POLYCOM®

KIRK Release Note

KWS600v3

Firmware Version 09-60700.89
Q4/2009

Table of Contents

1. General

This release note applies to IP firmware version 09-60700.89 and radio firmware PCS05Ag of the KWS600v3. This release replaces the 09-60700.87 release as the latest generally available (GA) release.

2. Important Notes

This firmware release has been tested on our automatic system against the following systems:

- Cisco Unified CallManager version 4.1.3 SR2
- Cisco Unified CallManager version 5.1 5.1.1.3
- Cisco Unified CallManager version 6.0 6.0.1.2000-3
- Cisco Unified CallManager version 6.1 6.1.2.1
- Cisco Unified CallManager version 7.0
- SIP Asterisk

Please notice, if you are using SIP there is no need to upgrade unless you are using the msf interface or user.dll.

Besides the system test, the msf interface has been tested with our automatic system and our msf stress tester.

This release has been interoperability tested on CUCM 7.0.0.39771-4
When you upgrade, follow the procedures described in the KWS600v3 upgrade guide.
Upgrade all units on the system with the new firmware files.

2.1 Skinny protocol: Removal of user data

If a user is removed from the CCM (user data deleted), but not removed from the KWS600v3 LDAP server (in this test the IP-Master), registration attempts are sent to the CCM.

This means that user-data removed from the CCM must always be removed from the LDAP-server; otherwise it will produce unnecessary signaling

2.2 SRST:

This firmware only supports SRST 4.0 and upwards. SRST 3x and lower is not supported.

2.3 System Requirements

Hardware Platform:	**Description**
KWS600v3	KWS600v3

2.4 Terms and Definitions

WCAG Web Content Accessibility Guidelines. W3C Recommendation. These guidelines explain how to make Web content accessible to people with disabilities.

3. Distribution Files

Click here >> to find the IP firmware 09-60700.89

Click here >> to find the BMC firmware: PCS05Ag

4. Changes

4.1 Version Q4/2009

4.1.1 Added or Changed Features

4.1.1.1 Skinny

- When adding (subscribing) a new handset to the system, the registration is now done immediately towards the Call manager.
- The registration procedure has been optimized and failover / fallback works faster than before.
  Note, if there is a firewall between the IP600v3 and the Callmanager, you may have to run a special command, to slow down the registration process, in order to not trigger the firewall.

  Command instructions (The command ends with a reset, and if you are using other / commands on the skinny line, you need to merge the commands on one line):

  http://172.29.200.201/!config change SKINNY /slowregmode

  http://172.29.200.201/!config write

  http://172.29.200.201/!config reset

4.1.2 Corrections

4.1.2.1 General system

- In our latest release the users.dll (used by 3rd party application) was not working correctly, this has been fixed and a new users.dll can be obtained by contacting Polycom Denmark or our service department.
- A new msf developer package is also available, and can also be obtained by contacting Polycom Denmark or our service department.

4.1.2.2 Skinny

- When making a call transfer from an external call to a number with call forward active, and the transfer ended up with voice mail contact, it was not possible to toggle back to the external call. This has been fixed.

4.1.3 Configuration File Parameter Changes

None

4.1.4 Outstanding Issues

- Unable to Resume from Hold or forward call in Mitel PBX
  https://jira.polycom.com:8443/browse/DECTESC-118
- Skinny callwaiting can end up with call transfer
- A MWI update is only delivered if the handset is in range when using SIP.
- Skinny Group pickup is not supported and has never been supported (by mistake it has been published as supported in our feature list), but a similar feature “Call pickup” is still fully supported.

4.2 Version Q3/2009

4.2.1 Added or Changed Features

4.2.1.1 Skinny

- Added new EMC: 0x0138B (05003) and 0x138A (05002).
- Delay on incoming call can now be disabled (the delay is by default enabled):

  On large installations it’s not recommended to disable delay on incoming call, because it could give a high cpu load on the ip master. On small installations this function can be disabled.

  We distinguish between large and small installations in the following way: if there are more than 64 bases on the system or more than 750 phones on the system, it’s a large installation - otherwise it’s a small installation.

  To disable delay on incoming call, navigate to this page on the ip master: Configuration->Dect->Master and enable the checkbox: “Disable broadcast delay”
  Now restart the ip master

  If you have configured an alternative ip master on the system, make sure to run with the same settings as the ip master.

4.2.2 Corrections

4.2.2.1 General system

- When enabling the checkbox “No Display of Date and Time” in Configuration->Dect->System it caused 100 % cpu usage in version 60700.55, this is fixed both in 60700.80 and 60700.83

4.2.2.2 Skinny

- Emc check on all msf events from skinny. Before 3rd party phones received those, and some 3rd phones cannot handle this.
- By default, we now disable standby text updating. Before enabling standby text update, ensure all Polycom handsets on the system support this feature.

