



**Avaya one-X™ Quick Edition**  
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System Administration Guide

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The executable installer program described in the "Upgrading Telephone Software" chapter uses the nullsoft scriptable install system ([nsis.sourceforge.net](http://www.sourceforge.net)) by Nullsoft, Inc.

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# Chapter 1: Technical Overview

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## Introduction

This guide provides instructions for administering all Avaya one-X™ Quick Edition IP telephones and analog gateways making up the one-X Quick Edition network. While administering these devices does not require the technical expertise of a traditional system administrator or IT professional, it is a good practice to make one person responsible for controlling the advanced system-wide features described in this guide.

## Symbolic Conventions

**Note:** This text precedes additional information about a topic.



**Tip:** This symbol highlights the benefits and capabilities of the product or makes you aware of alternative methods that can help you increase efficiency.



**CAUTION:** This symbol calls attention to situations that can result in harm to software, loss of data, or an interruption to service.

## Typographic Conventions

Convention	Description
<u>Document</u>	Underlined text indicates a section or subsection in this document containing additional information about a topic.
“Section”	Text enclosed in double-quotation marks indicates a reference to a specific chapter or section of another document.
<i>Italics</i>	Italic text indicates the title of another document.
<b>System Options</b>	Words shown in bold represent literal elements of the user interfaces.

## one-X Quick Edition Documentation

Avaya one-X Quick Edition product and related documentation is available online at the following URL: <http://support.avaya.com/QuickEdition>

## Key Features and Benefits

Avaya one-X Quick Edition software is embedded in each telephone. The system configures itself initially using default values at installation time, and as new devices are added to the Local Area Network (LAN).

All telephones connected to the same network participate in traffic routing, call handling, and other network-related processes automatically. As an option, a one-X Quick Edition network can be equipped with a PSTN gateway to enable access to traditional telephony systems.

Each telephone has its own software and system backup data, so if one telephone fails the others continue to work, providing embedded fail-over and business continuity. Avaya one-X Quick Edition is ideal for offices having two-to-ten users, scaling up to 20 users.

Installation is as simple as connecting the telephones to the existing company LAN. Through SIP-based, peer-to-peer (P2P) discovery protocols, the telephones discover each other, establish a network, assign themselves extension numbers, build a Corporate directory, and perform system configuration automatically. Afterward, the telephones maintain communications within their own private network. Calls between the private network and the outside world can be made through a thin-trunk interface (analog gateway) to the PSTN, or through a direct or Internet connection via SIP to a Voice over IP (VoIP) service provider.

## Telephone Features

Avaya one-X Quick Edition provides the most frequently used telephony applications, so even the smallest offices can gain operational efficiencies. Refer to the *Avaya one-X Quick Edition Telephone User Guide* (Document No. 16-601411) to learn how to use the features.

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## System Features and Options

System features can be accessed and configured, through any telephone or through a web-based configuration interface, to provision the entire one-X Quick Edition network. All system changes and updates made through system options are communicated to all telephones and gateways on the network automatically.

Telephone and system features include:

- [Changing the Administration Password](#) on page 9
- [Set Management](#) on page 17

- [System Localization](#) on page 28
- [Creating and Managing Telephone Groups](#) on page 30
- [Setting the System Date and Time](#) on page 34
- [System Gateways](#) on page 35
- [Dial Plan](#) on page 36
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- [System Networking Options](#) on page 39
- [Service Provider](#) on page 44
- [Configuring the Auto Attendant Feature](#) on page 53
- [Extending Services to Remote Users through the Teleworker Feature](#) on page 60

**Note:**

Analog gateway settings are also accessed through system options. To configure a gateway, see the *Avaya one-X Quick Edition Configuration Guide for Global Analog Gateways* (Document No. 16-601415) or *Avaya one-X Quick Edition G10 PSTN Gateway Installation Guide* (Document No. 16-600793).

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## Accessing System Options

To access system options through the web-based administration interface, a computer with a web browser must be connected to the same network address space as your telephones.


**Tip:**

To access the options associated with a specific telephone using a web browser, see [Appendix A: Web-based User Options](#) on page 77. Refer to [Appendix B: Options Menus](#) on page 83 for a list of Web-based and telephone interface configurable system options.

**CAUTION:**

Password protection is required to prevent unauthorized users from changing system options. If you are the person who manages system-wide options, change the existing administration password to a password that is known only by you.

### To access options using the web-based administration interface


To log in to the web-based System Options interface, use the IP address of any telephone listed in the Corporate directory as the Uniform Resource Locator (URL) in your web browser. To determine the IP address of a telephone, press the # key and then the Page Right () button.


After you switch to the **System Options** login page, the password that you enter is encrypted before the information is sent to the system. Refer to page [Security](#) on page 78 for protocol and security details.

1. Start the web browser on your computer.
2. In the **Address** field, enter the IP address of a telephone (for example, `https://192.168.0.10`). If you have not disabled the security alert, a message is displayed:
3. Click **OK**. If you have not installed the self-signed security certificate on your computer, a Security Alert message is displayed.
4. Perform one of the following actions:
  - To proceed without installing the security certificate, click **Yes**.
  - If you want to install the security certificate, click **View Certificate**. When the **Certificate** dialog box is displayed, click **Install Certificate** and follow the instructions.
5. Click **System Options** in the upper right corner of the dialog box.
6. In the **Password** field, type the administration password that is needed to access the one-X Quick Edition network.
7. Click **Login**.

The **System Options** home page is displayed. Links to additional pages are displayed in the navigation bar on the left side of the screen. Click an item on the System Options menu to view and edit system-wide settings.

### To access system options menu through buttons on the telephone

The **System Options** menu can be accessed through the Options (  ) button on any telephone.

1. At the telephone, press the Options (  ) button below and to the right of the display area.
2. Select **Options** on the **Main** menu, or press the indicated dialpad key.
3. Select **System Options** on the **Options** menu, or press the indicated dialpad key.
4. Enter the password (the default password, which should have been changed when the network was first created, is 54321). If you changed the default password previously to a password known by you only, enter that password instead.
5. Select the **Done** softkey. The **System Options** menu is displayed.





#### Tip:

Refer to [Appendix B: Options Menus](#) on page 83 for a list of Web-based and telephone interface configurable system options.



#### Tip:

To view options that are displayed on the next or previous telephone display screens, press the Page Right (  ) button or Page Left (  ) button.



---

## Changing the Administration Password

Using a password to access system options prevents unauthorized users from viewing or changing system-wide settings. A default password (54321) is factory installed on one-X Quick Edition devices. This password can be changed as often as necessary to prevent unauthorized access to the system.

The password that you use to access system options must contain at least five numbers up to a maximum of 32 numbers.

### To change the administration password using a web browser

1. Access the **System Options** home page.
2. In the upper right corner of the **System Options** home page, click **Change Admin Password**.
3. In the **Current Password** field, type the current password.
4. In the **New Password** field, type the new password.
5. In the **Confirm Password** field, re-type the new password.
6. Click **Submit**.

**CAUTION:**

If you forget the administration password, contact your technical support representative to have it reset.

### To change the administration password using a Quick Edition IP telephone

1. Display the **System Options** menu.
2. Select **Change Password** on the **System Options** menu.
3. Enter the password that you used to access the **System Options** menu, and then select the **Next** softkey.
4. Enter the new password. The password must contain at least five numbers.
5. Select the **Next** softkey.
6. When you are prompted, re-enter the new password and then select the **Next** softkey.
7. Select the **Ok** softkey.

---

## About the Multisite Provisioning Tool

The Multisite Provisioning Tool (MPT) enables you to simultaneously configure one or more one-X Quick Edition networks from a single central location. Any individual one-X Quick Edition network, a subset of selected networks, or all networks added to the Multisite Provisioning Tool can be configured at once.

## Obtaining and Installing the Multisite Provisioning Tool

You can obtain the MPT software free of charge from the Avaya Technical Support web site at the following address: <http://support.avaya.com/QuickEdition>. Follow the link to the “Download Center” to obtain the software. You can purchase the CD-ROM that also includes the software, language wizard, and documentation.

**Note:**

Refer to the *Avaya one-X Quick Edition Multisite Provisioning Tool Help* (Document number 16-601673) for details.

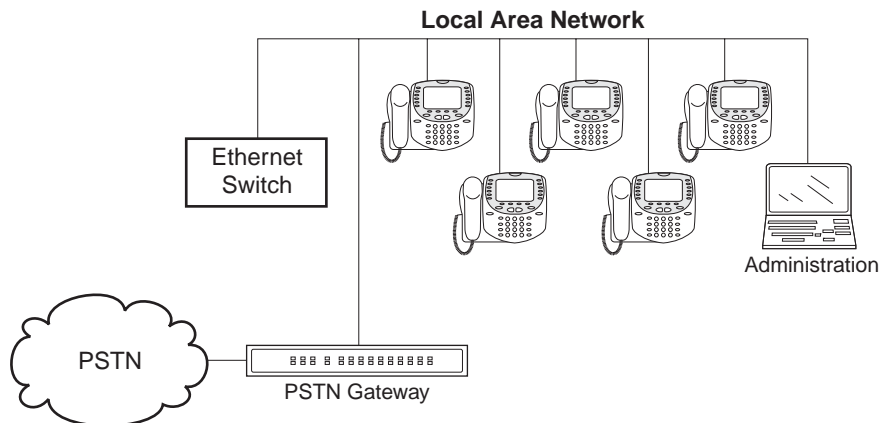
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## Typical Network Configurations

[Figure 1](#) shows a typical local network configuration. A typical local configuration consists of a customer-supplied 10/100 Base-T Ethernet/Fast Ethernet LAN with wall jacks or Ethernet switches, to which your telephones are connected. In addition, an administration computer may be connected to the LAN to provide web-based access to user-specific and system-wide settings.

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**Figure 1: Local Network Configuration**



**Note:**

Your one-X Quick Edition network must be equipped with a gateway to access the PSTN through PSTN lines supplied by your telephone service provider.

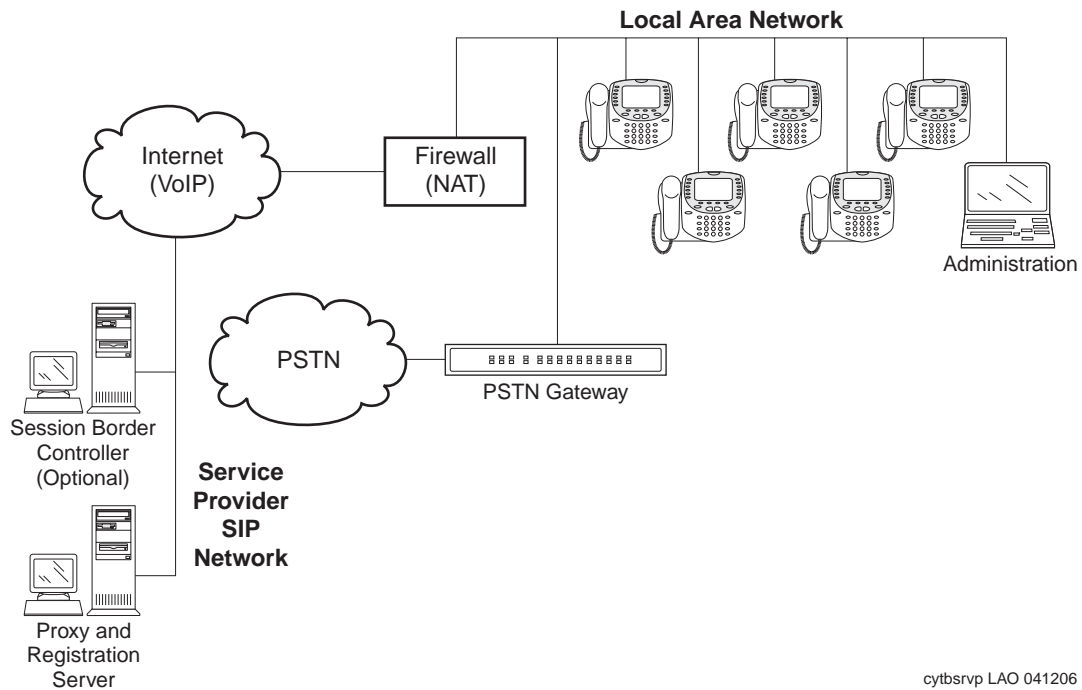
You can use either of the following methods to connect telephones to the Ethernet LAN:

- a direct connection to the LAN—connect the telephone through a wall jack or Ethernet switch
- a shared connection to a computer—the telephone shares the computer’s connection to the Ethernet LAN

Power is ideally supplied to telephones by plugging the telephones into 802.3af PoE-enabled Ethernet wall jacks using the supplied Cat5 modular line cords. You may optionally connect the telephone between a computer and the Ethernet LAN—the telephone and the computer share the connection to the LAN.

[Figure 2](#) shows a typical SIP service provider configuration. The service provider's SIP network provides a gateway to the Internet, a SIP proxy and registration server, and (optionally) a session border controller.

**Figure 2: SIP Service Provider Configuration**



cytbsrvp LAO 041206

*The service provider's SIP network provides a gateway to the Internet, a SIP proxy and registration server, and (optionally) a session border controller.*

## Important Networking Concepts

Regardless of whether you are creating a new network or adding telephones to an existing network, connect the telephones to the *same IP network segment*. If you have a complex network, this means:

- The IP addresses of all telephones and PSTN gateways must belong to the same network address space. Avaya one-X Quick Edition devices assign themselves IP addresses in the same network address space automatically if the LAN does not include a Dynamic Host Configuration Protocol (DHCP) server host. Avaya one-X Quick Edition devices try to obtain an IP address from a DHCP server before they assign themselves IP addresses.
- Unless assigned a static IP address previously, Avaya one-X Quick Edition devices assign themselves IP addresses each time they are connected to an Ethernet LAN. If you move devices making up the same one-X Quick Edition network from one LAN to another, they may assign themselves different IP addresses without disrupting communications between established network peers.
- Virtual LANs (VLANs) may be defined on an Ethernet switch. If your IP network includes VLANs, include all of the one-X Quick Edition devices that need to communicate with each other in the same VLAN. If multiple one-X Quick Edition networks need to be run on the same IP network segment, ask your LAN administrator to define the required number of VLANs on the Ethernet switch and assign a single one-X Quick Edition network to each VLAN.

---

## Performance Recommendations

To ensure optimum performance, the following operational guidelines are assumed:

- Supply RJ45 Category 5 (Cat5) or better (for example, Cat5e) Ethernet cabling to connect equipment such as PSTN gateways to the LAN.
- If your voice network equipment is capable of full-duplex transmission, select full-duplex instead of half-duplex transmission. Full-duplex devices enable callers on both ends of the connection to speak simultaneously.
- You can measure and correct delay, jitter, and packet loss on your network. Delays in speech delivery, variations in the amount of time needed for speech to travel across the network, and packet loss all have the potential to degrade the overall quality of one-X Quick Edition applications. To achieve the best voice quality performance, ensure that end-to-end delay, jitter, and packet loss on the network adhere to the following tolerances:
  - Packet loss less than 1%
  - Jitter less than 20 ms
  - End-to-end delay less than 50 ms
  - Ping delay less than 100ms

- Do not connect a network server PC (for example, a web server, file server, or database server) or a network printer to the PC port on a telephone.
- Connect all one-X Quick Edition devices to the same network segment—their IP addresses must belong to the same network address space. If your network includes VLANs, include all one-X Quick Edition devices in the same VLAN.
- Avaya one-X Quick Edition devices implement Quality of Service (QoS) at TCP/IP Layer 2 (OSI Reference Model). For details, see [Enabling Priority \(QoS\) Tagging](#) on page 40. If you are experiencing performance degradation with the one-X Quick Edition system due to the traffic load on your network, consider enabling QoS on the network.
- Multicast IP traffic must be permitted on the LAN. The default multicast address used by the one-X Quick Edition system is 239.192.228.123. This is a limited-scope address that is not routable over the public network (Internet).
- The one-X Quick Edition system supports IGMP v2. A number of higher-end Layer 2 switches support IGMP snooping by default. If the one-X Quick Edition devices making up the network cannot communicate with each other, turning off IGMP snooping on the switch may resolve the problem if the switch requires an IGMP querier to be present on the network but an IGMP querier is not available.
- If any one-X Quick Edition devices are connected to a switch that runs the Spanning Tree Protocol (STP), configure the one-X Quick Edition switch connections with Portfast (also known as “spantree start-forwarding” on a Cisco 1900XL). Portfast enables the switch to go into a forwarding state almost immediately after it is powered on.

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## Adding an Analog Gateway

Each analog gateway has four Foreign Exchange Office (FXO) ports that provide one-X Quick Edition networks with access to the Public Switched Telephone Network (PSTN) through PSTN lines managed by your telephone service provider.

