Avaya™ Quick Edition

Release 3.3

System Administration Guide

16-601412
Release 3.3
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Issue 4

Chapter 1: Overview

Introduction

This guide provides instructions for administering all of the Avaya™ Quick Edition IP telephones and gateways that make up the 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 system-wide features.

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
Document	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.
Italics	Italic text indicates the title of another document.
System Options	Words shown in bold represent literal elements of the user interfaces.

Quick Edition Documentation

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

What’s New in This Release

Release 3.3

- Call Pickup Groups: A user can answer a call that is ringing on another telephone in the same pickup group.
- Multi-Level Auto Attendant: MLAA provides for the programming of single digit routing. Callers can direct their call by dialing a single digit between 2 and 6.
- Transferring Calls Directly to Voicemail: A caller can dial a voice mailbox directly rather than dialing the extension number and being forwarded to voice mail.
- System Size Increase to 40 Devices.

Release 3.2

- Downloadable Music On Hold: System MOH can be provided by using an audio (.wav) file installed on a device.
- System Size Increase to 20 Devices.

Release 3.1

- G20 ISDN BRI Gateway: The gateway supports two VoIP calls on two ISDN BRI posts and allows your QE network to use Internet telephony on existing ISDN telephones.
- A10 Analog Telephone Adapter: The adapter combines IP routing, VPN/Security, and Quality of Service for voice and FAX calls over an IP or PSTN network.
- Programmable Softkeys: Users can program softkeys to access their most frequently-used telephone functions.
- Status Monitoring: You can monitor the status of other telephones to determine their availability to communicate with others.
- Call Detail Recording: CDR provides the ability to capture information about all phone calls.
- One Button Night Service: You can assign a programmable softkey to enable the activation of an alternate, after-hours, greeting.
- Database Backup and Restore: You can backup and restore system and user data.
- Authorization Codes: A user can dial a restricted number by using an authorization code.
- External Numbers in Corporate Directory: The directory can include external numbers.
- Security: Enhancements include password rules, password change enforcement, and authorization codes.

System Features and Options

System features can be accessed and configured, using the web-based administration interface or a telephone, to provision the entire Quick Edition network. System changes and updates are communicated to all telephones and gateways on the network.

Key Features and Benefits

Avaya Quick Edition software is embedded in each telephone to provide the most frequently used telephony applications. All telephones connected to the same network participate in traffic routing, call handling, and other network-related processes automatically. A 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 Quick Edition is ideal for offices having two-to-ten users. Refer to the *Avaya Quick Edition Telephone User Guide* (Document No. 16-601411) to learn how to use the features.

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 and public network 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.

Multisite Provisioning Tool

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

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

Call Flow Configuration Options

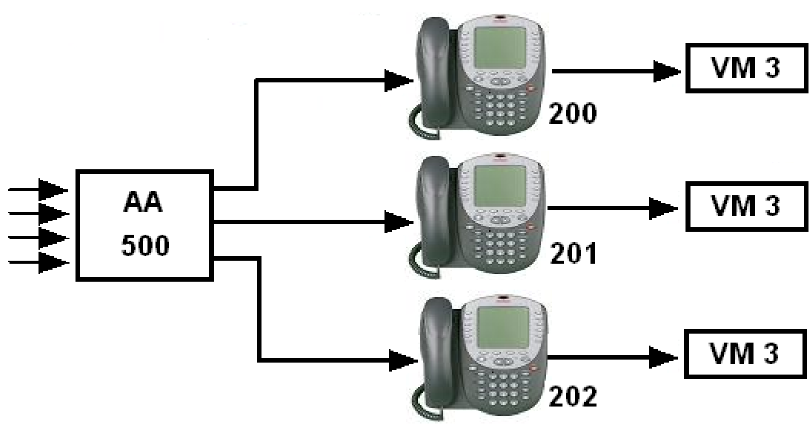
This section provides examples of call flow configurations.

- Default System Configuration
- Attendant with Auto Attendant Backup on page 5
- Attendant with After Hours Auto Attendant on page 6
- Attendant with General Mailbox Directed to Auto Attendant on page 7
- Incoming Call to Group Forwarded to Auto Attendant on page 8
- Two-Level Auto Attendant on page 9.

Default System Configuration

Incoming calls to the Public Switched Telephone Network (PSTN) gateway are directed to phones by the auto attendant. Unanswered calls are handled by voicemail on each telephone. In North America external calls are prefixed with a '9' and internal dial plan defaults to 3 digits.

Figure 1: Default system configuration



AA - Auto Attendant **CF** = Call forward	**CF D** = Call forward disabled **DIL** = Direct inward line	**VM 3/6** = Voicemail 3 or 6 rings **ZR** = Zero redirect

1. The auto attendant will answer incoming calls from all PSTN lines connected to the G11 gateway. 2. The AA will play the default greeting. You may record custom greetings (page 36). 3. Calls can be directed in one of three ways: dial by extension; dial by name; dial Operator. 4. The default operator extension, 200, can be changed (see page 40).	1. The Quick Edition telephones will forward unanswered calls to voicemail after three rings. 2. Each telephone has 20 minutes of voicemail storage. 3. Each mailbox has a default greeting that can be changed. Select Voicemail on the User Options Menu. 4. You can change the forwarding destination for voicemail to another internal or external number. Select Call Forward on the User Options Menu.

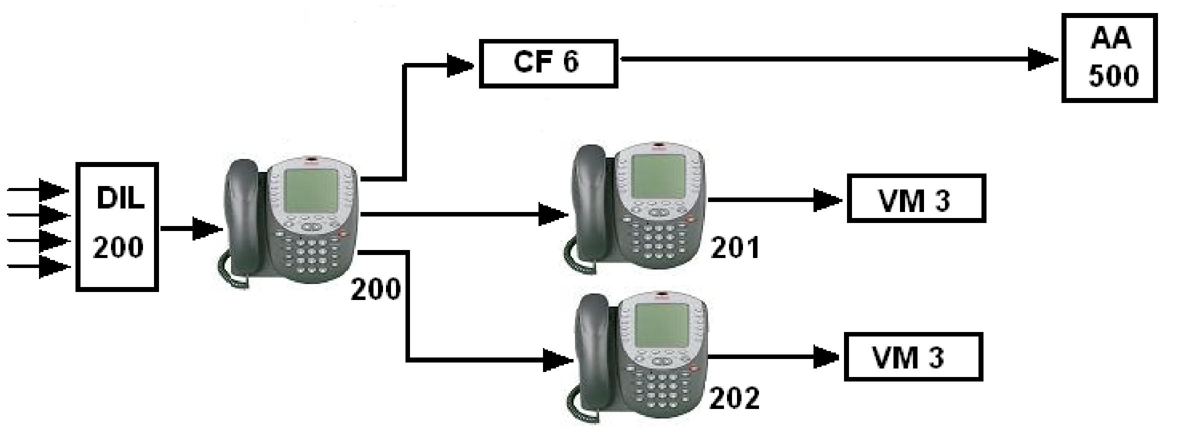
Attendant with Auto Attendant Backup

In Figure 2, extension 200 has been designated as the attendant position and also as the default operator extension. When ‘0’ is dialed within the system or from the auto attendant, calls will be routed to the operator extension.

The attendant can direct incoming calls to telephones in the system using transfer, call park, or conference. If the user at station 200 is not able to answer calls, because they are away from their desk or on another line, calls will be forwarded to the defined auto attendant, in this case after 6 rings. Calls can then be directed to telephones using the auto attendant.

Other telephones within the system will have calls handled after 3 rings by the voicemail on the telephone. This call handling can be changed by the station user if desired.

Figure 2: Attendant with auto attendant backup



AA - Auto Attendant **CF** = Call forward **CF D** = Call forward disabled		**DIL** = Direct inward line **VM 3/6** = Voicemail 3 or 6 rings **ZR** = Zero redirect	
1. You can configure direct ringing (direct inward line) at an extension. See DIL, Lines Settings on page 44.	1. Answered call can be transferred to any extension or external number.	1. The Quick Edition telephones will forward unanswered calls to voicemail after three rings. 2. You can change the forwarding destination for voicemail to another internal or external number. Select Call Forward on the User Options Menu.	1. Calls that reach the AA can be directed in one of three ways: dial by extension; dial by name; dial Operator. 2. The default operator extension, 200, can be changed (see Specify the Designated Operator Extension on page 40).

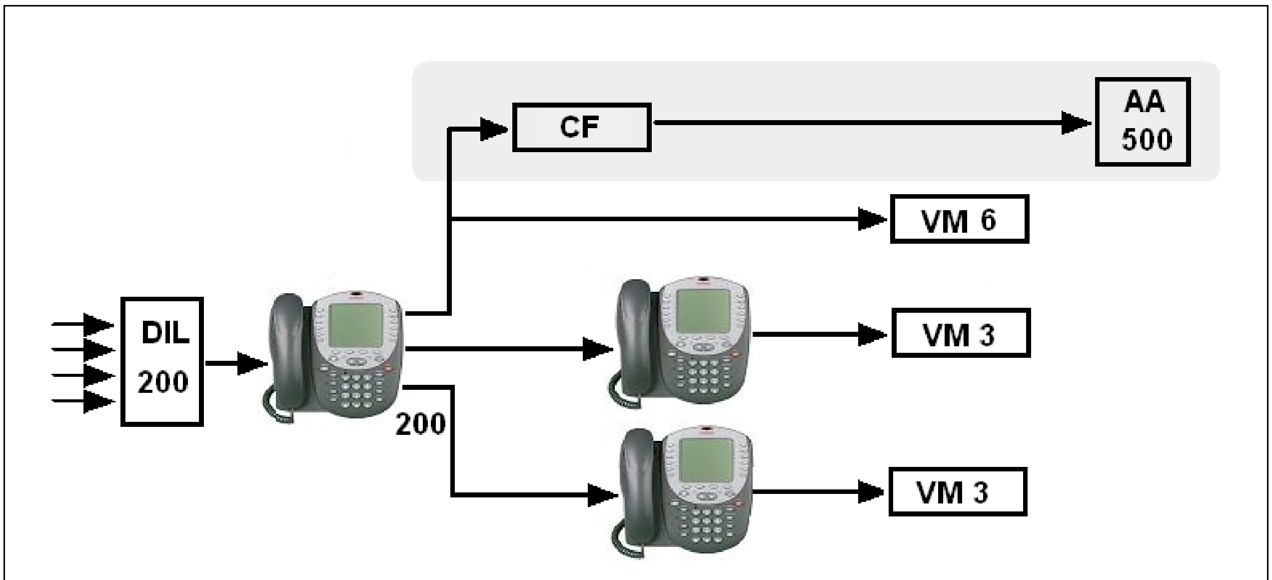
Attendant with After Hours Auto Attendant

You can route all calls from the PSTN to a single attendant extension in the network. In Figure 3 extension 200 is the attendant position and the operator extension. When '0' is dialed within the system or from the auto attendant, calls are routed to the operator extension, 200.

The attendant can direct incoming calls to other telephones using transfer, call park, or conference. Each telephone supports 2 forwarding configurations, call forward All and call forward N rings. During normal hours of operation the attendant will have call forward ALL disabled so calls would be forwarded to VM only after a user defined 6 rings. During off hours the user at extension 200 would enable call forward ALL to extension 500. Calls are then directed to all stations on the network using the auto attendant capabilities of the system.

Other stations will have calls handled after 3 rings by the default voicemail system.

Figure 3: Attendant with after hours auto attendant



<table>
<tr><td colspan="2">AA - Auto Attendant
CF = Call forward
CF D = Call forward disabled</td><td colspan="2">DIL = Direct inward line
VM 3/6 = Voicemail 3 or 6 rings
ZR = Zero redirect</td></tr>
<tr><td>1. You can configure, in Web Admin, to bypass the AA and directly ring (direct inward line) a telephone. See DIL, Lines Settings on page 44.</td><td>1. Answered call can be transferred to any extension or external number.
2. This configuration provides for night or off-hours service.</td><td>1. The Quick Edition telephones will forward unanswered calls to voicemail after three rings.
2. You can change the forwarding destination for voicemail to another internal or external number. Select Call Forward on the User Options Menu.</td><td>1. Calls that reach the AA can be directed in one of three ways: dial by extension; dial by name; dial Operator.
2. The default operator extension, 200, can be changed (see Specify the Designated Operator Extension on page 40).</td></tr>
</table>

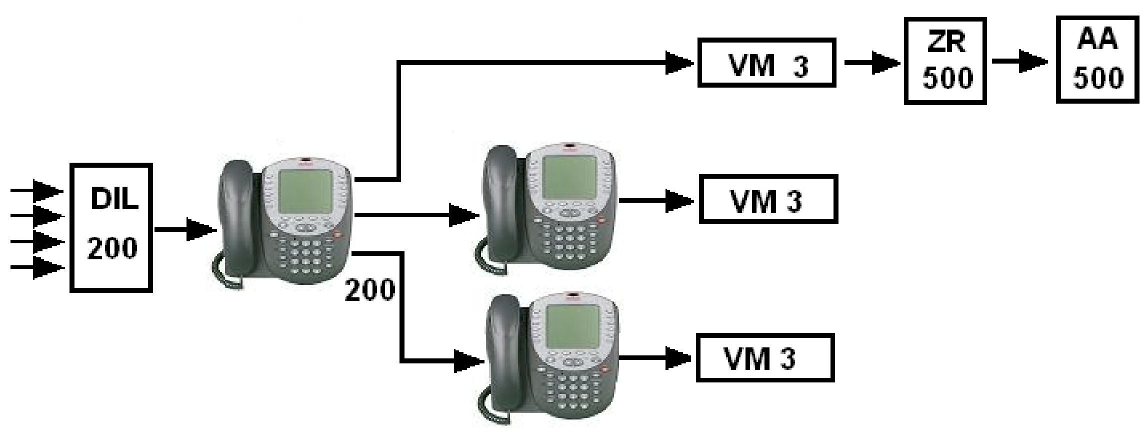
Attendant with General Mailbox Directed to Auto Attendant

In Figure 4 extension 200 has been designated as the attendant position and as the default operator extension. When ‘0’ is dialed within the system or from the auto attendant calls will automatically be routed to the operator extension, 200.

The attendant, like all other stations on the network, can then direct incoming calls to other telephones within the system using transfer, call park, or conference.

Calls that are not answered by extension 200 are directed to the voicemail for extension 200. Voicemail supports zero redirect capability that allows the user to remain within the voicemail system and leave a message, or press ‘0’ to be directed to a user-defined extension or external PSTN number. In this example the zero redirect number is the auto attendant that provides callers the ability to reach other extensions using dial by name or dial by number capabilities. The recorded greeting for the voicemail on station 200 will indicate that pressing ‘0’ will redirect the caller to the auto attendant or they may remain on the line to leave a message.

Figure 4: Attendant with general mailbox directed to auto attendant



AA - Auto Attendant **CF** = Call forward **CF D** = Call forward disabled		**DIL** = Direct inward line **VM 3/6** = Voicemail 3 or 6 rings **ZR** = Zero redirect	
1. You can configure direct ringing (direct inward line) at an extension. See DIL, Lines Settings on page 44.	1. Answered calls can be transferred to any extension or external number. 2. Any phone in the system will forward unanswered call to voicemail after 3 rings. 3. Extension 200 greeting can indicate general mailbox with the option to press 0 to reach the AA.	1. Voicemail functionality can provide a zero redirect phone number. Choose Zero Redirect in the User Voicemail Options. 2. Enter dialed number 500 and record the appropriate VM greeting. Callers will have the option to redirect to the AA at any time.	1. Calls that reach the AA can be directed in one of three ways: dial by extension; dial by name; dial Operator. 2. The default operator extension, 200, can be changed (see Specify the Designated Operator Extension on page 40).

Incoming Call to Group Forwarded to Auto Attendant

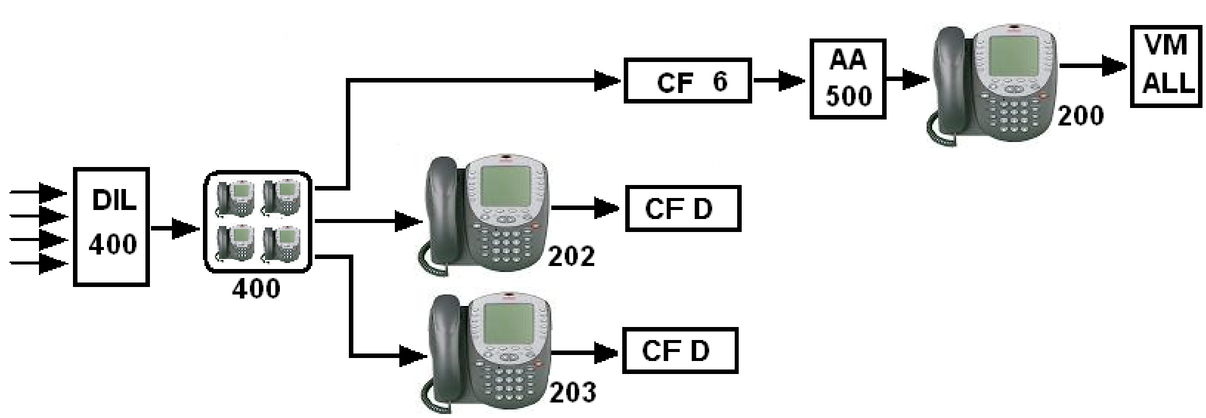
In Figure 5 ring group 400 is the destination for all incoming calls. The ring group can contain up to 10 telephones that can ring simultaneously when a call is directed to the group extension. With a DIL assignment for the lines on the gateway to extension 400, this provides a key system-like solution in which all telephones are presented with incoming calls at the same time.

When the call is answered, the call appearance remains on the telephone that is on the active call and all other telephones return to the idle screen. Calls can continue to be directed to other telephones using transfer, call park, and conference whether they are a group member or not.

Group extensions also provide a forwarding option should no member of the group be available. In this case call forwarding for the group directs unanswered calls to the auto attendant 500.

A further option from the auto attendant would be to have callers press ‘0’ to reach the general mailbox, in this example extension 200, or dial the ring group or extension, based on instructions given in the AA recording. In that case all calls directed to extension 200 will immediately be forwarded to voicemail, and calls to ring group 400 will still be presented at extension 200 as they would normally.

