



# Avaya B179 SIP Conference Phone

## Installation and Administration Guide

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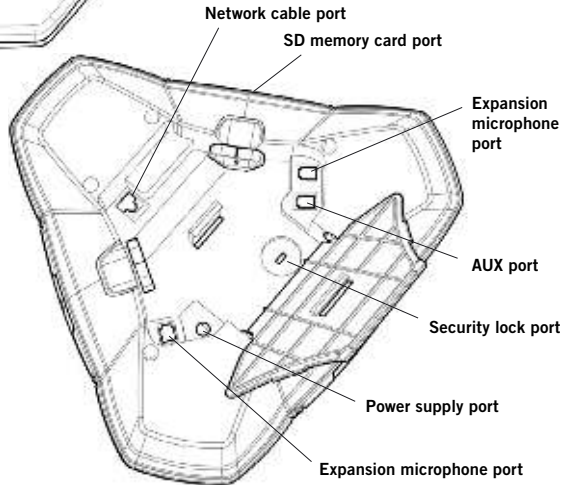
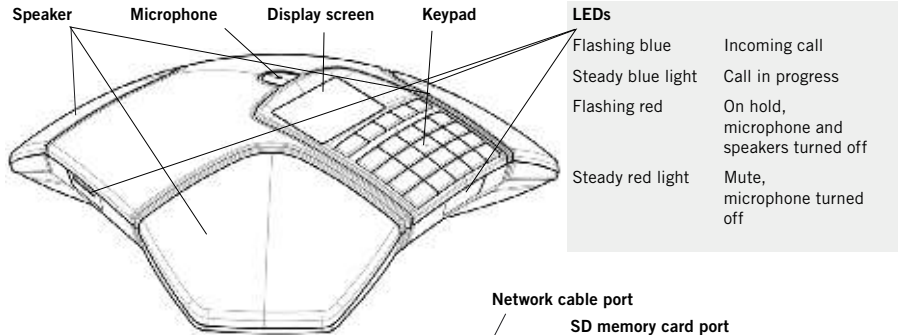
This document only includes setup, registration of accounts and configuration. The use of the conference phone is described in the *Avaya B179 SIP Conference Phone - Quick Reference Guide* (16-603916) and the *Avaya B179 SIP Conference Phone - User Guide* (16-603918). The latest version of all documentation can be downloaded from **support.avaya.com**.

Please note that there are also supporting **Application Notes** describing the steps to configure the Avaya B179 SIP Conference Phone to work with certain systems and also how to configure the systems (eg. administer SIP extensions).

# CONTENT

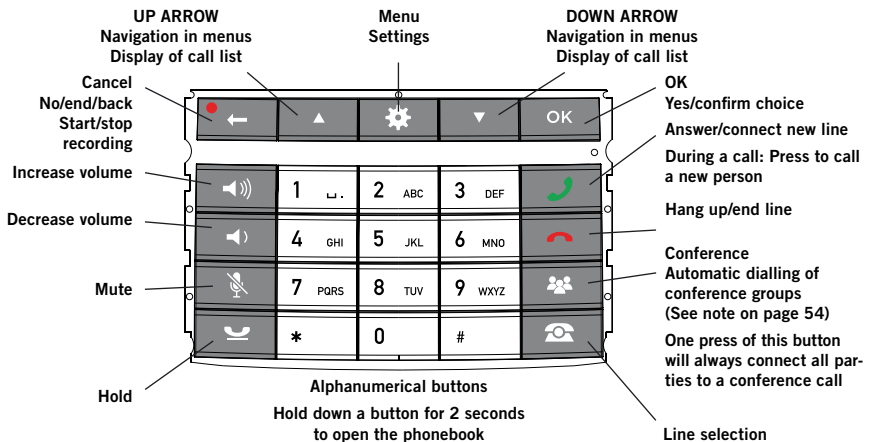
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# DESCRIPTION



**Maintenance**

Clean the equipment with a soft, dry cloth. Never use liquids.



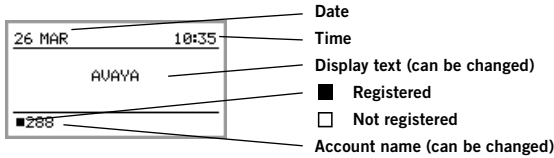
ⓘ Some Avaya B179 have a different keypad with other symbols. This does not affect the functions of the buttons.

# DISPLAY INFORMATION AND WEB INTERFACE


## DISPLAY SCREEN

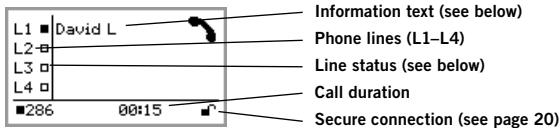
### On Hook

Press  to display this screen.



### Off Hook


Press  to display this screen.




#### Line status:

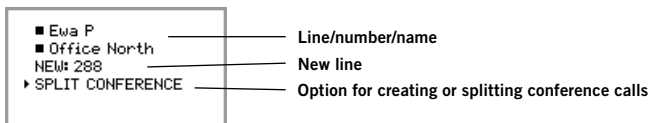
- Line free (Before account name – telephone not registered)
- Line connected (Before account name – telephone registered)
- Line on hold (“HOLD” displayed on the screen – all calls on hold)
- Line (called party) busy
- Own line put on hold by other party
- Recording call
- Secure connection

#### Information text displays one of the following:

- Number or name of each phone line  
(The name will be displayed if a number is in the phone book)
- Explanation of what you should do (For example **ENTER NUMBER**)
- Status (For example  when you place all calls on hold)

### Line menu

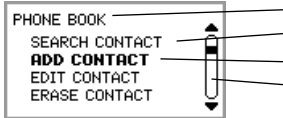
Press  to switch to and from this menu.



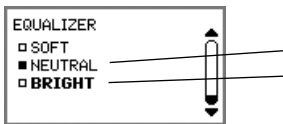
# DISPLAY INFORMATION AND WEB INTERFACE

## Menu

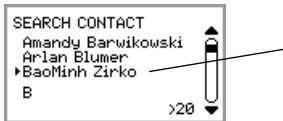
Press **⚙** to switch to and from a menu.



Current menu  
 Submenu  
 Marked option – open by pressing **OK** button  
 Scrolling list  
 (indication of where the marked option is in the list or menu)



List of setting options:  
 Existing settings  
 Marked option – select by pressing **OK** button



List of names:  
 Marked name – select by pressing **OK** button

## NAVIGATION AND SELECTION IN MENUS

- ⇒ Press **⚙**.
- ⇒ Select the option you want from the menu using the arrow buttons.
- ⇒ Confirm by pressing **OK** to select the marked option.
- ⇒ Cancel the setting or go back one level in the menu by pressing **• ←**.
- ⇒ Quit the menu by pressing **⚙** again.
- ① Note that after you have made changes to a setting, you must press **OK** to activate the setting.
- 🌀 It is possible to open a menu option directly by pressing the number button that corresponds to the position of the option in the menu (e.g. 2 to open **PHONE BOOK** and then 3 to select **EDIT CONTACT**).

## Writing style in instructions

In the instructions, **⚙ > SETTINGS (6)** means you should:

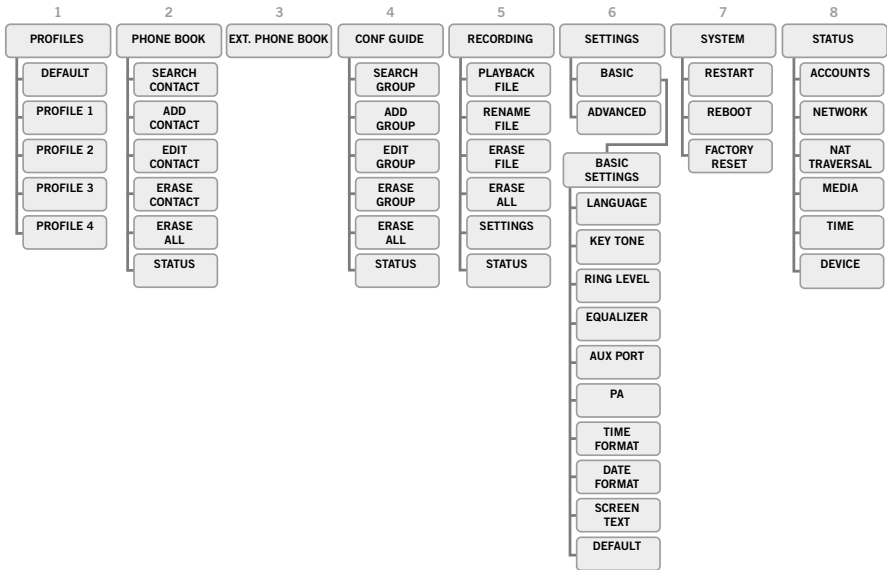
- ⇒ Press **⚙**.
- ⇒ Mark the **SETTINGS** option using the arrow buttons and confirm by pressing **OK** to open the menu (or press button number **6**).

Correspondingly, **Phone book > Conference Guide** in the web interface means you should select Menu Phone book and the Conference Guide tab.



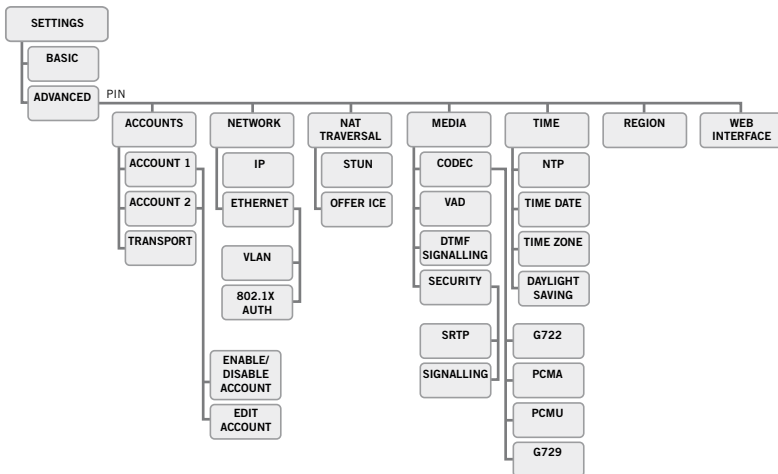
# DISPLAY INFORMATION AND WEB INTERFACE


## Menu tree



## Menu tree, advanced settings

The advanced settings are protected by administrator's PIN code. The default value is **1234**.



 The simplest way to make settings and edit contacts is using a PC and the Avaya B179 web interface.

# DISPLAY INFORMATION AND WEB INTERFACE

## USING THE WEB INTERFACE

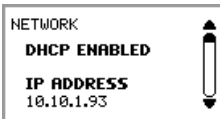
You can use the web browser of a PC connected to the same network to manage contacts, conference groups and settings in the Avaya B179.

For security reasons, recordings can only be managed directly on the Avaya B179. All other settings that can be made directly on the Avaya B179 can also be made via the web interface. It is also possible to import and export contacts and conference groups, name user profiles and change PIN codes, which can only be done via the web interface. The administrator can also view logs, update software and create a configuration file.

The default setting for the PIN code is **0000** for the user account (Default, Profile 1, Profile 2, Profile 3 and Profile 4) and 1234 for the administrator's account (Admin). We recommend that you change the PIN codes in order to protect the settings. The code may consist of eight digits. The administrator can always view and change the PIN codes to the user accounts. The administrator's PIN code can only be reset with a complete reset to factory settings.

### Checking IP address

- ⇒ Press **⚙** and select the sub menu **STATUS > NETWORK (8,2)**.
- ⇒ Check the conference phone's network address under the heading **IP ADDRESS**.



- ① Use this address to log into the web server in the conference phone.

### Login

- ⇒ Log into the web server in Avaya B179 by entering the phone's network address in your computer's web browser.
- ⇒ Select **Admin** as Profile and enter your PIN.

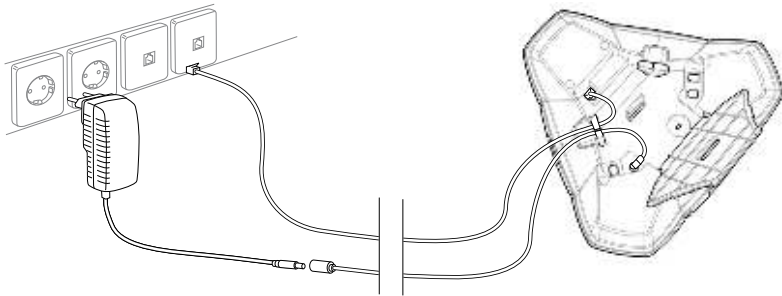


# INSTALLATION

The printed Installation Guide provides brief and simplified installation instructions. The guide includes the basic settings for a quick start and works in most cases.

