



## **Application Notes for Extenway Bedside Terminal with Avaya Communication Server 1000E SIP Line Release 7.6 – Issue 1.0**

### **Abstract**

These Application Notes describe a solution comprised of Avaya Communication Server 1000E SIP Line Release 7.6 and Extenway Bedside Terminal. During the compliance testing, Extenway Bedside Terminal is a SIP-based device that registers to Avaya Communication Server 1000E as SIP endpoints for this testing. The Extenway Bedside Terminal was able to place and receive calls from the Communication Server 1000 Release 7.6E non-SIP and SIP Line telephones.

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 provide detailed information on how to create SIP Endpoints for Bedside Terminal used during the compliance testing. Only basic applicable telephony feature test cases of Avaya Communication Server 1000E (CS1000) release 7.6 SIP line were executed on the Bedside Terminal, where applicable, to ensure interoperability with the CS1000.

## 2 General Test Approach and Test Results

The general test approach was to have the Bedside Terminal register to the CS1000 SIP line gateway successfully. From the CS1000 telephone calls were placed to and from Bedside Terminal as well as from PSTN. Telephony features such as No Answer, Hold, DTMF, MWI and codec negotiation were tested.

### 2.1 Interoperability Compliance Testing

The focus of this testing was to verify that the Extenway Bedside Terminal was able to interoperate with the CS1000 SIP line system. The following areas were covered:

- Registration of the Bedside Terminal to the CS1000 SIP Line Gateway.
- Voice call establishment of Bedside Terminal with Avaya CS1000 SIP and Avaya UNSTim telephones, to ensure two way speech paths.
- Telephony features: basic calls inbound and outbound, call display.
- Bedside Terminal does not support the following features internally; however these features were tested with Bedside Terminal as a participant: Voicemail (tested with Bedside Terminal leaving messages only), blind and consultative transfer (between Avaya phone), DTMF (dual tone multi frequency), call forward no answer (on Avaya phone).
- Codec negotiation – G.711 and G.722.

### 2.2 Test Results

The objectives outlined in **Section 2.1** were verified. The following observations were made during the compliance testing:

- Following telephony feature were not tested as Bedside terminal does not support: Message Waiting Indicator (MWI), Conference, Do not Disturb (DND) and other features such as Call Forward No Answer, Call Forward Busy, Call Forward All Calls, Call Park and Retrieve, Conference, Call Waiting, Calling Party Name Display.
- SIP Line features like Speed Dial, Group Call Pickup, Make Set Busy, Ring Again Busy and Ring Again No Answer were not tested since it is not relevant to Bedside Terminal.
- CallerID on Bedside Terminal will not update after a call is transferred between Avaya phones. No end user impact.

