Spectralink DECT Server 2500 and 8000

Release Notes 2016

Firmware Version PCS16B_

Table of Contents

Introduction

These release notes apply to the released firmware versions for the Spectralink DECT Server 2500 and the Spectralink DECT Server 8000. This version specifically applies to version PCS16B_ of the firmware. The release is a main release that replaces the PCS16A_ release as the latest generally available (GA) release.

Feature License and Platform Limitations

The following table summarizes features that require a particular hardware platform and / or a license key for activation.

Feature	*Comment*
Software Security Package (TLS, SRTP)	License required. Part number 14075280
Automatic Alarm Call	License required. Part number 14075450
Handset Sharing	License required. Part number 14075460
TAP Interface	License required. Part number 14075470
Cisco Unified CM Enhanced Features for DECT for DECT Server 2500	License required. Part number 14075491
Cisco Unified CM Enhanced Features for DECT for DECT Server 8000	License required. Part number 14075496
Microsoft Lync Interop for DECT server 2500 incl. Spectralink Software Security Package (SRTP)	License required. Part number 14075492
Microsoft Lync Interop for DECT server 8000 incl. Spectralink Software Security Package (SRTP)	License required. Part number 14075497
LAN Sync for DECT Server 2500	License required. Part number 14075580
LAN Sync for DECT Server 8000	License required. Part number 14075590

Distribution Files

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Changes

Version PCS16B_, 2016 - Q3

For this and future software releases, use only the WEB-GUI to configure your system. If you cannot access the WEB-GUI, please use Tiny OAM 0.3.1.0 for configuring IP settings.
If the previous OAM Administration Program is used, it might corrupt the configuration.

Added or Changed Features

- Added base station radio synchronization loop detection. It is of major importance that the configuration of the synchronization of the base stations does not contain loops. When Loop button (on the base station administration page) is pressed, a loop detection routine is started. Warnings will be presented directly to the user. Be aware that the loop detection might be unable to detect loops correctly if the configuration contains duplicate RPNs or repeaters are involved.
- Embedded RFP6 firmware PCS16E_.
- Added and changed some text in the service report (pp_statistic.txt).

Known limitations

- Shared lines (Handset Sharing) and internal switching cannot be combined.
- 3rd party call control does not work on handsets configured as DECT-to-DECT.
- Be aware: The 74 series handsets do not have the features required to have full SIP call handling functionality; ex. tones and display text will be missing in some situations.

Corrections and Improvements

- DECTESC-700 and DECTESC-689 - overlapped sending.
  A handset can make an outgoing voice call in two ways.
  1) En-bloc sending of digits (dial extension and press hook key - like GSM phones).
  2) Overlapped sending of digits (press hook key and dial extension (digits one by one)).
  This issue only addresses older handsets subscribed to SIP lines with older firmware that uses overlapped sending. The dialed extension was unpredictable.
- DECTESC-697 - Ringing only once.
  User subscribed to analogue lines that is connected to a PBX (with some country specific settings) could experience handsets only ringing once. The timing between start and stop ringing were too short. Before handset received ringing start (An actually begins to ring) the handset receives a stop ringing. Previously a start ringing signal contained only ringing information. During the last year a display text has been added to the ringing information signal. The ringing signal is now divided into two signals send to the handset. In general if problems with "short" ring tones are experienced tone number 6 from the handset can be

used, it has a much shorter acoustical ramp up than the other tones. Furthermore the length of the ring tones can be extended by adjusting the "minimum ring time" value (Configuration | DECT Server | Ringing mode) in the DECT Server WEB-GUI.

- DECTESC-646 – Don't receive XML_RPC messages.
  When messages was received fast via XML_RPC (but still within legal timing) sometimes messages was lost due to an internal race condition. The DECT Server was sending "ReleaseReason 84 [KWS unknown rel. reason]" to the 3th. party application. This problem has now been solved.
- Fixed problem with special characters in handset
  If "Internal CLIP and Presentation" is enabled (WEB-GUI - Configuration | DECT Server | Ringing mode), special characters was not shown correctly in handset (typical name and presentation text). The error was introduced in February 2015 and has now been corrected.

Version PCS16A_, 2016 - Q2

For this and future software releases, use only the web-GUI to configure your system. If you cannot access the web-GUI, please use Tiny OAM 0.3.1.0 for configuring IP settings.
If the previous OAM Administration Program is used, it might corrupt the configuration.

Added or Changed Features

- In order to reduce the number of NTP notice messages:
  - Increase the amount of time that can be adjusted from 400msec to 1sec.
  - Increase the minimum delay for the NTP response filter from 10msec to 20msec.
- Do not generate a new Cisco Unified Communications Manager SEP number when changing the username.
- New Digital RFP firmware PCS16B_ embedded.
- Add an additional SRTP SDP crypto line without MKI if MKI is enabled and update local MKI setting from remote.
- Lync DNS based auto discovery of the frontend server / pool. This changed the DNS procedures to be more like the Microsoft client, allowing for easier integration into a Lync environment.
- When Lync is enabled an Accept-Language header will be added to outgoing INVITE messages, allowing remote endpoints like voicemail and IVR to select which language to use. The value sent is controlled by the Phone Language setting.
- The SIP registration and re-registrations time interval now includes a random portion to help reduce the load of a server after restarts etc.
- Lync DNS based auto discovery of the frontend server / pool. This changed the DNS procedures to be more like the Microsoft client, allowing for easier integration into a Lync environment.

- Added special behaviour status (absent in charger / multi-charger) to service repor

Known limitations

- Shared lines (Handset Sharing) and internal switching cannot be combined.
- 3rd party call control does not work on handsets configured as DECT-to-DECT.
- Be aware: The 74 series handsets do not have the features required to have full SIP call handling functionality; ex. tones and display text will be missing in some situations.

Corrections and Improvements

- Fix dectesc-660: Was not able to replace a handset from a master handset subscribed to an analogue line.
- Remove memory leak when a SIP redirect loop is detected and remove logging noise.
- Remove buffer overflow when receiving very large DNS results.
- Fixed error, when a Text call is received shortly after an outgoing message (like Charger In/out).
- Fixed error in update of handsets over the air. In a scenario where a base station restarts during handset update and the user start dialling a number immediately after the handset is released can cause a lockup situation for this specific handset. The handset will not be able to receive or make calls afterwards. The only way to bypass that was by manual rebooting the DECT server.
- Download of service report could some time stop unexpectedly. When the number of entries in the RFP handover statistic list exceeded an internal level.