## CONNECTING

- ⇒ Connect the Avaya B179 to the network as illustrated below.
- ⇒ Plug the Avaya B179 into the mains using the power adapter as illustrated below.
- ⓘ The Avaya B179 can be driven directly from the network (Power over Ethernet, Class III) if the network supports this.



- ⇒ Place the conference phone in the middle of the table.

The Avaya B179 must obtain a network address and be registered in a SIP PBX before it can be used. The easiest way to register an account and make the settings in the Avaya B179 is using a computer connected to the same network and via the integrated web server.

# INSTALLATION


## OBTAINING A NETWORK ADDRESS

### Connecting to a network with DHCP

See “Check IP address” under “USING THE WEB INTERFACE” on page 6.

### Connecting to a network with static IP addresses

You need the IP address, host name, domain, netmask, gateway, DNS 1, and DNS 2. The host name can be set freely. The domain and secondary DNS can be left blank.

- ⇒ Press  and select **SETTINGS > ADVANCED** (6,2).
- ⇒ Enter the PIN code.
  - ① The default code is **1234**.
- ⇒ Select **NETWORK** (2)
- ⇒ Select **STATIC IP**.
- ⇒ Enter values for the IP ADDRESS.
  - ① Enter three digits (begin with 0 if necessary), press **OK**, enter three digits, and so on.
- ⇒ Enter HOST NAME
  - Default is avaya.*
- ⇒ Enter DOMAIN
- ⇒ Enter NETMASK
- ⇒ Enter GATEWAY
- ⇒ Enter DNS 1
- ⇒ Enter DNS 2
  - The display shows DONE.*

## LOGIN

See “Login” under “USING THE WEB INTERFACE” on page 6.

# INSTALLATION

## SOFTWARE UPGRADE AND BASIC SETTINGS

The following settings should be done during installation.

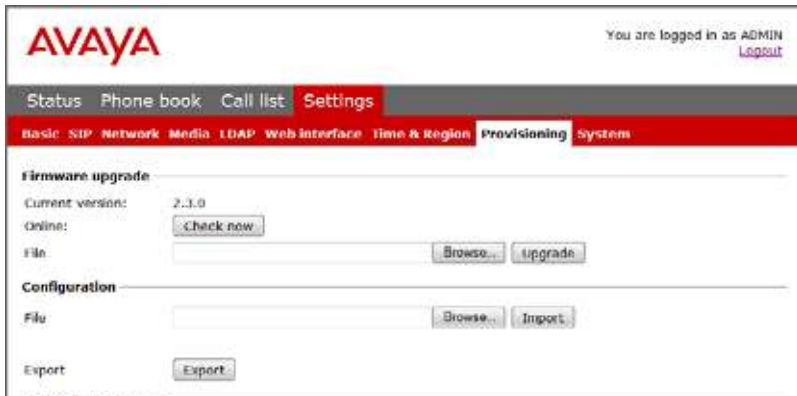
- ① See page 48 about Device management if you are responsible for installing or upgrading many phones.

Note that all settings on the Basic tab also affect the user profile Default. Other user profiles can be changed individually. The settings on the Basic tab, except the name and PIN for Admin, can be modified by any user. Other settings require a login as Admin.

### Upgrade software

See the heading “PROVISIONING – UPGRADE AND CONFIGURATION” on page 40 for a detailed description and upgrading options.

- ⇒ Select **Settings > Provisioning**.



- ⇒ Click on **Check Now**.
- ⇒ Compare the latest version with the current version (shown on the web page).
- ⇒ If you want to upgrade, select the desired version in the list box and click on **Upgrade**.  
*The browser window and the display on the Avaya B179 shows that the upgrade has begun.*
- ① The download and installation can take several minutes. Do not interrupt the upgrade and do not disconnect plugs to the Avaya B179 during the upgrade. Interrupting the upgrade may render the conference phone inoperable.
- ⇒ When installation is complete, the text “Upgrade Complete. The unit will be rebooted.” is shown in your browser, and after a while you hear the Avaya music signature, which indicates that the conference phone has started.

# INSTALLATION

## Setting time and region

⇒ Select **Settings > Time & Region**.

The screenshot shows the Avaya B179 SIP Conference Phone web interface. The top navigation bar includes 'Status', 'Phone book', 'Call list', and 'Settings'. The 'Settings' menu is expanded, showing 'Basic', 'SIP', 'Network', 'Media', 'LDAP', 'Web interface', 'Time & Region', and 'Provisioning system'. The 'Time & Region' page is displayed, with the following settings:

- Time**
  - Enable NTP:  On  Off
  - Time: 07:45:30
  - Date: 2011-02-08
  - Timezone: UTC+1 00
  - NTP Server: pool.ntp.org
- Region**
  - Region: SWE Sweden
- Daylight saving**
  - Enable DST:  Yes  No
  - DST Timezone (hh:mm): UTC+2 00
  - DST Mode:  Automatic  Manual
- Start fixed date**
  - Start fixed date:  Yes  No
- Start**
  - Start month: March
  - Start day of week: Sunday
  - Start week: 5
  - Start time (HH:mm): 02:00
- Stop fixed date**
  - Stop fixed date:  Yes  No
- Stop**
  - Stop month: October
  - Stop day of week: Sunday
  - Stop week: 5
  - Stop time (HH:mm): 02:00

Buttons: Save, Cancel

⇒ Select the time zone and, if you wish, correction for DST (Daylight saving).

① It is also possible to set the time and date manually or choose a different time server.

⇒ Select the region where you are.

① This setting affects the frequency and duration of the signaling tones (ring signal, busy tone, etc).

⇒ Save the setting.

*The Avaya B179 reboots with the new settings.*

# INSTALLATION

## Changing the language

⇒ Select **Settings > Basic**.

The screenshot shows the Avaya web interface. At the top, it says "AVAYA" and "You are logged in as ADMIN Logout". Below that is a navigation bar with "Status", "Phone book", "Call list", and "Settings". Under "Settings", there are tabs for "Basic", "SIP", "Network", "Media", "LDAP", "Web interface", "Time & Region", "Provisioning", and "System".

The "Profiles" section contains a table:

	Name	PIN	Edit	Set
Default	DEFAULT	****	Edit	Set
Profile 1	PROFILE 1	****	Edit	Set
Profile 2	PROFILE 2	****	Edit	Set
Profile 3	PROFILE 3	****	Edit	Set
Profile 4	PROFILE 4	****	Edit	Set
Admin	ADMIN	****	Edit	Set

The "Preferences" section includes:

- Phone language: English
- Ring level: Level 4
- Key tone:  On  Off
- Recording:  On  Off
- Recording tone:  On  Off
- Auxiliary port:  Headset  PA (Refer to the user guide before changing to PA)
- Time format:  12 Hour  24 Hour
- Date format:  YYYY-MM-DD  MM/DD/YYYY  MM-DD-YYYY
- Equalizer:  Soft  Neutral  Bright
- Screen text: AVAYA

At the bottom, there are "Save" and "Cancel" buttons.

⇒ Select the desired language in the list box after Language and save the setting.

① Note, this only affects the phone language, not the web interface.

## Changing the PIN

We recommend that you change the PIN code for Admin from the default setting to protect the settings. Make a note of the new PIN code and keep it in a safe place. The administrator's PIN code can only be reset by a full factory reset!

⇒ Select **Settings > Basic** and click the **Edit** button on the Admin line.

⇒ Enter a new PIN.

① The PIN code may consist of 8 digits.

⇒ Click on the **Set** and **Save** buttons.

# INSTALLATION


## REGISTERING AN ACCOUNT

The conference phone supports two accounts. The second account is a fallback account that is automatically used if the phone fail to register to the main account. Note that this account then will be used as long as the fallback account is available. If the phone fails to register to the fallback account, it will try to use the main account again.

To register your phone, you must have access to the account information and all necessary settings that the SIP PBX requires.

See the heading “SIP” on page 17 for a detailed description of all settings.

⇒ Select **Settings** > **SIP**.


You are logged in as ADMIN [Logout](#)

Status
Phone book
Call list
Settings

Basic
SIP
Network
Media
LDAP
Web interface
Time & Region
Provisioning
System

---

**Main account**

Enable account	<input checked="" type="radio"/> Yes <input type="radio"/> No	
Account name ⓘ	<input type="text" value="Conf 2"/>	Realm ⓘ <input type="text" value="*"/>
User ⓘ	<input type="text" value="140"/>	Authentication name ⓘ <input type="text" value="140"/>
Registrar ⓘ	<input type="text" value="192.168.0.1"/>	Password <input type="password" value="*****"/>
Proxy ⓘ	<input type="text"/>	Registration interval ⓘ <input type="text" value="1800"/>

**Fallback account**

Enable account	<input checked="" type="radio"/> Yes <input type="radio"/> No	
Account name	<input type="text" value="Conf 2"/>	Realm <input type="text" value="*"/>
User	<input type="text" value="140"/>	Authentication name <input type="text" value="140"/>
Registrar	<input type="text" value="192.168.0.2"/>	Password <input type="password" value="*****"/>
Proxy	<input type="text"/>	Registration interval <input type="text" value="1800"/>

**NAT Traversal**

STUN ⓘ	<input type="radio"/> On <input checked="" type="radio"/> Off	STUN host <input type="text"/>
Offer ICE	<input type="radio"/> Yes <input checked="" type="radio"/> No	
TURN ⓘ	<input type="radio"/> On <input checked="" type="radio"/> Off	TURN user <input type="text"/>
TURN host	<input type="text"/>	Password <input type="password"/>

**Advanced**

Enable SIP Replaces	<input type="radio"/> Yes <input checked="" type="radio"/> No
Enable Blind Transfer	<input type="radio"/> Yes <input checked="" type="radio"/> No
Allow contact rewrite	<input type="radio"/> Yes <input checked="" type="radio"/> No
Outbound proxy	<input type="text"/>

**Transport**

Protocol	<input type="radio"/> UDP <input checked="" type="radio"/> TCP <input type="radio"/> TLS <input type="radio"/> SIP5	Please check corresponding media signaling setting
Local TCP port	<input type="text" value="5060"/>	



# INSTALLATION

- ⇒ Click **Yes** at Enable account under Main account.
- ⇒ Enter the account information you have received.
- ① Account name can be chosen freely and is the name or phone number you want to appear in the phone display.
- ⇒ Leave the default values if you have no other information.
- ⇒ Select a method of NAT traversal if you have received this information.
- ⇒ Select a different transport protocol if you have received this information. See page 20 about using a secure transport protocol.
- ⇒ Save the settings by clicking the **Save** button.

*The Avaya B179 responds by showing REGISTERING. If registration is successful, your selected account name will appear at the bottom of the display screen next to a shaded square.*



## Make media settings

- ⇒ Select a different codec priority, if you do not accept the default settings. See page 26.
- ⇒ Select a SRTP option if you need a secure media protocol. See page 27. Note that this also requires a corresponding transport setting on the SIP tab.

# SETTINGS

Almost all settings can be done directly on the Avaya B179. See “NAVIGATION AND SELECTION IN MENU” on page 4 for using the menu system. We explain how to make settings using the web interface as this is the easiest method.

For safety reasons, recordings can only be managed directly on the Avaya B179. All other settings can be changed via the web interface. The web interface also allows you to import and export contacts and conference groups, rename user profiles and change PIN codes. As an administrator, you can also study logs, upgrade the software and create an XML based configuration file for easier management of a set of phones.

## LOGIN

See “USING THE WEB INTERFACE” on page 6 for a description of how to log in to the web server in the Avaya B179.

## BASIC

⇒ Select **Settings > Basic**.

The screenshot shows the Avaya B179 web interface. At the top, the Avaya logo is on the left, and "You are logged in as ADMIN" with a "Logout" link is on the right. Below the logo is a navigation bar with tabs: Status, Phone book, Call list, and Settings (which is highlighted). Underneath is a sub-menu bar with tabs: Basic (highlighted), SIP, Network, Media, LDAP, Web interface, Time & Region, Provisioning, and System.

The main content area is divided into two sections: Profiles and Preferences.