Figure 5: Call to group forwarded to auto attendant



AA - Auto Attendant **CF** = Call forward **CF D** = Call forward disabled	**DIL** = Direct inward line **VM 3/6** = Voicemail 3 or 6 rings **ZR** = Zero redirect

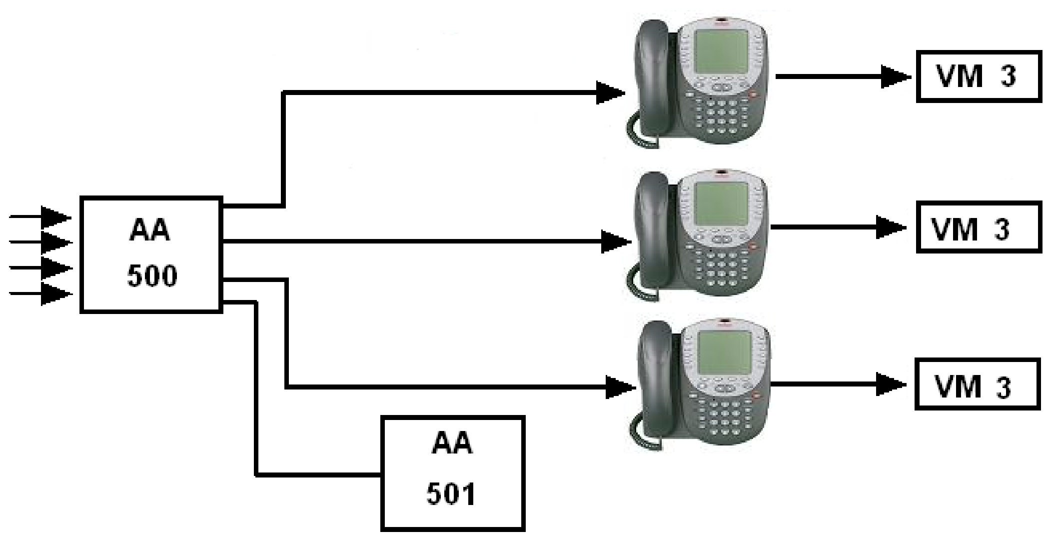
1. You can configure direct ringing (direct inward line) at a group. See DIL, Lines Settings on page 44.	1. Create a group. 2. Create a forwarding rule for the group to redirect to an extension.	1. You can change the forward destination to another number instead of voicemail.	1. Calls that reach the AA can be directed in one of three ways: dial by extension; dial by name; dial Operator. 2. The default operator extension, 200, can be changed (see Specify the Designated Operator Extension on page 40).	1. The greeting for extension 200 can be recorded to indicate that it is the general mailbox.

Two-Level Auto Attendant

This example of a two level auto attendant may be used in conjunction with any of the previous call flow examples or by itself. By default an auto attendant with extension 500 will answer incoming calls. The system supports multiple auto attendants and multiple custom recorded greetings. You can create a two level auto attendant to provide secondary messages to customers, for example, a message regarding directions to a location or notices that may not be applicable to all callers.

In addition to providing the standard custom greeting and call routing features, the primary auto attendant (500) can direct callers to dial 501 to hear the secondary message. Extension 501 could be recorded to direct callers to dial 500 to return to the main level of the attendant.

Figure 6: Two level auto attendant



AA - Auto Attendant **CF** = Call forward **CF D** = Call forward disabled	**DIL** = Direct inward line **VM 3/6** = Voicemail 3 or 6 rings **ZR** = Zero redirect

1. One of the auto attendants will answer incoming calls from the PSTN to the Gateway on ALL lines. 2. The AA will use a custom greeting, recorded on a telephone on the System Options Menu. 3. In this example, the greeting should inform the caller to dial ’501’ to reach the second AA. 4. Calls that reach either of the AA’s can be directed in one of three ways: dial by extension; dial by name; dial Operator.	1. The second AA greeting will provide information, such as directions to the location or seasonal information that may not be required in the main AA greeting. 2. The AA will use a custom greeting recorded on a telephone on the System Options Menu. 3. The greeting should inform the caller to dial ’500’ to reach the main AA.	1. The Quick Edition telephones will forward unanswered calls to voicemail after three rings. 2. Each telephone has 20 minutes of voicemail storage. 3. Each mailbox has a default greeting that can be changed. Select Voicemail on the User Options Menu. 4. You can change the forwarding destination for voicemail to another internal or external number. Select Call Forward on the User Options Menu.

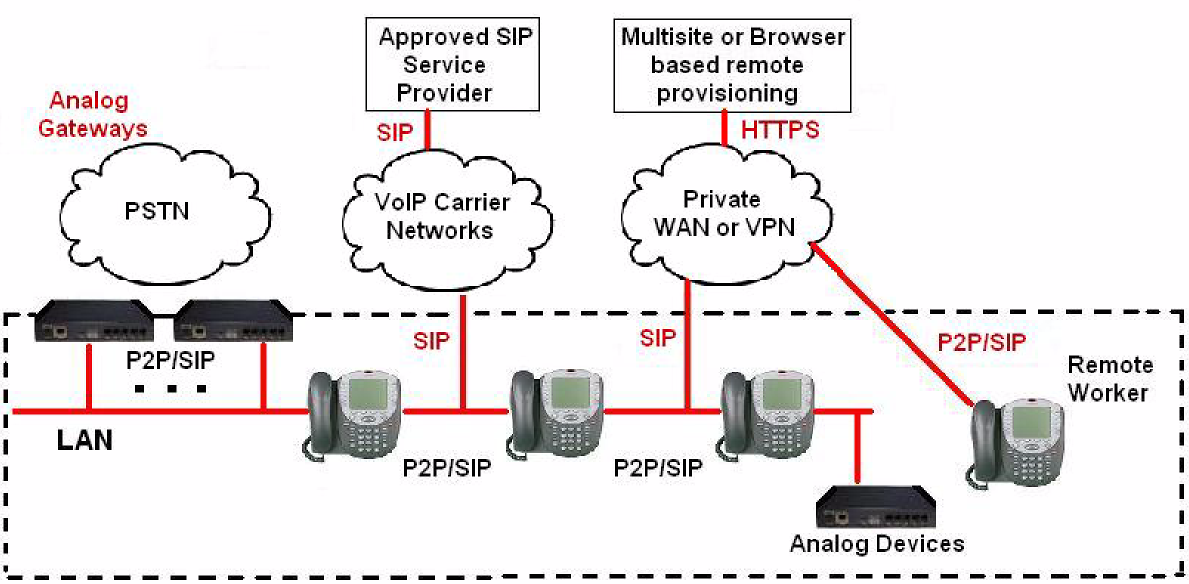
Multi-Level Auto Attendant

The Quick Edition voice mail application provides for single-digit routing at the auto attendant level. This enhancement will allow you to set up multiple levels of auto attendant, customize menu choices, and record corresponding auto attendant prompts.

Typical 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. An administration computer may be connected to the subnet to provide web-based access to user and system settings. Your network must be equipped with a gateway to access the PSTN through PSTN lines supplied by your telephone service provider. Power is ideally supplied to telephones by plugging the telephones into 802.3af PoE-enabled Ethernet wall jacks. You may optionally connect the telephone between a computer and the Ethernet LAN.

Figure 7: External Connectivity



Remote workers/telephones connected via VPN.
Interoperability with SIP Service providers.
Centralized PC-based application for networked upgrades and system provisioning.
Analog devices connect to available ports on the optional A10 Analog Telephone Adapter.

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. 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. Quick Edition devices try to obtain an IP address from a DHCP server before they assign themselves IP addresses.

Note:
The A10 Analog Telephone Adapter and the G20 ISDN BRI Gateway do not support Zeroconf. Refer to G20 BRI ISDN Gateway Setup on page 21 and A10 Analog Telephone Adapter Setup on page 23 to configure without a DHCP server.

- Avaya 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 Quick Edition devices that need to communicate with each other in the same VLAN. If multiple 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 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.
- Delay, jitter, and packet loss in speech delivery all have the potential to degrade the quality of 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 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 83. If you are experiencing performance degradation with the 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 Quick Edition system uses 239.192.228.123. This is a limited-scope address that is not routable over the public network (Internet).
- The Quick Edition system supports IGMP v2. A number of higher-end Layer 2 switches support IGMP snooping by default. If the 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 Quick Edition devices are connected to a switch that runs the Spanning Tree Protocol (STP), configure the 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.

Analog Gateway

Analog Gateway

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

ISDN BRI Gateway

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

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 Quick Edition system.

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

Figure 8: Teleworker over Internet (VPN)

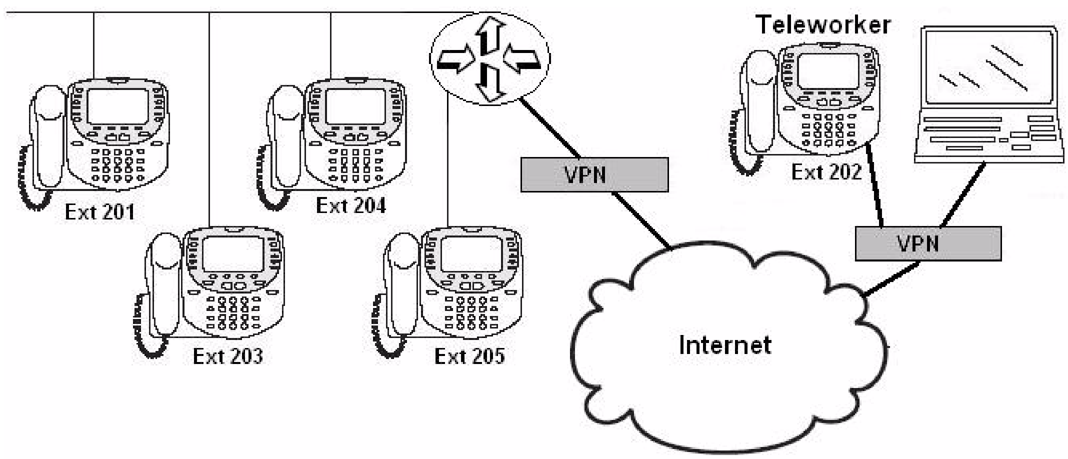
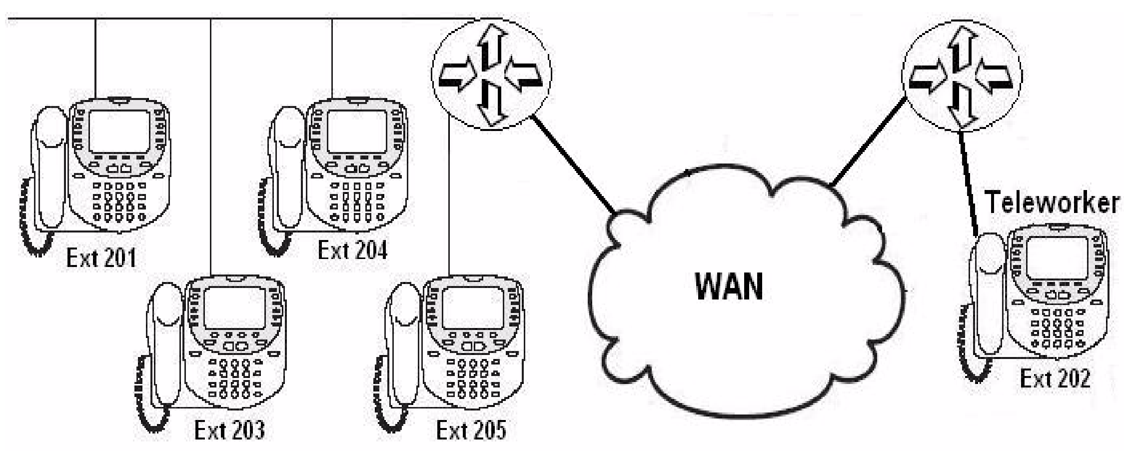


Figure 9: Teleworker over a Private WAN



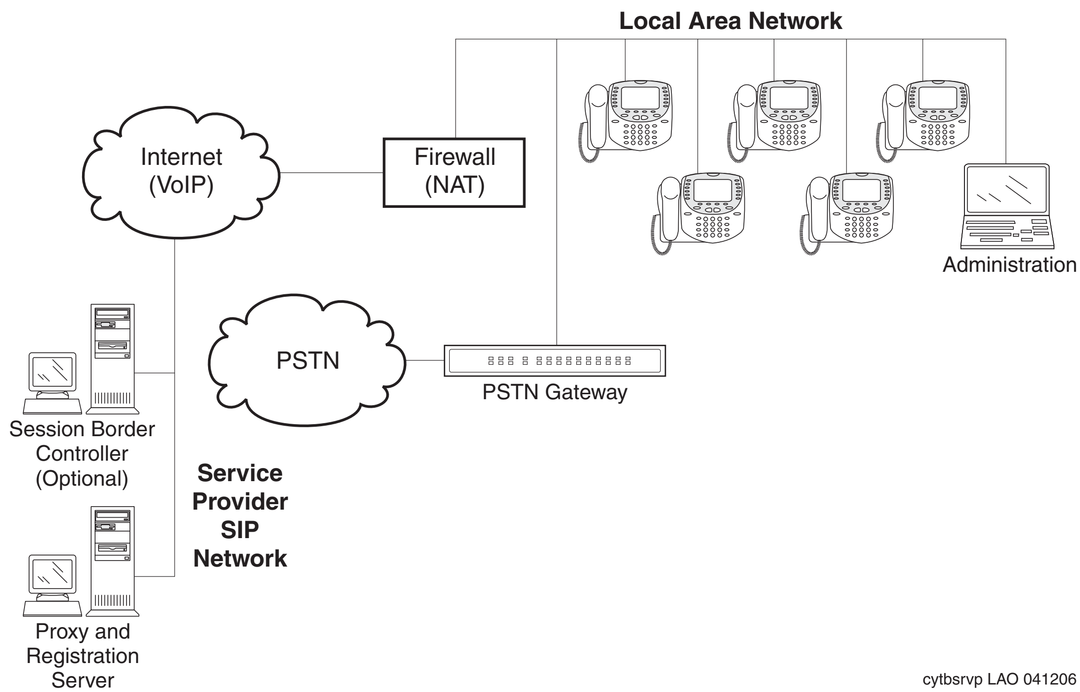
SIP (VoIP) Interoperability

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.

To support SIP interoperability, the telephones on your 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.

Figure 10 shows a typical SIP service provider configuration. The SIP network provides a gateway to the Internet, a SIP proxy and registration server, and a session border controller.

Figure 10: SIP Service Provider Configuration



Growing Your Network

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

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).

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).

To ensure the best quality of service without needing to optimize network performance, plan on a maximum of 40 telephones,10 PSTN gateways, and 10 Analog Telephone Adapters per subnet. Beyond 40 devices, the requirement for more complex telephony applications may increase.

Tip:
If you need more than 40 devices, have your network assessed first. The Avaya Intelligent Communications portfolio supports enterprises on a migration path from traditional telephony to IP, SIP, mobile platforms, Web-based services. Avaya Professional Services can help you manage your migration path.

Localization

Supported regions	Extension	AA	Emergency	Operator	PSTN	SIP	International	National	Country Code
Argentina	200-599	200-599	911	0	9	8	011	1	1
Australia	200-599	200-599	911	0	9	8	011	1	1
Austria	20-89	89-89	112	NIL	0	9	00	0	43
Belgium	200-599	500-599	112	9	0	8	00	0	32
Brazil	200-599	200-599	911	0	9	8	011	1	1
Chile	200-599	200-599	911	0	9	8	011	1	1
Columbia	200-599	200-599	911	0	9	8	011	1	1
Dominican Republic	200-599	200-599	911	0	9	8	011	1	1
El Salvador	200-599	200-599	911	0	9	8	011	1	1
France	2000-6999	6990-6999	112	9	0	7	00	0	33
Germany	20-89	89-89	112	NIL	0	9	00	0	49
Greece	200-599	200-599	911	0	9	8	011	1	1
Guatemala	200-599	200-599	911	0	9	8	011	1	1
Hong Kong	200-599	200-599	911	0	9	8	011	1	1
India	200-599	200-599	911	0	9	8	011	1	1
Italy	200-799	700-799	112	9	0	8	00	NIL	39
Jamaica	200-599	200-599	911	0	9	8	011	1	1

Supported regions	Extension	AA	Emergency	Operator	PSTN	SIP	International	National	Country Code
Japan	200-599	200-599	911	0	9	8	011	1	1
Mexico	200-599	200-599	911	0	9	8	011	1	1
Netherlands	200-599	500-599	112	9	0	8	00	0	31
New Zealand	200-599	200-599	911	0	9	8	011	1	1
North America	200-599	200-599	911	0	9	8	011	1	1
Norway	200-599	200-599	911	0	9	8	011	1	1
Panama	200-599	200-599	911	0	9	8	011	1	1
Peru	200-599	200-599	911	0	9	8	011	1	1
Russia	200-599	200-599	911	0	9	8	011	1	1
Singapore	200-599	200-599	911	0	9	8	011	1	1
South Africa	200-599	200-599	911	0	9	8	011	1	1
Spain	20-89	89-89	112	NIL	0	9	00	NIL	49
Switzerland	20-89	89-89	144	NIL	0	9	00	0	41
Thailand	200-599	200-599	911	0	9	8	011	1	1
United Kingdom	200-799	700-799	999	0	9	8	00	0	44

Supported languages
Dutch
English - British
English - US
French - Canadian
French - Parisian
German
Italian
Spanish - Castilian
Spanish - Latin American

Chapter 2: Basic Setup

Initial System Setup

Register your system at https://www.avaya.com/quickregistration.

Confirm your data network

Managed switch/router parameters may require adjustment; see the Troubleshooting Guide.

Connect the first telephone

- Connect the telephone to the LAN.
- With power applied correctly, the Message Waiting LED above the Avaya logo will light.
- The telephone display will show software information and the initialization process.

Select the system language

- The telephone will prompt for a Language Selection; select from the list provided.
  - Use the PAGE LEFT and PAGE RIGHT buttons to navigate through long list items.

Create the Site

- The telephone will prompt for the creation of a site; choose to create the site.
- Use the telephone keypad to enter a site name.
  - The site name will be used to identify all devices that join your local phone system.
  - Once created, the site name cannot be changed.

Enter a Password

- The telephone will prompt for a password; at least six digits, for example 665544.
- Confirm the new password by entering it a second time.
  - Make note of the password; it will be required for additional system configuration.
  - The telephone will indicate the system site number and site name.