To install an analog gateway, see the *Avaya one-X Quick Edition G10 PSTN Gateway Installation Guide* (Document No. 16-600796) or *Avaya one-X Quick Edition G11 Global Analog Gateway Installation Guide* (Document No. 16-601414). To configure a global analog gateway, see the *Avaya one-X Quick Edition Configuration Guide for Global Analog Gateways* (Document No. 16-601415) or *Avaya one-X Quick Edition G10 PSTN Gateway Installation Guide* (Document No. 16-600793).

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## Supporting Telecommuters

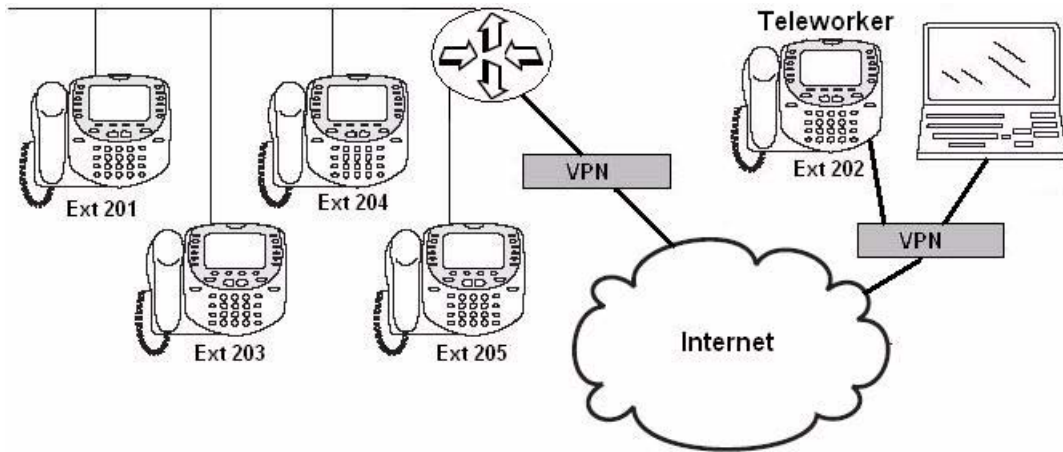
The Teleworker Application lets a worker at any remote office connect a telephone to a high-speed Internet connection and access the Corporate directory and most of the other features and services that are available to everyone who uses the one-X Quick Edition system.

**Note:**

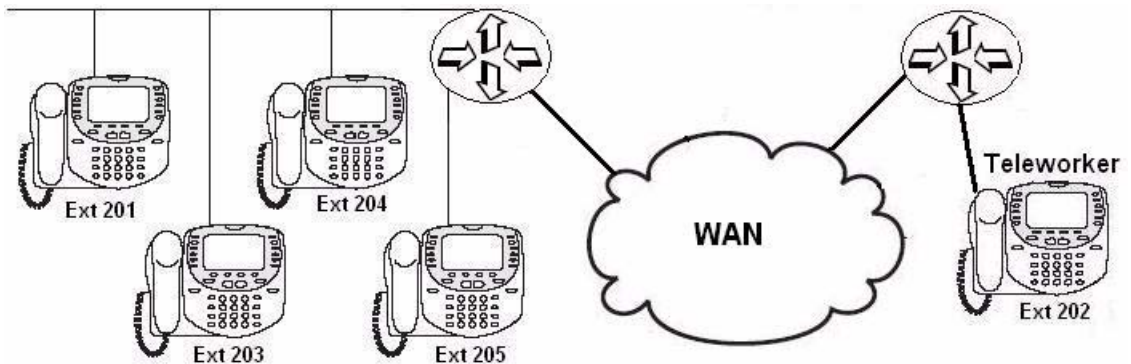
To provide a secure link to the one-X Quick Edition network from a remote office, a customer-supplied VPN configuration is required.

For information about how to configure and use the teleworker feature, see [Extending Services to Remote Users through the Teleworker Feature on page 60](#).

**Figure 3: Teleworker over Internet (VPN)**



**Figure 4: Teleworker over WAN**



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## SIP (VoIP) Interoperability

The Session Initiation Protocol (SIP) service provider feature supports interoperability with certified SIP-based (VoIP) service provider networks, and with the Avaya Communication Manager (CM) gateway and SIP Enablement Services (SES) server. For example, a CM S8300 media server module installed in a G700 media gateway is capable of running SES-edge and/or home server applications.

To support SIP interoperability, the telephones on your one-X Quick Edition network may be connected to the service provider's network directly, or through a Network Address Translation (NAT) device such as a VoIP gateway or router.

For more information, see [Configuring Interoperability with a SIP \(VoIP\) Service Provider](#) on page 44.

**Note:**

To configure an SES/CM system to communicate with a one-X Quick Edition network, see *Configuring SIP Trunking to a one-X Quick Edition Network on CCS/SES and CM* (Document No. TSS-000001) and *Configuring a one-X Quick Edition Network to Interoperate with CCS/SES and CM* (Document No. TSS-000002).

---

## Growing Your Network

In a small office, a single network serves the local user community. Users have telephones and can call each other by dialing an extension number from the Corporate directory.

PSTN lines may be connected to the network through one or more PSTN gateways. To place external calls to the PSTN, users must dial the 9 prefix (9 by default but flexible) before dialing the PSTN number. External calls are routed via PSTN service provider trunks (PSTN lines).

Calls to and from a SIP (VoIP) network are supported through the SIP interoperability feature. To place external calls to a SIP network, users must dial the 8 prefix before dialing the number. External calls are routed via a SIP service provider network (Internet).

You may consider including more than 20 devices in a single one-X Quick Edition network provided that the application needs of your user community are low. However, the configuration and capacity of your LAN may impose practical limitations with a larger than recommended number of devices on the network and as a result, quality of service could suffer.

To ensure the best quality of service without needing to optimize network performance, plan on a maximum of 20 telephones and PSTN gateways per network segment. Beyond 20 devices, the requirement for more complex telephony applications may increase.



**Tip:**

If you need to grow your network beyond 20 devices, it is a good idea to have your network assessed first. You may consider migrating to Avaya IP Office and/or Avaya Communication Manager.

The Avaya Intelligent Communications portfolio supports enterprises on a migration path from traditional telephony to IP, SIP, mobile platforms, Web-based services, and increasingly sophisticated architectures. Avaya Professional Services can help you manage your migration path.

A single central administration interface, the Multisite Provisioning Tool, is available from the Avaya Technical Support site to help simplify the tasks of configuring a large number of individual one-X Quick Edition networks. For example, you can use the Multisite Provisioning Tool to push software upgrades to the devices making up one or more one-X Quick Edition networks.



# Chapter 2: Configuring the System

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## Set Management

System options let you manage the telephones that can be accessed through the Corporate directory.



---

### Sets

#### Adding a Telephone to the System



You can simply connect the telephone to the LAN. Through SIP-based, P2P discovery protocols, the telephone will discover its peers, join the network, assign itself an extension number, and enter itself into the Corporate directory. You will be prompted to enter a name; comma-separated last and then first name.

#### To view information about a Quick Edition IP telephone

1. At the telephone, press the Options (  ) button.
2. Select **Set Details** on the **Main** menu, or press the indicated dialpad key. The extension number, name, version number of the software load, and site identifier are displayed.
3. Select the **Ext** softkey, or press the indicated dialpad key.  
The extension number, name, IP address, and MAC address are displayed.
4. Select the **Back** softkey to display the previous menu, or press the Phone/Exit (  ) button to clear the display area.



#### Tip:

To view options that are displayed on the next or previous telephone display screens, press the Page Right (  ) button or Page Left (  ) button.

#### Changing the Extension Number of a Telephone

When you connect a telephone to the LAN and direct it to join an existing network, the telephone automatically assigns itself a unique extension. The newly assigned extension number is added to the Corporate directory automatically.

Any changes are communicated to the system-wide Auto Attendant, and to telephone users through the Corporate directory. When a device is added to the network, it will be assigned a number in the regional dialing plan range. You can change an assigned extension number to an unused number in the extension range.

### To change the extension number of a telephone using a web browser

1. Access the **System Options** home page. The **Set Management** list is displayed.
2. In the **Name** column, click the name that corresponds to the extension number to be changed. The **Set Details** dialog box is displayed.
3. Click **Change Details**. The **Edit Set Details** dialog box is displayed.
4. Type an unused extension number in the **Extension** field.
5. Click **Submit**.

### To change an extension number using a Quick Edition IP telephone

1. Display the **System Options** menu.
2. Select **Set Management** on the **System Options** menu.
3. Select **Set Extension** on the **Set Management** menu, or press the indicated dialpad key.
4. Select the **Chg** softkey.
5. Select the **Bksp** softkey to move the cursor to the left and delete the existing number.
6. Enter an unused extension number in the regional extension number range.
7. Select the **Save** softkey.

## Changing the Name Assigned to an Extension Number

When a telephone is added to the network, the telephone prompts the installer for a name. The information is presented in the Corporate directory.

Enter the name of the person who will be using the telephone, last name first.

- The names in the Corporate directory are sorted alphabetically by last name.
- The dial-by-name function of the Auto Attendant feature attempts to match caller key presses to the Corporate directory name, starting with the first character of the last name.

### To change the name assigned to a telephone using a web browser

1. Access the **System Options** home page. The **Set Management** list is displayed.
2. In the **Name** column, click the name that needs to be changed. The **Set Details** dialog box is displayed.
3. Click **Change Details**. The **Edit Set Details** dialog box is displayed.
4. Type the new name in the **Name** field.

5. Click **Submit**.

### To change the name assigned to a telephone using a Quick Edition IP telephone

You cannot change the name of a telephone using the buttons on the telephone unless you know the password needed to access user options on that telephone. To change the name, refer to the *Avaya one-X Quick Edition Telephone User Guide*.

## Changing the Paging Zone Assigned to a Telephone

The paging feature can be used to broadcast an announcement to a predefined zone of telephones. All persons in the paging zone hear the broadcast, unless they are on an active call. The general zone includes all telephones on the network as well as any external paging equipment connected to the External Paging jack of a PSTN gateway. Members of the general zone cannot be changed; however, any telephone can be configured to belong to one additional zone.

### To add a telephone to a specific paging zone

1. Access the **System Options** web page. The **Set Management** list is displayed.
2. In the **Name** column, click the name associated with the telephone that you want to configure. The **Set Details** dialog box is displayed.
3. Click **Change Details**. The **Edit Set Details** dialog box is displayed.
4. Select a zone from the **Page Zone** list. The telephone will be added to the selected zone.
5. Click **Submit**.

## Resetting a User Password

You can reset a user's user-options and voicemail passwords to the default password (12345).

After you reset the password, advise the user to change the password (see the *Avaya one-X Quick Edition Telephone User Guide*).

### To reset a user's passwords using a web browser

1. Access the **System Options** home page. The **Set Management** list is displayed.
2. In the **Name** column, click the Corporate directory name of the telephone on which the password is to be reset. The **Set Details** dialog box is displayed.
3. Click **Reset Password**. The **Reset User Password** dialog box is displayed.
4. Click **Submit**. The user-options and voicemail passwords are reset to 12345.

### To reset a user's passwords using the Quick Edition IP telephone

1. Display the **System Options** menu.

2. Select **Set Management** on the **System Options** menu.
3. Select **Reset Password** on the **Set Management** menu.
4. When you are prompted to reset the password to the default value, select the **Yes** softkey.
5. Select the **Ok** softkey. The user-options and voicemail passwords are reset to 12345.

### Removing a Telephone or Gateway from the System

The following procedures remove a telephone or gateway from the system and free the associated extension number so that it can be reused. Removing an extension number from the system prevents all telephones and PSTN gateways associated with the system from recognizing that extension number until it is re-assigned. The extension number may be re-assigned automatically the next time a telephone is added to the one-X Quick Edition network.

**Note:**

These procedures will reset the device to factory default. To reset the entire system, complete the procedure for all devices (including gateways). The factory default network will prompt the administrator to enter a site name.

#### To remove a device from the system using a web browser

1. Access the **System Options** home page.
2. In the **Set Management** list, click **Remove Extension**.
3. From the **Extension To Remove** list, select the extension from the drop-down list.
4. Click **Validate**.
5. When you are prompted to remove the extension, click **Remove Extension**.

When the process completes, a confirmation message is displayed on the device that had its extension number removed.

6. Unplug the device from the network. Connect the telephone to the LAN and it will join the network.

#### To remove a telephone from the system using a Quick Edition IP telephone

1. Using any telephone, display the **System Options** menu.
2. Select **Set Management** on the **System Options** menu.
3. Select **Remove Extension** on the **Set Management** menu.  
The extension number of the telephone that you accessed is displayed.
4. Perform one of the following actions:
  - To remove the telephone that you accessed from the system, select the **Next** softkey.
  - To remove a different telephone, enter the extension number of the telephone, and then select the **Next** softkey.

5. When you are prompted to remove the extension, select the **Yes** softkey.

When the process completes, a confirmation message is displayed on the device that had its extension number removed.

6. Unplug the device from the network.

**Note:**

If you were performing this procedure to reset the device to factory defaults, connect the device to the LAN and it will join the network.

## Modifying Telephone Network Address Settings

You can specify the IP address of a telephone, the network mask of the IP address, and optionally, the IP addresses of a default IP gateway and Domain Name System (DNS) server if they reside in the one-X Quick Edition network address space.

When you change the IP address of a telephone, the new IP address is kept by the telephone through subsequent power cycles. This kind of IP address is said to be “static.” Any changes to network addresses are automatically communicated to all of the one-X Quick Edition devices.



**CAUTION:**

It is not usually necessary to configure network options. Do not change any network address settings unless you have advanced network knowledge. These capabilities are offered if required to support multi-branch network installations.

Before you begin, obtain IP addresses for the required network configuration.

### To change telephone network address settings using a web browser

1. Log in to the **System Options** interface.
2. In the **Name** column, click the Corporate directory name of the telephone that you want to configure. The **Set Details** dialog box is displayed.
3. Click **Networking**. The **Network Settings** dialog box is displayed.
4. Click **Change Details**. The **Edit Network Settings** dialog box is displayed.
5. To assign a static IP address to the telephone, enter an unused **IP address** in the IP Address field. The address must correspond to the network address space used by the connected network segment (for example, if the network address space is 198.16.10.0, you could enter an IP address such as **198.16.10.100**).
6. Verify that the network mask matches the telephone IP address (for example, **198.16.10.0**). If required, specify a different network mask to qualify the static IP address.
7. To specify the IP address of a default IP gateway (for routing traffic generated by the telephone to a next-hop router), enter the IP address of the gateway in the **Gateway** field.
8. Click **Submit**.

### To change network address settings using a Quick Edition IP telephone

1. At the telephone that you want to configure, display the **System Options** menu.
2. Select **Network Options** on the **System Options** menu.
3. Select **IP Address** on the **Network Options** menu. The current settings are displayed.
4. Select the **Chg** softkey.
5. To assign a static IP address to the telephone, enter an unused IP address in the IP Address field. The address must correspond to the network address space used by the connected network segment (for example, if the network address space is 198.16.10.0, you could enter an IP address such as **198.16.10.100**). Select the **.** softkey to add separator characters between the digits.
6. Select the **Next** softkey.
7. Verify that the network mask matches the telephone IP address (for example, **198.16.10.0**). If required, specify a different network mask to qualify the static IP address. Select the **.** softkey to add separator characters between the digits.
8. Select the **Next** softkey.
9. If you want to specify the IP address of a default IP gateway (for routing traffic generated by the telephone to a next-hop router), enter the IP address of the gateway. Select the **.** softkey to add separator characters between the digits.
10. Select the **Next** softkey.
11. If you want to specify the IP address of a DNS server (for resolving device names to IP addresses), enter the IP address of the DNS server host. Select the **.** softkey to add separator characters between the digits.
12. Select the **Next** softkey followed by the **Save** softkey.

---

## Viewing Registration Information for Optional Features

You can view registration information about special features through the web-based user options interface (Terminal Settings), the web-based administration interface, or the telephone's Optional Features button.

Features that have been activated on a telephone are displayed on the telephone **Feature List** with a check symbol. Features that are available but have not been activated are displayed on the telephone **Feature List** menu with an **X** symbol.



Available special features include:

- Email Notification of Voicemail
- Web-based System Administration
- Teleworker Application

### To view registration information for special features using a web browser

1. Log in to the **System Options** web interface.
2. In the Set Management **Name** column, click the Corporate directory name of the telephone in question.
3. Click **Features**. The Set Optional Features **Available** column indicates whether a feature has been activated. **N/A** means the feature has not been activated.

### To view registration information for special features using a Quick Edition IP telephone

1. At the telephone, press the Options (  ) button.
2. Select **Opt Features** on the **Main** menu.
3. Select **Email Fwd Options**, **WebAdm Sys Options**, or **Teleworker Options** on the **Feature List** menu. The registration code is displayed.
4. Select the **Exit** softkey to display the previous menu, or press the Phone/Exit (  ) button to clear the display area.