**Profiles**

	Name	PIN	Edit	Set
default	DEFAULT	****	Edit	Set
Profile 1	PROFILE 1	****	Edit	Set
Profile 2	PROFILE 2	****	Edit	Set
Profile 3	PROFILE 3	****	Edit	Set
Profile 4	PROFILE 4	****	Edit	Set
Admin	ADMIN	****	Edit	Set

**Preferences**

Phone language: English (dropdown)

Ring level: Level 4 (dropdown)

Key tone:  On  Off

Recording:  On  Off

Recording tone:  On  Off

Auxiliary port:  Headset  PA (Refer to the user guide before changing to PA)

Time format:  12 Hour  24 Hour

Date format:  YYYY-MM-DD  MM/DD/YYYY  MM-DD-YYYY

Equalizer:  Soft  Neutral  Bright

Screen text: AVAYA

Buttons: Save, Cancel

# SETTINGS

These settings affect the Admin and Default profiles. To change the basic settings of a user profile, you need to log in with that profile.

## Profiles – edit name and PIN

We recommend that you change the PIN code from the default setting to protect the settings.

⇒ Select **Settings** > **Basic** and click the **Edit** button on the account you want to change.

⇒ Enter a new PIN code.

① The PIN code may consist of 8 digits.

① You can also choose to change the name of a user profile.

⇒ Click on the **Set** and **Save** buttons.

① Make a note of the new PIN code and keep it in a safe place.

① The administrator's PIN code can only be reset with a complete reset to factory settings!

## Language

⇒ Select phone language using the list box and click on the **Save** button.

On phone: **⚙** > **SETTINGS** > **BASIC** > **LANGUAGE** (6,1,1).

## Ring level

There are six volume levels plus a silent mode. You will hear the ring tone for each level you select. If you select silent mode, only the blue LEDs on the phone flash when an incoming call is received.

⇒ Select level using the list box and click on the **Save** button.

On phone: **⚙** > **SETTINGS** > **BASIC** > **RING LEVEL** (6,1,3).

## Key tone

You can select whether or not you want a tone to be heard when you press a button.

⇒ Select On or Off and click on the **Save** button.

On phone: **⚙** > **SETTINGS** > **BASIC** > **KEY TONE** (6,1,2).

## Recording

It is possible to turn off the recording feature. This setting can only be done by the administrator and affects all profiles.

⇒ Select On or Off and click on the **Save** button.

# SETTINGS

## Recording tone

A short beep is heard every 20 seconds so that all the parties in the call know it is being recorded. This feature can be turned off.

⇒ Select On or Off and click on the **Save** button.

On phone: **⚙** > **RECORDING TONE** > **SETTINGS** (5,5).

## Settings when connecting external equipment (Aux)

The Avaya B179 can be connected to a wireless headset or an external PA system. An optional PA interface box is required for PA system connection.

⇒ Select the **PA** option to activate features for external microphone mixer and PA system.

ⓘ Do not select the PA option unless a PA system is connected. This option turns off the internal microphone and internal speakers as default. The HEADSET option may be selected whether or not a headset is connected.

## Time format

⇒ Select 12 hour or 24 hour and click on the **Save** button.

On phone: **⚙** > **SETTINGS** > **BASIC** > **TIME FORMAT** (6,1,7).

## Date format

⇒ Select date format and click on the **Save** button.

On phone: **⚙** > **SETTINGS** > **BASIC** > **DATE FORMAT** (6,1,8).

## Equalizer

The sound reproduction can be adjusted to the required pitch (SOFT, NEUTRAL or BRIGHT).

⇒ Select Soft, Neutral or Bright and click on the **Save** button.

On phone: **⚙** > **SETTINGS** > **BASIC** > **EQUALIZER** (6,1,4).

## Screen text

The text on the display screen is shown when the Avaya B179 is in stand-by mode (on hook).

⇒ Enter your new text in the text box and click on the **Save** button.

On phone: **⚙** > **SETTINGS** > **BASIC** > **SCREEN TEXT** (6,1,9).

# SETTINGS

## SIP

⇒ Select **Settings** > **SIP**

The screenshot displays the Avaya SIP Settings web interface. At the top, the Avaya logo is on the left, and the user is logged in as ADMIN with a Logout link on the right. The main navigation bar includes Status, Phone book, Call list, and Settings (which is active). Below this, a secondary navigation bar lists various settings categories: Basic, SIP (selected), Network, Media, LDAP, Web interface, Time & Region, Provisioning, and System.

The main content area is divided into several sections:

- Main account:** Includes fields for Enable account (Yes/No), Account name (Conf 2), User (140), Registrar (192.168.0.1), Proxy, Realm (\*), Authentication name (140), Password (masked), and Registration interval (1800).
- Fallback account:** Similar fields to the main account, but with Registrar set to 192.168.0.2.
- NAT Traversal:** Includes STUN (On/Off), Offer ICE (Yes/No), TURN (On/Off), STUN host, TURN user, and Password.
- Advanced:** Includes Enable SIP Replaces, Enable Blind Transfer, Allow contact rewrite (all Yes/No), and Outbound proxy.
- Transport:** Includes Protocol (UDP/TCP/TLS/SIPs), Local TCP port (5060), and a note to check corresponding media signaling settings.

At the bottom left, there are Save and Cancel buttons.

The conference phone supports two accounts. The second account is a fallback account that is automatically used if the phone fails to register to the main account. Note that this account then will be used as long as the fallback account is available. If the phone fails to register to the fallback account, it will try to use the main account again.

### Main account and Fallback account

**Enable account** It is possible to store account information for future use, but temporarily disable it.

# SETTINGS

Account name	This is the name displayed on the screen. It can be set according to company standards.
User	The account (customer) name.
Registrar	Shall contain the IP address or the public name of the SIP server where the account is registered (e.g. 10.10.1.100 for a local SIP server or sip.company.net for a public VoIP service provider)
Proxy	Shall contain the proxy server used for Internet communication, if any. Can be left blank.
Realm	The protection domain where the SIP authentication (name and password) is valid. This is usually the same as the registrar. If left blank, or marked with a “*”, the phone will respond to any realm. If specified, the phone will only respond to the specific realm when asked for credentials.
Authentication name	The name used for the Realm authentication. This may be the same as the user name, but must be filled in.
Password	The password used for the Realm authentication.
Registration Interval	This is a request to the SIP server for when the registration should expire. Avaya B179 automatically renews the registration within the time interval if the phone is still on and connected to the server. The default value is 1800 seconds.

On phone:  > **SETTINGS** > **ADVANCED** > (PIN) > **ACCOUNTS** (6,2,1).

## Nat traversal

NAT (*Network Address Translation*) is a firewall or router function that operates by rewriting the IP addresses in the IP headers as packets pass from one interface to the other. When a packet, for example, is sent from the inside, the source IP address and port are rewritten from the private IP address space into the address space on the outside (Internet).

NAT rewrites the addresses but leaves the packets themselves untouched. This kind of translation works fine for many protocols, but causes a lot of trouble for SIP packets that contain address information in their content (for example an INVITE request from one IP address to another).

NAT traversal solves this problem, providing a “view from the outside” that makes it possible to replace the IP address in the SIP requests with the address shown on the other side of the firewall.

Note that in some cases NAT traversal is not necessary. Some public service providers of IP telephony keep track of the actual IP address used to register a phone, and the one used in the SIP requests from the same phone, and then replaces the addresses in the SIP messages.

# SETTINGS

STUN	<p>STUN (<i>Simple Traversal of UDP through NATs</i>) is a protocol that assists devices behind a NAT firewall or router with their packet routing. STUN is commonly used in real-time voice, video, messaging, and other interactive IP communication applications..</p> <p>The protocol allows applications operating through a NAT to discover the presence and specific type of NAT and obtain the mapped (public) IP address (NAT address) and port number that the NAT has allocated for the application's <i>User Datagram Protocol</i> (UDP) connections to remote hosts. The protocol requires assistance from a 3rd-party network server (STUN server).</p> <p>STUN should be activated if an external SIP server cannot connect to the Avaya B179 behind a firewall NAT function and the SIP server supports STUN. A suitable STUN server is usually provided by the VoIP service provider.</p> <p>Note: STUN might also be referred to as <i>Session Traversal Utilities for NAT</i>.</p>
STUN host	The IP address or public name of the STUN server.
Offer ICE	ICE ( <i>Interactive Connectivity Establishment</i> ), is a STUN addition that provides various techniques to allow SIP-based VoIP devices to successfully traverse the variety of firewalls that may exist between the devices. The protocol provides a mechanism for both endpoints to identify the most optimal path for the media traffic to follow.
TURN	TURN ( <i>Traversal Using Relay NAT</i> ) TURN is an extension of the STUN protocol that enables NAT traversal when both endpoints are behind symmetric NAT. With TURN, media traffic for the session will have to go to a relay server. Since relaying is expensive, in terms of bandwidth that must be provided by the provider, and additional delay for the media traffic, TURN is normally used as a last resort when endpoints cannot communicate directly.
TURN User	User authentication name on the TURN server.
TURN host	The IP address or public name of the TURN server.
Password	User authentication password on the TURN server.

On phone:  > **SETTINGS** > **ADVANCED** > (PIN) > **NAT TRAVERSAL** (6,2,3).

## Advanced

Enable SIP Replaces	Default is <b>Yes</b> . Setting this option to <b>No</b> , will instruct the PBX not to use the <i>SIP replace</i> header. Some PBXes try to take over the bridging functionality from Avaya B179 using this command, which causes the calls to interrupt.
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# SETTINGS

- Enable Blind Transfer** Default is **Yes**. Setting this option to **No**, will disable the transfer function (**☒ > TRANSFER**) during a call. This may be used if the PBX does not support blind transfer.
- Allow contact rewrite** Default is **Yes**. When enabled, the B179 will store the IP address from the response of the register request. If a change is detected, the phone will unregister the current sip URI (contact), and update the sip URI with the new address.

## Transport

The transport setting only concerns the protocol to be used for SIP messages between the devices involved. These settings do not include the media (the actual call). The settings on the Media tab should be set accordingly.

Note that if you choose to use a secure connection, both units must support it. Otherwise they cannot negotiate a connection. If an incoming call demands a secure TLS or SIPS connection, the Avaya B179 uses the appropriate protocol even if you have set the phone to use UDP.

### Protocol

**UDP** (*User Datagram Protocol*) is a protocol on the transport layer in the Internet Protocol Suite. It is a stateless protocol for short messages – datagrams. Stateless implies that it does not establish any connection between sender and receiver in advance. UDP does not guarantee reliability or ordering in the way that TCP does. Datagrams may arrive out of order or go missing without notice. The advantages it offers are speed and efficiency.

UDP is the default protocol for SIP.

**TCP** (*Transmission Control Protocol*) is a protocol on the transport layer in the Internet Protocol Suite. TCP is the standard protocol for Internet communication. TCP keeps track of all individual packets of data, ensuring that they reach the receiver and are put together properly. TCP is not the default protocol for SIP, because it is slower and uses more bandwidth than UDP.

With UDP and TCP, SIP packets travel in plain text. **TLS** (*Transport Layer Security*) is a cryptographic protocol that provides security and data integrity for communications over TCP/IP networks. TLS encrypts the datagrams of the transport layer protocol in use. The secure connection may be to the end device or to the first server (usually the SIP server where the phone is registered). There is no guarantee that there is a secure channel to the end point, but because the SIP server is the only part receiving the user authentication, this is still a rather secure solution.



# SETTINGS

**SIPS** (*Secure SIP*) is a security measure that uses TLS to provide an encrypted end-to-end channel for the SIP messages. To use SIPS, however, both VoIP devices and the SIP server must support it.

- ① Even if Transport is set to TLS or SIPS, the Avaya B179 still accepts incoming UDP or TCP signalling.

On phone: **⚙️ > SETTINGS > ADVANCED > (PIN) > ACCOUNTS > TRANSPORT (6,2,1,3).**

## TLS Settings

If you select TLS or SIPS under the transport setting, this additional setting appears on the page.

It may be possible to use secure communication without a certificate and make changes to these settings. In some cases, if you choose TLS or SIPS, the SIP server requires a certificate for user/client verification. This should be specified in the account information. You can further increase security by requiring verification of the server, or the client when the Avaya B179 acts as a server for incoming calls.

