Avaya Solution & Interoperability Test Lab

Application Notes for Talkaphone VOIP-500 Series and VOIP-600 Series IP Call Stations with Avaya Communication Server 1000 - Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate the Talkaphone VOIP-500 Series and VOIP-600 Series IP Call Stations with Avaya Communication Server 1000. Talkaphone VOIP-500 Series and VOIP-600 Series IP Call Stations registered with Avaya Communication Server 1000 via SIP. Although not explicitly tested, these Application Notes would also apply to the Talkaphone Wide-Area Emergency Broadcast System (WEBS®) Series Devices, which leverage the same electronics and firmware with a similar subset of features (e.g. paging only with no two-way communication) as the VOIP-500 Series and VOIP-600 Series Phones but differ in form factor and packaging.

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 integrate the Talkaphone VOIP-500 Series and VOIP-600 Series IP Call Stations with Avaya Communication Server 1000 (hereafter referred as Avaya CS1000). Talkaphone VOIP-500 Series and VOIP-600 Series IP Call Stations registered with Avaya Communication Server 1000 SIP Line server via SIP. Although not explicitly tested, these Application Notes would also apply to the Talkaphone Wide-Area Emergency Broadcast System (WEBS®) Series Devices, which leverage the same electronics and firmware with a similar subset of features (e.g. paging only with no two-way communication) as the VOIP-500 Series and VOIP-600 Series Phones but differ in form factor and packaging.

2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between Talkaphone VOIP-500 Series and VOIP-600 Series IP Call Stations, Avaya SIP and Unistim telephones, and the PSTN, and exercising basic telephony features, such as hold, mute, transfer, and conference, from the Avaya IP phones. Additional telephony features, such as call forward and call coverage, were also verified.

The serviceability testing focused on verifying that the Talkaphone VOIP-500 Series and VOIP-600 Series IP Call Stations come back into service after re-connecting the Ethernet cable or rebooting the IP Call Station.

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

Interoperability compliance testing covered the following features and functionality:

- SIP registration of Talkaphone IP Call Station with Avaya CS100 SIP Line Server.
- Inbound and outbound calls between Talkaphone IP Call Station and Avaya SIP and Unistim telephones with Direct IP Media (Shuffling) enabled.
- Inbound and outbound calls between the Talkaphone IP Call Station and the PSTN.
- G.711 and G.729 codec support.
- Proper recognition of DTMF tones.
- Basic telephony features, including hold, mute, redial, transfer, and 3-way conference, initiated from the Avaya IP phone.
- Use of paging, speed-dial buttons, and number lists on the Talkaphone IP Call Station.
- Proper system recovery after a restart of the Talkaphone IP Call Station and loss of IP connectivity.

2.2. Test Results

All test cases passed with the following observation(s):

- Emergency calls cannot be terminated from the Talkaphone VOIP-500 Series and VOIP-600 Series IP Call Stations. The calls can only be disconnected by the destination phone or upon expiration of the Call Conversation Timer. The Talkaphone VOIP-500 Series and VOIP-600 Series IP Call Stations dial a list of programmed numbers in a round-robin fashion. If the first number in the list does not answer (i.e., Busy, Out of Order, Invalid number), it will call the next number in line and will keep doing so until the destination answers the call or until the 'Call Conversation Timer' expires.

- DTMF duration in Talkaphone VOIP station needs to be configured as “800ms” to work with Avaya Aura® Messaging system that was used to test for DTMF RFC2833; the detail configuration is mentioned in **Section 6.4.**

2.3. Support

For technical support and information on Talkaphone VOIP-500 Series and VOIP-600 Series IP Call Stations, contact Talkaphone support at:

Address	: 7530 North Natchez Ave. Niles, IL 60714
Telephone	: (773) 539-1100
Fax	: (773) 539-1241
Email	: info@talkaphone.com
Web	: www.talkaphone.com

3. Reference Configuration

Figure 1 illustrates a sample configuration with an Avaya SIP-based network that includes the following products:

- Avaya Communication Server 1000 Call Server running on CPPM card.
- Avaya Communication Server 1000 SIP Line server running on COST2 server.
- Avaya Communication Server 1000 connected to simulated PSTN via SIP trunk and PRI trunk.
- Avaya 2007 IP Unistim, 1140E SIP and 3904 digital Deskphones were used to place and receive call to/from Talkaphone VOIP station.
- Talkaphone VOIP-500 Series and VOIP-600 Series IP Call Stations registered with Avaya Communication Server 1000 SIP Line server.

