Avaya Solution & Interoperability Test Lab

Application Notes for Polycom SoundStation IP 6000 with Avaya Integral Enterprise - Issue 1.0

Abstract

These Application Notes document compliance testing the Polycom IP 6000 with Avaya IP and digital telephones controlled by Avaya Integral Enterprise. The Polycom IP 6000 communicates with Avaya Integral Enterprise via LAN and the VoIP Board or the new IP Media Resource Module (IPMR) using SIP (Session Initiation Protocol) and enables meeting or conference participants to simultaneously participate in a telephone conversation.

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 procedures for configuring Polycom SoundStation IP 6000 which was compliance tested with Avaya Integral Enterprise. The SoundStation IP 6000 is a SIP based IP conference phone that delivers superior performance for small to midsize conference rooms.

These Application Notes assume that the Avaya Integral Enterprise is already installed and basic configuration steps have been performed. Only steps relevant to this compliance test will be described in this document. For further details on configuration steps not covered in this document, consult [1].

1.1. Interoperability Compliance Testing

The interoperability compliance test included features and serviceability. The focus of the interoperability compliance testing was primarily on verifying call establishment on the Polycom SoundStation IP 6000. Polycom SoundStation IP 6000 operations such as inbound calls, outbound calls, hold, transfer, forward, conference, and Polycom SoundStation IP 6000 interactions with Avaya IP and digital telephones were verified. The serviceability testing introduced failure scenarios to see if Polycom SoundStation IP 6000 can recover from failures.

1.2. Support

Technical documentation and software downloads for the SoundStation IP 6000 can be found at: http://www.polycom.com/support/voice/soundstation_ip_series/soundstation_ip6000.html.

2. Reference Configuration

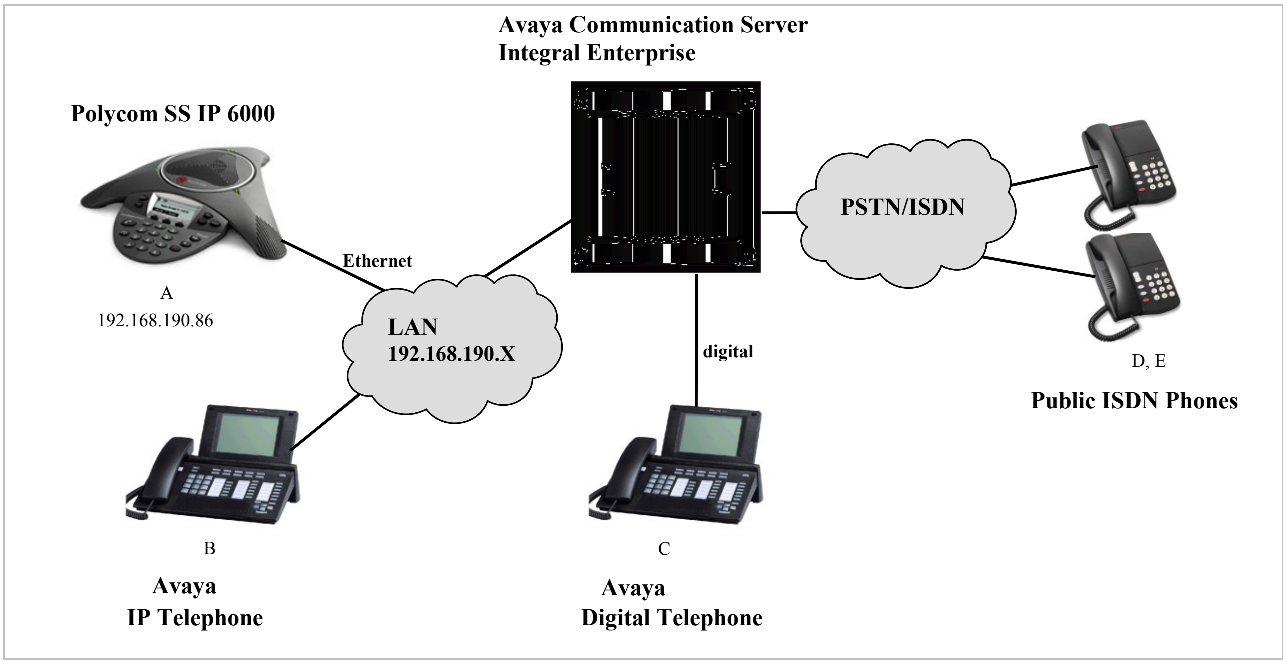


Figure 1: Test Configuration

The configuration that was used for testing consists of an Avaya Integral Enterprise including a VoIP board or an IPMR (IP Media Resource) module. Note that the Polycom IP 6000 is able to operate solely from the power that it received from an Avaya C364T-PWR Ethernet switch (in the LAN) to which it was attached. Therefore the unit is shipped without an optional power supply.

The following table contains additional information about each of the telephone endpoints depicted in the Test Configuration diagram:

Diagram	Ext	Endpoint
A	1319	Polycom IP 6000
B	1312	T3 IP
C	1500	T3.14
D	0 75009497	ISDN telephone
E	0 75056632	ISDN telephone

Table 1: Extensions Used for Testing

3. Equipment and Software Validated

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

Equipment	Software Version
Avaya S5500 (ACB) Integral Enterprise	IEEE5 (L050V00.2.0.1)
Avaya IPMR Module	IPMRSW15 (Sep 21, 2009)
Avaya VoIP Board	VOIPSW79 (Mar 05, 2009)
Avaya T3 IP Comfort Telephone	T542_0DE.a3i
Avaya T3.14 Comfort ISDN Telephone	T314_0DE.hxt
Avaya C364T-PWR	4.5.14
Polycom IP 6000	3.1.2.0392

Table 2: Version Numbers of Equipment and Software

4. Configure the Avaya Integral Enterprise

The configuration and verification operations illustrated in this section were performed using the Avaya Integral System Management (ISM) tool on a service PC. Access to the system was via the LAN.

The configuration of the interface to the PSTN/ISDN and the interfaces to the Avaya telephones are outside the scope of this document.

4.1. Configure Dial Plan

Launch ISM by selecting **Start -> Programs -> Integral 33 -> ISM** and enter default username n1 and password p1. To open a connection to the Avaya Integral Enterprise click on Customer and enter the necessary parameters to connect:

Customer name,
PABX: Integral Enterprise,
Software version: IEE5,
User name: EXPERT,
Password: ACCESS,
MML password: stepputtis,
IP Address of the Ethernet interface of the system,
Select Ethernet under TUX.