Version PCS16__, 2016 - Q1

For this and future software releases, use only the web-GUI to configure your system. If you cannot access the web-GUI, please use Tiny OAM 0.3.1.0 for configuring IP settings.
If the previous OAM Administration Program is used, it might corrupt the configuration.

Added or Changed Features

- It is now possible to set the Microsoft Lync presence status using the handset. The presence status will be sent to the Lync infrastructure and will affect the status on all connected devices for the Lync user. If a manual change of status is performed on the Spectralink handsets or on the Lync Client (e.g. manually setting the status to DND) this will be reflected on the Spectralink handset with a yellow/red/green icon. An automatic change of status on the Lync Client (e.g. making a phone call) will not be reflected on the Spectralink handset.
  Setting the presence status requires the newest handset platform and is supported on Spectralink 7522 handsets and on the upcoming 7532, 7622, 7642, 7722, 7742 handsets. It is not supported on the 7480, Butterfly and upcoming 7202 and 7212 handsets.
- The DECT Server is now able to join calls into a bridged ad hoc conference on Cisco Unified Communications Manager. The feature is available when two calls are active on the handset and is invoked by selecting “Join in conference” from the call Options menu. When the conference is established further participants can be added by adding a new call and joining it into the conference. Ad hoc conference requires the newest handset platform and is supported on Spectralink 7522, 7532, 7622, 7642, 7722 and 7742 handsets. It is not supported on the 7480, Butterfly, 7202 and 7212 handsets. Furthermore, the “Cisco Unified CM Enhanced features” license is required.
- Automatic Lync presence bootstrapping is now supported. Lync requires bootstrapping of presence for a user in order for a device to be able to publish presence. In previous releases of the firmware automatic bootstrapping was not supported and the administrator had to issue Power Shell commands or login with a Lync client in order to enable presence. Starting with this release the DECT Server will automatically perform this bootstrapping and no manual steps have to be taken in order to activate presence.
- The DECT Server now supports internal/external ring-patterns on Cisco Unified Communications Manager. CUCM supports using specific Alert-Info headers to indicate whether a call is internal or external. These headers are now parsed and recognized by the DECT Server. Spectralink handsets have two variants of each ring-pattern making it possible to hear whether a call is internal or external. The symbol for an incoming call will also reflect whether the call is internal or external.
- The handling of SUBSCRIBE NOTIFY signalling has been improved when connected to a Microsoft Lync server. The DECT Server now allows Lync to piggyback the first NOTIFY with the 200 OK for SUBSCRIBE. Furthermore the BENOTIFY request is now supported. BENOTIFY is best-effort NOTIFY, which is a NOTIFY that is sent without a transaction and which does not require a response. Both improves scalability as the amount of signalling required is reduced.

- The handling of SIP registrations has been improved when connected to a Microsoft Lync. The DECT Server now supports Lync keep-alive signalling. When SIP registrations are sent from the DECT Server it indicates to Lync that it supports Lync keep-alive, and extracts the keep-alive timeout from the register response and sends keep-alive signals at the rate specified by Lync. Also, when a REGISTER is sent to Lync the DECT Server will omit the "Expires" header from the REGISTER because Microsoft recommends that. When Lync keep-alive is used the re-register period is increased from 900 seconds to 7200 seconds by the Lync server. However, the DECT Server still has a default re-register period of 3600 seconds, which will be used unless configured otherwise.
- Change the SIP signalling with Cisco Unified Communications Manager to be more like the Cisco SIP Phones. This was required in order to be able to establish an ad hoc conference. The CUCM now uses remote call control and sometimes sends “playtonereq” and “statuslineupdatereq” instead of standard SIP responses. This is for example the case when calling a busy user. The DECT Server interprets the requests and plays a tone or displays the status text.
- Update the default CA bundle. This updates the list of Certificate Authorities known by the DECT Server, which ensures that the system knows new ones and insecure ones are removed.
- Several open source packages included in the firmware have been updated in order to keep track with upstream development to get improvements and security fixes.
- The Linux kernel for the media resource card and CPU-card-II (CPU cards after September 2012) has been updated to version 4.2.3.
- Use SIP Warning header in the log message if a transaction fails. Cisco Unified Communications Manager puts additional error information in the Warning header. This information is now added to the log message to make debugging easier.
- Add an additional SRTP SDP crypto line without MKI if MKI is enabled and update local MKI setting from remote. This is to improve the ability to negotiate SRTP encryption with various SIP endpoints. Specifically, this enables successful negotiation with the Pidgin client when connected to Lync.
- Added new information’s to service report
  - Microsoft LYNC settings.
  - Default IP configuration of IP base stations.
  - External antenna on IP base stations.
  - Language settings.
  - Cable delay values on pair 2 (8 channel base station).
- Allow and disallow bearer handovers to repeaters.
  Repeaters may have their eligibility in some situations. In general base stations are preferred whenever possible. “Allow bearer handovers to repeaters” is the default setting. With the default setting, the handset will know if rpn's are 64 apart it has to make bearer

handovers. That is why a repeater connected/repeating from base station RPN 0 must have/emit RPN 64. (Repeater on base station 1 must have 65 and so on...). In larger systems with more than 64 base stations, it is essential that base stations with an RPN offset of 64 not can be "seen" from a handset in any position. In theory this is easy, but in practice we have experienced more cases, where this has been an issue. The consequence of the "64 offset" issue is typically lost calls and/or noisy handovers. The handset will try to make bearer handover between two base stations – this will newer work. Disabling bearer handovers to repeaters means that the handset will always make connection handovers (like normal handover between two base stations).The"64 offset" issue is now out of the equation. Repeaters can still be connected, but it is NOT recommendable. Handovers will be slow, and probably noisy.