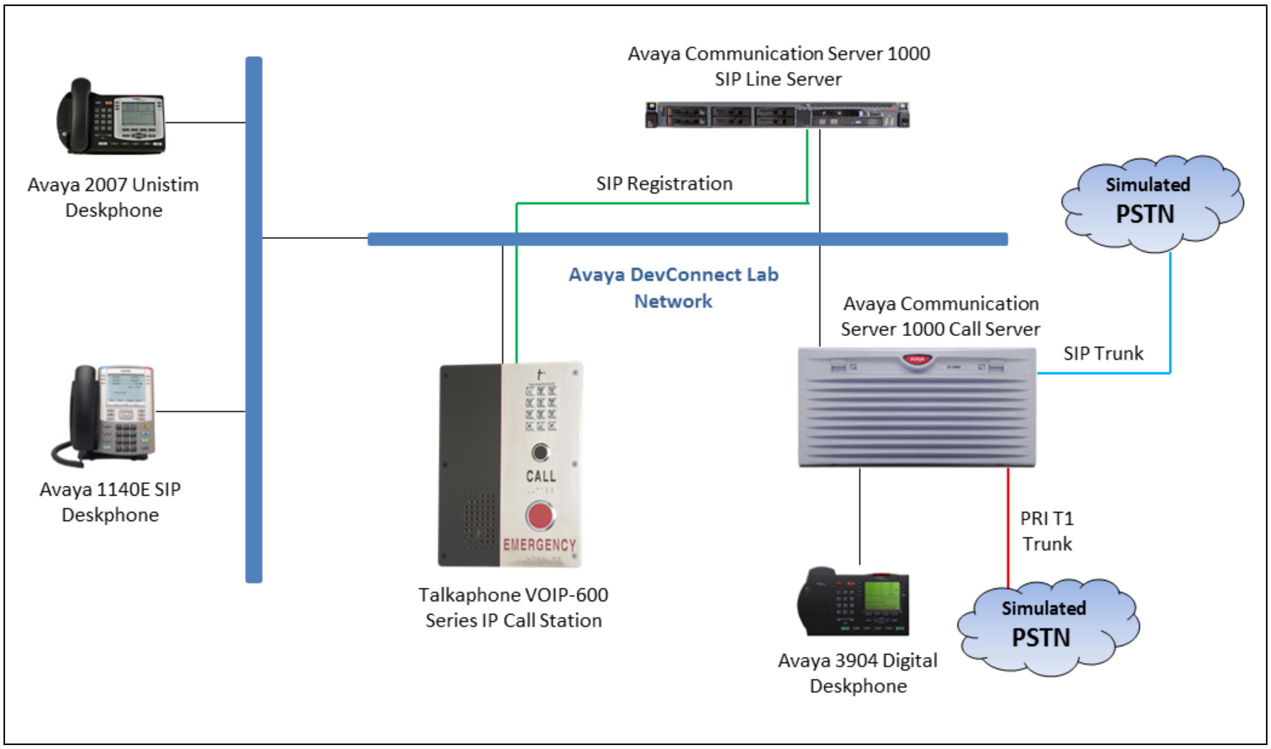


Figure 1: Avaya SIP Network with Talkaphone VOIP-500 Series and VOIP-600 Series IP Call Stations

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Communication Server 1000 Call server running on CPPM card	7.6 SP8 DepList 1: core Issue: 01 (created: 2016-06-20)
Avaya Communication Server 1000 SIP Line running on COST2 server	7.6 SP8
Avaya 2007 IP Unistim Deskphone	5.5
Avaya 1140E SIP Deskphone	04.04.28.00
Avaya 3409 Digital Deskphone	-
Talkaphone VOIP-500 Series IP Call Stations	Firmware Version : 1.0.2.7j Bootloader Version : 1.1.9
Talkaphone VOIP-600 Series IP Call Stations	Firmware Version : 1.0.2.7j Bootloader Version : 1.1.9

5. Configure Avaya Communication Server 1000

This section provides the procedures for configuring Avaya Communication Server 1000. The procedures include the following areas:

- Verify CS1000 Prerequisite.
- Configure SIP User.
- Configure Personal Call Assistant,
- Administer SIP Line Server

5.1. Verify CS1000 Prerequisite

This document assumes that the CS 1000 SIP Line server has been:

- Installed with CS 1000 Release 7.6 Linux Base.
- Joined CS 1000 Release 7.6 Security Domain.
- Deployed with SIP Line Application.

The following packages need to be enabled in the key code. If any of these features have not been enabled, please contact your Avaya account team or Avaya technical support at http://www.avaya.com.

Package Mnemonic	Package #	Descriptions	Package Type	Applicable market
SIP_LINES	417	SIP Line Service package	New package	Global
FFC	139	Flexible Feature Codes	Existing package	Global
SIPL_AVAYA	415	Avaya SIP Line package	Existing package	Global
SIPL_3RDPARTY	416	Third-Party SIP Line Package	Existing package	Global