**Transport**

Protocol  UDP  TCP  TLS  SIPS Please check corresponding media signaling setting

Local TLS port

---

**TLS settings**

Method  Verify server  On  Off

Negotiation timeout  Verify client  On  Off

Require client certificate  On  Off

Certificate

Root certificate

Private key

Private key password

- |                     |   |
|---------------------|---|
| Method              | The TLS includes a variety of security measures. The methods are defined in the versions of the standard (SSL, SSL v2, SSL v3, TLS v1, TLS v2). The default method is SSLv23, which accepts both SSL v2 and v3.           |
| Negotiation timeout | The TLS settings are negotiated during a call setup (both incoming and outgoing). If this negotiation does not succeed within the specified time (seconds) the negotiation is aborted. Timeout is disabled with 0 (zero). |
| Verify client       | When set to On, the Avaya B179 will activate peer verification for incoming secure SIP connections (TLS or SIPS).   |

# SETTINGS

## Require client certificate

When set to On, the Avaya B179 rejects incoming secure SIP connections (TLS or SIPS) if the client does not have a valid certificate.

## Verify server

When the Avaya B179 is acting as a client (outgoing connections) using secure SIP (TLS or SIPS) it will always receive a certificate from the peer. If Verify server is set to On, the Avaya B179 closes the connection if the server certificate is not valid.

## Certificate

Here you can upload a certificate to the Avaya B179 to be used for TLS or SIPS communication.

A certificate is a file that combines a *public key* with information about the *owner* of the public key, all signed by a trusted third party. If you trust the third party, then you can be sure that the public key belongs to the person/organization named in that file. You can also be sure that everything you decrypt with that public key is encrypted by the person/organization named in the certificate.

## Root certificate

The public key in the root certificate is used to verify other certificates. A root certificate is only needed if you have selected client or server verification.

A root certificate is signed by the same public key that is in the certificate, a so-called “self-signed” certificate. A typical root certificate is one received from a *Certificate Authority*.

## Private key

Here you can upload a private key to the Avaya B179 to be used for TLS or SIPS communication.

A private key is one of the keys in a key-pair used in *asymmetric cryptography*. Messages encrypted using the public key can only be decrypted using the private key.

## Private key password

The password used for encryption of the private key, if it is encrypted.

# SETTINGS

## NETWORK

⇒ Select **Settings** > **Network**.

The screenshot shows the Avaya B179 SIP Conference Phone Settings page, specifically the Network tab. The page is titled "AVAYA" and shows the user is logged in as ADMIN. The navigation menu includes Status, Phone book, Call list, Settings, Basic, SIP, network, media, LDAP, Web interface, Time & Region, Provisioning, and System. The Network section includes fields for DHCP (On/Off), IP address (192.168.1.100), Netmask (255.255.255.0), Gateway (192.168.1.1), Hostname (AvayaB179), Domain, Primary DNS, and Secondary DNS. The Quality of Service section includes fields for SIP DiffServ (0), Media DiffServ (0), VLAN (On/Off), VLAN ID (1), VLAN map enable (On/Off), VLAN prio SIP (0 - Best Effort), and VLAN prio media (0 - Best Effort). The 802.1x section includes fields for Enable 802.1x (On/Off), EAP method (MDS/TLS), Username, Certificate, Root certificate, Private key, and TLS password. The page ends with Save and Cancel buttons.

DHCP

*Dynamic Host Configuration Protocol* is used by network devices (clients) to obtain the parameters necessary for operation in the IP network. This protocol reduces system administration workload, allowing devices to be added to the network with little or no manual configuration.

DHCP should be set to On if no other information is given. When set to On, all information on this page will be set automatically.

IP address

IP address of the device (Avaya B179). The address is provided by the network administrator or service provider if DHCP is not in use.

Hostname

Set to avaya as default. Can be changed to suitable name.

# SETTINGS


Netmask	Usually set to 255.255.255.0 to limit network traffic to the subnet.
Domain	The domain where the device is located. May be left blank.
Gateway	The device or server used for Internet communication.
Primary DNS	The address to the primary DNS ( <i>Domain Name System</i> ) server - a program or computer that maps a human-recognisable name to its computer-recognisable identifier (IP address).
Secondary DNS	The address of an optional secondary DNS server.

On phone:  > **SETTINGS** > **ADVANCED** > (PIN) > **NETWORK** (6,2,2).

## Quality of Service

*Quality of service* is used in IP networks to provide different priority to different applications, users, or to guarantee a certain level of performance to a critical data flow such as voice or video. *Differentiated Services* or *DiffServ* is a networking architecture that specifies a simple mechanism for classifying network traffic using a 6-bit field in the header of the IP packets. *VLAN (Virtual LAN)* is a technology to logically divide a physical network into several logical nets and thus to differentiate traffic.

SIP DiffServ	Enter a value between 0 and 63 to prioritize the SIP messages.
Media DiffServ	Enter a value between 0 and 63 to prioritize the media packets (voice).
VLAN	By enabling this option, all communication to and from Avaya B179 is done via the VLAN specified under VLAN ID. Note that this VLAN also must be used to communicate with Avaya B179 via the web interface.
VLAN ID	The ID number to be used for the IP telephony VLAN.
VLAN map enable	Enabling VLAN priority mapping from the DiffServ setting.
VLAN prio SIP	Set a value between 0 and 7 to prioritize the SIP messages in the VLAN.
VLAN prio media	Set a value between 0 and 7 to prioritize the media packets in the VLAN.


On phone:  > **SETTINGS** > **ADVANCED** > (PIN) > **NETWORK** > **ETHERNET** > **VLAN** (6,2,2,2,1).

# SETTINGS

## 802.1x

IEEE 802.1X is an IEEE Standard for port-based Network Access Control and is part of the IEEE 802.1 group of networking protocols. It provides an authentication mechanism to devices wishing to attach to a LAN or WLAN.

Enable 802.1x	By enabling this option, Avaya B179 asks an authentication server for permission when connected to the LAN.
EAP method	Select which EAP ( <i>Extensible Authentication Protocol</i> ) method to use: MD5 or TLS.
Username	The device identity in the network.
MD5 password	Password for the device identity when using MD5.
Certificate	Here you can upload a certificate to the Avaya B179 to be used for authentication when using TLS.
Root certificate	The public key in the root certificate is used to verify other certificates when using TLS.
Private key	Here you can upload a private key to the Avaya B179 to be used for authentication when using TLS.
TLS password	The password used for encryption of the private key when using TLS.

On phone:  > **SETTINGS** > **ADVANCED** > (PIN) > **NETWORK** > **ETHERNET** > **802.1X AUTH**  
(6,2,2,2,2).

# SETTINGS

## MEDIA

⇒ Select **Settings** > **Media**.

The screenshot displays the Avaya B179 SIP Conference Phone web interface. At the top, the Avaya logo is on the left, and the user is logged in as ADMIN with a Logout link on the right. The navigation menu includes Status, Phone book, Call list, and Settings. The Settings menu is expanded to show Basic, SIP, Network, Media, LDAP, Web interface, Time & Region, Provisioning, and Systems. The Media settings page is shown, with sections for Codec, Security, VAD, DTMF, and Advanced. The Codec section lists G722, G711 Alaw, G711 ulaw, and G729 with their respective priorities. The Security section shows SRTCP, SRTCP, and Secure signaling settings. The VAD section shows Enable VAD. The DTMF section shows DTMF Signaling. The Advanced section shows First RTP port.

The media settings determine how audio is sent between the devices. The devices negotiate via SIP before a call is connected. All devices must support the same media types, codecs and security settings.

### Codec

Codecs are used to convert an analog voice signal to a digitally encoded version and vice versa. Codecs vary in the sound quality they deliver and the bandwidth required. The Avaya B179 supports the most common codecs and each codec can be given a precedence depending on your requirements for high quality audio or low bandwidth use.

The priority can be set to from 4 (high) to 1 (low) or 0 (disabled)

#### G722

G.722 is an *ITU-T* standard codec that provides 7 kHz wideband audio at a data rate within 64 kbit/s. It offers greatly improved speech quality compared with older narrowband codecs such as G.711, but requires a high quality network connection between the devices.

# SETTINGS

- G711 Alaw**                      G.711 is an ITU-T standard codec that uses audio companding. Companding algorithms reduce the dynamic range of an audio signal. In analog systems, this can increase the signal-to-noise ratio achieved during transmission and, in the digital domain, can reduce the quantization error.
- Two main compression algorithms are defined in the standard, the  $\mu$ -law algorithm (used in North America and Japan) and A-law algorithm (used in Europe and the rest of the world).
- G711 Ulaw**                      See G711  $\mu$ -law above.
- G729**                              G.729 is an ITU-T standard codec that operates at 8 kbit/s. It is mostly used in VoIP applications with low bandwidth requirement.

On phone:  > **SETTINGS** > **ADVANCED** > (PIN) > **MEDIA** > **CODEC** (6,2,4,1).

## Security

The media in VoIP calls is usually sent using the RTP protocol (*Real-time Transport Protocol*). RTP is a standardized packet format for delivering audio and video over the Internet.

SRTP (*Secure Real-time Transport Protocol*) is an extension of RTP to provide encryption, message authentication and integrity for the audio and video streams.

All devices must support SRTP to establish a connection. It is therefore possible to set SRTP as disabled, optional or mandatory.

RTCP can be used to control the RTP session. When using SRTP, the same security-related features are added to the control protocol. SRTCP (*Secure Real Time Control Protocol*) can be either encrypted or not encrypted.

- SRTP**                              If set to disabled, the media is sent using RTP. Note that despite this setting, the Avaya B179 will still use a secure channel if the opposite device demands it.
- If set to optional or mandatory, a padlock will be shown in the bottom right-hand corner of the screen. If the other devices support SRTP, the padlock will be locked. Otherwise, an open padlock will be displayed.
- If set to mandatory, the call will not be connected if the other devices do not support SRTP.
- SRTCP**                              Is automatically used when SRTP is enabled. Can be set to use encryption or not.

# SETTINGS

**Secure signalling**      The SIP messages (signalling) and the SRTP cipher key are sent on a different channel than the media and are not affected by the RTP/SRTP setting. To ensure a secure connection, the signalling must be secured using **TLS** or **SIPS**, see page 20. Note that the SIP transport setting must be set accordingly.

On phone: **⚙** > **SETTINGS** > **ADVANCED** > (PIN) > **MEDIA** > **SECURITY** (6,2,4,4).

## VAD

*Voice Activity Detection* (speech detection) is a technique used in speech processing to detect the presence or absence of human speech in regions of audio. In VoIP applications, VAD is mainly used to avoid unnecessary coding and transmission of silence packets, saving on computation and network bandwidth.

On phone: **⚙** > **SETTINGS** > **ADVANCED** > (PIN) > **MEDIA** > **VAD** (6,2,4,2).

## DTMF

DTMF (*Dual-tone multi-frequency*) signalling is used for telephone signalling over the line to the phone switch or PBX.

If the device itself generates the tones and they are sent in the voice-frequency band, the method is called **Inband**. This is not the best method when using VoIP. Low bit rate codecs may corrupt the signalling tones and make it difficult for the switch to identify them.

**RFC 2833** is a method of carrying DTMF signals in RTP packets using a separate RTP payload format. With this method a PSTN gateway reproduces the DTMF tones sent from the end device.

With **SIP Info** the DTMF signals are sent as SIP requests. The SIP switch creates the tones if the call is transferred to the PSTN.

Use RFC 2833 or SIP Info as preferred methods. Switch to inband only if you encounter problems using DTMF signalling with your PBX/SIP switch.

On phone: **⚙** > **SETTINGS** > **ADVANCED** > (PIN) > **MEDIA** > **DTMF SIGNALLING** (6,2,4,3).

## Advanced

**First RTP port**      If the RTP packets must be directed to a specific port series, the first port number is set here.



# SETTINGS

## LDAP

⇒ Select **Settings** > **LDAP**.

The screenshot shows the Avaya B179 Settings page. At the top, there is a navigation bar with tabs for Status, Phone book, Call list, and Settings. Below this is a sub-navigation bar with tabs for Basic, SIP, Network, Media, LDAP, Web interface, Time & Region, Provisioning, and System. The main content area is titled 'LDAP configuration' and contains the following fields:

- Enable LDAP:  Yes  No
- Name filter:  Number attributes:
- Server URL:  Country code:
- Search base:  Area code:
- Username:  External prefix:
- Password:  Min length for external prefix:
- Max hits:
- Display name:
- Sort results:  Yes  No

At the bottom of the form are 'Save' and 'Cancel' buttons.