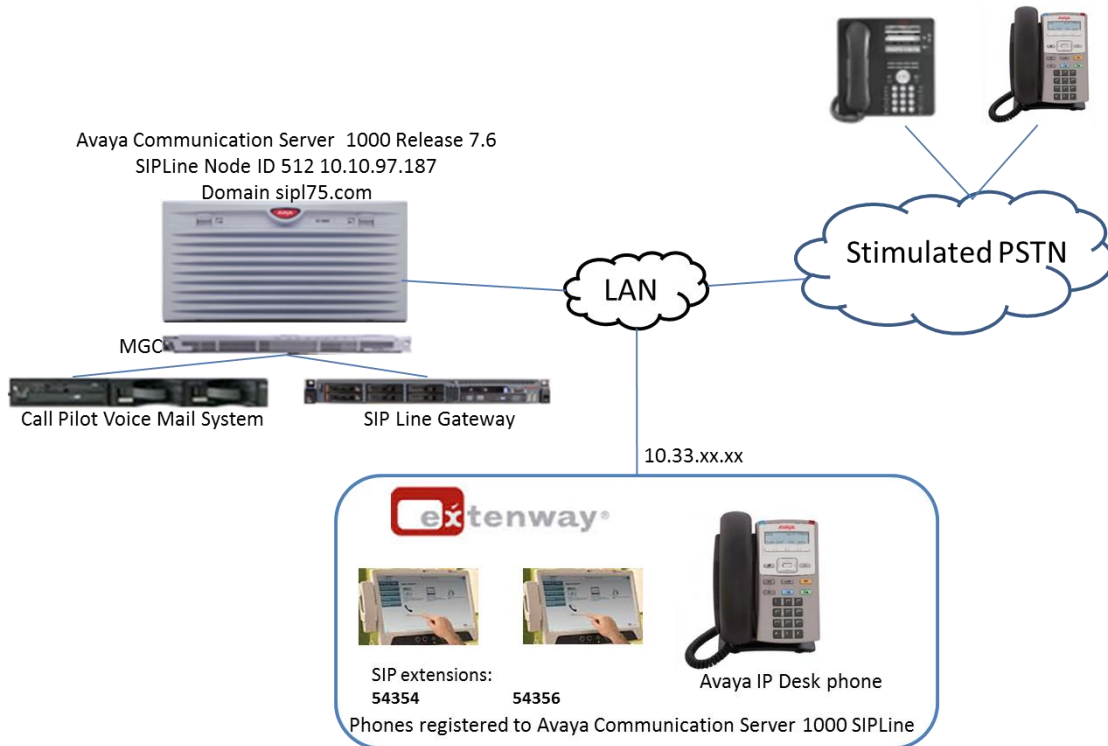
## 2.3 Support

For technical support on Bedside Terminal, please contact Extenway technical support team:

- **Telephone:** 1-819-485-1191
- **Email:**
- **Website:**

## 3 Reference Configuration

**Figure 1** illustrates the test configuration used during the compliance testing between the Avaya CS1000 and the Bedside Terminal.



**Figure 1: Network Configuration Diagram**

## 4 Equipment and Software Validated

The following equipment and software was used during the lab testing:

Equipment	Software Version
Avaya Communication Server 1000E Release 7.6	Call Server (CPPM): 7.65P+ Signaling Server (CPPM): 7.65.16
Avaya CallPilot® Messaging System	5.0.1
Avaya 1100 IP Deskphones (UNSTim phones)	0623C8Q
Avaya 1140 IP Deskphones (SIP)	04.03.09
esi-interface (Bedside Terminal Application)	V3.X
esi-phoneserver (Server-side Application)	V1.X

## 5 Configure SIP extension for Bedside Terminal on Avaya Communication Server 1000E

This section describes the steps to configure the Avaya CS1000 SIP Line phone using the CS1000 command line interface (CLI). For detailed information on how to configure and administer the CS1000 SIP Line, please refer to **Section 9**.

This document assumes that CS100 SIP Line configuration had been setup and operational. The following is a summary of tasks that need to be done to configure the CS1000 SIP User for Bedside Terminal: Create SIP Line phones.

### 5.1 Prerequisite

This document assumes that the CS1000 SIP Line server has:

- Been installed with CS1000 Release 7.6 Linux Base.
- Joined CS1000 Release 7.6 Security Domain.
- Been deployed with SIP Line Application.
- SIP Line Node has been setup and in operational state. The IP address of SIP Line used in this test is 10.10.97.187 and its domain is sip175.com, as displayed in **Figure 1**.

## 5.2 Create a SIP Line Phone for Bedside Terminal

To create a SIP Line phone on the Call Server, log in as administrator using the command line interface (CLI) and issue the overlay (LD) 11/20 as shown below.