Use the Transparent Console (TCO) of the ISM and the task WABE to configure the dial plan. Use the command **akze:13,intern,2,v;** followed by **zids:<,2;** to add a new entry, e.g. for a 4-digit extension number beginning with 13, covering the extension (1319 and 1317) used for the Polycom IP 6000. The commands **anzg;** (show) and **dwgr:2,v;** (display dial group 2) are used to display the current assignments for dial group 2 as shown in **Figure 2**.

```
PROL<1:wabe;
Kommando in Bearbeitung !
WABE<akze:13,intern,2,v;
zids:<,2;
WABE<anzg;
WABE<dwgr:2,v;
                                                           03.08.09  16:55:25
Anzeigen der Wahlbewertungsdaten zu einer Wahlgruppe
====================================================
 Wahlgruppe   :  2
 Wahlverfahren:  Vorwahl

AKZ        Wahl   Bndl  AKZ  SA    Co. LCR   Vorwahlzu.      ext. LCR- RI- Num.
           selek. num.  Info Grup. Nr. Daten Ziff.      Sel- Belg Rout SA  Plan
                                       satz  folge      ekt. art  Flg  Flg
-------------------------------------------------------------------------------

0          EXTERN 4     -    -     -   -                0    INIT ROFF -   -
                                                           Amt ueber Erdtaste
13         INTERN -     2    -     -   -                0     -    -    -   -
20         INTERN -     1    -     -   -                0     -    -    -   -
21         INTERN -     1    -     -   -                0     -    -    -   -
22         INTERN -     2    -     -   -                0     -    -    -   -
23         INTERN -     1    -     -   -                0     -    -    -   -
24         INTERN -     1    -     -   -                0     -    -    -   -
25         INTERN -     1    -     -   -                0     -    -    -   -
26         INTERN -     1    -     -   -                0     -    -    -   -
27         INTERN -     1    -     -   -                0     -    -    -   -
28         INTERN -     1    -     -   -                0     -    -    -   -
29         INTERN -     1    -     -   -                0     -    -    -   -
40         INTERN -     2    -     -   -                0     -    -    -   -
41         INTERN -     1    -     -   -                0     -    -    -   -
...
```

Figure 2: Dial Plan Display Form

4.2. Configure Interface to Polycom IP 6000

Use the TCO of the ISM and the task AOGD to configure the interface. Use the command **`aoei:1319,...;`** (create extension 1319) followed by the necessary parameters for a digital telephone, which is the basis for IP and SIP phones (**Table 3**). Use the command **`dnei;`** to create the service telephony (TLP) and in addition the command **`cdec:`** to set the appropriate codec type, echo cancellation mode and codec mode for IP and SIP phones. **Figure 3** shows an example command sequence for that configuration. The command **`anzg:`** followed by the appropriate extension number (e.g. **`anzg:1319;`**) is used to display the current settings as shown in **Figure 4**. Once the account is registered the device IP address and active coder appear in the General Data list.

Parameter	Usage
Extension number	Enter the extension number to be assigned to the subscriber, e.g. 1319 or 1317.
Hardware address (HWA)	Enter the designation for the port to which the unit is (virtually) attached, e.g. 01-01-03-xx.
Type (AO-Type)	Enter the station type for a digital telephone, i.e. DITN.
Name	Enter the name of the user which is to be associated with the telephone.
Protocol	Enter the protocol and version to be used by a digital telephone, i.e. ETSI, 0.
Codec	Enter the codec type, echo cancellation mode, and codec mode to be used.

Table 3: Configuration - Polycom IP 6000 Subscriber

```
PROL<praw:aogd;
AOGD<aoei:1319,01-01-03-29;
AOGD<aoty:DITN;
AOGD<alae;
AOGD<nako:...;
AOGD<prve:ETSI,0;
AOGD<agrp:1,,,,;
AOGD<uela:1,;
AOGD<exit;
AOGD<aoae;
AOGD<dnei:TLP;
AOGD<grda:2,1,0,0,1;
AOGD<cdec:1,on,n;
AOGD<dnzu:f;
AOGD<exit;
AOGD<aozu:f;
AOGD<exit;
```

Figure 3: Command Sequence for the Configuration of Polycom IP 6000 Subscriber

```
PROL<1:aogd;
Kommando in Bearbeitung !

AOGD<anzg:1319;
==============================================================================
                                                            03.08.09  17:18:51
Anschlussorgan
--------------
Rufnummer           : 1319
Steckplatz/HWA      : 01-01-03-29
AO-Typ              : DITN
==============================================================================

Allgemeine ADS-Daten
--------------------
Name                       :
Kostenstelle               : 00000
Protokolle                 :
                              Protokoll | Version | faulty | busy 2| error
                              ----------+---------+--------+-------+------
                               ETSI     | 0       | AUS    | AUS   | AUS
Ueberlastprioritaet        : 0
SPWKGR. Amtszugriff        : 0
SPWKGR. COLISEE            : 0
DISA-Gruppe                : 0
Haendlergruppe             : 0
Rufnr.zuord. HKZ u.QUE     :
Kategorie                  : -1
Wartefeld Maximum          : 0
Reservierte
Verbindungsspeicher        : 0
Dienstspeicher             : 2
AO-Zustand                 : IN BETRIEB
Service-Sperre             : sv-frei
Rufnummern-Sperre          : Aus
IP - Adresse               :
(V4)192.168.190.88:5060
Akt. Coder                 : g711alaw64k
Sichere Registrierung      : NEIN
==============================================================================

Dienstdaten
-----------

                |     TLP      |     GEN      |              |              |
----------------+--------------+--------------+--------------+--------------+
Zustand         | FREI         | FREI         |              |              |
Wahlgruppe      | 2            | 2            |              |              |
Verkehrsgruppe  | 1            | 1            |              |              |
Umschaltegruppe | 0            | 0            |              |              |
Codewahlgruppe  | 0            | 0            |              |              |
LCR-Gruppe      | 0            | 0            |              |              |
Wahlabruf       | DEAKTIV      | DEAKTIV      |              |              |
Rueckausloesen  | DEAKTIV      | DEAKTIV      |              |              |
Coder           | g711alaw64k  | g711alaw64k  | init         | init         |
Codermode       | Normal       | Normal       |              |              |
Echounterdrueck.|  Ein         |  Ein         |              |              |
==============================================================================

B-Kanal-Daten
-------------
Vergabekennung      :  -
Verhandlungskennung :  -

B-Kan- Buendel- Richtg  Zugr Zustd  |  B-Kan- Buendel- Richtg  Zugr Zustd
Nr.    nummer                       |  Nr.    nummer
----------------------------------  |  ----------------------------------
1        -        -      -     F    |  2        -        -      -     F
```

```
Anzahl der belegbaren B-Kanaele: 2

Belegungsrichtung                    |  Zustand
----------------------------------   |  ----------------------------------
G - gehend                           |  B  - BELEGT
K - kommend                          |  D  - DEFEKT
W - wechselseitig                    |  EB - EDSS1 BELEGT
                                     |  ER - EDDS1 RESERVIERT
                                     |  F  - FREI
                                     |  G  - GESTOERT
Zugriffsrecht                        |  R  - RESERVIERT
----------------------------------   |  S  - SPERRZUSTAND
M - mit                              |  T  - DEFEKT/GESPERRT
O - ohne                             |  V  - BELEGT/GESPERRT
=============================================================================
AOGD<
```

Figure 4: Polycom IP 6000 Subscriber Display Form - General and Service Data

4.3. Allow Access to an External Line (PSTN/ISDN)

To be able to make external calls into PSTN/ISDN this feature must be enabled at a system level (ISM-TCO, task AALM) as well as for the extension (ISM-TCO, task AOLM). Select the extension (command **`aoau:`**, e.g. **`aoau:1319;`**) and use the command **`falm:`** followed by the acronym for the feature or supplementary service to be enabled (e.g. **`falm:AMT;`**). **Figure 5** shows an example command sequence to enable the necessary features / supplementary services for the Polycom IP 6000. To display the set of supplementary services assigned to an extension use the command **`aalm;`** as shown in **Figure 6**.

```
PROL<praw:aolm;
AOLM<aoau:1319;
AOLM<falm:amt,rults,rwlts,kon;
AOLM<falm:rzc,ank,rnu,mak;
AOLM:exit;
```

Figure 5: Supplementary Services assignment to an Extension

```
PROL<1:aolm;
Kommando in Bearbeitung !
AOLM<aoau:1319;
AOLM<aalm;
                                                           13.02.08  17:04:04
  AO-Nummer  AO - Leistungsmerkmale ( Dienst : TLP )
-----------------------------------------------------------------------------

 1319        AMT   RUL   RULTS ARSTS ARR   AUF   CICL1 ANK   CIPL0 CWA
             EMU   API   RWLTS KON   RZN   RZC   ACO   ACOAT RNU   MAK
```

Figure 6: Supplementary Services assigned to an Extension

4.4. Configure the Integral Enterprise VoIP or IPMR board

The Integral System Management (ISM) is also used to configure the VoIP or IPMR board. Select pull down menu PABX Administration - Board - SW Exchange Config Data, enter the appropriate Board number and Execute "Change data".