---

## Email Notification of Voicemail

When configured and enabled, the email-notification-of-voicemail feature sends e-mail messages to alert users whenever new voicemail messages are received. The e-mail message contains call header information, including the caller name (if available), caller number, and the time and length of the call. A recording of the voicemail message is created as a G.729a-encoded audio file (in WAV file format) and the file is forwarded as an e-mail attachment.

**Note:**

To play the audio file, the user's PC must be equipped with a media player that supports the G.729a CODEC. Users may listen to the voicemail recording by opening the audio file using the media player. For more information, see [Playing an Audio File Attachment](#) on page 80.

For information about how to configure and enable this feature, see [Supporting E-mail Notification of Voicemail](#) on page 42.

---

## Web-based System Administration

If your telephones are connected to the same network as the administration computer, you can configure system-wide options using the web-based administration interface. The web-based administration interface can be accessed using the web browser on the computer (see [To access options using the web-based administration interface](#) on page 7).

## Teleworker Application

Using the teleworker application, a remote user can connect a telephone to a high-speed Internet connection at any remote location and access the Corporate directory and most of the other features and services that are available to everyone who uses the one-X Quick Edition system.

**Note:**

To provide a secure link to the one-X Quick Edition network from a remote office, a customer-supplied VPN configuration is required.

For information about how to configure and use the teleworker feature, see [Extending Services to Remote Users through the Teleworker Feature on page 60](#).

---

## Operator

### To edit the operator extension

1. Access the **System Options** home page. The **Set Management** list is displayed.
  2. Click **Operator**. The **Edit Operator Extension** dialog box is displayed.
  3. Select an operator extension from the drop-down list.
  4. Click **Submit**.
- 

## Upgrade Software Loads

All devices on a one-X Quick Edition network must be running the same version of software. If your system is based on one-X Quick Edition Release 2.0 (or earlier) and you would like to add an Avaya 46xx SW IP telephone or analog gateway that is running one-X Quick Edition Release 3.0 (or later) software, you must upgrade the software on your existing system to the current software version first.

You can obtain the software upgrade package from the Avaya Technical Support web site at the following address: <http://support.avaya.com/QuickEdition>

Follow the link to the “Download Center” to obtain the software upgrade package.

Before you download the software upgrade package, check the version numbers of the software loads on your telephones and analog gateways. Determine the version of software running on the network through the **Set Details** option on the **Main** menu.

- Minor version upgrades can only be applied to devices running the same major software version. For example, version 2.2 can be applied to a device running version 2.1 or 2.0.



- Major version upgrades can only be applied at an X.0 level. For example, version 3.0 can only be applied to a version 2.0 software load. A version 3.2 software load cannot be applied directly to version 2.0 software without installing version 3.0 software first.

**Note:**

Avaya one-X Quick Edition devices frequently check the software running on the network to determine whether all software loads are compatible.

If the major software version number of any software running on the network is not compatible (for example, one device has version 3.0 and one other device has version 2.0), the telephones having older build will display “Mismatched software versions”.

If the minor version number is not identical (for example, one device has version 2.2 and at least one other device has version 2.1), the telephones having older builds will display the software compatibility message “Newer software available”.

You cannot clear these notification messages—the system clears them automatically when different software versions are no longer detected.

You can upgrade the software load on a single telephone or on all devices making up the one-X Quick Edition network.

The software upgrade package must be installed on a computer that can access the one-X Quick Edition devices that you want to upgrade. When you start the installer program, a Trivial File Transfer Protocol (TFTP) server is started on the computer. Afterward, selecting system options commands the TFTP server to download the new software load to selected one-X Quick Edition devices.

**Note:**

If you have a firewall that does not allow TFTP communications, the firewall may block the software upgrade process. If necessary, configure the firewall to allow TFTP communications while you upgrade the software.

**CAUTION:**

Before you begin, read the README file that is provided with the software upgrade package and shut down all other applications on the TFTP PC. Next, start the TFTP server by starting the installer program.

During a software upgrade, active calls are dropped and new calls cannot be placed or taken until the process completes.

**To upgrade the software using a web browser**

1. Log in to the **System Options** interface (see [To access options using the web-based administration interface](#) on page 7).
2. In the **Set Management** list, click **Software Upgrade**.
3. In the **TFTP Server** field, type the IP address of the computer on which the TFTP server is running.
4. Perform one of the following actions:

- To upgrade all of the telephones and analog gateways in the network, select **Upgrade All Sets**.
  - To upgrade a single one-X Quick Edition device, clear **Upgrade All Sets** and select the extension number from the **Set To Upgrade** list.
5. Click **Prepare Upgrade**.

The **Start Software Upgrade** dialog box is displayed. The list shows what will be done and which components are included in the new load.
  6. Click **Start Upgrade**.
  7. When you are prompted to begin the upgrade process, click **Confirm Upgrade**.

A message is displayed in the web browser during the upgrade process. Meanwhile, progress messages are displayed in the display area(s) of the telephone(s). Wait two-to-three minutes for the network to re-initialize.
  8. After the upgrade process completes, in the web browser window, click **Click Here When Done**.



**Tip:**

If your network includes an analog gateway, check for a green Power LED on the front panel to verify that the analog gateway is ready—analog gateways take slightly longer than telephones to re-initialize.

9. Exit the TFTP server.

### To upgrade the software using a Quick Edition IP telephone

1. At the telephone that you want to upgrade, display the **System Options** menu (see [To access system options menu through buttons on the telephone](#) on page 8).
2. Select **Set Management** on the **System Options** menu.
3. Select **Upgrade** on the **Set Management** menu.
4. Select the **Upg** softkey.
5. The IP address of the administration computer (on which the TFTP server is running) should be displayed. If the IP address is not correct, select the **Chg** softkey and enter the correct IP address.



**Tip:**

To determine the IP address of the administration computer: on the Windows **Start** menu, select **All Programs > Accessories > Command Prompt**. At the command prompt, type `ipconfig /all`. Look for the IP address in the list of displayed information.

6. Select the **Next** softkey.

The **Upgrade Settings** menu displays the software components included in the load.

7. Select the **Next** softkey.

The **Upgrade Options** menu displays two options for performing the upgrade:

- **Clean Database**—When selected, the software is upgraded and all user-configurable settings and system-configuration data are erased (the settings revert to factory settings).



**CAUTION:**

Do not select **Clean Database** unless you intend to erase all existing user and system configuration data.

- **Upgrade all?**—When selected, the software on all telephones and analog gateways belonging to the same network as the telephone mentioned in Step 1 are upgraded. If you do not select this option, the software load on your telephone only is upgraded.
8. If you want to erase all existing user and system configuration data (that is, return the telephone and/or system to factory configured settings), select **Clean Database** on the **Upgrade Options** menu, or press 1 on the dialpad. The option is selected.
  9. If you want to upgrade all of the telephones and analog gateways in the network, select **Upgrade all?** on the **Upgrade Options** menu, or press 2 on the dialpad. The option is selected.
  10. Select the **Next** softkey.
  11. When you are prompted to start the upgrade process, select the **Yes** softkey.  
A progress message is displayed. When the process completes, the telephone restarts.
  12. At the administration computer, exit the installer program.

---

## Language Packs

The software on each Quick Edition IP telephone includes languages for the telephone user and system interfaces. Language packs will provide languages to the web interfaces, auto attendant prompts, and voicemail prompts to match those already selected for the system and user interfaces on the devices.

1. Insert the software and documentation CD-ROM into the drive on a computer that can access the one-X Quick Edition devices. The language wizard will auto-run.
2. Select a language from the drop-down list.
3. Read the Software License Terms and click **I Agree**.
4. Click **Next** after confirming that you have satisfied the **Check list**.
5. Enter the IP address of any device in the one-X Quick Edition network. You can change the language for that single device or select **Upgrade ALL devices**, to update languages on all telephones in the network.
6. Enter the Quick Edition system administration password and click **Next**.

7. Click **Next** to start the TFTP server.
8. Select the language to use for your telephones from the drop-down list and click **Next**.
9. Confirm the update settings and click **Next**.
10. Click **Start**. The wizard will display a progress bar during the initialization of the upgrade process.
11. Click **Next**.
12. Click **Next** to **Confirm update complete** when the phones return to idle state and the gateways display a green LED.
13. Click Yes in the Warning window.
14. Click **Close** to exit the Language Setup Wizard.

---

## System Localization

You can select country and language settings that affect the language displayed in the interfaces, the language in which Auto Attendant and voicemail prompts are played, the default dial plan, and the ringer and feedback tones generated by the system.

The following regions are supported:

- North America
- United Kingdom
- Italy
- Germany

The following languages are supported:

- American English
- British English
- German
- Italian
- Canadian French

---


## System Language Settings

The language settings procedures will change the system language and the telephone user interface (UI). Refer to [Language Packs](#) on page 27 to change the voicemail and auto attendant prompts, and web interface.

### To change system language settings using a web browser

1. Log in to the **System Options** interface.
2. Select **Localization** in the System Options menu.
3. Click **Change Details** in the **View System Language** field.
4. Select the **System Language** from the drop-down list.
5. Click **Submit**. The language changes on all telephones in the one-X system.

### To change the system language settings using a Quick Edition IP telephone

1. At the telephone, display the **System Options** menu (see [To access system options menu through buttons on the telephone](#) on page 8).
2. Select **System Language** on the **System Options** menu.
3. Click the **Change** softkey.
4. Select the language and click the **Yes** softkey.
5. Click the **OK** softkey to display the previous menu, or press the Phone/Exit (  ) button.

## Changing the System Region Settings

Changing the System Region and resetting the Dial Plan will load the default Dial Plan for your region. The procedure may change the DN range, Auto Attendant range, SIP and PSTN access codes, Operator code, and Emergency contact numbers. Changing the region is a two step process:

- Change the region
- Reset the dial plan

### To change system region settings using a web browser

1. Log in to the **System Options** interface.
2. Select **Localization** in the **System Options** menu.
3. Click **Change Details** in the **View System Region** field.
4. Select the **System Region** from the drop-down list.
5. Click **Submit**.

### To reset the dial plan using a web browser

1. Select **Dial Plan** in the **System Options** menu.
2. Click **Reset Dial Plan**.
3. Click **Commit** on the **Change Dial Plan Settings** screen.

## Creating and Managing Telephone Groups

You can create groups of telephones to help simplify administration tasks and enable the selection of certain telephones. For example, you can configure all of the telephones in a group to ring at the same time given an incoming call—the call can then be answered using any of the telephones in the group. This type of configuration is useful to support customer service groups, technical support groups, help desks, and so on.

When you configure a group of telephones, you have the option to define group-dialing rules for the group. Group dialing rules either enable or prevent users from dialing certain types of calls. Group dialing rules contain dialing patterns, which identify the specific sequences of numbers that group members may dial.

When a call to a group goes unanswered, the group call-forwarding rules that you specify take effect. For example, you can choose to forward the call to a single extension number, where the call can either be answered by a person, or handled by voicemail or the Auto Attendant according to the configuration of that telephone. A call cannot be forwarded from one group to another group.

**Note:**

Groups are created and administered only using the System Options web page of the Multi-Site Provisioning Tool (MPT).

---

## Creating a Group and Adding Members

You can create groups without adding members right away. The settings associated with a group can be edited at any time. A maximum of nine groups, each having a maximum of 10 members, is supported.

### To create a group of telephones

1. Log in to the **System Options** interface.
2. On the **System Options** menu, select **Groups**. The **Groups** list is displayed. Initially, no group information is available because no groups have been defined.
3. In the upper-right corner of the dialog box, click **Add Group**. The **Add Group** dialog box is displayed.
4. In the **Group Name** field, type a meaningful name for the group.
5. In the **Extension** field, type an extension number that can be used to ring all members of the group. The number must be in the system DN range.

**Note:**

You cannot assign the same extension number to more than one group.

6. Click **Submit**. The new group is displayed in the **Groups** list.

### To view or edit group settings

1. Log in to the **System Options** interface.
2. On the **System Options** menu, select **Groups**. The **Groups** list is displayed.
3. In the **Group Name** column, click the name of a group.  
The **View Group Details** dialog box is displayed.

### To add members to an existing group

1. In the **Group Name** column, click the name of a group to which you will add members.
2. With the **View Group Details** dialog box on display, click **Members**.
3. Click **Change Details**. A list of all of the telephones in the network is displayed.
4. Select the corresponding check box in the **Members** column to add a member.
5. Click **Submit**. The selected members are displayed in the **View Group Members** list.

### To delete a group

1. In the **Group Name** column, click the name of a group that you want to delete.
2. With the **View Group Details** dialog box on display, click **Delete Group**.
3. Click **Submit**.

### To change the name of a group

1. In the **Group Name** column, click the name of a group for which you want to change the name.
2. With the **View Group Details** dialog box on display, click **Change Details**.
3. In the **Group Name** dialog box, type a different name for the group.
4. Click **Submit**.

## Defining Group-Dialing Rules

Dialing rules specify numeric dialing patterns that the system can distinguish and process in a certain way (allow or disallow). You can define any number of rules before you assign processing actions to them. If you wish to restrict any calls, you must define a dialing rule.

### To view or create group-dialing rules

1. Log in to the **System Options** interface.
2. On the **System Options** menu, select **Groups**.

3. With the **View Group Details** dialog box on display, click **Dialing Rules**.  
The **View Group Dialing Rules** list is displayed. Initially, the list is empty because no dialing rules have been defined.
4. Click **Add Rule**. The **Edit Group Add Dialing Rules** dialog box is displayed.
5. From the **Add Rule** list, select one of the following options. Your choice determines which calls will be affected by the rule. You can apply rules, depending on region, to:
  - all calls placed to the emergency number; select **Emergency**.
  - all calls placed to an extension in the Corporate directory; select **Extension - ???**.
  - all calls to the PSTN; select **External**.
  - all calls redirected to the designated Operator telephone, select; **Operator**.
  - all calls to a SIP (VoIP) network; select **VoIP E164**.
6. Click **Submit**.

**Note:**

The options available to you may be different. You can specify the dialing patterns to use in a rule and when you create a new dialing pattern, a corresponding option is added to the **Add Rule** list. For more information, see [Specifying Patterns for Dialing Rules on page 38](#).

7. Perform one of the following actions:
  - Select **Allow** to accept incoming and outgoing calls of the selected type.
  - Select **Disallow** to refuse incoming and outgoing calls of the selected type.
8. Click **Submit**.

The new rule is added to the **View Group Dialing Rules** list.

### To delete a custom group-dialing rule

1. With the **View Group Details** dialog box on display, click **Dialing Rules**.  
The **View Group Dialing Rules** list is displayed.
2. Click **Remove Rule**.

**Note:**

The **Remove Rule** link is not available unless custom group-dialing rules have been added to the **View Group Dialing Rules** list.

The **Edit Group Remove Dialing Rule** dialog box is displayed.

3. From the **Delete Rule From Group** list, select the rule that you want to delete.
4. Click **Submit**.



### To edit a custom group-dialing rule

1. With the **View Group Details** dialog box on display, click **Dialing Rules**.  
The **View Group Dialing Rules** dialog box is displayed.
2. Click **Change Details**.

**Note:**

The **Change Details** link is not available unless custom group-dialing rules have been added to the **View Group Dialing Rules** list.

The **Edit Group Dialing Rules** dialog box is displayed.

3. Select **Allow** or **Disallow** beside the rule that you want to edit.
4. Click **Submit**.

---

## Defining Call-forwarding Rules for a Group

Call-forwarding features let you specify rules for handling calls that are routed to a telephone group. You can forward calls to:

- an extension number listed in the Corporate directory,
- the telephone number that you specify (for example, an outside line), or
- the designated operator extension.

**Note:**

If your network has a PSTN gateway and you forward an incoming outside call to a PSTN number, the call will consume two PSTN lines (one incoming, and one outgoing) on the analog gateway while the call is active.

### To define call-forwarding rules for a group

1. With the **View Group Details** dialog box on display, click **Forwarding**.
2. Click **Change Details**. The **Edit Group Forwarding** dialog box is displayed.
3. Select **Enable Forwarding**.
4. From the **Forward After # Rings** list, select the number of rings that must occur before an unanswered call is forwarded.
5. Select one of the following options:
  - To forward the call to an extension number, select **Extension**, and then select the extension number from the adjacent list.
  - To forward the call to the number you specify, select **Dialed** and type the number into the adjacent field. Include the prefix to forward the call to an outside (PSTN) line or to a SIP (VoIP) network.

- To forward the call to the designated operator extension, select **Operator Assist**. The operator extension has to be specified separately (see [Specifying the Designated Operator Extension on page 54](#)).
6. Click **Submit**.

---

## Setting the System Date and Time

If you subscribe to “Caller ID” to obtain date and time information from your service provider, date and time information is delivered to your one-X Quick Edition system from the PSTN. A PSTN gateway uses this information to synchronize its clock.