The bold fields must be properly inputted as they are configured on the Call server, for other fields hit enter to leave it at default values:

```
REQ: prt
TYPE: UEXT - Universal extension type for SIP Line phone

TN 104 0 0 14
DES Extway
TYPE UEXT
CDEN 8D
CTYP XDLC
CUST 0
UXTY SIPL - Universal extension type is SIP Line
MCCL YES
SIPN 0 - For SIP phone third party, enter 0 in this field
SIP3 1 - For SIP phone third party, enter 1 in this field
FMCL 0
TLSV 0
SIPU 54356 - SIP phone user name
NDID 512 - Node ID of SIP Line
SUPR NO
UXID
NUID
NHTN
CFG_ZONE 00001
CUR_ZONE 00001
SCPW 1234 - Password of SIP Line user. Need to register Extenway to SIP Line Node
VSIT NO
KEY 00 SCR 54356 0 MARP
    CPND
        CPND_LANG ROMAN
        NAME Extw 1
        XPLN 13
        DISPLAY_FMT FIRST, LAST
    01 HOT U 2654356 MARP 0 - Key 1 hot U with prefix + DN
```

Repeat same step for Extensions: **54354, 54504**.

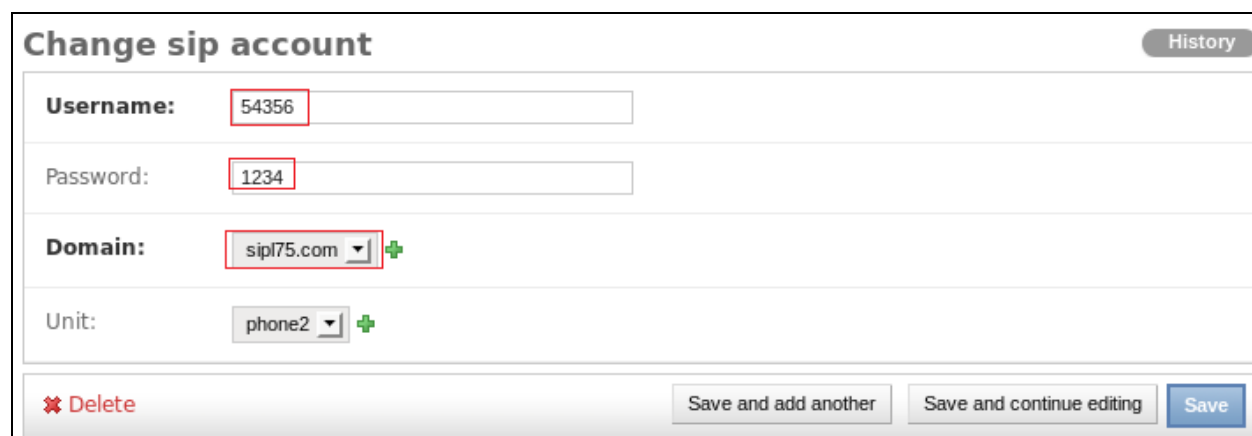
## 6 Configure Extenway Bedside Terminal

The configuration and setup had been completed by Extenway technician. This section only shows the screenshot captured that displays the configuration used during compliance test.

### 6.1 Administer Bedside Terminal SIP Account

Select **Setting** (not shown) from the top menu, to display the screen below. Configure the **SIP Account**, enter the following values for the specified fields, and retain the default values in the remaining fields.

- **Username:** The **SIPU** value created in **Section 5.2**.
- **Password:** The **SCPW** value created in **Section 5.2**.
- **Domain:** The SIP Line Node domain, during compliance test, domain of SIP Line Node 512 is **sip175.com**.



**Change sip account** History

**Username:** 54356

**Password:** 1234

**Domain:** sip175.com +

**Unit:** phone2 +

✘ Delete Save and add another Save and continue editing Save

## 7 Verification Steps

### 7.1 Verify SIP Line phones are registered

To verify the status of a SIP Line phone on the Call Server, log in as administrator using the command line interface (CLI) and issue the overlay (LD) 32, verify the status of SIP Phone is registered as shown below:

```
>ld 32
.stat 104 0 0 13
IDLE REGISTERED 00
.stat 104 0 0 14
IDLE REGISTERED 00
```

To verify the status of SIP Line phone on SIP Line Signaling server, log in to the SIPLine Signaling server as administrator and issue slgSetShowAll command:

```
[admin@sip175 ~]$ slgSetShowAll

=== VTRK ===
UserID          AuthId          TN              Clients  Calls  SetHandle  Pos ID
-----
-----
----- IPv4 Endpoints -----
54008           54008           104-00-00-01    1        0    0x9083560
54354           54354           104-00-00-14    1        0    0x91f9ad8
54504           54504           104-00-01-04    1        1    0x9210428
54356           54356           104-00-00-20    1        0    0x9202658

Total User Registered = 4   V4 Registered = 4   V6 Registered = 0
```