- The statistics web GUI has been extended with statistics for devices. Every time a voice call is released Spectralink handsets supporting statistics, will send statistics to the server, which stores the information. The following statistics are available in the web GUI and in the service report.
    - Bearer handover cancelled. The percentage and the number of succeeded / started bearer handovers.
      A bearer handover is when the handset changes channel (frequency/timeslot) but stays on the same base station. This is typically performed if the handset detects disturbances on the current channel.
    - Connection handover cancelled. The percentage and the number of succeeded / started connection handovers.
      A connection handover is when the handset changes base station. This is typically performed if the handset detects a better base station than the one it is currently on.
    - Call setup failed. The percentage and the number succeeded / started call setups.
      An outgoing call setup can fail for different reasons. Some examples are: If the radio connection drops while establishing the call or if the user presses on-hook/off-hook quickly.
    - Text setup failed. The percentage and the number of failed outgoing text call setups.
    - Out of coverage. The number of times and the duration in hh:mm the handset has been out of coverage.
    - Voice muted. The number of times and the duration in hh:mm the handset's voice path has been muted.
    - Frame errors. The percentage of voice/data packets with frame errors detected while in an active radio connection.

Known limitations

- Shared lines (Handset Sharing) and internal switching cannot be combined.

- 3rd party call control does not work on handsets configured as DECT-to-DECT.
- Be aware: The 74 series handsets do not have the features required to have full SIP call handling functionality; ex. tones and display text will be missing in some situations.

Corrections and Improvements

- DECTESC – 628: In some situations a master handset was not working as expected.
- DECTESC - 643: Fix attended transfer to unreachable or busy number for non-NG handsets. If an attended transfer was attempted with a non-NG handset and the transfer target was unreachable or busy it was not possible to return to the original call or attempt a new transfer.
- DECTESC-641: Customer experience intermittent call disconnection. Problem fixed.
- When using trace filter via web GUI (Diagnose | Trace | Trace filter), it is now possible to see both the too and from base station in a handover scenario.
- Set the current peer when completing ICE, this eliminates one-way audio problems in some call forking scenarios.
- An error in surveillance of cable delay values on BIF02 cards (Base station interface cards with only 2 ports) has been corrected. The values was previously not shown.
- System could crash if a handset assigned to an analogue line dialled a number longer than 36 digits.

Version PCS15J_, 2015 - November

For this and future software releases, use only the web-GUI to configure your system. If you cannot access the web-GUI, please use Tiny OAM 0.3.1.0 for configuring IP settings.
If the previous OAM Administration Program is used, it might corrupt the configuration.
This version is based on PCS15H_.

Added or Changed Features

- Due to an error in a production test of the base station (RFP6), an EEPROM setting was left with a wrong value. The invalid value caused the base station to transmit with 10 dB less power. When a base station (RFP6) starts up, the DECT server will modify this EEPROM setting if necessary. The DECT server will automatically reboot the base station if EEPROM has been modified.

Version PCS15H_, 2015 - Q4

For this and future software releases, use only the web-GUI to configure your system. If you cannot access the web-GUI, please use Tiny OAM 0.3.1.0 for configuring IP settings.
If the previous OAM Administration Program is used, it might corrupt the configuration.
Added or Changed Features

- Provide DECT Server generated Music-on-Hold (MoH). This adds the ability to play music when a call is put on hold on a SIP server that does not provide MoH itself. This is for example the case with Microsoft Lync, which relies on the endpoints to generate MoH. The feature is enabled by uploading a music file to the DECT Server via the web GUI at Configuration | SIP | Call status | Provide Music-on-Hold. The music file must be an 8 kHz, 16-bit, mono WAV file with a maximum size of 7MB. If MoH is also provided by the SIP server the MoH provided by the DECT Server might conflict and the behaviour is unpredictable.
- Radio/LAN Gateway functionality added to IP-base station web GUI.
  Support for a new type of base station synchronization has been added. If configured as Radio/LAN Gateway an IP base station will use an over-the-air synchronization as the source of synchronization and act as a LAN sync master. With this type it is possible to add a LAN synchronized segment of IP base stations to an existing synchronization chain of base stations. As an example it is now possible to augment an existing DECT Server 8000 system with digital base stations with a segment of IP base stations which use sync over LAN.
- In web GUI (Base Station | Digital Base Station) a description field has been added. The description field is meant to store a physical location description for each digital base station.
- Added and changed SIP stack:
  - Improve handling of SIP transaction timeouts. The default client transaction timeout has been increased from 4000ms to 16000ms. This is to be more forgiving to SIP servers and endpoints that sometimes fails to respond quickly. The increased timeout will make the communication more robust but at the price of prolonged detection of failure. If a faster detection is required the timeout can be decreased by changing the setting Configuration | SIP | Client transaction timeout.
  - Starting with this Firmware the DECT Server will re-register earlier to make room for the increased client transaction timeouts and still maintain a valid registration to a secondary SIP server in a failover situation. The Re-register timeout now depends on the client transaction timeout.
  - On timeout only blacklist if SIP request is INVITE outside a dialog or REGISTER and no response is received at all. Previously the blacklisting of a SIP proxy could be too effective and blacklist a server due to a missing response from another endpoint. This was observed at an external site with a large CUCM deployment.
  - Respond 481 to SUBSCRIBE for dialog state if no dialog exists. Previously a new dialog was created but this caused the CSeq number to be inconsistent.

