



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) by Nullsoft, Inc.

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Chapter 1: Technical Overview

Introduction

This guide provides instructions for administering all of the Avaya one-X™ Quick Edition IP telephones and gateways that make 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.
System Options	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 using default values. All telephones connected to the same network participate in traffic routing, call handling, and other network-related processes automatically. A one-X Quick Edition network can be equipped with a 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 Local Area Network (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 Public Switched Telephone Network (PSTN), through an ISDN BRI Gateway, 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.

Documentation List

- Avaya one-X™ Quick Edition System Administration Guide
- Avaya one-X Quick Edition Telephone User Guide
- Avaya one-X Quick Edition Telephone Safety and Quick Installation Instructions
- Avaya one-X Quick Edition 46xx SW IP Telephone Quick Reference Guide
- Avaya one-X Quick Edition Safety and Quick Installation Instructions for: G11 Global Analog Gateway, G20 ISDN BRI Gateway, A10 Analog Telephone Adapter
- Avaya one-X Quick Edition Multisite Provisioning Tool Help.

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. System changes and updates are communicated to all telephones and gateways on the network.

Note:

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

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.

Typical Network Configurations

[Figure 6](#) on page 105 and [Figure 9](#) on page 113 show typical local network configurations. 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 subnet to provide web-based access to user-specific and system-wide settings.

Note:

Your one-X Quick Edition network must be equipped with a gateway to access the Public Switched Telephone Network (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 subnet.

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 subnet*. 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 Zeroconf IP addresses (see Note below) 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.
- Avaya one-X Quick Edition devices assign themselves IP addresses each time they are connected to an Ethernet LAN. Static IP addresses can be assigned to the devices.
- Virtual LANs (VLANs) may be defined on an Ethernet switch. If your network has 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 subnet, 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.

Note:

Zero Configuration Networking (Zeroconf), based on the RFC3927 standard, is a set of techniques that automatically create a usable IP network without configuration or special servers.

Performance Recommendations

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

- Supply RJ-45 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.
- Delay, jitter, and packet loss in speech delivery all have the potential to degrade the quality of one-X Quick Edition system. To achieve the best voice quality performance, ensure that end-to-end delay, jitter, and packet loss on the network are within 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.
- 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 78. 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 A10/G20 is 224.0.1.75; the rest of the one-X Quick Edition system uses 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.

Adding an Analog Gateway

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 Safety and Quick Installation Instructions for: G11 Global Analog Gateway, G20 ISDN BRI Gateway, A10 Analog Telephone Adapter* (Document No. 16-601414).

ISDN BRI Gateway

The G20 ISDN BRI Gateway supports two VoIP calls on two ISDN BRI ports. This gateway allows your one-X Quick Edition network to use Internet Telephony on existing ISDN phones for Single office/Home office and branch office voice and data connectivity.

To install a BRI gateway, see *Avaya one-X Quick Edition Safety and Quick Installation Instructions for: G11 Global Analog Gateway, G20 ISDN BRI Gateway, A10 Analog Telephone Adapter* (Document No. 16-601414).

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 [Add a Teleworker on page 48](#).

Figure 1: Teleworker over Internet (VPN)

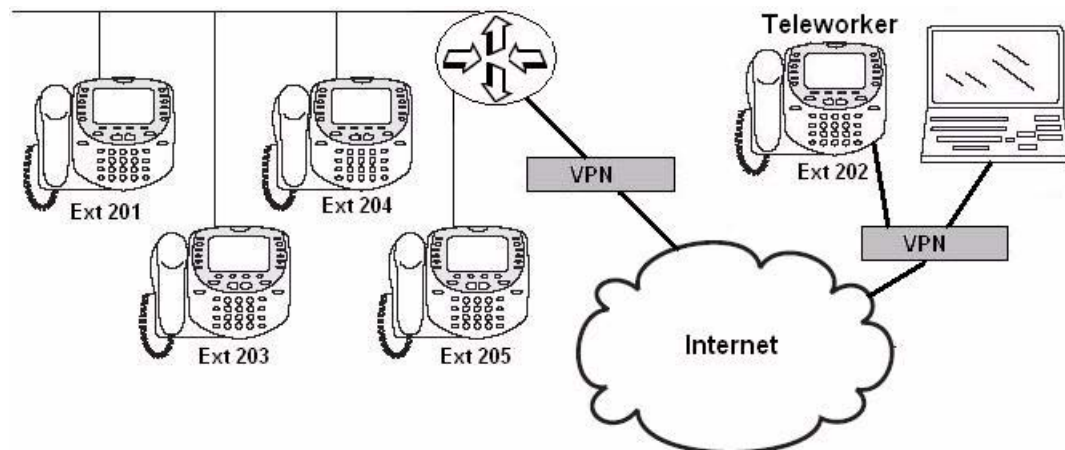
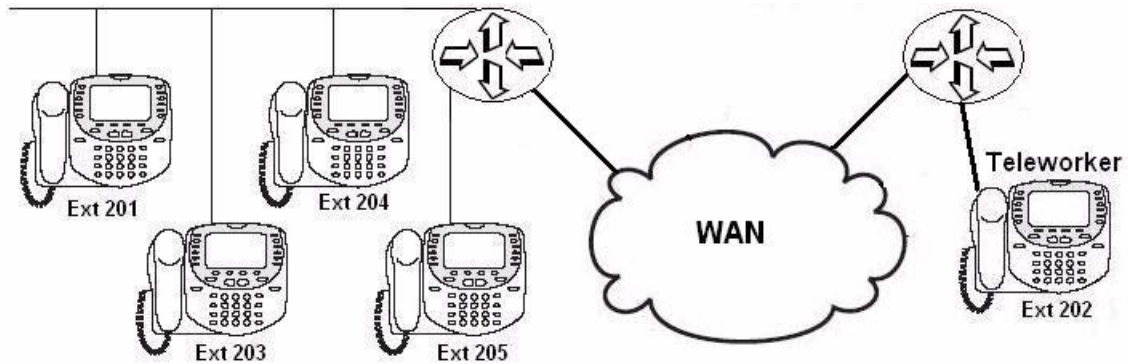


Figure 2: Teleworker over a Private WAN

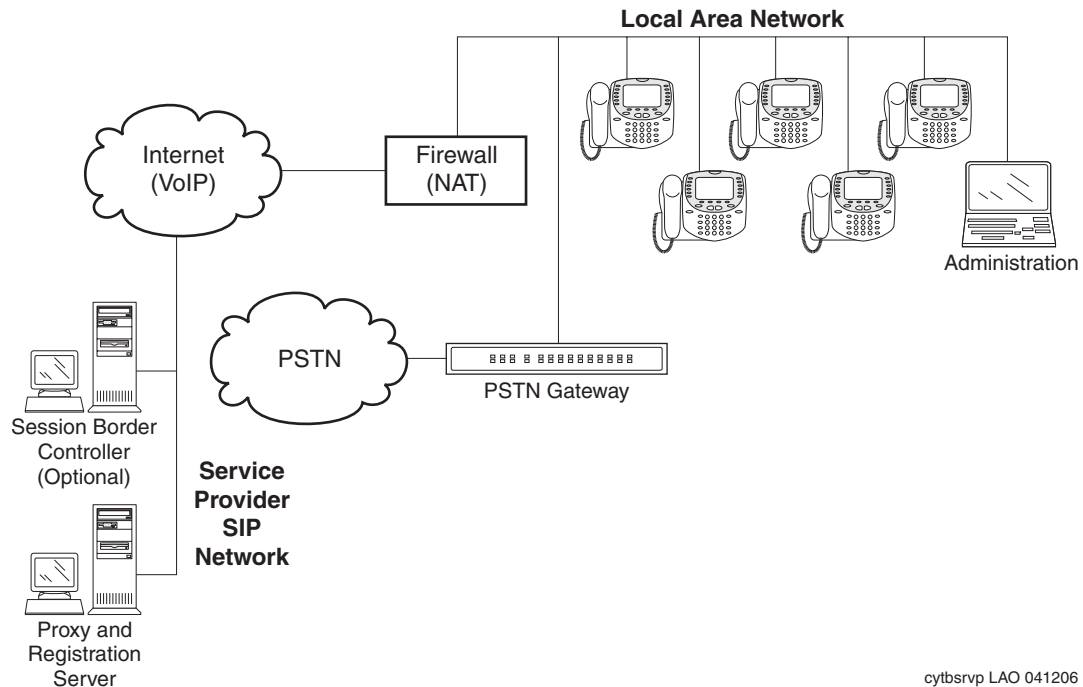
SIP (VoIP) Interoperability

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 details, see [Service Provider](#) on page 59.

[Figure 3](#) 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 a session border controller.

Figure 3: SIP Service Provider Configuration



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 prefix 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 prefix before dialing the number. External calls are routed via a SIP service provider network (Internet).



Tip:

Refer to [Dial Plan](#) on page 56 for your dialing prefix.

To ensure the best quality of service without needing to optimize network performance, plan on a maximum of 20 telephones and PSTN gateways per subnet. 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.

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.

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

Installation Overview

1. Register your system at <https://www.avaya.com/quickregistration>.
2. Connect the first Quick Edition telephone to the LAN.
3. Choose a system language.
4. Enter a site name for the network.
5. Enter and confirm an administration password.
6. Enter a user name for the telephone.
7. Connect additional devices to the LAN.

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 [Web-based User Options](#) on page 91. Refer to [System Options](#) on page 97 for a list of Web-based and telephone interface configurable system options.



CAUTION:

Password protection will prevent unauthorized users from changing system options. If you are the person who manages system-wide options, set a password that is known only to 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.

The password that you enter to log in is encrypted before the information is sent to the system. Refer to [Security](#) on page 71 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. The password must be set when the first telephone is installed on the subnet.
7. Click **Login**.
The **System Options, Device Management** 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 that you set at startup.
5. Select the **Done** softkey. The **System Options** menu is displayed.



Tip:

To view options that are displayed on the next or previous telephone display screens, press the **PAGE LEFT** or **PAGE RIGHT** () button.

Account lockout after failed login attempts

There are two settings that affect a telephone user account login:

1. Login delay after three failed attempts: after every three failed login attempts, this setting will delay the next attempt (default 60 seconds).
2. Attempts before account lockout: the total number of failed attempts allowed before a user is permanently locked out (default 0). The system administrator must reset the user account.

**Tip:**

Refer to [Security](#) on page 71 for details.

Changing the Administration Password

The password that you use to access system options must contain at least four numbers up to a maximum of 32 numbers (default is 6 numbers). The password can be changed as often as necessary to prevent unauthorized access to the system.

Note:

You can change the complexity of passwords, expiration requirements, the frequency of required password changes, etc. For more information, see [Security](#) on page 71.

To change the administration password using a web browser

1. Access the **System Options** web page.
2. In the top right corner of the **System Options** 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**. The **Security** dialog box launches. See [Security](#) on page 71 for descriptions of the **Password Rules**.

**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. Press the **OPTIONS** () button below and to the right of the display.
2. Select **Options** on the **Main** menu.
3. Select **System Options** on the **Options** menu, and log in.
4. Select **Change Password** on the **System Options** menu.
5. Enter the **new password**. See [Security](#) on page 71 for details.
6. Select the **Next** softkey.
7. When you are prompted, re-enter the **new password** and then click **Next**.
8. When you are prompted, enter the **old password** and then click **Next**.
9. Select the **Ok** softkey.

To set a new password when the old one expires using a Quick Edition IP telephone

1. Select **System Options** on the **Options** menu, and log in. You are informed of password expiry and prompted to set a new one.
2. Select the **Ok** softkey.
3. Enter the **new password**, and then select the **Next** softkey.
4. Re-type the **new password** to confirm it and then select the **Next** softkey.
5. Enter your **old password**, and then select the **Next** softkey.
6. Select the **Ok** softkey.
7. Press the PHONE/EXIT () button.

Device Management

Device Management, on the System Options web page, displays all physical devices that make up the Quick Edition network. From this screen, you can select a device for editing; initiate a software upgrade; begin a backup or restore operation.



Tip:

To add a telephone to the system simply connect the telephone to the LAN.

Using 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. See [To add an Auto Attendant configuration using a Quick Edition IP telephone](#) on page 47 for keypad name entry instructions.

To enter a name using the telephone keypad

- 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.

To add or configure devices, refer to the following sections:

- [Add a Teleworker](#) on page 48
- [Gateway Details \(G10/G11\)](#) on page 25
- [A10 Analog Telephone Adapter \(ATA\) Details](#) on page 28
- [G20 ISDN BRI Gateway Details](#) on page 30.

Device Details

To view the list of devices that make up the network

1. Access the **System Options** web page. The **Device Management** list is displayed.
2. Select the sort criterion (Name, Ext., IP Address, Type, Version, Status). For example, click **Ext.** to sort by extension number.
3. Click any device to display the details screen.

To view telephone information from the telephone

1. At the telephone, press the OPTIONS () button.

Note:

To view options that are displayed on the next or previous telephone display screens, press the PAGE LEFT or PAGE RIGHT () 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.

Set Details

Any extension number 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.

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.