5.2. Administer SIP User

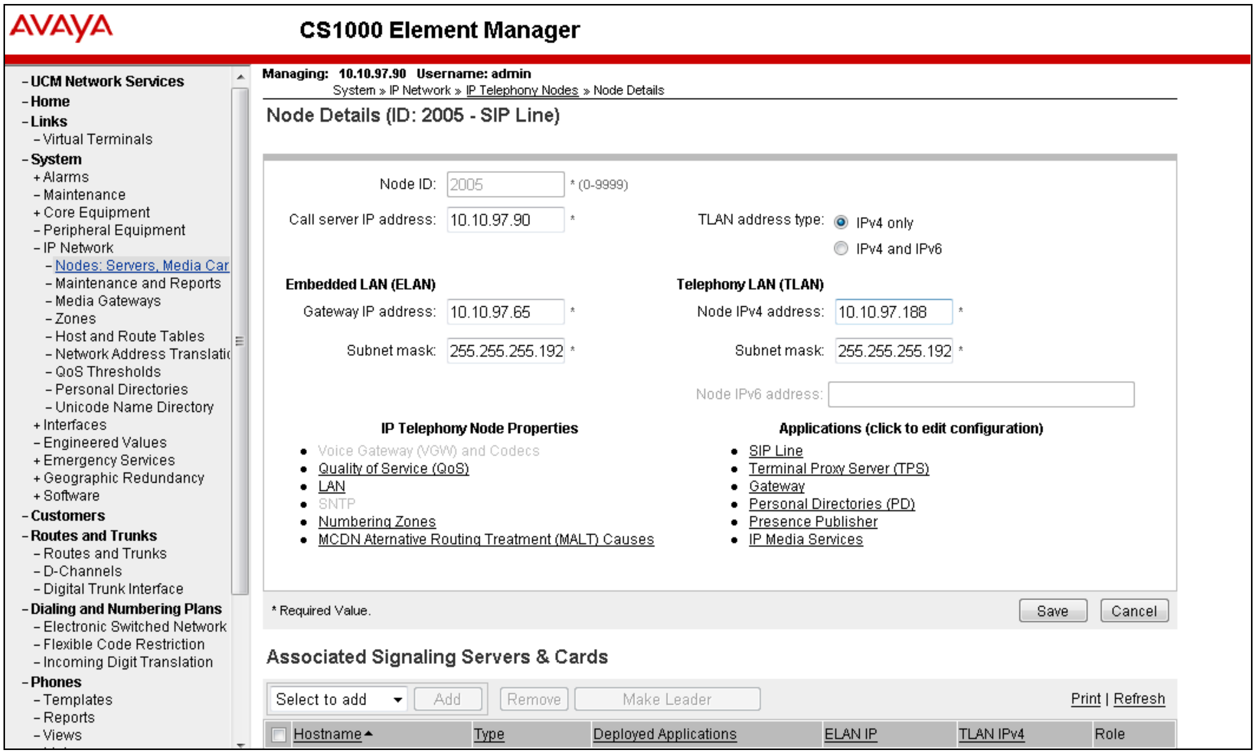
Access to the overlay command in the Avaya CS 1000 call server, use the overlay command LD 20/11 to create a new terminal number for a SIP user. The screen below shows the previously configured SIP user that used by Talkaphone VOIP IP Call station, in the detail of the terminal number configuration the "SIP3" set to **1** as this is 3rd party SIP endpoint, SIP user (SIPU) set to **4606**, Node ID (NDID) set to **2005** the node ID will be mentioned in the next section, station control password (SCPW) set to **1234** this is the password of SIP user, and the extension is configured for this SIP user in the Key 0 which is **4606**. The **Key 1 Hot U** with the prefix **111** in the front of extension is used for SIP user.

```
TN   108 0 01 09  VIRTUAL
TYPE UEXT
CDEN 8D
CTYP XDLC
CUST 1
UXTY SIPL
MCCL YES
SIPN 0
SIP3 1
FMCL 0
TLSV 0
SIPU 4694
NDID 2005
SUPR NO
UXID
NUID
NHTN
CFG_ZONE 00001
CUR_ZONE 00001
MRT
ERL  0
ECL  0
VSIT NO
FDN
TGAR 1
LDN  NO
NCOS 0
SGRP 0
RNPG 0
SCI  0
SSU
XLST
SCPW 1234
SFLT NO
CAC_MFC 0
CLS  CTD FBD WTA LPR MTD FND HTD TDD HFD CRPD
    MWD LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
```

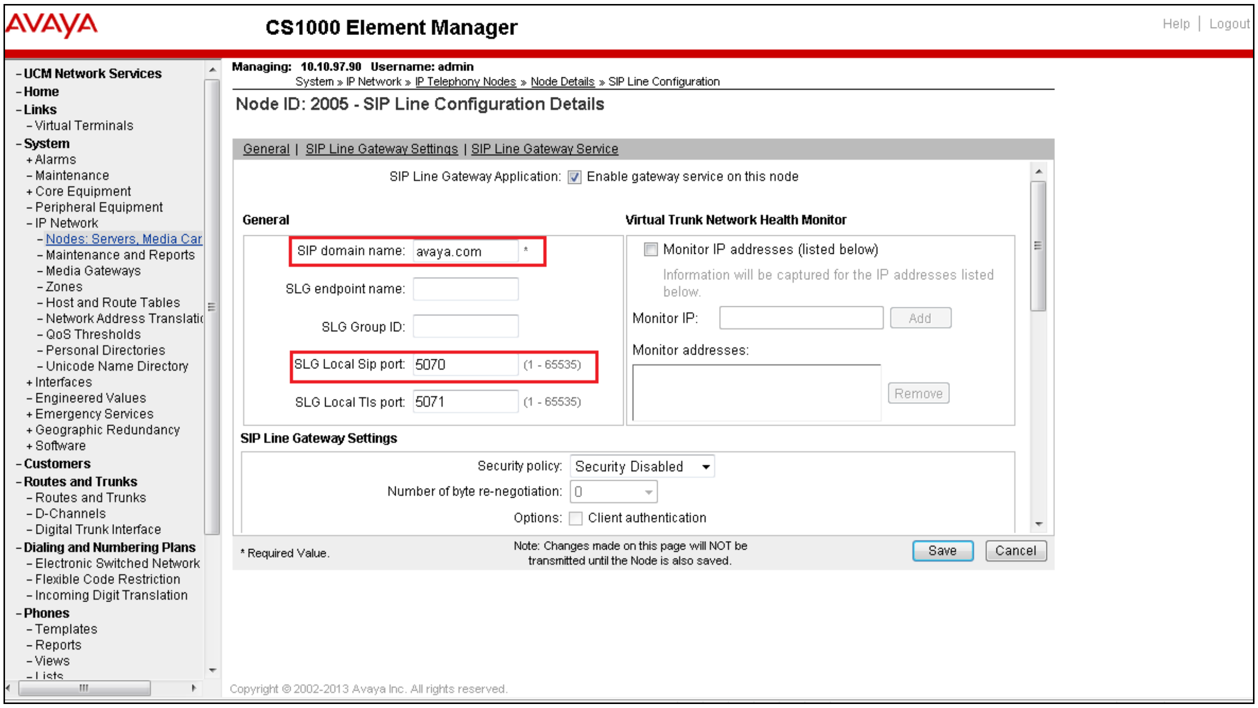
```
    POD SLKD CCSD SWD LND CNDA
    CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBD
    ICDD CDMD LLCN MCTD CLBD AUTU
    GPUD DPUD DNDA CFXD ARHD CLTD ASCD
    CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
    UDI RCC HBTD AHA IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
    DRDD EXR0
    USMD USRD ULAD CCBD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3
MCBN
    FDSD NOVD VOLA VOUD CDMR PRED RECD MCDD T87D SBMD ELMD
    MSNV FRA  PKCH MWTD DVLD CROD ELCD VMSA
CPND_LANG ENG
HUNT
PLEV 02
PUID
UPWD
DANI NO
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY  00 SCR 4606 0     MARP
       CPND
         CPND_LANG ROMAN
           NAME Talkaphone 600
           XPLN 14
           DISPLAY_FMT FIRST,LAST
    01 HOT U 1114606 MARP 0
    02
    03
    04
    05
    06
```