If the time on the PSTN gateway is two minutes fast or slow compared to PSTN time, the analog gateway resets its clock to match PSTN time and resynchronizes all of the telephones in the network. Afterward, the date and time information associated with internal and external calls is obtained from the one-X Quick Edition network.

**Note:**

Caller ID information does not include the current year. If the factory settings on your telephone do not specify the current year, you must set the current year on your telephone manually. The current year may need to be specified whenever your telephone goes through a power cycle.

The values that you specify in these procedures override the time that a gateway may derive from the PSTN.

**Note:**

The one-X Quick Edition system does not make use of any date and time information that may be obtained from a SIP network. If your one-X Quick Edition system internetworks with a SIP network and not the PSTN, you must set the date and time manually.

### To change the system time using a web browser

1. Log in to the **System Options** interface.
2. On the **System Options** menu, select **Time & Date**.
3. Select **Change Details** in the **View System Time** area.  
The **Edit System Time** dialog box is displayed.
4. In the **Time** fields, enter the system time and select an **AM** or **PM** setting.
5. Click **Submit**.

### To change the system date using a web browser

1. Log in to the **System Options** interface.
2. On the **System Options** menu, select **Time & Date**.
3. Select **Change Details** in the **View System Date** area.
4. In the **Date** fields, enter the system date.
5. Click **Submit**.

### To change the system date using a Quick Edition IP telephone

1. Using any telephone connected to the network, display the **System Options** menu.
2. Select **Date and Time** on the **System Options** menu.
3. Select **Date** on the **Date & Time Options** menu.
4. Enter the date in MM/DD/YYYY format. Pressing the / softkey enters a month-day-year separator character.
5. Select the **Save** softkey.

### To change the system time using a Quick Edition IP telephone

1. Using any telephone connected to the network, display the **System Options** menu.
2. Select **Date and Time** on the **System Options** menu.
3. Select **Time** on the **Date & Time Options** menu.
4. Enter the time in HH:MM (24-hour) format (for example, 15:36). Pressing the : softkey enters an hour-minute separator character.
5. Select the **Save** softkey.

---

## System Gateways



You can access, manage, and upgrade a Quick Edition IP through its web-based administration interface. See *Avaya one-X Quick Edition G10 PSTN Gateway Installation Guide* (Document No. 16-600793) or *Avaya one-X Quick Edition Configuration Guide for G11 Global Analog Gateway* (document number 16-601415) for complete details.

A subset of the configuration options available through the web-based administration interface can be accessed through the **System Options** menu on any Quick Edition IP Telephone on the one-X Quick Edition network. Related procedures are given throughout this chapter where applicable.

### To access global analog gateway options through the web-based administration interface

1. Log in to the **System Options** interface.
2. On the **System Options** menu, select **Gateways**.  
The **Gateways** list appears with information about all of the Quick Edition IPs in the network.
3. In the **Extension** column, click the extension number of the Quick Edition IP that you want  
Configuration options related to the selected Quick Edition IP are displayed in the **View Gateway Details** dialog box.

### To access analog gateway options through buttons on the telephone

1. At the telephone, press the Options (  ) button below and to the right of the display area.
2. Select **Options** on the **Main** menu, or press 1 on the dialpad.
3. Select **System Options** on the **Options** menu, or press the indicated dialpad key.
4. When you are prompted to enter a password, press the keys on the dialpad to enter the administration password.
5. Select the **Done** softkey.  
The **System Options** menu is displayed.
6. Select **Gateways** on the **System Options** menu, or press the indicated dialpad key.
7. Using a Line/Feature (  ) button or numeric key on the dialpad, extension number of the Quick Edition IP that you want to configure.

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## Dial Plan

Avaya one-X Quick Edition uses an internal numbering system, a dial plan, to assign extension numbers to network devices and to enable calls outside the one-X Quick Edition network.

Initially, the system configures itself using the default North America set of extension numbers, numbers to permit access to emergency services, the PSTN (dial 9 followed by the PSTN number) and if available, a SIP (VoIP) network (dial 8 followed by the SIP network number).

When you select another region and reset the dial plan, the extension numbers, emergency, operator, PSTN, and SIP codes will change to the default numbers for the region.

**Note:**

When you change the system region, you must reset the dial plan. See [Changing the System Region Settings](#) on page 29.

Because the dial plan is configurable, you can then specify your choice of access numbers or extension numbers for the following services:

- local telephones—extension numbers can be from one-to-six digits long in the range that you specify
- local Auto Attendant configurations—extension numbers can be from one-to-six digits long in the range that you specify, a range that is within the extension number range
- the PSTN—the default access number can be reassigned
- a SIP service provider network—the default access number can be reassigned
- help during an emergency—a specific number can be programmed
- local extension numbers used to identify analog gateways—specific numbers can be programmed.

A change to the dial plan may mean that some manual configuration is required for the following features:

- Auto Attendants
- Zero Redirect
- call forwarding
- speed dials
- personal directory
- group members (the group DN will be correct)
- SIP Option table
- TTI Option table.

### To change a dial plan using a web browser

1. Log in to the **System Options** interface.
2. On the **System Options** menu, select **Dial Plan**.
3. Click **Change Dial Plan** on the **View Dial Plan Settings** dialog box.
4. Make your changes, keeping the auto attendant range within the extension range.
5. Click **Validate**.

**Note:**

Any change to the dial plan may result in a change to the gateway incoming lines configuration. If you had modified the Forwarding option, it will return to the default, OPERATOR.

## Dialing Rules

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### Specifying Patterns for Dialing Rules

Dialing patterns provide a way for you to describe a specific sequence of key presses that the one-X Quick Edition system can recognize and process in a certain way. These patterns, when added to a group-dialing rule (see [Defining Group-Dialing Rules](#) on page 31), either allow or prevent users from making certain types of calls.

#### To view dialing rule patterns

1. Log in to the **System Options** interface.
2. On the **System Options** menu, select **Dialing Rules**.

#### To create a custom dialing rule pattern

1. Log in to the **System Options** interface.
2. On the **System Options** menu, select **Dialing Rules**.
3. Click **Create Rule**.
4. In the **Rule Description** field, type a description of the rule.
5. In the **Pattern** field, type a rule pattern. The pattern must be a numeric string. The following special characters can be used in the pattern:
  - \* matches any digit and any number of digits to follow
  - ? is a wildcard character that matches any single digit in that specific position

For example, to create a dialing pattern to match a 1 800 number in an area that uses the 9 PSTN-access code, you can type **91800\*** or **91800???????**.

6. Click **Submit**. The new rule is added to the Rules list.

#### To delete a custom dialing rule pattern

1. Log in to the **System Options** interface.
2. On the **System Options** menu, select **Dialing Rules**.
3. Click **Delete Rule**. The **Delete Rule** dialog box is displayed.
4. From the **Delete Rule** drop-down list, select the rule that you want to delete. You cannot delete the default dialing rules.

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# System Networking Options

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## Adjusting Audio Bandwidth Settings

Audio quality may be improved by adjusting the audio bandwidth and/or enabling priority tagging on the one-X Quick Edition network.

The COder/DECoder (CODEC) used to encode the audio path (either G.711 or G.729a) can be selected through the web-based administration interface. By default, the audio path is encoded using the standard G.711 (64 Kbps) CODEC, and G.729a (8 Kbps) is negotiated automatically if required to support calls initiated at the far end.

**Note:**

The G.729a CODEC is also used to store Voicemail data.

You can specify an audio quality setting for the entire network. The high setting (64 Kbps) provides better sound but occupies more audio bandwidth compared to the low setting (8 Kbps).

### To adjust the audio bandwidth

1. Log in to the **System Options** interface.
2. On the **System Options** menu, select **Networking**. The **View SMTP Server** dialog box is displayed.
3. Click **Audio Bandwidth**. The **View Audio Bandwidth** dialog box is displayed.
4. Click **Change Details**. The **Edit Audio Bandwidth** dialog box is displayed.
5. From the **Audio Bandwidth** drop-down list, select one of the following options:
  - **High**—Sound is digitally encoded using a high-quality format that consumes approximately 64 kilobits per second of bandwidth on the network.
  - **Low**—Uses a compressed format for sound that occupies approximately 8 kilobits per second of bandwidth on the network.
6. Select **Submit**.

## Adjusting VLAN Settings

### Enabling Priority (QoS) Tagging

Depending on the configuration of your LAN and the amount of traffic being processed, you may configure Virtual LANs (VLANs) on your LAN routers or switches to implement more than one logical network on the same physical network segment and/or improve the Quality of Service (QoS) throughout your network. Priority tagging improves QoS when one-X Quick Edition traffic is routed over a network segment that implements VLANs.

To support one-X Quick Edition priority tagging, the Ethernet devices to which your one-X Quick Edition devices are connected must support IEEE 802.1pQ frames and be configured properly to permit the forwarding of one-X Quick Edition voice and data traffic to one-X Quick Edition devices.

Avaya one-X Quick Edition devices support IEEE 802.1p (priority value tagging), within the framework of the IEEE 802.1Q *Virtual Bridged Local Area Networks* standard. The feature allows you to assign priority levels to one-X Quick Edition voice and data traffic in order to ensure QoS at OSI Layer 2. Specifying a priority such as 5 for voice traffic and 3 for data ensures that one-X Quick Edition voice traffic has priority over data. Your LAN administrator must determine the required settings.

You can use the web-based administration interface to assign priority settings. Priority assignments can be applied to voice traffic leaving telephones through their LAN ports, and to data entering telephones through their PC ports. The settings are system-wide, and affect all devices making up the one-X Quick Edition network.

Initially, the priority tagging of voice traffic is disabled. You can enable the generation of priority tags for voice traffic and set a priority. Similarly, telephones initially pass data received on their PC ports to the LAN without modifying priority tags. You can disable the generation of priority tags for data sent to computers through the telephones' PC ports (for example, for computers that do not support priority tagging), or enable priority tagging of the data, which causes the telephones to replace the priority tags received on their PC ports with the value that you specify.

**Note:**

Priority tagging on teleworker client telephones is disabled. Priority tagging can be configured only when teleworker mode is disabled.

If you decide to use one-X Quick Edition priority tagging, you do not have to configure the ports on supporting Ethernet routers or switches in any special way. When priority tagging is enabled, the VLAN identifier is set to 0, which means that one-X Quick Edition devices will accept any VLAN identifier that is currently assigned to the ports on your LAN routers and switches. Simply verify that all one-X Quick Edition devices are included in the same VLAN.



**CAUTION:**

Before you enable priority tagging for the PC port on a Quick Edition IP telephone, verify that the device connected to the PC port supports 802.1pQ tagging.

A computer may be connected to the PC port to enable priority tag settings. If the network interface card on the computer does not support 802.1pQ tagging, make sure that the PC port setting is explicitly set to “disabled” before enabling priority tag settings for the LAN port. We strongly recommend that you do not change the PC port setting from “disabled” to “enabled” or “pass-through,” as this would cause the computer to lose its connection to the web-based administration interface on the telephone. If this happens, you will only be able to recover the lost connection by:

1. Replacing the network interface card in the PC with one that supports 802.1pQ tagging.
2. Connecting the computer to a port on the LAN switch that has connectivity to the Quick Edition IP telephone and at the same time untags any outgoing frames.

### To configure priority tagging for a one-X Quick Edition network

1. Log in to the **System Options** interface.
2. On the **System Options** menu, select **Networking**.
3. Click **VLAN Settings**. The **View VLAN Tagging** dialog box is displayed.
4. Click **Change Details**. The **Edit VLAN Settings** dialog box is displayed.
5. To enable the priority tagging of voice traffic:
  - Select **Enabled** from the **Audio Tagging** drop-down list.
  - Select the priority level to apply to voice traffic from the **Audio Priority** drop-down list. Choose a value in the 0 to 7 range, where 7 assigns the highest priority.
6. To enable the priority tagging of data traffic select from the **Data Tagging** drop-down list:
  - Select **Enabled** to enable the priority tagging of data traffic received on the telephone PC ports and routed towards LAN by the telephones. The tagging associated with data traffic received on LAN port and sent out through the PC port is not changed.
  - Select **Pass Through** to have the telephones leave any priority tags for data traffic received or sent through their PC ports unchanged.
  - Select **Disabled** to explicitly disable the priority tagging of data traffic by the telephones—data traffic sent through their PC ports is not tagged. If the data traffic received on their PC ports is tagged, the priority value is changed to 0. If voice tagging is also disabled, the data traffic sent out to the LAN is not tagged.
7. If you enabled the priority tagging of data traffic, from the **Data Priority** list, select the priority level to apply to data traffic. Choose a value in the 0 to 7 range, where 7 assigns the highest priority.

### 8. Click **Submit**.

The device into which you are logged updates its own priority settings and verifies that it can still communicate with the administration computer and all other devices making up the network.

When communications remain normal, the new priority settings are downloaded to all other devices making up the network. If some of the devices can no longer be reached (for example, because the switches connecting some or all of the one-X Quick Edition devices do not support IEEE 802.1p priority tagging), the changes are cancelled and a status message is displayed.

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## SMTP

### Supporting E-mail Notification of Voicemail

When enabled, the email-notification-of-voicemail feature sends e-mail messages to alert users whenever new voicemail messages addressed to them are received. The e-mail message contains call header information, including the caller name (if available), caller number, and the time and length of the call. The e-mail message also contains an attached audio file having a recording of the voicemail message.

**Note:**

See [Email Notification of Voicemail](#) on page 23 for information about the audio file format.

The information associated with the call is sent to a Simple Mail Transfer Protocol (SMTP) recipient. Specifically, SMTP is used to send e-mail messages from voicemail on a telephone to an SMTP server, which in turn forwards the e-mail messages to the specified user's e-mail address.

There are two parts to configuring the email-notification-of-voicemail feature:

1. Enable SMTP on the one-X Quick Edition network, which enables a telephone to forward e-mail messages from voicemail on the telephone to an SMTP server. You can specify the IP address of the SMTP server using a web browser or buttons on a telephone as described in the procedures below.

By default, the IP address of the SMTP server host is set to 0.0.0.0, and the TCP/IP port on which SMTP communications is assumed to take place is port 25. In most cases, only the IP address has to be changed.

2. Enable the feature and specify the e-mail address of the person to whom notification will be sent. This can only be done through the web-based **User Options** interface as described in the procedure below. The audio component of the voicemail message is sent as an audio file attachment when you enable the feature. To play the audio file attachment, see [Playing an Audio File Attachment](#) on page 80.

Before you begin, obtain the host IP address and port number of the SMTP server. If the SMTP server does not reside in the one-X Quick Edition network address space, the one-X Quick Edition network must have a route to the SMTP server. You must also obtain the e-mail address to which e-mail notifications will be sent.

### To enable SMTP using a web browser

1. Log in to the **System Options** interface (see [To access options using the web-based administration interface](#) on page 7).
2. On the **System Options** menu, select **Networking**. The **View SMTP Server** dialog box is displayed.
3. Click **Change Details**. The **Edit SMTP Server** dialog box is displayed.
4. Select **Enable SMTP**.
5. In the **Server Address** field, type the IP address of the SMTP server host.
6. If your network uses a different IP port number for SMTP communications, specify the port number in the **Server Port** field.
7. Click **Submit**.

### To enable SMTP through telephone buttons

1. Using any telephone connected to the network, display the **System Options** menu.
2. Select **Network Options** on the **System Options** menu.
3. Select **SMTP Settings** on the **Network Options** menu.
4. Select the **On** softkey to enable SMTP on the telephone.
5. Select the **Chg** softkey.
6. Enter the IP address of the SMTP server host. Select the **.** softkey to add separator characters between the digits.
7. Select the **Next** softkey.
8. If your network uses a different IP port number for SMTP communications, specify the port number.
9. Select the **Next** softkey.
10. Select the **Save** softkey.

### To enable notification and specify the e-mail address of the SMTP recipient

1. If you are logged in to the web-based system administration interface, log out.
2. Using the extension number of the telephone that receives voicemail messages on behalf of the user and the password needed to access user options on that telephone, log in to the web-based **User Options** interface.
3. On the **User Options** menu, click **Voice Mail**.

4. In the **SMTP** area, click **Change**.
5. In the **To Address** field, type the e-mail address of the person to whom notifications will be sent (for example, `user1@mycompany.com`).
6. In the **From Address** field, type an e-mail address that the telephone can use to place in the **From** field of the e-mail header. This value has to look like an e-mail address, but it does not have to be a real e-mail address (for example, `email@user1phone.com` is acceptable). Choose a value that is meaningful to the party who will be receiving e-mail messages from the telephone.
7. Select **Enable Notification**.
8. Click **Submit**.
9. Log out.

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## Service Provider

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### Configuring Interoperability with a SIP (VoIP) Service Provider

Certified SIP (VoIP) service provider options can be configured through the web-based administration interface. This feature supports internetworking with the Avaya Communication Manager (CM) gateway and a SIP Enablement Services (SES) server. For example, a CM S8300 media server module installed in a G700 media gateway is capable of running SES-edge and/or home server applications.