  We use standby text updating when:
    - a phone is forwarded. The standby text is then updated to: “Forwarded to xxxx” where xxxx is the number the phone is forwarded to.

- a phone has a MWI. The envelope on the phone is controlled with standby text update.

To enable standby text updating run these commands on the ip master (and alternative ip master if available):

http://172.29.200.201/!config change SKINNY /msfupdate
http://172.29.200.201/!config write

Then restart the base:

http://172.29.200.201/!reset

To disable standby text updating run these commands on the ip master (and alternative ip master if available):

http://172.29.200.201/!config change SKINNY
http://172.29.200.201/!config write

Then restart the base:
http://172.29.200.201/!reset

Note: Replace the ip address with the ip address of you ip master.

The following is a list of required firmware versions in handsets that support standby text update:

Handset type	**Firmware version**	**Comment**
5020 and 5040	Earlier than PCS05Ja	Standby text update not supported, update handset firmware, before enabling the feature.
5020 and 5040	PCS05Ja	Support standby text update, stored in eeprom.
5020 and 5040	PCS 06Ha	Support standby text update, not stored in eeprom.
4020 and 4040	PCS 06Ba	Supported
4020 and 4040	Earlier than PCS06Ba	Standby text update not supported, update handset firmware, before enabling the feature.
3040	13309910.06Cc	Supported
3040	Earlier than 13309910.06Cc	Standby text update not supported, update handset firmware, before

		enabling the feature.
OEM handsets		Check with your supplier.

4.2.3 Outstanding Issues

- Unable to Resume from Hold or forward call in Mitel PBX
  https://jira.polycom.com:8443/browse/DECTESC-118
- Skinny callwaiting can end up with call transfer

4.3 Version Q2/2009

4.3.1 Added or Changed Features

4.3.1.1 General

- FREQUENCY: The frequency band BRAZIL has been added. (Subset of South America).

4.3.1.2 Skinny

- STANDBY TEXT: When activating Call Forward All, a forward text is now shown as StandByText.

4.3.2 Removed Features

- SIP MWI

4.3.3 Corrections

4.3.3.1 General

- PERFORMANCE IMPROVEMENT: To improve performance in the IP-master. permanent TCP connections are setup between the IP-master and slaves.
- No relevant status messages removed from SYSLOG:
- Incoming call fix: Sometimes incoming calls didn't alert in the handset. This happened because of hanging instances in the radio(bmc), now we have implemented cleanup functionality. (Jira: DECTSEC-98)

4.3.3.2 Skinny

- REPLICATION DATA ERROR: If "No Display of Date and Time" in system settings was marked, this lead to a replication error. User data was not replicated to other IP600v3. After this setting an earlier downloaded configuration file could not be uploaded to the IP600v3.
- KEEP ALIVE: The time between keep alives sent from the KWS600v3 to the Call Manager could be changed when failing over to an Alternative Call Manager and back again
- EDIT USER: If the user name was changed, registration was removed, and the phone was never registered again.

- HUNTGROUP: Two incoming calls to a hunt group failed. One call was not released correct after call release
- MWI: MWI shown in display was removed after PP onhook/offhook
- MWI: If MWI was received during pp out of range or during pp powered down, then MWI was not shown after pp in range again or after pp power on.
- CALL FORWARD ALL: Correct code for enable and disable Call Forward All must now be used before the code is accepted. Earlier the enable code could be used to disable the feature. After enable/disable the pp returns to Idle after approx 4 sec

4.3.3.3 MSF

- If the PP was in the charger at the time of sending an msf, the right release reason was not sent, leading to retransmissions due to missing releases.
- Hanging msf call fixed. (Two different problems with PPStatusReq, and PPOutOfRange).
- Potential "Handset to handset MSF" trap removed.
- IP master trap fixed, we were able to reproduce the state that could cause the trap, but we never reproduced the actual trap, however removing the error state just before the trap should also remove the trap.
- Improvements on the msf interface to be more tolerant for 3rd. apps not following the protocol.
- Queue system in the radios removed, which caused all the bases to trap.
- Clean up hanging object in the radios, in the old versions handsets could be “locked” and a system restart was needed in order to make msf work again on the handset.
- IP master queue counter could be decremented from zero causing the system to be msf busy.
- IP master paging queue counter could miss some decrement which made the system slower to handle msf.

4.3.3.4 SIP

- Nonce error fixed

4.3.4 Configuration File Parameter Changes

None

4.4 Version Q4/2008

4.4.1 General

4.4.1.1 Corrections

None.