Avaya B179 has support for an external phone book, which means it can communicate with a directory server using LDAP (*Lightweight Directory Access Protocol*). The built in search function dynamically filters the content from the LDAP database, based on the search characters the user enter.

To make the LDAP phone book available, the administrator has to activate and configure the LDAP feature.

- |             |  |
|-------------|--|
| Enable LDAP | The LDAP feature is disabled by default because it has to be configured.   |
| Name filter | Defines how the entered search characters are used. The filter is designed conforming to the string representation of LDAP search filters described in <i>RFC2254</i> . The character % in the filter string will be replaced with the search character entered by the user.<br>Example:<br><b>{{(sn=%*)(cn=%*)}}</b> - All entries with the search characters in the beginning of the <b>sn</b> OR <b>cn</b> attribute are presented to the user. |
| Server URL  | The IP address of the LDAP server host. Supports <i>ldap</i> and <i>ldaps</i> .  |
| Search base | The DN ( <i>distinguished name</i> ) of the search base<br>Example:<br><b>dc=domain, dc=com</b> .  |
| Username    | Leave this field blank if the LDAP server does not require a username.   |

# SETTINGS

Password	Leave this field blank if the LDAP server does not require username and password.
Max hits	The maximum number of hits to return for each LDAP search.
Display name	Specifies how the search hits shall be presented on the display in Avaya B179. Example: <b>%cn</b> - shows the <b>cn</b> attribute. <b>%givenName %sn</b> - shows the <b>givenName</b> attribute and the <b>sn</b> attribute with a space in between.
Sort results	Sorts the search hits based on the Display name.
Number attributes	Here you define the attributes that shall be displayed for a selected search hit. Example: <b>mobile telephoneNumber</b> - shows the mobile phone number and office phone number on separate rows for the selected Display name. (Refer to the LDAP administrator for the actual names of the fields in the LDAP database.)
Country code	By entering the country code where the phone is located, the country code in any phone number attribute is ignored, if it is identical.
Area code	By entering the area code where the phone is located, the area code in any phone number attribute is ignored, if it is identical.
External prefix	If a special prefix is needed to dial external numbers, it should be added here. Use this if you for example need to dial 0 to get a dialing tone.
Min length for external prefix	Restricts the external prefix to be added only if the phone number is longer than the min length. This makes it possible to use short internal numbers.
Exact length for no external prefix	The external prefix is not added if the phone number has exactly the entered length.
Number prefix for no external prefix	All numbers that starts with this number will not have the external prefix added. Useful if you know that all internal numbers start with a certain number.

# SETTINGS

## WEB INTERFACE

⇒ Select **Settings** > **Web interface**.



The web server in the Avaya B179 supports secure connections using HTTPS.

**Enable HTTPS**            Set *Enable HTTPS* to On if you need a secure communication between the PC used for setup and the phone.

**Certificate**            To use HTTPS you need to upload a certificate to the phone.

On phone: ⚙ > **SETTINGS** > **ADVANCED** > (PIN) > **WEB INTERFACE** (6,2,7).

# SETTINGS

## TIME & REGION

⇒ Select **Settings > Time & Region**.

The screenshot shows the Avaya web interface for the 'Time & Region' settings. The page is titled 'AVAYA' and shows the user is logged in as 'ADMIN'. The navigation menu includes 'Status', 'Phone book', 'Call list', 'Settings', 'Basic', 'SIP', 'Network', 'Media', 'LDAP', 'Web interface', 'Time & Region', 'Provisioning', and 'System'. The 'Time & Region' section is active and contains the following fields:

- Time**
  - Enable NTP:  On  Off
  - Time:  Date:
  - Timezone:
  - NTP Server:
- Region**
  - Region:
- Daylight saving**
  - Enable DST:  Yes  No
  - DST Timezone (hh:mm):
  - DST Mode:  Automatic  Manual
  - Start fixed date:  Yes  No
    - Start month:  Start week:
    - Start day of week:  Start time (HH:mm):
  - Stop fixed date:  Yes  No
    - Stop month:  Stop week:
    - Stop day of week:  Stop time (HH:mm):

At the bottom of the form are 'Save' and 'Cancel' buttons.

### Time

Enable NTP

NTP (*Network Time Protocol*) is a protocol for distributing the *Coordinated Universal Time* (UTC) by means of synchronizing the clocks of computer systems over packet-switched, variable-latency data networks.

Time

This field shows the actual time if NTP is enabled. Otherwise enter the correct time (hh:mm:ss) and save the setting.


Date

This field shows the actual date if NTP is enabled. Otherwise enter the correct date (yyyy-mm-dd) and save the setting.

Timezone

Select the UTC time zone in your country.

# SETTINGS

- Daylight saving** Select the Yes radio button if DST (*Daylight Saving Time* or *Summer Time*) is currently used in your country. Note that this setting only adjusts the time by one hour and does not change the time automatically when the DST starts and ends.
- NTP Server** The *NTP pool* is a dynamic collection of networked computers that volunteer to provide highly accurate time via NTP to clients world-wide. These computers are part of the pool.ntp.org domain and part of several subdomains divided by geographical zones. They are distributed to NTP clients via round robin DNS.
- On phone:  > **SETTINGS** > **ADVANCED** > (PIN) > **TIME** (6,2,5).

## Region

Select the region where you are. This setting determines the signalling (disconnect tone, busy tone, etc).

On phone:  > **SETTINGS** > **ADVANCED** > (PIN) > **REGION** (6,2,6).

## Daylight saving

- Enable DST** Select the Yes radio button if DST (*Daylight Saving Time* or *Summer Time*) is used in your country.
- DST Timezone** Select the offset from UTC time when daylight saving is in use.
- DST Mode** When set to Automatic, OmniTouch 4135 IP uses dates stored in the phone to adjust for DST. When set to Manual, you need to manually set the offset two times a year.
- Start/Stop fixed date** Set to Yes if DST changes the same date every year in your country. Then select the time and date it changes.
- Set to No if DST changes a specific week and day each year. (For instance third sunday in March.) Then select the month, week and time it changes.

## PROVISIONING

See “PROVISIONING – UPGRADE AND CONFIGURATION” on page 40.

# SETTINGS

## SYSTEM

⇒ Select **Settings > System**.



### Application restart

The **Restart** button restarts the phone application. This takes less than 30 seconds.

On phone: **⚙️ > SYSTEM > RESTART (7,1)**.

### System reboot

The **Reboot** button reboots the conference phone. The starting procedure may take about two minutes.

On phone: **⚙️ > SYSTEM > REBOOT (7,2)**.

### Factory reset

The **Reset** button resets the Avaya B179 to factory default settings. All personal settings, including account information, are erased.

On phone: **⚙️ > SYSTEM > FACTORY RESET (7,3)**.

### Hard reset to factory settings

See page 39 about resetting the phone if you have forgotten the Admin PIN code.

# HEADSET AND PA INSTALLATION AND SETTINGS

## CONNECTING A WIRELESS HEADSET

⇒ Connect the headset to the Aux port on Avaya B179.

The microphones from the Avaya B179 and the wireless headset will work simultaneously and transmit the call to other participants in the phone conference.

Please refer to the headset manual for further information.

## Turning off the internal speakers when using a headset

The internal speakers can be turned off temporarily if you wish to use the Avaya B179 as a personal telephone with a headset.

⇒ During a call, select **⚙** > **HEADSET**.

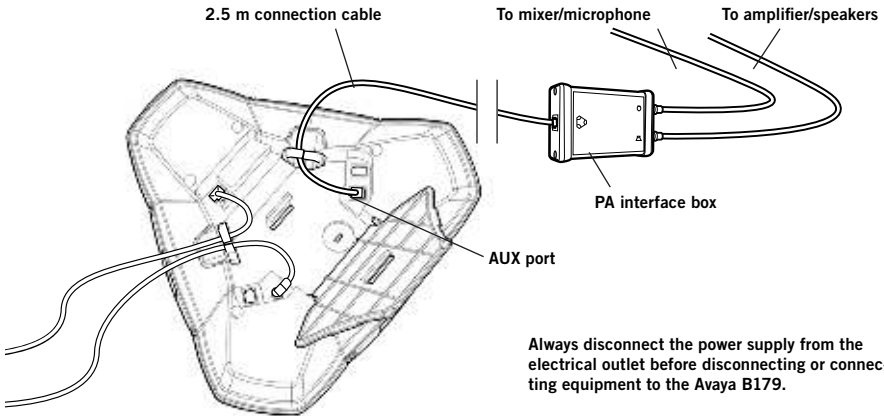
⇒ Select YES when asked “SPEAKER OFF?”.

① The speakers come on automatically when the call is ended.

# HEADSET AND PA INSTALLATION AND SETTINGS

## CONNECTING A PA INTERFACE BOX

The Avaya B179 can be connected to an external PA system using a PA interface box.



- ⇒ Connect the PA-box to the AUX port on Avaya B179 with the included cable.
- ⇒ Connect the external amplifier to the RCA connector marked with a speaker.
- ⇒ Connect the microphone mixer to the RCA connector marked with a microphone.

## Changing the auxiliary port setting

- ⇒ Select **Settings > Basic**.
- ⇒ Select **PA** under the heading **Auxiliary port** to activate the functions for external microphones and speaker system.



- ⇒ Click on **Save**.
- ⓘ Do not select the PA option unless a PA system is connected. This option turns off the internal microphone and internal speakers as default.

On phone: **⚙ > SETTINGS > BASIC > AUX PORT (6,1,5)**.



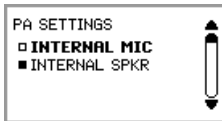
# HEADSET AND PA INSTALLATION AND SETTINGS

## PA SETTINGS

To match several types of situations and equipment, there are some settings available in the Avaya B179 menu.

### Activating internal microphone and speakers

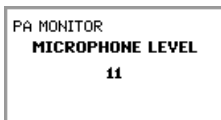
- ① These settings are not available via the web interface.
- ⇒ Select **⚙️ > SETTINGS > BASIC > PA (6,1,6)**.



- ⇒ Select INTERNAL MIC and press **OK** to switch between on (shaded box) and off.
- ① To ensure maximum audio quality, do not use the internal microphone and external microphones connected via the PA box at the same time.
- ① Only the internal microphone is turned off. Any external microphones connected to the Avaya B179 are still turned on.
- ⇒ Select INTERNAL SPKR and press **OK** to switch between on (shaded box) and off.
- ① To ensure maximum audio quality, do not use the internal speakers and external speakers connected via the PA box at the same time.

### Adjusting microphone volume from PA

- ⇒ During a call, select **⚙️ > PA > PA MONITOR**.



- ⇒ Adjust the microphone volume from the mixer so that the level on the display screen is around 10–12 when speaking in a normal tone.

### Adjusting PA calibration manually

It is possible to calibrate the duplex performance of the conference phone when it is connected to a PA system. The calibration level can be set automatically by the Avaya B179 or adjusted manually to any value between 0 and 5 (0 being full duplex).

- Increase the calibration if the other party experiences disturbing echo.
- Decrease the calibration if the other party experiences low duplex, i.e. your voice is muted or clipped when the other party is speaking.

# HEADSET AND PA INSTALLATION AND SETTINGS

① The position of the PA system's microphones and speakers and the amplifier's settings may affect full duplex performance.

⇒ Select **⚙️ > PA > CALIBRATION.**



① AUTO is the default setting and is recommended in most cases. The figure shown in brackets is the measured calibration value.

⇒ Select different levels and compare the audio quality to achieve your preferred setting.

① NB. You must ask the person you are calling to assess the effect of the adjustments you make.

# HARD SYSTEM RECOVERY

## Reset configuration

If you have forgotten the Admin PIN code, the only way to reset it to default is to do a hard factory reset. This is the same as the Factory reset in the system menu (**⚙ > SYSTEM > FACTORY RESET**).

- ① This erases all settings including account information and contacts!
- ⇒ Disconnect the power supply cable. Note that this is the same as the network cable if the phone uses Power over Ethernet.
- ⇒ Press and hold the **⚙** button while you connect the cable again (i.e. starts the Avaya B179). Hold the button pressed until the SYSTEM RECOVERY menu is shown on the display.
- ① You can press any other button than **1**, **2**, or **3** to start the phone without resetting.
- ⇒ Press **1** to select **Reset configuration** and confirm with **OK**.
- ⇒ Upgrade to the latest version of the software when the phone has started and redo the setup of account and other settings (see page 7).