**Note:**

To configure SES/CM to support a one-X Quick Edition network, see *Configuring SIP Trunking to Quick Edition on CCS/SES and CM* (Document No. TSS-000001). To configure a one-X Quick Edition network to operate with an SES/CM system, see *Configuring a one-X Quick Edition Network to Interoperate with CCS/SES and CM* (Document No. TSS-000002). Both documents can be downloaded from the Avaya Technical Support web site, <http://support.avaya.com/QuickEdition>.

To support SIP interoperability, the telephones on your company network may be connected to the service provider's network directly, or through a Network Address Translation (NAT) device, as shown in [Figure 2: SIP Service Provider Configuration](#) on page 11. When the one-X Quick Edition network is connected through a NAT device, a Session Border Controller (SBC) must be present on the service provider's SIP network. Your service provider will configure the SBC properly based on the information you provide about your network.

**Note:**

*The SIP proxy server must support INVITES, Re-INVITES, basic response codes, and SDP session media negotiations throughout the life cycle of the call.*

Consult your service provider to obtain information about the service provider SIP network. Your service provider may give you information that is similar to all or some of the information shown below:

<b>Proxy:</b>	sip.sipexample.com	Port: 5060 <sup>1</sup>
<b>Registrar:</b>	sip.sipexample.com	Port: 5060
<b>Session Border Controller:</b>	outproxyhost.sipexample.com	
<b>Domain:</b>	sipexample.com	
<b>Realm:</b>	auth_domain.sipexample.com	

1. If the SIP servers in the service provider's network have backup configurations known as "redundancy" based on a Domain Name System (DNS) algorithm, DNS service (DNS SRV) records will be used to specify the location of a specific SIP server. In this case, you must set the proxy port number to 0.

In addition, your service provider determines which (E.164) telephone numbers to assign to your network. These numbers are needed to support communications between your telephones and the SIP devices on the service provider's network.

Service providers require you to assign "SIP identities" (usernames and passwords) to the devices on your company network so that calls coming in through the SIP network can be mapped and delivered to a specific telephone and/or handled by the system-wide Auto Attendant feature. You can assign SIP identities through Service Provider options. The one-X Quick Edition system makes each identity that you assign available to any telephone in the network.

Consult your service provider to obtain SIP identities (usernames and passwords) for your telephones. The user name is typically a telephone number. For example, if you have three telephones that have been configured to receive calls directly (through a direct inward line), your service provider may send you the following information (one SIP identity and password per telephone):

<b>SIP identity 1:</b>	1715557654	Password: 12000
<b>SIP identity 2:</b>	1715557655	Password: 12001
<b>SIP identity 3:</b>	1715557656	Password: 12002

Alternatively, if one person receives all calls to the one-X Quick Edition network and redirects the calls to the individual telephones manually, your service provider would send you a single SIP identifier:

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<b>SIP identity:</b>	1715557600	Password: 12700
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SIP identities and passwords may be mapped to individual telephone extensions or group extensions. When an individual telephone is assigned a SIP identity, the telephone registers with the SIP proxy server on the service provider's network in order to handle both incoming and outgoing calls. When a SIP identity is assigned to a group, one of the telephones in the group registers with the SIP proxy server on the service provider's network. In both cases, the registration process is done automatically.

As an option, the system-wide Auto Attendant feature can be assigned to a SIP identity. When you assign an Auto Attendant configuration to a SIP identity, incoming calls are routed to the extension number associated with the Auto Attendant configuration.

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## Planning Your Configuration

After you receive the required service provider configuration information and SIP identities from your service provider, you can configure the settings you need to support SIP interoperability. There are two high-level steps to configuring the network:

1. Define the SIP network configuration. See [To add a service provider configuration on page 49](#).
2. Assign SIP identities. The assignments will depend on how your office is set up:
  - Are all calls answered by a receptionist and redirected manually? If so, see [Case A: External Incoming Calls are Answered by a Receptionist on page 47](#).
  - Does everyone have a direct line? If so, see [Case B: Everyone has a Direct Line on page 47](#).
  - Do you have a sales or support group that takes calls on a dedicated line whose telephone number differs from that used to call the receptionist? If so, see [Case C: Incoming Calls are Routed to a Group on page 48](#).
  - Should all incoming external calls be answered by the Auto Attendant feature, which prompts callers to dial an extension number? If so, see [Case D: Incoming Calls are Handled by the Auto Attendant Feature](#) on page 48.

## Case A: External Incoming Calls are Answered by a Receptionist

If external calls to your company arrive on a single line to a receptionist who answers and redirects all incoming calls manually, you need to configure one SIP identity that corresponds to the receptionist's extension number (see [To assign a SIP identity on page 51](#)). The receptionist's telephone would be responsible for handling all incoming calls.

To enable all telephones connected to the network to make outgoing calls, you assign the same SIP identity to those telephones (see [To assign a SIP identity on page 51](#)). To make configuring outgoing calls easy, the "global" option lets you select all of the telephones on the network.

In summary, you would specify information similar to the following values when you create the SIP identity, where extension number 200 belongs to the receptionist, and the **Global** setting refers to all of the telephones connected to the network:

<b>SIP Identity:</b>	1715557600
<b>Password:</b>	12700
<b>Member (Outgoing Calls):</b>	Global
<b>Member (Incoming Calls):</b>	200

After you configure the telephones to interoperate with the service provider's SIP network, the receptionist's telephone will register with the SIP registration server to receive all incoming calls, and one of the other telephones on the network will register to forward outgoing calls on behalf of all other telephones on the network.

Registration takes place automatically on a continual basis. You can specify how often telephones register with the SIP registration server (for example, every 3600 seconds).

## Case B: Everyone has a Direct Line

When everyone has a direct line, you need to assign a unique SIP identity to each telephone (see [To assign a SIP identity on page 51](#)). In this case, each telephone handles its own incoming and outgoing calls.

To configure SIP interoperability in this case, you would specify information similar to the following when you create the SIP identities:

<b>SIP identity:</b>	1715557654	1715557655	1715557656
<b>Password:</b>	12000	12001	12002

<b>Member (Outgoing Calls):</b>	201	202	203
<b>Member (Incoming Calls):</b>	201	202	203

After you configure the telephones to interoperate with the service provider's SIP network, each telephone will register with the SIP registration server to receive its own incoming and outgoing calls. Registration takes place automatically on a continual basis. You can specify how often telephones register with the SIP registration server (for example, every 3600 seconds).

### Case C: Incoming Calls are Routed to a Group

It is possible to route external incoming calls to a group of telephones based on the dialed number. First, you must create a group that contains the extension numbers of those telephones (see [Creating and Managing Telephone Groups](#) on page 30). Next, configure one SIP identity that corresponds to the group extension number (see [To assign a SIP identity on page 51](#)). When you specify the extension number of the group to handle incoming and outgoing calls, one of the telephones in the group registers on behalf of the group.

For example, the following configuration causes all incoming and outgoing calls to be routed to the group extension 405:

<b>SIP Identity:</b>	1715557600
<b>Password:</b>	12700
<b>Member (Outgoing Calls):</b>	405
<b>Member (Incoming Calls):</b>	405

After you configure the telephone group to interoperate with the service provider's SIP network, one of the telephones in the group will register with the SIP registration server to receive incoming and outgoing calls. Registration takes place automatically on a continual basis. You can specify how often telephones register with the SIP registration server (for example, every 3600 seconds).

### Case D: Incoming Calls are Handled by the Auto Attendant Feature

If you want the Auto Attendant feature to answer all calls from an external network automatically, you need to configure one SIP identity that corresponds to the entire one-X Quick Edition network in general (see [To assign a SIP identity on page 51](#)). To enable an Auto Attendant configuration to handle all incoming calls, you specify the “global” option for the incoming extension.



To enable all telephones connected to the network to make outgoing calls, you assign the same SIP identity to those telephones (see [To assign a SIP identity on page 51](#)). To make configuring outgoing calls easy, the “global” option lets you select all of the telephones on the network.

In summary, you would specify information similar to the following values when you create the SIP identity, where the **Global** setting refers to the Auto Attendant configuration assigned to the incoming PSTN line:

<b>SIP Identity:</b>	1715557600
<b>Password:</b>	12700
<b>Member (Outgoing Calls):</b>	Global
<b>Member (Incoming Calls):</b>	Global

After you configure the telephones to interoperate with the service provider's SIP network, one of the telephones on the network will register to receive all incoming calls, and another telephone will register to forward outgoing calls on behalf of all other telephones on the network.

Registration takes place automatically on a continual basis. You can specify how often telephones register with the SES server (for example, every 3600 seconds).

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## Configuring SIP Interoperability

Configuring SIP interoperability involves four steps:

1. Obtaining information from your service provider, such as SIP identities, the fully qualified domain names, or IP addresses of the SIP proxy server, registration server, and SBC (if an SBC is used), the domain name of the SIP network and any associated realm information.
2. Creating a service provider configuration. See [To view or edit service provider configurations on page 50](#).
3. If required to support Case C (see [Case C: Incoming Calls are Routed to a Group on page 48](#)), configuring a group of telephones. See [Creating and Managing Telephone Groups](#) on page 30.
4. Assigning SIP identities. See [To assign a SIP identity on page 51](#).

### To add a service provider configuration

1. Log in to the **System Options** interface.
2. On the **System Options** menu, select **Service Provider**.

The **Configurations** list is displayed. Initially, the list is empty because no configurations have been created.

3. Click **Add Configuration**. The **Add Service Provider Configuration** dialog box is displayed, with the following fields:
  - In the **Domain Name** field, type the service provider domain name to use for registration purposes (for example, `sipexample.com`). This value is included in the “From” header of outgoing SIP calls.
  - In the **Proxy Host** field, type the IP address or fully qualified domain name of the proxy server through which outgoing P2P calls will be routed (for example, `sip.sipexample.com`).
  - In the **Proxy Port** field, type the port number through which the proxy server listens for SIP communications (for example, `5060`). Set the **Proxy Port** value to 0 if the service provider network contains redundant SIP servers where redundancy is based on a DNS algorithm.
  - In the **Registrar Host** field, type the IP address or fully qualified domain name of the registration server that your telephones have to contact in order to receive incoming VoIP calls (for example, `sip.sipexample.com`). If your service provider did not give you any information about a registration server, re-type (in the **Registrar Host** field) the same value that you entered for the **Proxy Host** field.
  - In the **Registrar Port** field, type the port number through which the registration server listens for SIP communications (for example, `5060`).
  - If your service provider uses a separate outbound proxy server or session border controller to manage calls leaving the SIP network, in the **Outbound Proxy Host** field, type the IP address or fully qualified domain name of the SBC host.
  - In the **Outbound Proxy Port** field, type the port number through which the outbound proxy server or SBC listens for SIP communications. Set the **Outbound Proxy Port** value to 0 if the service provider network contains redundant SIP servers where redundancy is based on a DNS algorithm.
  - In the **Realm** field, type the realm information that is used to locate authorization credentials (for example, `auth_domain.sipexample.com`).
  - In the **Register Expiry Time** field, type the amount of time that telephones wait between sending updates to refresh registration information (for example, `3600`). The value is measured in seconds.
4. Click **Submit**.

### To view or edit service provider configurations

1. Log in to the **System Options** interface.
2. On the **System Options** menu, select **Service Provider**.

The **Configurations** list is displayed. When a service provider configuration exists, the domain name associated with the configuration is displayed in the list.
3. Click a domain name to access the associated service provider configuration settings.

The **View Configuration Details** dialog box is displayed.

4. If you want to edit the information, click **Change Details**. See above for field descriptions in the procedure, [To add a service provider configuration](#).
5. When you are finished, click **Submit**.

### To delete a service provider configuration

1. Log in to the **System Options** interface.
2. On the **System Options** menu, select **Service Provider**.
3. Click the name of the configuration that you want to delete, in the **Configurations** list.
4. Click **Delete Configuration**. The **Delete Configuration** dialog box is displayed.
5. Click **Submit**.
6. When you are prompted to confirm the deletion, click **Confirm**.

### To assign a SIP identity

This procedure requires that you have already created a service provider configuration.

1. Log in to the **System Options** interface.
2. On the **System Options** menu, select **Service Provider**.  
The **Configurations** list is displayed.
3. In the Domain column, click the name of the service provider configuration.  
The **View Configuration Details** dialog box is displayed.
4. Click **Identities**.  
The **Identities** dialog box is displayed. Initially, the list is empty because no SIP identities have been assigned. When SIP identities have been assigned, a list of SIP identities is displayed, along with the extension number(s) of the telephone(s) that will be handling incoming and outgoing calls.
5. Click **Add Identity**. The **Add Configuration Identity** dialog box is displayed.
  - In the **Identity** field, type the user name that your service provider assigned (for example, 1715557600).
  - In the **Password** field, type the password that is associated with the user name (for example, 12700).
  - From the **Incoming Extension** list, select the telephone extension or group extension that will be handling incoming calls for the specified identity. If you want the Auto Attendant feature to answer incoming calls instead, select **Global**.
  - From the **Outgoing Extension** list, select the telephone extension or group extension that will be allowed to make outgoing calls using the specified identity. If you are configuring a system in which a receptionist redirects incoming external calls manually or the Auto Attendant feature answers incoming external calls automatically, select the **Global** option.

## Chapter 2: Configuring the System

- If you want to use a custom Auto Attendant configuration instead of the default configuration, select the name of the custom configuration from the **AA Script** list.
6. Click **Submit**.

### To view or edit a SIP identity assignment

1. Log in to the **System Options** interface.
2. On the **System Options** menu, select **Service Provider**.
3. Click **Identities**. The **Identities** dialog box is displayed.
4. In the **Identity** column, click a user name to access the associated identity assignment.

### To delete a SIP identity

1. Log in to the **System Options** interface
2. On the **System Options** menu, select **Service Provider**.  
The **Configurations** list is displayed.
3. Click **Identities**.  
The **Identities** dialog box is displayed.
4. In the **Identity** column, click the user name of the identity that you want to delete.  
The **View Identity Details** dialog box is displayed.
5. Click **Delete Identity**. The **Delete Configuration Identity** dialog box is displayed.
6. Click **Submit**.
7. When you are prompted to confirm the deletion, click **Confirm**.

## Placing a Call to a SIP Network

To place a call from a telephone to the SIP network, a user must dial the number including a prefix (8 by default). For example, a user dialing 85551002 will result in signalling being routed to the proxy host that resides on the service provider's SIP network. If the fully qualified domain name of the proxy host is proxy.example.com, a destination address of 5551002@proxy.example.com is used to route the call to the SIP network.

The DNS server on the telephone converts the domain name to a routable IP address. If required, you can specify the host IP address of the SIP proxy/registrar server instead of a domain name when you create the service provider configuration.

---

## Configuring the Auto Attendant Feature

Your one-X Quick Edition network is equipped with a default Auto Attendant that answers calls to the one-X Quick Edition network from the PSTN or a SIP (VoIP) network. A different Auto Attendant configuration can be assigned to each PSTN gateway or SIP identity.

**Tip:**

The default Auto Attendant can be started from any telephone in the one-X Quick Edition network by dialing the lowest number in the auto attendant extension range (default is 500); however, to reach someone who has a telephone on the network, it may be faster to dial that person's extension number through the Corporate directory.

---

## How the Auto Attendant Feature Works

The Auto Attendant answers calls and instructs callers to dial the extension number, enter the name of the person, or press 0 to reach the operator.

A caller who does not know the extension number can use the dialpad keys to spell a name and have the call transferred. For example, to reach someone named Mary Young, a caller would press 1 to access the Corporate directory, and then press the keys on the dialpad to enter the last name (96864) followed by the # key.

If the caller does not respond to Auto Attendant prompts, or dials zero, the call is transferred to the designated operator. Callers who use rotary or pulse telephones are redirected to the designated operator because rotary/pulse telephones are unable to generate tones. The system interprets this lack of tones as "no response".

The Auto Attendant can play the default greeting all the time, or you can record two custom greetings and manually switch the recording to play at different times. For example, you can use one greeting during the day when you are open for business, and one when the office is closed.

You can define multiple Auto Attendant configurations per one-X Quick Edition network. You can use the Auto Attendant configuration that is supplied with one-X Quick Edition, or you can create and enable one or more custom Auto Attendant configurations.

## Configuring the Auto Attendant Feature to Answer Calls from the PSTN

The Auto Attendant feature can answer as many calls from the PSTN simultaneously as there are PSTN lines connected to the one-X Quick Edition network.

One Auto Attendant configuration per PSTN gateway can be enabled. Initially, incoming calls on all PSTN lines connected to the PSTN gateway are routed to the default Auto Attendant configuration at extension 500 (the default).

There may be reasons for disabling the Auto Attendant feature on one or more PSTN lines:

- to enable a fax machine to answer and place facsimile transmissions,
- to make a PSTN line a “direct inward line,” in which case calls arriving on that line are routed directly to the specified extension instead of the Auto Attendant feature, and/or
- to make a PSTN line a “private outgoing line,” which forwards outgoing calls from the specified extension to the PSTN line.