4.4.1.2 Added or Changed Features

None.

4.4.2 CUCM 6.1

4.4.2.1 Corrections

- VOICE MAIL. Beep tone was missing when setting up a voice mail. After comfort noise was received the dsp did not start decoding RTP-packets fast enough. Loosing RTP-packets was depending on the time between comfort noise and receiving RTP-packets again. The problem has now been resolved.
- REGISTRATION: When running with many users and SKINNY protocol, problems with ether-net breakdown have occurred at the ip-master. The problem was that the users tried to set up a socket to the CUCM simultaneously causing load problems. This problem occured from 700-1500 users. Now, the SKINNY tcp "socket connect" is carried out and controlled from one timer. This means that one socket must be connected, before the next socket connect is requested. With 1500 users, it can last for up to 5-6 minutes before all sockets are connected on both CUCM. With this change the reliability has improved.
- REGISTRATION: Re-registration has been fixed. It has now been ensured that the registration is always done if requested.
- KEEPALIVE: Resetting users from the Call Manager (CUCM) with many registered users caused keep alive problems. Users were unregistered because the keep alive time was changed if the users registered to the secondary CUCM. This was seen after adding/editing users with following reset from the CUCM. The problem occured from 700-1500 users. The problem has now been resolved using only the primary CUCM keep alive time. To give better performance on the system a further change has been made. Now keep alive is sent in groups. With each group 5 keep alives are sent. Between each group is a fixed delay of 100 msec.

4.4.2.2 Added or Changed Features

None

4.4.2.3 Removed Features

None

4.4.2.4 Configuration File Parameter Changes

None

4.4.3 SIP (Asterisk)

4.4.3.1 Added or Changed Features

None

4.4.3.2 Removed Features

None.

4.4.3.3 Configuration File Parameter Changes

4.4.4 SIP (Broadsoft)

4.4.4.1 Added or Changed Features

None

4.4.4.2 Removed Features

None.

4.4.4.3 Configuration File Parameter Changes

4.5 Q2/2008

4.5.1 General

4.5.1.1 Corrections

None.

4.5.1.2 Added or Changed Features

None.

4.5.2 CUCM 6.0

4.5.2.1 Corrections

- DECT TRACE: Unknown events now shown.
- RING BACK: If remote user disconnect was received ring back (remote user on hold and local user went on hook) it could happen that ring back was still done. This has now been stopped.
- PERFORMANCE IMPROVEMENT. Internal signaling CALL-PROC from RADIO-base to IP-master removed for incoming calls. This has been done to reduce internal signaling between the KWS600v3
- MSF: Handles MSF when no registered KWS600v3 slaves.
- DECT/BMC INTERFACE: NicSubsAndUserDatCfm added. If user data don't read by BMC, data is retransmitted. Parameter returned to see if any conflict.
- CCM FAILOVER/FAILBACK. Failback from SRST to CUCM2 failed
- SRST DATA: Old SRST data now removed from config file if new SRST data is used.
- IP MASTER MENU RADIOS: Busy field in IP Master "Menu" and "Radios" turned red if KWS600v3 lost connection to the IP Master and calls were tried. Now BUSY is only set if all radio channels on the KWS600v3 are occupied.
- PP CLOCK: Update location registration has now been implemented.

4.5.2.2 Added or Changed Features

- MSF BROADCAST: Support of MSF broadcast sent to a single user, to a group or sent to all users.
- STANDBY TEXT: Standby text can now be updated without turning the handset power off and on.
- FREQUENCY BAND: Changing frequency band for Europe, South America and North America implemented.

4.5.2.3 Removed Features

None.

4.5.2.4 Configuration File Parameter Changes

None.

4.5.3 CUCM 4.1.3

4.5.3.1 Corrections

- KEEP ALIVE: Has been changed in order to improve the CPU performance of the KWS600v3, and keep alive is not sent to ALT-CUCM if CUCM connection is OK.
- DELAYED INFO: At incoming calls, info is only sent to the KWS600v3 with alerting handset. In the previous version, info containing additional display- and ringer info was sent to each handset in the system making unnecessary signaling.
- MISSING RINGING: When CPU load was high, the handset did not always ring when an incoming call was received. This problem has now been solved.
- BCT: A-Blind Call Transfer number at C-user was not always displayed correctly when BCT was activated by B-user. This problem has now been solved, and C-user is now always displayed correctly.
- REGISTRATION: User Name is now checked in the edit user menu of the KWS600v3. Registration is only performed if name starts with “SEP” and consists of 15 characters (SEP + 12 digits). If not, no registration is performed.

