Avaya™ Quick Edition

Release 4.0
Troubleshooting Guide

16-602443
Release 4.0
April 2008
Issue 3

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

Chapter 1: Logging and Performance

Contacting Technical Support

If you are unable to resolve the problem using the troubleshooting guidelines provided in this chapter, contact your technical support representative. If required, you may contact Avaya Technical Support by using one of the methods available through the Avaya Technical Support web site: http://support.avaya.com/QuickEdition. If you have an Avaya support plan, call the number associated with that plan to obtain service according to the terms of your agreement.

Logging Options

The event logging and reporting feature will capture and report event log data at the 1600-series telephone level. There are five reporting levels, in order of severity: info, minor, major, fatal, and support. Log entries are preserved through a reboot but are not part of a backup.

Each log file will have a unique name that includes time/date and 1616 phone MAC/DN.

The following event alarms will be displayed on the idle screen:

- SIP SP identity registration failure
- Gateway down
- SIP Proxy Registration.

Table 1: Logging Events and Levels

Event	Info	Minor	Major	Fatal	Support
Peer up and down					✔
Denial of service notification					✔
Duplicate IP address					✔
Admin user logging in					✔
Log in failure					✔
Auto attendant hosting		✔			
Teleworker hosting		✔			

Table 1: Logging Events and Levels

Button presses	✔				
IGMP join/query		✔			
CDR buffer overflow		✔			
Backups change		✔			
SIP datagram		✔			
TTI/GAG line up, line down		✔			
Call status		✔			

To set the logging levels

1. Press the MENU button to the left of the keypad.
2. Press the **System** softkey.
3. Type the system administration password.
4. Navigate to **Logging Options** and press OK.
5. Navigate to level that you want to configure; Log To Console, Fatal, Major, Minor, Support.
6. Press **Save** and navigate to the selected **Logging Mask** and press **Save**.

To specify data to be included in the log

1. Press the MENU button to the left of the keypad.
2. Press the **System** softkey.
3. Type the system administration password.
4. Navigate to **Logging Options** and press OK.
5. Navigate to level that you want to configure; Log To Console, Fatal, Major, Minor, Support.
6. Press **Save** and navigate to the selected **Logging Mask**
7. Press OK to select (✔) or deselect (✖) each logging mask.
8. Press **Save**.

Performance Metrics

The following 1600-series telephone performance measurements can be viewed on the screen:

- Packet loss: faulty transmit and receive packets

- Uptime:
- Memory status, RAM and CPU
- SIP Registration Status
- Number of SIP identities
- CPU Utilization

To display performance measurements

1. Press the MENU button to the left of the keypad.
2. Press the **System** softkey.
3. Type the system administration password.
4. Navigate to Performance Metrics and press OK.
5. Navigate to the metric that you want to view.
6. Press OK.

Syslog

After you collect the requested logs you will need a syslog server to forward the logs to support, for example: http://www.kiwisyslog.com/kiwi-syslog-daemon-download/

Syslog is a standard for forwarding log messages in an IP network. Syslog messages can be sent via UDP and/or TCP.

To change the syslog settings

1. Press the MENU button to the left of the keypad.
2. Press the **System** softkey.
3. Type the system administration password.
4. Navigate to **Syslog** and press OK.
5. Press **Change** to change the **Protocol**, the **IP address**, and/or the **Port**.

TFTP

To send the syslog to support

1. Press the MENU button to the left of the keypad.
2. Press the **System** softkey.
3. Type the system administration password.
4. Navigate to **TFTP**.
5. Press **Change** to change the **IP address** or **Port**.
6. Press **SendNow** to send the logs.

Chapter 2: Telephones

Avaya Quick Edition IP telephones are relatively trouble free. This chapter provides information and charts that walk you through resolving problems. This guide also includes recommended LAN operating parameters and an introduction to analog trunks.

Troubleshooting Telephones

If you cannot resolve a problem on your own, contact your technical support representative (see Troubleshooting Telephones on page 5).

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

Note:
If it becomes necessary to replace a telephone and you do not have a user data backup, you will need to reprogram the device. You will have replace, for example, speed dials, forwarding, personal directory, the voicemail greeting.

Table 2: Basic Troubleshooting Procedures for Quick Edition IP Telephones

Problem/Symptom	Suggested Solution(s)
Telephone does not become active immediately after a power interruption.	Allow a few minutes for the telephone to initialize if the telephone became unplugged from the LAN or experienced a power interruption.
Telephone does not power on after connecting it to the LAN.	Verify that the telephone is receiving power through the Ethernet LAN connection: ● Verify that the switch or router to which the telephone is connected supports 802.3af PoE. If all other connected Quick Edition IP telephones exhibit the same problem, the switch or router may need to be replaced. ● If the LAN is not 802.3af PoE enabled, it is your responsibility to provide a suitable inline power supply. You can order an Avaya PoE injector to power the telephone. ● Check the Ethernet LAN connection to the telephone to ensure that it is properly connected. If required, swap the cable to determine whether the cable is defective. If you can determine that power is being supplied to the telephone (the display area or any of the indicators on the front panel of the telephone are lit) but the telephone still does not become operational, report the problem to your technical support representative.
Telephone does not display the option to join an existing site when connected to an existing network.	When a telephone is connected to the LAN for the first time, it searches for P2P peers. If the existing P2P devices do not use IP addresses in the same network address space (for example, static IP addresses may have been assigned), a newly connected telephone will not be able to detect peers connected to the same subnet. As a result, the telephone will prompt you to create a new network. For more information, see IP Addresses on page 23. To resolve the problem, see To re-assign a telephone to a different network address space and/or network on page 24.
Speaker does not seem to work.	Press SPEAKER. Use the Volume Control buttons to set the volume to an audible level. If you do not hear a dial tone, contact your sales representative.
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Table 2: Basic Troubleshooting Procedures for Quick Edition IP Telephones (continued)