## Restore firmware

This replaces the current software with the one supplied with the phone. All settings are erased.

- ⇒ Export a configuration file (page 47) if local settings should be saved and export the contacts (page 53) if these are stored in the unit.
- ⇒ Disconnect the power supply cable. Note that this is the same as the network cable if the phone uses Power over Ethernet.
- ⇒ Press and hold the **⚙** button while you connect the cable again (i.e. starts the Avaya B179). Hold the button pressed until the SYSTEM RECOVERY menu is shown on the display.
- ① You can press any other button than **1**, **2**, or **3** to start the phone without restoring the firmware.
- ⇒ Press **3** to select **Restore firmware** and confirm with **OK**.  
*All content in the phone's memory is erased and the firmware supplied with the phone is written to the memory.*
- ⇒ Upgrade to the preferred version of the firmware when the phone has started.
- ⇒ Import the local configuration (page 47) and previous contacts (page 54) or do a manual account setup (page 7).

# PROVISIONING – UPGRADE AND CONFIGURATION

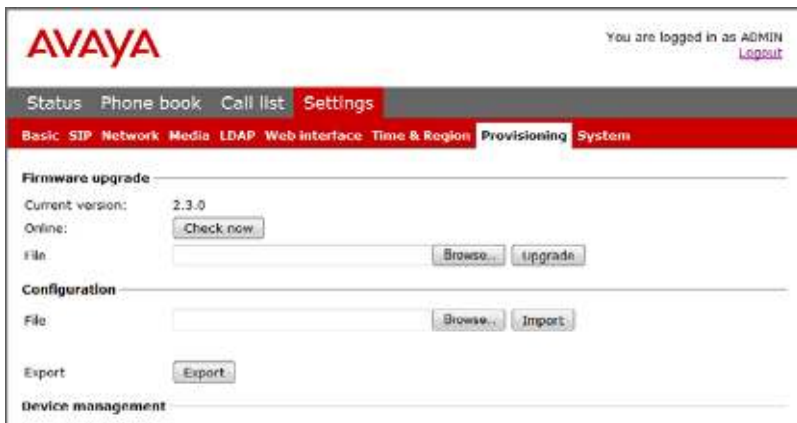
## FIRMWARE UPGRADE ON A SINGLE PHONE

The easiest way to upgrade the Avaya B179 is via a computer connected to the same network. Via the web interface, you can check for a more recent version and then automatically install it.

It is also possible to download the latest version, via the Avaya website ([support.avaya.com](http://support.avaya.com)), and then install the file via the web interface or using a SD card.

### Using the web interface

⇒ Select **Settings > Provisioning**.



⇒ Click on the **Check Now** button.

⇒ Compare the latest software version with the current version (shown on the same page).



⇒ If you choose to upgrade, select version in the list box and click on the **Upgrade** button.  
*The browser window and the Avaya B179 display shows that the upgrading has begun.*

ⓘ The download and installation can take several minutes. Do not interrupt the upgrade and do not disconnect plugs to the Avaya B179 during the upgrade. Interrupting the upgrade may render the conference phone inoperable.

# PROVISIONING – UPGRADE AND CONFIGURATION

When installation is complete, the text “Upgrade Complete. The unit will be rebooted.” is shown in your browser, and after a while you hear the Avaya music signature, which indicates that the conference phone has started.

## Upgrading from downloaded file

It is possible to download a firmware file from support.avaya.com and install it on the Avaya B179 from the local hard disk.

- ⇒ Download the firmware file from **support.avaya.com**.
- ⇒ Click on the **Browse...** button and locate and select the downloaded file.
- ⇒ Click on the **Upgrade** button.

## Upgrading from SD card

Upgrading from SD card may be suitable if you have many phones to upgrade. The phones do not have to be connected to the network.

- ⇒ Download the latest firmware as above and save it on a SD card.
- ⇒ Put the SD card in the phone you want to upgrade.
- ⇒ Disconnect the power supply cable. Note that this is the same as the network cable if the phone uses Power over Ethernet.
- ⇒ Press and hold the **⚙** button while you connect the cable again (i.e. starts the Avaya B179). Hold the button pressed until the SYSTEM RECOVERY menu is shown on the display.
- ① You can press any other button than **1**, **2**, or **3** to start the phone without any change.
- ⇒ Press **2** to select **SD-card upgrade**.

*The Avaya B179 is upgraded with the firmware file on the SD card and starts when the upgrade is done.*

## After upgrading

If DHCP is used in the network, the IP address may have been changed. If the web browser loses contact with Avaya B179, check the IP address on the conference phone (see “USING THE WEB INTERFACE” on page 6).

# PROVISIONING – UPGRADE AND CONFIGURATION

## USING A CONFIGURATION FILE

It is possible to save a configuration xml file to be used as:

- Backup (i.e. if the system has been reset to factory default)
- Configuration interface (there are some settings that are not configurable via the web interface)
- Management tool (export, edit and import settings to a set of phones instead of doing the settings on each phone)
- Use with a Device Management server, see page 48.

The structure of the xml file is as follows:

### <locale>

<region>

### <recording>

<enable>

### <logging>

<level>

The phone application logs messages to log facility LOCAL0. Log level 1–5 (equivalent to Fatal–Trace)

<log\_sip>

Log SIP messages to log facility LOCAL1. Default is true.

<remote\_log>

Log messages to a remote log server. Default is false.

<remote\_host />

Remote log server.

### <network>

<net>

<dhcp>

Specify if DHCP should be used to obtain network settings. If so, the other network settings won't be used.

<ip>

Specify the IP address of the Avaya B179.

<netmask>

The netmask of the IP address.

<gateway>

Specify the default gateway to be used.

<dns1>

Specify at most two Domain Name Servers to be used.

<dns2>

<hostname>

Specify host name.

<domain />

Specify domain name.

<vlan>

<enable>

Virtual LAN enabled if set to true

<id>

VLAN ID.

<std\_prio\_map>

<sip\_priority>

<media\_priority>

<ether\_8021x>

<enable>

# PROVISIONING – UPGRADE AND CONFIGURATION

```

    <username />
    <eap_md5>
        <enable>
        <password>
    <eap_tls>
        <enable>
        <password>
<qos>
    <dscp_sip>
    <dscp_media>
<time>
    <ntp>
    <timezone>
    <daylight_save>
    <ntps>

```

## <sip>

<code>&lt;udp_transport&gt;</code>	Specify if UDP shall be used as transport.
<code>&lt;udp_port&gt;</code>	Specify the UDP port to listen to.
<code>&lt;tcp_transport&gt;</code>	Specify if TCP shall be used as transport.
<code>&lt;tcp_port&gt;</code>	Specify the TCP port to listen to.
<code>&lt;tls_transport&gt;</code>	Specify if TLS shall be used as transport.
<code>&lt;sips_transport&gt;</code>	Specify if SIPS shall be used as transport.
<code>&lt;tls_port&gt;</code>	Specify the TLS port to listen to.
<code>&lt;rtp_port&gt;</code>	Specify the start port for RTP traffic.
<code>&lt;outbound_proxy /&gt;</code>	Specify the URL of outbound proxies to visit for all outgoing requests. The outbound proxies will be used for all accounts and will be used to build the route set for outgoing requests. The final route set for outgoing requests will consist of the outbound proxies and the proxy configured in the account.
<code>&lt;use_stun&gt;</code>	Use Simple Traversal of UDP through NATs (STUN) for NAT traversal. Default is no.
<code>&lt;stun_domain /&gt;</code>	Specify domain name to be resolved with DNS SRV resolution to get the address of the STUN servers. Alternatively application may specify <code>stun_host</code> and <code>stun_relay_host</code> instead.
<code>&lt;stun_host /&gt;</code>	Specify STUN server to be used in "HOST[:PORT]" format. If port is not specified, default port 3478 will be used.
<code>&lt;use_turn&gt;</code>	Use Traversal Using Relay NAT (TURN) for NAT traversal. Default is no.
<code>&lt;turn_host /&gt;</code>	Specify TURN relay server to be used.
<code>&lt;turn_tcp&gt;</code>	Use TCP connection to TURN server. Default is false.
<code>&lt;turn_user /&gt;</code>	TURN username.
<code>&lt;turn_passwd /&gt;</code>	TURN password.
<code>&lt;nat_type_in_sdp&gt;</code>	Support for adding and parsing NAT type in the SDP to assist troubleshooting. The valid values are: 0: no information will be added in SDP and parsing is disabled

# PROVISIONING – UPGRADE AND CONFIGURATION

	1: only the NAT type number is added 2: add both NAT type number and name
<require_100rel>	Specify whether support for reliable provisional response (100rel and PRACK) should be required by default. Note that this setting can be further customized in account configuration.
<use_srtp>	Specify default value of secure media transport usage. Note that this setting can be further customized in account configuration. 0: SRTP will be disabled, and the transport will reject RTP/SAVP offer. 1: SRTP will be advertised as optional and incoming SRTP offer will be accepted. 2: The transport will require that RTP/SAVP media shall be used.
<srtp_secure_signaling>	Specify whether SRTP requires secure signalling. This option is only used when use_srtp option above is non-zero. Note that this setting can be further customized in account configuration. 0: SRTP does not require secure signalling 1: SRTP requires secure transport such as TLS 2: SRTP requires secure end-to-end transport (SIPS)
<codec>	
<type>	Codec type
<name>	Codec name
<prio>	Codec priority (0-4)
<dtmf>	DTMF signalling. Default is 2. 0: In-band 1: SIP message 2: RTP message
<no_vad>	Disable VAD. Default is VAD enabled.
<ec_tail>	Echo canceller tail length, in milliseconds.
<enable_ice>	Enable ICE?
<enable_relay>	Enable ICE relay?
<enable_presence>	Enable the use of presence signalling.
<enable_sip_replaces>	
<enable_blind_transfer>	
<allow_contact_rewrite>	
<b>&lt;tls&gt;</b>	
<tls_password />	Password for the private key
<tls_method>	TLS protocol method from pjsip_ssl_method, which can be: 0: Default (SSLv23) 1: TLSv1 2: SSLv2 3: SSLv3 23: SSLv23
<tls_verify_server>	Verify server certificate.
<tls_verify_client>	Verify client certificate.
<tls_require_client_cert>	Require client certificate.
<tls_neg_timeout>	TLS negotiation timeout in seconds to be applied for both outgoing and incoming connections. If zero, no timeout is used.



# PROVISIONING – UPGRADE AND CONFIGURATION

## <account1>

<valid>	If this account information is valid or not.
<name>	User defined name of the account
<id>	The full SIP URL for the account.
<registrar>	This is the URL to be put in the request URI for the registration.
<publish_enabled>	If this flag is set, the presence information of this account will be published to the server where the account belongs.
<initial_auth>	If this flag is set, the authentication client framework will send an empty Authorization header in each initial request.
<initial_algo />	Specify the algorithm to use when empty Authorization header is to be sent for each initial request (see above).
<pidf_tuple_id />	Optional PIDF tuple ID for outgoing PUBLISH and NOTIFY. If this value is not specified, a random string will be used.
<force_contact />	Optional URI to be put as Contact for this account. It is recommended that this field is left empty, so that the value will be calculated automatically based on the transport address.
<require_100rel>	Specify whether support for reliable provisional response (100rel and PRACK) should be required for all sessions of this account.
<proxy_uri />	Optional URI of the proxies to be visited for all outgoing requests that are using this account (REGISTER, INVITE, etc).
<reg_timeout>	Optional interval for registration, in seconds. If the value is zero, default interval will be used.
<cred>	Array of credentials. Normally, if registration is required, at least one credential should be specified to successfully authenticate the service provider. More credentials can be specified, for example when it is expected that requests will be challenged by the proxies in the route set.
<realm>	Realm. Use "*" to make a credential that can be used to authenticate any challenges.
<scheme />	Scheme (e.g. "digest").
<username>	Authentication name
<cred_data_type>	Type of data (0 for plaintext password).
<cred_data>	The data, which can be a plaintext password or a hashed digest.
<auto_update_nat>	This option is useful for keeping the UDP transport address up to date with the NAT public mapped address. When this option is enabled and STUN is configured, the library will keep track of the public IP address from the response of REGISTER request. Once it detects that the address has changed, it will unregister current Contact, update the UDP transport address and register a new Contact to the registrar.
<ka_interval>	Set the interval for periodic keep-alive transmission for this account. If this value is zero, keep-alive will be disabled for this account. The keep-alive transmission will be sent to the registrar's address after successful registration.
<ka_data />	Specify the data to be transmitted as keep-alive packets. Default: CR-LF.
<use_srtp>	Specify whether secure media transport should be used for this account. 0: SRTP will be disabled and the transport will reject RTP/SAVP offer. 1: SRTP will be advertised as optional and incoming SRTP offer will be accepted.