4.5.3.2 Added or Changed Features

- MWI: Message Waiting Indication
- NAME: Called/calling name is now shown for outgoing/incoming calls.
- CONFERENCE. “Conference” text now shown when conference established.
- SRST: Can now be entered into cfg file if tftp is not supported.

4.5.3.3 Removed Features

None.

4.5.3.4 Configuration File Parameter Changes

None.

None.

4.5.4 SIP (Asterisk)

4.5.4.1 Added or Changed Features

The DNS features are added in this release:

- DNS SRV lookup: DNS SRV is a mechanism used for high availability and load balancing.
- DNS fail over: If primary server is failing, the secondary server takes over.
- Fully Qualified Domain Name (FQDN)

MWI subscription:

- Unsolicited: Nothing is written in the “Message Center” field under Configuration/DECT features and the mail icon now turns on when a voice mail is received.
- R button: Dial tone is now always available after R button has been pressed.
- The page “Master” under Configuration/DECT has been updated with new fields and some fields have been removed as well.

4.5.4.2 Removed Features

None.

4.5.4.3 Configuration File Parameter Changes

None.

4.5.4.4 Important Notes

The following tests carried out in the SIP qualification test report “SIP_QT_KWS600” failed:

Test Step	Description	Expected Result	Actual Result
90-93	A Off Hook calls b and B leaves range during alerting.	Sending cancel and receiving 487.	KWS600v3 sends a 603 DECLINE when out of range.
274-283	Blind transfer	Verify that the Refer to Header in the REFER request has no replace parameters.	Refer to Header in the REFER request has a replace parameter.
298-304	Blind transfer: Transferor to unknown user	A receives error tone	A receives busy tone
396-379	Attended transfer: Transferor to	A receives error tone	A receives busy tone

Test Step	Description	Expected Result	Actual Result
	unknown user		
454-461	DTMF: Enable DTMF with info: Press digit 0-9 and *, # from X	Verify that the pressed digits are received	KWS600v3 is not able to receive DTMF tone via INFO
543-549	Codec G711 u-Law: Enable only G711 u-Law with 30ms on A and 30ms on X	Verify in RTP that payload is 30ms	Payload is 20ms instead of 30ms when sent from KWS600v3
591-598	Codec G729: Enable only G729 with 30ms on A and 30ms on X	Verify in RTP that payload is 30ms	Payload is 20ms in stead of 30ms when sent from KWS600v3
654-664	Make sure that the primary server is failing: Make a call from A to B and answer the call.	Verify that the INVITE request is sent to the primary SIP server and retransmitted.	The KWS600v3 does not follow the signaling in this test case, however the way that the KWS600v3 is signaling is not wrong.

4.5.5 SIP (Broadsoft R13)

4.5.5.1 Added or Changed Features

The DNS features are added in this release:

- DNS SRV lookup: DNS SRV is a mechanism used for high availability and load balancing.
- DNS fail over: If primary server is failing, the secondary server takes over.
- Fully Qualified Domain Name (FQDN)

MWI subscription:

- Unsolicited: Nothing is written in the “Message Center” field under Configuration/DECT features and the mail icon now turns on when a voice mail is received.
- R button: Dial tone is now always available after R button has been pressed.
- The page “Master” under Configuration/DECT has been updated with new fields and some fields have been removed as well.

4.5.5.2 Removed Features

None.

4.5.5.3 Configuration File Parameter Changes

None.

5. Important Notes

The following tests carried out in the SIP qualification test report “SIP_QT_KWS600 against Broadsoft” failed:

Test Step	Description	Expected Result	Actual Result
91-94	A Off Hook calls b and B leaves range during alerting.	Sending cancel and receiving 487.	KWS600v3 receives CANCEL followed by time out (408) and sends 487 request terminated.
279-288	Blind transfer	Verify that either the transferor or the transferee sends a BYE request from the transferee. Verify that the Refer to Header in the REFER request has no replaces parameter.	Cannot verify.
545-550	Codec G711 u-Law: Enable only G711 u-Law with 30ms on A and 30ms on X	Verify in RTP that payload is 30ms	Payload is 20ms instead of 30ms when sent from KWS600v3
592-598	Codec G729: Enable only G729 with 30ms on A and 30ms on X	Verify in RTP that payload is 30ms	Payload is 20ms instead of 30ms when sent from KWS600v3
658-664	Make sure that the primary server is failing: Make a call from A to B and answer the call.	Verify that the INVITE request is sent to the primary SIP server and retransmitted.	The KWS600v3 does not follow the signaling in this test case, however the way that the KWS600v3 is signaling is not wrong.