Initial Telephone Setup

Enter a name for the telephone

- The telephone will prompt for a set name.
- Press the dialpad keys to enter a name of a maximum 32 characters.
  - Press the key 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.
  - Press PAGE LEFT or PAGE RIGHT to move the cursor to the left or to the right without deleting a character.
  - Press PAGE RIGHT to add a space to the end of a line.
  - Press **Case** to change a character to upper- or lower-case. The first character in a line and the first character after a space are capitalized automatically.
  - Press the 1 dialpad key to enter the special characters . , ’ & - and @.
- Press **Save** to assign the name to the telephone.
- The set name for any telephone can be changed at any time.

No Peers

- The No Peers indication will appear on the top line of the display.
  - The indication will remain until another Quick Edition device has been added to the system.

Assigned extension number

- The first telephone added to the network will be assigned an extension number.
  - The extension number shown in the upper left of the display is used to call this telephone locally within the system.
  - Press the **Dir** softkey to display a list of the telephones and extensions that have been added to the network.

Change the extension number

- The system administrator can change the extension number from the telephone or web-based administration interface at any time.
  - Access Options -> System Options -> Device Management -> Set Extension

Additional System Setup

Connect a second telephone

- With power applied, the Message Waiting LED above the Avaya logo will light.
- When the application begins to load, the LED will turn off.
- The telephone display will show software information and the initialization process.

Connect additional devices

- Additional devices do not require language selection or site creation.
  - **Simply wait and allow the device to join the site that has been created**.
  - The telephone will indicate the site name associated with the system it has joined.
- You can access system options from any device using the system password.

Additional Telephone Setup

Enter a name for additional telephones

- Enter the name that will be used for the telephone in the corporate directory and auto attendant.
- Press the dialpad keys to enter a name of a maximum 32 characters.
- Press **Save** to assign the name to the telephone.
- The set name for any telephone can be changed at any time.

Assigned extension number for additional telephones

- The extension number shown in the upper left of the display is used to call this telephone locally within the system.
- Press the **Dir** softkey to display a list of the telephones and extensions that have been added to the network.

Change the extension number for additional telephones

- The system administrator can change the extension number from the telephone or web-based administration interface at any time.
- Duplicate extension numbers cannot be assigned within the system; if another device is using the number that you are attempting to establish, you must first change the extension number for that device.
  - Access Options -> System Options -> Device Management -> Set Extension

Add additional telephones

- Add additional telephones by following the same process as above to join the site.

Default user password

- In software release 3.1 and later, the default user password is 1-2-3-4-5-6.
- In software release 3.0 and earlier, the default user password is 1-2-3-4-5.

Analog PSTN Gateway Setup

A G11 has four Foreign Exchange Office (FXO) ports that provide your network with access to the Public Switched Telephone Network (PSTN) through PSTN lines managed by your telephone service provider.

Add G11 PSTN gateways to the network

- With power applied correctly, the system LED will turn RED.
- When the device has joined the Quick Edition site, the LED will turn GREEN.
- The device will be assigned a non-dialable extension number in the web-based administration interface.

Change the gateway extension number

- You can view and change the extension number from Device Management in the web-based administration interface.
- Duplicate extension numbers cannot exist in the system; if another device is using the number that you are attempting to establish, you must first change the extension number for that device.

Ground the gateway

- Ground the gateway using the provided grounded power supply OR ground using a grounding cable (not provided) between the gateway grounding lug and EARTH ground.

  CAUTION:
  Use only one grounding method. Using both grounding methods will create a ground loop and may introduce line echo.

Adjust loop length to compensate for line echo

- By default, incoming ports are set to medium loop length with gain settings appropriate to most PSTN lines.
- Loop length can be changed by the system administrator from the telephone or the web-based administration interface (refer to page 44 for details).
- If echo or noise exists on external calls, ensure that the device is properly grounded and adjust the loop length to short to compensate for gain differences.

G20 BRI ISDN Gateway Setup

The G20 ISDN BRI Gateway is a compact VoIP gateway router that supports two VoIP calls on each of two ISDN BRI ports. This gateway allows your 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.

Add a G20 BRI ISDN gateway to the network

- With power applied correctly, the power LED will blink GREEN.
- The G20 gateway does **not** support PoE (power over Ethernet). Use the wall adapter to provide power to the device.

G20 IP address assignment

DHCP server

- Connect the G20 to the same subnet as the local Quick Edition system.
- The G20 will broadcast to a known multicast address.
- In response to the broadcast, Quick Edition will add the G20 to Device Management in the web-based administration interface.

No DHCP server

- The G20 BRI ISDN gateway does not support Zeroconf IP address assignment.
- Connect G20 RJ-45 LAN port directly to an Ethernet port on your PC.
- Configure the PC with a static IP address.
- Use a web browser directed to 192.168.123.10 to assign static IP, subnet mask, and default G20 IP address.
- Save running configuration and reload.
- Reset assigned IP address on the PC.
- Connect the G20 to the same subnet as the local Quick Edition system.
- The G20 will broadcast to a known multicast address.
- In response to the broadcast, Quick Edition will add the G20 to Device Management in the web-based administration interface.

See Assigning a Static IP Address to the A10 or G20 on page 50 for detailed information.

G20 IP activity

- When the device has joined the Quick Edition site, the power LED will be solid GREEN and the device will be listed in Device Management in the web-based administration interface.
- By default, the name given is “G20”. The device is not provided with an extension number.

BRI ISDN channel assignment

Configure SIP Proxy Identities

- Create internal system identities to be associated with the BRI channels provided by the Service Provider.
  - Program on the SIP Proxy menu in the web-based administration interface.

Associate Proxy Identities to ports on the G20

- Associate the previously created identity with a specific physical port on the BRI gateway.
  - Program on the G20 Device Management menu in the web-based administration interface.

Note:
Refer to G20 ISDN BRI Gateway Details on page 47 for further information.

A10 Analog Telephone Adapter Setup

The A10 Analog Telephone Adapter combines IP routing, VPN/Security, and Quality of Service (QoS) for voice and FAX calls over any IP or PSTN network.

Add an A10 ATA to the network

- With power applied correctly, the power LED will blink GREEN.
- The A10 gateway does **not** support PoE (power over Ethernet). Use the wall adapter to provide power to the device.

A10 ATA IP address assignment

DHCP server

- Connect the A10 to the same subnet as the local Quick Edition system.
- The A10 will broadcast to a known multicast address.
- In response to the broadcast, Quick Edition will add the A10 to Device Management in the web-based administration interface.

No DHCP server

- The A10 ATA does not support Zeroconf IP address assignment.
- Connect A10 RJ-45 LAN port directly to an Ethernet port on your PC.
- Configure the PC with a static IP address.
- Use a web browser directed to 192.168.123.10 to assign static IP, subnet mask, and default A10 IP address.
- Save running configuration and reload.
- Reset assigned IP address on the PC.
- Connect the A10 to the same subnet as the local Quick Edition system.
- The A10 will broadcast to a known multicast address.
- In response to the broadcast, Quick Edition will add the A10 to Device Management in the web-based administration interface.
- See Assigning a Static IP Address to the A10 or G20 on page 50 for more information.

A10 ATA IP activity

- When the device has joined the Quick Edition site, the power LED will be solid GREEN and the device will be listed in Device Management in the web-based administration interface.
- By default, the name given is “A10”. The device is not provided with an extension number.

A10 ATA port assignment

Configure SIP Proxy Identities

- Create internal system identities to be associated with the ATA ports for connecting analog devices.
  - Program on the SIP Proxy menu in the web-based administration interface.

Associate Proxy Identities to ports on the A10

- Associate the previously created identity with a specific physical port on the A10 ATA.
  - Program in A10 Device Management in the web-based administration interface.

System Options Configuration

Note:
Refer to Accessing System Options on page 27 for log on instructions.

Create groups

- Ring groups provide the ability to call many telephones at the same time.
- Create groups using the web-based administration interface.
- Incoming calls from the PSTN can be directed immediately to a group.
- You can create a maximum of ten groups, each having a maximum of 10 members.
- Unanswered calls to the group can be forwarded to a system defined destination. Refer to Create a Group on page 30 for details.

Create auto attendant extensions and record greetings

- The auto attendant can direct calls to any telephone in the system by dialing the extension or using the dial by name functionality.

- Create auto attendant extensions by using the web-based administration interface.
- A default auto attendant extension is provided with a default recorded greeting.
- Record custom greetings from any telephone by accessing the System Options menu. Refer to Auto Attendant on page 34 for additional information.

Assign the operator extension

- The operator extension allows users to reach the assigned telephone by pressing zero.
- The system administrator can change the operator extension from the telephone or web-based administration interface at any time.

Assign incoming extension on G11 gateway ports

- By default all incoming ports on the gateway are directed to the auto attendant (extension 500 in North America).
- The system administrator can configure the ports to be directed to any active extension on the network, including telephones, auto attendants, or ring groups.
- Incoming line assignments are configured using the web-based administration interface.

User Options Configuration

Assign call forwarding destination

- By default all incoming calls to a telephone are forwarded to voicemail after 3 rings.
- Configure the call forward destination using the telephone or web-based user interface.
- Users can configure calls to be forwarded to any external or internal telephone number.

Record a voicemail greeting

- Record the custom voicemail greeting from the telephone user interface.
- A default greeting is provided for each telephone on the network.

Record a voicemail name

- Record the custom voicemail name from the telephone user interface.

- This name is used by the dial-by-name functionality provided by the auto attendant.

Assign zero redirect

- Zero redirect allows callers to reach a user-defined extension by dialing 0.
- Configure the zero redirect destination using the telephone or web-based user interface.
- By default zero redirect is disabled for every telephone on the network.
- Refer to the telephone user guide for additional configuration options.

Note:

When you forward an incoming outside call to a PSTN number, the call will consume two PSTN lines (one incoming, and one outgoing) and two ports on the PSTN gateway while the call is active. Ensure appropriate PSTN resources are available to support this.

Chapter 3: Advanced System Configuration

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 as your telephones. Refer to Features on page 101 for a list of Web-based and telephone interface configurable system options.

CAUTION:
Password protection will prevent unauthorized users from changing system options. Set a password that is known only to you.

To access options using the web-based administration interface

The password that you enter to log in is encrypted before the information is sent to the system. Refer to Security on page 77 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`**). To determine the telephone IP address, press the **#** key and then the PAGE RIGHT ( ) button to toggle the display between IP address and date and time.
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:
   - Click **Yes** to proceed without installing the security certificate.
   - Click **View Certificate** if you want to install the security certificate. Click **Install Certificate** and follow the instructions.
5. Click **System Options** on the **Login** screen.
6. Type the administration password, set when the first telephone was installed.
7. Click **Login**.
   The **Device Management** page is displayed. Links to additional pages are in the list on the left side of the screen. Click an item in the list to view and edit system-wide settings.

To access system options menu through buttons on the telephone

1. At the telephone, press OPTIONS ( ) button below and to the right of the display.

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. Press **Done**. The **System Options** menu is displayed.

Tip:
To view options that are displayed on the next or previous telephone display screen, 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 is 0 for no lockout). The system administrator must reset the user account.

Changing the Administration Password

A password must contain at least four numbers up to a maximum of 32 numbers (default is 6). The password can be changed as often as necessary. You can change the complexity of passwords, expiration requirements, the frequency of required password changes, etc.

Tip:
For more information, see Security on page 77.

There are two ways to change the administration password using a web browser:

A) From the Device Management page:

1. Access the **System Options** web page.
2. Click **Change Admin Password** in the top right corner of the **Device Management** page.
3. Type the current password in the **Current Password** field.
4. Type the new password in the **New Password** field.
5. Re-type the new password in the **Confirm Password** field.
6. Click **Submit**.

B) From the Security page:

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

> **CAUTION:**
> If you forget the administration password, refer to “System Problems and Solutions” in the *Troubleshooting Guide*, and 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**.
6. Press **Next**.
7. Re-enter the **new password** when you are prompted, and then click **Next**.
8. Enter the **old password** when you are prompted, and then click **Next**.
9. Press **Ok**.

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. Press **Ok**.
3. Enter the **new password**, and then press **Next**.
4. Re-type the **new password** to confirm it and then press **Next**.
5. Enter your **old password**, and then press **Next**.
6. Press **Ok**.
7. Press the PHONE/EXIT ( ) button.

Create a Group

A maximum of ten 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 web-based administration interface or the Multi-Site Provisioning Tool (MPT).

To add a group

1. Log in to the **System Options** web page.
2. Select **Corporate Directory**, on the **System Options** menu.
3. Click **Add Entry**.
4. Select **Group** as the **Entry Type**.
5. In the **Extension** field, type an extension number that will be used to ring all members of the group. The number must be unique and in the system DN range.
6. Type a name for the group in the **Group Name** field.
7. Click **Create**. The new group is displayed in the **Groups** list.

To delete a group

1. Click the selected group extension in the **Corporate Directory page**.
2. Click **Delete Group**.
3. Click **Submit**.

To edit group details

1. Click **Change Details** in **View Group Details**.
2. You can change the **Group Name**.
3. Click **Submit**.

Group Members

To add members to a group

1. Click the selected group extension in the **Corporate Directory** page.
2. Click **Members** in **View Group Details**.
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:
A group cannot forward a call to another group. 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. Click **Forwarding** in **View Group Details**.
2. Click **Change Details**.
3. Select **Enable Forwarding**.
4. In 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 in 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 page 40).
6. Click **Submit**.

Note:
To Apply, Delete, Enable or Disable a Rule, refer to Dialing Rules on page 73.

Group Dialing Rules

Dialing rules specify numeric dialing patterns that the system can distinguish and process in a certain way (allow or disallow). These patterns, when applied globally or to a group, either allow or prevent users from making certain types of calls. 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. Refer to Group Dialing Rules on page 74 for configuration details.

Call Pickup Groups

Call pickup provides the ability for a user to answer a call that is ringing on another telephone in the same pickup group.The execution and receiving telephones must be running software release 3.3 or later. Call pickup alert is OFF by default. Each member of a group can choose to enable a visual or audio alert to signal when there is a call to the group; refer to Call Pickup in the Telephone User Guide.

Access Call Pickup Groups Programming

To access call pickup groups programming in the web interface

1. Access the **System Options** web page.
2. Select **Corporate Directory** on the **System Options** menu.

To access call pickup groups programming on the 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 **Call Pickup Group** on the **System Options** menu.

Add a Call Pickup Group

To add a call pickup group and group members using a web browser

1. Click **Add Entry** in the **Corporate Directory** window.
2. Select **Call Pickup Group** in the **Entry Type** field.
3. Type a name for the group in the **Pickup Group Name** field.
4. Click the **Notification of External Calls Only** box if required; leave the box clear to allow notification of all calls.
5. In **Notification After # Rings**, enter the number of rings to delay before initiating a call pickup alert.
6. Select an extension and click **Add Member** after each selection, in the **Members** field.
7. The **Remote Pickup** check box, enabled by default, allows pickup of calls to that extension.
8. Click **Create**. The new pickup group is displayed in the **Corporate Directory** list.

To add a call pickup group using a Quick Edition IP telephone

1. Click **Add** on the **Call Pickup Groups** screen.
2. Enter a name for the new pickup group (see page 18 for keypad instructions).
3. Press **Next** to navigate to the **Ring Delay** screen.
4. Enter the number of rings to delay before initiating a pickup alert.
5. Press **Done** to navigate to the **Notification Type** screen.
6. Select **All Calls** to allow notification of all calls to the group or **External Calls** to allow notification of only external calls to the group.

To add extensions to a pickup group using a Quick Edition IP telephone

1. Select the pickup group on the **Call Pickup Groups** screen.
2. Press **AddEx** on the **Group Details** screen.
3. Press **FrDir** on the **Add Extension to Group** screen, to select an entry from the corporate directory or type the extension number.
4. Press **Done**.

Delete a Call Pickup Group

To delete a call pickup group using a web browser

1. Click **Remove Entry** in the **Corporate Directory** window.
2. Select the **Call Pickup Group** in the **Entry To Remove list**.
3. Click **Remove** and then click **Remove Entry** to confirm.

To delete a pickup group using a Quick Edition IP telephone

1. Select the pickup group on the **Call Pickup Groups** screen.
2. Press **DelGr** on the **Group Details** screen.
3. Press **Yes** to confirm and then press **Ok**.

Delete Call Pickup Group Members

To delete members from a call pickup group using a web browser

1. Click **Call Pickup Group** in the **Corporate Directory** window.
2. Select the group in the **Number** field.
3. Click **Delete** for each member that you want to delete.
4. Click **Submit**.

To delete an extension from a pickup group using a Quick Edition IP telephone

1. Select the pickup group on the **Call Pickup Groups** screen.
2. Select the extension on the **Group Details** screen.
3. Press **DelEx** on the **Extension Detail** screen.

Set Pickup Permission Status

To set permission status of a pickup group member using a web browser

1. Select the group in the **Corporate Directory** window.
2. Click the **Remote Pickup** check box to allow or deny the pickup of calls to that extension.
3. Click **Submit**.

To set permission status of a pickup group member using a telephone

1. Select the pickup group on the **Call Pickup Groups** screen.
2. Select the extension on the **Group Details** screen.
3. Press the Line/Feature button beside **Rmt Pickup** on the **Extension Detail** screen.
4. Select **Allow** or **Block** to allow or deny permission for others to pick up calls to this extension.

Auto Attendant

By default, the auto attendant answers incoming calls from PSTN or a SIP (VoIP) network connected to the gateway and instructs callers to dial the extension number, enter the name of the person, or press 0 to reach the operator. The default operator extension (200 in North America) can be changed.

You can create a two level auto attendant to provide secondary messages or create a multi-level auto attendant configuration.

You can record a custom greeting 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.

A caller can use the dialpad keys to spell a name and have the call transferred. For example, to reach someone named “Young, Mary”, 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, dials zero, or uses a rotary or pulse telephone, the call is transferred to the designated operator. Calls from rotary or pulse telephones are directed to the operator because rotary/pulse telephones do not generate tones.

Tip:
To configure the auto attendant to answer calls from a PSTN gateway, see Lines Settings on page 44. To configure the auto attendant to answer calls from a SIP network, see To assign a SIP identity on page 88.

Basic

There is no configuration required to use the default auto attendant. In North America the auto attendant is assigned extension 500. The default greeting instructs callers to dial the extension number, enter the name of the person, or press 0 to reach the operator. You may want to record a custom greeting to replace the default greeting.