The paging feature can be used to broadcast an announcement to a predefined zone of telephones. All persons in the paging zone (except teleworkers) hear the broadcast, unless they are on an active call. The general zone includes all phones on the network and any external paging equipment connected to the 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 edit the name, extension number, or page zone using a web browser

1. Access the **System Options** web page. The **Device Management** list is displayed.
2. In the **Name** column, click the name that corresponds to the device to be changed.
3. In the **Set Details** dialog box, click **Change Details**.
4. In the **Edit Set Details** dialog box, type the new name in **Name** field, type an unused extension number in the **Extension** field, or select a zone from the **Page Zone** list to add the device to that page zone.
5. Click **Submit**.

To change an extension number using the 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.

Note:

You cannot change the name of a telephone from 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*.

Reset Password

To reset the system administration password

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

To reset a user's password using a web browser

1. Access the **System Options** web page. The **Device Management** list is displayed.

2. 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 new user-options and voicemail password is reset to a random value and displayed. Advise the user of the new password which must be changed at first login.

To reset a user's password 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, select the **Yes** softkey.
5. Select the **Ok** softkey.
6. The new user-options and voicemail password is reset to a random value and displayed. Advise the user of the new password which must be changed at first login.

Network Settings

You can specify the IP address of a telephone, the network mask of the IP address, and optionally, the IP addresses of a G10 or G11 gateway and Domain Name Server (DNS). Before you begin, obtain IP addresses for the required network configuration.

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 or customer-specified installations.

To change device network address settings using a web browser

1. Log in to the **System Options** web page. The **Device Management** list is displayed.
2. In the **Name** column, click the Corporate directory name of the device that you want to configure. The **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 device, enter an unused **IP address** in the IP Address field. The address must correspond to the network address space used by the connected subnet (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 encompasses the telephone IP address (for example, **255.255.255.0**). If required, specify a different network mask to qualify the static IP address.
7. To specify the IP address of a 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 subnet (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 encompasses the telephone IP address (for example, **255.255.255.0**). 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.

Optional Features

You can view registration information about special features through the web-based user options interface, the web-based administration interface, or the telephone's Optional Features menu item.

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 (Email Fwd Options, page 54)
- Web-based System Administration (WebAdm Sys Options)
- Teleworker Application (Teleworker Options, page 48)

To view registration information using a web browser

1. Log in to the **System Options** web page.
2. In the Device 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 using a 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 ().

Gateway Details (G10/G11)

See *Avaya one-X Quick Edition G10 PSTN Gateway Installation Guide* (Document No. 16-600793) for complete installation and configuration details; *Avaya one-X Quick Edition Safety and Quick Installation Instructions for: G11 Global Analog Gateway, G20 ISDN BRI Gateway, A10 Analog Telephone Adapter* (Document No. 16-601414) for installation 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.

Note:

If your gateway/adaptor is not a G10/G1, refer to [A10 Analog Telephone Adapter \(ATA\) Details](#) on page 28 or [G20 ISDN BRI Gateway Details](#) on page 30.

To edit the gateway extension number using a web browser

1. Log in to the **System Options** web page.
2. On the **System Options** menu, select **Device Management**.
3. Select the gateway in the **Name** list.
4. Configuration options related to the selected Quick Edition IP are displayed in the **View Gateway Details** dialog box.
5. Click **Change Details** to edit the extension number.

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, 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, select the extension number of the Quick Edition IP that you want to configure.

Lines Settings

Configure the Auto Attendant 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.

By default, incoming calls on all PSTN lines connected to the PSTN gateway are routed to the default Auto Attendant configuration.

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

- 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,
- to make a PSTN line a “private outgoing line,” which forwards outgoing calls from the specified extension to the PSTN line,
- if the port is not used.

For more information and procedures related to disabling the Auto Attendant feature on one or more PSTN lines, see *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 edit an Auto Attendant configuration using a Quick Edition IP telephone](#) on page 45.

To modify gateway line settings

You must assign an Auto Attendant configuration to a PSTN gateway using the web-based System Options interface.

1. Log in to the **System Options** web page.
2. On the **System Options** menu, select **Device Management**.
3. Select the gateway in the **Name** list.
4. Click **Lines** on the **Gateway Details** dialog box.
5. Click **Change Details** on the **View Gateway Lines** dialog box.

Music on Hold (MOH)

There can be only one music source per system. Enable music on the gateway to which the source is connected.

To enable or disable music on hold

1. Log in to the **System Options** web page.
2. On the **System Options** menu, select **Device Management**.
3. Select the gateway in the **Name** list.
4. Click **MOH** on the **Gateway Details** dialog box.
5. Click **Change Details** on the **View Gateway Music On Hold** dialog box to enable or disable MOH.

Network Settings

To modify gateway settings

1. Log in to the **System Options** web page.
2. On the **System Options** menu, select **Device Management**.
3. Select the gateway in the **Name** list.
4. Click **Networking** on the **Gateway Details** dialog box.
5. Click **Change Details** on the **View Gateway Network Info** dialog box.
6. You can change the IP Address, Netmask, and Gateway default IP Address.

Advanced Settings

To modify the loop detect debounce and incoming call guard timer

1. Log in to the **System Options** web page.
2. On the **System Options** menu, select **Device Management**.
3. Select the gateway in the **Name** list.
4. Click **Advanced** on the **Gateway Details** dialog box.
5. Click **Change Details** on the **View Gateway Network Info** dialog box.
6. Enter new values for **Loop Detect Debounce** (default 250ms for G10, 80ms for G11) or **Incoming Call Guard Timer** (default 2000ms).
7. Click **Submit**.

A10 Analog Telephone Adapter (ATA) Details

There are four major steps involved in the installation and configuration of the A10:

1. Install the Quick Edition telephones (refer to the installation document in the phone box).
2. Install the A10 on the LAN (refer to [Figure 4: A10 Analog Telephone Adapter Installation.](#))
3. Assign [SIP Proxy Identities](#) (refer to page 68 for complete SIP procedures).

Note:

The A10 can support up to three analog telephones on each port by using an RJ-11 two or three-port jack adapter (splitter).



Tip:

The A10 is shipped with a default IP address (192.168.123.10).

4. Configure the A10 (see [To configure the A10](#) on page 29).

To add a SIP Proxy identity for A10

1. Log in to the **System Options** web page.
2. On the **System Options** menu, select **SIP Proxy**. The **SIP Proxy Configurations** window is displayed.
3. Click **Identities**.
4. Click **Add**. The **Add SIP Proxy Identity** dialog box displays.
5. Complete the following fields:
 - For **Type**, select **Subscriber**.
 - Enter a unique **Name** (for example, a user name).

- Enter an **Identity**, an internal extension number. The number is used when you assign an extension number in your one-X Quick Edition network to an A10 port.
 - Select a **Domain**.
 - Select an **Authorized User**.
6. Click **Submit**.

To configure the A10

1. Log in to the **System Options** web page.
2. On the **System Options** menu, select **Device Management**.
3. Select the A10 to be configured. The **Edit ATA Gateway Details** dialog includes read-only fields for Device Type, MAC Address, and Firmware (date of the firmware build).
4. Name the A10 and select the SIP Identity for each port. For example, map the SIP Proxy Identity added in the previous procedure, to Port 1. You can continue to configure the other three ports.
5. If you need to change the network information, clear the **DHCP Address** check box. Refer to [Assigning a Static IP Address](#) on page 89 for the procedure to set a static IP address.

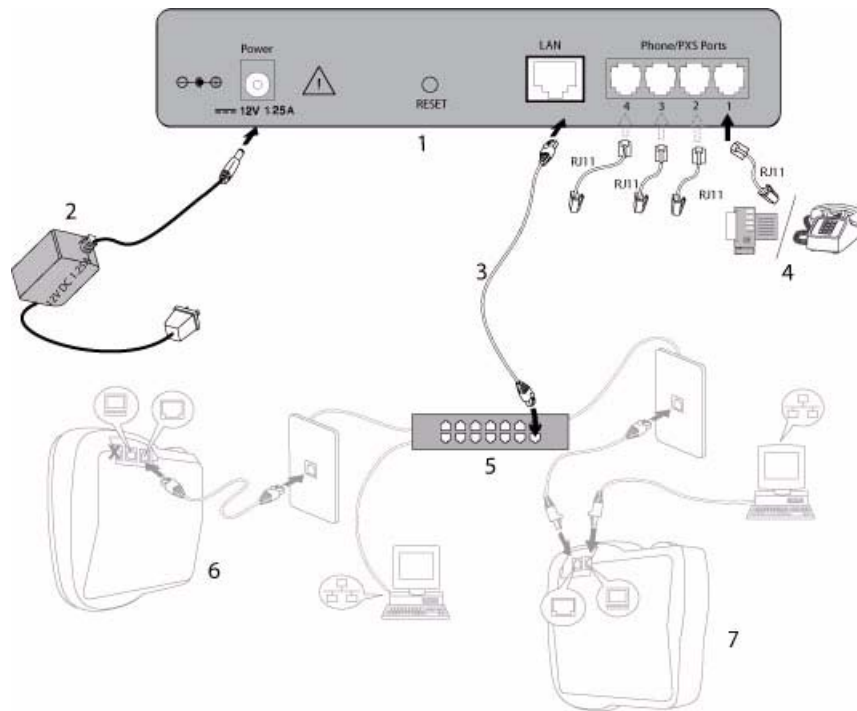


WARNING:

Changing the network settings to assign a static IP address could cause loss of network connectivity with the A10. Each time you clear or select **DHCP Address**, and click **Submit**, the A10 will be reset.

6. Click **Submit**.

Figure 4: A10 Analog Telephone Adapter Installation



G20 ISDN BRI Gateway Details

There are five major steps involved in the installation and configuration of the G20 ISDN BRI Gateway (BRI):

1. Install the Quick Edition telephones (refer to the installation document in the phone box).
2. Purchase DDI/MSN lines and protocol information from your service provider.
3. Install the G20 on the LAN (refer to [Figure 5: G20 ISDN BRI Gateway Installation](#) or the installation document in the G20 box).
4. Assign [SIP Proxy Identities](#) (refer to page 68 for complete SIP procedures).



Tip:

The G20 is shipped with a default IP address (192.168.123.10).

5. Map the identities (see [To configure the G20](#) on page 31).

To add a SIP Proxy identity for G20

1. Log in to the **System Options** web page.
2. On the **System Options** menu, select **SIP Proxy**. The **SIP Proxy Configurations** window is displayed.
3. Click **Identities**.
4. Click **Add**. The **Add SIP Proxy Identity** dialog box displays.
5. Complete the following fields:
 - For **Type**, select **Trunk**.
 - Enter a unique **Name** (name assigned to the user of this DDI/MSN number).
 - Enter an **Identity**; the telephone number (no dashes or spaces) in the DDI/MSN range supplied by your service provider. This will enable calls coming in through the SIP network to be delivered to a specific telephone and/or handled by the system-wide Auto Attendant feature. The one-X Quick Edition system makes each identity that you assign available to any telephone in the network.

Note:

Typically, the first number in the DDI/MSN range of numbers from your service provider is assigned as the Primary Registerer for each port.

- Accept the default **Domain** and **Authorized User**.
 - Select the **Primary Registerer**. Select **Create As Registerer** to turn the current identity number into a primary registerer or select another identity number to be the primary registerer. When a primary registerer identity is assigned to a G20 port, all identities associated with the primary registerer are assigned to that port.
 - Select an **Incoming Extension** number. Select the auto attendant, the telephone extension, the group extension, or the SIP identity that will be handling incoming calls for the specified identity.
 - Select an **Outgoing Extension** number. Select the telephone extension, group extension, or the SIP identity 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 or the Auto Attendant answers incoming external calls automatically, select the **Global** option.
6. Click **Submit**.

To configure the G20

1. Log in to the **System Options** web page.
2. On the **System Options** menu, select **Device Management**.
3. Select the G20 to be configured from the **Name** list. The **Edit ISDN Gateway Details** dialog includes read-only fields for Device Type, MAC Address, and Firmware (date of the firmware build).

4. Name the ISDN BRI and select identity and protocol values for **Port** 1 and 2.

- The Device Name should be the Primary Registerer for that port.

Layer 2

Layer 2 allows a station to reliably send messages to another station using the D channel. Layer 2 implements flow control, error detection and correction (retransmission) as well as addressing mechanism to direct messages to individual devices.

If you select DID (Direct Inward Dialing), the G20 port is using point-to-point topology.

If you select MSN (Multiple Subscriber Number), the G20 port is using point-to-multipoint topology.

Layer 3

Layer 3 does send and receive application level messages (i.e. call control). It sends broadcast messages and collects the individual results of the attached devices. It also handles the assignment of the B channels.

If you select DSS1, layer 3 uses DSS1 signalling protocol.

If you select DMS-100, layer 3 uses Digital Multiplex System signalling.

Layer 3 UniSide

User is selected by default, which means that the G20 port runs user side part of signalling protocol.