For more information and procedures related to disabling the Auto Attendant feature on one or more PSTN lines, see the *Avaya one-X Quick Edition Configuration Guide for G11 Global Analog Gateways* (Document No. 16-601415) or *Avaya one-X Quick Edition G10 PSTN Gateway Installation Guide* (Document No. 16-600793). To assign an Auto Attendant configuration to an analog gateway, see [To view or edit an Auto Attendant configuration using a Quick Edition IP telephone](#) on page 56.

You must assign an Auto Attendant configuration to a PSTN gateway using the web-based System Options interface. Log in and click Gateways, select the extension, and click Lines. Click Change Details and select the auto attendant from the drop-down list.

## Configuring the Auto Attendant Feature to Answer Calls from a SIP Network

When your one-X Quick Edition network can be accessed by callers on a SIP network, you can arrange to invoke the Auto Attendant feature whenever the number associated with a SIP identity is called. In this case, the number of calls the Auto Attendant feature can answer depends on how many SIP identities have been configured:

- When a call from the SIP network is forwarded to a SIP identity that belongs to a single telephone, a maximum of three or four calls from the SIP network can be answered simultaneously, depending on the maximum number of active calls the telephone can handle (three for 4610 SW IP telephones, and four for 4621 SW IP telephones).
- When a SIP identity is associated with a group of telephones, the capacity increases and is directly proportional to the number of telephones in the group, and the number of active calls each telephone can handle.

To configure the Auto Attendant feature to answer calls from a SIP network, see [To assign a SIP identity on page 51](#).

---

## Specifying the Designated Operator Extension

You can choose the telephone that you want to designate as the operator extension. Initially, the telephone at extension number 200 receives these redirected calls. When a call is forwarded to the operator extension but remains unanswered, the call-forwarding rules on that telephone ultimately determine how the call will be handled.

### To specify the operator extension using a web browser

1. Log in to the **System Options** interface.
2. In the **Set Management** list, click **Operator**. The **View Operator Extension** dialog box is displayed.
3. Click **Change Details**. The **Edit Operator Extension** dialog box is displayed.
4. From the **Operator Extension** list, select the extension.
5. Click **Submit**.

### To specify the operator extension using a Quick Edition IP telephone

1. Using any telephone connected to the network, display the **System Options** menu.
2. Select **Network Options** on the **System Options** menu.
3. Select **Operator Extension** on the **Network Options** menu.
4. Select the **Chg** softkey.
5. Perform one of the following steps:
  - If you know which extension number to use, enter the number and then select the **Next** softkey.
  - To choose an extension number from the Corporate directory, select the **FrDir** softkey and then select the entry.
6. Select the **Save** softkey.

---

## Recording Custom Greetings

You can record two different greetings for each Auto Attendant configuration.

### To record a custom greeting

1. Using any telephone connected to the network, display the **System Options** menu.
2. Select **Auto Attendant** on the **System Options** menu, or press the indicated dialpad key.
3. Select **Custom Greeting 1** or **Custom Greeting 2** on the **Auto Attendants** menu.
4. When you are prompted to record the custom greeting, lift the handset and prepare yourself to make the recording.
5. Select the **Rec** softkey, and speak the greeting clearly into the microphone.
6. When you are finished, select the **Stop** softkey.

7. Perform one of the following actions:
  - To listen to your recording, select **Play** on the **AA Prompt** menu.
  - If you are satisfied with the recording, select the **Save** softkey followed by the **Ok** softkey.
  - If necessary, select the **Rec** softkey to re-record the greeting.

---

## Viewing and Editing Auto Attendant Configurations

You can view and/or edit the default Auto Attendant configuration and/or create custom Auto Attendant configurations.

You cannot change the extension number that is associated with an existing Auto Attendant configuration—instead, delete the configuration and create another one having the required extension number.

### To view or edit an Auto Attendant configuration using a web browser

1. Log in to the **System Options** interface.
2. On the **System Options** menu, select **Auto Attendant**.

The **Auto Attendant Extensions** list is displayed. The default Auto Attendant configuration is the only one displayed when no custom Auto Attendant configurations have been created.
3. In the **Extension** column, click the extension number of the Auto Attendant configuration that you want to view or edit. The **View Extension Details** dialog box is displayed.
4. If you want to change the name of the configuration or the greeting that is played, or the language of the recording, click **Change Details**.

The **Edit Extension Details** dialog box is displayed.
5. To change the name of the configuration, type a new name in the **Name** field.
6. To change the greeting that is played when the Auto Attendant is invoked, select the greeting from the **Prompt** list.

**Note:**

You must record a custom greeting before it can be played. See page [55](#).

7. Click **Submit**.

### To view or edit an Auto Attendant configuration using a Quick Edition IP telephone

1. Using any telephone connected to the network, display the **System Options** menu.
2. Select **Auto Attendant** or press the indicated dialpad key.



3. Select **Auto Attendant** on the **Auto Attendants** menu, or press the indicated dialpad key.  
All Auto Attendant configurations are displayed.
4. Select the Line/Feature (▶) button beside the configuration that you want to view, or press the corresponding key on the dialpad.
5. To change the name of the selected configuration:
  - Select **Name** on the **AA Details** menu, or press the indicated dialpad key.
  - Using the keys on the dialpad and the softkeys, enter a new name for the Auto Attendant configuration (see Step 7 in [To add an Auto Attendant configuration using a Quick Edition IP telephone](#) on page 58 for detailed instructions for entering a name).
  - Select the **Next** softkey.
6. To change the prompt that is played when the selected configuration is invoked:
  - Select **Prompt** on the **AA Details** menu, or press the indicated dialpad key, and then perform one of the following actions:
    - To select the default greeting, select **Default** on the **Select AA Prompt** menu, or press the indicated dialpad key.
    - If you recorded a custom greeting or intend to record a custom greeting (see [To record a custom greeting on page 55](#)), press the indicated dialpad key to select the appropriate greeting.
7. To assign or remove an association with a PSTN gateway:
  - Select **Gateway** on the **AA Details** menu, or press the indicated dialpad key.  
The **Select AA Gateways** menu displays the extension number of all PSTN gateways connected to the one-X Quick Edition network. If a PSTN gateway is associated with the selected Auto Attendant configuration, its extension number is displayed with a check symbol. Extension numbers displayed with an **X** symbol are not associated with the selected Auto Attendant configuration.
  - Select the Line/Feature (▶) button beside the extension number of the PSTN gateway that you want to associate with the Auto Attendant configuration, or press the corresponding key on the dialpad.

**Note:**

If the PSTN gateway that you selected was associated with a different Auto Attendant configuration, that association is removed automatically.

---

## Creating a Custom Auto Attendant Configuration

You can create a custom Auto Attendant configuration using the web-based administration interface or a telephone.

### To create a custom Auto Attendant configuration using a web browser

1. Log in to the **System Options** interface.
2. On the **System Options** menu, select **Auto Attendant**.
3. Click **Add Auto Attendant Extension**. The **Add New Auto Attendant Extension** dialog box is displayed.
4. In the **Extension** field, type the extension number that will be used to invoke the Auto Attendant configuration. The number must be in the system extension range.
5. In the **Name** field, type a name for the configuration.
6. From the **Prompt** list, select the greeting to play when the Auto Attendant configuration is invoked.







**Tip:**

You must record a custom greeting before it can be played. For more information, see [To record a custom greeting on page 55](#).

7. Click **Create**. The new configuration is displayed in the **Auto Attendant Extensions** list.

### To add an Auto Attendant configuration using a Quick Edition IP telephone

1. Using any telephone connected to the network, display the **System Options** menu.
2. Select **Auto Attendant** on the **System Options** menu, or press the indicated dialpad key.
3. Select **Auto Attendant** on the **Auto Attendants** menu, or press the indicated dialpad key.
4. Select the **Add** softkey.
5. Using the keys on the dialpad, enter an unused extension number in the auto attendant range.
6. Select the **Next** softkey.
7. Using the keys on the dialpad and the softkeys, enter a name for the Auto Attendant configuration:
  - Press the dialpad key with the corresponding letter on it—press once to enter the first letter, twice for the second letter, three times for the third letter, and four times for the fourth letter. For example, to type “R”, press the dialpad key “7” three times. To enter the next character, wait for the cursor to move to the right automatically or press the Page Right (  ) button.
  - To move the cursor to the left without deleting a character, press the Page Left (  ) button.
  - To move the cursor to the right without deleting a character, press the Page Right (  ) button.
  - To add a space to the end of a line, press the Page Right (  ) button.

- To change a character to upper- or lower-case, select the **Case** softkey. The first character in a line and the first character after a space are capitalized automatically.
  - The special characters . , ' & - and @ are entered by pressing the 1 dialpad key.
8. Select the **Next** softkey.
  9. Perform one of the following actions:
    - To select the default greeting, select **Default** on the **Select AA Prompt** menu, or press the indicated dialpad key.
    - If you recorded a custom greeting or intend to record a custom greeting (see [To record a custom greeting on page 55](#)), press the indicated dialpad key to select the appropriate greeting.

A summary of the configuration settings is displayed.
  10. Select the **Save** softkey.

---

## Deleting an Auto Attendant Configuration

You cannot delete an Auto Attendant configuration when it is assigned to a PSTN gateway—you must first assign a different Auto Attendant configuration to the gateway. You cannot delete the default Auto Attendant configuration, which is assigned to extension number 500, by default.

**Note:**

To select an Auto Attendant configuration for a PSTN gateway, see [Configuring the Auto Attendant Feature to Answer Calls from the PSTN on page 53](#)).

### To delete a custom Auto Attendant configuration using a web browser

1. Log in to the **System Options** interface.
2. On the **System Options** menu, select **Auto Attendant**.
3. In the **Extension** column, click the extension number of the Auto Attendant configuration that you want to delete. The **View Extension Details** dialog box is displayed.
4. Click **Delete Extension**.

**Note:**

The **Delete Extension** link is not available when the Auto Attendant configuration is assigned to a PSTN gateway. You cannot delete an Auto Attendant configuration when it is associated with a PSTN gateway.

5. Click **Submit**.

### To delete a custom Auto Attendant configuration using a Quick Edition IP telephone

1. Using any telephone connected to the network, display the **System Options** menu (see [To access system options menu through buttons on the telephone](#) on page 8).
2. Select **Auto Attendant** on the **System Options** menu, or press the indicated dialpad key.
3. Select **Auto Attendant** on the **Auto Attendants** menu, or press the indicated dialpad key.  
All Auto Attendant configurations are displayed.
4. Select the Line/Feature (▶) button beside the configuration that you want to delete, or press the corresponding key on the dialpad.  
The **AA Details** menu is displayed.
5. Select the **Del** softkey.
6. When you are prompted to confirm deletion, select the **Yes** softkey.  
A confirmation message is displayed.

### To bypass the default Auto Attendant functionality on a PSTN line

When you disable incoming lines in gateway configuration, the Auto Attendant will not answer.

1. Access the gateway configuration (see [To access options using the web-based administration interface](#) on page 7).  
The **View Gateway Details** dialog box displays.
2. Click **Lines**. The **View Gateway Lines** dialog box displays.
3. Click **Change Details**. The **Edit Gateway Lines** dialog box displays.
4. From the **Incoming** list that is associated with the line, select your operator extension.
5. Select **Submit**.

---

## Extending Services to Remote Users through the Teleworker Feature

The software allows workers at remote offices to have access to the Corporate directory and most of the other features and services that are available to everyone who uses the one-X Quick Edition system.

## How the Teleworker Application Works

The teleworker client telephone at the remote office connects to a teleworker server host on the one-X Quick Edition network—any telephone or PSTN gateway on the one-X Quick Edition network can act as a teleworker server host.

After you enable teleworker mode on a telephone, the telephone becomes a teleworker client.

Teleworker client remote office requirements:

- a data network
- high-speed (cable/T1/ E1/ADSL) Internet connection
- a secure Virtual Private Network (VPN) tunnel to the corporate office.

### Specifying a Teleworker Server Host

It is usually not necessary to specify a teleworker server host in the Quick Edition network unless the teleworker client telephone is unable to establish a teleworker session, or it is a factory-default telephone that has never been installed.

Given a connection to the one-X Quick Edition network through a VPN, the teleworker client telephone at the remote office attempts to connect to one of the telephones or PSTN gateways (a teleworker server host) on the one-X Quick Edition network.

If the selected teleworker server host is unavailable, the teleworker client telephone tries another one-X Quick Edition device one-by-one, until all possibilities are exhausted. This process is cyclical and may be repeated indefinitely in order for the teleworker client telephone to establish a teleworker session.

## Configuring the Teleworker Application

You can add a teleworker client telephone to the one-X Quick Edition network locally, or from a remote office.

### Adding a teleworker client to the network, locally

1. Connect the new, factory configured telephone to the one-X Quick Edition network.
2. Add the telephone to the LAN (see page [Adding a Telephone to the System](#) on page 17).
3. Move the teleworker client telephone to the remote office.
4. At the remote office, configure a VPN tunnel to the company one-X Quick Edition network. Refer to the manufacturer's documentation to configure the VPN. Ask your LAN administrator to configure complementary VPN settings at the company office to enable access to the one-X Quick Edition network.
5. Verify that the VPN tunnel is operational. For example, if the one-X Quick Edition network and certain company resources (such as a file server or web server) both use the same network address space, use a client application at the remote office (for example, an FTP client or web browser) to access the company file server or web server.

6. At the remote office, connect the teleworker client telephone to the local LAN.


**Note:**

The LAN at the remote office must be equipped with a router that forwards traffic to the one-X Quick Edition network through a VPN tunnel.

The teleworker client telephone at the remote office attempts to connect to one of the devices on the one-X Quick Edition network to establish a teleworker session. One of the telephones or PSTN gateways on the one-X Quick Edition network will act as a teleworker server to authenticate the teleworker client and establish the session.

**Note:**

You can view the status of the connection in the display area of the teleworker client telephone. The information is updated dynamically as the connection status changes. For more information, see [To view the connection status of the teleworker client telephone on page 64](#).

7. To enable teleworker mode on the telephone, select **Teleworker** on the **User Options** menu.
8. Select **Mode** on the **Teleworker** menu.
9. Select the **Chg** softkey to enable (or disable) teleworker mode.
10. Press the Phone/Exit (  ) button to clear the display area.

After a session has been established, the teleworker client telephone operates as if it were connected to the one-X Quick Edition network directly.

### Adding a teleworker client to the network from a remote location

There are two procedures involved in adding a teleworker client from a remote location:

- programming at the one-X Quick Edition network locally, and
- configuration and programming at the remote location.



### Programming at the one-X Quick Edition network locally

1. Log in to the **System Options** interface.
2. In the **Set Management** list, click **Add Extension**.
3. In the **Extension** field, type an unused extension number for the teleworker client phone.
4. In the **Name** field, type a user name to associate with the extension number.
5. In the **MAC Address** field, type the 12-digit MAC address of the teleworker client telephone (for example, 00:00:5A:99:62:50).
6. Click **Validate**.
7. When you are prompted to verify the information, click **Add Extension**.
8. At the teleworker client telephone, access the **User Options** menu to configure the teleworker client telephone to connect to a teleworker server host.

**Note:**

If a computer is connected to the local network, you can use the web browser on the computer to specify the host IP address of a teleworker server instead of following this procedure. For more information about the web-based options, see [Options on the Teleworker Options Page on page 81](#).

### Configuration and programming at the remote location

1. At the remote office, configure a VPN tunnel (see Step 4, above).
2. Verify that the VPN tunnel is operational (see Step 5, above).
3. Move the teleworker client telephone to the remote office. The telephone does not require factory configured settings.
4. At the remote office, connect the telephone to the local LAN.
5. After the teleworker client telephone initializes and a name has been assigned to the telephone (if the telephone is in a factory configured state, you will also have to assign a site name), make a note of its MAC address.
6. Select **Teleworker** on the **User Options** menu.
7. Select **Default Server** on the **Teleworker** menu.
8. Select the **Chg** softkey and enter the IP address. Press the **.** softkey to add separator characters between the digits.
9. Select the **Exit** softkey to display the previous menu, or press the Phone/Exit (  ) button to clear the display area.
10. To enable teleworker mode on the telephone, select **Teleworker** on the **User Options** menu.
11. Select **Mode** on the **Teleworker** menu.
12. Select the **Chg** softkey to enable (or disable) teleworker mode.
13. Press the Phone/Exit (  ) button to clear the display area

If the site identifier teleworker phone is not identical to that of the company one-X Quick Edition network, the teleworker server uploads stored system settings to the remote telephone—the telephone receives a new site identifier (identical to the one belonging to the company one-X Quick Edition network), a new extension number, and the user name associated with that extension number.

The teleworker client accepts the new site identifier along with additional configuration information sent by the teleworker server. The new configuration will be put into effect when the teleworker client telephone is restarted.