Two-Level

In addition to providing the standard custom greeting and call routing features, the primary auto attendant can direct callers to dial a number to hear the secondary message; for example, a message regarding directions to a location or notices that may not be applicable to all callers. Refer to Two-Level Auto Attendant on page 9 for an illustrated example.

Multi-Level

Multi-level auto attendant functionality provides for the programming of single digit routing. Callers can be directed to a telephone, group, voicemail, another auto attendant, or external number by dialing a single digit between 2 and 6. The total recording time available for all auto attendant prompts is four minutes. The maximum number of auto attendants is 20.

Table 1: Multi-Level Auto Attendant User Call Flow Example

A caller telephones your company. The caller reaches your default auto attendant, extension number 500 in North America.	PSTN to Your Company Inc. to AA 500 Auto attendant 500 presents options.
Caller hears instructions from AA 500: If you know the name of the party that you wish to reach, press 1. To reach the Sales Department, press 2. To reach Customer Support, press 3. For immediate assistance, press 0.	Caller presses 2 and is directed to AA 502, Sales Department.
Caller hears instructions from AA 502: To reach the first available agent, press 1. For shipment tracking information, press 2. To reach the Sales Manager, press 3.	Caller presses 1 and is directed to the first available member of the Sales Group.

Record Greetings

Record a custom greeting for each auto attendant configuration.

To record an auto attendant greeting using a Quick Edition telephone

1. Using any telephone connected to the network, display the **System Options** menu.
2. Select **Auto Attendant** on the **System Options** menu.
3. Select **Auto Attendant Prompts** on the **Auto Attendants Menu**.
4. Select the prompt that you want to record.
5. Lift the handset and press **Rec** to record the prompt. Press **Stop** when you are finished.
6. Perform one of the following actions:
   - Press **Play** to listen to your recording.
   - Press **Save** if you are satisfied with the recording.
   - Press **Chg** to re-record the greeting, if necessary.

Add an Auto Attendant

Create as many menu choices as you need for your multi-level auto attendant configuration. For example, 2 for Sales, 3 for Support, 4 for Shipping. Menu choice configuration begins with step 5 in To add and configure an auto attendant using a Quick Edition telephone or step 5 in To add and configure the menu for an auto attendant using a web browser.

The following numbers/symbols are reserved:
0. Operator
1. Corporate directory
7. Future use
8. Back to main menu
9. Back to previous menu
*. Repeat prompt.

To add and configure an auto attendant using a Quick Edition telephone

1. Using any telephone connected to the network, display the **System Options** menu.
2. Select **Auto Attendant** on the **System Options Menu** screen.
3. Select **Auto Attendant** on the **Auto Attendants Menu** screen.
4. Click **Add**.
5. Type the extension number, in the auto attendant range, on the **Enter AA Extension** screen.

6. Click **Next**.
7. Type the name of the auto attendant (see Enter a name for the telephone on page 18).
8. Click **Next**.
9. Click **Menu Configuration** on the **AA Details** screen.
10. Press the Line\Feature button to select a **Menu Choice** on the **Menu Configuration** screen.
11. Click **Edit** on the **Edit or Delete?** screen.
12. Select the destination for the menu choice from the **Corporate Dir**.
13. Select **1.Dial** to enable transfer to the destination DN or **2.VDial** to enable transfer to the destination DN mail box.
14. Configure other menu choices as needed.
15. Click **Next** to return to the **AA Details** screen.
16. Click **Save** on the **AA Details** screen.
17. Click **Timeout Action** on the **AA Details** screen if you want to change from the default setting.
18. Press the Line\Feature button to select a **Timeout Action**.
19. Click **Next** and then click **Save**.

To add and configure the menu for an auto attendant using a web browser

1. Log in to the **System Options** web page.
2. Click **Corporate Directory** on the **System Options** menu.
3. Click **Add Entry**.
4. Select **Auto Attendant** in the **Entry Type** list.
5. Type the auto attendant **Extension** number and **Name**.
6. For each **Menu Configuration** option, select an extension in the **Transfer** list.
7. Select the **Direct to VoiceMail** check box to enable transfer to the destination DN mail box; leave the check box clear to enable transfer to the destination DN.
8. Select the **Timeout** option; Default, Disconnect Caller, Main Menu, or Transfer and choose a number.
9. Click **Create**.
10. From an IP telephone on the system, **Record the greeting** (see To record an auto attendant greeting using a Quick Edition telephone on page 36).

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.

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.
5. Click **Name** on the **AA Details** screen.
6. Change the name for the auto attendant configuration. Click Bksp to erase characters; use the keys on the dialpad and the softkeys to enter new characters (see Enter a name for the telephone on page 18 for detailed instructions).
7. Click **Next**.
8. Click **Menu Configuration** on the **AA Details** screen.
9. Select the Line/Feature ( ) button beside the menu option that you want to change.
10. Click **Edit** to select a new DN, or click **Cancel** or **Delete**.
11. Click **Save**.

To edit an auto attendant configuration using a web browser

1. Log in to the **System Options** web page.
2. Select **Corporate Directory** on the **System Options** menu.
3. Select the auto attendant that you want to edit.
4. Click **Change Details** in the **Auto Attendant Details** window.
5. You can change the name, transfer options, and timeout selection.
6. Click **Submit**.

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 (assigned to extension 500 in North America).

To delete a custom auto attendant configuration using a telephone

1. Use a telephone in the network to display the **System Options** menu (see page 27).
2. Select **Auto Attendant** on the **System Options** menu, or press the indicated dialpad key.
3. Select **Auto Attendants** on the **Auto Attendants Menu**.
4. Select the Line/Feature ( ) button beside the configuration that you want to delete.
5. Press **Del**.
6. Press **Yes** to confirm.

To delete a custom auto attendant configuration using a web browser

1. Log in to the **System Options** web page.
2. Select **Corporate Directory** on the **System Options** menu.
3. Select the auto attendant configuration that you want to delete.
4. Click **Delete Extension** in the **Auto Attendant Details** window.
5. Click **Submit**.

Auto Attendant Switching (Night Service)

The operator or an authorized user can assign a programmable softkey on any telephone to enable the activation of a specific, after-hours, greeting. Typically, this greeting would be separate from a multi-level auto attendant configuration.

To create night mode for after-hours call routing

1. Using any telephone connected to the network, display the **System Options** menu.
2. Select **Auto Attendant** on the **System Options Menu** screen.
3. Select **Auto Attendant** on the **Auto Attendants Menu** screen.
4. Click **Add**.
5. Type the extension number, in the auto attendant range, on the **Enter AA Extension** screen.
6. Click **Next**.
7. Type the name of the auto attendant, for example, Night.
8. Record the greeting (see Record Greetings on page 36).

To select a prompt for after-hours call routing

1. Press the **Grtn** softkey.
2. Select the after-hours prompt from the **Auto Attendant Prompts** list.

To program a Grtn (night switching) key

1. On the telephone that will enable the Night greeting, press a softkey for two seconds.
2. Press the Line/Feature button beside the **Greeting** function. Press PAGE LEFT and PAGE RIGHT to view all options.
3. Enter the Admin Password and press **Ok**.

Disable the Default Auto Attendant

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. Log in to the **System Options** web page.
2. Select **Corporate Directory** on the **System Options** menu.
3. Click the number for the **Gateway** that you want to bypass.
4. Click **Lines**. The **View Gateway Lines** window displays.
5. Click **Change Details**. The **Edit Gateway Lines** window displays.
6. Select your operator extension in the **Incoming** list that is associated with the line.
7. 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. Click **Dialing Configuration** in the **System Options** list.
3. Click **Operator** on the **Dial Plan Settings** window.
4. Click **Change Details**. The **Edit Operator Extension** window is displayed.
5. Select the extension in the **Operator Extension** list and then 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. Press **Chg**.
5. Enter the extension number or press **FrDir** and select the entry, and then click **Next**.
6. Press **Save**.

To change the operator extension number

- Refer to the procedures in Set Details on page 54 to change the operator extension number from the default 200 (in North America).

Music On Hold (.wav file)

System Music on Hold (MOH) can be provided by using an audio file on a device or by using a music source connected to an analog gateway (see Music on Hold (physical device) on page 45).

To supply system music on hold by using an audio (.wav) file:

- Use the install wizard to transfer the .wav file to any or all telephones
- Use the wizard to enable up to two sources
- Use the web-based administration interface or a telephone to edit the device MOH.

Note:
Because an audio file will reduce the voicemail capacity on the host telephone, we recommend that the .wav file be installed on a telephone with a limited voicemail requirement, for example, a lobby telephone.

Audio source file size	Voicemail storage capacity
No audio file	20 minutes
Less than 30 seconds	15 minutes
Between 30 seconds and 1 minute	10 minutes
Between 1 minute and 2 minutes	0 minutes.

To transfer the .wav file to a device on the system

Note:
You can obtain the install wizard from the Avaya Technical Support web site at the following address: http://support.avaya.com/QuickEdition. Click **Downloads** to obtain the QE install wizard that includes Music on Hold.

1. Launch the QE_Install_Wizard_vxxxxx.exe.
2. Select **Music-On-Hold Configure**.

Note:
All current music sources are disabled before the transfer begins.

3. Click **Start the TFTP Server**.
4. Follow instructions to transfer the file to the telephones and enable up to two devices.
   - You can choose from three files shipped with the wizard (30 seconds, one minute, two minutes), or transfer your own .wav file.
   - You can test play the .wav file through your PC speakers.
   - An existing audio file will be replaced when a new file is transferred.

To modify the audio tag or remove the .wav file using a web browser

Note:
This procedure applies only to telephones. The gateway cannot host system music with a .wav file.

1. Log in to the **System Options** web page.
2. Select the device to be configured on the **Device Management** window.
3. Click **MOH** in the **Device Details** window.
   – Click **Change Details** to change the audio tag name and click **Submit**;
   – Click **Remove MOH File** to remove the MOH source file from the device, and then click **Remove** and **Confirm**.

To assign or unassign MOH using a web browser

Note:
This procedure applies to telephones and analog gateways, wav file and physical source.

1. Log in to the **System Options** web page.
2. Select **Applications** on the **System Options** menu.
3. Click **MOH** to launch the **View MOH Settings** window.
4. Click **Change Details** to launch the **Edit MOH Settings** window.
5. Select **Assign** or **Unassign** in the list and then click **Submit**.

To modify MOH using a Quick Edition telephone

Note:
This procedure applies to telephones and analog gateways

1. Log in to the **System Options** menu.
2. Select **Music On Hold** on the **System Options** menu.
3. Press the corresponding Line/Feature button for the telephone that you want to modify.
   - The MOH Details screen will display devices that are hosting the .wav file or the physical music source.
4. Click **Edit** to change the audio tag name (telephone only) or to enable or disable the status.
5. Click **Done** and then PHONE/EXIT.

Analog Gateway Details

See *Avaya 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 a Quick Edition telephone on the network. Related procedures are given throughout this chapter where applicable.

Note:
If your gateway/adapter is not a G11, refer to A10 Analog Telephone Adapter (ATA) Details on page 49 or G20 ISDN BRI Gateway Details on page 47.

To edit the gateway extension number using a web browser

1. Log in to the **System Options** web page.
2. Select the gateway in the **Name** list on the **Device Management** screen.
3. Click **Change Details** to edit the extension number.

To access analog gateway options through buttons on the telephone

1. Press OPTIONS on the telephone.
2. Select **Options** on the **Main** menu.
3. Select **System Options** on the **Options** menu.
4. Type the administration password when you are prompted.
5. Press **Done**. The **System Options** menu is displayed.

6. Select **Gateways** on the **System Options** menu, or press the indicated dialpad key.
7. Select the extension number of the gateway that you want to configure or press the corresponding Line/Feature ( ) button.

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 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 several reasons that you may want to have one or more of your PSTN lines bypass the auto attendant feature:

- 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 (for example, to a Fax machine or other analog device),
- to make a PSTN line a “private outgoing line,” which reserves outgoing ports for the specified extension to the PSTN line,
- if the port is not used.

To assign an auto attendant configuration to an analog gateway, see Edit Auto Attendant Configurations on page 38.

To modify gateway line settings

You must assign an auto attendant configuration to a PSTN gateway using the web-based administration interface.

1. Log in to the **System Options** web page.
2. Select **Device Management** on the **System Options** menu.
3. Select the gateway in the **Name** list.
4. Click **Lines** on the **Gateway Details** window.
5. Click **Change Details** on the **View Gateway Lines** window.
6. You can change the incoming and outgoing settings.
7. Click **Submit**.

To adjust loop length to compensate for line echo

1. Log in to the **System Options** web page.
2. Select **Device Management** on the **System Options** menu.
3. Select the gateway in the **Name** list.

4. Click **Lines** on the **Gateway Details** window.
5. Click **Change Details** on the **View Gateway Lines** window.
6. Select **Short** in the **Loop Length** list and then click **Submit**.

Music on Hold (physical device)

There can be only one physical music source per system. Use the Music on Hold audio input stereo mini-jack on the gateway to support the music-on-hold feature. You may also provide system music on hold through the use of a .wav file (see Music On Hold (.wav file) on page 41).

To modify, enable or disable a physical music source

Refer to the .wav file programming procedures on page 42.

Network Settings

To modify gateway settings using a web browser

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

To modify gateway settings using buttons on the telephone

1. Log in to the **System Options** menu.
2. Select **Gateways**.
3. Select the gateway that you want to modify.
4. Click **IP Address**.
5. Click **Chg** and then click **Next** at the Warning message.
6. You can change the IP Address, Netmask, and Gateway default IP Address.
7. Click **Save** and then **Exit**.

Advanced Settings

To modify loop detect debounce, incoming call guard timer, and impedance

1. Log in to **System Options** web page and select **Device Management**.
2. Select the gateway in the **Name** list and click **Advanced** on the **Gateway Details** window.
3. Click **Change Details** on the **View Gateway Network Info** window.
4. Enter new values for **Loop Detect Debounce** (default 80ms for G11; the wait time before a call is disconnected locally) or **Incoming Call Guard Timer** (default 2000ms; the wait time before detecting ringing again).
5. You can change the **Impedance** for each line. Contact your service provider before selecting an alternate match Line Impedance for your system.
6. Click **Submit**.

Impedance name	Line Impedance	Country/Region
ATTComplex	350 Ω + (1000 Ω || 210 nF)	North America, Singapore, India, Thailand, Hong Kong, Russia, Mexico, Argentina, Japan
600Complex2	600 Ω + 1 μF	Alternate match
900Complex2	900 Ω + 2.16 μF	Alternate match
900Complex1	900 Ω + 1 μF	Alternate match
600R	600	Alternate match
600Complex1	600 Ω + 2.16 μF	Alternate match
900R	900	Alternate match
TBR21	270 Ω + (750 Ω || 150 nF) (TBR21) and 275 Ω + (780 Ω || 150 nF)	Switzerland, Austria, France, Spain, Netherlands, Belgium, Norway, Italy
Complex1	320 Ω + (1050 Ω || 230 nF)	United Kingdom
Complex2	370 Ω + (820 Ω || 110 nF)	Alternate match
Complex3	275 Ω + (780 Ω || 115 nF)	Alternate match
Complex4	120 Ω + (820 Ω || 110 nF)	Alternate match
Complex5	200 Ω + (680 Ω || 100 nF)	Alternate match
ANZComplex1	220 Ω + (820 Ω || 120 nF) and 220 Ω + (820 Ω || 115 nF)	Australia, South Africa, Germany
ANZComplex2	370 Ω + (620 Ω || 310 nF)	New Zealand
GlobalComplex	Global impedance	Alternate match

G20 ISDN BRI Gateway Details

There are five major steps involved in the installation and configuration of the G20:

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 *Avaya Quick Edition Safety and Quick Installation Instructions for: G11 Global Analog Gateway, G20 ISDN BRI Gateway, A10 Analog Telephone Adapter* (Document No. 16-601414)).
4. Assign SIP proxy identities (see next procedure).

   Tip:
   The G20 is shipped with a default IP address of 192.168.123.10.

5. Map the identities (see To configure the G20 on page 48).

To add a SIP Proxy identity for G20

1. Log in to the **System Options** web page.
2. Select **SIP Proxy** on the **System Options** menu.
3. Click **Identities** in the **SIP Proxy Configurations** window.
4. Click **Add**. The **Add SIP Proxy Identity** window 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 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. Select **Unassigned** to configure the trunk for incoming calls only.

6. Click **Submit**.

To configure the G20

1. Log in to the **System Options** web page.
2. Select **Device Management** on the **System Options** menu.
3. Select the G20 to be configured in the **Name** list. The **Edit ISDN Gateway Details** window 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 Identity 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. **Note**: The selection is dependent on the service that you purchase from your service provider. For example, select DID if you have purchased a DDI trunk.
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 (Digital Signaling System #1) signaling protocol. If you select DMS-100, layer 3 uses Digital Multiplex System signaling.
Layer 3 UniSide	User is selected by default, which means that the G20 port runs the user side part of signaling protocol.

5. If you need to change the network information, clear the **DHCP Address** check box. Refer to Assigning a Static IP Address to the A10 or G20 on page 50.

CAUTION:
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**.

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 *Avaya Quick Edition Safety and Quick Installation Instructions for: G11 Global Analog Gateway, G20 ISDN BRI Gateway, A10 Analog Telephone Adapter* (Document No. 16-601414)).
3. Assign SIP Proxy Identities (see next procedure).

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

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

4. Configure the A10.

To add a SIP Proxy identity for A10

1. Log in to the **System Options** web page.
2. Select **SIP Proxy** on the **System Options** menu.
3. Click **Identities** in the **SIP Proxy Configurations** window.
4. Click **Add**. The **Add SIP Proxy Identity** window 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 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. Select **Device Management** on the **System Options** menu.
3. Select the A10 to be configured. The **Edit ATA Gateway Details** window 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 to the A10 or G20 on page 50.

CAUTION:
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 reset.

6. Click **Submit**.

Assigning a Static IP Address to the A10 or G20

This procedure describes how to configure the static IP address of the A10 or G20 device when a DHCP server is not available.

1. Using the included Ethernet cable, connect the G20 or A10 RJ-45 LAN port directly to the Ethernet port on your PC.
2. 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, and select **Properties**.
- In the **Local Area Connection Properties** window, select **Internet Protocol (TCP/IP)** and click **Properties** (note the current settings; you will need to set them back later).