General data, such as the board IP address are set under "General" (Figure below).
Note that Packet size for both Coder types G.711 and G.729A has to be set to 20ms.

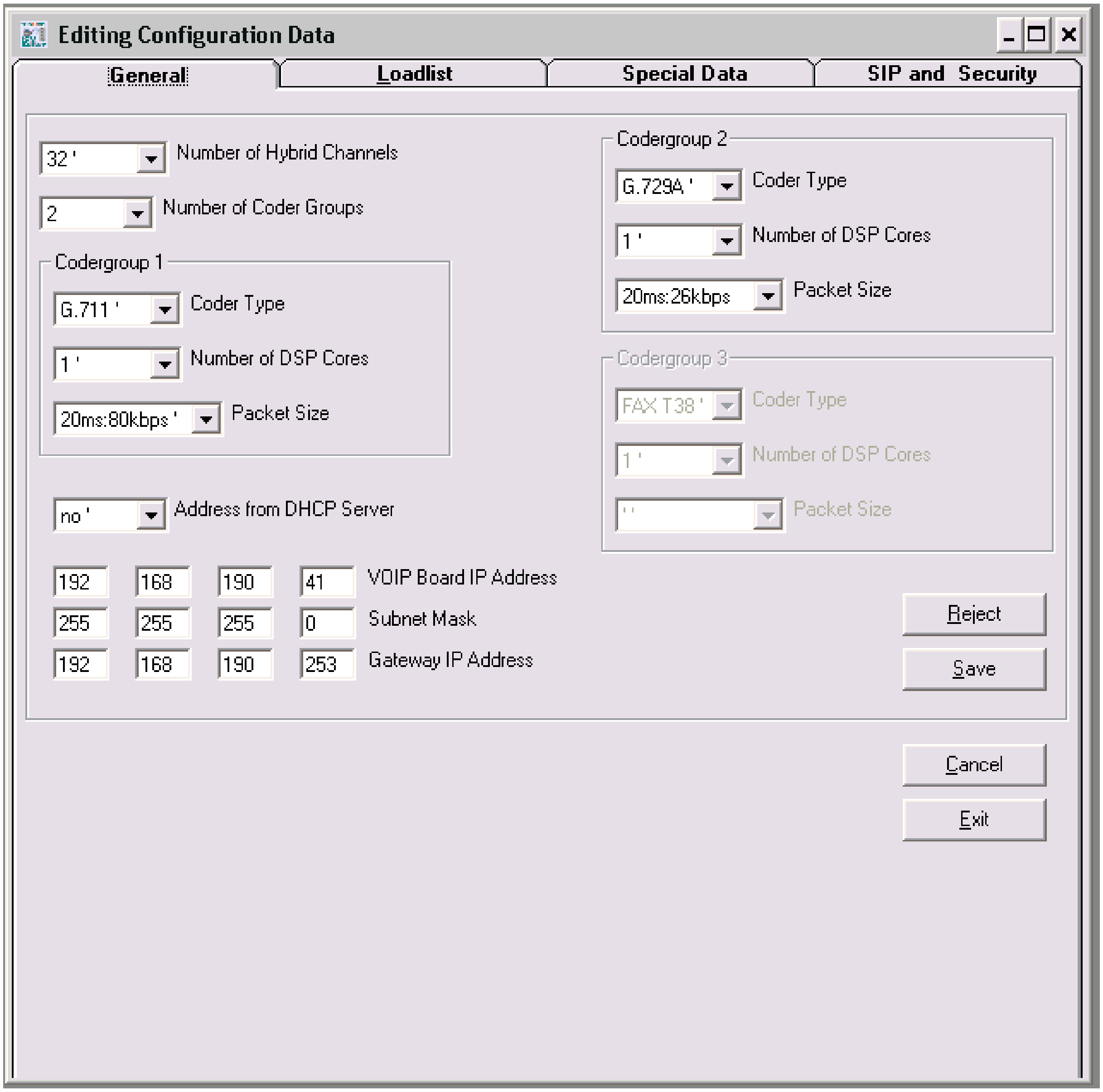


Figure 7: IPMR / VoIP board Configuration Data - General

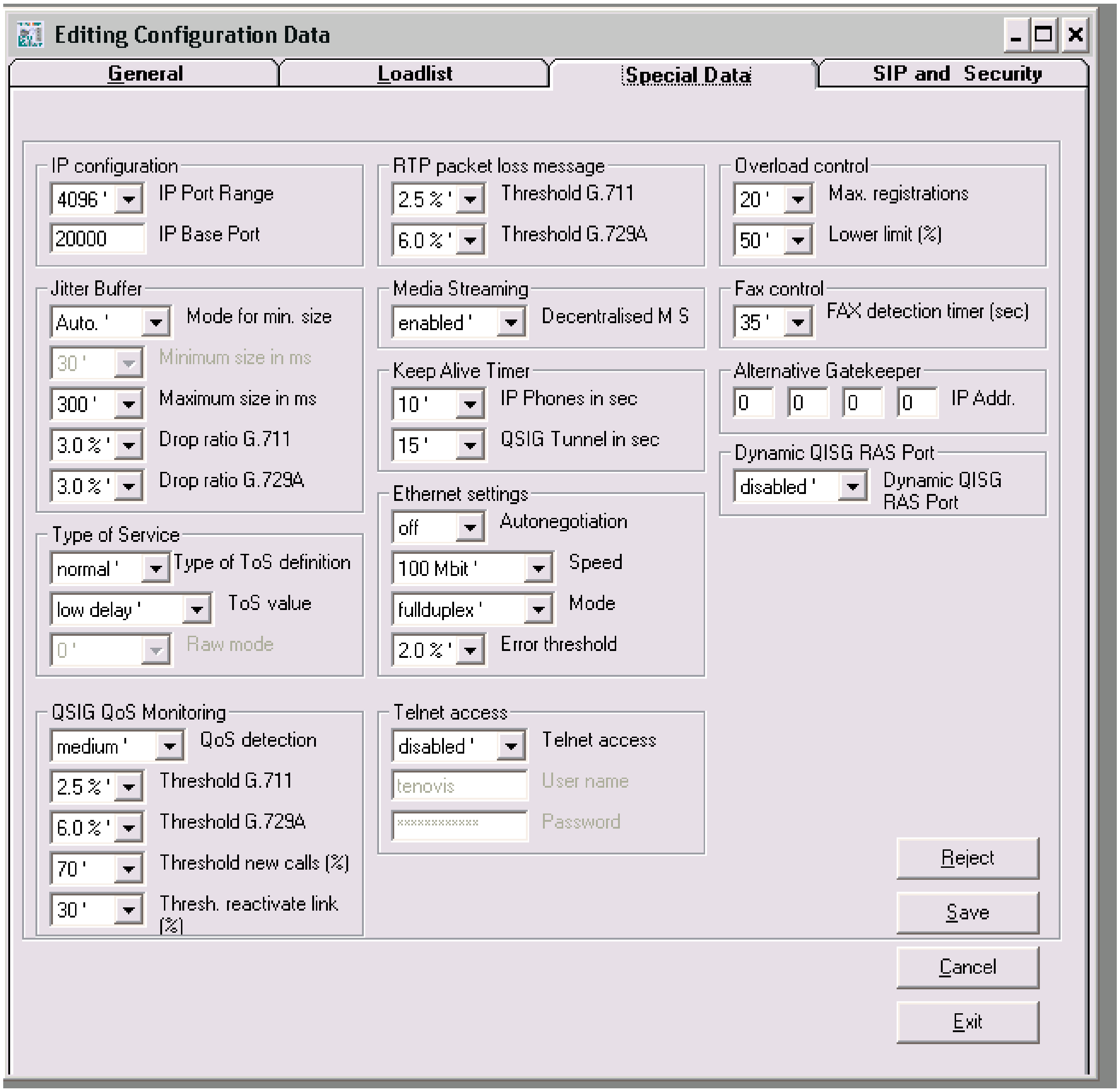


Figure 8: IPMR / VoIP board Configuration Data - Special Data

Under "SIP and Security", depending on the transport protocol to be used for SIP, the SIP Client UDP interface, SIP Client TCP interface, or SIP Client TLS interface respectively must be enabled (Figure below).
If SRTP is to be used for media encryption, S-RTP encryption must be enabled.
If SIP Client UDP and/or TCP Port are left at default (0), the default port 5060 will be assumed (5061 for TLS).

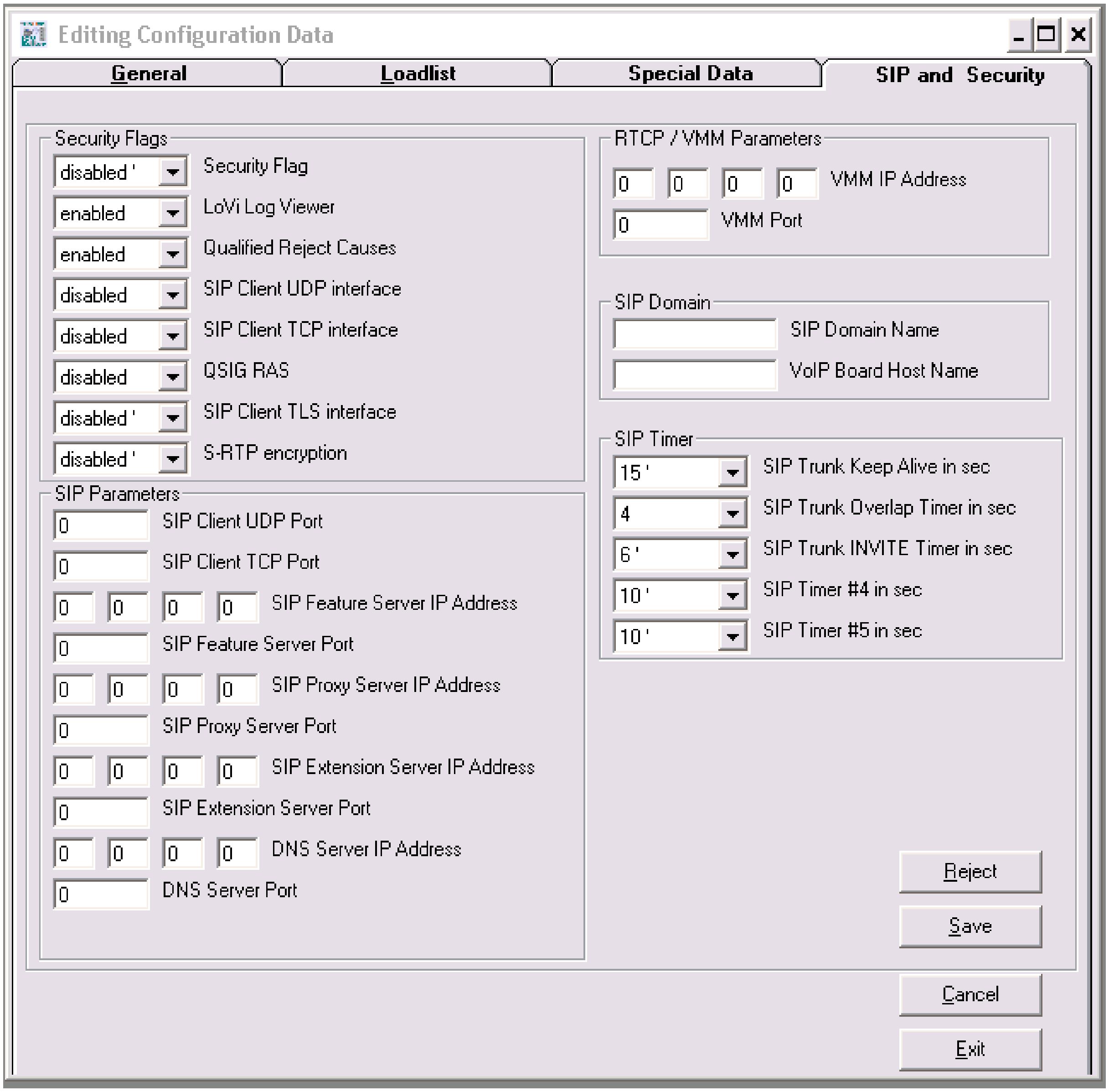


Figure 9: IPMR / VoIP board Configuration Data - SIP and Security

Save changes, Exit, and Execute "Send new data" and "Reset board".

5. Configure the Polycom IP 6000

The Polycom IP 6000 can be assigned an IP address either manually, or via DHCP. Configuration via the keypad interface on the Polycom IP 6000, using the following sequence:

- Press the "Menu" key.
- Select "Settings".
- Select "Advanced", and enter the administrator PIN when prompted (default 456).
- Select "Network".
- Select either "DHCP" or "STATIC" followed by the IP address, Netmask and Gateway address to be assigned to the unit.

Once the Polycom IP 6000 has been assigned an IP address, the configuration procedure can be preformed either with the keyboard/display of the Polycom IP 6000, or via Web browser, as illustrated by this section of the document.

To use a web browser, enter the IP address of the Polycom IP 6000 into the URI "Address" field of the browser, which causes the screen shown below to appear. Enter User name (default Polycom) and Password (default 456), and click "OK".