# PROVISIONING – UPGRADE AND CONFIGURATION

- 2: The transport will require that RTP/SAVP media is used.
- <use\_srtp> Specifies whether encryption or no encryption is used for the control protocol. Valid values: true or false.
- <srtp\_secure\_signaling> Specify whether SRTP requires secure signalling. This option is only used when use\_srtp option above is non-zero.
- 0: SRTP does not require secure signalling  
 1: SRTP requires secure transport such as TLS  
 2: SRTP requires secure end-to-end transport (SIPS)

## <account2>

Same as above for account 2

## <provisioning>

- <upgrade>
- <url> Place to find software upgrades. The supported URL types are: HTTP, FTP, and TFTP.
- <dev\_mgnt>
- <enable> Device management enabled, true or false.
- <use\_dhcp\_option> Use DHCP option for DM server address.
- <dhcp\_option> Specification of which DHCP option to use.
- <file\_server\_address> DM server address if not provided by DHCP option.
- <pagename /> Base name of configuration files to download
- <type /> Configuration file type specification
- <update\_interval> Timing for downloading files. Shall be entered in crontab format: \* \* \* \* \* where the \* stands for minute (0–59), hour (0–23), day of month (1–30), month (1–12), day of week (0–7) (Sunday = 0 or 7)  
 Example:  
 0 6 \* \* \* = the files are downloaded daily at 6:00.
- <https\_check\_srv\_cert> Controls server certificate, true or false.
- <https\_protocol> Possibility to set https protocol if open-ssl auto detection fails.

## <www>

- <enable\_https> Secure communication to the Avaya B179 web server. Default is false.

## <pa>

- <enable\_pa> PA enabled, true or false
- <enable\_internal\_mic> Internal mic enabled when PA set to true.
- <enable\_internal\_sprk> Internal speakers enabled when PA set to true.
- <calibration> Calibration value. Note that 0 is auto, 1 is calibration value 1, 2 is calibration value 1, etc.

## <ldap>

- <enable> LDAP enabled, true or false.
- <name\_filter> Name filter according to RFC2254
- <server\_url> LDAP server address
- <search\_base> The DN (distinguished name) of the search base
- <username />

# PROVISIONING – UPGRADE AND CONFIGURATION

```
<password />
<max_hits>
<country_code>
<area_code>
<external_prefix />
<min_length_for_ext_prefix />
<exact_length_for_no_ext_prefix />
<number_prefix_for_no_ext_prefix />
<number_attributes>
<display_name>
<sort_results>
```

## Export configuration

- ⇒ Select **Settings > Provisioning**.
- ⇒ Click on the **Export** button under **Configuration**.  
*The configuration file is shown in the web browser.*
- ⇒ Choose to save the page as an xml file.  
*The xml file is as default saved in your folder for downloaded files.*
- ⇒ If necessary, edit the xml file in a suitable editor.

## Import configuration

- ⇒ Click on the **Browse...** button under **Configuration**.
- ⇒ Select the xml file and choose to open it.
- ⇒ Click on the **Import** button.

# PROVISIONING – UPGRADE AND CONFIGURATION


## USING A DEVICE MANAGEMENT SERVER

Using *Device management* facilitates the upgrading and configuration of multiple conference phones. To use this feature, the Device management needs to be enabled (default) and configured and the appropriate files must be located on a server reachable from all phones, here called a *device management server*.

The configuration and firmware download are controlled with a configurable frequency. The default value is once every 30 minutes. (Note: The interval can only be edited directly in the configuration file.)

### Configuration priorities

Because the same configuration parameters can be entered in multiple locations, there is a need for priorities. The local configuration files have the highest priority followed by the global configuration file. Configuration entered on the unit itself, via the web interface or directly on the phone, is overridden the next time the configuration files are downloaded.

 Note one exception. Phone language entered on the unit will take precedence.

### Files on the device management server

#### Global configuration file

The global configuration file contains the basic configuration – all settings that are common for all conference phones on your location. The easiest way to create this file is simply to configure one phone and export the configuration file or use the built in configuration file creator.

The default name for this file is *avaya.xml*, but it is possible to create a custom name by using the *pagename* element in the configuration file. It is also possible to refer to a cgi, php, asp, js or jsp file, instead of the xml file, if this is declared using the *type* element in the configuration file.

Avaya B179 searches for configuration files in the following order:

"	Type parameter value	Result
1	<nothing>	<pagename>.xml
2	cgi	<pagename>.cgi?phone_model=avaya_b179>
3	%%	<pagename>.php?phone_model=avaya_b179>
4	asp	<pagename>.asp?phone_model=avaya_b179>
5	E	<pagename>.js?phone_model=avaya_b179>
6	E %	<pagename>.jsp?phone_model=avaya_b179>
:	<any name>	<pagename>.<any name>? phone_model=avaya_b179
=	auto	1, 2, 3, 4, 5 and 6 will be tried in that order

# PROVISIONING – UPGRADE AND CONFIGURATION

## Local configuration file

The local configuration file contains configuration parameters that are unique for every conference phone. The settings in this file takes precedence over the settings in the global configuration file.

The default name for this file is *avaya-<MAC>.xml*, where <MAC> is the MAC address of the specific conference phone. The MAC address should be written without colons.

It is possible to create a custom name by using the *pagename* element in the configuration file. It is also possible to refer to a cgi, php, asp, js or jsp file, instead of the xml file, if this is declared using the *type* element in the configuration file.

Avaya B179 searches for configuration files in the following order:

"	Type parameter value	Result
1	<nothing>	<pagename>-<MAC>.xml
2	cgi	<pagename>.cgi? phone_model=avaya_b179&eth=<MAC>
3	%""%	<pagename>.php? phone_model=avaya_b179&eth=<MAC>
4	asp	<pagename>.asp? phone_model=avaya_b179&eth=<MAC>
5	E	<pagename>.js? phone_model=avaya_b179&eth=<MAC>
6	E %	<pagename>.jsp? phone_model=avaya_b179&eth=<MAC>
:	<any name>	<pagename>.<any name>? phone_model=avaya_b179&eth=<MAC>
=	auto	1, 2, 3, 4, 5 and 6 will be tried in that order

## Firmware binary

Contains the firmware binary that will be downloaded and installed by Avaya B179 if the metadata file shows that this is a newer version than the present installed. The binary file can be downloaded from [support.avaya.com](http://support.avaya.com).

## Firmware metadata file

A metadata file in xml format with information of the firmware version in the binary file. The file is used to check if the binary file should be downloaded to the phone or not.

The name of this file shall be *avaya\_fw\_version.xml*. The file shall contain the following elements in xml format.

```
<firmware_version>
<version>X.X.X </version>   Eg. 2.3.0
<filename>xxxx </filename>  Eg. AVAYA_B179_v2.3.0.kt
<checksum>XXXX </checksum> MD5 checksum of the firmware binary
</firmware_version>
```

# PROVISIONING – UPGRADE AND CONFIGURATION

## Device management configuration in Avaya B179

⇒ Select **Settings > Provisioning**.

The screenshot displays the Avaya B179 provisioning web interface. At the top, the Avaya logo is on the left, and the user is logged in as ADMIN with a Logout link. The navigation menu includes Status, Phone book, Call list, Settings, Basic, SIP, Network, Media, LDAP, Web interface, Time & Region, Provisioning, and System. The 'Provisioning' section is active, showing 'Firmware upgrade' (Current version: 2.3.0, Check now button) and 'Configuration' (File, Browse, Import buttons). The 'Device management' section is expanded, showing 'Enable' (On/Off radio buttons), 'Use DHCP option' (On/Off radio buttons), 'DHCP option' (dropdown menu set to 67: Bootfile-name), 'File server address' (text field with http://10.10.1.30/config), 'HTTPS protocol' (dropdown menu set to auto), 'Check server cert.' (On/Off radio buttons), 'Certificate', 'Root certificate', and 'Private key' (each with a Browse... button), and a 'Save' button.

### Device management

Enable	On enables Device management.
Use DHCP option	Set to on if you want to use DHCP option for DM server address.
DHCP option	Select the DHCP option used for the DM server address. 242: Avaya specific option (default) 43: Vendor specific 56: DHCP message 60: Class Id 61: Client Id 66: Server-name 67: Bootfile-name
File server address	DM server address if not provided by DHCP option.
HTTPS protocol	Default is auto, but can be set to SSLv2 or SSLv3 if open-ssl auto detection fails.

# PROVISIONING – UPGRADE AND CONFIGURATION

Check server cert.	Enable authentication with certificate.
Certificate	Here you can upload a certificate to the Avaya B179 to be used for authentication when using Device management.
Root certificate	The public key in the root certificate is used to verify other certificates when using Device management.
Private key	Here you can upload a private key to the Avaya B179 to be used for authentication when using Device management.

① Read more about certificates on page 22.

## Setting up a Device management server

This is a description of a manual method to create the configuration files.

- ⇒ Select **Settings > Provisioning**.
- ⇒ Enable Device management and enter the server information.

### Creating a global configuration file

- ⇒ Configure one phone with the basic configuration.
- ⇒ Click on **Export** to create a configuration file.
- ⇒ If necessary, edit the xml file in a suitable editor.
- ① Some parameters can't be entered via the web interface (update frequency, page-name, and filetype).
- ① To avoid confusion, it may be wise to delete the local information from the file (eg. account information).
- ⇒ Save the file with the name **avaya.xml** on the *File server address* specified above.

### Creating local configuration files

- ⇒ Save a copy of the configuration file for each conference phone on your location with content only in the elements that shall be unique for each conference phone (eg. account information).

The default name for each file is *avaya-<MAC>.xml*, where <MAC> is the MAC address of the specific conference phone.

- ⇒ Place the configuration files on the *File server address* specified above.

### Firmware binary

- ⇒ Place the firmware binary file on the Provisioning server.
- ⇒ Create a Firmware metadata file according to page 49 and place it on the *File server address* specified above.
- ① Depending on the server used, and the security settings, there might be necessary to add the file type *.kt* to the MIME settings on the server. This is easily checked by trying to download the kt file from a web browser.

# PROVISIONING – UPGRADE AND CONFIGURATION

## UPGRADING IN IP OFFICE

Avaya B179 supports the IP Office check-sync message with file id 4 for firmware upgrades. Please see the IP Office documentation for more information.

## HOW TO DO A DOWNGRADE

The only way to “downgrade” – install a previous version of the firmware - is to restore the firmware to factory default (page 39) and then install the preferred firmware.



# MANAGING PHONE BOOK AND CONF. GUIDE


## IMPORTING AND EXPORTING CONTACTS

You can import contacts from a comma separated values (CSV) file. One way of creating a CSV file is using Microsoft Excel and saving the file in CSV format.

Enter the names of the contacts in the first column and their phone numbers or URIs in the second. Do not use hyphens or spaces in the number. Note that Excel ignores zeros at the beginning of numbers. The cells must therefore be formatted as text.

	A	B	C	D	E	F
1	Name	Telephone				
2	Allen, Jerry	+461007954884				
3	Andersson, Justin	+461007954955				
4	Andrews, Fanny	+461007954883				
5	Berg, David	+461007954893				
6	Berlin office	+49116603087451				
7	Bewers, Darren	+461007954884				
8	Bjork, Markus	+461007954949				
9	Branshaw, Liw	+461007954871				
10	Carling, Richard	+461007954868				
11	Carlsson, Julia	+461007954884				
12	Claesson, Nicole	+461007954886				
13	Collins, David	+46111599581				
14	Cordin, Justin	+461007954898				
15	Crown, Juanito	+461007954896				
16	Evalders, Julie	+461007954881				
17	Gardellus, Stefan	+461007954950				
18	Hellberg, Mark	+461007954884				
19	Konrads, Ray	+461007954870				
20	Langdon, Steve	+461007954890				
21	Leander, Adam	+461007954879				
22	Lowendahl, Roger	+461007954885				
23	Luong, Xi	+461007954878				
24	Magret, Robin	+461007954895				
25	Mowat, Leo	+461007954872				
26	Mowji, Ali	+461007954866				
27	Nelson, Mike	+461007954880				
28	Nyberg, Paul	+461007954867				

 It is normally possible to export contact books stored in your PC in CSV format.