- Select **Use the following IP address**.
  IP address: enter an IP address that conforms to the default IP address (192.168.123.10) 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**.

3. Open your web browser and connect to https://192.168.123.10.
4. Type **nimdbg** for User Name and **54321** for Password.
5. On the displayed interface, go to **Configuration** menu.
6. Under Network, click **IP/DNS**.
7. On the displayed page, click the **Interfaces** tab.
8. Click **eth0** under **Name**.
9. On the displayed page, click the **Configuration** tab.
10. For IP Address, select the **User Defined IP Address** radio button.
11. Enter the assigned IP address and IP mask for the device.
12. Click **Apply** at the bottom of the page.
13. Select **Save** on the **Configuration** menu.
14. On the displayed **Save Configuration** window and click **Save**.
15. On the displayed **Reload Device** window, click **Reload**.
16. Reset the IP address assigned to your PC to its previous setting (see step 2).
17. From Device Management on the Web-based administration interface, open the G20 or A10.
18. Clear the **DHCP Address** check box.
19. Enter the assigned static IP address and netmask address you set in step 11.
20. For Gateway, enter the gateway IP address for A10.
21. Click **Submit**.

Chapter 4: Additional Configuration

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.

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

- Teleworker on page 65
- Analog Gateway Details on page 43
- A10 Analog Telephone Adapter (ATA) Details on page 49
- G20 ISDN BRI Gateway Details on page 47.

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. Press **Ext** or press the indicated dialpad key.

   The extension number, name, IP address, and MAC address are displayed.

4. Press **Back** 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. All telephones, by default, belong to paging zone 2.

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. If there is more than one match, the auto attendant will you prompt you to enter a number for your selection.

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. 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. Click **Change Details** in the **Set Details** window.
4. In the **Edit Set Details** window, type the new name in **Name** field, type an unused extension number in the **Extension** field, or select a zone in 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. Press **Chg**.
5. Press **Bksp** 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. Press **Save**.

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 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** window is displayed.
3. Click **Reset Password**. The **Reset User Password** window 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, press **Yes**.
5. Press **Ok**.
6. The new user-options and voicemail password is reset to a random value and displayed for Release 3.1 and later. For versions prior to Release 3.1, the password will be reset to 12345. 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 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 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** window is displayed.
3. Click **Networking**. The **Network Settings** window is displayed.
4. Click **Change Details**. The **Edit Network Settings** window 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. Press **Chg**.
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`**). Press the **`.`** softkey to add separator characters between the digits.
6. Press **Next**.
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. Press the **`.`** softkey to add separator characters between the digits.
8. Press **Next**.
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. Press the **`.`** softkey to add separator characters between the digits.
10. Press **Next**.

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. Press the **.** softkey to add separator characters between the digits.
12. Press **Next** followed by **Save**.

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.

Available special features include:

- Email Notification of Voicemail (Email Fwd Options, page 68)
- Web-based System Administration (WebAdm Sys Options)
- Teleworker Application (Teleworker on page 65)

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 OPTIONS.
2. Select **Opt Features** on the **Main** menu.
3. Select **Email Fwd Options**, **WebAdm Sys Options**, or **Teleworker Options** in the **Feature List**. The registration code is displayed.
4. Press **Exit** to display the previous menu, or press the PHONE/EXIT ( ).

Software Upgrade

All devices on a Quick Edition network must be running the same version of software. If your system is based on Quick Edition Release 2.0 (or earlier) and you would like to add a device that is running 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 installation wizard from the following address: http://support.avaya.com/QuickEdition. Click **Downloads** to obtain the software, language packs, and Music on Hold.

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 web-based administration 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 Quick Edition network. You can perform an upgrade using one of these tools:

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

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:
During a software upgrade, active calls are dropped and new calls cannot be placed or received 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.

Tip:
Upgrade is complete when the phones return to idle state. 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.

2. After the upgrade process completes, reset the System Time and Date (see page 81).

To upgrade the software using a web browser

Note:
You must first run the wizard and then, as part of the upgrade process, select **Upgrade with another interface**.

1. Log in to the **System Options** web page.
2. Click **Software Upgrade** on the **Device Management** window.
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. Click **Confirm Upgrade** when you are prompted to begin the upgrade process.
8. After the upgrade process completes, select **Click Here When Done**.

Tip:
Upgrade is complete when the phones return to idle state. 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.

To upgrade the software using a Quick Edition IP telephone

Note:
You must first run the wizard and then, as part of the upgrade process, select **Upgrade with another interface**.

1. At the telephone that you want to upgrade, display the **System Options** menu.
2. Select **Set Management** on the **System Options** menu.
3. Select **Upgrade** on the **Set Management** menu.
4. Press **Upg**.
5. The IP address of the administration computer (on which the TFTP server is running) should be displayed. If necessary, press **Chg** and enter the correct IP address.

Tip:
To determine the IP address of the administration computer: on the Windows **Start** menu, click **Run**, type **cmd**, and click **OK**. At the command prompt, type **`ipconfig /all`**. Look for the IP address in the list of displayed information.

6. Press **Next**. The **Upgrade Settings** menu displays the software components.
7. Press **Next**.
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). After a clean database upgrade, reset any A10s or G20s in the network.

   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 in the network are upgraded. If not selected, the software on only this telephone is upgraded.
9. Press **Next**.
10. When you are prompted to start the upgrade process, press **Yes**.
11. At the administration computer, exit the installer program.
12. Reset the system time and date.

To upgrade the software using the Multisite Provisioning Tool

The Multisite Provisioning Tool is a Java-based software application that communicates with Quick Edition networks over an HTTPS link. The Multisite Provisioning Tool enables you to simultaneously configure one or more Quick Edition networks from a single central location. Any individual 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. Click **Downloads** to obtain the QE Install Wizard that includes software, language packs, and Music on Hold.

1. On the Windows **Start** menu, point to **All Programs > Avaya Multisite Provisioning Tool**, and then click **Avaya Multisite Provisioning Tool**.
2. Follow on screen instructions to upgrade all or selected devices.
3. Reset the System Time and Date (see page 81).

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 software version is 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
Call pickup configurations	Call detail recording settings
Service provider configurations	A10 and G20 configuration data
Gateways and their settings	

To back up system configuration data using a web browser

1. Log in to the **System Options** web page.
2. Click **Backup & Restore** on the **Device Management** home page.
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.
2. Click **Backup & Restore** on the **Device Management** home page.
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 In on page 99).
2. Select **Backup & Restore** on 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.
2. Select **Backup & Restore** on the **User Options** menu.
3. **Browse** to the backup file.
4. Click **Restore**.

Corporate Directory

The Corporate Directory window will display all of the dialable numbers (maximum 100 entries), 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 external numbers.

Add an Entry

1. Log in to the **System Options** web page.
2. Click **Corporate Directory** on the **System Options** menu.
3. Click **Add Entry** to launch the **Add Corporate Directory Entry** page.
4. Select an **Entry Type** in the list.
5. Complete the type-specific fields that will appear:
   - Groups (see Create a Group on page 30)
   - Call Pickup Groups (see Call Pickup Groups on page 32)
   - Auto Attendants (see Auto Attendant on page 34)
   - Extension (see Teleworker on page 65)
   - External Entries (see External Number on page 63).

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. Click **Corporate Directory** on the **System Options** menu.
3. Click **Add Entry** to launch the **Add Corporate Directory Entry** page.
4. Select **External Entry** in the **Entry Type** list.
5. Type the telephone number including required PSTN or SIP prefixes.
6. Type the name.
7. Click **Create**.

Remove an Entry

The following procedures remove an extension for telephones or G11 gateways 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 remove that entry from the corporate directory.

Note:
To remove a G20 ISDN BRI Gateway or A10 Analog Telephone Adapter, refer to the *Quick Edition Troubleshooting Guide*.

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. Click **Corporate Directory** on the **System Options** menu.
3. Click **Remove Entry** to launch the **Remove Corporate Directory Entry** page.
4. Select the entry to be removed from the 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.

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, press **Next**.
   - To remove a different device, type its extension number, and press **Next**.
5. When you are prompted to remove the extension, press **Yes**.
6. Disconnect the phone that corresponds with the removed extension.

Teleworker

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

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 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 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 Quick Edition network.

If the selected teleworker server host is unavailable, the teleworker client telephone tries another 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 Quick Edition network.
2. Move the teleworker client telephone to the remote office.
3. At the remote office, configure a VPN tunnel to the company 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 Quick Edition network.
4. Verify that the VPN tunnel is operational and passing traffic in both directions.
5. 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 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 Quick Edition network to establish a teleworker session. One of the telephones or PSTN gateways on the 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 68.

6. Press **Teleworker** on the **User Options** menu.
7. Select **Mode** on the **Teleworker** menu.
8. Press **Chg** to enable (Teleworker) or disable (Local) teleworker mode.
9. Press PHONE/EXIT to clear the display area.

   After a session has been established, the teleworker client telephone operates as if it were connected to the 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 Quick Edition network locally, and
- configuration and programming at the remote location.

Programming at the Quick Edition network locally

1. Log in to the **System Options** web page.
2. Click **Add Entry** in the **Corporate Directory** window.
3. Select **Extension** in the **Entry Type** 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` (you **must** include **:** separators in the MAC address)).

Note:
To view the MAC address, press OPTIONS, select **Set Details**, select **Ext**, and press PAGE RIGHT to see the MAC and IP addresses. The MAC address is also on a sticker on the underside of the telephone.

6. Click **Validate**.
7. Click **Add Extension**.
8. At the teleworker client telephone, access **Teleworker** on the **User Options** menu to configure the teleworker client telephone to connect to a teleworker server host.

Configuration and programming at the remote location

1. At the remote office, configure a VPN tunnel to the company 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 Quick Edition network
2. Verify that the VPN tunnel is operational and passing traffic in both directions.
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 VPN device and apply power.
5. Configure the telephone with the same language settings, site name and password as the LAN network-based Quick Edition system and provide a user name. Configure the telephone with a unique extension to ensure the extensions assigned within the Quick Edition system do not shuffle.
6. Select **Teleworker** on the **User Options** menu.
7. Select **Default Server** on the **Teleworker** menu.
8. Press **Chg** 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. Press **Exit** to display the previous menu, or press the PHONE/EXIT ( ) button to clear the display area.
10. To enable teleworker mode, select **Teleworker** on the **User Options** menu.
11. Select **Mode** on the **Teleworker** menu.
12. Press **Chg** to enable (or disable) teleworker mode.
13. Press PHONE/EXIT to clear the display area

    If the site identifier teleworker phone is not identical to that of the company 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 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, press **OK**.

To view the connection status of the teleworker client telephone

1. At the teleworker phone, access the **User Options** menu.
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 Quick Edition network. The teleworker server host IP address is also displayed in the display area.
4. Press **Exit** 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 the Troubleshooting Guide.

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.

This forwarding method does not remove the message from your mailbox; you will have to manually delete the forwarded messages.

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

1. Enable SMTP on the Quick Edition network to permit 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.

Before you begin, obtain the host IP address and port number of the SMTP server. If the SMTP server does not reside in the Quick Edition network address space, the 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.
2. Select **Applications** on the **System Options** menu.
3. Click **Change Details** on the **View SMTP Server** window.
4. Select the **Enable SMTP** check box.
5. Type the IP address of the SMTP server host in the **Server Address** field.
6. If your network uses a different IP port number for SMTP communications, specify the port number in the **Server Port** field (2 - 4 digits; 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. Press **On** to enable SMTP on the telephone.
5. Press **Chg**.
6. Enter the IP address of the SMTP server host. Press the **.** softkey to add separators.
7. Press **Next**.
8. If your network uses a different IP port number (default port 25) for SMTP communications, specify the port number.
9. Press **Next** and then **Save**.

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

1. Log in to the **User Options** interface using the extension number of the telephone that receives voicemail messages; you will need to use the phone user’s password.
2. Click **Voicemail** on the **User Options** menu.
3. In the **SMTP** area, click **Change**.

4. In the **To Address** field, type the e-mail address of the person to whom notifications will be sent (for example, **user1@mycompany.com**).
5. In the **From Address** field, type an e-mail address that the telephone can use to place in the From field of the e-mail header. This value has to look like an e-mail address but it does not have to be a real one (for example, **email@user1phone.com** is acceptable).
6. Select **Enable Notification**.
7. Click **Submit** and log out.

To play an audio file attachment

User PC requirements:

- a plug-in for the Microsoft Windows Media Player (a free download from the Avaya Technical Support site and installed on each user PC - see Notes, below)
- Microsoft Windows XP or 2003
- audio playback capability
- a media player that supports the G.729a CODEC.

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

Note:
The plug-in is not compatible with "N" editions of the Microsoft Windows XP operating system.

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

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

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

Call Detail Recording

Avaya Quick Edition provides the ability to capture information about all phone calls. Refer to CDR Record Fields on page 107 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. Select **Applications** on the **System Options** menu.
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 (must not be less than the Max CDR Per Batch).
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 dial plan will change to the default numbers for that region. Refer to Localization on page 15 for the default list of numbers.

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

- Extension Range—extension numbers can be from two-to-six digits long in the range that you specify.
- Auto Attendant Extension Range—extension numbers can be from two-to-six digits long in the range that you specify, a range that is within the extension number range.
- Emergency Code—a specific number can be programmed for help during an emergency.

- Operator Code—a region-specific default number that acts as an alias to forward calls to the operator extension. For example in North America, when you dial zero you are actually calling the operator extension, 200 by default.
- PSTN Code—the default PSTN access number can be reassigned.
- SIP Code—a SIP service provider network—the default access number can be reassigned.
- International Direct Dialing Prefix—the prefix used when dialing out of your country to another country, 1 to 6 digits.
- National Direct Dialing Prefix—the prefix used when making a long distance call from one city to another within your country. Some countries do not have a NDD prefix requirement.
- Country Code—the number used to represent a country or integrated numbering plan. When dialing out to another country, the country code (destination) follows the IDD prefix.
- Area Codes—Regional area codes are defined by each country, usually between 2 and 5 digits. When dialing out to another country, the area code for the specific city (in the destination country) follows the country code (destination).

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. Select **Dialing Configuration** on the **System Options** menu.
3. Click **Edit Dial Plan** on the **View Dial Plan Settings** window.
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 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.

To view dialing rule patterns

1. Log in to the **System Options** web page.
2. Select **Dialing Configuration** on the **System Options** menu.
3. Click **Dialing Rules** in the **Dial Plan Settings** window.

To create a custom dialing rule pattern

1. Log in to the **System Options** web page.
2. Select **Dialing Configuration** on the **System Options** menu.
3. Click **Dialing Rules** in the **Dial Plan Settings** window.
4. Click **Create Rule**.
5. Type a description of the rule in the **Rule Description** field (up to 21 characters).
6. Type a rule pattern in the **Pattern** field. 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 specifc 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. Select **Dialing Configuration** on the **System Options** menu.
3. Click **Dialing Rules** in the **Dial Plan Settings** window.
4. Click **Delete Rule**.
5. Select the rule that you want to delete in the **Delete Rule** list. You cannot delete the default dialing rules. 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.

To apply a dialing rule to a group

1. Log in to the **System Options** web page.
2. Select **Corporate Directory** on the **System Options** menu.
3. Click the **Group** extension on the **Corporate Directory** page.
4. Click **Dialing Rules** in the **View Group Details** window.
5. Click **Add Rule**. The **Edit Group Add Dialing Rules** window is displayed.
6. Select one of the options in the **Add Rule** list. Your choice determines which calls will be affected by the rule. You can apply rules, depending on region, to:
   - select **Emergency**; all calls placed to the emergency number.
   - select **Extension**; all calls placed to an extension in the Corporate directory.
   - select **External**; all calls to the PSTN.
   - select **Operator**; all calls redirected to the designated Operator telephone.
   - select **VoIP E164**; all calls to a SIP (VoIP) network.

   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.
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** window on display, click **Dialing Rules**.
2. Click **Remove Rule**. The **Edit Group Remove Dialing Rule** window 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. Select the rule that you want to delete in the **Delete Rule From Group** list.
4. Click **Submit**.

To enable or disable a dialing rule for a group

1. With the **View Group Details** window on display, click **Dialing Rules**.
2. Click **Change Details**. The **Edit Group Dialing Rules** window 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**.

Global Dialing Rules

You will apply a dialing rule from a list. When you configure a global dialing rule, it will apply to all telephones in the system.

To apply a dialing rule for all telephones

1. Log in to the **System Options** web page.
2. Select **Corporate Directory** on the **System Options** menu.
3. Click **Global Dialing Rules** on the **Corporate Directory** page.
4. Click **Add Rule** in the **View Global Dialing Rules** window.
5. Select one of the options in the **Add Rule** list. Your choice determines which calls will be affected by the rule.
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 Global Dialing Rules** list.

To delete a dialing rule from all telephones

1. With the **View Global Dialing Rules** window on display, click **Remove Rule**.
2. Select the rule in the **Delete Global Rule** list.
3. Click **Submit**.

To enable or disable a dialing rule for all telephones

1. With the **View Global Dialing Rules** window on display, click **Change Details**.
2. Select **Allow** or **Disallow** beside the rule that you want to edit.
3. 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. Click **Dialing Configuration** in the **System Options** list.
3. Click **Operator** on the **Dial Plan Settings** window.
4. Click **Change Details**. The **Edit Operator Extension** window is displayed.
5. Select the extension in the **Operator Extension** list.
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. Press **Chg**.
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, press **FrDir** and then select the entry.
6. Press **Save**.

To change the operator extension number

- Refer to the procedures in Set Details on page 54 to change the operator extension number from the default 200 (in North America).

Security

General

Password Rules

To modify the password rules

1. Log in to the **System Options** web page.
2. Select **Security** on the **System Options** menu.
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; 0 to 3600 seconds; 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 Call Detail Recording 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 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** on 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** on the **System Options** menu.
3. Click **Authorization Codes** in the **Security** page.
4. Click **Delete Code**.
5. Select the code from the list.
6. Click **Submit**

Admin Password

Refer to Changing the Administration Password on page 28 for the procedure.

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.

System Language Settings

Changing language settings will change the system and telephone user interface languages. 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** on the **System Options** menu.
3. Click **Change Details** in the **View System Language** field.
4. Select the **System Language** from the list.
5. Click **Submit**. The language changes on all telephones in the system.

