Release Notes

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Quick Edition Release 3.3 GA

Release Notes

These release notes provide a high level description of new features, along with resolved and existing issues in Avaya Quick Edition Release 3.3 GA.

Contents

Release 3.3 GA 1
New Features in Release 3.3 2
Issues Resolved in Release 3.3 3
Incidents Reports and Workarounds 4
Installation and Upgrades 4
System and Network Configuration 4
General 6
A10 Analog Telephone Adapter 7
Connecting Quick Edition to CM/SES or Service Provider 7
Multisite Provisioning Tool 8
Web Admin 8

Release 3.3 GA

Device	Firmware	Build & Date	Bootloader
Avaya 4610	7.3.2153	2153 – March 15 2008	2.2.14
Avaya 4621	7.3.2153	2153 – March 15 2008	2.2.14
Avaya G11 PSTN gateway	7.3.2153	2153 – March 15 2008	n/a
Avaya Quick Edition Upgrade Wizard	n/a	3.3.1222008	n/a
MPT	3.3	A-033108	n/a
A10 Analog Telephone Adapter	3.21.0	Wed Jan 13 2007	n/a
G20 ISDN BRI gateway	3.21.0	2007-5-14	n/a

New Features in Release 3.3

More details on the new features can be found in the Release 3.3 documentation available at http://support.avaya.com/QuickEdition.

Call Pick-up

Call Pick-Up is a feature that enables a user to answer (pick up) a call that is alerting (ringing) on another. There are three variations of this feature.

Call Pick Up – Any This feature allows a user to answer calls that are ringing on other sets with a pickup group. The user attempting the pickup has to be part of the same pickup group as the ringing set, for authorization to pickup the call. Once the user triggers Call-Pickup-Any, the system arbitrarily selects the set and the particular call targeted for pickup.

Call-Pickup - Group This feature allows a user to answer a particular call that is ringing on a particular set (extension) in a pickup group. A user attempting the pickup has to be part of the same pickup group as the ringing set, to be authorized to pickup the call. This feature provides a list of all the current ringing calls to the users in the pickup group, thus allowing the pickup user to select a particular call for pickup based on the calling or called party.

Call Pick Up – Extension This variation allows any user to pick up a call ringing at any other station, specifying the station by selection from a list. A user attempting the pickup has to be part of the same pickup group as the ringing set, to be authorized to pickup the call.

Multilevel Auto Attendant

The Quick Edition Auto Attendant application has been enhanced to support multilevel menus. This capability will allow small business, specifically retail and professional establishments, to direct callers to the appropriate department or service desk.

Hotkey transfer to VM

Hotkey transfer to voice mail supports call termination directly to a voice mailbox. Doing so improves call handling times and presents a professional image in offices where a person is involved in handling customers' calls. For example, a receptionist, knowing that the CEO is not to be disturbed, transfers a caller directly to the CEO's voice mail. The feature may be used in conjunction with the status feature introduced in Release 3.1, but is not intended to be exclusive to that environment. Call-forward options could support voice mail boxes as valid destinations. This would support forwarding of a user's cordless phone (on ATA) to their QE desk phone voice mail. It could be used to facilitate forwarding of multiple phones to a common mailbox.

Service Provider Keep Alive

Typically, Quick Edition would maintain connectivity to a SIP service provider by sending registration request within a specified time range, ensuring that the pin hole through the service provider's firewall is kept open, maintaining registrations and allowing RTP traffic to pass between phone and proxy. Release 3.3 sees the introduction of Keep Alive feature. This feature allows Quick Edition to maintain connectivity with the SIP service provider by using a simple ping-like command that maintains connectivity through the firewall. This maintains registration and allows for transmitting and receiving of voice packets, while putting less demand on the service provider's softswitch.

Increased Capacity

Due to the increased demand from our business partners, Quick Edition can now scale up to:

- Supports up to 40 Quick Edition telephones (4621, 4610).
- Supports up to 10 PSTN Gateways (G11 or G20).
- Supports up to 10 Analog Telephone Adapters (A10).
- Supports 40 FXS type SIP extensions (soft SIP clients or SIP phone).
- Supports 1 FXO Gateway (TI/E1/PRI 3rd party gateway) up to 32 channels.