  - Improve detection of transport errors when connecting to SIP servers. The SIP stack did in certain situations not immediately detect transport errors while connecting. This is fixed.
  - On registration expire and connection failure do not issue a re-register if a register is pending. Before this change a new useless REGISTER was put in the queue and send after the pending registration was finished.
  - The DECT Server now supports adding a ms-endpoint-location-data header to SIP INVITE in a Microsoft Lync configuration. The remote peer can utilize this to do bandwidth management according to our network location. This was reported in DECTESC-595 in a Microsoft Lync setup. The Ms-Subnet header is now included in REGISTER messages to Lync. This enables support for Location Based Routing by the LYNC server.
  - The supported G.726 codec is now advertised and displayed as AAL2-G726-32. This allows for low bandwidth consuming calls between a Lync Client and Spectralink DECT, as the Lync Client recognizes and supports this codec.
  - Improvements to the SIP dialog handling for the dialog event package, RFC4235, which is used by CUCM and others to monitor the endpoint call state. These improvements are to better handle creation, deletion and inconsistencies in the SIP dialog state and avoid error messages in the log.
  - In the case of a Lync trusted server configuration, the DECT Server now supports fail back to the primary proxy when registered with the secondary proxy. Previously the DECT Server would stay with the discovered proxy server when re-registering as Lync trusted server. With regard to registration Lync behaves differently when configured for NTLM or trusted server. When configured for NTLM the client is redirected to the home server and it must keep registering towards this one. When configured for trusted server any Lync server will accept the registration and the request is routed to the home server internally within the Lync environment.
  - Improve handling of receiving DTMF tones with SRTP enabled. When SRTP was enabled the incoming RFC2833 DTMF events affected the jitter buffer and the overall sound quality of the call could be degraded.
- New RFP6 firmware (PCS15L_) has been embedded in this version.
- RFP6 base stations (with firmware PCS 15L_ or newer) with an external antenna connected can via the web GUI be configured in 3 modes.
  - Use internal & external antenna (default setting).
  - Use internal antenna only.
  - Use external antenna only.
- Use a more efficient encoding of phase of neighbour base stations when sent from the base station to the DECT Server. This introduces an incompatibility that will not allow older revisions of DECT Server firmware to receive phase information from PCS15D_ or newer revisions of IP base station firmware. The Offset column in the RSSI map available when

selecting a base station in the Base Station | IP Base Station web GUI, will in this case be empty.

- Changes in service report:
    - To the list of foreign and own detected base stations a warning have been added if a duplicated RPN is seen, and if a RPN with an offset of 64, 128 or 192 is seen.
    - Max number of active connection on each base station has been added to statistics.txt in service report.
- A new tap Users | Phones has been added to the web GUI. New tap is showing only handset related information like IPEI, HW PCS, model number, product name and SW PCS.
- In the service report, printout regarding bad communication on 2nd pair on 8 channel digital base stations has been added.

Known limitations

- Shared lines (Handset Sharing) and internal switching cannot be combined.
- 3rd party call control does not work on handsets configured as DECT-to-DECT.
- Be aware: The 74 series handsets do not have the features required to have full SIP call handling functionality; ex. tones and display text will be missing in some situations.

Corrections and Improvements

- DECTESC – 623: Spontaneous restart of digital base stations (RFP6) solved in new RFP firmware (PCS 15L_).
- DECTESC – 618: Issue with MSF not delivered to shared handset fixed.
- DECTESC – 614: Buffer overflow in PCS 15B_ fixed, it was not an error that caused malfunction.
- DECTESC – 610: Issue with call transfer failing using shared handsets solved.
- DECTESC – 621: If the remote end sends a Remote-Party-ID containing an invalid URI (specifically an URI without a host part which is not allowed as per RFC3261), the DECT server could crash. Now an invalid Remote-Party-ID header is ignored.
- Updated help text and mouse over in WEB GUI.
- DECTESC – 617: When a subscribed handset is powered on a LOCATE_ind is send to the server. The server responds with a LOCATE_res. The length of the LOCATE_res signal is reduced for all other than PP7 and newer handsets. The change is implemented as result of problems with customer 3rd party DECT handsets.
- DECTESC – 619: The supported G.726 codec payload type is now offered dynamically.
- In a special situation where an outgoing and incoming MSF are crossing each other; sometimes it was not possible to send keypads from handset to third party application.

- If multiple licenses were loaded and one of them was "Cisco Unified CM". The "Cisco Unified CM" was overwritten, and not shown in the summary feature list. Also the extra features associated with "Cisco Unified CM" was not shown in the web GUI.
- When sending MSF to more than one shared handset on the same extension could cause writing outside allocated buffer. The server behaviour afterwards was unpredictable. In a similar situation a MSF instance was not cleared and handsets could not receive MSF's.
- If an external antenna is connected to an IP Base Station it is visible in web GUI for IP Base Station.
- Possible buffer overflow error corrected. If Company Name (Installation | Company Info) is added and system size was 3 or more shelves, the total length could be too large.
- An error in using click-to-dial events has been corrected. The DECT server did not react on a received click-to-dial event containing the special ringing pattern 0x42 (knock-knock). The event is typical received in off-hook state.
- Added handset model number and product name to "mouse over" in web GUI.

Version PCS15F_, 2015 - Q3

For this and future software releases, use only the web-GUI to configure your system. If you cannot access the web-GUI, please use Tiny OAM 0.3.1.0 for configuring IP settings.
If the previous OAM Administration Program is used, it might corrupt the configuration.

Added or Changed Features

- The server part of supporting LAN based synchronization of the IP DECT base stations has been implemented. A license is required in order to enable this new feature. If the beta release of LAN sync is used and this new firmware is installed the synchronization will be reverted to be radio based. Therefore it is highly recommend to load the license before upgrading the firmware.
- The DECT Server can now be configured to SIP de-register the associated user, when a handset is powered off (the handset tells the server it is powered off by sending a DETACH to the server). This is controlled by a new setting "Handset power off action", available in Configuration | SIP | General. Default for this setting is to ignore the power off event and keep the registration. The handset power off state is not persisted in flash, and after a reboot the DECT Server will assume that all handsets are powered on. The server will learn the state when handsets are powered off.
- In DECT Server WEB GUI, the table in menu Base Station | IP Base Station has been expanded with 2 columns - hardware PCS and serial number. Furthermore offset values has been added to the configuration menu of the individual IP DECT base stations (IP base station Configuration menu is reached by left clicking on the desired row).

- DECT Server WEB GUI main and sub-menus has been moved to new locations. The new GUI menu/sub-menu change has been made to meet future features and more logical and systematic design.

Known limitations

- Shared lines (Handset Sharing) and internal switching cannot be combined.
- 3rd party call control does not work on handsets configured as DECT-to-DECT.
- Be aware: The 74 series handsets do not have the features required to have full SIP call handling functionality; ex. tones and display text will be missing in some situations.