 The way the number can be written may depend on the SIP PBX being used, but normally you can use:

- Complete phone number, including country code
- Phone number, including area code
- Local phone number only
- Internal speed dial number (with company's own PBX)
- URI, e.g. **sip:user@company.com**
- URI with IP address, e.g. **sip:10.10.1.100** (within a local network)

# MANAGING PHONE BOOK AND CONF. GUIDE

## Importing contacts

- ⇒ Select **Phone Book**.
- ⇒ Click on the **Scroll...** button under the heading Import in the web window.
- ⇒ Open your CSV file.
- ⇒ Click on **Import**.
- ① The name is limited to 15 characters, since the Avaya B179 screen cannot display more than 15 characters.

## Exporting contacts

You can export your contacts as a CSV document in order to import them into another phone.

- ⇒ Click on **Export**.
- ⇒ Save the document.

## IMPORTING AND EXPORTING CONFERENCE GROUPS

- ① The Conference group feature requires that your system allows multiple call appearances. Avaya Aura Communication Manager supports this while Avaya IP Office and Communication Server 1000 do not.

The conference groups can be imported and exported in the same way as the contacts in the phone book, but use a three column csv instead of a two column csv.

	A	B	C	D	E	F	G
1	Group	Name	Number				
2	Sales	Carlsson, Julia	+4616017954884				
3	Sales	Berg, David	+4616017954893				
4	Sales	Berlin office	+4966023687451				
5	Sales	UK office	+4416057953687				
6	Development	Bjork, Markus	+4616017954949				
7	Development	Branshaw, Llew	+4616017954871				
8	Development	Luong, Xi	+4616017954878				
9	Development	Lowendahl, Roger	+4616017954885				
10							
11							
12							
13							
14							
15							

# CHECKING STATUS AND LOGS

The tabs under Status show the settings on corresponding tabs plus device info and logs.

## DEVICE

⇒ Select **Status** > **Device**.

The screenshot shows the Avaya B179 SIP Conference Phone web interface. The top navigation bar includes 'Status', 'Phone book', 'Call list', and 'Settings'. The 'Status' tab is active, and the 'Device' sub-tab is selected. The 'Device' sub-tab has a dropdown menu with options: 'Device', 'Network', 'Time & Region', 'SIP', 'Media', 'Log', and 'Licenses'. The main content area is divided into two sections: 'Hardware' and 'Software'. The 'Hardware' section lists the following information:

Hardware	
Product name	Avaya B179
Serial Number	03212548
MAC address	00:17:7D:12:0A:36

The 'Software' section lists the following information:

Software	
Application	2.3.0

The top right corner of the interface indicates 'You are logged in as DEFAULT' with a 'Logout' link.

The Device tab shows phone information including serial number, network port and current software version.

On phone: ⚙ > **STATUS** > **DEVICE** (8,6).

## NETWORK

⇒ Select **Status** > **Network**.

The screenshot shows the Avaya B179 SIP Conference Phone web interface. The top navigation bar includes 'Status', 'Phone book', 'Call list', and 'Settings'. The 'Status' tab is active, and the 'Network' sub-tab is selected. The 'Network' sub-tab has a dropdown menu with options: 'Device', 'Network', 'Time & Region', 'SIP', 'Media', 'Log', and 'Licenses'. The main content area is divided into two sections: 'Network' and 'Quality of Service'. The 'Network' section lists the following information:

Network			
DHCP	<input checked="" type="radio"/> On	<input type="radio"/> Off	
IP address	10.10.1.64	Hostname	AvayaB179
Netmask	255.255.255.0	Domain	
Gateway	10.10.1.1		
Primary DNS	10.10.1.10		
Secondary DNS	10.10.1.10		

The 'Quality of Service' section is currently empty. The top right corner of the interface indicates 'You are logged in as ADMIN' with a 'Logout' link.

On phone: ⚙ > **STATUS** > **NETWORK** (8,2).

# CHECKING STATUS AND LOGS

## TIME & REGION

⇒ Select **Status > Time & Region**.

**AVAYA** You are logged in as ADMIN [Logout](#)

**Status** Phone book Call list Settings

Device: Network **Time & Region** SIP Media Log Licenses

**Time**

Enable NTP	<input checked="" type="radio"/> On <input type="radio"/> Off	Date	2012-02-08
Time	10:10:59	Daylight saving	<input type="radio"/> Yes <input checked="" type="radio"/> No
Timezone	UTC+01:00		
NTP Server	pool.ntp.org		

**Region**

Country	Sweden (SWE)
---------	--------------

On phone: ⚙️ > **STATUS** > **TIME** (8,5).

## SIP

⇒ Select **Status > SIP**.

**AVAYA** You are logged in as ADMIN [Logout](#)

**Status** Phone book Call list Settings

Device: Network Time & Region **SIP** Media Log Licenses

**Account 1**

Status	NOT REGISTERED		
Account name	100	Realm	*
User	100	Authentication name	100
Registrar	192.168.0.1	Password	*****
Proxy		Registration interval	300

**Account 2**

Status	NOT REGISTERED		
Account name	200	Realm	*
User	200	Authentication name	200
Registrar	192.168.0.1	Password	*****
Proxy		Registration interval	300

**NAT Traversal**

STUN	<input type="radio"/> On <input checked="" type="radio"/> Off	STUN host	
Offer ICE	<input type="radio"/> Yes <input checked="" type="radio"/> No		
TURN	<input type="radio"/> On <input checked="" type="radio"/> Off	TURN user	
TURN host		Password	

**Advanced**

# CHECKING STATUS AND LOGS

## Account 1 and Account 2

On phone: **⚙** > **STATUS** > **ACCOUNTS** (8,1,1 and 8,1,2).

## NAT traversal

On phone: **⚙** > **STATUS** > **NAT TRAVERSAL** (8,3).

## Transport

On phone: **⚙** > **STATUS** > **ACCOUNTS** > **TRANSPORT** (8,1,3).

## MEDIA

⇒ Select **Status** > **Media**.

**AVAYA** You are logged in as ADMIN [Logout](#)

Status Phone book Call list Settings

Device Network Time & Region SIP **Media** Log Licenses

**Codec**

Codec	Priority
G722	4
G711 Alaw	3
G711 Ulaw	2
G729	1

**Security**

SRTP  Disabled  Optional  Mandatory

Secure signaling  No  TLS  SIPS

**DTMF**

DTMF Signaling  RFC 2833  SIP Info  Inband

**Advanced**

First RTP port 1000

On phone: **⚙** > **STATUS** > **MEDIA** (8,4).

# CHECKING STATUS AND LOGS

## LOG

⇒ Select **Status** > **Log**.

The screenshot shows the Avaya web interface. At the top left is the Avaya logo. At the top right, it says "You are logged in as ADMIN" with a "Logout" link. Below this is a navigation menu with "Status", "Phone book", "Call list", and "Settings". Underneath is a red bar with "Device", "Network", "Time & Region", "SIP", "Media", "Log", and "Licenses". The "Log" tab is selected. Below this bar, there is an "Application" dropdown menu, a "Change" button, and an "Refresh" button. The main section is titled "Application log" and contains a "Log level" dropdown menu set to "Info", a "Set" button, and a "Clear Log" button. The log content shows several lines of text, including:
 

```
Feb 8 07:03:55: INF: SIP: Using not established account 100
Feb 8 07:03:55: INF: sip_manager: UA_OSC-NAMESERVER_SOUND = 3
Feb 8 07:03:55: INF: sip_manager: UA_OSC-NAMESERVER_ID = 10.10.1.10
Feb 8 07:03:55: INF: sip_manager: UA_OSC-NAMESERVER_ID = 10.10.1.10
Feb 8 07:03:55: INF: sip_manager: UA_OSC-NAMESERVER_PROXY_OSC = 0
Feb 8 07:03:56: INF: SIP: Version 1.0-trunk for SIP-526-11000-000 INITIALIZED
Feb 8 07:03:57: INF: Registration sent
Feb 8 07:03:57: INF: Registration sent
Feb 8 07:03:57: INF: sip_manager: Setting codec G722, prio 4
Feb 8 07:03:57: INF: sip_manager: Setting codec PCMA, prio 5
Feb 8 07:03:57: INF: sip_manager: Setting codec PCMU, prio 2
Feb 8 07:03:57: INF: sip_manager: Setting codec G729, prio 1
Feb 8 07:04:00: INF: SIP: Using not established account 100
Feb 8 07:04:05: INF: SIP: Using not established account 100
Feb 8 07:04:10: INF: SIP: Using not established account 100
Feb 8 07:04:30: WRN: SIP registration failed, status=408 (Request Timeout)
Feb 8 07:04:30: WRN: SIP registration failed, status=408 (Request Timeout)
Feb 8 07:04:30: INF: Registration sent
Feb 8 07:04:30: INF: Registration sent
Feb 8 07:10:03: WRN: SIP registration failed, status=408 (Request Timeout)
```

The Log tab contains five types of log messages and can be useful for trouble shooting.

⇒ Select the log you want to review and click on the **Change** button.

① The **Refresh** button adds all new messages sent since the present log was chosen.

### Application log

This shows the phone application messages. The log can be filtered from “Fatal” (only the fatal error messages) to “Trace” (all messages).

The **Clear log** button erases all content in the log.

### SIP Trace

The SIP Trace logs the communication between the phone and the SIP PBX.

The **Clear log** button erases all content in the log.

It is possible to disable the SIP trace log

⇒ Select SIP logging **Off** and click on the **Set** button.

# CHECKING STATUS AND LOGS

## **System log**

Shows the phone system messages.

## **Device management log**

Logs the device management activities.

## **Upgrade log**

Logs the upgrade procedure.

## TECHNICAL DATA

Size	Diameter 240 mm, height 77 mm
Weight	1 kg
Color	Liquorice black
Display screen	Illuminated graphics (LCD), 128x64
Keypad	Alphanumerical 0–9, *, on, off, mute, hold, volume up, volume down, 5 buttons for menu navigation, line mode, conference guide
Anti-theft protection	Kensington security slot
Memory	Support for SD and SDHC memory cards (Recording capacity 35h/GB)

### Connectivity

Network connection	Modular 8P8C (RJ45), Ethernet 10/100 Base T
Power supply	AC adapter 100–240 V AC/14 V DC IEEE 802.3af Power over Ethernet, Class III.
Extra microphones	2 modular 4P4C
Auxiliary	Modular 4P4C for wireless headset

### Network and communication

Network addressing	DHCP and static IP
NAT traversal	STUN, ICE and TURN
Connection protocol	SIP 2.0 (RFC 3261 and companion RFCs)
Transport	UDP, TCP, TLS and SIPS
Security	SRTP and TLS
Quality of Service	DiffServ, VLAN and 802.1x
Audio support	Codecs: G722, G711 A-law, G711 $\mu$ -law, G729ab
DTMF tone generation	RFC, SIP INFO, In-band
Time servers	NTP and SNTP
Daylight saving: Configurable for automatic adjustments.	
Configuration	Via integrated web server

### Configuration and provisioning

Configuration	Via integrated web server, HTTP or HTTPS Separate user and administrator login for secure configuration.
Device management	Support for device management for easy configuration and updating of multiple conference phones.



# TECHNICAL DATA

## Directory

Internal phone book	1,000 entries per profile (4 password protected profiles) Export/import of directory Call list
External directory	Support for LDAP

## Sound

Technology	OmniSound® Wideband
Microphone	Omni-directional
Reception area	Up to 30 metres <sup>2</sup> , >10 people
Speaker	Frequency band 200–7000 Hz,
Volume	90 dB SPL 0.5 m
Equalizer	Three pitches: soft, neutral, bright

## Environment

Temperature:	5°–40°C
Relative humidity:	20-80% condensation free
Recommended acoustic conditions:	Reverberation period: 0.5 S Rt 60 Background noise: 45 dBA

## Approvals

Electrical safety	EN 60950-1:2006, ANSI/UL 60950-1-2002, CAN/CSA-C22.2, No. 60950-1-03
EMC/Radio	EN 301 489-3 V1.4.1 (2002-08), EN 301 489-1 V1.6.1 (2005-09), FCC Part 15 subpart B class A, FCC Part 15 subpart C, EN 300220-1:2000, EN 300220-2:2000 RoHS