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

1. At the telephone, display the **System Options** menu.
2. Select **System Language** on the **System Options** menu.
3. Press **Change**.
4. Select the language and press **Yes**.
5. Press **OK** 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. Refer to Localization on page 15 for the default list of 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** on the **System Options** menu.
3. Click **Change Details** in the **View System Region** field.
4. Select the **System Region** from the list.
5. Click **Submit**.

To reset the dial plan using a web browser

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

Conference Tones

The conference tones provides you with an audible indication of a party joining your conference. The conference tone is disabled by default and can be enabled only for Italy.

To modify the conference tones

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

Language Packs

The software on each Quick Edition IP telephone includes languages for the telephone and system interfaces. Language packs provide languages to the web interfaces and auto attendant and voicemail prompts.

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 Quick Edition devices. The language wizard will auto-run.
2. Select a language from the 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 Quick Edition network. You can choose to change the language for a single device or 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 in the list and click **Next**.
9. Confirm the update settings and click **Next**.
10. Click **Start** and then click **Next**.
11. Click **Next** when the phones return to idle state and the gateways display a green LED, after approximately one to two minutes.
12. Click **Yes** in the **Warning** window and then click **Close** to exit the 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. Select **Localization** on the **System Options** menu.
3. On the **Language & Region** page, click **Time & Date**.
4. Click **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. Press the **/** softkey to add a separator.
   Enter the time in HH:MM (24-hour) format (for example, **`15:36`**). Press **:** to add a separator.
5. Press **Save**.

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

Note:
For other configurations, if your 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 changing the audio bandwidth and/or enabling priority tagging. The COder/DECoder (CODEC) used to encode the audio path can be selected. By default, the audio path is encoded using the standard G.711 (64 Kbps) CODEC. G.729a (8 Kbps) is negotiated automatically if required to support calls initiated at the far end. The G.729a CODEC is also used to store Voicemail data.

To adjust the audio bandwidth

1. Log in to the **System Options** web page.
2. Select **Networking** on the **System Options** menu.
3. Click **Change Details** in the **View Audio Bandwidth** window.
4. Select one of the following options in the **Audio Bandwidth** list:
   - **High**—Sound is digitally encoded using a high-quality format, G.711, that consumes approximately 64 Kbps of bandwidth on the network.
   - **Low**—Uses a compressed format, G.729a, for sound that consumes approximately 8 Kbps of bandwidth on the network.

   Note:
   Your service provider may have specific bandwidth requirements.
5. 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 routers or switches to implement more than one logical network on the same physical subnet and/or improve Quality of Service (QoS). Priority tagging improves QoS when Quick Edition traffic is routed over a subnet that implements VLANs.

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

Avaya 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 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 Quick Edition voice traffic has priority over data. Your LAN administrator must determine the required settings.

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. Settings affect all devices in the 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.

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.

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 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 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 Quick Edition devices are included in the same VLAN.

To configure priority tagging for a Quick Edition network

1. Log in to the **System Options** web page.
2. Select **Networking** on the **System Options** menu.
3. Click **VLAN Settings** and then click **Change Details**.
4. To enable the priority tagging of voice traffic:
    - Select **Enabled** in the **Audio Tagging** list.
    - Select the priority level to apply to voice traffic, in the **Audio Priority** list.
5. To enable the priority tagging of data traffic select from the **Data Tagging** 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.
6. If you enabled the priority tagging of data traffic, select the priority level to apply to data traffic.
7. 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 Quick Edition devices do not support IEEE 802.1p priority tagging), the changes are cancelled and a status message is displayed.

Chapter 5: Service Provider

Service Provider

Certified SIP (Session Initiation Protocol) VoIP service provider feature supports interoperability 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 Quick Edition Network to Interoperate with CCS/SES and CM* (Document No. TSS-000002), available at http://support.avaya.com/QuickEdition.

Refer to Figure 10: SIP Service Provider Configuration on page 14 for a configuration example. When the 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

Obtain information about their SIP network from your service provider. You will be provided with information that is similar to all or some of the information shown below:

Proxy:	sip.sipexample.com	Port: 5060
Registrar:	sip.sipexample.com	Port: 5060
Session Border Controller:	outproxyhost.sipexample.com	
Domain:	sipexample.com	
Realm:	auth_domain.sipexample.com	
Register Expiry Time:	time, in seconds	
Keep-Alive Time:	time, in seconds	

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

You must assign SIP identities (usernames and passwords) to the devices on your network so that calls coming in through the SIP network can be mapped and delivered to a specific telephone and/or handled by the auto attendant feature. Assign SIP identities through Service Provider options to make each identity available to any telephone in the network.

Consult your service provider to obtain SIP identities 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

If one person receives all calls to the 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

SIP identities and passwords may be mapped to individual telephone extensions or group extensions. When a telephone is assigned a SIP identity, the telephone registers with the SIP proxy server on the service provider's network to handle 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. 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 service provider information (page 85), 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 realm information.
2. Add a service provider configuration.

3. If you intend to route incoming calls to a group, refer to Case D: Incoming Calls are Routed to a Group on page 95.
4. Assign SIP identities. See To assign a SIP identity on page 88.

Note:
When adding service provider configurations, Quick Edition supports only one configuration per network.

To add a service provider configuration

1. Log in to the **System Options** web page.
2. Select **Service Provider** on the **System Options** menu.
3. Click **Add Configuration**. The **Add Service Provider Configuration** window is displayed, with the following fields:

- **Domain Name**: type the service provider domain name or IP address for registration (for example, `sipexample.com or 10.10.10.10`). This value is included in the “From” header of outgoing SIP calls.
- **Proxy Host**: 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 or 10.10.10.10`).
- **Proxy Port**: 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.
- **Registrar Host**: type the IP address or fully qualified domain name of the registration server that your telephones have to contact to receive incoming VoIP calls (for example, `sip.sipexample.com or 10.10.10.10`). If your service provider did not give you any information about a registration server, type the same value that you entered for the **Proxy Host** field.
- **Registrar Port**: type the port number through which the registration server listens for SIP communications (for example, `5060`).
- **Outbound Proxy Host**: type the IP address or fully qualified domain name of the SBC host if your service provider uses a separate outbound proxy server or session border controller to manage calls leaving the SIP network.
- **Outbound Proxy Port**: type the port number through which the outbound proxy server or SBC listens for SIP communications. Set the value to 0 if the service provider network contains redundant SIP servers where redundancy is based on a DNS algorithm.
- **Realm**: type the realm information that is used to locate authorization credentials (for example, `auth_domain.sipexample.com`).
- **Register Expiry Time**: type the amount of time that telephones wait between sending updates to refresh registration information (for example, `3600`). The value is measured in seconds and is provided by your SIP service provider.

- **Keep-Alive Time**: type the amount of time between sending Keep Alive. Keep Alive maintains connectivity with the SIP service provider by using a simple ping-like command that maintains connectivity through the firewall.
- **International Notation (+)**: change this to enabled only if your service provider required the + sign before the country code.

4. Click **Submit**.

> **CAUTION:**
> If the **Register Expiry Time** and **Keep-Alive Time** parameters are not set as specified by your service provider you will experience service outages.

To edit service provider configurations

1. Log in to the **System Options** web page.
2. Select **Service Provider** on the **System Options** menu.
3. Click a domain name to access the associated service provider configuration settings.
4. Click **Change Details** to edit the information. See above for field descriptions.
5. Click **Submit**.

To delete a service provider configuration

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

To assign a SIP identity

When your 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 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. Select **Service Provider** on the **System Options** menu.
3. In the Domain column, click the name of the service provider configuration.
4. Click **Identities**. When SIP identities have been assigned, a list of SIP identities is displayed, along with the extension numbers of the telephones that will handle incoming and outgoing calls.
5. Click **Add Identity**. The **Add Configuration Identity** window is displayed.

- In the **Identity** field, type a logical user name for the identity.
- In the **Authorized User** field, type the number that your service provider assigned.
- In the **Password** field, type the password that is associated with the user name.
- In the **Incoming Extension** list, select the telephone extension or group extension that will handle incoming calls for the specified identity. If you want the auto attendant feature to answer incoming calls instead, select **Global**.
- In 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 want a receptionist or the auto attendant feature to answer incoming external calls, select **Global**.
- Select the **Register** check box when each set is registering, i.e. no DID. If you are configuring PBX mode, register the Primary Identity (PID) and clear the box for the Virtual Identities (VID).
- If you want to use a custom auto attendant configuration instead of the default configuration, select the name of the custom configuration in the **AA Script** list.

6. Click **Submit**.

To edit a SIP identity assignment

1. Log in to the **System Options** web page.
2. Select **Service Provider** on the **System Options** menu.
3. Click **Identities**.
4. In the **Identity** column, click a user name to access the associated identity assignment.
5. Click **Change Details**.
6. Click **Submit**.

To delete a SIP identity

1. Log in to the **System Options** web page.
2. Select **Service Provider** on the **System Options** menu.

   The **Configurations** list is displayed.

3. Click **Identities**.

   The **Identities** window is displayed.

4. In the **Identity** column, click the user name of the identity that you want to delete.

   The **View Identity Details** window is displayed.

5. Click **Delete Identity**. The **Delete Configuration Identity** window 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, the number (alpha characters are not supported) including a prefix. For example, a user in North America 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 (To add a service provider configuration on page 87).
2. Assign SIP identities. The assignments will depend on how your office is set up:
   - Case A: SIP Trunking using the IP PBX Model on page 91
   - Case B: External Incoming Calls are Answered by a Receptionist on page 92
   - Case C: Everyone has a Direct Line on page 94
   - Case D: Incoming Calls are Routed to a Group on page 95
   - Case E: Incoming Calls are Handled by the Auto Attendant Feature on page 96.

Case A: SIP Trunking using the IP PBX Model

Some SIP service providers assign their customers a main SIP identity, the Primary Identity (PID), and assign multiple line appearances (called DID's) to that identity. The additional line appearances for this account, Virtual Identity (VIDs), have an implied registration when the PID registers the main account with the SIP service provider. Incoming calls are then proxied through the PID Quick Edition set before being routed to the correct QE extension. This emulates a single central IP PBX. This case explains how to configure a Quick Edition network to interoperate with a SIP service provider that requires a single device to handle all registrations and complete call routing.

IP PBX Model Functionality

Because Quick Edition employs peer to peer technology and does not have a single central server (single IP) to register all of the SIP accounts, it utilizes the phone that has the PID registered as a proxy for all subsequent VID incoming calls. To the service provider it appears that there is a single device registering and handling all SIP registrations and invites (much like a VoIP PBX) while maintaining the reliability and redundancy of a distributed peer to peer network.

When an incoming call comes into a VID, the SIP service provider sends the SIP invite to the registered device (PID) which then routes the call to the correct Quick Edition extension.

Requirements:

- Network consisting of one or more QE phones running firmware version 3.2.5 or higher
- Service provider accounts that require the IP PBX SIP trunking model. (Check with your SIP service provider to see if this is required.)

To configure SIP trunking using the IP PBX model

1. Configure domain information; refer to To add a service provider configuration on page 87.
2. Provision Identities for your new service provider entry; refer to To assign a SIP identity on page 88.
   – Click the **Register** check box when you add the primary identity.
   – For all subsequent virtual identities, click to clear the **Register** check box.

Note:
The Primary Identity (PID) is set to register and each Virtual Identity (VID) is set to not register (proxy).

Case B: 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 88).

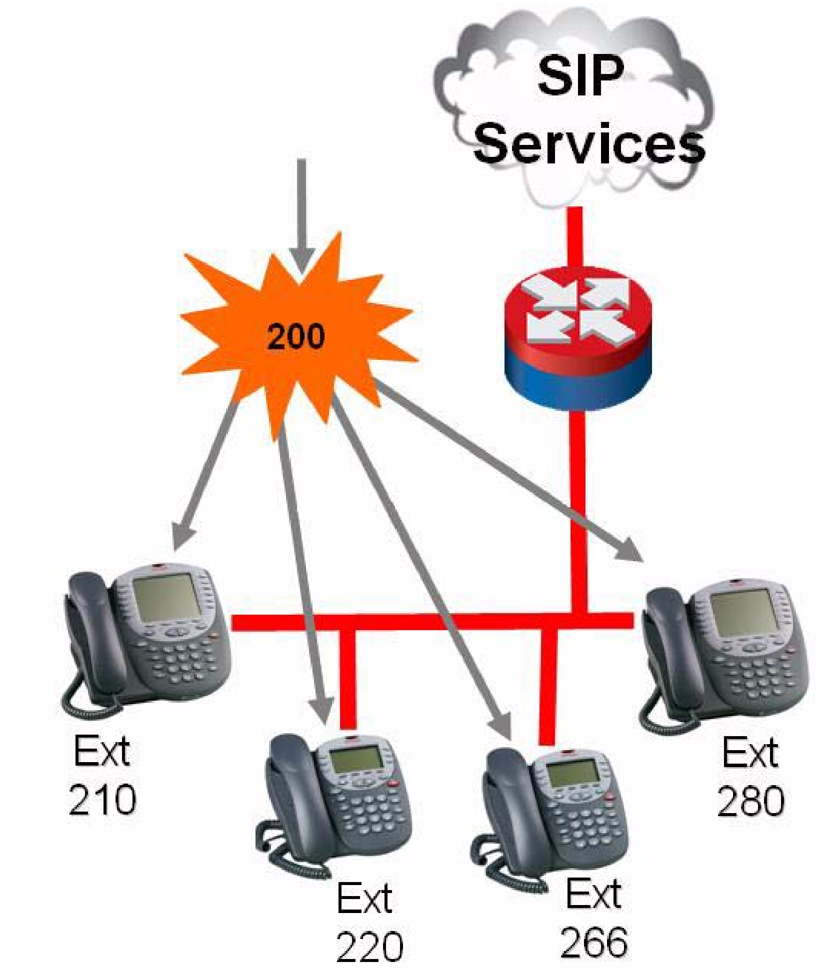
To enable all telephones connected to the network to make outgoing calls, assign the same SIP identity to those telephones. In summary, you would specify information similar to the following values when you create the SIP identity, where extension number 200 belongs to the receptionist, and the **Global** setting refers to all of the telephones connected to the network:

SIP Identity:	1715557600
Password:	12700
Member (Incoming Calls):	200
Member (Outgoing Calls):	Global

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).

Figure 11: Incoming calls are answered by the receptionist



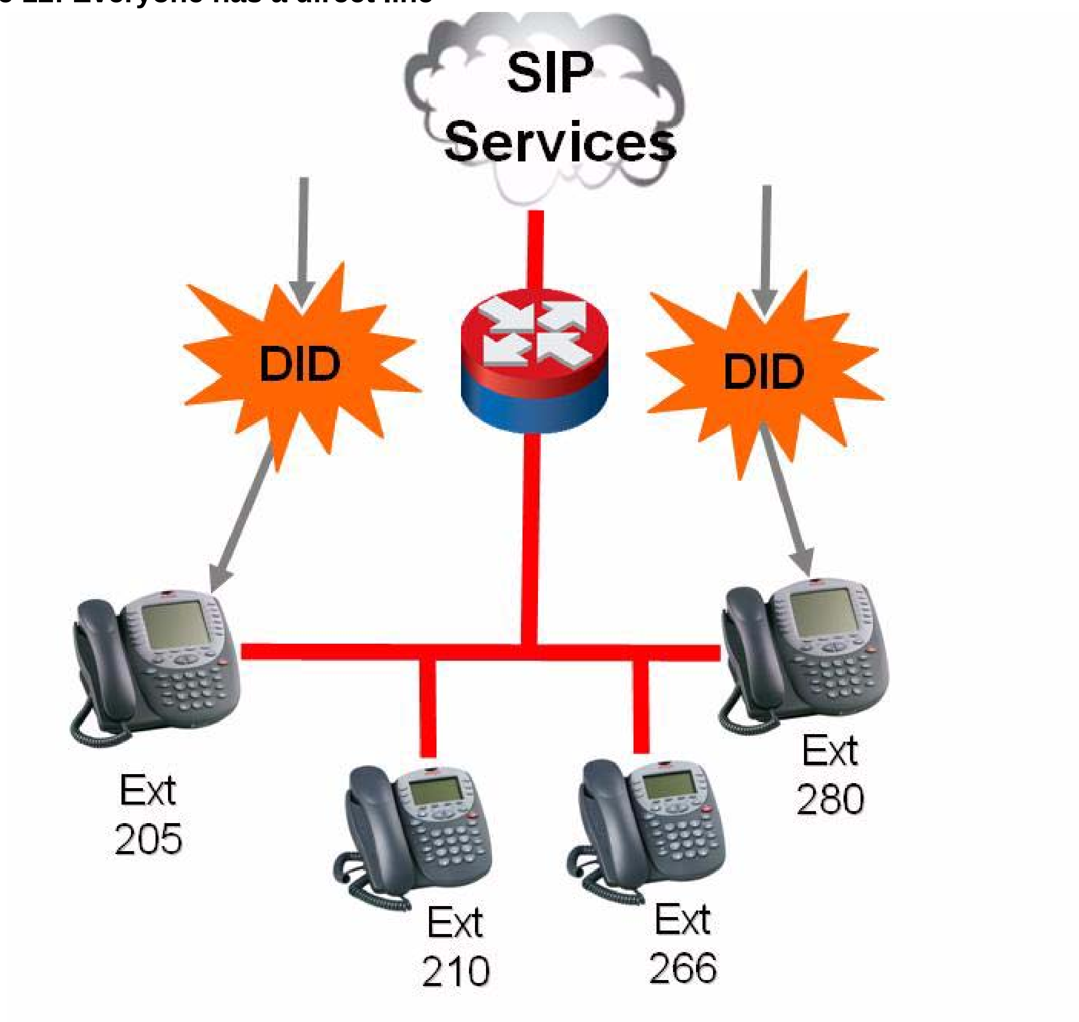
Case C: Everyone has a Direct Line

When everyone has a direct line, assign a unique SIP identity to each telephone. To configure SIP interoperability, refer to the example below:

SIP identity:	1715557654	1715557655	1715557656
Password:	12000	12001	12002
Member (Incoming Calls):	201	202	203
Member (Outgoing 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 incoming and outgoing calls. You can specify how often telephones register with the SIP registration server.