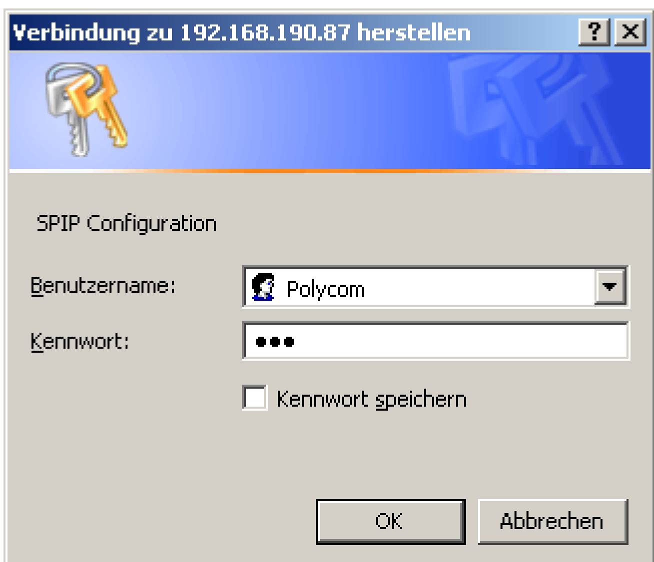


Figure 10: Polycom IP 6000 Login Screen

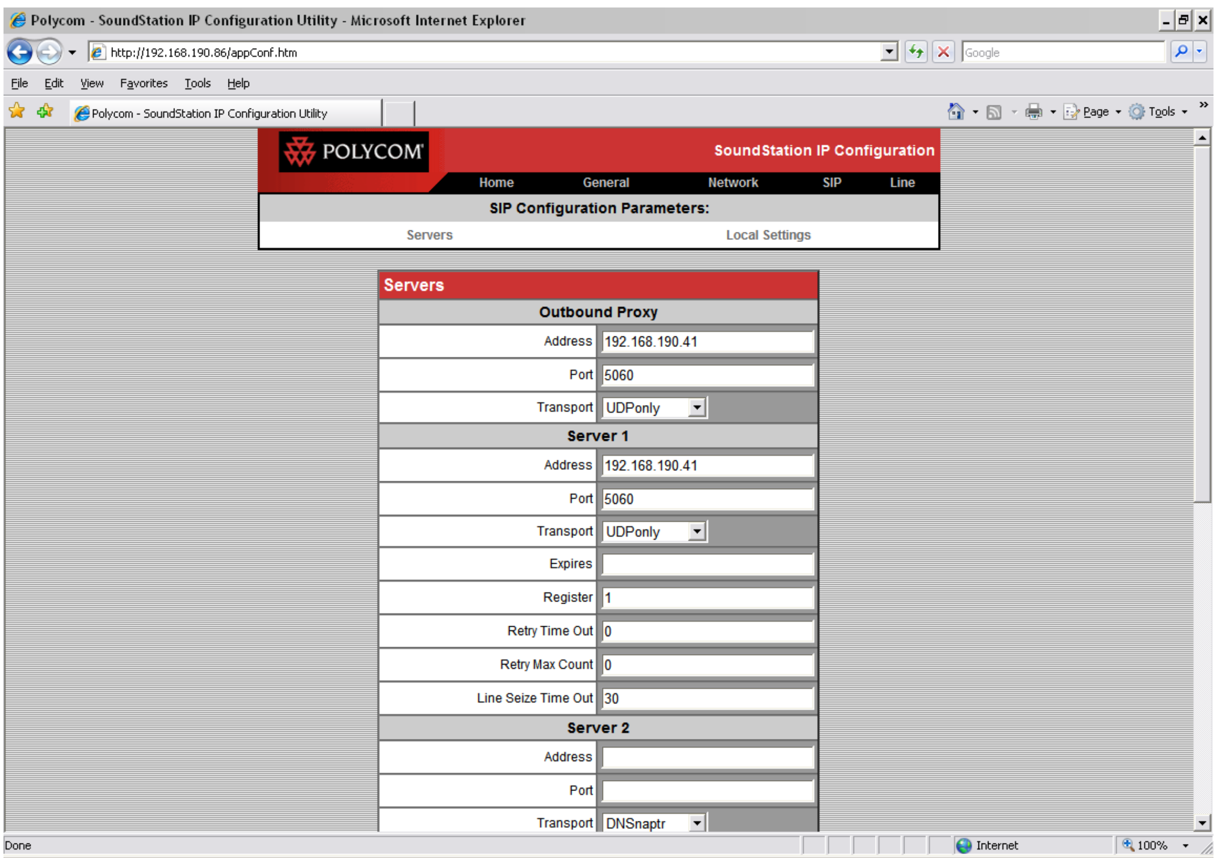
5.1. Configure SIP

Select the “Settings” tab from the top of the screen, and then “SIP” from row of the underlying set of tabs. Enter the parameters shown in the following table for "Outbound Proxy" and "Server 1" and click "Submit" at the bottom of the "Servers" box.
Other values on that page are left as default.

Parameter	Usage
Address	Enter the IP Address of the VoIP/IPMR board.
Port	Enter the Port related to the selected transport Protocol.
Transport	Select the Transport protocol (See Note below).
Register	1

Table 4: SIP Configuration Parameters

Note: For SIP via UDP select UDPonly and Port 5060, SIP via TCP TCPonly and Port 5060, for SIP via TLS select TLS and Port 5061.