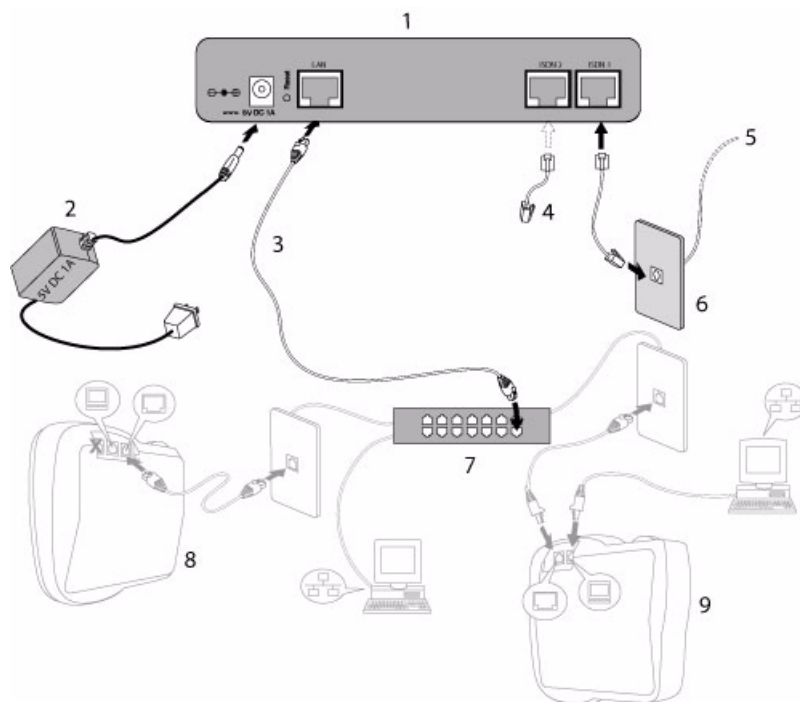
5. If you need to change the network information, clear the **DHCP Address** check box. Refer to [Assigning a Static IP Address](#) on page 89 for the procedure to set a static IP address.



WARNING:

Changing the network settings assigns a static IP address and could cause loss of network connectivity with the G20. Each time you clear or select **DHCP Address**, and click **Submit**, the G20 will be reset.

6. Click **Submit**.

Figure 5: G20 ISDN BRI Gateway Installation


Software Upgrade

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 QE Install Wizard that includes the software and language packs.

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.

Note:

When you have used the system web interface and then decide to change to another language or upgrade to a new software version, you must clear your browser cache (history).

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. You can perform an upgrade using one of these tools:

- Install Wizard
- System Administration web interface
- Quick Edition telephone
- Multisite Provisioning Tool.

In most cases the software upgrade package must be installed on a computer that can access the 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 the Install Wizard

The Install Wizard is the preferred method for upgrades because the wizard will discover the network and it will launch the TFTP server.

1. Launch the QE_Install_Wizard_vxxxxx.exe and follow on screen instructions to upgrade all or selected devices in the Quick Edition network.
2. After the upgrade process completes, reset the [System Time and Date](#) (see page 76).

To upgrade the software using a web browser

1. Log in to the **System Options** web page (see [To access options using the web-based administration interface on page 17](#)).
2. On the **Device Management** dialog, click **Software Upgrade**.
3. In the **TFTP Server** field, type the IP address of the computer on which the TFTP server is running.
4. Upgrade all or selected devices in the Quick Edition network.
5. Click **Prepare Upgrade**.
6. Click **Start Upgrade**.
7. When you are prompted to begin the upgrade process, click **Confirm Upgrade**.
8. After the upgrade process completes, 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.
10. Reset the [System Time and Date](#) (see page 76).

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 18](#)).
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 necessary, 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.
7. Select the **Next** softkey.
8. Choose the **Upgrade Option**:

- **Clean Database**—When selected, the software is upgraded and all user-configurable settings and system-configuration data are erased (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 devices 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.
9. Select the **Next** softkey.
 10. When you are prompted to start the upgrade process, select the **Yes** softkey.
 11. At the administration computer, exit the installer program.
 12. Reset the [System Time and Date](#) (see page 76).

To upgrade the software using the Multisite Provisioning Tool

The Multisite Provisioning Tool is a Java-based software application that communicates with one-X Quick Edition networks over an HTTPS link. The Multisite Provisioning Tool 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 represented in the Multisite Provisioning Tool database can be configured at once.

You can obtain the 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.

1. On the Windows **Start** menu, point to **All Programs > Avaya one-X Multisite Provisioning Tool**, and then click **Avaya one-X Multisite Provisioning Tool**.
2. Follow on screen instructions to upgrade all or selected devices.
3. Reset the [System Time and Date](#) (see page 76).

Backup and Restore

Use the System Options web interface to backup and restore system data. You can also backup and restore any or all user data from the User Options web interface. Keep a current data backup on your PC to replace lost data, in the case of inadvertently deleting a file or replacing defective hardware, or to restore data when you upgrade software. New software provides some generic values; restoring your data will eliminate the need to customize those values.

The transfer of data is secure and the files stored on the PC are encrypted. The date and time and the software version are included as part of the backup. Passwords are not backed up, nor is the data for a device that is not connected.

The file to restore must have been created by a software load no older than one major release (i.e. if the active software load is R5.x.y then the restore is supported for any backup file created by a software load back to R4.0.0). However a restore from a file created by a newer software load than the active load shall not be supported (i.e. if R4.0.0 is the active software load then the backup file must not have been created after R4.0.0).

System Configuration Data

The encrypted system backup file will include the following information:

Language	Audio bandwidth
Region	Email notification settings
Site name	Authorization codes
Dialing plan	Security policy
Auto attendants and their settings	Applied dialing rules
Groups and their settings	Customizable prompts
Service provider configurations	Call detail recording settings
Gateways and their settings	A10 and G20 configuration data

To back up system configuration data using a web browser

1. Log in to the **System Options** web page (see [To access options using the web-based administration interface on page 17](#)).
2. On the **Device Management** home page, click **Backup & Restore**.
3. Click **Download Backup** to launch the **Save as** window.
4. Select the location for the backup file and click **Save**.

To restore system configuration data using a web browser

1. Log in to the **System Options** web page (see [To access options using the web-based administration interface on page 17](#)).
2. On the **Device Management** home page, click **Backup & Restore**.
3. **Browse** to the backup file.
4. Click **Restore**.

Telephone Configuration Data

The encrypted user data backup file will include the following information:

Extension	Call forward settings
Name	Current ring tone
Language	Page zone
Personal directory	Presence monitoring list
Speed dial directory	Configured presence status
Voice mail settings	Programmable soft keys

To back up telephone user configuration data using a web browser

1. Log in to the User Options interface (see [Logging On](#) on page 91).
2. Select **Backup & Restore** in the User Options menu.
3. Click **Download Backup** to launch the **Save as** window.
4. Select the location for the backup file and click **Save**.

To restore telephone user configuration data using a web browser

1. Log in to the User Options interface (see [Logging On](#) on page 91).
2. Select **Backup & Restore** in the User Options menu.
3. **Browse** to the backup file.
4. Click **Restore**.

Corporate Directory

The Corporate Directory dialog box will display all of the dialable numbers, internal and external, that make up the Corporate Directory. The Corporate Directory includes numbers associated with physical devices such as telephones, and numbers associated with virtual applications such as groups and auto attendants.

Note:

Selecting a phone in the Corporate Directory dialog launches the [Set Details](#) dialog (page 21) in the Device Management list; selecting a gateway launches the [Gateway Details \(G10/G11\)](#) dialog (page 25) in the Device Management list.

Add an Entry

1. Log in to the **System Options** web page.
2. On the **System Options** menu, click **Corporate Directory**.
3. Click **Add Entry** to launch the **Add Corporate Directory Entry** page.
4. Select an **Entry Type** from the drop-down list.
5. Complete the type-specific fields that will appear:
 - Groups (see [Group Details on page 40](#))
 - Auto Attendants (see [Auto Attendant Details](#) on page 43)
 - Extension (see [Extension](#) on page 48)
 - External Entries (see [External Number](#) on page 52).

Remove an Entry

The following procedures remove an extension for devices from the system and free the extension number so that it can be reused. This is especially important when replacing a defective device. Removing an auto attendant, group, or external entry will removal of that entry from the corporate directory.

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 an entry using a web browser

1. Log in to the **System Options** web page.
2. On the **System Options** menu, click **Corporate Directory**.
3. Click **Remove Entry** to launch the **Remove Corporate Directory Entry** page.
4. Select the entry to be removed from the drop-down list.
5. Click **Remove**.
6. Click **Remove Entry** to confirm.
7. Continue if you are removing a telephone extension:
 - At the telephone, press **Ok** to reset. All configuration changes will have been lost and the device will be in a factory default state.
 - Enter a name for the telephone.

To remove an entry 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 device, enter the extension number of the device, and then select the **Next** softkey.
5. When you are prompted to remove the extension, select the **Yes** softkey.
6. At the message “Set Has Been Removed From Network”, press **Reset**. After the reset, all configuration changes will have been lost; the device will be in a factory default state.
7. Enter a name for the telephone.

Group Details

A maximum of nine groups, each having a maximum of 10 members, is supported. 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. Create and administer groups using the System Options web tool or the Multi-Site Provisioning Tool (MPT).

To add a group

1. Log in to the **System Options** web page.
2. On the **System Options** menu, select **Corporate Directory**.
3. In the upper-right corner of the dialog box, click **Add Entry**.
4. Select **Groups** from the drop-down **Entry Type** list.
5. In the **Group Name** field, type a name for the group.
6. 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. You cannot assign the same extension number to more than one group.
7. Click **Submit**. The new group is displayed in the **Groups** list.

To delete a group

1. In the **Corporate Directory** page, navigate to the selected group.
2. With the **View Group Details** dialog box on display, click **Delete Group**.
3. Click **Submit**.

To edit group details

1. In the **View Group Details** area, click **Change Details**.
2. You can change the **Group Name** in this edit screen.
3. Click **Submit**.

Group Members

To add members to a group

1. In the **Corporate Directory** page, navigate to the selected group.
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.

Group Forwarding

Call-forwarding features let you specify rules for handling calls that are routed to a group.

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.

Note:

Group forwarding will not work if all members are analog extensions connected to an A10. At least one member of the group must be a Quick Edition phone.

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 [Specify the Designated Operator Extension on page 58](#)).
6. Click **Submit**.

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.



Tip:

To create a group that includes all telephones in the system, refer to [Global Dialing Rules](#) on page 53.

To apply a dialing rule to a group

1. Log in to the **System Options** web page.
2. On the **System Options** menu, select **Corporate Directory**.
3. On the **Corporate Directory** page, click the **Group** extension.
4. In the **View Group Details** dialog box, click **Dialing Rules**.
5. Click **Add Rule**. The **Edit Group Add Dialing Rules** dialog box is displayed.
6. From the **Add Rule** drop-down list, select one of the 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**.

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 [Dialing Rules on page 57](#).

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 dialing rule from a group

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

Note:

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

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

To enable or disable a dialing rule for a group

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

Note:

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

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

Auto Attendant Details

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

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 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.

**Tip:**

To configure the Auto Attendant feature to answer calls from a PSTN gateway, see [Lines Settings](#) on page 26.

**Tip:**

To configure the Auto Attendant feature to answer calls from a SIP network, see [To assign a SIP identity on page 63](#).

Record Custom Greetings

You can record two custom 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.
5. Select the **Rec** softkey, and speak the greeting clearly into the microphone. The greeting must be a minimum of 3 seconds and a maximum of 2 minutes in length.
6. When you are finished, select the **Stop** softkey.
7. Perform one of the following actions:
 - To listen to your recording, select **Play**.
 - If you are satisfied with the recording, select **Save** followed by **Ok**.
 - If necessary, select the **Rec** softkey to re-record the greeting.

Apply Custom Greetings - Auto Attendant Switching (Night Service)

The operator or an authorized user can change the auto attendant greeting from default to a custom greeting. You can assign a programmable softkey on any telephone, to enable the activation of an alternate greeting, typically for night switching to an after-hours greeting.

To assign greeting to a Quick Edition IP telephone softkey

1. At the Operator or other assigned telephone, press the selected softkey for two seconds to access the programmable softkeys menu.
 - The screen will display all possible options. A check mark indicates the current function assigned to the key and a 'D' indicates the default function of the key.
2. Select the Line/Feature button beside the **Greeting** function. Press the PAGE LEFT or PAGE RIGHT (◀▶) button to view all options.

- See “Programmable Softkeys” in the Telephone User Guide to program a softkey.
- 3. Enter the system administration password at the prompt.
- 4. Click **OK** at the “key assigned” message.

Edit Auto Attendant Configurations

You can 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 edit an Auto Attendant configuration using a web browser

1. Log in to the **System Options** web page.
2. On the **System Options** menu, select **Corporate Directory**.
3. On the **Corporate Directory** dialog, select the Auto Attendant that you want to 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, click **Change Details**.
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 [44](#).

7. Click **Submit**.

To 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.
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 6 in [To add an Auto Attendant configuration using a Quick Edition IP telephone](#) on page 47 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 44](#)), 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.

Create a Custom Auto Attendant Configuration

Create a custom Auto Attendant configuration using the web-based interface or a telephone.

To create a custom Auto Attendant configuration using a web browser

1. Log in to the **System Options** web page.
2. On the **System Options** menu, click **Corporate Directory**.
3. Click **Add Entry**.
4. Select **Auto Attendant** from the **Entry Type** drop-down list.
5. 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.
6. In the **Name** field, type a name for the configuration.
7. In 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 44](#).

8. 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 and click **Add**.
4. Using the keys on the dialpad, enter an unused extension number in the auto attendant range.
5. Select the **Next** softkey.
6. Using the keys on the dialpad and the softkeys, enter a name for the Auto Attendant configuration (see [To enter a name using the telephone keypad](#) on page 20).
7. Select the **Next** softkey.
8. Perform one of the following actions:
 - To select the default greeting, select **Default** on the **Select AA Prompt** menu.
 - If you recorded or intend to record a custom greeting (see [To record a custom greeting on page 44](#)), press the indicated dialpad key to select the greeting.