Figure 12: Everyone has a direct line



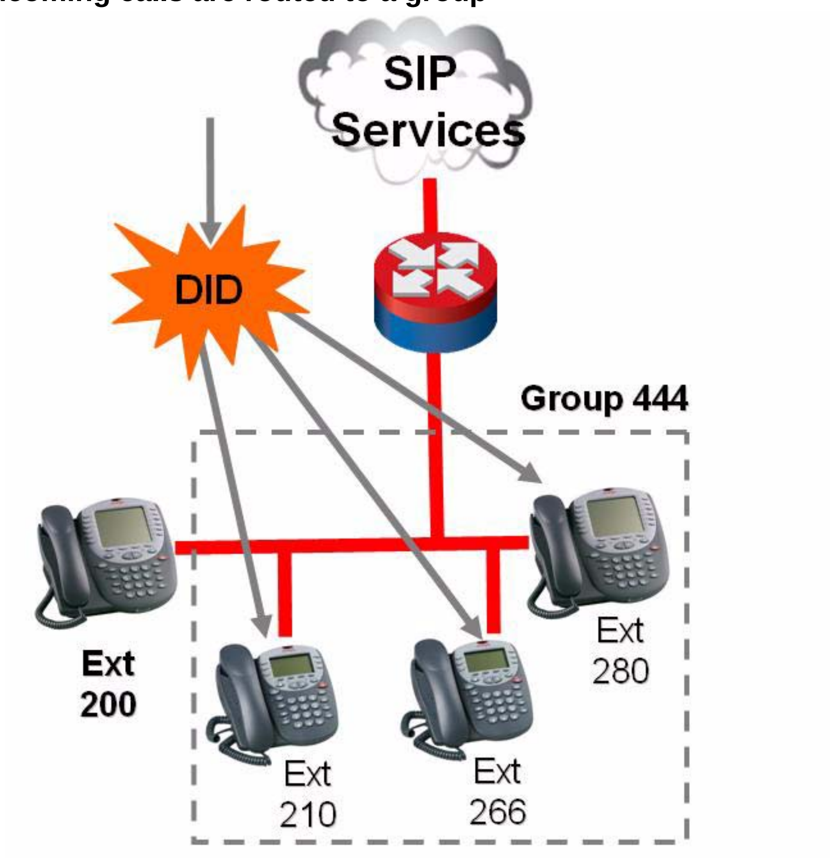
Case D: Incoming Calls are Routed to a Group

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 Create a Group on page 30). Next, configure one SIP identity that corresponds to the group extension number (see To assign a SIP identity on page 88). 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 (Incoming Calls):	405
Member (Outgoing Calls):	405

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).

Figure 13: Incoming calls are routed to a group



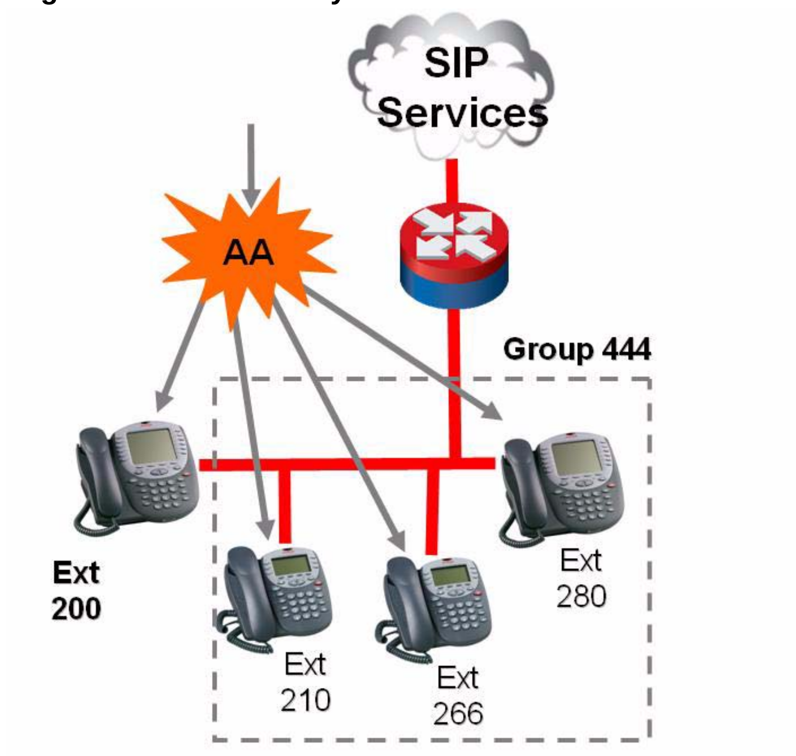
Case E: Incoming Calls are Handled by the Auto Attendant Feature

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 Quick Edition network in general (see To assign a SIP identity on page 88). To enable all telephones connected to the network to make outgoing calls, assign the same SIP identity to those telephones. **Global** lets you select all of the telephones on the network. 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 (Incoming Calls):	Global
Member (Outgoing Calls):	Global

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

Figure 14: Incoming calls are handled by an auto attendant



Chapter 6: SIP Proxy

The SIP Proxy option allows you to add the values that are used to configure SIP devices, A10 Analog Telephone Adapter, G20 BRI ISDN Gateway, softphone, and WiFi on the 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.

To edit a SIP Proxy configuration

1. Log in to the **System Options** web page.
2. Select **SIP Proxy** on the **System Options** menu.
3. Click the **Domain Name** that you want to modify.
4. Make the changes to the **Realm** and **Expiry** time fields and 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. Select **SIP Proxy** on the **System Options** menu.
3. Click **Identities** and then click **Add**.
4. 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 Quick Edition network to an A10 port.
     - **G20:** Assign identities (typically a telephone number) from your service provider to devices on your network. The 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:
  - **Primary Registerer**. Select **Create As Registerer** to make the current identity number the 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
  - **Incoming Extension** number. Select the auto attendant, telephone, group extension, or SIP identity that will handle incoming calls for this identity. Choose **Global** if a receptionist or the auto attendant answers all incoming external calls.
  - **Outgoing Extension** number. Select the extension or the SIP identity that can make outgoing calls using this identity. Select **Unassigned** to configure the trunk for incoming calls only.

5. Click **Submit**.

To edit a SIP Proxy identity

1. Log in to the **System Options** web page.
2. Select **SIP Proxy** on the **System Options** menu, and then click **Identities**.
3. Select the Identity that you want to modify, **Subscriber** (A10) or **Trunk** (G20).
4. Make the changes and then click **Submit**.

To remove a SIP Proxy identity

1. Log in to the **System Options** web page.
2. Select **SIP Proxy**, click **Identities**, and then click **Delete**.
3. Select the identity to be deleted, from the list and then click **Submit**.

SIP Proxy Authorized Users

A SIP authorized user account, added to a proxy domain, is used for authenticating SIP-based communications between the A10 or G20 and the Quick Edition telephones. The admin authorized user is there by default; you can choose to add another authorized user.

To edit a SIP Proxy Authorized User account

1. Log in to the **System Options** web page.
2. Select **SIP Proxy** on the **System Options** menu.
3. Click **Authorized Users**.
4. Select the authorized user account that you want to modify.
5. Make the changes and then click **Submit**.

Appendix A: Web-based User Options

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

Logging In

To display the IP address of the telephone, press # and PAGE LEFT or PAGE RIGHT.

To access telephone and user options using a web browser

1. In the browser **Address** field, enter the IP address of a telephone (for example, **`https://192.168.0.2`**). If you have not disabled it, a Security Alert message is displayed.
2. Click **OK**. If you have not installed the self-signed security certificate on your computer, a Security Alert action request is displayed.
3. To proceed without installing the security certificate, click **Yes**, or to install the certificate, click **View Certificate**, click **Install Certificate** and follow the on-screen instructions.
4. Type the telephone extension number and password on the **Login** window.
5. Click **Login**. The **User Options Home** page for the telephone is displayed.

Logging Off

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

Changing the Password

1. Click **Change Password** on the **User Options** web page.
2. Enter the current password, the new password, and confirm the new password.
3. Click **Submit**. The same password is used for voicemail and for user options.

Home Page

To modify call forwarding option settings

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

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

- Click **Change** in the **Do Not Disturb (DND)** window.

To create or modify Speed Dial assignments

- In the **Speed Dial** window, click a site to add or modify a speed dial assignment.

To enable or disable Call Pickup Audio and Visual Alert

1. In the **Call Pickup Alert** dialog box, click **Change**.
2. Select or clear the appropriate **Alert** box and then click **Create**.

Caller's Log

- Click **Clear Log** to clear the incoming calls list or dialed calls list.
- Click **Reset Missed Call Counter** to clear the counter.
- Click **Clear All Logs** to reset both lists and the call counter.

Terminal Settings

- Click **Change** in the **Terminal Settings** window to change the name. Enter the last name then first name to ensure that all names are sorted alphabetically by last name.
- For information about special features, see Optional Features on page 57.

Voice Mail

- Click **Change** to enable or disable **Zero Redirect**.
- Click **Change** to enable or disable voice mail notification to email, **SMTP**.

Teleworker

For more information about this feature, see Teleworker on page 65.

- Click **Change** in the **Teleworker Options** window to enable (Teleworker) or disable (Local) teleworker mode and specify the IP address of a teleworker server host.

Backup & Restore User Options

Refer to Telephone Configuration Data on page 62 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: Features

This appendix includes the following tables:

- Table 2: Telephone Options - Main Menu
- Table 3: Telephone Options - Telephone User Option Menu on page 102
- Table 4: Telephone Options - Web Interface Menu on page 102
- Table 5: System Options - Telephone Menu on page 103
- Table 6: System Options - Web Interface Menu on page 104.

Telephone Options

Table 2: 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.
10. Voice Mail Dial	Dials the called party's voice mail.

Table 3: 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.
5. Language	Select a user language in the 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	To connect your telephone to high-speed Internet and access the Corporate directory.
11. Reset Softkeys	Reset softkeys to factory defaults.
12. Call Pickup Alert	Select Audio or Visual Alert to display the status screen for audio/visual pickup alerts. Displays the status screen for audio pickup alerts.

Table 4: 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 `https://192.168.0.2`).		

System Options

Table 5: 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. Music on Hold	Edit the audio tag name or enable or disable the status.		
9. Auto Attendant	**1. Auto Attendant**	View extension number, name, selected greeting (prompt), prompt language, and extension number of associated PSTN gateway. Create an extension and edit a name.	
	2. Custom Greeting 1	Record, play, and/or save a custom greeting.	
	3. Custom Greeting 2	Record, play, and/or save a custom greeting.	
10. Call Pickup Groups	**Add or delete a pickup group; add extensions to the pickup group.**		

Table 6: 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 in the entry type 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 6: 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 window 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 in the 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.

Appendix C: Call Detail Recording

Call Detail Recording Fields

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.

Table 7: 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.
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; 10 - PSTN.
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.
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).

Table 7: Call Detail Record Fields

Recorded data	Description
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 the table.

Table 8: CDR example

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; 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:

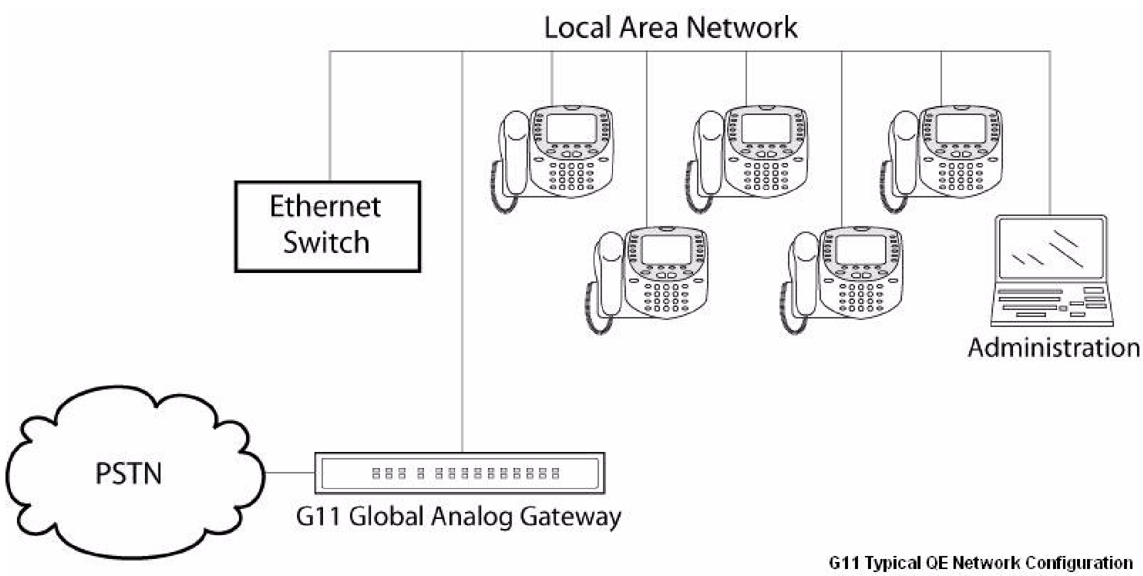
```
00:04:0D:9C:93:42,1,f4801968fa9df098a3f1936fad916af3@135.20.161.73,6093515a350
be2f,e639f466f9ae299,01,03,Adam,200@135.20.161.73,200@135.20.161.73,Bob,202@13
5.20.160.181,202@135.20.160.181,01,2007/01/03 10:23:24,2007/01/03
10:23:26,0,10,2007/01/03 10:23:43,17,2,,,0,,C,,135.20.161.73:0,,
```

Appendix D: G11 PSTN Gateway, Analog

Introduction

A G11 has four Foreign Exchange Office (FXO) ports that provide your network with access to the Public Switched Telephone Network (PSTN) through PSTN lines managed by your telephone service provider. Figure 15 shows the position of a global analog gateway in a typical company telephone network. Up to ten gateways can be connected to your Quick Edition network if required to increase the number of PSTN connections into your office.

Figure 15: Typical 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 telephones are connected. A computer may be connected to the network to provide web-based configuration.

If your Quick Edition IP telephones are already connected to the LAN and running software version 3.0 or greater, adding a global analog gateway to the system is easy. Simply connect the global analog gateway LAN port to an unused port on the IP router or switch, and connect the global analog gateway FXO ports to the PSTN lines supplied by your service provider. Power is delivered to the global analog gateway through the IETF 802.3a Power over Ethernet (O0E) enabled router or switch, and the global analog gateway configures itself automatically.

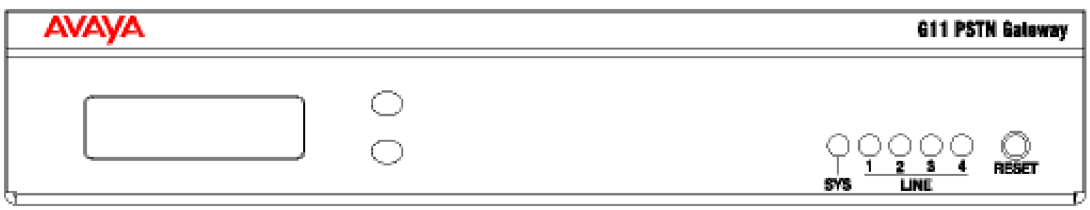
Hardware Features

This section describes the hardware features of the global analog gateway.

Front Panel

See Figure 16 for a diagram of the global analog gateway front panel.

Figure 16: Global Analog Gateway Front Panel



LCD and Control Buttons

You can use the control buttons and LCD to navigate the gateway and display information about the gateway. The function of the control buttons varies according to the content of each LCD screen display.

After a successful startup, the LCD displays the following information:

- Date and time.
- IP address of the gateway.
- Extension number of the gateway.
- Menu.

No Peers alarm will appear when there are no networks for the gateway to join.

To display the gateway details

1. After a successful startup, select **Menu**.
2. Select **Details**.
3. Follow the LCD prompts.

   The global analog gateway information such as the extension number, IP address, MAC address, software version, site name, and site ID display.

To display the gateway system configuration settings

1. After a successful startup, select **Menu**.
2. Select **Configuration**.
3. Follow the LCD prompts.

   The region, language, and music on hold information display.

To set the gateway to factory settings

Reset a global analog gateway to factory settings when you want to move the global analog gateway to a different Quick Edition network.

This procedure assumes that no telephone cords (PSTN lines) are connected to the FXO ports on the rear panel of the global analog gateway.

> **Note:**
> Before you begin, verify that the Power LED on the global analog gateway is green. If the Power LED is red, disconnect the supply of power to the global analog gateway, and then reconnect the supply of power.

1. After a successful startup, do one of the following:
   - Select **Menu > Options > Freshstart**.
   - Press and hold the **Reset** button until **Begin?** appears on the LCD.
2. Follow the LCD prompts.
3. When “Freshstart Done” appears, disconnect the power supply to the gateway.

LEDs

There are five LEDs on the front panel of the global analog gateway. The left-most LED is the System LED. It turns red when the global analog gateway is powered on, and changes to green when the global analog gateway is ready to service calls to and from the PSTN.

The Line 1, Line 2, Line 3, and Line 4 line LEDs reflect the status of PSTN lines:

- When a line LED is off, no PSTN line is connected.
- When a line LED is green, the connected PSTN line is available but not in use.
- When a line LED flashes green, the connected PSTN line is in use.

Reset Button

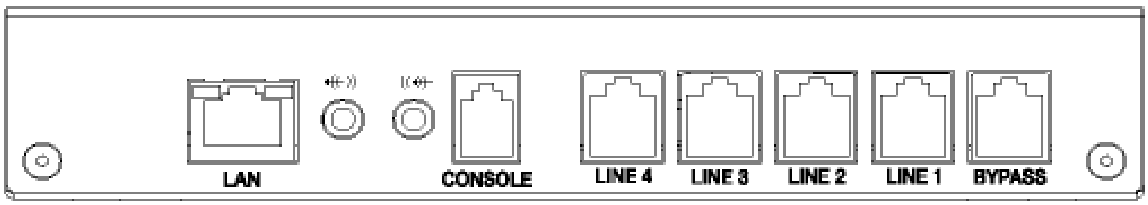
Use the Reset button to:

- restart the global analog gateway by pressing the button once.
- reset the global analog gateway to factory settings (see To set the gateway to factory settings on page 111).

Rear Panel

See Figure 17 for a diagram of the global analog gateway rear panel.

Figure 17: Global Analog Gateway Rear Panel



Power Option

The global analog gateway can operate from an 802.3af PoE LAN.

LAN Port

The global analog gateway has one 10/100 Base-T Ethernet connector: an RJ45 LAN port. Using a customer-supplied CAT 5 (or better) Ethernet cable, you connect the LAN port to an unused port on an IP router or switch that is connected to your Ethernet LAN.