Problem/Symptom	Suggested Solution(s)
Cannot add an entry to the Speed Dial list.	You can have a maximum of nine Speed Dial button assignments. If all assignments are in use, delete at least one entry before attempting to add a new entry.
REDIAL () button does not dial the last number called.	Check the year in the system date and time setting (refer to the System Administration Guide). If the factory set year is different compared to the current year, the year setting will be incorrect after the Quick Edition IP telephone goes through a power cycle. Calls in the Call Log **Outgoing** list are sorted according to date, and the redial function dials number at the top of the list. If the system date is incorrect, the redial function may not be dialing the most recent outgoing call.
Telephone no longer communicates with peers.	Verify the telephone connection to the LAN. Review the network performance recommendations to ensure that the network adheres to minimum requirements (refer to the System Administration Guide). When previously configured and reconnected to the network, a telephone needs a few minutes to initialize and become operational. During this time, the telephone attempts to communicate with all other Quick Edition devices in the same network address space. Verify the following items by viewing **Set Details** on the **Main** menu: ● Are all Quick Edition devices running the same or compatible software versions? ● Do all devices belong to the same network address space? If required, see To re-assign a telephone to a different network address space and/or network on page 24. ● Do all peers belong to the same Quick Edition site? When you install a Quick Edition IP telephone, you can choose to have the telephone join an existing network or create a new network. Unless the telephone is the first telephone to be installed, you should have the telephone join the existing network. To determine to which network a Quick Edition IP telephone belongs, look up the site identifier, which is displayed in the **Site** field under **Set Details**. All peers making up the same network have the same site identifier and name. If required, see To re-assign a telephone to a different network address space and/or network on page 24 to enable the telephone to communicate with existing peers.
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Table 2: Basic Troubleshooting Procedures for Quick Edition IP Telephones (continued)

Problem/Symptom	Suggested Solution(s)
Dialing hidden identities (e.g. anonymous) by using the incoming call log does not reach the calling party.	When a caller chooses to block Caller ID for privacy reasons, the Quick Edition log will not display the actual name and number. Instead, the log will present, for example, “anonymous” or “unknown”. Because the central office or service provider did not pass the caller ID information, the anonymous and unknown call log entries can not be redialed. To avoid reaching other parties, do not dial a number from the call log list that does not display a complete telephone number string.
No dial tone on a Quick Edition IP telephone.	Verify that the handset is securely connected. Use the Volume Control buttons to set the volume to an audible level.
Cannot dial an external number using any Speed Dial button or the Call Log.	See “Cannot make an external call” below.
Telephone does not ring on an incoming call.	Use the Volume Control buttons set the ringer volume to an audible level. From another telephone, place a call to your extension to test the ringer volume after adjustment.
Information is not displayed in the display area.	See “Telephone does not power on after connecting it to the LAN” above. Check all connections to the telephone to ensure that it is properly connected and receiving power.
The display area appears to be frozen and there is no dial tone.	When a menu is not being accessed, labels for the **SDial**, **Log**, **Dir**, and **Vmail** softkeys are displayed at the bottom of the display area. If a **Back**, **Exit**, or **Done** softkey is displayed instead, select the softkey. Alternatively, press the PHONE/EXIT ( ) button to restore the display area.
Cannot reach the party at a known extension through the Corporate directory.	Verify the dial tone (see “No dial tone on a Quick Edition IP telephone” above). Next, verify that the party you are trying to call has not enabled: ● call forwarding (see “Redirecting Calls through Call Forwarding” in the *Avaya Quick Edition Telephone User Guide*), and/or ● the do-not-disturb feature (see “Enabling/Disabling the Do-Not-Disturb Feature” in the *Avaya Quick Edition Telephone User Guide*). If required, contact your LAN administrator to determine whether there is a problem with the Ethernet LAN.
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Table 2: Basic Troubleshooting Procedures for Quick Edition IP Telephones (continued)

Problem/Symptom	Suggested Solution(s)
Cannot make an external call.	Verify the dial tone (see “No dial tone on a Quick Edition IP telephone” above). If your network includes a PSTN gateway and all PSTN lines are busy, you will hear a low-pitched, rapid (busy) tone repeated 60 times per minute. Wait for a few minutes until a PSTN line becomes free, and then try your call again. To place a call to the PSTN, dial the prefix followed by the number. To call a SIP network, dial the prefix followed by the number. If the problem persists, verify that the PSTN gateway is connected and accessible, and that at least one PSTN line is connected to line port L1 on the rear panel of the gateway. A PSTN line must be connected to port L1 using a customer-provided telephone cord. Before you call your technical support representative, contact your service provider to verify that the PSTN lines are working as expected (for example, loop-start signaling is required).
Audio quality is poor—specifically, you hear: ● sudden silences or gaps in speech, and/or ● clipped or garbled speech.	Network problems are the most likely cause. Contact your LAN administrator and provide a complete description of the problem. If you are the LAN administrator, refer to the System Administration Guide for a list of recommendations. **Tip:** You can specify an audio quality setting for the entire network (refer to the System Administration Guide).
An application or feature does not work as described in the product documentation.	Verify the procedure and retry. If this action does not produce the desired result, the Quick Edition system might need additional programming to enable certain features. Consult the product documentation for additional configuration instructions (refer to the System Administration Guide). If you would like to report a documentation error or omission, contact your technical support representative.
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Table 2: Basic Troubleshooting Procedures for Quick Edition IP Telephones (continued)