Corrections and Improvements

- Make the LDAP reader of the central phonebook more robust. If the LDAP server returned some unexpected data the central phonebook could crash the DECT Server immediately after saving the configuration. This issue is related to DECTESC-581. In rare instances the actual use of the phonebook could also crash the DECT Server if phonebook was used. Therefore it is highly recommend to use this firmware when LDAP phonebook is being used.
- DECTESC-601 has been fixed. The bug caused that a present and available handset sometimes was reported as not found internally and SIP stack reported "481 Call/Transaction Does Not Exist" to the calling party. The risk of experiencing this error was higher when multiple user types was used (SIP, Analogue or Dect to Dect). This error was introduced in PCS15D_ (previous GA release).
- Call control event - The sub-event SendDigitsReq was not working properly. The mute bit was not working in all situations.
- Import and export of certificates could sometimes fail from a GUI perspective. The GUI feedback is now corresponding to the requested action.
- SIP stack has been updated. (General SIP features and integration with Cisco Unified CM, Microsoft Lync and other has been improved).
- In regards to internationalized system messages sent to handsets an error has been found when using Cyrillic characters in the server-to-handset character encoder. Problem has been fixed.
- Fixed issue with display of call duration. Sometimes either in the trace window or in the service report the duration of a short call could be measured to several hours instead of maybe 2 minutes. As a consequence the total call duration of calls system wide was also calculated wrong.
- If a shared handset was replaced via a master handset, the shared group number was not replaced correctly. This caused that shared handset could not be used without correcting the group numbers via the WEB GUI.

- A shared handset could not be called right after it has logged in. The "window" for not being reachable was between 1 - 5 seconds. This "window" is now reduced 100 - 500 ms.
- Service reported adapted to LAN base stations and other minor corrections has been made.
- In WEB GUI menu Installation | Network | Date & Time the user was sometimes met by an error message when trying to save NTP server (both as name or IP address). Other times there were no error message, but the feedback in the “NTP server” text field could be rubbish. This has been fixed.

Version PCS15D_, 2015 - Q2

For this and future software releases, use only the web-GUI to configure your system. If you cannot access the web-GUI, please use Tiny OAM 0.3.1.0 for configuring IP settings.
If the previous OAM Administration Program is used, it might corrupt the configuration.

Added or Changed Features

- The DECT server now supports internationalized system messages sent to handsets. The status and error texts send from the DECT Server to the handsets are available in Danish, Dutch, English, French, German, Italian, Norwegian, Portuguese, Russian, Spanish and Swedish. This is to give a better user experience for the end users. The language can be configured on the “System | Configuration” Server GUI page. This does neither replace - nor affect - the handset language configured locally in the phone.
- The embedded RFP6 firmware in the DECT server has been updated to the Q2-2015 release - PCS15C_.
- New and more intuitive web GUI layout for updating handset over the air and updating wired base stations. Previously it was only possible to update Butterfly handsets, now two further handset types are possible (which is not yet released).
- The DECT server is now able to receive handset product and model number during location registration (when a handset is powered on). A new facility in coming handsets.
- When creating or editing a user it is now possible to set special behavior on incoming call when in single or multi charger. If checkbox is selected all incoming call is handled as the handset were detached when the handset is placed in a charger.
- Add endpoint_partiel_clear function to XML_RPC interface. The function is used to clear messaging and calling lists in the handset.

Known limitations

- From this software version and forth a backup of user data can not be restored on previous software versions (older than PCS15D_).
- Shared lines (Handset Sharing) and internal switching cannot be combined.
- 3rd party call control does not work on handsets configured as DECT-to-DECT.
- Be aware: The 74 series handsets do not have the features required to have full SIP call handling functionality; ex. tones and display text will be missing in some situations.

Corrections and Improvements

- DECTESC-569, DECTESC-563 and DECTESC-553, DECTESC-581 & DECTESC-576 has been addressed.
- The XML-RPC function "endpoint_heartbeat_config_callback" has been made more robust to different 3.rd party implementations of the interface.
- EMD password now supports special characters.
- Add support for SIP username up to 63 characters. Previously only 24 characters. Maximum length of XML-RPC password has been changed from 12 to 31 characters.
- DECTESC-582: Shared handset - faulty assignments will be deleted upon startup and write to database.
- Master handset - corrected error that caused system to crash - when an IPEI for a shared handset was changed from a master handset, a vital process in the server was restarted. The consequence was base station reboot, lost calls and no contact to WEB GUI for up to 30 seconds. Furthermore the IPEI change executed from the master handset was not completed.
- Added extra trace message when alien handsets are making location subscription attempts.
- Fixed issue with faulty error message in log during startup of many wired base stations.
- Fixed problem in WEB GUI not being able to show timeslot 'Follow RFP'. (Error introduced in PCS15A_) Only relevant for base stations RFP4 and RFP 5 ("Even" timeslot was incorrectly shown instead).
- When SIP user credentials are changed (SIP username, SIP Auth username or SIP Auth password) a new SIP registration is now performed. Previously this was only done when "service status" was changed from "Disable" to "Allow".
- Added counter for call to detached handsets in statistics.
- A handset assigned to an analogue line with no physical interface card present will after off-hook show the warning text "AB card is not present". This is typically a wrong configuration. In some special configuration the warning text was overwritten by the handsets presentation text.

- In a case of two consecutive incoming analogue calls, the called handset would previously only show clip from the first call. The handset will now show clip from the latest call.
- Service report improvements
  - Problem with reading IP status from more than one MR32 card solved by implementing a queue system.
  - readme.txt updated
  - Introduced PP7 product name and model number
- Handling of analogue interface cards was previously consuming a lot CPU cycles. This has been changed to a minimum.

Version PCS15A_, 2015 - Q1

For this and future software releases, use only the web-GUI to configure your system. If you cannot access the web-GUI, please use TinyOAM 0.3.1.0 for configuring IP settings.
If the previous OAM Administration Program is used, it might corrupt the configuration.

Added or Changed Features

- Possibility to restore the system configuration file (config.xml) via WEB GUI (System / (Restart/Reset/Restore)). The configuration file is available from the service report. Previously settings like “mail info”, “company info”, “max RS232 level” and other settings where stored in separate files. All these settings are today included in the configuration file.
- Handset sharing has been improved with more feedback to the user. Possibility to login/logout via “long press” of key 8 on the handset. When handsets are logged out all voice “message waiting” indications in the handset are deleted (LED and icon). This improvement also covers the DECTESC-557.

Known limitations

- Shared lines (Handset Sharing) and internal switching cannot be combined.
- 3rd party call control does not work on handsets configured as DECT-to-DECT.
- Be aware: The 74 series handsets do not have the features required to have full SIP call handling functionality; ex. tones and display text will be missing in some situations

Corrections and Improvements

- Improving performance. The response time when reading many users or base stations has been improved significantly. In a linked system with old CPU’s its also easy to see improvements of response time.