The LAN port has two LEDs: a green link-and-activity LED (right of the port) and an amber speed LED (left of the port). The link-and-activity LED indicates a link when solid green, and an activity when flashing green. The speed LED indicates 10 Base-T when off, and 100 Base-T when solid yellow.

Page

The global analog gateway provides a Paging 3.5 mm stereo mini-jack that you can connect to a customer-supplied audio amplifier with speakers for making announcements. The output is monophonic (single-channel). The External Paging jack provides a typical output of 500mV$_{rms}$ across a 600 ohms load.

Music on Hold

The global analog gateway has a Music on Hold audio input stereo mini-jack. The jack accommodates standard 3.5 mm connectors. Set the volume at the audio source to a comfortable level; adjust the volume to produce an audible level at the handset. The music jack supports a maximum input level of 25mV$_{rms}$ to 1V$_{rms}$ across an input impedance of 1 kohms.

Console Port

This is a dedicated port used by Avaya technicians for troubleshooting purpose.

FXO Ports

The four FXO ports (Line 1, Line 2, Line 3, Line 4) are designed to receive analog telecommunications lines from a Central Office (CO) of the PSTN. Your service provider delivers one or more PSTN lines to your place of business through telephone jacks on the wall, or a patch panel in a telephone closet.

You can connect the global analog gateway to a telephone jack using standard 4-wire modular line cords (telephone cords) equipped with a RJ-11 connector at one end and a country-specific connector at the other end. Plug the RJ-11 connector into port Line 1 of the global analog gateway, and plug the country-specific connector into the telephone jack.

The global analog gateway works with loop-start PSTN lines. The term “loop start” describes the way the system applies analog signaling to obtain a dial tone. When you arrange to lease services from your local telephone company, order loop-start lines.

Each PSTN line from your service provider supports a single conversation at a time. To allow more than one user on the Quick Edition network to place or receive calls to or from the PSTN simultaneously, you will need to order more than one PSTN line from your service provider. You can connect a maximum of four PSTN lines to a single global analog gateway.

A PSTN line must always be connected to port Line 1 using a telephone cord to support the analog telephone bypass feature (see Analog Telephone Bypass Port on page 113). You may connect additional PSTN lines to ports Line 2, Line 3, and Line 4 using customer-supplied telephone cords.

Analog Telephone Bypass Port

You will need a Plain Old Telephone Service (POTS) analog telephone to connect to the analog telephone bypass port on the global analog gateway. The analog telephone bypass port is a type of power fail transfer port.

To maintain an emergency connection to the PSTN at all times, plug the analog telephone into the analog telephone bypass port on the power fail-transfer port and keep a PSTN line (via a modular line cord) connected to port Line 1 on the rear panel of the power fail-transfer port. In the event of a power failure, the analog telephone can then be used to place and receive calls to and from the PSTN.

Normally (when the power fail-transfer port has power), the bypass port is disabled—there is no connection to the PSTN, so a dial tone cannot be heard through the analog telephone handset.

When no power is being delivered to the power fail-transfer port, an internal connection between the Line 1 port and the bypass port is closed automatically, which causes a connection to the PSTN through the bypass port. When the bypass port has a connection to the PSTN, you can hear a dial tone through the analog telephone handset, and you can place and receive calls using the analog telephone. No access code is required for placing calls.

Technical Specifications

Table 9 provides a summary of the technical specifications associated with G11 PSTN gateway. The G11 is Restriction of Hazardous Substances (RoHS) Directive compliant.

Table 9: Technical Specifications

FXO Ports	Four analog ports configured for loop-start operation; RJ-11 connectors
Power Fail Transfer Port	One analog telephone bypass port; RJ-11 connector
Network Port	10/100 Base-T; RJ-45 connector
Audio Input	Supports a maximum input level of 25mV$_{rms}$ to 1 V$_{rms}$ across an input impedance of 1 kohms; standard 3.5 mm stereo mini-jack connector
Audio Output	Provides a typical output of 500mV$_{rms}$ across a 600 ohms load; PCM data input to -3dB; standard 3.5 mm stereo mini-jack connector
CODEC Options	G.711 PCM default for live conversation; G.729a (Annex B) can be negotiated if required to support calls initiated at the far end. G.729a (Annex B) is used for voicemail storage.
Power Options	IETF 802.3af Power over Ethernet
Certifications	UL/cUL UL60950-1: 2003, First Edition CSA C22.2 No. 60950-1-03 1st Ed. April 1, 2003 FCC Part 15, Subpart B Class B, and FCC Part 68 ANSI C63.4:2003
Operating Temperature/ Humidity	32° F to 104° F (0° C to 40° C) 10% to 95% non condensing
Storage Temperature/ Humidity	14° F to 104° F (-10° C to 40° C) 5% to 90% non condensing
MAC Address	See the label affixed to the bottom of the device.

Table 10: Ringing Characteristics

Ringer Detect Threshold	13.5Vrms Min - 16.5Vrms Guaranteed (configurable)
Ring Assertion Time	10Hz -> 83.3Hz (configurable)
Ring Confirmation Count	256ms (configurable)

Table 11: DTMF

DTMF Detection Range	-30dB to 2dB
DTMF Generation	Country specific
Frequency Deviation	1% or less
DTMF Twist / Tone Duration / Inter digit pause	Country specific

Caller ID

- Both Type I and Type II supported
- Telcordia (North America)
- ETSI (Europe)
- BT (UK)
- NTT (Japan)

Table 12: Other

Echo Cancellation	64ms G.168 echo cancellation
Silence Detection	-50dB (configurable)
Software Input / Output Level control	-35dB to +32dB (-15 to +12dB dynamic)
Min battery voltage required	24V

Table 13: Line Impedance

North America	350R+(1000R//210nF)
United Kingdom	320R+(1050R//230nF)
Italy	270R+(750R//150nF) TBR21
Australia	220R+(820R//120nF)

Table 13: Line Impedance (continued)

Singapore	350R+(1000R//210nF)
Thailand	350R+(1000R//210nF)
Hong Kong	350R+(1000R//210nF)
South Africa	220R+(820R//115nF)
India	350R+(1000R//210nF)
Russia	350R+(1000R//210nF)
Mexico	350R+(1000R//210nF)
New Zealand	370R+(620R//310nF)
Switzerland	270R+(750R//150nF) TBR21
Austria	270R+(750R//150nF) TBR21
Germany	220R+(820R//115nF)
France	270R+(750R//150nF) TBR21
Spain	270R+(750R//150nF) TBR21
Netherlands	270R+(750R//150nF) TBR21
Belgium	270R+(750R//150nF) TBR21
Argentina	350R+(1000R//210nF)
Japan	350R+(1000R//210nF)
Norway	270R+(750R//150nF) TBR21

Appendix E: A10 Analog Telephone Adapter

Introduction

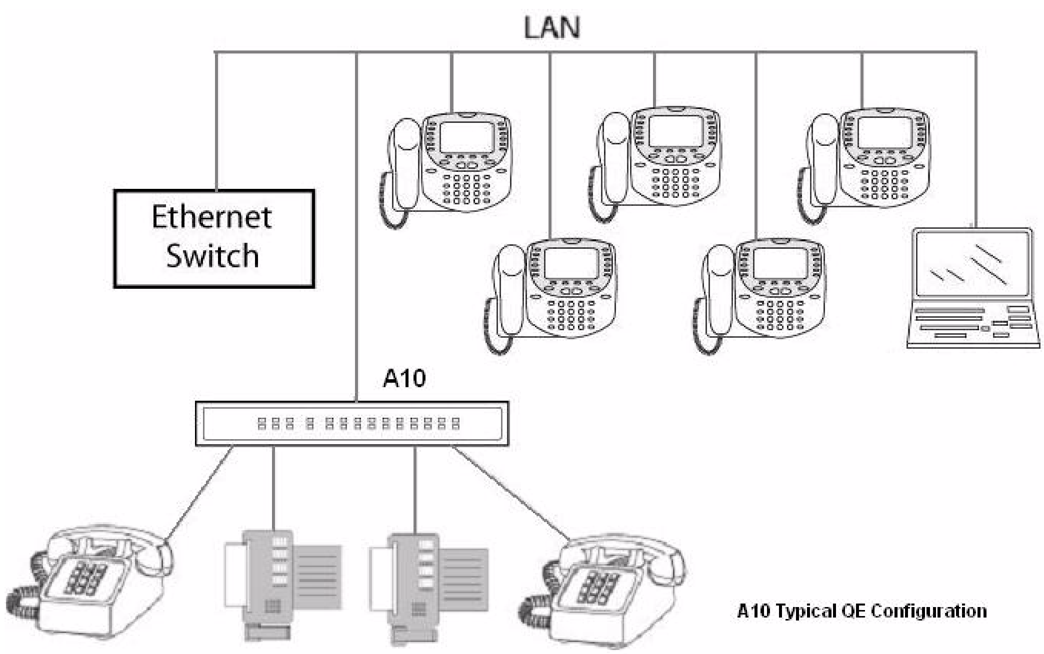
The A10 Analog Telephone Adapter combines IP routing, VPN/Security, and Quality of Service (QoS) for voice and FAX calls over any IP or PSTN network.

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 protocols.
- Internet access and IP routing with IP QoS support for voice and data traffic.

Up to five adapters can be connected to your Quick Edition network to increase the number of PSTN connections into your office.

Figure 18: Typical 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 also be connected to the network.

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

Hardware Features

Front Panel

Figure 19: A10 Analog Telephone Adapter Front Panel

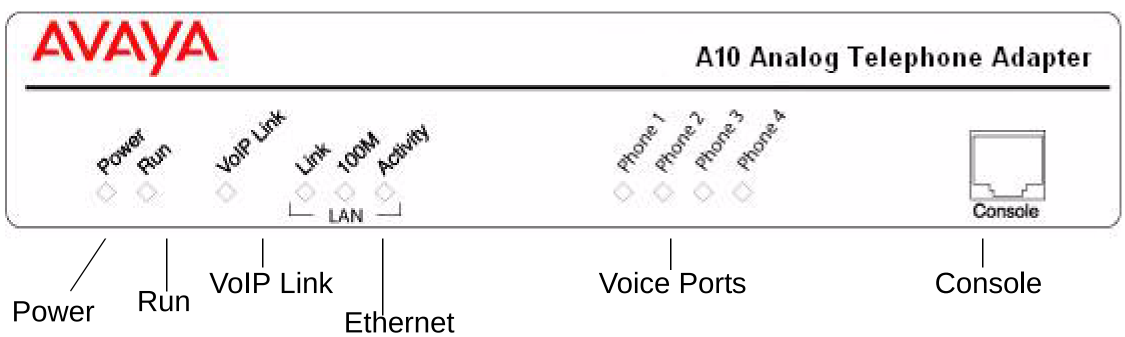


Table 14: Front panel descriptions

Power LED	When lit, indicates power is applied. Off indicates no power is applied.
Run LED	When lit, indicates normal operation. Flashes 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.
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.
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 20: A10 Analog Telephone Adapter Rear Panel

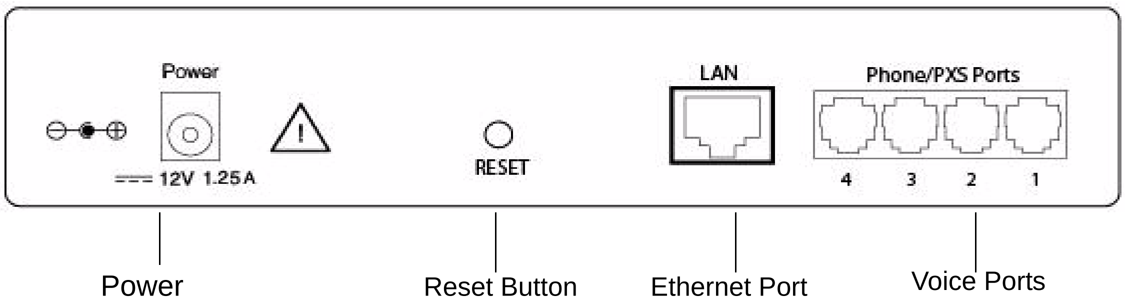


Table 15: Back panel descriptions

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 RESET button. ● 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)—With the unit powered off, press and hold the RESET button as you apply power. 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 16: 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
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 17: 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.

Voice processing (signalling dependent)	• Voice codes:
	  • G.711 A-Law/µ-Law (64 kbps)
	  • G.726 (ADPCM 40, 32, 24, 16 kpbs)
	  • 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 routing—session router	• Local switching; Interface hunt groups
	• Routing Criteria
	  • Interface
	  • Calling/called party number
	  • Time of day, day of week, date
	• Number manipulation functions
	  • Replace numbers; Add/remove digits
	  • Multiple remote gateways; PLAR

Table 17: 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 • 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.

Call holding

To place a call on hold from an analog telephone, do one of the following

- If the analog telephone set has a hold button, press it.

- If the set has only a link/flash button, then press the button once to put the call on hold. You will have 6 seconds to dial another party, and then you will hear 3 consecutive beeps every four seconds to remind you that someone is on hold.
- If the set does not have a hold or link/flash button, press the hook flash once to put the call on hold. You will have 6 seconds to dial another party, and then you will hear 3 consecutive beeps every four seconds to remind you that someone is on hold.

Note:
The caller placed on hold will not hear music on hold.

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 or external number

- Press flash-hook and dial the number for any analog or IP phone extension in the system or to any external telephone number.

Call Forward

Call forwarding is defined through the Group Forwarding rules (see page 31 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). Do not use this feature 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 1 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 call

The analog phone can dial any telephone number or extension that can be dialed by a QE set.

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

To make a second call

1. Press flash-hook to put your current call on-hold and give you dial tone for the second call.
2. Dial your second call after hearing the dial tone.

Table 18: 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.

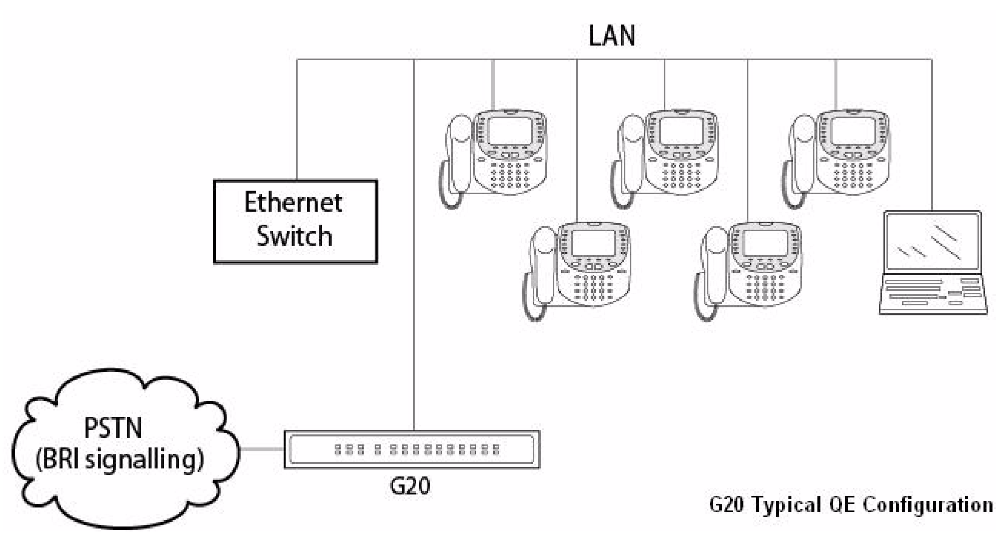
Appendix F: G20 ISDN BRI Gateway

Introduction

The G20 ISDN BRI Gateway is a compact VoIP gateway router that supports two VoIP calls on each of two ISDN BRI ports. This gateway allows your 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 21 shows the position of the gateway in a typical company telephone network. Up to five gateways can be connected to your Quick Edition network if required to increase the number of PSTN connections into your office.

Figure 21: Typical 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 22: G20 ISDN BRI Gateway Front Panel

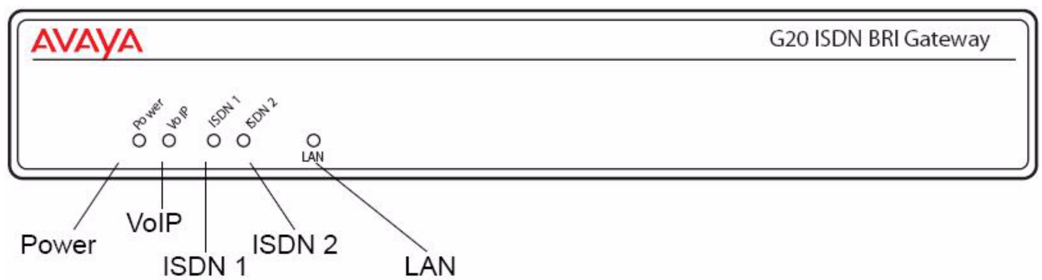


Table 19: G20 ISDN BRI Gateway front panel LED indications

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 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 23: G20 ISDN BRI Gateway Rear Panel

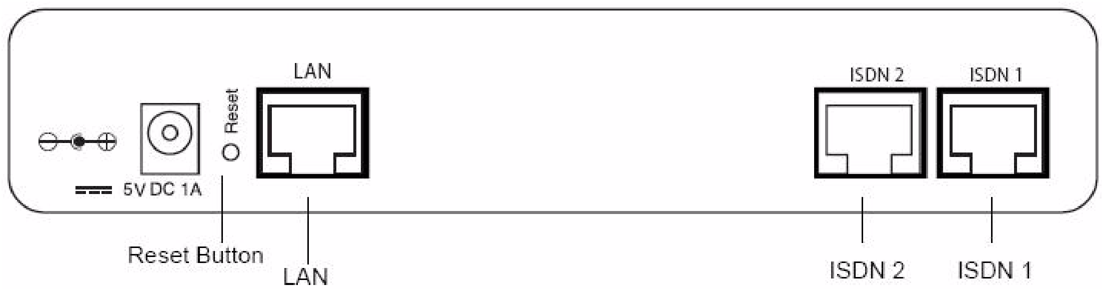


Table 20: G20 ISDN BRI Gateway back panel descriptions

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.
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 21: 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).
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 22: 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.

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 • SIP call transfer, redirect • Overlap or en-bloc dialing • DTMF in-band, out-of-band • Configurable progress tones

Table 22: Technical Specifications

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

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 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

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