Problem/Symptom	Suggested Solution(s)
Teleworker: Teleworker client telephone cannot connect to the remote company network.	Determine the connection status of the teleworker client telephone (refer to the System Administration Guide). Verify that the VPN tunnel is operational. If the IP address of the company LAN has changed recently and the VPN configuration has not been updated, the teleworker client telephone may be unable to connect: ● Update the private IP address of the company office network in the VPN configuration. ● If a host IP address is specified for the teleworker server, change the IP address specification if required. A teleworker client telephone cannot act as a teleworker server. The teleworker server host IP address must belong to a telephone or PSTN gateway on the Quick Edition network.
	No Server message - specify the teleworker server host IP address.
	Not Authorized message - the teleworker server host IP does not recognize the client. Verify the **Add Extension** programming.
Restart Required message	The teleworker server host has uploaded new software to the client telephone. The configuration will take effect when the client telephone is restarted. Press the **OK** softkey to restart.
Connection is successful, but the call was dropped during a conversation.	If someone unplugs a Quick Edition IP telephone that is acting as the teleworker server for the connected teleworker client telephone, the call will be dropped.
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Chapter 3: System

Table 3: Avaya Quick Edition System Problems and Solutions

Problem/Symptom	Suggested Solution(s)
Web-based interface: Cannot access the web-based interface.	Compare the IP address of the telephone and its P2P peers to the IP address of the PC on which the web browser is installed. To determine the IP address of a PC, refer to the System Administration Guide. If the telephone uses a different network address space compared to the address space used by its P2P peers, change the IP address of the telephone to match the network address space used by its peers (refer to the System Administration Guide). If the network address space used by the telephones does not match the network address space used by the PC, ask your LAN administrator to add a route to the Quick Edition network from the PC. Verify that you can access user options and/or system options using the Options ( ) button on the telephone. If you cannot access these options using telephone buttons, the telephone is not working properly. Troubleshoot the problem using the guidelines given in Table 2: Basic Troubleshooting Procedures for Quick Edition IP Telephones on page 6. Check the LAN settings in your web browser. Proxy server settings could be preventing you from accessing the web-based interface. In Internet Explorer, select **Tools > Internet Options**, and then click the **Connections** tab. Click the **LAN Settings** button to view the proxy server settings. Contact your LAN administrator for more information and help if required.
Web-based interface fails after upgrade: Login after upgrading from 3.0 or earlier versions may fail if the interface loads a cached copy of the web page.	In release 3.1, changes were made to the way the web pages are loaded and decompressed to improve performance. To prevent loading a cached copy of a prior release web page, clear the cache. **To clear the cache in Internet Explorer and disable proxy server setting** 1. Start Internet Explorer and click **Internet Options** in the **Tools** list. 2. On the **General** tab, click **Delete Files** in the **Temporary Internet files** section. **Note:** This will **NOT** remove web cookies or browsing history. 3. Click **LAN Settings** in the **Connections** tab to verify that your proxy server is not caching old files. 4. Ensure that **Use a proxy server for your LAN** is **NOT** selected. 5. Click **OK**, close all dialog boxes, and refresh the page. All user and system options will be available. **Note:** If your organization requires the use of a proxy server for external browsing, you may need to re-select **Use a proxy server for your LAN** before being able to access resources outside of your LAN.

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Table 3: Avaya Quick Edition System Problems and Solutions (continued)

Problem/Symptom	Suggested Solution(s)
Multiple call appearances for a single incoming VOIP call. This is a SIP service provider specific issue.	You may need to adjust the registration of the Service Provider SIP Identities. If you are using a service provider that maintains a record of the registered SIP identities even after they expire, dialing the identity might result several line appearances instead of one incoming call. It is recommended to create the SIP identity as virtual rather than primary. Clear the **Register** check box.
System password lost or forgotten	Obtain the Password Reset Key: 1. On a telephone in the system that requires the password reset, - press OPTIONS > Options > System Options. - press the unlabeled soft-key (3rd from the left) between Bksp and Done. 2. The set displays an alpha numeric, case sensitive 19-character key. Record the string to provide to Avaya Support. **Note:** You have only 15 minutes in which to reset your password before the password key changes. Provide the Password Reset key to Avaya Support: 1. Contact Avaya support with the following information at hand: - Firmware version of the devices in the system - Duration since the reset key was gathered - The Reset Key (double check to ensure it is correct) 2. Avaya Support provides you with a password reset string. - The password reset string is 16-digits long - This string is entered as the system password (without spaces) - The login will then be complete and the administrator will be prompted to update the system password
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Table 3: Avaya Quick Edition System Problems and Solutions (continued)

Problem/Symptom	Suggested Solution(s)
System Password is not recoverable	You can reset an entire system to factory default if the system password is not recoverable. This procedure must be performed on all devices to remove the Site ID information. If there is even a single active Site ID on the network, devices will join it and assume the password associated with that Site ID. We strongly recommended that you use the procedure, refer to the System Administration Guide, to reset and remove devices from the system when access to system admin is available. This procedure should only be used after all other options have been explored. **Reset a Quick Edition Telephone** 1. Remove power from the device. 2. Apply power to the device and press and hold the OPTIONS button. The device will reboot. 3. Continue to press and hold the OPTIONS button. 4. The telephone will prompt with the option to reset to factory default. 5. Select the factory default option. The device will indicate when the reset process is complete. 6. Remove power from the device. **Reset a G11 PSTN Gateway when the Power LED is Green** 1. Press and hold the Reset button on the back of the G11 gateway. The power LED turns from Green to Red (approximately 10s). 2. Release the Reset button when L1 turns Green. 3. Wait for L3 and L4 to turn solid Green. 4. Remove power from the device. **Reset a G11 PSTN Gateway when the Power LED is Red** 1. Cycle power on the G11 gateway. 2. Immediately press and hold the Reset button on the back of the G11 gateway. 3. Release the Reset button when L1 turns Green. 4. Wait for L3 and L4 to turn solid Green. 5. Remove power from the device.
Cannot redial from call log	If you cannot use the call log to redial from G20 BRI: Depending on the Called Party OTN provided by your service provider, you may not be able to use the call log or voice mail functionality to return calls. If your service provider uses National or International Called Party TON, and your region does not use 0 or 00, you may be impacted. For example, an incoming number shows the calling party number as 001707118118 and TON national. When redialed, the G20 inserts a 0 which is not needed and so the call fails. See the examples table below.
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Table 3: Avaya Quick Edition System Problems and Solutions (continued)