Quick Edition Re-Purposing Wizard

Provides the ability to convert Quick Edition phones to an H.323 format so that the phones can be re-used for migration to Avaya IP Office, Communication Manager or Distributed Office.

Issues Resolved in Release 3.3

7143: My Status translation is required for non-English languages.

7395: Incorrect Italian Translation on Web Admin build 1774

7560: No audio after a consult transfer to QE with Gamma proxy.

7604: Tone support Argentina and other South American Countries.

7606: Proper error message is required when license limitation is reached.

7644: RMS: Scheduling a macro to create a new AA fails.

7678: RMS: AMPT does not work in JRE 1.6.

7692: Memory leak for BRI or ATA registrar.

7694: Call Log: Incorrect entries in Incoming Call Log in case of Transfer.

7706: DAA forwarder changes codec in re-Invite results one way audio.

7707: Cbeyond SP: Virtual ID registration fails.

7723: AMPT: Truncated line on installation wizard.

7778: Deleting SIP proxy Identities does not remove entries from AA name tree.

7792: SIP: Server Returns 486 Busy and QE Sends Audio.

7873: Web Admin: Need to handle '+' dialing in all applicable fields.

7938: CDR: Wrong call type for DAA calls from G11.

7943: Documentation: (QE_SysAdmin_R3.2).

7988: Re-registration not occurring within configured expiration period

8053: Web Admin: Cannot disable SIP identity thru change screen

8059: Field Issue: DIL to Group or Ext fails if answered on first ring.

8060: Cannot Delete SIP Identities

8061: Field Issue: Outbound SIP call /w Voice Flex missing first second of audio

8069: Edit Call Pickup Groups – delete group member

8093: "Keep alive" feature needs to be configurable from Web Admin

8147: SP: Sometimes SIP Identity remains in state ADDING after upgrade

8225: SP interoperability - QE registration refresh does not work in some cases

Incidents Reports and Workarounds

Installation and Upgrades

7577: A10 ATA or G20 BRI not useable after an upgrade.

If the upgrade of an A10 ATA or G20 BRI is interrupted (i.e. network issue), it may be unusable after the upgrade.

Workaround: *There are three potential workarounds.*

1. *if no provisioning changes have been made, power cycle the A10 or G20,*
2. *change the DHCP setting and save, then set it back to its original setting and save,*
3. *factory reset the device by pressing and holding the Reset button until the "Run" or "Power" LED begins to flash and then release.*

7740: Downgrades fail from Release 3.2

Workaround: *Initiate the software downgrade using the installation wizard. When prompted to "Select Upgrade Interface", select "Upgrade using another interface (MPT, Phone or Web interface). Completing the downgrade using one of these methods will avoid the installation wizard's version check.*

7962: BRI incoming call failed after downgrade

If a system downgrade is required, BRI identities cannot receive calls upon restart.

Workaround: *Delete and re add the identities.*

8164: Phone remained in initialization mode after fresh start upgrade

When executing a clean database installation, some sets may remain in initialization mode.

Workaround: *Reboot the affected sets.*

System and Network Configuration

4985: You can not change the SMTP port in the system to a 5 digit number.

The help text shows that the valid Port Numbers for SMTP mail is 25, or in the range 1025-65535. The system allows the entry of a 5 digit port number, but strips the fifth digit when the port number is subsequently displayed.

Workaround: *Set your SMTP server to use a port numbered less than 10000.*

6284: Some dialing rules not visible when a dialing rule name has more than 20 characters.

If a dialing rule has been created with more than 20 characters, any dialing rules that fall after that rule alphabetically will not be available when applying the rules to a group.

Workaround: *Use 20 characters or less in the names for dialing rules.*

7108: SIP identity can not be modified in Web Admin.

Workaround: *In order to make changes to a SIP identity, the identity must be deleted and recreated with the changes.*

7219: Web Admin does not open with Firefox version 2.0.