5.3. Administer Communication Server 1000 SIP Line Server

The CS 1000 SIP Line server can be accessed and configured via Element Manager, the screen below shows the **Node Details** configuration of the SIP Line server that was used for the compliance test. The Node ID is **2005** this node ID is matched with the node ID configured in the SIP user **4606** in **Section 5.2** and the node IP address is **10.10.97.188**.



Click on the **SIP Line** link under the **Application** as shown in the screen above to edit or display the detail configuration of the SIP Line. The screen below shows the **SIP Line Configuration Details** page, ensure the “SIP Line Gateway Application” check box is checked to enable gateway service on this node. In the **General** section, enter a sip domain in the **SIP domain name** field in the testing the SIP domain name used as “avaya.com” and the **SLG local SIP port** set to **5070**.



6. Configure Talkaphone VOIP-500 Series and VOIP-600 Series IP Call Stations

This section covers the configuration of the Talkaphone VOIP-500 Series and VOIP-600 Series IP Call Stations. The following procedures are covered:

1. Launching the Web Administration Interface
2. Network Configuration
3. SIP Configuration
4. Configure Audio Settings
5. Configure Call Parameters
6. Configure Buttons

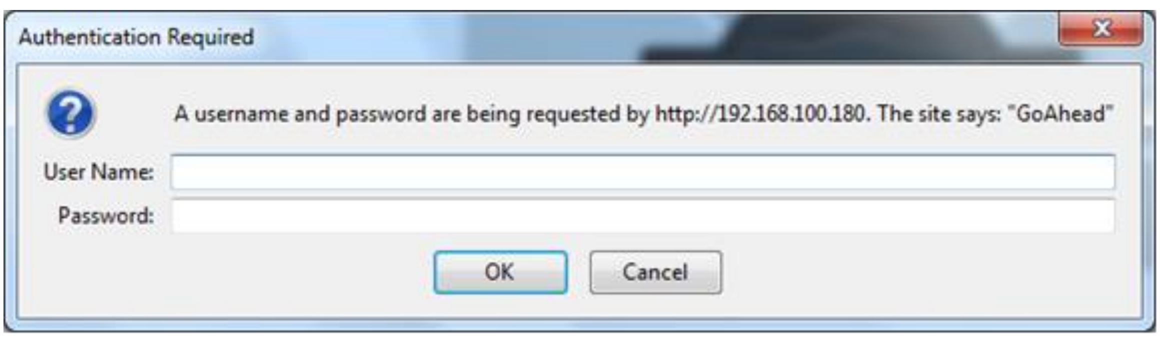
For more information on configuring other features of the Talkaphone IP Call Stations, refer to **[3].**

6.1. Launching the Web Administration Interface

The Talkaphone VOIP Call Stations are pre-configured with the following default values:

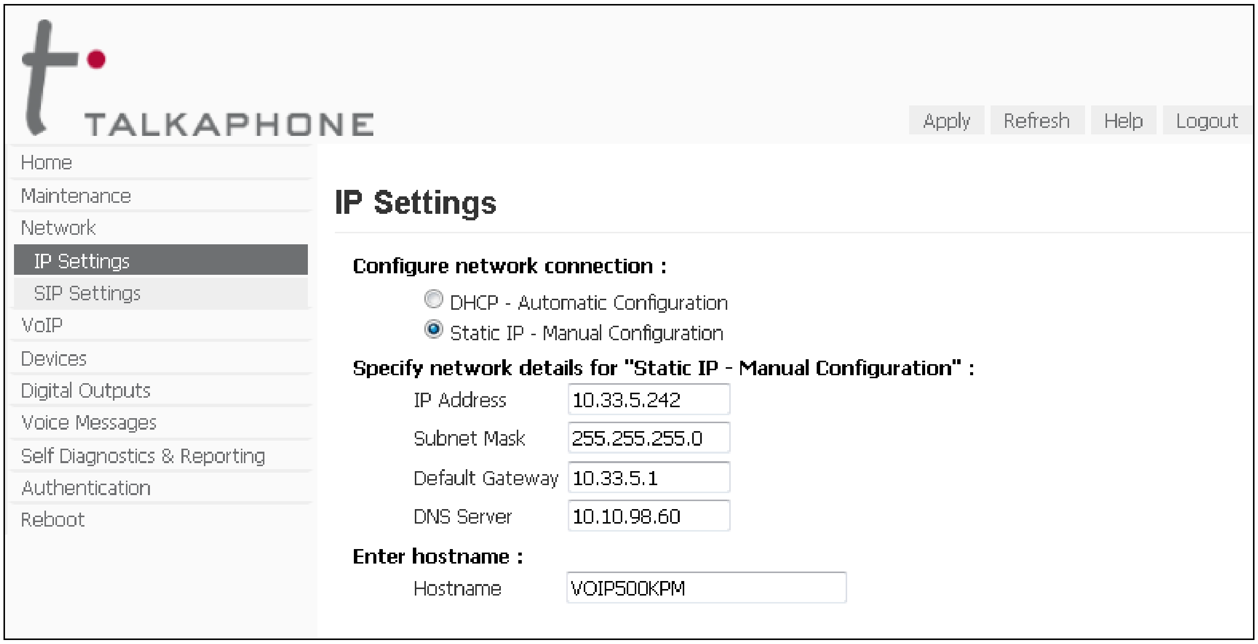
- **IP Address:** 192.168.1.10
- **Username:** admin
- **Password:** admin@123

Ensure that the administration PC and Talkaphone IP Call Station are connected to the LAN. Open a web browser and enter the IP address of the Talkaphone IP Call Station in the URL field. The browser prompts for authentication. Log in with the appropriate credentials.