14. When a message prompting you to restart the teleworker client telephone is displayed, select the **OK** softkey.

**Tip:**


If you choose to not proceed at this point, disconnecting the source of power to the teleworker client telephone and then reconnecting the supply of power will restore the configuration of the telephone.

The next time a session is established, the teleworker client telephone operates as if it were connected to the one-X Quick Edition network directly.

### To view the connection status of the teleworker client telephone

1. At the teleworker client telephone, access the **User Options** menu (see “Displaying the User Options Menu” in the *Avaya one-X Quick Edition Telephone User Guide*, Document No. 16-601411).
2. Select **Teleworker** on the **User Options** menu.
3. Select **Status** on the **Teleworker** menu.

One of the following messages is displayed:

- When **Local** is displayed, teleworker mode has not been enabled.
  - When **Idle** is displayed, teleworker mode is enabled and the teleworker client is idle.
  - When **Connecting...** is displayed, the teleworker client telephone is attempting to connect to a teleworker server host—a session has not been established. The host IP address of the teleworker server is also displayed in the display area.
  - When **Connected** is displayed, the teleworker client telephone has connected to the teleworker server host successfully and can access the remote one-X Quick Edition network. The teleworker server host IP address is also displayed in the display area.
4. Select the **Exit** softkey to display the previous menu, or press the Phone/Exit (  ) button to clear the display area.

**Tip:**

For additional information related to troubleshooting connectivity problems, see [Chapter 3: Troubleshooting](#).



# Chapter 3: Troubleshooting

Avaya Quick Edition IP telephones are relatively trouble free. This chapter provides helpful troubleshooting information, and basic troubleshooting charts that walk you through resolving problems.

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## Basic Troubleshooting for Telephones

If you cannot resolve a problem on your own, contact your technical support representative (see [Contacting Technical Support on page 75](#)).






**Tip:**

If the suggestions given below do not resolve the problem, power cycle the telephone by disconnecting the PoE LAN connection (or switching off local power to the telephone) and then reconnecting the telephone (or power source). There will be a slight operational delay when you power cycle the telephone.

**Table 1: Basic Troubleshooting Procedures for Quick Edition IP Telephones**

Problem/Symptom	Suggested Solution(s)
Telephone does not power on after connecting it to the LAN.	<p>Verify that the telephone is receiving power through the Ethernet LAN connection:</p> <ul style="list-style-type: none"> <li>● Verify that the switch or router to which the telephone is connected supports 802.3af PoE. If all other connected Quick Edition IP telephones exhibit the same problem, the switch or router may need to be replaced.</li> <li>● If the LAN is not 802.3af PoE enabled, it is your responsibility to provide a suitable inline power supply. You can order an Avaya PoE injector (Part # 1151B1) to power the telephone.</li> <li>● Check the Ethernet LAN connection to the telephone to ensure that it is properly connected. If required, swap the cable to determine whether the cable is defective.</li> </ul> <p>If you can determine that power is being supplied to the telephone (the display area or any of the indicators on the front panel of the telephone are lit) but the telephone still does not become operational, report the problem to your technical support representative.</p>
Telephone does not become active immediately after a power interruption.	<p>Allow a few minutes for the telephone to initialize if the telephone became unplugged from the LAN or experienced a power interruption.</p>
Telephone does not display the option to join an existing site when connected to an existing network.	<p>When a telephone is connected to the LAN for the first time, it searches for P2P peers. If the existing P2P devices do not use IP addresses in the same network address space (for example, static IP addresses may have been assigned), a newly connected telephone will not be able to detect peers connected to the same network segment. As a result, the telephone will prompt you to create a new network. For more information, see <a href="#">A Word about IP Addresses on page 73</a>.</p> <p>To resolve the problem, see <a href="#">To re-assign a telephone to a different network address space and/or network on page 75</a>.</p>

Table 1: Basic Troubleshooting Procedures for Quick Edition IP Telephones (continued)

Problem/Symptom	Suggested Solution(s)
Telephone no longer communicates with peers.	<p>Verify the telephone connection to the LAN.</p> <p>Review the network performance recommendations to ensure that the network adheres to minimum requirements (see <a href="#">Performance Recommendations</a> on page 12).</p> <p>When previously configured and reconnected to the network, a telephone needs a few minutes to initialize and become operational. During this time, the telephone attempts to communicate with all other one-X Quick Edition devices in the same network address space. Verify the following items by viewing <b>Set Details</b> on the <b>Main</b> menu:</p> <ul style="list-style-type: none"> <li>● Are all one-X Quick Edition devices running the same or compatible software versions?</li> <li>● Do all devices belong to the same network address space? If required, see <a href="#">To re-assign a telephone to a different network address space and/or network on page 75</a>.</li> <li>● Do all peers belong to the same one-X Quick Edition site? When you install a Quick Edition IP telephone, you can choose to have the telephone join an existing network or create a new network. Unless the telephone is the first telephone to be installed, you should have the telephone join the existing network. To determine to which network a Quick Edition IP telephone belongs, look up the site identifier, which is displayed in the <b>Site</b> field under <b>Set Details</b>. All peers making up the same network have the same site identifier and name. If required, see <a href="#">To re-assign a telephone to a different network address space and/or network on page 75</a> to enable the telephone to communicate with existing peers.</li> </ul>
Speaker does not seem to work.	<p>Press the Speaker (  ) button. Use the Volume Control (  ) buttons to set the volume to an audible level. If you do not hear a dial tone, contact your sales representative.</p>
Cannot add an entry to the Speed Dial list.	<p>You can have a maximum of nine Speed Dial button assignments. If all assignments are in use, delete at least one entry before attempting to add a new entry.</p>
Cannot dial an external number using any Speed Dial button or the Call Log.	<p>See “Cannot make an external call” below.</p>
No dial tone on a Quick Edition IP telephone.	<p>Verify that the handset is securely connected.</p> <p>Use the Volume Control (  ) buttons to set the volume to an audible level.</p> <p>If your network includes a PSTN gateway, verify that the PSTN line(s) managed by your service provider are securely connected to the gateway through customer-provided telephone cords.</p>

**Table 1: Basic Troubleshooting Procedures for Quick Edition IP Telephones (continued)**





Problem/Symptom	Suggested Solution(s)
Telephone does not ring on an incoming call.	Use the Volume Control (  ) buttons set the ringer volume to an audible level. From another telephone, place a call to your extension to test the ringer volume after adjustment.
Information is not displayed in the display area.	See “Telephone does not power on after connecting it to the LAN” above. Check all connections to the telephone to ensure that it is properly connected and receiving power.
The display area appears to be frozen and there is no dial tone.	When a menu is not being accessed, labels for the <b>SDial</b> , <b>Log</b> , <b>Dir</b> , and <b>Vmail</b> softkeys are displayed at the bottom of the display area. If a <b>Back</b> , <b>Exit</b> , or <b>Done</b> softkey is displayed instead, select the softkey. Alternatively, press the Phone/Exit (  ) button to restore the display area.
Cannot reach the party at a known extension through the Corporate directory.	<p>Verify the dial tone (see “No dial tone on a Quick Edition IP telephone” above).</p> <p>Next, verify that the party you are trying to call has not enabled:</p> <ul style="list-style-type: none"> <li>● call forwarding (see “Redirecting Calls through Call Forwarding” in the <i>Avaya one-X Quick Edition Telephone User Guide</i>), and/or</li> <li>● the do-not-disturb feature (see “Enabling/Disabling the Do-Not-Disturb Feature” in the <i>Avaya one-X Quick Edition Telephone User Guide</i>).</li> </ul> <p>If required, contact your LAN administrator to determine whether there is a problem with the Ethernet LAN.</p>
Cannot make an external call.	<p>Verify the dial tone (see “No dial tone on a Quick Edition IP telephone” above). If your network includes a PSTN gateway and all PSTN lines are busy, you will hear a low-pitched, rapid (busy) tone repeated 60 times per minute. Wait for a few minutes until a PSTN line becomes free, and then try your call again.</p> <p>To place a call to the PSTN, dial 9 followed by the number. To call a SIP network, dial 8 followed by the number.</p> <p>If the problem persists, verify that the PSTN gateway is connected and accessible, and that at least one PSTN line is connected to line port L1 on the rear panel of the gateway. A PSTN line must be connected to port L1 using a customer-provided telephone cord.</p> <p>Before you call your technical support representative, contact your service provider to verify that the PSTN lines are working as expected (for example, loop-start signaling is required).</p>

Table 1: Basic Troubleshooting Procedures for Quick Edition IP Telephones (continued)

Problem/Symptom	Suggested Solution(s)
Redial (  ) button does not dial the last number called.	Check the year in the system date and time setting (see <a href="#">Setting the System Date and Time</a> on page 34). If the factory set year is different compared to the current year, the year setting will be incorrect after the Quick Edition IP telephone goes through a power cycle. Calls in the Call Log <b>Outgoing</b> list are sorted according to date, and the redial function dials number at the top of the list. If the system date is incorrect, the redial function may not be dialing the most recent outgoing call.
<p>Audio quality is poor—specifically, you hear:</p> <ul style="list-style-type: none"> <li>● static when the line is active,</li> <li>● sudden silences or gaps in speech, and/or</li> <li>● clipped or garbled speech.</li> </ul>	<p>Network problems are the most likely cause. Contact your LAN administrator and provide a complete description of the problem. If you are the LAN administrator, see <a href="#">Performance Recommendations</a> on page 12 for a list of recommendations.</p> <p> <b>Tip:</b> You can specify an audio quality setting for the entire network (see <a href="#">To adjust the audio bandwidth</a> on page 39).</p>
An application or feature does not work as described in the product documentation.	<p>Verify the procedure and retry.</p> <p>If this action does not produce the desired result, the one-X Quick Edition system might need additional programming to enable certain features. Consult the product documentation for additional configuration instructions (see <a href="#">one-X Quick Edition Documentation</a> on page 6).</p> <p>If you would like to report a documentation error or omission, contact your technical support representative.</p>

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Table 1: Basic Troubleshooting Procedures for Quick Edition IP Telephones (continued)


Problem/Symptom	Suggested Solution(s)
<p><b>Teleworker:</b> Teleworker client telephone cannot connect to the remote company network.</p>	<p>Determine the connection status of the teleworker client telephone (see <a href="#">“To view the connection status of the teleworker client telephone on page 64”</a>).</p> <p>Verify that the VPN tunnel is operational. If the IP address of the company LAN has changed recently and the VPN configuration has not been updated, the teleworker client telephone may be unable to connect:</p> <ul style="list-style-type: none"> <li>● Update the private IP address of the company office network in the VPN configuration.</li> <li>● If a host IP address is specified for the teleworker server, change the IP address specification if required.</li> </ul> <p>A teleworker client telephone cannot act as a teleworker server. The teleworker server host IP address must belong to a telephone or PSTN gateway on the one-X Quick Edition network.</p> <p><b>No Server</b> message - specify the teleworker server host IP address.</p> <p><b>Not Authorized</b> message - the teleworker server host IP does not recognize the client. Verify the <b>Add Extension</b> programming.</p>
<p><b>Restart Required</b> message</p>	<p>The teleworker server host has uploaded new software to the client telephone. The configuration will take effect when the client telephone is restarted. Press the <b>OK</b> softkey to restart.</p>
<p><b>Server Busy</b> message</p>	<p>The teleworker server host is occupied with connections to two other client telephones and cannot accept a third. Supply each teleworker with a different teleworker server host IP.</p>
<p><b>No Response</b> message</p>	<p>The connection request timed out.</p> <p>Verify your Internet connection. Verify that the router and firewall have been configured to pass P2P packets through a VPN tunnel.</p> <p>Verify that the teleworker server host is connected to the one-X Quick Edition network. However, with the physical connection working properly, the teleworker client telephone will contact another teleworker server host for a connection automatically</p>
<p><b>Late Approval</b> message</p>	<p>The message may be sent, after a No Response message, to the client telephone by a teleworker server host that was unable to acknowledge a previous connection request.</p>
<p>Connection is successful, but the call was dropped during a conversation.</p>	<p>If someone unplugs a Quick Edition IP telephone that is acting as the teleworker server for the connected teleworker client telephone, the call will be dropped.</p>
<p><b>5 of 5</b></p>	

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## System Problems and Solutions

The information in [Table 2: Avaya one-X Quick Edition System Problems and Solutions](#) below may help you to resolve a problem with the one-X Quick Edition system.

Table 2: Avaya one-X Quick Edition System Problems and Solutions

Problem/Symptom	Suggested Solution(s)
<p><b>Web-based interface:</b> Cannot access the web-based interface.</p>	<p>Compare the IP address of the telephone and its P2P peers to the IP address of the PC on which the web browser is installed. To determine the IP address of a PC, see the Tip under <a href="#">To upgrade the software using a Quick Edition IP telephone</a> on page 26. If the telephone uses a different network address space compared to the address space used by its P2P peers, change the IP address of the telephone to match the network address space used by its peers (see <a href="#">To change network address settings using a Quick Edition IP telephone</a> on page 22). If the network address space used by the telephones does not match the network address space used by the PC, ask your LAN administrator to add a route to the one-X Quick Edition network from the PC.</p> <p>Verify that you can access user options and/or system options using the Options (  ) button on the telephone. If you cannot access these options using telephone buttons, the telephone is not working properly. Troubleshoot the problem using the guidelines given in <a href="#">Table 1: Basic Troubleshooting Procedures for Quick Edition IP Telephones</a> on page 66.</p> <p>Check the LAN settings in your web browser. Proxy server settings could be preventing you from accessing the web-based interface. In Internet Explorer, select <b>Tools &gt; Internet Options</b>, and then click the <b>Connections</b> tab. Click the <b>LAN Settings</b> button to view the proxy server settings. Contact your LAN administrator for more information and help if required.</p>
<p><b>Auto Attendant:</b> The Auto Attendant feature does not answer incoming calls, or an incorrect greeting is played.</p>	<p>Access the <b>Auto Attendant</b> menu under <b>System Options</b>, and verify that the Auto Attendant configuration that you want to use has been created. See <a href="#">Configuring the Auto Attendant Feature</a> on page 53.</p> <p>If you have a PSTN gateway, verify that an Auto Attendant configuration has been assigned to the gateway (see <a href="#">To view or edit an Auto Attendant configuration using a Quick Edition IP telephone</a> on page 56). Verify that the greeting you want to use has been recorded and assigned to the associated Auto Attendant configuration (see <a href="#">Viewing and Editing Auto Attendant Configurations</a> on page 56).</p> <p>If your one-X Quick Edition system interoperates with a SIP network, you can assign an Auto Attendant configuration to each SIP identity through the web-based administration interface. Ensure that the Auto Attendant feature has been enabled for the identity (see <a href="#">To assign a SIP identity on page 51</a>).</p>

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Table 2: Avaya one-X Quick Edition System Problems and Solutions (continued)

Problem/Symptom	Suggested Solution(s)
<p><b>Paging:</b> Not all of the telephones in a paging zone are receiving announcements.</p>	<p>If the telephone in question is in use while you make the announcement, the announcement is suppressed on that telephone—this is normal.</p> <p>Verify that the telephone in question has been added to the paging zone (see <a href="#">To add a telephone to a specific paging zone</a> on page 19).</p> <p>Verify that the telephone in question is on-hook, and that the do-not-disturb feature is disabled (see “Enabling/Disabling the Do-Not-Disturb Feature” in the <i>Avaya one-X Quick Edition Telephone User Guide</i>).</p>
<p><b>Voicemail:</b> Callers cannot leave voicemail because the storage area is full.</p>	<p>When a Voicemail inbox is full, callers hear a message indicating that Voicemail storage has been filled and no more messages can be saved. To avoid this situation, users are encouraged to delete voicemail messages on a regular basis.</p>
<p><b>Email Notification of Voicemail:</b> Voicemail messages are not being forwarded to an e-mail account.</p>	<p>Access the <b>Network Options</b> menu under <b>System Options</b> to verify that the IP address of an SMTP server has been specified (see <a href="#">Supporting E-mail Notification of Voicemail</a> on page 42).</p> <p>Using the web-based <b>User Options</b> interface (see <a href="#">Appendix A: Web-based User Options</a>), select <b>Voice Mail</b> and verify that “To” and “From” addresses have been specified, and that notification status has been enabled.</p>

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## A Word about IP Addresses

Every telephone has an IP address. If a DHCP (Dynamic Host Configuration Protocol) server resides on the Ethernet LAN, the telephones and any PSTN gateways that you connect to the LAN will request network addresses from the DHCP server.

On networks without a DHCP server, telephones and PSTN gateways assign themselves unique addresses in the 169.254/16 (Zeroconf) network address space with a network mask of 255.255.0.0.

Unless assigned a static IP address previously, Avaya one-X Quick Edition devices assign themselves IP addresses this way each time they are connected to an Ethernet LAN. If you move devices making up the same one-X Quick Edition network from one LAN to another, they may assign themselves different IP addresses without disrupting communications between established network peers.