Workaround: *Web Admin is supported with Internet Explorer versions 6.0 & 7.0.*

7291: Web Admin pages do not load correctly with proxy server enabled in Internet Explorer.

With a proxy server set in Internet Explorer, some pages on Web Admin may not load correctly.

Workaround: *Temporarily disable proxy server settings in Internet Explorer while using Web Admin.*

7407: Active G10 or G11 PSTN gateway may not be visible on Web Admin device management after removing another G10 or G11 gateway.

This will occur if a G10 or G11 PSTN gateway is deleted from the network while a Quick Edition phone set is disconnected from the network. When the Quick Edition phone is reconnected to the network, other G10 or G11 gateways in the network may not be visible on the Web Admin Device Management screen.

Workaround: *Remove the G10 or G11 gateway that was previously deleted from the network again. This can be done on a phone from "System Options/Set Management/Remove Extension" or on the Web Admin from the Corporate Directory page by selecting "Remove Entry".*

7433: A10 ATA or G20 BRI does not function with VLAN tagging enabled.

Workaround: *VLAN tagging must be disabled if you have an A10 or G20 in your network.*

8145: Call Pickup Alerts not in User Backup/Restore

Backup and Restore does not backup alerts.

Workaround: *Manually note all alerts.*

8144: Call Pickup Groups not in System Backup

Call Pickup groups do not appear in a System Backup.

Workaround: *Manually note all entries.*

7682: "Audio Bandwidth" is reset to default value after rebooting system

If the audio bandwidth is changed to "low from the default of high and the system is restarted for any reason, the value of the audio bandwidth changes back to the default of high.

Workaround: *Manually change the value back to low.*

8217: MLAA: Unable to Add Menu Configuration to a new AA through the phone UI

When adding an AA through the phone UI then adding Menu Configuration the user is presented with the error "Unable to save this AA".

Workaround: *Use Web Admin to Add Menu Configuration to a new AA. Alternately if Web Admin is unavailable, firstly add the AA through the phone UI and save and then edit the AA through the UI to add the Menu Configuration.*

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General

5642: Caller ID not available on DIL (direct inward line) calls from PSTN if forwarded or answered before the Caller ID is made available from the PSTN.

Workaround: *Answer or forward calls after the PSTN has provided the Caller ID information. This is typically received from the PSTN between the 1st and 2nd rings.*

6216: Transfer could not be completed after going on-hook to complete the transfer.

If you attempt to complete a transfer by going on-hook, the transfer may fail.

Workaround: *Instead of completing the transfer by going on-hook, complete the transfer by pressing the Transfer hardkey on the phone set.*

6228: Blind transfer to a set with Call Forward All to an external number enabled is not reliable.

Workaround: *The transferor should wait for ring-back before completing the transfer.*

6238: Auto Attendant will not detect DTMF if caller is connecting with G.711 codec.

Workaround: *If you are using a 3rd party gateway to connect to a SIP service provider, this gateway must support both G.729 and G.711. If you use a gateway that only supports G.711 then you will not be able to use the Auto Attendant functionality of the Quick Edition system.*

6288: Calls to a large group may fail with Call Forward No Answer set to less than 3 rings.

If a large group (i.e. 10 members) has call forward no answer set to less than 3 rings, the call may fail if answered from one of the group members.

Workaround: *For large groups, use at least 3 rings for the call forward no answer setting*

6904: Web Admin device management links may not redirect properly after a system admin password change.

After changing the system admin password, the device management links to other devices may present an “Access Denied” error, followed by requesting the system admin password. You will then be required to log in again to access the device.

Workaround: *Changing to a different password resolves the issue.*

7094, 7095: Auto Attendant dial by name feature won't accept entire name or spell back entire name with special characters.

Workaround: *Names entered on the Quick Edition sets should only include alpha-numeric characters. You should not use accents, commas, or other special characters.*

7621: A10 ATA or G20 BRI provisioning may not be restored after a backup restoration.

Workaround: *After performing a system restoration, ensure that all identity mappings are correct for A10 and G20 devices.*

8212: Maximum number of SIP Subscriber Identities Reached @ 8

The original limit set for SIP proxy identities was hard coded at 8, any new installation with a software version greater than 3.1 will not encounter this issue.