6.2. Network Configuration

To modify the IP network configuration of the Talkaphone VOIP Call Station, navigate to the **Network → IP Settings** page. Configure the IP settings so that it conforms to the customer network requirements. Click **Apply** when done.



6.3. SIP Configuration

Navigate to **Network → SIP Settings** to configure the SIP setting of the Talkaphone IP Call Station. Configure the following parameters.

Under **Assign a phone number**:

- **Phone Number:** Specify the SIP number (e.g., *4606*) configured in **Section 5.2**

Under **Specify SIP Server FQDN/IP Address**:

- **Primary SIP Server FQDN/IP Address:** Specify the IP address of CS1000 SIP Line server (e.g., *10.10.97.188*) or the SIP domain (e.g. *avaya.com*). For the compliance test, the CS1000 SIP Line IP address was used.

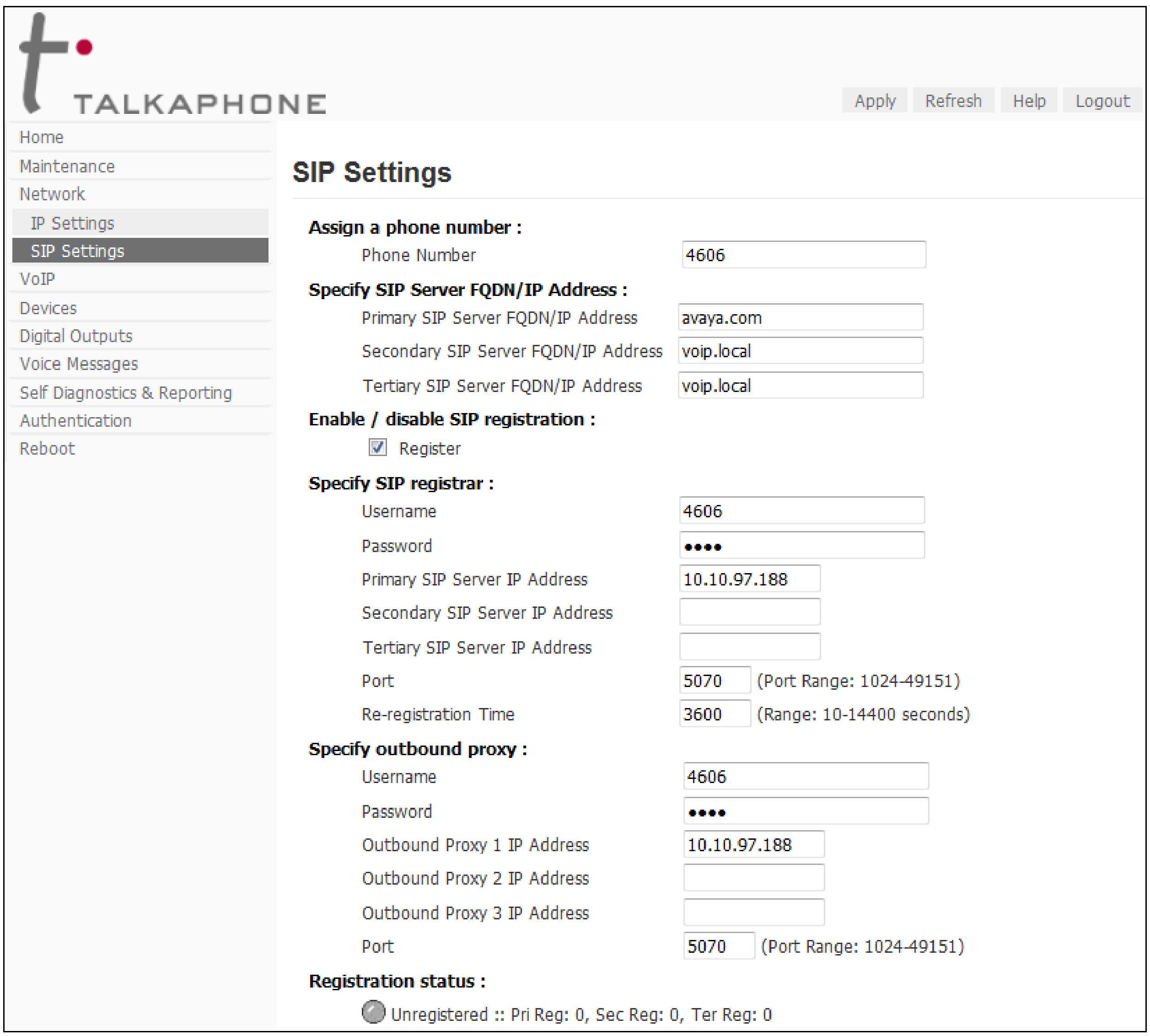
Under **Enable / disable SIP registration**:

- **Register:** Select the checkbox.