- Previously there have been some instances with handling of special characters in the WEB GUI. Also some cases of not supporting ciphers, alphanumeric and special characters in an input field. All this has been fixed in this release. This fix also covers DECTESC-550 (handling of special characters in LDAP password).
- New features in TAP interface. It is now possible via WEB GUI to add trailing “pound digit” in call back numbers and set max load of simultaneous MSF.
- When handsets are detached (powered down in coverage area), they will no longer be paged in a incoming voice or MSF call. The caller will immediately know that the user is not available. For MSF the “caller” is typical a 3rd party application.
- The DECTESC-559 – “Can not parse a SIP OK message from an Avaya SM” has been fixed.
- An error with handling pause digits in a called number on the SIP interface has been corrected. Error was introduced in 2014-Q2 release.
- Fixed bug with interpreting call back number in an incoming MSF if the number starts with ‘sip:’ or ‘sips:’.
- When importing a corrupted CSV file a process in the server was restarting, and resulting in no contact to WEB/EMD interface for several seconds. This issue has been addressed.
- General Web GUI improvements.
- SEP numbers (special Cisco identification numbers “Selsius Ethernet Phone”) is added to service report.

Version PCS14C_, 2014 - Q4

For this and future software releases, use only the web-GUI to configure your system. If you cannot access the web-GUI, please use TinyOAM 0.3.1.0 for configuring IP settings.
If the previous OAM Administration Program is used, it might corrupt the configuration.

Added or Changed Features

- Support for RFP6 generation base stations. The latest Digital base stations (successor for RFP5) come in a 4 and 8 channels variant. Firmware upgrade from the server is off course also possible as in previous base stations. Info about RFP6 version (4 or 8 channels) is available through the WEB GUI. If an external antenna is connected it will be shown in WEB GUI. Also if only 1 pair of wires out of the 2 pairs is connect on an 8 channel RFP, this will be shown as well.
- System message logs are now available from the WEB GUI in the “Diagnose” tab. System logs contains historical information since the last reboot of the system.
- Changed max possible IP base stations from 256 to 1024.

- Support of Multicast (one-to-many or many-to-many distribution) is implemented because number of possible IP RFP has been increased.

Known limitations

- Shared lines (Handset Sharing) and internal switching cannot be combined.
- 3rd party call control does not work on handsets configured as DECT-to-DECT.
- Be aware: The 74 series handsets do not have the features required to have full SIP call handling functionality; ex. tones and display text will be missing in some situations

Corrections and Improvements

- DECTESC-543: VLAN setting on MR32 card is now read correctly back to user interface.
- DECTESC-544: Problem with importing users via CSV file, if comma in stand by text. Issue fixed.
- Previously during the handover phase it was sometimes possible to hear short interruptions/cracks in the voice path during a conversation. These interruptions/cracks has now been reduced to a minimum.
- Backplane and CPU production ID can now be read in a system with no interface cards.
- Shared handset is working again with SIP extensions.
- Allow an empty name field in a imported CSV file.
- Miscellaneous WEB GUI layout improvements.
- When a "switched off" handset is called, the log would previously have a lot of extra entries and the statistic showed several incoming calls - (roughly one incoming call is counted per ringback tone in the calling line). This only applies to handset subscribed to an analogue line.

Version PCS14B_, 2014 – Q3

For this and future software releases, use only the web-GUI to configure your system. If you cannot access the web-GUI, please use TinyOAM 0.3.1.0 for configuring IP settings.
If the previous OAM Administration Program is used, it might corrupt the configuration.

Added or Changed Features

- Major update of SIP stack, SIP application and Media Resource handling. Aligned with DECT Server 6500 release 2014-Q2.
- Integration with Cisco Unified CM has been improved significantly. The DECT Server can now be connected to the Cisco Unified CM as a known phone type instead of just being a third party SIP device.
  This gives handsets connected to the DECT Server access to additional features not supported for third party SIP devices:

- Music-On-Hold: The handsets can put the remote party on hold with Music-On-Hold.
- Call pickup: The handsets have access to various kinds of call pickup by dialling feature codes.
- Meet-Me Conferencing: The handsets can initiate a Meet-Me based conference by dialling a feature code.
- Call Forward Unconditional: Call Forward Unconditional is controlled within the Cisco Unified CM instead of locally in the DECT Server. This means that if Call Forward Unconditional is enabled from the DECT handset other devices sharing the same line in the Cisco Unified CM will also be forwarded. Similarly if Call Forward Unconditional is enabled on a shared line device it will be displayed on the DECT handset.
- Furthermore, administration of many DECT handsets on a Cisco Unified CM has been improved. The DECT Server now supports exporting CSV files, which can be imported directly into the Cisco Unified CM Bulk Administration.
- The improved integration requires a COP file to be loaded into the Cisco Unified CM and a license to be loaded into the DECT Server. Please refer to the Cisco Unified CM integration application note for further details. If no license is loaded into the DECT Server it can still be connected to the Cisco Unified CM as a third party SIP device and nothing is changed.

Known limitations

- Shared lines (Handset sharing) will NOT work with SIP extensions in this version.
- Shared lines (Handset Sharing) and internal switching cannot be combined.
- 3rd party call control does not work on handsets configured as DECT-to-DECT.
- Be aware: The 74 series handsets do not have the features required to have full SIP call handling functionality; ex. tones and display text will be missing in some situations

Corrections and Improvements

- Phonebook configuration via WEB GUI now allows strings up to 256 bytes (previously it was 64 bytes).
- Number of allowed Media resource cards has been changed from 32 to 64.
- Display trace/status from provisioning process is now being sent to trace window in WEB GUI.
- WEB GUI button layout has been changed. When a button is clicked it changes to a yellow colour (waiting for action to be executed). Depending on the result, the colour will change to red or green. Previously the user had to click an "OK" button in a separated confirm box. The main reason for this change has been to reduce the number of mouse clicks.
- Layout of WEB GUI tables has been changed. The new table design provides a more intuitive "look and feel" and has excellent search and sort functions.