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

Delete 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 that is assigned to extension 500.

Note:

To select an Auto Attendant configuration for a PSTN gateway, see [Lines Settings](#) on page 26).

To delete a custom Auto Attendant configuration using a web browser

1. Log in to the **System Options** web page.
2. On the **System Options** menu, select **Corporate Directory**.
3. In the **Number** 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:

You cannot delete a configuration that is associated with a PSTN gateway.

5. Click **Submit**.

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

1. Use a telephone on to the network to display the **System Options** menu (see page [18](#)).
2. Select **Auto Attendant** on the **System Options** menu, or press the indicated dialpad key.
3. Select **Auto Attendant** on the **Auto Attendants** menu.
4. Select the Line/Feature (►) button beside the configuration that you want to delete.
5. Select the **Del** softkey.
6. When you are prompted to confirm deletion, select the **Yes** softkey.

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 17](#)).
2. Click **Corporate Directory** and select the Gateway number.
3. Click **Lines**. The **View Gateway Lines** dialog box displays.
4. Click **Change Details**. The **Edit Gateway Lines** dialog box displays.
5. From the **Incoming** list that is associated with the line, select your operator extension.
6. Select **Submit**.

Extension

Add a Teleworker

Workers at remote offices can have access to the Corporate directory and most of the features and services that are available to the one-X Quick Edition system. You can add a teleworker client telephone locally or from a remote office.

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.

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.

**Tip:**

See [Supporting Telecommuters](#) on page 12 for high-level teleworker configuration examples.

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 new 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.

Adding a teleworker client to the network, locally

1. Connect the new telephone to the one-X Quick Edition network.
2. Add the telephone to the LAN (see page 20).
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 and passing traffic in both directions.
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 52](#).

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** web page.
2. In the **Device Management** list, click **Add Device**.
3. Select **Phone** from the **Device Type** drop-down list.
4. In the **Extension** field, type an unused extension number for the teleworker client phone.
5. In the **MAC** field, type the 12-digit MAC address of the teleworker client telephone (for example, **00:00:5A:99:62:50**).

Note:

The MAC address is on a sticker on the underside of the telephone. You may have to remove the base to view the sticker.

6. Click **Add Device**.
7. 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 [Teleworker on page 93](#).

Configuration and programming at the remote location

1. At the remote office, configure a VPN tunnel (see Step 4 in [Adding a teleworker client to the network, locally](#)).
2. Verify that the VPN tunnel is operational (see Step 5 in [Adding a teleworker client to the network, locally](#)).
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 of any active Quick Edition telephone on the network in the corporate (internal) location. 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.
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](#).

External Number

Like all corporate directory entries, external numbers will appear in the Directory of each telephone in the system. The name and number can be a maximum of 32 characters.

To add an external telephone number to the corporate directory

1. Log in to the **System Options** web page.
2. On the **System Options** menu, click **Corporate Directory**.
3. Click **Add Entry** to launch the **Add Corporate Directory Entry** page.
4. Select **External Entry** from the **Entry Type** drop-down list.
5. Type the telephone number including required PSTN or SIP prefixes.
6. Type the name.
7. Click **Create**.

SIP Identity

Clicking a SIP Identity (A10 extension) on the Corporate Directory dialog will launch the Edit [SIP Proxy Identities](#) dialog. See page 69 for add, edit, and remove SIP Proxy Identity procedures.

Global Dialing Rules

You will apply a dialing rule from a drop-down list; first, you may want [To create a custom dialing rule pattern](#) on page 57. When you configure a global dialing rule, it will apply to all telephones in the system.

Note:

To create a group of up to ten telephones, see [Group Details on page 40](#).

To apply a dialing rule for all telephones

1. Log in to the **System Options** web page.
2. On the **System Options** menu, select **Corporate Directory**.
3. On the **Corporate Directory** page, click **Global Dialing Rules**.
4. In the **View Global Dialing Rules** dialog box, click **Add Rule**.
5. From the **Add Rule** drop-down list, select one of the options. Your choice determines which calls will be affected by the rule. You can apply rules, depending on region; refer to [To apply a dialing rule to a group](#) on page 42.
6. 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.
7. Click **Submit**. The new rule is added to the **View Group Dialing Rules** list.

To delete a dialing rule from all telephones

1. With the **View Global Dialing Rules** dialog box on display, click **Remove Rule**.
2. Select the rule from the **Delete Global Rule** drop-down list.
3. Click **Submit**.

To enable or disable a dialing rule for all telephones

1. With the **View Global Dialing Rules** dialog box on display, click **Change Details**.
2. Select **Allow** or **Disallow** beside the rule that you want to edit.
3. Click **Submit**.

Applications

SMTP

E-mail Forwarding of Voicemail

SMTP (Simple Mail Transfer Protocol) is used to send email messages from devices on the network. SMTP settings are used to configure communications with your SMTP server. When configured and enabled, the email-forwarding-of-voicemail feature sends e-mail messages containing 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 forwarded as an e-mail attachment.

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

1. Enable SMTP on the one-X Quick Edition network to enable a telephone to forward e-mail messages from voicemail on the phone to an SMTP server. You can specify the IP address of the SMTP server using a web browser or buttons on a telephone.
2. Enable the feature and specify the e-mail address of the person to whom notification will be sent, through the web-based **User Options** interface. To play the audio file attachment, see the *Avaya one-X Quick Edition Telephone User Guide*, “Email Forwarding of Voicemail”.

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.

To configure SMTP using a web browser

1. Log in to the **System Options** web page (see [To access options using the web-based administration interface on page 17](#)).
2. On the **System Options** menu, select **Applications**. 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 (default port 25).
7. Click **Submit**.

To configure 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.
7. Select the **Next** softkey.
8. If your network uses a different IP port number (default port 25) for SMTP communications, specify the port number.
9. Select the **Next** softkey and then the **Save** softkey.

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

- Refer to the *Avaya one-X Quick Edition Telephone User Guide*, “Email Forwarding of Voicemail”.

To play an audio file attachment

- Refer to the *Avaya one-X Quick Edition Telephone User Guide*, “Email Forwarding of Voicemail”.

Call Detail Recording

Avaya one-X Quick Edition provides the ability to capture information about all phone calls. Refer to [Call Detail Recording Fields](#) on page 100 for record categories, details, and examples.

You can provide the captured ASCII CSV format data to a designated Call Detail Records (CDR) collecting server to generate reports. Such reports can help eliminate internal telephone abuses, allocate communication costs across an organization, guard against external telephone frauds, provide centralized and local visibility of telecommunication costs for worldwide networks, measure leased line (private wire usage), and identify missed and unanswered calls.

To configure call detail recording using a web browser

1. Log in to the **System Options** web page.
2. On the **System Options** menu, select **Applications**.
3. In the **SMTP Settings** page, click **CDR**.
4. Select **Enable CDR**.
5. Enter the IP address and port for the **Primary CDR Server**.

6. Enter the **Collection Interval**.
7. Enter the **Max CDR Per Batch**; the maximum number of records to be sent to the server.
8. Enter the **Max Records**; the local storage limit for the device.
9. Click **Submit** and log out.

Dialing Configuration

Dial Plan

When you set the region during the install, the system configures itself using the default set of extension numbers, numbers to access emergency services, the PSTN prefix, and a SIP (VoIP) network prefix.

If 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 that region. See [System Region Settings](#) on page 75.

Because the dial plan is configurable, you can 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 require some manual configuration 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** web page.
2. On the **System Options** menu, select **Dialing Configuration**.
3. Click **Edit 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 OPERATOR.

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 applied globally or to a group, either allow or prevent users from making certain types of calls. See [Group Dialing Rules](#) on page 42 and [Global Dialing Rules](#) on page 53.

To view dialing rule patterns

1. Log in to the **System Options** web page.
2. On the **System Options** menu, select **Dialing Configuration**.
3. Click **Dialing Rules** in the **Dial Plan Settings** dialog box.

To create a custom dialing rule pattern

1. Log in to the **System Options** web page.
2. On the **System Options** menu, select **Dialing Configuration**.
3. Click **Dialing Rules** in the **Dial Plan Settings** dialog box.
4. Click **Create Rule**.
5. In the **Rule Description** field, type a description of the rule.
6. 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???????**.

7. Click **Submit**. The new rule is added to the Rule Description list.

To delete a custom dialing rule pattern

1. Log in to the **System Options** web page.
2. On the **System Options** menu, select **Dialing Configuration**.
3. Click **Dialing Rules** in the **Dial Plan Settings** dialog box.
4. Click **Delete Rule**.
5. From the **Delete Rule** drop-down list, select the rule that you want to delete. You cannot delete the default dialing rules. Click **Submit**.

Operator

Specify the Designated Operator Extension

You can choose the telephone that you want to designate as the operator extension. Initially, a region-specific default extension number receives 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** web page.
2. In the **System Options** list, click **Dialing Configuration**.
3. Click **Operator** on the **Dial Plan Settings** dialog box.
4. Click **Change Details**. The **Edit Operator Extension** dialog box is displayed.
5. From the **Operator Extension** list, select the extension.
6. 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 click **Next**.
 - To choose an extension number from the Corporate directory, select the **FrDir** softkey and then select the entry.
6. Select the **Save** softkey.

To change the operator extension number

- Refer to the procedures in [Set Details](#) on page 21 to change the operator extension number from the default 200.

Service Provider

Certified SIP (Session Initiation Protocol) VoIP service provider options can be configured through the web-based administration interface. The 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:

See also, *Configuring SIP Trunking to Quick Edition 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). 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 3: SIP Service Provider Configuration](#) on page 14. 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.

Required Service Provider Information

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:

Proxy:	sip.sipexample.com	Port: 5060 ¹
Registrar:	sip.sipexample.com	Port: 5060
Session Border Controller:	outproxyhost.sipexample.com	
Domain:	sipexample.com	
Realm:	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):

SIP identity 1:	1715557654	Password: 12000
SIP identity 2:	1715557655	Password: 12001
SIP identity 3:	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:

SIP identity:	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.

SIP Interoperability

Configuring SIP interoperability involves four steps:

1. Obtain [Required Service Provider Information](#) on page 60 information, 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. Create a service provider configuration. See [To add a service provider configuration on page 61](#).
3. If you intend to route incoming calls to a group, refer to [Case C: Incoming Calls are Routed to a Group on page 67](#) and [Group Details on page 40](#).
4. Assign SIP identities. See [To assign a SIP identity on page 63](#).

To add a service provider configuration

1. Log in to the **System Options** web page.
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** web page.
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 on page 61](#).
5. When you are finished, click **Submit**.

To delete a service provider configuration

1. Log in to the **System Options** web page.
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

Configure the Auto Attendant 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.

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

1. Log in to the **System Options** web page.
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.
 - 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** web page.
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** web page.
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. 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.

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 61](#).
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 65](#).
 - Does everyone have a direct line? If so, see [Case B: Everyone has a Direct Line on page 66](#).
 - 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 67](#).
 - 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 67.

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 63](#)). 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 63](#)). 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:

SIP Identity:	1715557600
Password:	12700
Member (Outgoing Calls):	Global
Member (Incoming Calls):	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 63](#)). 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:

SIP identity:	1715557654	1715557655	1715557656
Password:	12000	12001	12002
Member (Outgoing Calls):	201	202	203
Member (Incoming Calls):	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 [Group Details on page 40](#)). Next, configure one SIP identity that corresponds to the group extension number (see [To assign a SIP identity on page 63](#)). 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:

SIP Identity:	1715557600
Password:	12700
Member (Outgoing Calls):	405
Member (Incoming Calls):	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 63](#)). 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 63](#)). 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:

SIP Identity:	1715557600
Password:	12700

Member (Outgoing Calls):	Global
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Member (Incoming Calls):	Global
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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).

SIP Proxy

The SIP Proxy option allows you to add the values that are used to configure the SIP devices (that is, A10 and G20) on the one-X Quick Edition network.

SIP Proxy Configuration

SIP proxy configurations define the domains with which SIP identities associate. When you select a domain for an identity, the identity belongs to that domain.



Tip:

The A10 and G20 devices register to the Quick Edition network using the SIP Proxy functionality.

To edit a SIP Proxy configuration

1. Log in to the **System Options** web page.
2. On the **System Options** menu, select **SIP Proxy**. The **SIP Proxy Configurations** window is displayed.
3. Click the **Domain Name** that you want to modify.
4. Make the changes to the **Realm** and **Expiry** time fields.
5. Click **Submit**.