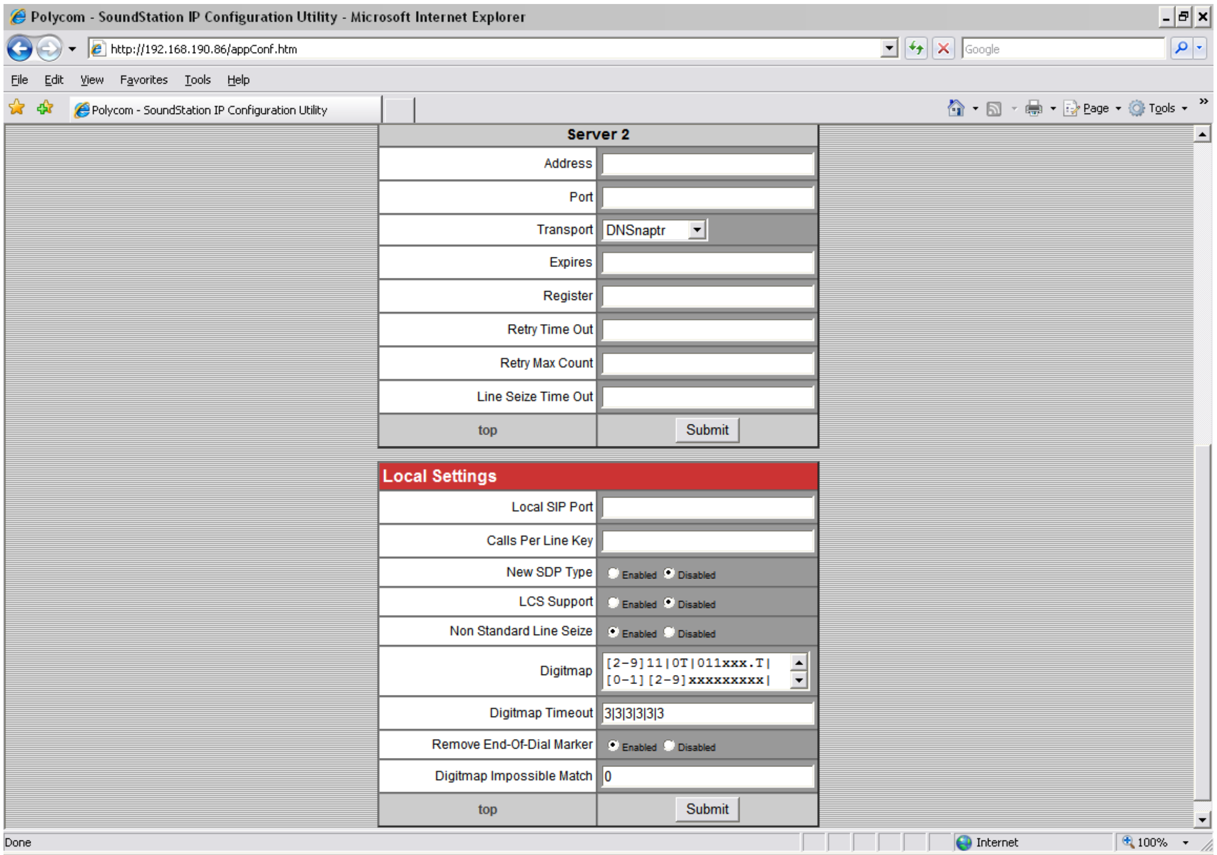


Figure 11: Polycom IP 6000 SIP Settings Screen

5.2. Configure Line

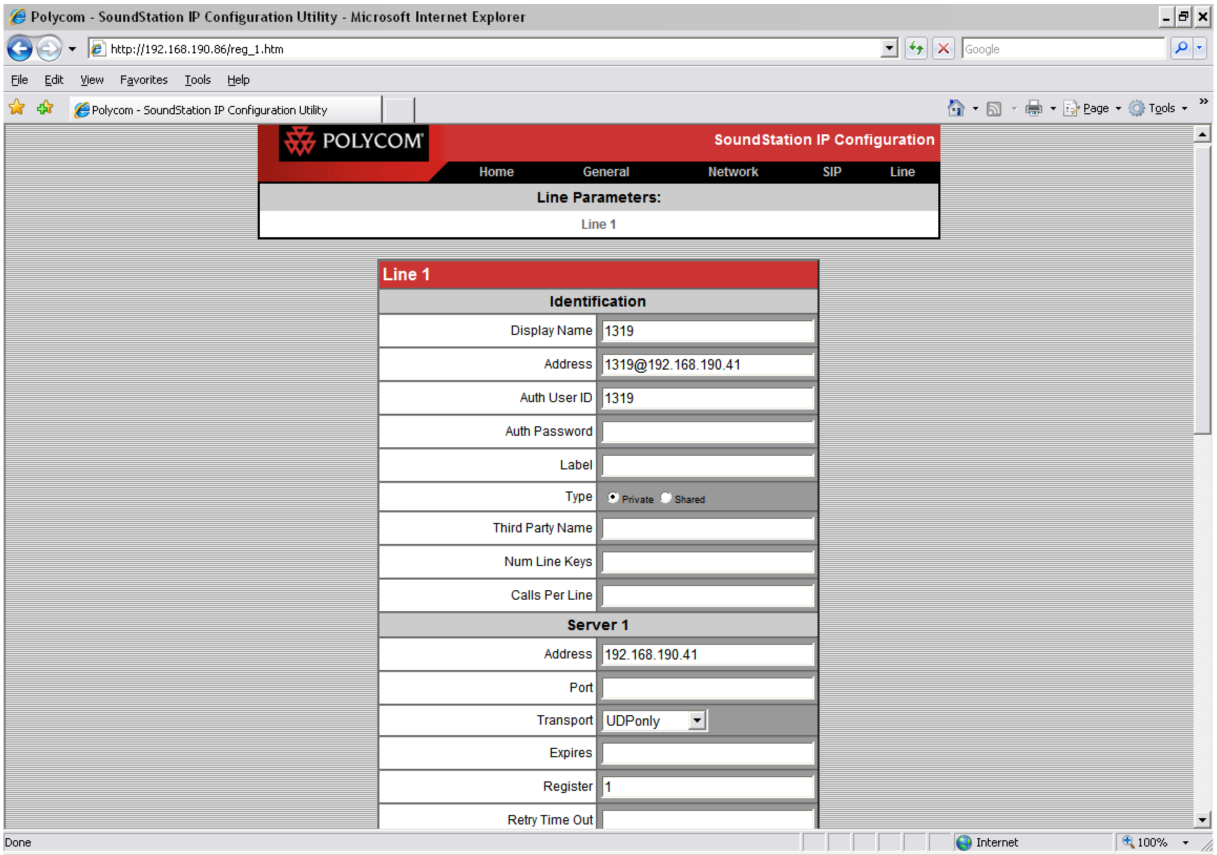
Select the “Settings” tab from the top of the screen, and then “Line” from row of the underlying set of tabs. Enter the parameters shown in the following table for "Identification" and "Server 1" and click "Submit" at the bottom of the page.

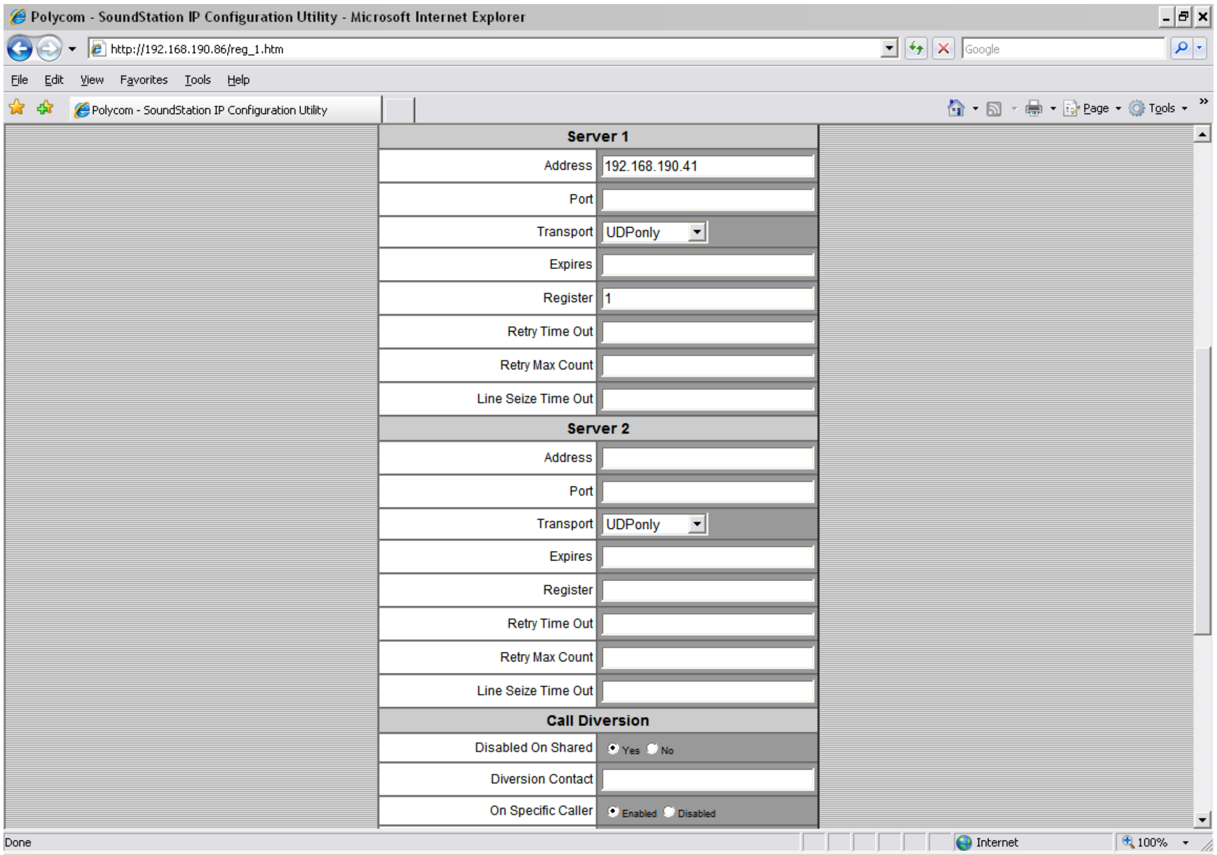
Parameter	Usage
Identification:	
Display name	Enter a descriptive name for the account.
Address	Enter User@IP address of the VoIP/IPMR board.
Auth User ID	Enter the extension number of the user.
Auth Password	Enter the “Password” which corresponds with the extension number.
Server 1:	
Address	Enter the IP Address of the VoIP/IPMR board.
Port	Enter the Port related to the selected transport Protocol.
Transport	Select the Transport protocol (Note 1).
Register	1

Table 5: Line Configuration Parameters

Note 1: For SIP via UDP select UDPonly and Port 5060, SIP via TCP TCPonly and Port 5060, for SIP via TLS select TLS and Port 5061.

In addition, it is recommended to disable Call Diversion On Specific Caller, Forward All, On Busy, and On No Answer. The default being enable causes the Polycom IP 6000 to offer a soft key Forward, which is not supported by the Integral Enterprise.