- DECTESC-525: When an IP DECT base station is disconnected from the system it previously showed as connected in the WEB GUI for about 15 minutes. This time is now reduced to typically 30 seconds (and up to 120 seconds). The reason for this delay has to be found in the way the used Linux kernel (2.6.34.1) is working.
- DECTESC-526: With newer handset firmware, the handset keeps the connection open after sending a PPStatusInd and rely on the DECT Server / third party application to release the connection. This does not work well with old Messaging applications. To comply with this changed behavior the DECT Server releases the CISS connection immediately, if OldKWS1500 or TAP1.8 protocol is chosen on the RS232 port.
- Added support for dead man button on 7640 & 7740 handsets.
- Mouse over and help texts in WEB GUI has been improved.
- DECTESC-508: Fixed issue with incorrect time stamp in the beginning of some log files.

Version PCS14A_, 2014 - Q2

For this and future software releases, use only the web-GUI to configure your system. If you cannot access the web-GUI, please use TinyOAM 0.4.1 for configuring IP settings.
If the previous OAM Administration Program is used, it might corrupt the configuration.

Added or Changed Features

- Now getting P-ID from handsets in both subscription and location registration. P-ID is written into service report.
- Added XML-RPC facility to read firmware part number & PCS from all handsets.
- Broadcast group number read from handsets and written into service report.

Known limitations

- Shared lines (Handset Sharing) and internal switching cannot be combined.
- 3rd party call control does not work on handsets configured as DECT-to-DECT.

Corrections and Improvements

- DECTESC-523: Solved issue with provisioning. Now check for duplicate local number, duplicate username and duplicate LID.
- Changed (factory default) default user name & password for the WEB interface to admin, admin. (XML-RPC credentials are unchanged.)
- DECTESC-510: Some problem with unpacking TCPdump (pcap) have been solved.
- DECTESC-518: Provisioning of StandByText is now working.
- DECTESC-516: Sound switching error on RFP3 (error introduced in PCS13B_) solved.
- WEB GUI: Function to find next vacant analogue port is working.

- Moscow NTP error solved. Changed from UTC+3 & daylight saving to UTC+4 & permanently
- WEB GUI: Error in function number in MSFFunction solved.
- Let MSFKeyPadInd get though in TEXT CALLs, even that the TEXT CALL has not been acknoledge by PP user. This can give the 3rd part application a "reason" if user is pressing ex. the hook key.
- Import of users can now also be done from comma seperated file, and still from semicolon seperated files). Improved trace feddback when importing users from *.csv file.
- Added total number of suppl. service calls to text in email.
- Timeout on getting RFP increased from 5 to 120 seconds.
- In WEB GUI docs: Added specs for PpNvData-Req &-Cfm.
- In WEB GUI docs: Added SK & PU bit to SetupSpec2 in desription of MSFSetupReq.
- Now saving and reading the pause (digit) multiplier, which can be set from command mode.
- Improved mouse over and help texts.

Version PCS14__, 2014 - Q1

For this and future software releases, use only the web-GUI to configure your system. If you cannot access the web-GUI, please use TinyOAM 0.4.1 for configuring IP settings.
If the previous OAM Administration Program is used, it might corrupt the configuration.

Added or Changed Features

- The PCM clock system has been redesigned.
  Previously, the clock system could cause different problems with the voice path in the DECT Server 8000. Typically, there was no audio or bad audio (noisy and crackling sounds). The problems are hardware dependent but not related to hardware editions (PCS). With special selected cards and DECT Server 8000 shelf (8000 shelf almost full), the issues are easier to reproduce but still typically require several restarts to get a bad situation. The problems are potentially on all cards (Base Station Interface, Analogue Line interface, Media resource and CPU cards).
  The redesigned PCM clock system also seems to give an improved handover performance on some installations.
  The corrections are implemented in the main FPGA on the CPU card and on the FPGAs located at the interface cards. The FPGA redesigns are embedded in the quarterly release and will automatically update the FPGAs. Refer to JIRA cases DECTESC-494, DECTESC-499 and DECTESC-501.
- The card hot plug has been improved. Previously, it failed frequently, especially when reseating Analogue line cards.

Known limitations

- Shared lines (Handset Sharing) and internal switching cannot be combined.
- 3rd party call control does not work on handsets configured as DECT-to-DECT.

Corrections and Improvements

- Improved MSF statistics (especially release indication reasons).
- Improved statistics if no Media Resource are present or all channels are busy.
- Off hook dialing was not possible for SIP users on 74-Series handsets. (bug introduced in PCS13E_) DECTESC-505.
- Handling of pause digits in a SIP user scenario has been improved. Previously, the server waited one second per pause digit. Now, the server waits until the user is connected before it sends the remaining digits. After the call is connected, remaining pause digits still correspond to approx. 1 sec pause.
- A command mode menu for enable/disable edit of users via RS232 has been added. This is to avoid user database mismatch if somebody accidentally creates users via the OAM Administration Program.
- Corrected error that caused a hanging XML-RPC instance when using EMD Call Control events.
- General improvements of the web-GUI:
  - Error in application demo - MSF format III demo, an empty “DN” field was not completely empty (an ‘-‘ (0x2D) was wrongly send). Also, the “Area Receiver Identifier” field was always sending a ‘0’ (0x30) even though it was supposed to be empty.
  - Added info about behavior of front LEDs in help text.
  - Updated help text regarding required license for frequency swap on IP-Base Stations.
  - In the Network menu, a host name can be entered for test purpose (Ping, Tracert or Nslookup). The size of the Host name field has been adjusted to 128 characters (previously it was only 16 characters).

Version PCS13E_, 2013 - Q4

For this and future software releases, use only the WEB-GUI to configure your system. If you cannot access the WEB-GUI, please use TinyOAM 3.0.0 for configuration settings. If previous OAM Administration Program is used it might corrupt configuration.