SIP Proxy Identities

To add a SIP Proxy identity for A10 or G20

1. Log in to the **System Options** web page.
2. On the **System Options** menu, select **SIP Proxy**. The **SIP Proxy Configurations** window is displayed.
3. Click **Identities**.
4. Click **Add**. The **Add SIP Proxy Identity** dialog box displays.
5. Complete the following fields:
 - For **Type**, select **Subscriber** (A10) or **Trunk** (G20).
 - Enter a unique **Name**.
 - Enter an **Identity** in numeric format.
 - **A10**: Add internal extension numbers as identities. The values are used when you assign an extension number in your one-X Quick Edition network to an A10 port.
 - **G20**: Assign identities (typically a telephone number), from by your service provider, to devices on your network. This will enable calls coming in through the BRI PSTN network to be delivered to a specific telephone and/or handled by the system-wide Auto Attendant feature. The one-X Quick Edition system makes each identity that you assign available to any telephone in the network.
 - Accept the default **Domain** and **Authorized User**.
 - The following three settings apply only to the G20:
 - Select the **Primary Registerer**. Select **Create As Registerer** to turn the current identity number into a primary registerer or select another identity number to be the primary registerer. When a primary registerer identity is assigned to a G20 port, all identities associated with the primary registerer are assigned to that port
 - Select an **Incoming Extension** number. Select the auto attendant, the telephone extension, the group extension, or the SIP identity that will be handling incoming calls for the specified identity
 - Select an **Outgoing Extension** number. Select the telephone extension, group extension, or the SIP identity 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 or the Auto Attendant answers incoming external calls automatically, select the **Global** option.
6. Click **Submit**.

To edit a SIP Proxy identity

1. Log in to the **System Options** web page.

2. On the **System Options** menu, select **SIP Proxy**. The **SIP Proxy Configurations** window is displayed.
3. Click **Identities**.
4. Select the Identity that you want to modify, **Subscriber** (A10) or **Trunk** (G20).
5. Make the changes.
6. Click **Submit**.

To remove a SIP Proxy identity

1. Log in to the **System Options** web page.
2. On the **System Options** menu, select **SIP Proxy**. The **SIP Proxy Configurations** window is displayed.
3. Click **Identities**.
4. Click **Delete**. The **Delete SIP Proxy Identity** dialog box displays.
5. Select the identity to be deleted from the drop-down list.
6. Click **Submit**.

SIP Proxy Authorized Users

A SIP authorized user account, is added to a proxy identity, is used for authenticating SIP-based communications between the A10 or G20 and the one-X Quick Edition telephones.

To edit a SIP Proxy Authorized User account

1. Log in to the **System Options** web page.
2. On the **System Options** menu, select **SIP Proxy**. The **SIP Proxy Configurations** window is displayed.
3. Click **Authorized Users**.
4. Select the authorized user account that you want to modify.
5. Make the changes.
6. Click **Submit**.

Security

General

Password Rules

To modify the password rules

1. Log in to the **System Options** web page.
2. On the **System Options** menu, select **Security**.
3. Click **Change Details** to modify any of the following rules:
 - Minimum password length: (4 to 32 digits; default is 6)
 - Password history depth: (0 - 12 previous passwords that must not match; default is 1)
 - Days before password expiry: (customer specified; default is 0 for no expiration)
 - Days of warning before password expiry: (customer specified; default is 10)
 - Attempts before account lockout: (customer specified number of failed attempts allowed before a user is permanently locked out; default 0 for no lockout)
 - Login delay after three failed attempts: (customer specified delay time after every three failed login attempts; default is 60 seconds)
 - Enable password-strength protections. Default is enabled (entirely sequential and repeated numbers are not allowed.)
4. Click **Submit**.

Examples:

If the “Attempts before account lockout” remains at 0, the user will never be locked out.

If the “Attempts before account lockout” is less than 4, the “Login delay after three failed attempts” will never be invoked.

Password history depth:

- 0 means that you can reuse any password without restriction;
- 1 means that you cannot reuse the current password;
- 2 means that you cannot reuse the current password or the one before it.
- 6 means that you would have to go back in 'history' to the seventh past password.

Note:

Password expiry warning appears at log in.

Authorization Codes

Authorization codes can be used to override dialing restrictions. The authorization codes are generated and assigned in the System Options web page. Authorization code information will be tracked as part of the [Call Detail Recording](#) feature (see page 55) and, for security reasons, displayed as a “user” or “description” string instead of the real authorization code.

When authorization codes is enabled, the user will dial a restricted number and will be prompted by a stutter tone to enter the authorization code. The call will complete after entry of the code. The code is created as a random 6-10 digits with a unique authorization user name ID. A maximum of 50, 10-digit codes is permitted. Emergency calls will always override a restriction.

To create an authorization code

1. Log in to the **System Options** web page.
2. Select **Security** in the System Options menu.
3. Click **Authorization Codes** in the **Security** page.
4. Click **Create Code**.
5. Enter a name for the authorization code.
6. Enter a random 6-10 digit numeric authorization code.
7. Click **Submit**.

Note:

You may not edit an authorization code; you must delete it and create a new one.

To delete an authorization code

1. Log in to the **System Options** web page.
2. Select **Security** in the System Options menu.
3. Click **Authorization Codes** in the **Security** page.
4. Click **Delete Code**.
5. Select the code from the drop-down list.
6. Click **Submit**.

Admin Password

To change the administration password

1. Log in to the **System Options** web page.
2. Select **Security** in the System Options menu.

3. Click **Admin Password** in the **Security** page.
4. Type the **Current Password**.
5. Type the **New Password**.
6. **Confirm Password**.
7. Click **Submit**.

Note:

You can also change the password using the **Change Admin Password** link on the top right of each web page.

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
- France
- Singapore
- Thailand
- Spain
- Netherlands
- Belgium
- Switzerland
- South Africa
- Austria
- Mexico

The following languages are supported:

- American English
- British English

- German
- Italian
- Canadian French
- Parisian French
- Castilian Spanish
- Latin American Spanish
- Dutch

System Language Settings

The language settings procedures will change the system language and the telephone user interface (UI). Refer to [Language Packs](#) on page 76 to change the voicemail and auto attendant prompts, and web interface.

Note:

When you have used the system web interface and then decide to change to another language or upgrade to a new software version, you must clear your browser cache (history).

To change system language settings using a web browser

1. Log in to the **System Options** web page.
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 18](#)).
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.

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** web page.
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.

Conference Tones

To modify the conference tones

1. Log in to the System Administration web interface.
2. Click **Localization** in the System Options menu.
3. Click **Change Details** in the **View System Region** section of the **Language & Region** web page.
4. Click in the **Conference Tone** check box to enable conference tones.

Note:

The conference tone is disabled by default and can be enabled only for Italy

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.

Note:

Use this procedure if you chose not to install the language packs during the install or an upgrade.

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 Time and Date

To change the system time or date using a web browser

1. Log in to the **System Options** web page.
2. On the **System Options** menu, select **Localization**.
3. On the **Language & Region** page, select **Time & Date**.

4. Select **Change Details** in the **View System Time** or **View System Date** area.
5. In the **Date** fields, enter the system date.
In the **Time** fields, enter the system time and select an **AM** or **PM** setting.
6. 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** or **Time** 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.
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.

Note:

If you have a G10 or G11 gateway and subscribe to Caller ID from your service provider, date and time information is delivered to your one-X Quick Edition system from the PSTN with an incoming call.

Note:

For other configurations, if your one-X Quick Edition system clock is no longer synchronized with that of your service provider, use these procedures to revise system time. If your region observes Daylight Saving Time, for example, you will have to change your system time to correspond with your service provider.

Networking

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** web page.
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 subnet 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 subnet 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** web page.
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.

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.

Basic Troubleshooting for Telephones

If you cannot resolve a problem on your own, contact your technical support representative (see [Contacting Technical Support on page 90](#)).



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.	Verify that the telephone is receiving power through the Ethernet LAN connection: • 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. • 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. • 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. 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.

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

Problem/Symptom	Suggested Solution(s)
Telephone does not become active immediately after a power interruption.	Allow a few minutes for the telephone to initialize if the telephone became unplugged from the LAN or experienced a power interruption.
Telephone does not display the option to join an existing site when connected to an existing network.	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 subnet. As a result, the telephone will prompt you to create a new network. For more information, see IP Addresses on page 88. To resolve the problem, see To re-assign a telephone to a different network address space and/or network on page 89.
Telephone no longer communicates with peers.	Verify the telephone connection to the LAN. Review the network performance recommendations to ensure that the network adheres to minimum requirements (see Performance Recommendations on page 10). 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 Set Details on the Main menu: • Are all one-X Quick Edition devices running the same or compatible software versions? • Do all devices belong to the same network address space? If required, see To re-assign a telephone to a different network address space and/or network on page 89. • 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 Site field under Set Details. All peers making up the same network have the same site identifier and name. If required, see To re-assign a telephone to a different network address space and/or network on page 89 to enable the telephone to communicate with existing peers.
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Table 1: Basic Troubleshooting Procedures for Quick Edition IP Telephones (continued)

Problem/Symptom	Suggested Solution(s)
Speaker does not seem to work.	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.
Cannot add an entry to the Speed Dial list.	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.
Cannot dial an external number using any Speed Dial button or the Call Log.	See “Cannot make an external call” below.
No dial tone on a Quick Edition IP telephone.	Verify that the handset is securely connected. Use the Volume Control ( ) buttons to set the volume to an audible level. 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.
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 SDial , Log , Dir , and Vmail softkeys are displayed at the bottom of the display area. If a Back , Exit , or Done 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.	Verify the dial tone (see “No dial tone on a Quick Edition IP telephone” above). Next, verify that the party you are trying to call has not enabled: • call forwarding (see “Redirecting Calls through Call Forwarding” in the <i>Avaya one-X Quick Edition Telephone User Guide</i>), and/or • 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>). If required, contact your LAN administrator to determine whether there is a problem with the Ethernet LAN.
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Table 1: Basic Troubleshooting Procedures for Quick Edition IP Telephones (continued)

Problem/Symptom	Suggested Solution(s)
Cannot make an external call.	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. To place a call to the PSTN, dial the prefix followed by the number. To call a SIP network, dial the prefix followed by the number. 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. 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).
REDIAL () button does not dial the last number called.	Check the year in the system date and time setting (see System Time and Date on page 76). 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 Outgoing 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.
Audio quality is poor—specifically, you hear: • static when the line is active, • sudden silences or gaps in speech, and/or • clipped or garbled speech.	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 Performance Recommendations on page 10 for a list of recommendations.  Tip: You can specify an audio quality setting for the entire network (see To adjust the audio bandwidth on page 78).
An application or feature does not work as described in the product documentation.	Verify the procedure and retry. 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 one-X Quick Edition Documentation on page 7). If you would like to report a documentation error or omission, contact your technical support representative.
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Table 1: Basic Troubleshooting Procedures for Quick Edition IP Telephones (continued)

Problem/Symptom	Suggested Solution(s)
Teleworker: Teleworker client telephone cannot connect to the remote company network.	Determine the connection status of the teleworker client telephone (see “To view the connection status of the teleworker client telephone on page 52”). 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: • Update the private IP address of the company office network in the VPN configuration. • If a host IP address is specified for the teleworker server, change the IP address specification if required. 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. No Server message - specify the teleworker server host IP address. Not Authorized message - the teleworker server host IP does not recognize the client. Verify the Add Extension programming.
Restart Required message	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 OK softkey to restart.
Server Busy message	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.
No Response message	The connection request timed out. Verify your Internet connection. Verify that the router and firewall have been configured to pass P2P packets through a VPN tunnel. 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
Late Approval message	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.
Connection is successful, but the call was dropped during a conversation.	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.
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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)
Web-based interface: Cannot access the web-based interface.	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 To upgrade the software using a Quick Edition IP telephone on page 35. 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 To change network address settings using a Quick Edition IP telephone on page 24). 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. 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 Table 1: Basic Troubleshooting Procedures for Quick Edition IP Telephones on page 81. 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 Tools > Internet Options, and then click the Connections tab. Click the LAN Settings button to view the proxy server settings. Contact your LAN administrator for more information and help if required.
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Table 2: Avaya one-X Quick Edition System Problems and Solutions (continued)

Problem/Symptom	Suggested Solution(s)
Web-based interface: Cannot access the G20 ISDN BRI Gateway or A10 Analog Telephone Adapter.	- Using the included Ethernet cable, connect the G20 or A10 RJ-45 LAN port directly to the Ethernet port on your PC. - Configure the PC with a static IP address. Note: The procedures below are for Microsoft Windows. For another operating system, refer to the documents that came with your computer. - On the PC, go to My Network Places > View Network Connections. - Right-click Local Area Connection or the designated connection used for the ethernet port to connect to the device, then select Properties. - In the Local Area Connection Properties window, select Internet Protocol (TCP/IP) and click Properties (take note of the current settings as you will need to set it back to this later). - Select Use the following IP address. IP address: enter an IP address that conforms to the default IP address of the device, for example, 192.168.123.11. Subnet mask: enter a value that conforms to the default subnet mask of the device, for example, 255.255.255.0. - Click OK. - Troubleshoot the G20 or A10. - On your computer, telnet 192.168.123.10, and press Enter. User:nimdbg Password:54321 192.168.123.10>enable 192.168.123.10>configure 192.168.123.10(cfg)#context ip router 192.168.123.10<ctx-ip>[router]#interface 0 192.168.123.10<if-ip>[0]#ipaddress <static-ip> <subnet-mask> 192.168.123.10<if-ip>[0]#copy running-config startup-config 192.168.123.10<if-ip>[eth1]#reload Press 'yes' to restart, 'no' to cancel: yes - Reset the IP address assigned to your PC to its previous setting. - From the Web-based interface, retry the connection.
2 of 3	

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

Problem/Symptom	Suggested Solution(s)
Auto Attendant: The Auto Attendant feature does not answer incoming calls, or an incorrect greeting is played.	Access the Auto Attendant menu under System Options, and verify that the Auto Attendant configuration that you want to use has been created. See Auto Attendant Details on page 43. If you have a PSTN gateway, verify that an Auto Attendant configuration has been assigned to the gateway (see To edit an Auto Attendant configuration using a Quick Edition IP telephone on page 45). Verify that the greeting you want to use has been recorded and assigned to the associated Auto Attendant configuration (see Edit Auto Attendant Configurations on page 45). 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 To assign a SIP identity on page 63).
Paging: Not all of the telephones in a paging zone are receiving announcements.	If the telephone in question is in use while you make the announcement, the announcement is suppressed on that telephone—this is normal. Verify that the telephone in question has been added to the paging zone (see Set Details on page 21). 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>).
Voicemail: Callers cannot leave voicemail because the storage area is full.	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.
Email Notification of Voicemail: Voicemail messages are not being forwarded to an e-mail account.	Access the Network Options menu under System Options to verify that the IP address of an SMTP server has been specified (see E-mail Forwarding of Voicemail on page 54). Using the web-based User Options interface (see Appendix A: Web-based User Options), select Voice Mail and verify that “To” and “From” addresses have been specified, and that notification status has been enabled.
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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.