<table>
<tr><th>Problem/Symptom</th><th>Suggested Solution(s)</th></tr>
<tr><td>ISDN Incoming calls do not connect</td><td>If the Called Party TON (type of number) is National or International, prefix with 0 or 00 respectively. If the Called Party TON is subscriber or unknown, include no prefix.
<table>
<tr><th colspan="4">SIP Proxy Identities dialog</th></tr>
<tr><th>Service Provider Called Party TON</th><th>Prefix</th><th>Identity Entry Example</th><th>Number to Dial</th></tr>
<tr><td>National</td><td>0</td><td>001707118118</td><td>01707118118</td></tr>
<tr><td>International</td><td>00</td><td>0001707118118</td><td>01707118118</td></tr>
<tr><td>Subscriber</td><td>None</td><td>01707118118</td><td>01707118118</td></tr>
<tr><td>Unknown</td><td>None</td><td>01707118118</td><td>01707118118</td></tr>
</table></td></tr>
<tr><td>Auto Attendant:
The Auto Attendant feature does not answer incoming calls, or an incorrect greeting is played.</td><td>Access the Auto Attendant menu under System Options, and verify that the Auto Attendant configuration that you want to use has been created.
If you have a PSTN gateway, verify that an Auto Attendant configuration has been assigned to the gateway (refer to the System Administration Guide). Verify that the greeting you want to use has been recorded and assigned to the associated Auto Attendant configuration (refer to the System Administration Guide). If your Quick Edition system interoperates with a SIP network, you can assign an Auto Attendant configuration to each SIP identity through the web-based administration interface. Ensure that the Auto Attendant feature has been enabled for the identity (refer to the System Administration Guide).</td></tr>
<tr><td>Paging:
Not all of the telephones in a paging zone are receiving announcements.</td><td>If the telephone in question is in use while you make the announcement, the announcement is suppressed on that telephone—this is normal. Verify that the telephone in question has been added to the paging zone (refer to the System Administration Guide). Verify that the telephone in question is on-hook, and that the do-not-disturb feature is disabled (see “Enabling/ Disabling the Do-Not-Disturb Feature” in the Avaya Quick Edition Telephone User Guide).</td></tr>
<tr><td>Voicemail:
Callers cannot leave voicemail because the storage area is full.</td><td>When a Voicemail inbox is full, callers hear a message indicating that Voicemail storage has been filled and no more messages can be saved. To avoid this situation, users are encouraged to delete voicemail messages on a regular basis. A telephone that is a music-on-hold .wav file host has reduced voicemail storage capacity. If necessary, remove the .wav file to another telephone.</td></tr>
<tr><td>Email Notification of Voicemail:
Voicemail messages are not being forwarded to an e-mail account.</td><td>Access the Network Options menu under System Options to verify that the IP address of an SMTP server has been specified.
Using the web-based User Options interface (see the System Administration Guide), select Voice Mail and verify that “To” and “From” addresses have been specified, and that notification status has been enabled.</td></tr>
<tr><td colspan="2" align="right">4 of 6</td></tr>
</table>

Table 3: Avaya Quick Edition System Problems and Solutions (continued)

Problem/Symptom	Suggested Solution(s)
TFTP transfer - upgrade failure: A message indicates that you can't connect through the firewall. TFTP uses UDP; QE is hard-coded to use port 69.	● Temporarily disable the firewall if that is an option. ● Add a firewall rule to allow TFTP transfer: – if you have privileges to add a rule but not to turn off the firewall, – for security reasons, you do not wish to turn off the firewall during upgrade. This solution includes three examples for creating a firewall rule: Microsoft Windows XP (automatic and manual), and McAfee Desktop **Automatically add a firewall rule to Microsoft Windows XP firewall** 1. On the Windows Firewall window, **General** tab, confirm that the **Don't allow exceptions** check box is clear. 2. Run the upgrade wizard. When Windows firewall displays a warning, click **Unblock**. 3. Proceed with the upgrade. **Manually add a firewall rule to Microsoft Windows XP firewall** 1. On the Windows Firewall window, **Exceptions** tab, click **Add Port**. 2. Name the rule (e.g. 69), click the **UDP** traffic type, and then click **OK**. 3. Verify that the rule appears on the exceptions list and enabled. 4. Proceed with the upgrade. **Add a firewall rule to the McAfee Desktop firewall** 1. Open McAfee Desktop Firewall and click **Add** in the **Firewall Policy** tab. 2. Click New Rule. 3. Create the rule: – Description: Give the rule a meaningful name, e.g. TFTP – Action: Permit – Protocol: UDP – Direction: Either – Local and remote service: single - tftp – Address: Any – Options: Active 4. Click **OK** and verify that the new rule appears in the firewall policy list. 5. Proceed with the upgrade.
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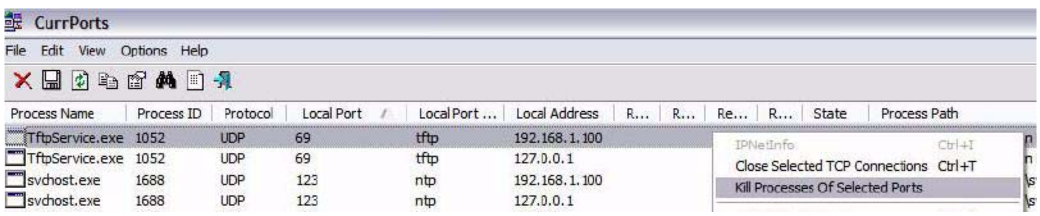
Table 3: Avaya Quick Edition System Problems and Solutions (continued)