Polycom - SoundStation IP Configuration Utility - Microsoft Internet Explorer
http://192.168.190.86/reg_1.htm
POLYCOM
SoundStation IP Configuration
Home
General
Network
SIP
Line
Line Parameters:
Line 1
Line 1
Identification
Display Name 1319
Address 1319@192.168.190.41
Auth User ID 1319
Auth Password
Label
Type Private Shared
Third Party Name
Num Line Keys
Calls Per Line
Server 1
Address 192.168.190.41
Port
Transport UDPonly
Expires
Register 1
Retry Time Out
Done
Internet
100%


Polycom - SoundStation IP Configuration Utility - Microsoft Internet Explorer
http://192.168.190.86/reg_1.htm
File Edit View Favorites Tools Help
Polycom - SoundStation IP Configuration Utility
Page Tools
Server 1
Address 192.168.190.41
Port
Transport UDPonly
Expires
Register 1
Retry Time Out
Retry Max Count
Line Seize Time Out
Server 2
Address
Port
Transport UDPonly
Expires
Register
Retry Time Out
Retry Max Count
Line Seize Time Out
Call Diversion
Disabled On Shared Yes No
Diversion Contact
On Specific Caller Enabled Disabled
Done
Internet
100%

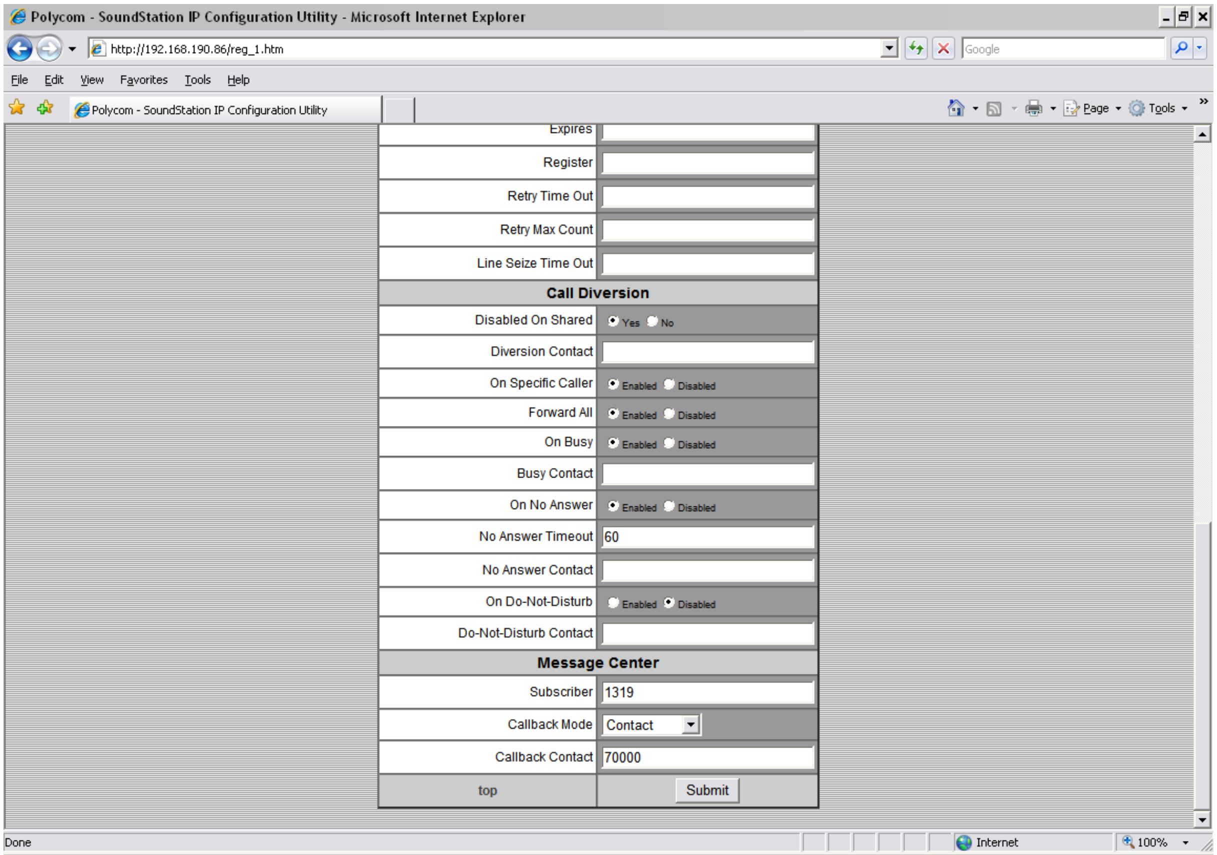


Figure 12: Polycom IP 6000 Line Settings Screen

5.3. Configure Media Settings

Select the embedded "Media" tab from within the "Settings" tab. The codec selected by Polycom IP 6000 users is dependent on fidelity requirements and bandwidth availability. Most of the testing was done with the G.711A codec, although other codec combinations were tested to ensure proper codec interoperation. The codec selection configured here must be compatible with the codecs configured for Avaya Integral Enterprise in **Figure 4**. Click "Save" to complete the configuration sequence.