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.

Assigning a Static IP Address

If required, you may assign fixed network addresses to telephones or PSTN gateways through network options (see [Network Settings](#) on page 23). Once assigned, a static IP address persists until it is changed manually or another device having the same IP address is detected at boot time.

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

- A device will find no P2P peers and 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 re-assign the device.

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 [Network Settings](#) on page 23).
2. Remove the extension number of the telephone from the network (see [Remove an Entry](#) on page 39), 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.



Tip:

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.

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

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* for feature details. Refer to [Appendix B: Menus, Call Detail Recording, Specifications](#) on page 95 for a list of Web-based and telephone interface-configurable options.

Logging On

To log on to the web-based **User Options** interface, you need to know the IP address, extension number, and the user options password for the telephone. To display the IP address of the telephone, press # and PAGE LEFT or PAGE RIGHT () button.

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, **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. On the **Login** dialog box, enter the telephone extension number and password.
6. Click **Login**.

The **User Options Home** page for the telephone is displayed.

Logging Off

To log off from the web-based User Options interface

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

Changing the Password

You can change, enable, or disable the password through the **User Options** menu. 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 of the **User Options** web page.
2. Enter the current password, the new password, and confirm the new password.
3. Click **Submit**.

Home Page

The **Home** page displays Call Forwarding, Do Not Disturb, and Speed Dial dialog boxes.

To modify call forwarding option settings

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

To enable or disable Do-Not-Disturb (DND)

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

To create or modify Speed Dial assignments

- In the **Speed Dial** dialog box, click a vacant square to add a speed dial assignment.
- Click a programmed square to modify a speed dial assignment.

Caller's Log

The **Caller's Log** dialog box displays a list of incoming calls and a list of dialed calls.

- Click **Clear Log** to clear the incoming calls 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

The **Terminal Settings** dialog box displays the Corporate directory name assigned to the telephone, and registration information for telephone optional features.

- Click **Change** in the **Terminal Settings** dialog box to change the name that has been assigned to the telephone extension number. Enter your last name then first name to ensure that all names in the Corporate directory are sorted alphabetically by last name.
- For information about special features, see [Optional Features](#) on page 25.

Teleworker

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

For more information about this feature, see [Add a Teleworker on page 48](#).

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.

Backup & Restore User Options

Refer to [Telephone Configuration Data](#) on page 38 for a detailed description of the option.

To restore user data

- Browse to the file location and click **Restore**.

To back up user data

- Click **Download** and select the **Save in** location for the backup file.

Appendix B: Menus, Call Detail Recording, Specifications

This appendix includes the following tables:

- [Table 3: Telephone Options - Main Menu](#)
- [Table 4: Telephone Options - Telephone User Option Menu](#) on page 96
- [Table 5: Telephone Options - Web Interface Menu](#) on page 96
- [Table 6: System Options - Telephone Menu](#) on page 97
- [Table 7: System Options - Web Interface Menu](#) on page 98
- [Table 9: Call Detail Record Fields](#) on page 100
- [CDR Example](#) on page 101.

Telephone Options

Table 3: Telephone Options - Main Menu

1. Options	To log into user or system options.
2. Call Log	View and clear incoming and outgoing log entries.
3. DND	Enable and disable Do Not Disturb.
4. My Status	Set your user status.
5. Paging	Initiate a page.
6. Speed Dial	Add, edit, and delete speed dial entries.
7. Optional Features	View.
8. Set Details	View.
9. Language	View settings.

Table 4: Telephone Options - Telephone User Option Menu

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

Table 5: Telephone Options - Web Interface Menu

Change Password	Enter existing password, enter new password, and confirm new password.	
Home	Call Forwarding.	Enable, disable, and modify call forwarding settings.
	Do Not Disturb (DND)	Enable and disable audio notification for incoming calls.
	Speed Dial	Create, modify, and delete personal speed dial numbers.
Caller's Logs	Incoming Calls	View details, Clear, and Reset Missed Call Counter.
	Dialed Calls	View details and Clear Log.
Terminal Settings	Name	Change name.
	Set Optional Features	View e-mail Fwd, Teleworker, and WebAdm Sys Options.
Voice Mail	Zero Redirect	Enable and disable zero redirect and enter the number to which a call will be redirected.
	SMTP	Enable or disable SMTP, specify the IP address that will receive messages and the IP address for the email header.
Teleworker Options	Working Mode	Disable (Local) or enable (Teleworker) teleworker.
	Preferred Server	Enter the IP address of the teleworker server host.
Backup & Restore	Backup & restore user configuration data.	
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, type ht tps : //192 . 168 . 0 . 2).		

System Options

Table 6: System Options - Telephone Menu

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

Table 7: System Options - Web Interface Menu

Logout	Change Admin Password		Help
Device Management	Devices	View all devices; add or remove a device.	
		Set Details	View Extension, Name, Status, Page Zone, Software Version, Network Name, Network ID. Change extension number, name, and page zone. Reset user password back to default.
		Networking	View and edit IP address, Netmask, and Gateway.
		Features	View optional features and registration information.
		Gateway Details	View extension, IP address, software version, status, MAC address, network name and ID. Edit the extension.
		Lines	View and edit the incoming, outgoing, and loop length settings.
		MOH	View the current status and enable/disable music on hold.
		Networking	View and edit IP address, Netmask, and Gateway.
		A10 Details	View the MAC address; edit the name and the configurations for each port/SIP Identity.
		G20 Details	View the MAC address; edit the name and the configurations for each SIP Identity; configure the primary identity; enable/disable Caller ID.
	Software Upgrade	Upgrade Quick Edition devices.	
	Backup & Restore	Perform a backup and restore of all system configurations.	
Corporate Directory	Add or Remove Entry	Select from the entry type drop-down list: Sets (Teleworker), Gateways, Groups, Auto Attendants, External Entries.	
	Phones		
	Gateways		
	Groups		Add or delete a group; change the group name.
		Members	Add and remove members.
		Forwarding	Enable and disable forwarding for the group. Provide a number to which group calls will be forwarded.
		Dialing Rules	Add and remove dialing rules.

Table 7: System Options - Web Interface Menu (continued)

	Auto Attendants	View the extension, name, and prompts. Add, modify, and delete a custom auto attendant.	
	External Entries	View, add, modify, or delete an external entry.	
	SIP Identity	Launch the SIP Proxy Identities dialog for editing.	
	Global Dialing Rules	Configure a global dialing rule, it will apply to all telephones in the system.	
Applications	SMTP	Edit SMTP server settings.	
	CDR	Edit CDR server settings.	
Dialing Configuration	Dialing Plan	Extension ranges, Emergency Code, Operator Code, PSTN Code, and SIP Code are applied, by default, depending on the region specified at install.	
	Dialing Rules	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.	
Service Provider	Configuration	You can add a service provider with the following configuration components to create an Identity: Domain Name, Proxy Host, Proxy Port, Registrar Host, Registrar Port, Outbound Proxy Host, Outbound Proxy Port, Realm, and Register Expiry Time.	
	Identities	Add, edit, or delete an identity created under Configuration.	
SIP Proxy	Configurations	Edit SIP configuration entries.	
	Identities	Add, edit, or delete SIP identities.	
	Authorized Users	Edit users authorized to challenge communications between 3rd party devices and the QE proxy counterpart.	
Security	General	Password Rules	View and modify the configurations governing the system security.
	Authorization Codes	Add or delete codes that must be used for any call made from the system.	
	Admin Password	Change Admin Password.	
Localization	Localization	Select a system language and region from the drop-down lists.	
	Time & Date	Edit and save system time and date.	
Networking	Audio Bandwidth	Select High or Low. High provides a better quality voice transmission but with a higher bandwidth consumption.	
	VLAN Settings	Select enable and priority or disable for voice traffic. Select disabled or pass-through or enabled and priority for data traffic.	

Call Detail Recording Fields

Avaya one-X Quick Edition provides the ability to capture information about all phone calls. The call detail records are in ASCII (American Standard Code for Information Interchange) CSV (comma-separated values) format. Use TCP (transmission control protocol) to send data packets over IP from the telephone or gateway to a computer running a collection system.

A “call” may involve more than one “dialog”, including transfer, call forward, conference, etc. A Call Detail Record (CDR) will be identified by a combination of Call-ID, From-Tag, and To-Tag.

Table 8: Call Detail Record

DevID	No. of Dialogs	Dialog1	Dialog2			
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Table 9: Call Detail Record Fields

Recorded data	Description
CDR Record Fields	
DevID	The unique ID on a set, typically the MAC Address, to identify a call. In a normal call, both parties generate a CDR record on their own device, and the call IDs of these two records are not associated.
Number of dialogs	The total number of dialogs, i.e. transfer, hold, conference.
Dialog Fields	
Call-ID	SIP call ID: unique identifier for a SIP session.
From-Tag	The From-Tag of the SIP session.
To-Tag	The To-Tag of the SIP session.
Call type	01 - Normal; 02 - Conference; 03 - Group; 04 - Backup; 05 - DAA (Distributed Auto Attendant) host; 06 - Same set; 07 - SP; 08 - ATA; 09 - BRI.
CallDirection	01 - Incoming; 02 - Outgoing; 03 - Intranet (two devices on the same subnet).
CallerName	Display name of the caller.
Caller	Caller's URL.
CallerContact	Contact URL where this caller could be reached.
CalleeName	Display name of the callee.
Callee	Callee's URL.
CalleeContact	Contact URL where this callee could be reached.
Remote DeviceType	01 - Set; 02 - GW; 03 - Other.

Table 9: Call Detail Record Fields

Recorded data	Description
Start Time	Call start time; YYYY/MM/DD HH:MM:SS in local time.
Connection Time	Time of call connection; YYYY/MM/DD HH:MM:SS in local time.
TotalLocalHoldDuration	Total hold time duration in seconds.
TotalRmtHoldDuration	Total held time duration in seconds.
End Time	Time of call end; YYYY/MM/DD HH:MM:SS in local time.
ConnectDuration	Duration of call connection in seconds.
RingDuration	Ring duration in seconds.
Account Code	Code for customer account administration (not yet supported).
Authorization Code	Code of authorizing outgoing call or long distance call.
FailureCode	Numeric SIP error code if a call fails abnormally.
FailureReason	SIP response reason (corresponding to response code).
TerminationCode	C - Call completed; F - Call forwarded; I - Call in progress; M - Call missed; P - Call parked; R - Call requested; T - Call transferred; V - Call routed to voicemail; X - Call failed; ? - Unknown.
3rdPartyNumber	Third party involved in the dialog (i.e. transfer target, forward target, etc.)
Via	Last hop (previous hop or next hop)
TrunkID	TrunkID (on a GW) used in this dialog (not yet supported).

CDR Example

The CDR example, below, is based on the telephones described in [Table 10](#).

Table 10: Available Devices

MAC	DN	IP	Status	Name	Version	Dev_ Type
00:04:0D:4F:67:AA	200	135.20.161.73	Up	Adam	7.1.10	NIM
00:04:0D:9C:93:42	202	135.20.160.181	Up	Bob	7.1.10	NIM

Adam calls Bob – Call connected – Adam puts Bob on hold for sometime and then Adam ends the call

Note:

In the following CDR example output strings, Call type (01), Call direction (03), and Call completion (C) codes are shown in **bold** text.