Problem/Symptom	Suggested Solution(s)
TFTP server fails to launch: Using the upgrade wizard, this message appears: Failed to create listening socket. The port may be in use by another application.	The Avaya TFTP server will launch but will be offline. The **Server is running** check box will be clear. If there are any other TFTP servers running, close them during the upgrade. If you are unaware of any other servers/applications using port 69, you can use a third party tool such as CurrPorts to find the offending application. CurrPorts downloads are available at: http://www.nirsoft.net/utils/cports.html. For the English version go to: http://www.nirsoft.net/utils/cports.zip. **To locate and stop applications using port 69** 1. Install and then launch CurrPorts to find any applications using port 69. 2. Right-click the offending application and select Kill Processes Of Selected Ports. Repeat for each process using port 69. 3. Continue with the Quick Edition upgrade.



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Chapter 4: G11 Global Analog Gateway

G11 Global Analog Gateway

Table 4: Global Analog Gateway Problems and Solutions

Problem/Symptom	Suggested Solution(s)
Power: The power LED on the global analog gateway is not lit.	When first plugged into the LAN, the Quick Edition IP needs a few minutes to initialize and become operational. During this period, the Power LED will be red. If this is not the case, verify that the global analog gateway is receiving power: ● Verify whether the IP switch or router to which the Quick Edition IP is connected supports 802.3af PoE. If all connected Quick Edition IP Telephones are not receiving power, the switch or router may need to be replaced. ● If applicable, check the PoE LAN connection to the global analog gateway to ensure that it is properly connected. If required, swap your CAT5 Ethernet cable to determine whether the cable is defective. If you can determine that power is being supplied to the global analog gateway but the gateway is not operational, report the problem to your technical support representative.
The power LED on the global analog gateway stays red.	If you connected the global analog gateway to the LAN for the first time without having previously installed Quick Edition IP Telephones, the Power LED will stay red. If the devices on the network are running incompatible software versions, the Power LED will stay red. If Telephones were already installed when you connected the global analog gateway, the Power LED should turn green in two to three minutes. To remedy the situation, reset the global analog gateway to factory settings. Refer to the System Administration Guide. Afterward, disconnect the global analog gateway from the network, install your Quick Edition IP Telephones, and then install the global analog gateway.
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Table 4: Global Analog Gateway Problems and Solutions (continued)

Problem/Symptom	Suggested Solution(s)
The power LED flashes red and green five times followed by a pause.	The global analog gateway is off-line. To remedy the situation, reset the global analog gateway to factory settings. Refer to the System Administration Guide. Afterward, connect the global analog gateway to the network. The Power LED will turn red initially, and then turn green when the global analog gateway is ready.
PSTN Lines: All calls to and from the PSTN have a loud hum or an echo.	The gain setting (loop length) on connected PSTN lines may need to be adjusted. Refer to the System Administration Guide.
No lines are available. You hear a fast-busy tone.	All lines may be in use. When you forward an incoming outside call to a PSTN number (zero-redirect), 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.
The line LED associated with a connected PSTN line is off.	Verify that the associated PSTN line is securely connected to the FXO port through a customer-provided working telephone cord. At the demarcation point of the associated PSTN line, listen for a dial tone from your CO by connecting an analog (POTS) telephone. If there is no dial tone, contact your phone company. If there is a dial tone, the global analog gateway has a hardware problem.
Incoming calls reach the G11, the G11 line indicator blinks, but calls do not reach the telephone. Callers coming into the system hear continuous ring-back but QE phones receive no indication of an incoming call.	In order for the G11 to properly detect incoming ringing to the system, the ringing signal from the provider must have a DC offset. The G11 may expect gain settings for analog loop lengths which are different than those provided from an IAD router. Adjust the following IAD parameters on each port connecting to a G11 FXO port: ● Ring frequency: 30 ● Ring dc-offset: 24-volts ● Idle-voltage: low ● Impedance: 900c
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Table 4: Global Analog Gateway Problems and Solutions (continued)

Problem/Symptom	Suggested Solution(s)
Music on Hold: When a calling party is placed on hold, the Music on Hold feature does not seem to work.	Verify that the feature is enabled. Refer to the System Administration Guide. Ensure that an audio source is plugged into the **Music on Hold** port of the global analog gateway. Verify that the audio source is working by connecting a headset to the source's output and listening to the audio output through the headset. Due to their remote locations, Teleworker client telephones cannot play audio input through the Music on Hold feature to a caller should the remote user put the call on hold. When the remote user puts a call that originates from the company Quick Edition system on hold, the caller hears silence. If the remote user calls a company extension and the user at the company office puts the call on hold, audio input is played to the remote user. For more information about the Teleworker feature, see the *Avaya Quick Edition System Administration Guide*.
The audio input associated with the Music on Hold feature sounds distorted or cannot be heard.	Verify that the audio source is located on a stable platform. If the audio source is shaken or bumped, the audio signal may be disrupted. If you are using a radio as the audio source, verify that the tuner is dialed to a radio station properly. Ensure that the volume control on the audio source is set to an audible volume.
Paging: External paging is not working during an announcement.	Ensure that an amplifier with speaker is plugged into the **Paging** port of the global analog gateway. Select **General Page** on the **Select Paging Zone** menu to make the announcement. See “Broadcasting Announcements through Paging” the *Avaya Quick Edition Telephone User Guide* (Document No. 16-601411).
The audio output associated with an external page cannot be heard.	Select **General Page** on the **Select Paging Zone** menu and make an announcement. See “Broadcasting Announcements through Paging” the *Avaya Quick Edition Telephone User Guide*. During the announcement, adjust the volume control on the customer-supplied amplifier. Verify that the amplifier is receiving power and that a customer-supplied speaker is connected to the amplifier.
The audio output associated with an external page sounds distorted.	Connect the external amplifier with speaker to a different audio source and test the equipment off-line.
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Table 4: Global Analog Gateway Problems and Solutions (continued)