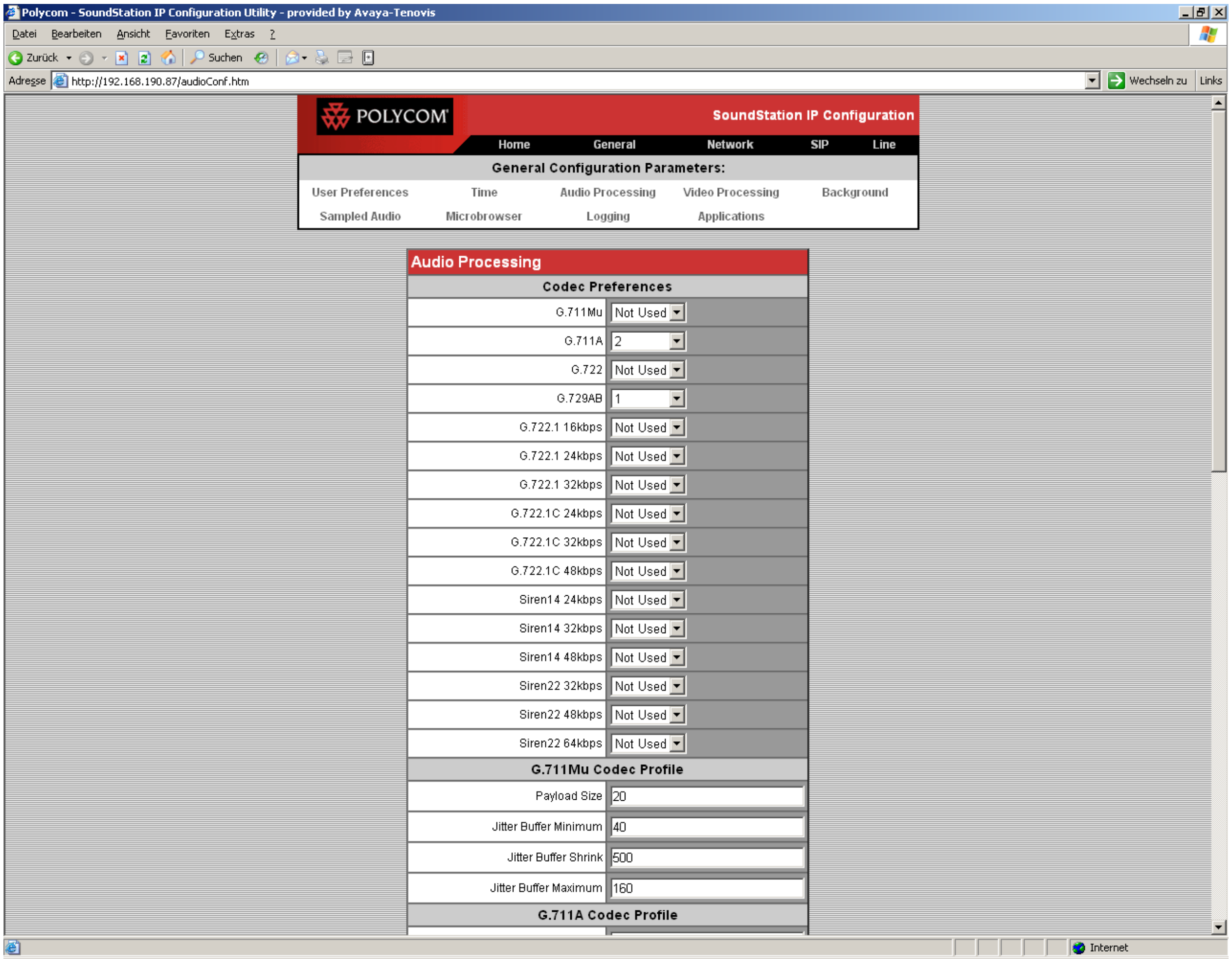


Figure 13: Polycom IP 6000 Media Settings Screen

5.4. Configure TLS and SRTP

For **TLS** select Transport TLS and Port 5061 under SIP - Outbound Proxy and Server 1 as well as under Line - Server 1.

The handshake process between server and client (IP 6000) to establish a TLS connection requires that a certificate has been downloaded into the client.
The IP 6000 offers two ways to download a certificate:

1) Menu/Settings/Advanced/PIN (456)/Admin Settings/SSL Security/CA Certificates:
Configure CA Certificates:
e.g. tftp://server.name/certificate.crt
Install Custom CA Certificates:
[x] All Certificates

(Technical Bulletin 17877)

2) The certificate can be inserted in the bootFix.cfg file:

```
... device.sec.SSL.customCert="XXX...XXX"/>
</device>
```

Note: The certificate must not contain any linefeeds!

To force download of the bootFix.cfg file (containing the certificate), e.g. from a TFTP server, an appropriate entry must be made in the 0004f2e41db3.cfg file (0004f2e41db3 being the MAC address of the IP 6000):

```
CONFIG_FILES="bootFix.cfg, phone1.cfg, sip.cfg"
```

The Server Menu must be configured for TFTP, e.g. as follows:

Menu/Settings/Advanced/PIN (456)/Admin Settings/Network Configuration/Server Menu:

Type:	TFTP	
Address:	192.168.190.3	*(TFTP server)*
User:	polycom	*(directory-name*.cfg)*
Password:		*(not required, depends on TFTP server)*

The following configuration files are being downloaded upon reboot (Menu/Settings/Advanced/PIN (456)/Reboot Phone):

0004f2e41db3.cfg	*(0004f2e41db3 being the MAC address of the IP 6000)*
0004f2e41db3-phone.cfg	
bootFix.cfg	*(containing the certificate)*
phone1.cfg	
sip.cfg	

SRTP can only be configured via the configuration file sip.cfg which is downloaded from a TFTP server:

Menu/Settings/Advanced/PIN (456)/Admin Settings/Network Configuration/Server Menu:

Type:	TFTP	
Address:	192.168.190.3	*(TFTP server)*
User:	polycom	*(directory-name\sip.cfg)*
Password:		

Two scenarios can be distinguished:

1) Scenario One: Mixed SRTP and non-SRTP endpoints.

The following sip.cfg configuration file changes are suggested:

```
sec.srtp.enable="1"
sec.srtp.leg.enable="1"
sec.srtp.offer="1"
sec.srtp.leg.allowLocalConf="0"
```

2) Scenario Two: Full Security.

In this scenario, the assumption is made that all elements of the solution are SRTP-capable. So security is mandated.

The following sip.cfg configuration file changes are suggested:

```
sec.srtp.enable="1"
sec.srtp.leg.enable="1"
sec.srtp.offer="1"
sec.srtp.require="1"
```

(Technical Bulletin 25751)

For compatibility reasons with the Avaya Integral Enterprise the following settings concerning RTCP are required in the sip.cfg configuration file:

```
sec.srtp.sessionParams.noEncrypRTCP.offer="1"
sec.srtp.sessionParams.noEncrypRTCP.require="1"
sec.srtp.sessionParams.leg.noEncrypRTCP.offer="1"
sec.srtp.sessionParams.leg.noEncrypRTCP.require="1"
```

6. General Test Approach and Test Results

The test method employed can be described as follows:

- Avaya Integral Enterprise was configured to support various local ISDN and H.323 IP telephones, as well as both the SIP accounts of the Polycom IP 6000.
- The Polycom IP 6000 was configured to use its two SIP accounts to act as separate SIP telephone endpoints.
- Various telephony operations involving the Polycom IP 6000 and Avaya Integral Enterprise were performed manually.
- The SIP protocol exchanges were monitored with a protocol trace program to verify the correct protocol exchanges.

7. Verification Steps

- Verify that the Polycom IP 6000 web configuration site can be opened by a web browser after having assigned an IP address to the device.
- Verify that the Polycom IP 6000 can register and re-register with the VoIP or IPMR board of the Avaya Integral Enterprise.
- Verify that the Polycom IP 6000 can make and receive calls from its SIP account line.
- Verify that the Polycom IP 6000 can make multiple simultaneous calls and toggle between those calls.
- Verify that the Polycom IP 6000 hold/retrieve feature is compatible with Avaya Integral Enterprise.
- Verify that the Polycom IP 6000 can create conferences manually.
- Verify that the Polycom IP 6000 codec support is compatible with Avaya Integral Enterprise.
- Verify that the Polycom IP 6000 DTMF facility is compatible with Avaya Integral Enterprise.
- Verify that the Polycom IP 6000 can operate from both its external power supply or from power that it received via its Power Over Ethernet connection.
- Verify that the Polycom IP 6000 can recover from interruptions to its Ethernet interface.
- Verify that the Polycom IP 6000 can use both UDP and TCP as transport protocols for SIP.
- Verify that the Polycom IP 6000 can use security features (encryption) such as TLS for SIP and SRTP for media.

8. Conclusion

The Polycom IP 6000 can be used with Avaya Integral Enterprise to enable those present in a room to participate in a telephone conversation. The configuration described in these Application Notes has been successfully compliance tested.

9. Additional References

This section references documentation relevant to these Application Notes. The Avaya product documentation is available at http://support.avaya.com.

[1] *Administrator's Guide for the Polycom SoundPoint IP/SoundStation IP Family, August 2008 Edition, 1725-11530-310 Rev. A, SIP 3.1*

[2] *User Guide for the Polycom SoundStation IP 6000 Phone, August 2008 Edition, 1725-15645-001 Rev. B, SIP 3.1*

[3] *Quick Tip 44011: Register Standalone SoundPoint IP, SoundStation IP, ...*

[4] *Technical Bulletin 25751: Secure Real-Time Transport Protocol on SoundPoint IP Phones*

[5] *Technical Bulletin 17877: Using Custom Certificates With SoundPoint® IP Phones*

All available at:
http://www.polycom.com/support/voice/soundstation_ip_series/soundstation_ip6000.htm

Several Internet Engineering Task Force (IETF) standards track RFC documents were referenced within these Application Notes. The RFC documents may be obtained at: http://www.rfc-editor.org/rfcsearch.html.

[6] RFC 3261 - *SIP (Session Initiation Protocol)*, June 2002, Proposed Standard

[7] RFC 2833 - *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*, May 2000, Proposed Standard

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.