To verify the detail information of SIP Line phone on SIP Line Signaling server, log in SIPLine Signaling server as administrator and issue slgSetShowByUID command:

```
[admin@sip175 ~]$ slgSetShowByUID 54504

=== VTRK ===
UserID          AuthId          TN              Clients  Calls  SetHandle  Pos ID  SIPL Type
-----
-----
-----
54504           54504           104-00-01-04    1        1    0x9210428          SIP Lines
-----
-----
StatusFlags = Registered Controlled KeyMapDwld SSD
FeatureMask =
CallProcStatus = 8

Current Client = 0, Total Clients = 1

== Client 0 ==
IPv4:Port:Trans = [REDACTED].163.98:65476:udp
Type = Unknown
UserAgent = Sip Express Media Server (1.4.3-33-g94d410d (x86_64/linux))
x-nt-guid = a178b686495f355097c508a2fe7869aa
RegDescrip =
RegStatus = 1
PbxReason = OK
SipCode = 200
hTransc = (nil)
Expire = 1000
Nonce = 7d17705170fb5ea0c55518de9ae9931e
NonceCount = 2
hTimer = 0x8fe0358
TimeRemain = 727
Stale = 0
Outbound = 0
ClientGUID = 0
MSec CLS = MSNV (MSEC-Never)
Contact = sip:54504@[REDACTED].163.98:65476
KeyNum = 255
AutoAnswer = NO
```

## 7.2 Verify Bedside Terminal is registered

Log on Extenway software. Verify all the extensions are successfully registered to CS1K.

Username	Domain	Registration state	Unit
<input type="checkbox"/> 54354	sipl75.com	ACTIVE 0:08:34	(None)
<input type="checkbox"/> 54356	sipl75.com	ACTIVE 0:08:33	phone2
<input type="checkbox"/> 54504	sipl75.com	ACTIVE 0:08:33	phone1

3 sip accounts

## 7.3 Make a call from Bedside Terminal to Avaya phone

Touch the screen to initiate call to Avaya phone, Avaya phone is ringing then pick up the phone to establish the call. Verify that the call is established with good quality of voice. Below is a screenshot of Bedside Terminal in a call.

Bed Phone1 Phone 1:42 PM

Service Information

On call

Current call

You are talking to  
**Unknown Caller**  
**54504**

Call Duration  
**00:00:08**

Hang Up

extenway

Need help using your terminal?  
Call customer support at 819-485-1191



## 8 Conclusion

All of the executed test cases have passed and met the objectives outlined in **Section 2.1**, with some exceptions outlined in **Section 2.2**. The Bedside Terminal firmware V3.X is considered to be in compliance with Avaya Communication Server 1000E SIP Line System Release 7.6.

## 9 Additional References

Product documentation for Avaya Communication Server 1000E products may be found at:  
<https://support.avaya.com/css/Products/>

- Avaya Communication Server 1000 Installation and Commissioning Release 7.6; NN43041-310, March 2013.
- Avaya Communication Server 1000 SIP Line Fundamental, Release 7.6; NN43001-508, March 2013
- Avaya Communication Server 1000 Element Manager System Reference – Administration; NN43001-632, November 2013.
- Avaya Communication Server 1000 Co-resident Call Server and Signaling Server Fundamentals, Release 7.6; NN43001-509, March 2013.
- Avaya Communication Server 1000 Unified Communications Management Common Services Fundamentals, Release 7.6; NN43001-116, March 2013.
- Avaya Communication Server 1000 ISDN Primary Rate Interface Installation and Commissioning, Release 7.6; NN43001-301, March 2013.

Product documentation for the Bedside Terminal products may be found at:

- <http://extenway.com/en/patients/technology/>

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