Problem/Symptom	Suggested Solution(s)
Feedback occurs during an external page.	If the Quick Edition IP Telephone that you are using to make the announcement is too close to another Quick Edition IP Telephone (that is, the Telephones are side by side), feedback will occur. Move the Telephones away from each other and try again.
Bypass Port: No dial tone on the telephone that is connected to the global analog gateway.	You can place or receive calls using an analog telephone connected to the analog telephone (bypass) port only when the Quick Edition IP is not receiving power. To place or receive calls using the analog telephone, a PSTN line must be connected to port **Line 1** on the rear panel of the global analog gateway using a customer-provided telephone cord. For more information, refer to the System Administration Guide.
Forwarding fails: Cannot forward a call; fast-busy.	If your network has a PSTN gateway and you forward an incoming outside call to a PSTN number, the call will consume two PSTN lines (one incoming, and one outgoing) on the analog gateway while the call is active.
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Chapter 5: G20 ISDN BRI Gateway or A10 Analog Telephone Adapter

Table 5: G20 and A10 Problems and Solutions

Problem/Symptom	Suggested Solution(s)
Replace defective device: The G20 or A10 must be replaced in the system.	**From the System Options web interface:** 1. Record the IP address of any active telephone on the network. 2. Get the MAC address of the G20 or A10 to be removed: 3. Click the gateway in the **Name** list. 4. Record the MAC address. 5. Record the **Identity** for each port (up to four identities for the A10 and 20 for the G20). 6. Set the **Identity** for each port to **Unassigned**. 7. Click **Submit** and then click **Confirm**.
Replace;	**Telnet to the QE telephone IP address:** 1. At the VxWorks login prompt, type **nimdbg**. 2. At Password, type the system password. 3. At the **->** prompt, type **nx** and press **Enter**. 4. At [nimcat]>, type **misc** and press **Enter**. 5. At [misc]>, type **deletedevice** and the MAC address of the A10/G20 (without separators) and then press **Enter** (for example, type **deletedevice 0050C233505F**).
Replace;	**Install the replacement G20 or A10:** 1. Create the identities that you recorded and removed from the failed device. 2. Associate the identities with the physical port on the new G20 or A10. **Note:** Refer to the *System Administration Guide* for additional details.
Unable to make or receive calls	If you have moved your Quick Edition system from one region to another or if you have changed the region programmed for your system, you will have to recreate your SIP Proxy Trunk and Subscriber Identities with the new telephone or extension numbers specific to the new region.
A call from an A10 analog phone to a QE phone with a SIP identity fails.	When configuring SIP identities for a Service Provider, make sure that you map one SIP identity to the A10 extension (i.e. DID) or create a Global identity to the QE branch.

Table 5: G20 and A10 Problems and Solutions

Problem/Symptom	Suggested Solution(s)
Web-based interface: Cannot access the G20 ISDN BRI Gateway or A10 Analog Telephone Adapter.	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, then select Properties. ● Select **Internet Protocol (TCP/IP)** and click **Properties** (note the current settings). ● 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. Troubleshoot the G20/ A10. Telnet 192.168.123.10, press Enter. `User:`**`nimdbg`**` Password:`**`54321`** `192.168.123.10>`**`enable`** `192.168.123.10>`**`configure`** `192.168.123.10(cfg)#`**`context ip router`** `192.168.123.10<ctx-ip>[router]#`**`interface eth0`** `192.168.123.10<if-ip>[0]#`**`ipaddress <static-ip> <subnet-mask>`** `192.168.123.10<if-ip>[0]#`**`copy running-config statup-config`** `192.168.123.10<if-ip>[eth1]#`**`reload`** `Press 'yes' to restart, 'no' to cancel: `**`yes`** 4. Reset the IP address assigned to your PC to its previous setting. 5. On the **Device Management** page, click the **G20** or **A10**. 6. On the **Edit Gateway Details** screen, clear the **DHCP Address** check box. 7. Enter the static **IP Address** and **Netmask** of the G20 or A10 (for A10, we recommend that you enter the gateway IP address). 8. Click **Submit**.
The G20 fails to come up after a software upgrade.	Power-cycle the gateway using procedure number 1 below. If the gateway still fails to come up, use procedure 2. 1. Restart the unit with the current startup configuration—Disconnect power and then reconnect. 2. 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.

Chapter 6: Music and IP Address

Embedded Music on Hold

Table 6: System Music on Hold Problems and Solutions

Problem/Symptom	Suggested Solution(s)
Music on Hold: Description of device music status.	a. Not supported (default for G20 ISDN BRI Gateway and A10 Analog Telephone Adapter). b. Supported - but no source available (default for a telephone). c. Source available - but not enabled (default for a G11 Global Analog Gateway). d. Enabled - but not providing the MOH. e. Active - providing the MOH. f. Source invalid (format does not match region). g. Teleworker. h. Unavailable - the telephone is down.
Download fails:	There may be insufficient storage on the device. Free some storage by, for example, deleting unneeded voicemail messages. The voicemail storage limit of 20 minutes is reduced by the MOH source file: - audio file of <30 seconds limits voicemail to 15 minutes; - audio file between 30 seconds and 1 minute limits voicemail to 10 minutes; - audio file between 1 minute and 2 minutes limits voicemail to 0 minutes.
Audio file is rejected:	An audio file larger than 2 minutes will be rejected. An audio file smaller than 3 seconds will be rejected.