Under **Specify SIP registrar** and **Specify outbound proxy**:

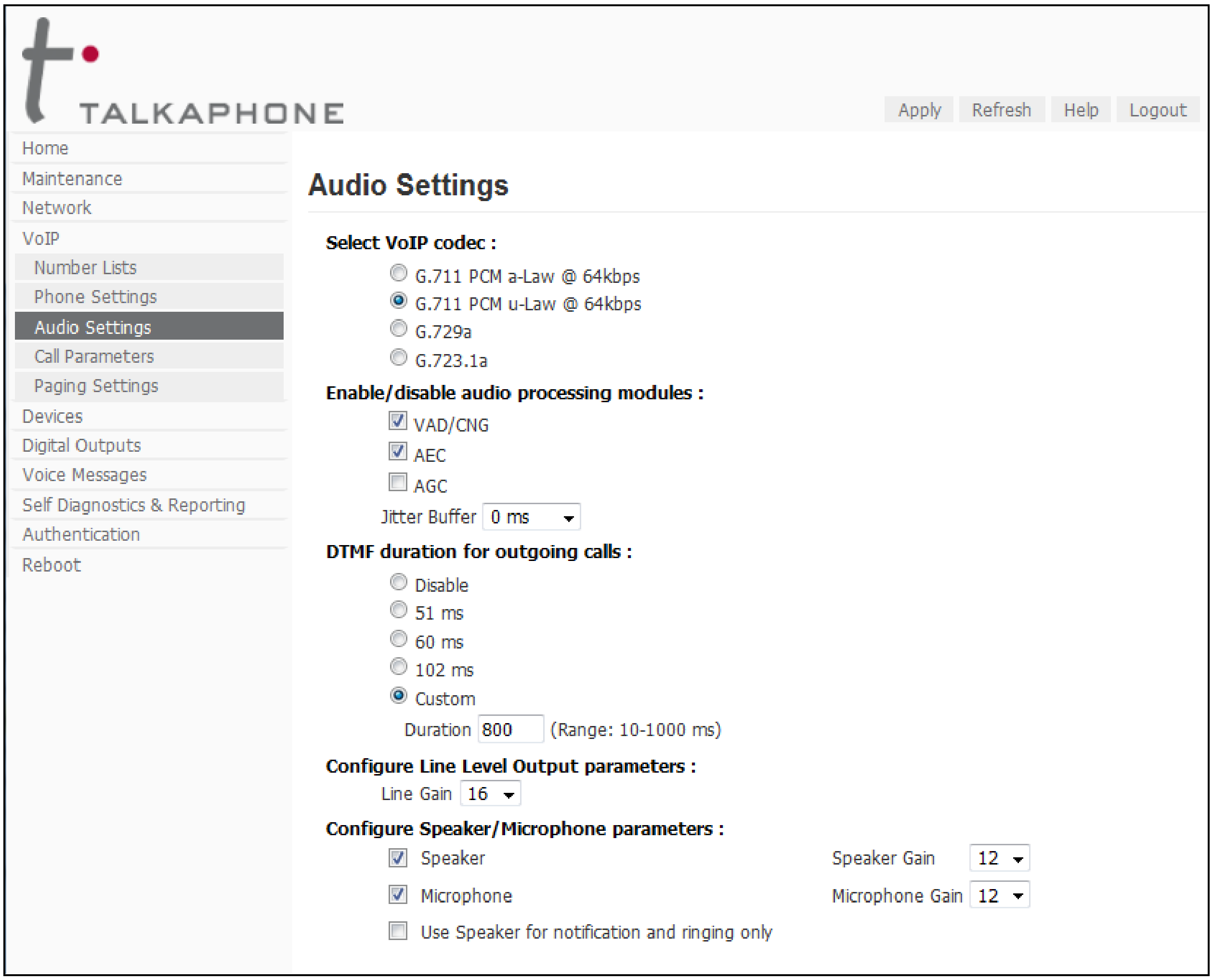
- **Username:** Specify the SIP number of the Talkaphone IP Call Station (e.g. *4606)*.
- **Password:** Specify the SIP password configured in **Section 5.5**.
- **Primary SIP Server IP Address:** Specify the LAN IP address of CS1000 SIP Line (e.g., *10.10.97.188)*
- **Port:** Specify the SIP port (e.g., *5070*) as configured in **Section 5.3**.
- **Replicate these settings in the 'Outbound Proxy' section on the same page.**

Accept the default values for the remaining fields and click **Apply** when done.