CDR in Adam's telephone:

```
00:04:0D:4F:67:AA,1,f4801968fa9df098a3f1936fad916af3@135.20.161.73,6093515a350
be2f,e639f466f9ae299,01,03,Adam,200@135.20.161.73,200@135.20.161.73,Bob,202@13
5.20.160.181,202@135.20.160.181,01,2007/01/03 10:23:20,2007/01/03
10:23:22,10,0,2007/01/03 10:23:39,17,2,,0,,C,,,
```

CDR in Bob's telephone:

```
00:04:0D:9C:93:42,1,f4801968fa9df098a3f1936fad916af3@135.20.161.73,6093515a350
be2f,e639f466f9ae299,01,03,Adam,200@135.20.161.73,200@135.20.161.73,Bob,202@13
5.20.160.181,202@135.20.160.181,01,2007/01/03 10:23:24,2007/01/03
10:23:26,0,10,2007/01/03 10:23:43,17,2,,0,,C,,135.20.161.73:0,,
```

IP Voice Bandwidth

Table 11: RTP audio codec bandwidth usage

Link TType	Single Call Bandwidth	
	G711 (High)	G729a (Low)
VoIP on Ethernet	96.8	40.8
VoIP on AAL5	106	42.4
VoIP on Frame Relay	82.4	26.4
VoIP on PPP	82.8	26.8
Assumes: Voice Frame Size of 20mx, Full Duplex connections		

IP Port Usage

Table 12: IP Port Usage

Protocol	Transport	Port
SIP	UDP	5060
RTP	UDP	20000-40000
Music on Hold	Multicast	24430
Paging	Multicast	24400
P2P	UDP	5071

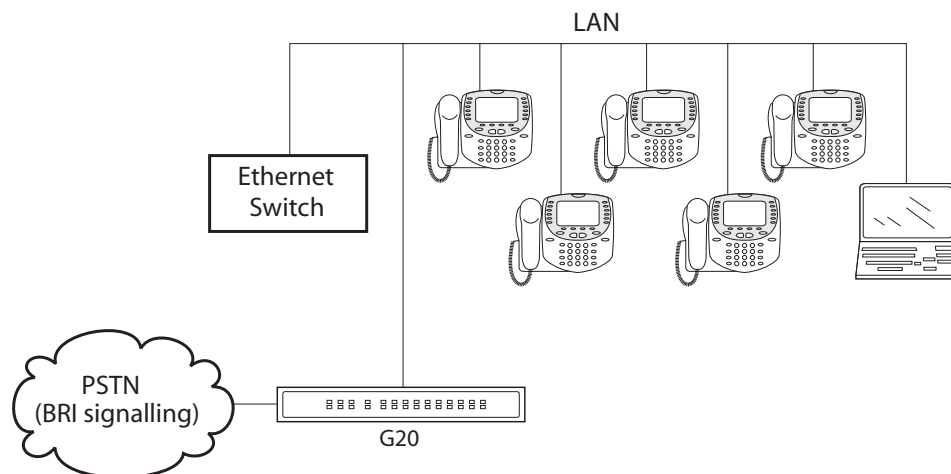
Appendix C: G20 ISDN BRI Gateway

Introduction

The G20 ISDN BRI Gateway is a compact VoIP gateway router that supports two VoIP calls on two ISDN BRI ports. This gateway allows your one-X Quick Edition network to leverage low-cost Internet Telephony on existing ISDN phones for complete Single office/Home office and branch office voice and data connectivity.

[Figure 6](#) shows the position of the gateway in a typical company telephone network. Up to five gateways can be connected to your one-X Quick Edition network if required to increase the number of PSTN connections into your office.

Figure 6: Typical one-X Quick Edition Network Configuration



A typical configuration consists of a customer-supplied 10/100 Base-T Ethernet Local Area Network (LAN) with a connected IP router or switch, to which the gateway and your telephones are connected. An administration computer may be connected to the network to provide web-based access to gateway configuration settings.

If your Quick Edition IP telephones are already connected to the LAN, adding a gateway to the system is easy. Simply connect the gateway LAN port to an unused port on the IP router or switch, and connect the ISDN BRI ports to the PSTN lines supplied by your service provider.

Hardware Features

Front Panel

Figure 7: G20 ISDN BRI Gateway Front Panel

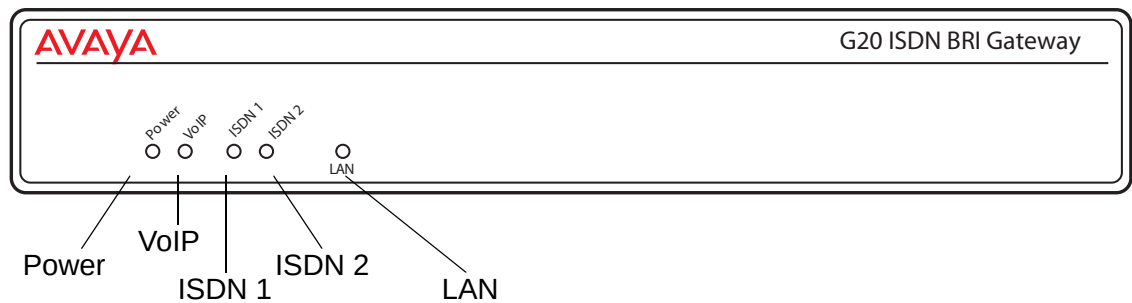


Table 13: G20 ISDN BRI Gateway front panel LED indications

LED	Description
Power	When lit, indicates power is applied and the unit is in normal operation. Off indicates no power is applied. Flashes once per second during boot (startup).
VoIP	On indicates the gateway is registered to an H.323 gatekeeper/SIP server, or, in the case of direct routing, has at least one active VoIP connection. Off indicates the unit is not configured or registered, or has no active direct-routed VoIP connection. Flashing green indicates that the unit is attempting to register or has failed to register.
ISDN 1 and 2	Off indicates no active calls. Blinking when one or two B-channels are connected.
LAN	On when the Ethernet connection on the corresponding port has a link indication. Flashes when data is received or transmitted at the corresponding Ethernet port.

Note:
If an error occurs, all LEDs will flash once per second.

Rear Panel

Figure 8: G20 ISDN BRI Gateway Rear Panel

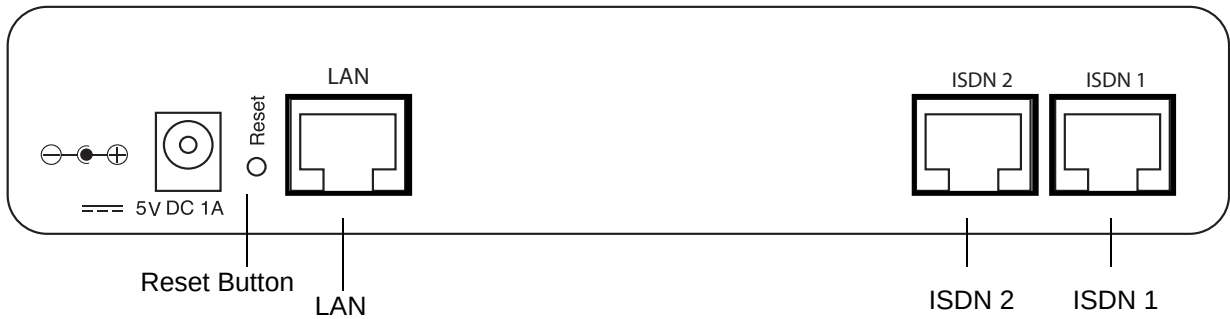


Table 14: G20 ISDN BRI Gateway back panel descriptions

Port	Description
5V DC, 1A	The power input for the supplied 5V wall adapter.
Reset	The reset button has three functions: ● Restart the unit with the current startup configuration—Press (for less than 1 second) and release the Reset button to restart the unit with the current startup configuration. ● Restart the unit with factory default configuration—Press the Reset button for 5 seconds until the Power LED starts blinking to restart the unit with factory default configuration. ● Restart the unit in bootloader mode (to be used only by trained G20 ISDN BRI Gateway technicians)—Starting with the unit powered off, press and hold the Reset button as you apply power to the unit. Release the Reset button when the Power LED starts blinking so the unit will enter bootloader mode.

Table 14: G20 ISDN BRI Gateway back panel descriptions

Port	Description
LAN	Auto-MIDX Fast-Ethernet port, RJ-45, connects the unit to an unused port on an IP router or switch that is connected to your Ethernet LAN.
ISDN 2 and 1	ISDN BRI TE (Usr) port, RJ-45 S0 (S/T)-interface, connects the unit to an ISDN NT. Point-to-point or point-to-multipoint configurable.

Compliance

Table 15: G20 ISDN BRI Gateway Compliance

EMC	● FCC Part 15, Class A ● EN55022, Class A ● EN55024 ● EN61000-3-2 ● EN61000-3-3
Safety	● EC/EN 60950-1 ● AS/NZS 60950-1
PSTN Regulatory	● TBR 3 ● AS/ACIF S031:2001 ● AS/ACIF S003 ● L3 supplement (NZ)
Radio and TV interference	The gateway generates and uses radio frequency energy, and if not installed and used properly-that is, in strict accordance with the manufacturer's instructions-may cause interference to radio and television reception. The gateway has been tested and found to comply with the limits for a Class A computing device in accordance with specifications in Subpart B of Part 15 of FCC rules, which are designed to provide reasonable protection from such interference in a commercial installation. However, there is no guarantee that interference will not occur in a particular installation. If the gateway does cause interference to radio or television reception, which can be determined by disconnecting the unit, the user is encouraged to try to correct the interference by one or more of the following measures: moving the computing equipment away from the receiver, re-orienting the receiving antenna and/or plugging the receiving equipment into a different AC outlet (such that the computing equipment and receiver are on different branches).

Table 15: G20 ISDN BRI Gateway Compliance

CE notice	We certify that the apparatus identified in this document conforms to the requirements of Council Directive 1999/5/EC on the approximation of the laws of the member states relating to Radio and Telecommunication Terminal Equipment and the mutual recognition of their conformity.
ISDN compliance	The device identified in this document is approved for connection to the public ISDN telecommunication network over a BRI/So interface.
MAC Address	See the label affixed to the bottom of the device.

Technical Specifications

Table 16: Technical Specifications

DSP	One 2-channel DSP.
Voice connectivity	● 2 ISDN BRI So (S/T), 4-wire RJ45 ● One Usr (TE) port labeled Line 1, one Net (NT) port labeled Line 0 ● Point-to-point, point-to-multipoint configurable ● Life-line cut-through relay between Line 1 and Line 0 ports ● Power feed-through between Line 1 and Line 0 ports
Fax and modem support	● Automatic fax and modem detection ● Codec fallback for modem-bypass ● T.38 Fax-Relay (Gr. 3 Fax, 9.6 k, 14.4 k) ● G.711 Fax-Bypass
Data connectivity	Full duplex, autosensing, and auto-MIDX 10/100Base-TX Ethernet LAN port.

Table 16: Technical Specifications

Voice processing (signalling dependent)	• 2 full-duplex channels of Voice CODECS: • G.711 A-Law/μ-Law (64 kbps) • G.726 (ADPCM 40, 32, 24, 16 kbps) • G.723.1 (5.3 or 6.3 kbps) • G.729ab (8 kbps) • Transparent ISDN data • G.168 echo cancellation • DTMF detection and generation • Carrier tone detection and generation • Silence suppression and comfort noise • Configurable jitter buffer • Configurable tones (dial, ringing, busy) • Configurable transmit packet length • RTP/RTCP (RFC 1889)
Voice signalling	• SIPv2 • H.323v4 • SIP call transfer, redirect • Overlap or en-bloc dialing • DTMF in-band, out-of-band • Configurable progress tones
Voice routing—session router	• Local switching (hairpinning) • Interface hunt groups • Call-Distribution groups • Call Routing Criteria: • Interface • Calling/called party number • Time of day, day of week, date • ISDN bearer capability • Various other information elements (IEs) of the ISDN setup • Wildcard and regular expression matching • Number manipulation functions: • Replace numbers • Add/remove digits • Pattern matching and replacement

Table 16: Technical Specifications

IP services	● IPv4 router; RIPv1, v2 (RFC 1058 and 2453) ● Programmable static routes ● ICMP redirect (RFC 792); Packet fragmentation ● DiffServ/ToS set or queue per header bits ● Packet Policing discards excess traffic ● 802.1p VLAN tagging ● IPSEC AH & ESP Modes ● Manual Key; IKE optional ● AES/DES/3DES Encryption
Operating environment	● Operating temperature: 32–104°F (0–40°C) ● Operating humidity: 5–80% (non condensing)
System	● CPU Motorola MC875 operating at 66 MHz ● Memory: ● 16 MB SDRAM ● 4 MB Flash
Dimensions	7.3W x 1.6H x 6.1D in. (18.5H x 4.1W x 15.5D cm)
Weight and power dissipation	● Weight: 30.5 oz./500 g ● Maximum power dissipation: 5W

Appendix D: A10 Analog Telephone Adapter

Introduction

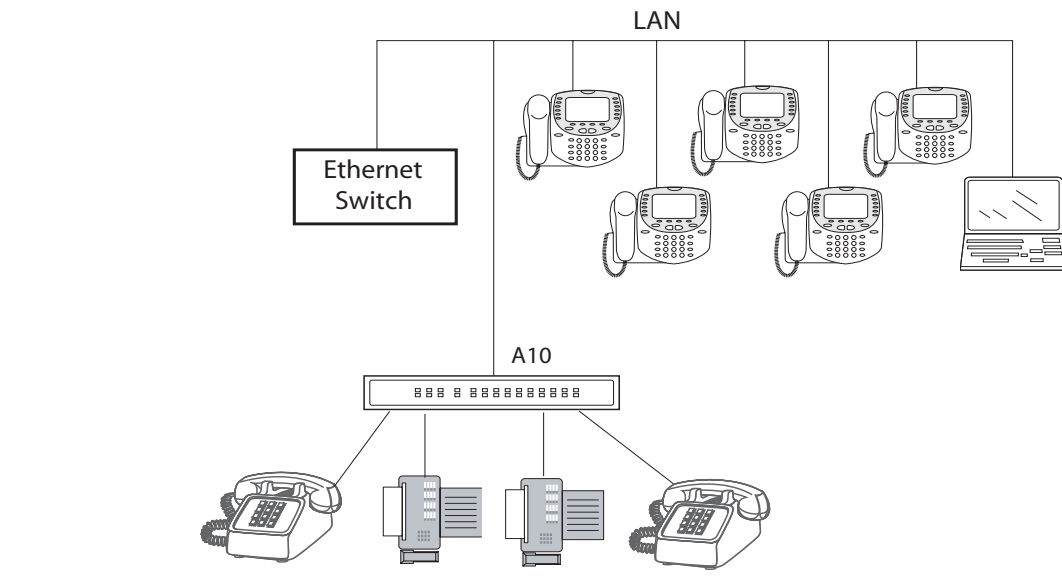
The A10 Analog Telephone Adapter combines IP routing, VPN/Security, and Quality of Service for voice and FAX calls over any IP or PSTN network. The one-X Quick Edition network can leverage low-cost IP services with packet-voice for branch office voice and data connectivity.

