07:09 > N009 Loading d:\nvt\.\nvdial_s1         1       (local)       1       VolP       [nvastart]       waitring         2       (local)       2       VolP       [nvastart]       waitring         3       (local)       3       VolP       [nvastart]       waitring         4       (local)       4       VolP       [nvastart]       waitring         5       (local)       5       VolP       [nvastart]       waitring         6       (local)       6       VolP       [nvastart]       waitring         7       (local)       8       VolP       [nvastart]       waitring         9       (local)       9       VolP       [nvastart]       waitring         10       (local)       10       VolP       [nvastart]       waitring	1     [local]     1     VolP     [nvastart]     waitring       2     [local]     2     VolP     [nvastart]     waitring       3     [local]     3     VolP     [nvastart]     waitring       4     [local]     4     VolP     [nvastart]     waitring       5     [local]     5     VolP     [nvastart]     waitring       6     [local]     6     VolP     [nvastart]     waitring       7     [local]     7     VolP     [nvastart]     waitring       8     [local]     8     VolP     [nvastart]     waitring       9     [local]     9     VolP     [nvastart]     waitring		Voi	e			🔮 Refresh⊻iew	-> Found NVTEVT1 end of scan.	
5     [local]     5     VolP     [nvastart]       6     [local]     6     VolP     [nvastart]       7     [local]     7     VolP     [nvastart]       8     [local]     8     VolP     [nvastart]       9     [local]     9     VolP     [nvastart]	5     [local]     5     VolP     [nvastart]       6     [local]     6     VolP     [nvastart]       7     [local]     7     VolP     [nvastart]       8     [local]     8     VolP     [nvastart]       9     [local]     9     VolP     [nvastart]	)1 )2 )3	(local) (local) (local)	1 2 3	VolP VolP VolP	[nvastart] [nvastart] [nvastart]	waitring waitring waitring	12.01.00 / Hobb Ebdaing a. white writing si	
9 (local) 9 VolP [nvastart] waitring	9 (local) 9 VolP [nvastart] waitring	)5 )6 )7	(local) (local) (local)	5 6 7	VolP VolP VolP	[nvastart] [nvastart] [nvastart]	waitring waitring waitring		
		9	(local)	9	VolP	[nvastart]	waitring		
							>		

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

0	New Voice Dialer	- 0 ×
Dial phonenumber		
DialNumber	💌 with Rings 🤞 🛨	Dial Dial
<ul> <li>Use automatic Prefix</li> <li>If the phonenumber is long</li> </ul>	gerihen 9 digits, a 0 is automatically added as prefix	for externel calls.
(*) Set Type of Call manually		
It Use Postiob to start Dialo	of on any Line	
C Use Balchjob to start Dial	out on Line 🗏 🚊	
🗂 Send Messages		
Tricky resources reading		
Declay Message dimonst	- Il supp	oited by PBX connection)
Marchinesague 3,000 Divity with 12 org 410(1	1	
F" Stat Program on Answer		
F Play / Record Way File:	in [	+

SJW; Reviewed: SPOC 12/18/2015 Solution & Interoperability Test Lab Application Notes ©2015 Avaya Inc. All Rights Reserved. 20 of 22 Mobicall8\_IPO91 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 <u>http://support.avaya.com</u>.

[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 <u>www.mobilisierung.com</u>

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