6.4. Configure Audio Settings

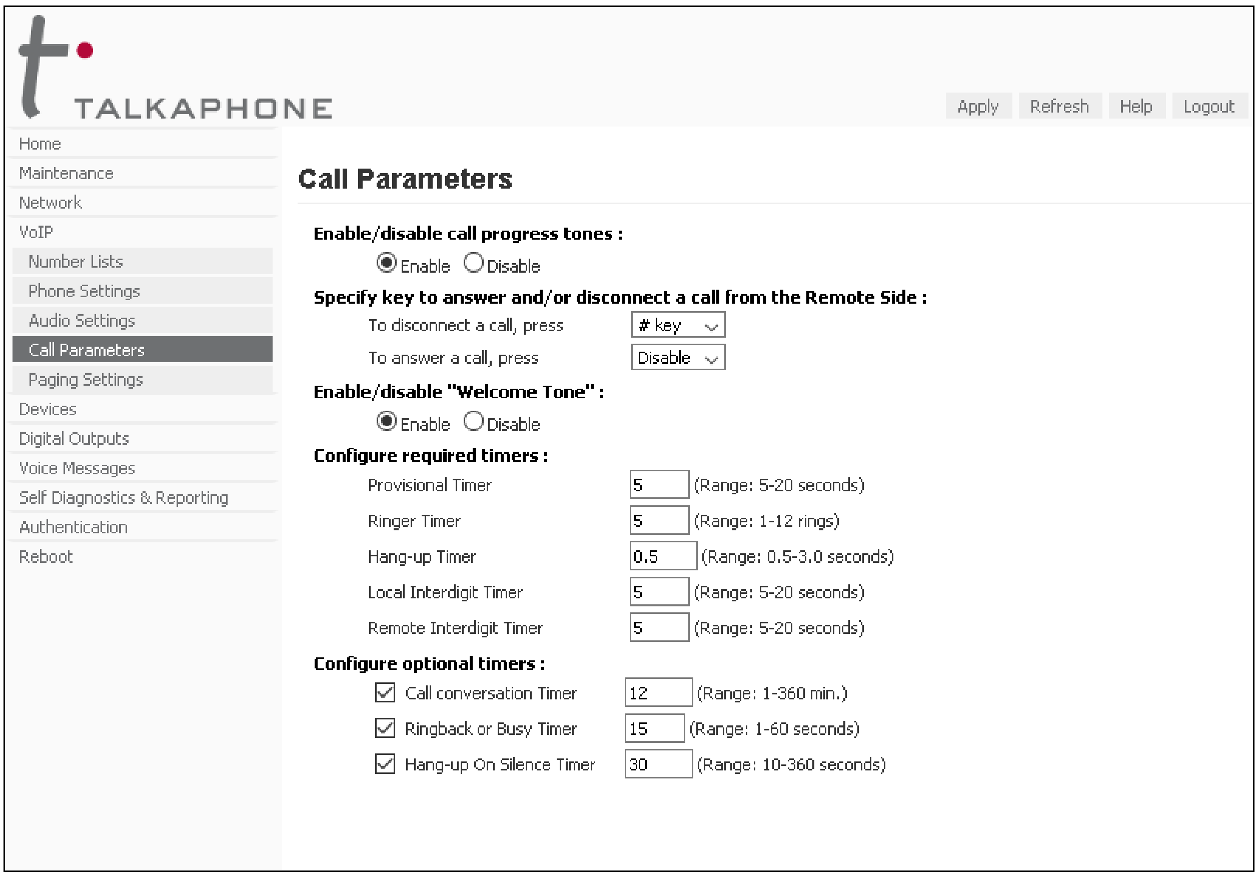
Navigate to **VoIP → Audio Settings** to configure the preferred codec, outbound DTMF duration, and microphone and speaker parameters. For the compliance test, the **DTMF duration for outgoing calls** was set to **Custom** with **Duration** of **800** ms. This is required so that a user can navigate through Avaya Aura® Messaging system using DTMF tones. In addition, the Speaker Gain can be adjusted to control the volume. All other fields were left at the default values. Click **Apply** when done.



6.5. Configure Call Parameters

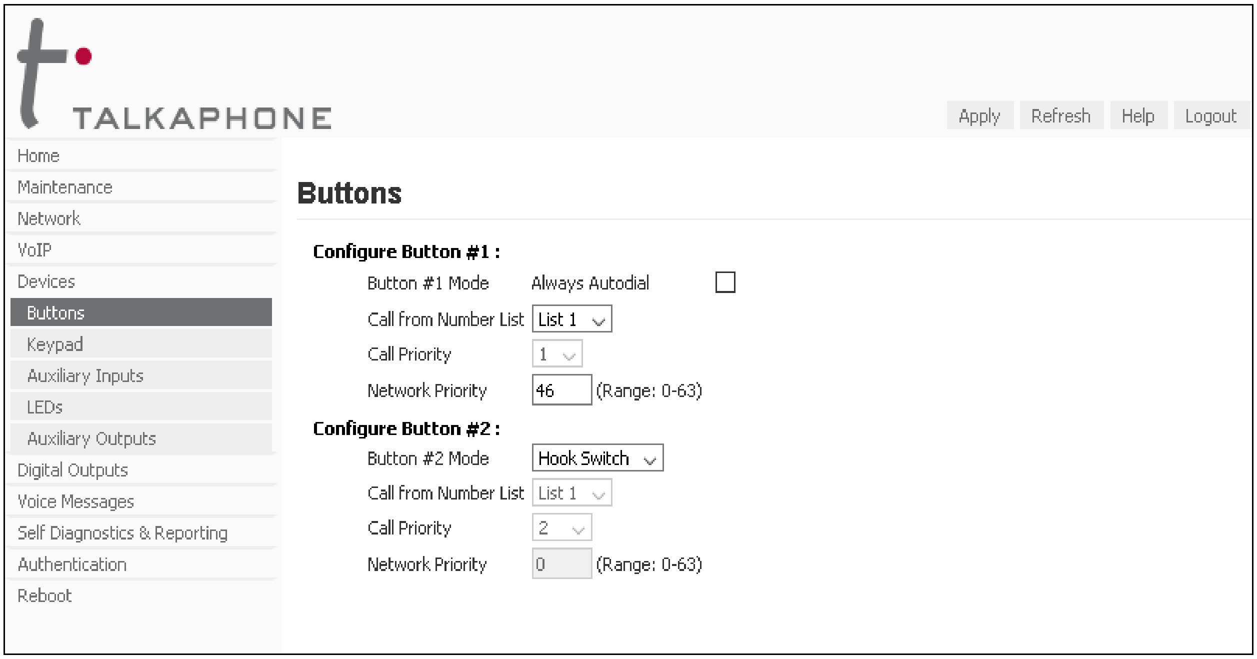
Navigate to **VoIP → Call Parameters** to view and customize any of the call parameters, such as **Local Interdigit Timer**, which dictates how long to wait before initiating a call after the user dials the digits, or the **Call conversation Timer**, which specifies how long an emergency call should remain active, unless the far-end drops the call. The following screen shows the default values for the call parameters.

Note: After a number is dialed on the Talkaphone IP Call Station, the **Local Interdigit Timer** must expire before the call is initiated. The minimum value for the **Local Interdigit Timer** is 5 secs.