The adapter performs the following major functions:

- Voice over IP and local switching via 4 analog phone ports (FXS).
- Standard compliant VoIP conversion in accordance with SIP and H.323 protocols.
- Internet access and IP routing with IP Quality of Service (QoS) support for mixed voice and data traffic.

[Figure 9](#) shows the position of an adapter in a typical company telephone network. Up to ten adapters can be connected to your one-X Quick Edition network to increase the number of PSTN connections into your office.

Figure 9: Typical one-X Quick Edition Network Configuration



A typical configuration consists of a customer-supplied 10/100 Base-T Ethernet Local Area Network (LAN) with a connected IP router or switch, to which the adapter and your telephones are connected. An administration computer may be connected to the network to provide web-based access to adapter configuration settings.

The FXS or Foreign eXchange Subscriber interfaces (Voice Ports) on the A10 Analog Telephone Adapter provide for analog telephony, also known as Plain Old Telephone Service or POTS.

Hardware Features

Front Panel

Figure 10: A10 Analog Telephone Adapter Front Panel

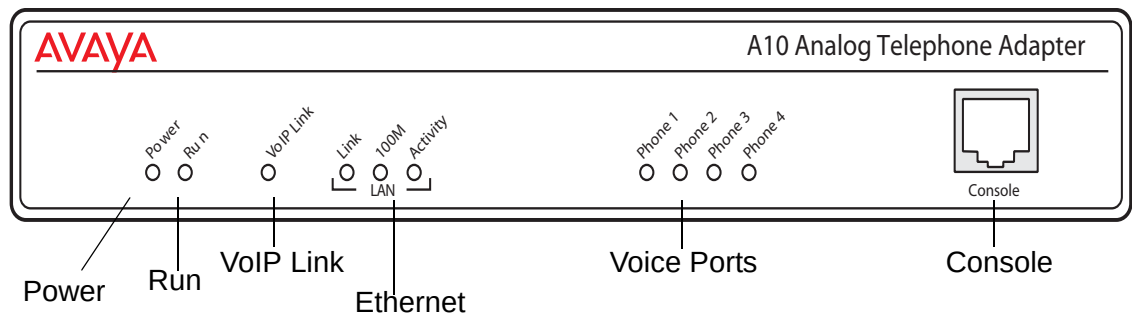


Table 17: Front panel descriptions

Power LED	When lit, indicates power is applied. Off indicates no power is applied.
Run LED	When lit, indicates normal operation. Flashes once per second during boot (startup).
VoIP Link LED	When lit, indicates the adapter is registered on a gatekeeper, media gateway controller, associated to a remote unit, or has an active VoIP connection. Off indicates the adapter is not configured or registered and has no active VoIP connection. Flashing green indicates that the adapter is attempting or has failed to associate/register.

Table 17: Front panel descriptions

Ethernet LED (each port)	● Link: Lit when Ethernet link is up. ● 100M: On when 100-Mbps Ethernet is selected. ● Activity: Flashes when data is received or transmitted from the unit to the LAN.
FXS Voice port LEDs	Off indicates on-hook condition. Solid green when off-hook. Flashes to follow ring cadence.
Console	Used for service and maintenance, the Console port, an RS-232 RJ-45 connector, connects the adapter to a serial terminal such as a PC or ASCII terminal (also called a dumb terminal).

Note:

If an error occurs, all LEDs will flash once per second.

Rear Panel

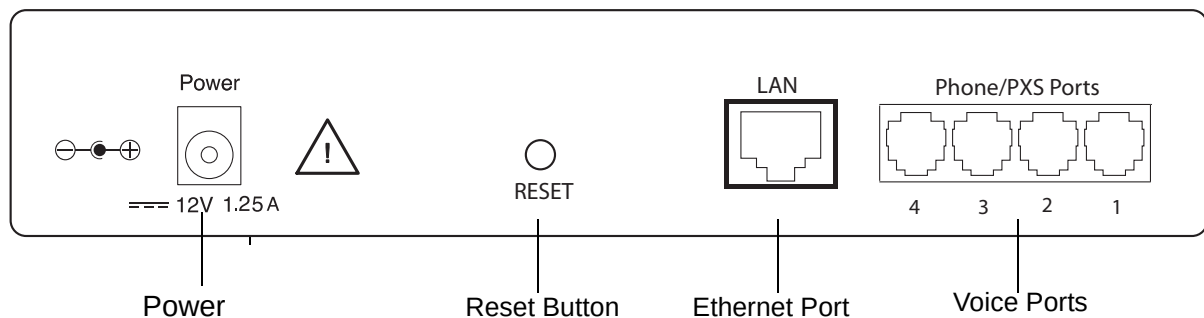
Figure 11: A10 Analog Telephone Adapter Rear Panel

Table 18: Back panel descriptions

Power	The power input for the supplied 12V 1.25A wall adapter.
RESET	The reset button has three functions: ● Restart the unit with the current startup configuration—Press (for less than 1 second) and release the RESET button to restart the unit with the current startup configuration. ● Restart the unit with factory default configuration—Press the RESET button for 5 seconds until the Power LED starts blinking to restart the unit with factory default configuration. ● Restart the unit in bootloader mode (to be used only by trained technicians)—Starting with the unit powered off, press and hold the RESET button as you apply power to the unit. Release the RESET button when the Power LED starts blinking so the unit will enter bootloader mode.
10/100 Ethernet port	RJ-45 connectors that connect the adapter to an Ethernet device (e.g., a cable or DSL modem, LAN hub or switch).
Analog voice ports, FXS	FXS RJ-11(6 position, 4 wire) connectors that connect the adapter with an analog terminal (a telephone, for example) FXO port. EuroPOTS support (ETSI EG201 188). FXS on-hook voltage is 48V for each FXS port.

Compliance

Table 19: Compliance

EMC	● FCC Part 15, Class A ● EN55022 plus Amd1 ● EN61000-3-2 ● EN61000-3-3 ● EN55024 ● AS/NZS CISPR 22 ● ICES 003
Safety	● EN60950 ● EC60950 ● AS/NZS 60950 ● CAN/CSA-C22.2 No. 60950 ● UL 60950

Table 19: Compliance

Radio and TV interference	The adapter generates and uses radio frequency energy, and if not installed and used properly-that is, in strict accordance with the manufacturer's instructions-may cause interference to radio and television reception. The adapter have been tested and found to comply with the limits for a Class A computing device in accordance with specifications in Subpart B of Part 15 of FCC rules, which are designed to provide reasonable protection from such interference in a commercial installation. However, there is no guarantee that interference will not occur in a particular installation. If the adapter does cause interference to radio or television reception, which can be determined by disconnecting the unit, the user is encouraged to try to correct the interference by one or more of the following measures: moving the computing equipment away from the receiver, re-orienting the receiving antenna and/or plugging the receiving equipment into a different AC outlet (such that the computing equipment and receiver are on different branches).
CE notice	The CE symbol on your adapter indicates that it is in compliance with the Electromagnetic Compatibility (EMC) directive and the Low Voltage Directive (LVD) of the European Union (EU).
MAC Address	See the label affixed to the bottom of the device.

Technical Specifications

Table 20: Technical Specifications

DSP	One 4-channel DSP.
Voice connectivity	• 2-wire Loopstart, RJ-11/12 • Short haul loop 1.1 km @3REN • EuroPOTS (ETSI EG201 188) • Programmable AC impedance, feeding, and ring voltage; On-Hook Voltage 29VDC • Caller-ID Type-1/2 FSK and ITU V.23/Bell 202 generation
Data connectivity	One 10/100 Full Duplex/Autosensing Ethernet RJ-45.

Table 20: Technical Specifications

Voice processing (signalling dependent)	• Voice codes: • G.711 A-Law/μ-Law (64 kbps) • G.726 (ADPCM 40, 32, 24, 16 kbps) • G.723.1 (5.3 or 6.3 kbps) • G.729ab (8 kbps) • Transparent pass through • G.168 echo cancellation • 8 parallel voice connections • DTMF detection and generation • Carrier tone detection and generation • Silence suppression and comfort noise • Configurable jitter buffer • Configurable tones (dial, ringing, busy) • Configurable transmit packet length • RTP/RTCP (RFC 1889)
Fax and modem support	• G.711 transparent FAX • Fax over IP (FoIP) • T.38 Fax relay (9.6 k, 14.4 k)
Voice signalling	H.323v4 • RAS, H.225, H.245 • Fast-connect, early H.245 • Gatekeeper auto discovery • Alias registration • Overlap sending • Empty capability set (call transfer, hold) • H.323v1 call transfer, hold ISDN over IP (ISoIP) • H.323 GW and GK compatible • H.323 Annex M3 • ISDN/Q-SIG feature tunneling • ISDN speech, audio and data (Fax Gr 4, UDI 64, RDI 64)

Table 20: Technical Specifications

Voice routing—session router	• Local switching; Interface hunt groups • Routing Criteria • Interface • Calling/called party number • Time of day, day of week, date • ISDN bearer capability • Number manipulation functions • Replace numbers; Add/remove digits • Multiple remote gateways; PLAR
IP services	• IPv4 router; RIPv1, v2 (RFC 1058 and 2453) • Programmable static routes • ICMP redirect (RFC 792); Packet fragmentation • DiffServ/ToS set or queue per header bits • Pocket Policing discards excess traffic • 802.1p VLAN tagging • IPSEC AH & ESP Modes • Manual Key; IKE optional • AES/DES/3DES Encryption
Operating environment	• Operating temperature: 32–104°F (0–40°C) • Operating humidity: 5–80% (non condensing)
System	• CPU Motorola MC859 operating at 50 MHz • Memory: • 32 MB SDRAM • 4 MB Flash
Dimensions	7.3W x 1.6H x 6.1D in. (18.5H x 4.1W x 15.5D cm)
Weight and power dissipation	• Weight: 30.5 oz./500 g • Maximum power dissipation: 5W
Power supply	12 VDC, 1.25 A Power is to be provided by an agency-approved external SELV source which provides reinforced insulation from the AC mains power and where the DC connector is the disconnect device. The source must have a rating of 12 VDC, 1.25 A.

FXS supplementary services

The voice port supplementary services are locally terminated (i.e. no other device is involved in providing the services), and they can be enabled/disabled separately on a POTS (plain old telephone service) phone. The services are:

- Call holding
- Call waiting
- Making a second call while holding first call

Call holding

It is possible to have two open calls on one POTS terminal: one active, the other on hold.

To toggle between an active and held call

- Press flash-hook, followed by the 2 key.

Call Transfer

To transfer an active call to an extension of external number

- Press flash-hook and dial the number.

Call Forward

Call forwarding is defined through the [Group Forwarding](#) rules (see page 41 for details).

Call waiting

When the analog line is busy, a second incoming call on the same interface announces itself using a special tone, the waiting tone. The user can then decide whether to accept the new call (put the current on hold or terminate it), or to reject it (keep the current call).

Note:

This feature is not used when the connected analog equipment is a fax, answering machine or similar device.

To reject a waiting call, do one of the following:

- Press flash-hook, followed by the 0 key,
- Ignore the call waiting signal. The waiting call will be rejected.

To accept a waiting call, do one of the following:

- Press flash-hook, followed by the 2 key to put the current call on hold and accept the waiting call. Call hold service must be enabled for this to work.
- Press flash-hook, followed by the 1 key to terminate the current call and accept the waiting call.
- Go on-hook to terminate the current call. The terminal will ring to indicate that a call is waiting (when this call is accepted by going off-hook again, the waiting call is connected).

Making a second call while holding first call

To make a second call while the current call is active, you will put the current call on hold, then make your second call.

Note:

Call holding service must be enabled for this service to work.

To make a second call

1. Press flash-hook which puts your current call on-hold and provides you a dial tone for the second call.
2. Dial your second call after hearing the dial tone.

Table 21: Command Summary

Action	Result
Press flash-hook, then 0	Rejects waiting call.
Press flash-hook, then 1	Drops the current call and accepts the incoming waiting call.
Press flash-hook, then 2	Toggles from the current call to the on-hold call if you have another call on-hold.
Press flash-hook, then 2	Toggles to the waiting call and puts the current call on-hold if you hear the waiting tone (to indicate a new incoming call).
Press flash-hook and wait for dial tone	If you want to make a second outgoing call, this action puts the current call on-hold and provides you dial tone for making another call. ● If the second call is through, you can toggle between the calls by using the flash-hook, 2 action. ● If the second call turns out to be busy, hang up the phone and pick it up when it rings to toggle to the call on-hold.

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