If a telephone shares a network connection with a PC and the network does not have a DHCP server, it is possible that the telephone and PC will have IP addresses in different network address spaces. As a result, the browser on the PC will be unable to connect to the telephone through the web-based interface. To resolve the problem, ensure that the one-X Quick Edition network and the PC network share the same network address space. Make sure that you use unused IP addresses for the telephones and PSTN gateways. For more information, see [Assigning a Static IP Address](#) below.

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## Assigning a Static IP Address

If required, you may assign fixed network addresses to telephones or PSTN gateways through network options (see [Modifying Telephone Network Address Settings](#) on page 21 in this document or the changing addresses topics in the *Avaya one-X Quick Edition G10 PSTN Gateway Installation Guide* or *Avaya one-X Quick Edition G11 Global Analog Gateway Installation Guide*. To change the IP address of a telephone or PSTN gateway, you must know the administration password.

Once assigned, a static IP address persists until it is either changed manually, or another device having the same IP address is detected at boot time. Because the devices making up a one-X Quick Edition network communicate configuration changes to each other automatically, any IP address changes that you make are communicated to all of the devices making up the network.

If the network does not have a DHCP server and you add a telephone or PSTN gateway to the network, the telephone or gateway will assign itself a unique address in the Zeroconf network address space. Because the self-assigned network address may not match the network address space being used by other P2P devices (for example, due to static IP address assignments), the newly added telephone or PSTN gateway may not be able to communicate with those P2P devices:

- A telephone will find no P2P peers and as a result, will create a new network when connected to the LAN.
- You will be unable to access the web-based interface of the telephone using a PC connected to the LAN, even when the telephone and PC share the connection to the LAN.

To resolve this type of issue, see [To re-assign a telephone to a different network address space and/or network](#) below.

To enable telephones or PSTN gateways with manually configured (static) IP addresses to go back to requesting IP addresses from a DHCP server or assigning themselves Zeroconf addresses automatically at boot time, you must change the static IP address of the telephone or gateway to 0.0.0.0.

### To re-assign a telephone to a different network address space and/or network

1. If the IP address of the telephone does not match the IP address space being used by other P2P devices, change the IP address of the telephone to an unused IP address in the required network address space. To determine the IP address of a telephone, press the Page Right (⏪) button. To change the IP address, see [Modifying Telephone Network Address Settings](#) on page 21).
2. Remove the extension number of the telephone from the network (see [Removing a Telephone or Gateway from the System](#) on page 20), but do not disconnect the telephone from the network. When the telephone initializes, have the telephone join the target network. The procedure does not change the manually assigned (static) IP address.

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## Contacting Technical Support

If you are unable to resolve the problem using the troubleshooting guidelines provided in this chapter, contact your technical support representative. If required, you may contact Avaya Technical Support by using one of the methods available through the Avaya Technical Support web site:

<http://support.avaya.com/QuickEdition>



**Important:**

If you have an Avaya support plan, call the number associated with that plan to obtain service according to the terms of your agreement.



# Appendix A: Web-based User Options

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## Introduction

When a computer is connected to the one-X Quick Edition network, you can use the web browser on the computer to access and manage some telephone and user options through the web-based **User Options** interface. For best results, choose Microsoft Internet Explorer 6.0 (or later) or Mozilla Firefox 1.0 (or later).

**Note:**

Refer to the *Avaya one-X Quick Edition Telephone User Guide* (Document No. 16-601411) to configure the following features. Refer to [Appendix B: Options Menus](#) on page 83 for a list of Web-based and telephone interface-configurable options.


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## Logging in to the Web-based User Options Interface

To log in to the web-based **User Options** interface, you need to know the IP address, extension number, and the user options password for the telephone.



**Tip:**

To display the IP address of the telephone, press the Options (  ) button and select **Set Details** on the **Main** menu.

### To access telephone and user options using a web browser

1. Start the web browser on your computer.
2. In the **Address** field, enter the IP address of the telephone (for example, if the IP address is 192.168.0.2, type `https://192.168.0.2`).  
If you have not disabled the first security alert, a Security Alert message is displayed.
3. Click **OK**. If you have not installed the self-signed security certificate on your computer, a Security Alert action request is displayed.
4. To proceed without installing the security certificate, click **Yes.**, or to install the security certificate, click **View Certificate**. When the Certificate dialog box is displayed, click **Install Certificate** and follow the on-screen instructions.

5. Enter the extension number of the telephone and the password for accessing user options.

The **User Options Home** page for the telephone is displayed. Links to additional pages for viewing and setting call log options, terminal settings, and voicemail options, and teleworker options are displayed in the navigation bar on the left side of the screen.

---

## Logging Out

### To log out from the web-based User Options interface

- Click **Logout** in the upper right corner of the **User Options** web page.

## Security

When you log in to the **User Options** login page, the system uses the Secure Sockets Layer (SSL) protocol to secure communications between the web browser on your computer and the system. SSL encrypts the information that you enter when you log in before that information is transmitted to the system. When the web browser connects to the telephone network, SSL is used to verify the identity of your computer to the telephone. As part of the exchange, the telephone generates and downloads a self-signed security certificate to your computer.

Your telephone uses a self-signed security certificate to authenticate itself to a web browser whenever the web browser initiates a secure connection through SSL. When the certificate is downloaded, two security messages are displayed in the web browser:

- The first message informs you that a secure link is going to be set up through SSL. You can select an option to suppress the display of this message.
- The second message prompts you to accept and optionally install the telephone's self-signed security certificate. If you do not accept the certificate, the telephone refuses the connection. When you accept the certificate, the **User Options** login page is displayed. If you choose to install the certificate, the prompt is not displayed again.

---

## Changing the Password

You can change, enable, or disable the password through the **User Options** menu on a telephone. The same password is used for voicemail and for user options.

### To change the telephone and voicemail password

1. Click **Change Password** in the upper right corner of the **User Options** web page.
2. Enter the current password, the new password, and confirm the new password.
3. Click **Submit**.

---

## Home Page Options

The **Home** page displays call-forwarding, do-not-disturb, and Speed Dial options. Refer to the *Avaya one-X Quick Edition Telephone User Guide* for feature details.

## Call-Forwarding Option Settings

Click **Change** in the **Call Forwarding** dialog box to modify call-forwarding option settings.

## Do-Not-Disturb Feature

Click **Change** in the **Do Not Disturb (DND)** dialog box to select the **Enable Do Not Disturb** option.

## Speed Dial Assignments

The dotted squares in the **Speed Dial** dialog box to display the name and extension or external telephone number that has been assigned to a Speed Dial button. Click a vacant square to add a speed dial assignment.

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## Caller's Log Page Options

The **Caller's Log** page displays a list of incoming calls and a list of dialed calls. Refer to the *Avaya one-X Quick Edition Telephone User Guide* for feature details.

- Click **Clear Log** to clear the Incoming list or dialed calls list.
- Click **Reset Missed Call Counter** to clear the counter that records the number of unanswered/ignored calls.
- Click **Clear All Logs** to resets both lists and the counter of missed/ignored calls.

---

## Terminal Settings Page Options

The **Terminal Settings** page displays the Corporate-directory name that has been assigned to the telephone, and also provides registration information for telephone features.

For general information about the names associated with telephone extension numbers, see "Editing Your Name in the Corporate Directory" in the *Avaya one-X Quick Edition Telephone User Guide*. For information about special features, see [Viewing Registration Information for Optional Features](#) on page 22.

Click **Change** in the **Terminal Settings** dialog box to change the name that has been assigned to the telephone extension number.

---

## Voicemail Options

The voicemail page displays information about the designated operator (see [Enabling/Disabling Zero-Redirect on page 80](#)) and provides a way for you to specify the settings needed to finish configuring the email-notification-of-voicemail feature (see [Specifying an e-mail Address to Receive Voicemail Notifications on page 80](#)).

### Enabling/Disabling Zero-Redirect

Click **Change** in the **Zero Redirect** dialog box to enable or disable the zero-redirect feature and specify an extension or telephone number. The zero-redirect feature permits a caller to redirect an unanswered call to the number you specify while listening to your personal greeting.

For general information about this feature, see “Setting Up the Zero-Redirect Feature” in the *Avaya one-X Quick Edition Telephone User Guide*.

### Selecting the Language of Voicemail Prompts

The system language setting determines which set of prompts are used for voicemail on telephones.

### Specifying an e-mail Address to Receive Voicemail Notifications

The complete procedure for configuring e-mail notification of voicemail is described under [Supporting E-mail Notification of Voicemail](#) on page 42. An overview of the associated web-based user options is provided in this section.

**Note:**

The audio component of the voicemail message is sent as an audio file attachment when the feature is enabled. For information about how to obtain the software needed to play the audio file attachment, see [Playing an Audio File Attachment](#) below.

Click **Change** in the SMTP dialog box, to specify the e-mail address of the SMTP recipient and enable the feature.

### Playing an Audio File Attachment

To permit users to play audio files when they arrive as e-mail attachments, you must download from the Avaya Technical Support site and install (on each user's PC) a free plug-in for the Microsoft Windows Media Player. The users' PCs must be running Microsoft Windows XP or 2003, and have audio playback capability.



**Note:**

The plug-in is not compatible with “N” editions of the Microsoft Windows XP operating system.

In addition, the plug-in requires that you install Microsoft DirectX 9.0c (or later) runtime software and the Microsoft Installer installation program. DirectX is an addition to the Microsoft Windows operating system.

A link to the installation program and the plug-in (<http://support.avaya.com/QuickEdition/MediaPlayer>) is conveniently included in every e-mail message. Install the Microsoft Installer program and run the installation program once to install the plug-in. The installation program will remind you to download and install the required DirectX runtime software.

After the plug-in has been installed (and provided you installed the DirectX runtime software), double-clicking the WAV file attachment directly from within the e-mail message will start the playback of the WAV file through the Windows Media Player.

---

## Options on the Teleworker Options Page

The **Teleworker Options** page displays information about teleworker application settings.

For more information about this feature, see [Extending Services to Remote Users through the Teleworker Feature on page 60](#).

When **Working Mode** is set to **Local**, teleworker mode is disabled. When **Working Mode** is set to **Teleworker**, teleworker mode is enabled.

Click **Change** in the **Teleworker Options** dialog box to enable or disable teleworker mode, and if required, specify the IP address of a teleworker server host.



# Appendix B: Options Menus

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## Telephone Options

**Table 3: Telephone Options - Telephone User Option Menu**

<b>1. Password</b>	Change password or turn password on or off.
<b>2. Call Forward</b>	Enable, disable, and modify call forwarding settings. Also available through a softkey.
<b>3. Voicemail</b>	Enable and disable operator redirect and the number to which a call will be redirected. Record your name and greeting. Also available through a softkey on the 4621 SW IP.
<b>4. Call Log</b>	View details, Clear, and Reset Missed Call Counter. There are additional options available from Call Logs on the Main Menu.
<b>5. Language</b>	Select a user language from the drop-down list.
<b>6. Name</b>	Enter your name in the corporate directory.
<b>7. Personalized Ringing</b>	Review and select your ringer tone.
<b>8. Call Waiting Tone</b>	Enable and disable the call waiting tone.
<b>9. Contrast Level</b>	Adjust the contrast level of your telephone display.
<b>10. Teleworker</b>	Enable to connect your Quick Edition IP telephone to a high-speed Internet connection and access the Corporate directory.

**Table 4: Telephone Options - Web Interface Menu**

<b>Change Password</b>	Enter existing password, enter new password, and confirm new password.	
<b>Home</b>	<b>Call Forwarding.</b>	Enable, disable, and modify call forwarding settings.
	<b>Do Not Disturb (DND)</b>	Enable and disable audio notification for incoming calls.
	<b>Speed Dial</b>	Create, modify, and delete personal speed dial numbers.
<b>Caller's Logs</b>	<b>Incoming Calls</b>	View details, Clear, and Reset Missed Call Counter.
	<b>Dialed Calls</b>	View details and Clear Log.
<b>Terminal Settings</b>	<b>Name</b>	Change name.
	<b>Set Optional Features</b>	View e-mail Fwd, Teleworker, and WebAdm Sys Options.
<b>Voice Mail</b>	<b>Zero Redirect</b>	Enable and disable Operator redirect and enter the number to which a call will be redirected.
	<b>SMTP Settings</b>	Enable or disable SMTP on network, and specify IP address of SMTP server host and/or SMTP port for support of e-mail notification of voice mail.
<b>Teleworker Options</b>	<b>Working Mode</b>	Disable (Worker) or enable (Teleworker) teleworker.
	<b>Preferred Server</b>	Enter the IP address of the teleworker server host.

# System Options

**Table 5: System Options - Telephone Menu**

<b>1. Change Password</b>	Enter existing password, enter new password, and confirm new password.	
<b>2. Set Management</b>	<b>1. Set Extension</b>	Change telephone extension number.
	<b>2. Remove Extension</b>	Specify extension to remove.
	<b>3. Reset Password</b>	Reset password on accessed telephone.
	<b>4. Upgrade</b>	View software version and/or confirm upgrade. Specify IP address of TFTP server host and start upgrade.
<b>3. System Language</b>	The language change will not apply to voicemail or auto attendant prompts, or Web admin.	
<b>4. System Region</b>	The region selection will apply the correct regional tone package.	
<b>5. Date and Time</b>	Edit and save system date and time.	
<b>6. Network Options</b>	<b>1. IP Address</b>	View or change IP address or network mask of telephone, IP address of default gateway for telephone, and/or IP address of DNS server.
	<b>2. SMTP Settings</b>	Enable or disable SMTP on network, and specify IP address of SMTP server host and/or SMTP port.
	<b>3. Operator Extension</b>	View or change the designated operator extension.
<b>7. Gateways</b>	<b>1. Details</b>	View PSTN gateway IP address, software version, connection status, and MAC address.
	<b>2. Lines</b>	Select a PSTN line and view or change its loop length setting.
	<b>3. Music On Hold</b>	Enable or disable the playing of audio input.
	<b>4. IP Address</b>	View or change IP address or network mask of PSTN gateway, and/or IP address of default IP gateway.
<b>8. Auto Attendant</b>	<b>1. Auto Attendant</b>	View extension number, name, selected greeting (prompt), prompt language, and extension number of associated PSTN gateway.
	<b>2. Custom Greeting 1</b>	Record, play, and/or save a custom greeting.
	<b>3. Custom Greeting 2</b>	Record, play, and/or save a custom greeting.

**Table 6: System Options - Web Interface Menu**

<b>Change Admin Password</b>	Enter existing password, enter new password, and confirm new password.	
<b>Set Management</b>	<b>Sets &amp; Operator</b>	Add or remove a telephone and its extension or select a new operator extension number.
	<b>Software Upgrade</b>	Upgrade all sets and analog gateways in the network or upgrade only one telephone.
	<b>Set Details</b>	View and change the extension number, name, and page zone. Reset the user password back to default.
	<b>Networking</b>	You can assign a static IP address.
	<b>Features</b>	View e-mail Fwd, Teleworker, and WebAdm Sys Options.
<b>Localization</b>	<b>Language and Region</b>	Select a system language and region from the drop-down lists.
<b>Groups</b>	<b>Group Details</b>	Add or delete a group; change the group name.
	<b>Members</b>	Add and remove members.
	<b>Forwarding</b>	Enable and disable forwarding for the group. Provide a number to which group calls will be forwarded.
	<b>Dialing Rules</b>	Add and remove dialing rules. Change rule details.
<b>Time &amp; Date</b>	Edit and save system time and date.	
<b>Gateways</b>	<b>Details</b>	View PSTN gateway IP address, software version, connection status, and MAC address.
	<b>Lines</b>	Select a PSTN line and view or change its loop length setting.
	<b>Music On Hold</b>	Enable or disable the playing of audio input.
	<b>Networking</b>	View or change IP address or network mask of PSTN gateway, and/or IP address of default IP gateway.
<b>Dial Plan</b>	Extension ranges, Emergency Code, Operator Code, PSTN Code, and SIP Code are applied, by default, depending on the region specified at install.	
<b>Dialing Rules</b>	You cannot delete or modify the emergency, extension, external, operator, or VoIP E164 rules. You can create a new rule that may be modified or deleted.	
<b>Networking</b>	<b>SMTP</b>	Enable or disable SMTP; define the SMPT server IP address and port.
	<b>Audio Bandwidth</b>	Select High or Low. High provides a better quality voice transmission but with a higher bandwidth consumption.
	<b>VLAN Settings</b>	Select enable and priority or disable for voice traffic. Select disabled or pass-through or enabled and priority for data traffic.
<b>Service Provider</b>	You can add a service provider with the following configuration components: Domain Name, Proxy Host, Proxy Port, Registrar Host, Registrar Port, Outbound Proxy Host, Outbound Proxy Port, Realm, and Register Expiry Time.	
<b>Auto Attendant</b>	You can add, modify, and delete a custom auto attendant.	

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