6.6. Configure Buttons

Navigate to **Devices → Buttons** to verify the appropriate settings. For the compliance test, the **Buttons** were configured as shown below.



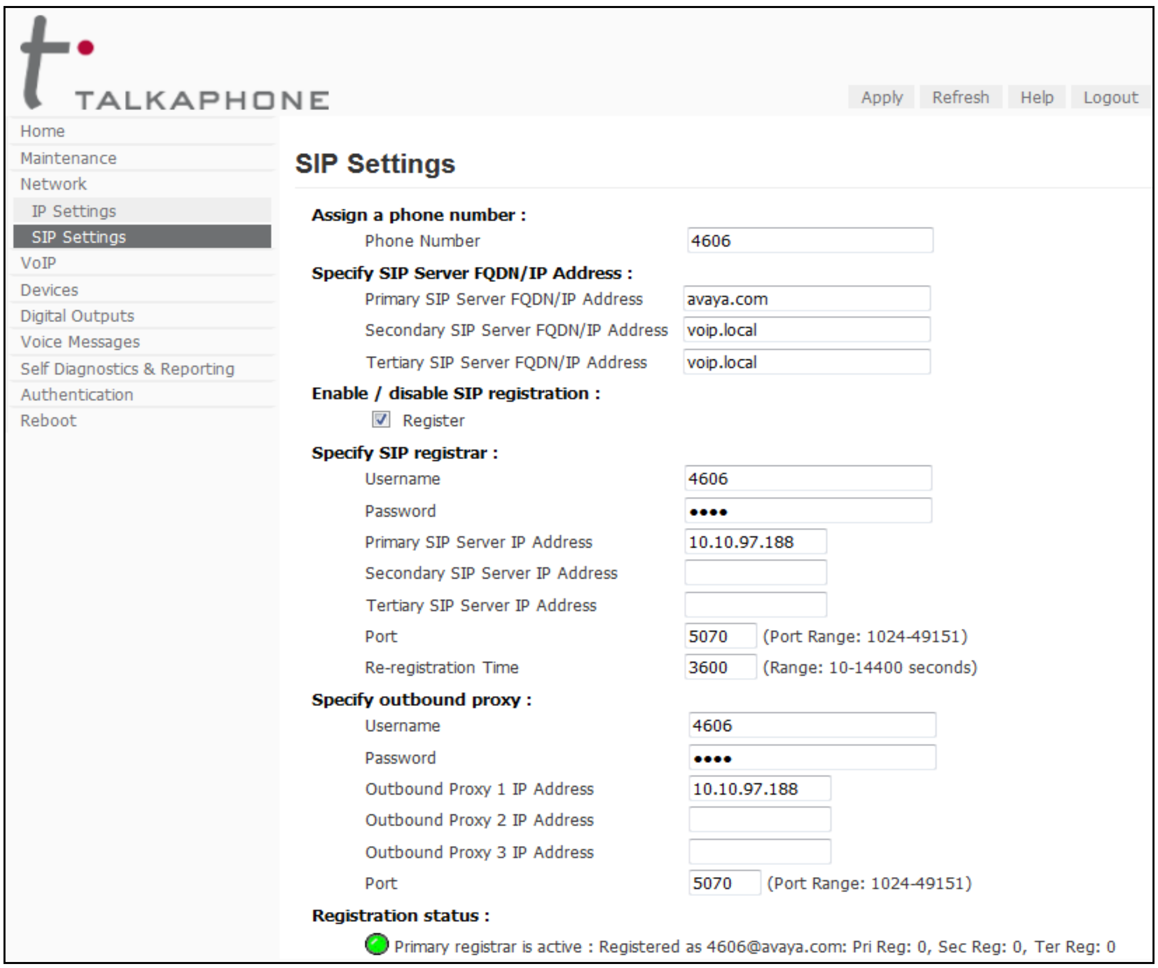
7. Verification Steps

This section provides the tests that can be performed to verify proper configuration of the Talkaphone VOIP-500 Series and VOIP-600 Series IP Call Stations with Avaya CS1000.

1. Verify that the Talkaphone IP Call Station has successfully registered with Avaya CS1000 SIP Line server by using Overlay (LD) 32 command in the call server to check status it should be shown as "REGISTERED".

```
>ld 32
NPR000
.stat 96 0 0 11
IDLE REGISTERED 00
```

Alternatively, the **SIP Settings** screen on the Talkaphone IP Call Station also shows the **Registration Status** with the green circle to indicate the registration status successfully.



2. Verify 2-way audio and proper call termination.

8. Conclusion

These Application Notes have described the administration steps required to integrate the Talkaphone VOIP-500 Series and VOIP-600 Series IP Call Stations with Avaya Communication Server 1000. Talkaphone IP Call Stations successfully registered with Avaya Communication Server 1000 and basic telephony features were verified. All test cases passed with observations noted in **Section 2.2.**

9. Additional References

This section references the Avaya and Talkaphone documentation relevant to these Application Notes. The following Avaya product documentation is available at support.avaya.com.

Avaya Communication Server 1000 Documents:

[1] Administering Avaya IP Office™ Platform with Manager, Release 10, Issue 10.33, October 2016.

[2] Deploying Avaya IP Office™ Platform Servers as Virtual Machines, Release 10, November 20156.

[3] IP Office™ Platform 9.1 Using IP Office System Monitor, Release 10, September 2016.

[4] Administering Avaya IP Office with Manager, Release 10, September 2016.

The following Talkaphone documentation may be found at www.talkaphone.com.

[5] *Talkaphone VOIP-500 Series Phone Configuration and Operation Manual v3.0.2*, Rev 7/31/2012.

[6] *Talkaphone VOIP-600 Series Configuration and Operation Manual v1.0.1*, Rev 9/17/2014.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.