IP Addresses

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

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

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

Assigning a Static IP Address

If required, you may assign fixed network addresses to telephones or PSTN gateways through network options (refer to the System Administration Guide). Once assigned, a static IP address persists until it is changed manually or another device having the same IP address is detected at boot time.

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

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

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

1. If the IP address of the telephone does not match the IP address space being used by other P2P devices, change the IP address of the telephone to an unused IP address in the required network address space. To determine the IP address of a telephone, press the PAGE RIGHT ( ) button. To change the IP address, refer to the System Administration Guide).
2. Remove the extension number of the telephone from the network (refer to the System Administration Guide), but do not disconnect the telephone from the network. When the telephone initializes, have the telephone join the target network. The procedure does not change the manually assigned (static) IP address.

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

Chapter 7: Data Network

To ensure optimum performance, we recommend the following LAN operating parameters.

Cabling

Supply RJ-45 Category 5 (CAT5) or better (for example, CAT5e) Ethernet cabling to connect equipment such as Quick Edition IP Telephones and G11 PSTN Gateways to the LAN.

Power

Quick Edition Telephones and G11 PSTN Gateways support 802.3af PoE (Power over Ethernet). To connect Quick Edition equipment to a switch that does not support 802.3af, use an Avaya PoE injector to power the telephone.

Table 7: Power Consumption

Device	Device Power Usage	
	Peak	Nominal
4610	6W	4W
4621	6.45W	4.9W
G10	9W	6W
G11	6W	4W
G20	5W	3W
A10	8W	5W

Adding a Zeroconf Route

In some cases without a DHCP server on the network a static Zeroconf route will have to be added to the PC in order to remotely access the Quick Edition system.

1. Open a command line with **start** -> **run**.
2. Type ’**cmd**’ to open a command lane prompt.
3. Type ’**route ADD 169.254.0.0 MASK 255.255.0.0. x.x.x.x**’. The x.x.x.x represents the customer’s real IP address bound to their NIC.

The static route will now be added to the PC to provide network access to the Quick Edition system with Zeroconf addresses.

VLAN and QoS tagging

VLAN tagging or QoS methods are not required. The network element (switch, router) used to connect Quick Edition can be a simple unmanaged device which supports IP multicast traffic. Certain networks will not require a QoS strategy, for example if bandwidth usage is low.

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. You can assign priority levels to Quick Edition voice and data traffic to ensure QoS at OSI Layer 2. Specifying a priority such as 5 for voice traffic and 3 for data ensures that voice traffic has priority over data. Your LAN administrator must determine the required settings. If your network has VLANs, include all Quick Edition devices in the same VLAN.

If you experience performance degradation with the Avaya Quick Edition System due to the traffic load on your network, you can physically segment the voice network from the data network, or configure QoS on the network switches, routers, etc. 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 an Avaya Quick Edition Telephone.

IGMP

The Quick Edition system supports IGMP v2. A number of Layer 2 switches support IGMP snooping by default. If the Quick Edition devices in 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.

STP

If any Quick Edition devices are connected to a switch that runs the Spanning Tree Protocol (STP), configure the communications 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.

About IP Addresses

If a DHCP (Dynamic Host Configuration Protocol) server is connected to the Quick Edition network, the Quick Edition telephones and G11 PSTN gateways will request network addresses from the DHCP server. On networks without a DHCP server, Quick Edition telephones and G11 PSTN gateways assign themselves unique addresses in the 169.254/16 (Zeroconf) network address space with a network mask of 255.255.255.0.

You may assign fixed network addresses to Quick Edition telephones or G11 gateways through Network Options settings. Once assigned, the settings persist until they are either changed manually, or another device having the same IP address is detected at boot time. To re-enable automatic IP Address assignment for telephones or G11 PSTN gateways which previously had manually assigned IP addresses, you must reset the telephone or G11 PSTN Gateway IP address to 0.0.0.0.

IP Multicast

Multicast IP traffic must be permitted on the LAN. The default multicast address used by the Avaya Quick Edition System is 239.192.228.123. This is a limited-scope address that is not routable over the public network (Internet). Connecting all Quick Edition devices to the same network segment—their IP addresses must belong to the same network address space—will help to ensure that IP multicast can be passed between the devices. However, certain high-end network devices do provide the ability to pass multicast traffic between IP segments if configured to do so.

The Quick Edition system (telephones sets and G11 PSTN gateway) use IP multicast to discover each other and detect when a peer is down. The multicast IP address that the Quick Edition uses is 239.192.228.123. Each peer multicast's a hello message to the rest of the system saying they are "alive". If the system detects that a peer has not sent 3 of the hello messages in a row, it will flag that peer as being down.

The Quick Edition also uses IP multicast to stream MOH (Music On Hold) and when paging a group of sets. The handling of IP multicast packets is not identical between all of the Layer 2 switch vendors. Most unmanaged switches simply broadcast IP multicast packet out every port and that is not a problem for the Quick Edition.

The problem arises when a managed switch requires an IGMP querier to be present on the network to forward IP multicast packets to other ports, but one is not present. Symptoms of this are that the Quick Edition sets will display "No Peers Available" and that set to set calls or set to G11 calls will fail with an audible "fast busy".

The IGMP querier would typically be a router which has an interface in the same subnet as that of the Quick Edition. The router needs to be configured to also route IP multicast. Because not all IP multicast implementations require that they be routed across multiple subnets, some vendors have added the option to have their managed L2 switch do the querying.

When the Quick Edition is installed on such a switch (with no IGMP querier present on the network), four options can be implemented to resolved the issue:

- Disable IGMP snooping
- Statically program the membership of the ports that have a Quick Edition device to be part of the multicast group (239.192.228.123)
- Add an IGMP querier (either configure the switch to do it or a router)

Disabling IGMP snooping on a switch means that the switch will no longer "snoop" the IP packets to see what ports are part of a particular multicast group. This has the affect of broadcasting IP multicast packets out to every port. This is the recommended solution because of its simplicity.