Workaround: *Contact P2P support for CLI command to reset the limit.*

A10 Analog Telephone Adapter

6766: Dial Plan reset does not propagate to registration for an A10 ATA.

After changing or resetting the dial plan, the extension range for the system may change. This will automatically update and change all extensions in the network for Quick Edition phones, but will not for the extension numbers associated with an A10 ATA device.

Workaround: *After changing the dial plan you may be required to manually edit and update the extension numbers associated with the A10 ATA devices if the extensions do not fall within the new extension range.*

7113: One extension number can be assigned to multiple A10 ATA ports.

Workaround: *While it is possible to assign the same number to multiple ports, each A10 ATA port must be assigned a unique extension number (SIP Identity).*

7244: A10 ATA configuration changes don't take effect if A10 is disconnected.

If an A10 ATA is powered down or disconnected from the network when you make provisioning changes, these will not take effect after the device is enabled back to the network.

Workaround: *After the device is enabled back to the network, you must access the System Options on Web Admin, alter a provisioned value, and submit the provisioning changes.*

7637: A10 ATA phone cannot retrieve parked call.

Workaround: *Calls can be transferred to an A10 ATA phone, but parked calls cannot be retrieved.*

7749: ATA to QE FWD ALL to SIP Device fails (no DAA)

If an ATA extension initiates a basic call to a Quick Edition set that is setup to forward all calls to an external number over the SIP service providers network, the call will fail. The calls fails as there is not registration associated between the ATA extension and the service provider.

Workaround: Configure 1 Global identity on the Quick Edition network.

Connecting Quick Edition to CM/SES or Service Provider

5121: If a Service Provider Identity is configured with a Domain Name different from the Proxy Host, outgoing INVITE messages will use 'user@proxy' in the "To" and "From" URI's instead of "user@domain".

Workaround: *The domain name of the registrar and proxy must be the same.*

5207: CM trunks used to connect to Quick Edition systems must have Direct IP-IP Media Shuffling disabled.

If Direct IP-IP Media Shuffling is not disabled, there can be an extended delay (up to one minute) in opening the audio path on calls forwarded from CM sets to Quick Edition Sets.

Workaround: *Disable Direct IP-IP Media Shuffling on CM trunks used to connect to Quick Edition systems.*

5913: Changes to identities on QE or SES do not force a re-registration.

Changes to username, domain name, registrar name/IP, registrar port, outbound proxy name/IP, or outbound proxy port will trigger a re-registration. Changes to other data may not trigger a re-registration.

Workaround: *When making other data changes, forcing a re-registration is recommended by power cycling the phone.*

6301: Conference calls may fail when conference legs are in different regions.

If you attempt to create a conference with parties that use different codec selections (G.711A vs. G.711MU), one of the conference legs will be dropped.

Workaround: *Disabling Media Shuffling on CM may solve the issue.*

7518: DID/DDI service provider calls to a group of only ATA phones does not function.

In order for a group to be registered on SES for DID/DDI, it must contain at least one Quick Edition set.

Workaround: *Include at least one Quick Edition set in the group, or use Auto Attendant for routing calls to the group of ATA phones.*

7972: Telenor SP: Dialing QE Group generates multiple incoming calls

This bug is specific to Telenor, because of strict registration requirements ring groups on cannot be registered as PID.

Workaround: *Ensure that group registrations to Telenor are VID and not PID.*

7970: SP VoiceFlex DAA doesn't work sometimes

This bug is specific to VoiceFlex and cause because the local firewall is blocking the RTP stream.

Workaround: *Set the local firewall set the port range for RTP to within the range of 10,000 to 40,000.*

Multisite Provisioning Tool

5739: Cannot reach site with no operator defined.

If the operator set has been removed from the network, the MPT will not be able to connect to the site.

Workaround: *Set a valid operator and attempt to reconnect with the MPT.*

Web Admin

8379: Web Admin: Changing/creating/deleting a SIP proxy identity displays "Application error" on webpage

Workaround: *Although the Application error is displayed, the change has actually been committed to the database.*

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