Statically programming the membership of the ports to a particular IP multicast group is an alternative that requires more administration.

Changing the query interval timer is yet another alternative but remember that when the timer expires, the Quick Edition system will no longer "see" each other.

Adding an IGMP querier is an alternative to disabling IGMP snooping if the broadcast of the Quick Edition IP multicast packets poses a problem. As mentioned above, there are two methods to implement this; configuring the L2 switch to be the IGMP querier (if supported by the switch) or add a router (L3) and configure it appropriately.

IP Voice Bandwidth

Table 8: RTP audio codec bandwidth usage

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

IP Port Usage

Table 9: IP Port Usage

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

Compatible L2 switches

The list of compatible switches can be found at: http://support.avaya.com/QuickEdition

If you find the Quick Edition is not working properly with your switch, ensure that the switch and network follow the network recommendations in this Appendix. Specifically, make sure the switch ports are configured to be on the same VLAN. If they are on the same VLAN yet the telephones display the message “No peers available”, then try disabling “IGMP snooping” on the L2 switch.

Chapter 8: PSTN Considerations

This appendix provides an introduction to analog trunks, required to connect a Quick Edition network to the Public Switched Telephone Network (PSTN) in North America. Contact your local service provider for information. Analog trunks connect to the G11 PSTN gateway to provide a link between the traditional telephone network and Avaya Quick Edition network.

What is an analog trunk?

The analog trunk is a traditional analog telephone line that is delivered to your home or business. In North America the trunk provides a 10-digit number and connects to your G11 gateway to provide access outside of the local Avaya Quick Edition network.

What type of trunk is supported by the G11 PSTN gateway?

The G11 supports loop-start (LS) analog trunks from the PSTN that connect to the FXO port on the gateway. We do not recommend the use of other trunk types (for example, ground start, Centrex, or digital lines).

Can I connect my G11 PSTN gateway to an IP service provider’s Analog Terminal Adaptor (ATA)?

Yes, this is an acceptable configuration. The RJ-11 connector from the back of the ATA will connect to one of the L1-L4 connectors on the G11 gateway. The loop length setting for any lines connected to the ATA should be configured as “short”. This configuration is performed using system web administration in the Avaya Quick Edition network.

How do I connect a G11 PSTN gateway to the PSTN?

The RJ-11 connector is used to terminate a single PSTN trunk at your location. RJ-11 connectors are also used to connect to fax machines or analog telephones. Multiple connections will exist for multiple lines at a single location. The connections to the G11 PSTN gateway to provide access to the PSTN to place calls outside the local Avaya Quick Edition network.

Can callers use a single telephone number to reach my business when I have multiple trunks?

Yes, a hunt group associates a single number with multiple analog PSTN trunks and ‘hunts’ for available lines in that group. Hunt groups may contain as few as 2 lines. Your service provider can provide multiple hunt groups if required.

How would I share a fax machine with the G11 gateway?

This would require a line splitter on the local PSTN trunk at the location. The PSTN gateway has to be configured not to answer the line that is being shared with the fax machine; programming is performed using Quick Edition web system interface. In this way the fax machine will answer incoming calls that are placed to that number. The line can also be used for placing outgoing calls, however, the line is only available to one device at a time.

Do I need to subscribe to a voicemail service from my service provider?

Each telephone on the Avaya Quick Edition network contains a voice mailbox. An external mailbox from the service provider would only be accessed by incoming callers when all PSTN analog trunks are in use. The Avaya Quick Edition system does not provide a message waiting tone or indicator when there are new messages on a service provider voicemail. Some businesses may wish to subscribe to this service in order to provide additional call handling when all lines from the service provider are in use.

Should I subscribe to call waiting on my trunks from the service provider?

The G11 PSTN gateway does not support call waiting functionality from the PSTN. The Quick Edition telephones provide multiple call appearances that are not directly associated with the PSTN lines connected to the network. These call appearances are associated with multiple physical PSTN lines and more lines will not be available to the Avaya Quick Edition by subscribing to call waiting from the service provider.

How do I determine how many lines I need to support my office?

A ratio one PSTN analog trunk for three to five Quick Edition Telephones would be typical. An environment where users are placing many external calls will required a higher ratio of PSTN analog trunks to Quick Edition Telephones. The Avaya Quick Edition network is scalable and by adding more gateways and subscribing to additional analog trunks from the service provider it is possible to provide more than one line per Quick Edition telephone.

Each trunk provides a connection back to the Central Office (CO) of your service provider to support a single conversation at a time. If your business requires multiple conversations simultaneously, you will require multiple lines from the service provider. How many lines will depend on the number of simultaneous external calls you expect.

Can I associate a single specific number from the PSTN to an internal extension or internal ring group on my Quick Edition System?

Yes, this is configured on the G11 PSTN gateway by assigning a Direct Inward Line (DIL) to a specific user or group without accessing the Auto Attendant. When creating a single DIL for a user we recommend that the PSTN line not be part of a hunt group. DIL relationships are assigned to the extension or ring group on a per line basis.

In what order are lines seized when placing outbound calls?

When outbound calls are placed from the Avaya Quick Edition network they capture lines on the G11 gateway polling from L4 to L1. If one of the lines is in use or disconnected it looks to the next available line. For configuration purposes, note that the PSTN line connected closest to L4 will be used the most for outbound calls.

How do I assign a line on the G11 gateway so only certain users have access to that resource for outbound calls?

Programming is performed using Quick Edition web system interface. Lines L1 through L4 on the G11 gateway can be assigned as Private Lines (PL) to reserve the PSTN resource on that gateway port for the assigned telephone or group. This ensures that only the specific users will capture this line for outbound calls. The extensions will still have access to the other lines, however the PL assigned ensures they are given exclusivity to that line.

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