Added or Changed Features

- Rebranded to Spectralink DECT Server.
- Based on customer experience, a new service code has been implemented for reading HTTP port number using a subscribed Spectralink 7000-Series handset (***999*05). Service codes in Spectralink DECT Server 8000 and Spectralink DECT Server 2500

(summary)
***999*00 IP address
***999*01 MAC address
***999*02 Firmware PCS
***999*04 VLAN ID
***999*05 HTTP port number

- XML-RPC CallControl (voice) functions and events have been improved, and trace output has been added.
- Text length in normal MSF and MSF format III has been increased to 180 characters. If the total length (text+ call back number + control byte) exceeds the limit, then text is truncated and the last characters in text are changed to "…".
- Import and Export of users (csv format) have been significant improved with regard to stability, flexibility and feedback.
- User_data.csv in service report has all user data information. User_data.txt is no longer a part of the service report.
- Help texts in WEB GUI has been improved.

Known limitations

- Shared lines (Handset Sharing) and internal switching cannot be combined.
- 3rd party call control does not work on handsets subscribed as DECT-to-DECT.

Corrections and Improvements

- Now all licenses will be shown in WEB GUI, even if more than the first 5 licenses exist.
- IE8: Added missing button for updating firmware in KIRK IP Base Stations.
- Timeouts in WEB GUI are increased, so now export of users and generation of service report on systems with 4095 users works.
- Fixed length issue in EMD event - ReadUniqeProductionIdCfm.
- Fixed issue with wrong error text, when number of SIP users exceed limit.(DETESC-482)
- PCM clock drive on NEW CPU card type has been improved in software. A few shelves, equipped with many interface cards and the new CPU card, have had issues with periodic bad or missing voice on some cards.

Version PCS13D_, 2013 – August

During the last months, it has come to our attention that a few sites have experienced problems with installations with analogue interface cards. Most sites have been running smoothly, however, without any problems. We have received different end-user explanations for the same problem, and the service reports that we have received have contributed with technical facts and helped us find the root cause of the problem.

Corrections and Improvements

- A system with an analogue interface card could be brought in a fail mode where the only solution was to reboot the system or reset the faulty interface card. The problem only occurs if 3 or 4 users on the same connector start an outgoing call (simultaneously) and en-bloc dial a long number. When the error occurs, all 4 users on a connector are affected and are not able to make or receive any calls.

Version PCS13C_, 2013 - Q3

For this and future software releases, use only the WEB-GUI to configure your system. If the WEB-GUI cannot be accessed, please use TinyOAM 3.0.0 for configuration settings.
If previous OAM Administration Program is used it might corrupt configuration.

Added or Changed Features

- Support for IP-DECT base stations. Connecting IP-DECT base stations with a KWS8000/2500 is done very easily.

Connect an IP-DECT base station to same network as the DECT server. From WEB-GUI use “IP Base Stations Discovery” to find IP-DECT base station on network, click on link, Enter server IP address in IP-DECT base station and you are ready.

Please note, that the use of IP-Base stations requires a Media resource card.

IP-DECT base stations and DECT base stations can co-exist in the same system. Connection handover can be made between all kinds of base stations. An existing system with a number of DECT base stations can easily be expanded with IP-DECT base stations. IP-DECT base stations can advantageously use DECT base stations as sync source. It is also possible to update the firmware in the IP-DECT base stations via WEB-GUI. IP-DECT base stations must have Q3-2013 firmware or newer to work with KWS8000/2500.

The KWS8000/2500 must all ways have a media resource card installed to work with IP-DECT base stations.

The Media Resource card has 32 channels. Each connection on an IP_DECT base station uses one channel for the connection between DECT Server and IP-DECT base station and each SIP call also uses one channel for the connection between DECT server and IP-PBX.
One SIP call connected via an IP-DECT base station uses 2 Media Recourse channels.
One SIP call connected via a wired digital DECT base station uses 1 Media Recourse channel.
One single (analogue) line call connected via an IP-DECT base station uses 1 Media Recourse channel.

One single (analogue) line call connected via a wired digital DECT base station uses 0 Media Recourse channel.

- New Media Resource firmware that handles RTP security. When the Q3 firmware is uploaded to the KWS8000/2500, the server automatically updates the media resource cards in the system. A license must be obtained in order to use RTP security on the IP connection between server and IP-base stations.

- The WEB-GUI has been updated and improved to accommodate the new quarterly features and corrections. The changes are mainly focused on IP-DECT base stations and derived changes to statistics.
- Added support for new way of handling MSF in BGR139 scenarios.
- Import and export a user file in CSV format. The preconditions for importing a (comma or semicolon separated user) file to the DECT server is usually an empty KWS user database. In case the KWS user database is not empty, the import will fail if data content are overlapping. (for example, if handset x in the database has extension 100 and IPEI 00077 0012345, none of the “handsets” in the comma separated user file can have the same extension or IPEI). When exporting the KWS user database, the content is dumped in a semicolon separated file which will be downloaded via your browser.
- If a pause digit is detected in an outgoing SIP call, the first part is send in a CC_SETUP and the rest is send as CC_INFO. Every pause digit results in a 1 second pause before next part is send.
- Based on customer experience a new service code has been implemented for reading VLAN ID on a KIRK subscribed handset (***999*04).
  Service codes in KWS8000/2500 (summary)
  ***999*00 IP address
  ***999*01 MAC address
  ***999*02 Firmware PCS
  ***999*03 test
  ***999*04 VLAN ID
- Minimum provisioning time has been changed from 1 to 5 minutes.
- If Over The Air update is set to delayed start, the remaining time will now be shown every minute.
- Support for handset collected statistics and info. Some handsets are sending relevant statistics when voice calls are released. When a handset makes a location registration, extended handset info (production date, HW PCS….) is received. (Handset collected statistics and info are not fully implemented in all handsets).
- Added (forward and rewind) arrows in upper left corner in edit user page. An easy way to go back and forth in the user list without going to the main list every time.
- When creating a new analogue user, the KWS suggest the first vacant position (shelf, slot and channel) if an analogue line interface card is present in the system.
- Demo Apps page: Added new alert pattern type (Pager) in MSF Format III and Normal MSF.

Known limitations

- Shared lines (Handset Sharing) and internal switching cannot be combined.
- 3rd party call control does not work on handsets subscribed as DECT-to-DECT.

Corrections and Improvements

- Fixed error that caused a system with more than 30 analogue users to block users above PPID (index) 30. In WEB-GUI users appears as “no license” in the “Service Status” field. The costumer must manually change service status to “allow” on every user above PPID 30 before the handset is working correctly. Error was introduced in Q2 release.
- Synchronization between shelves has been improved and extra data added to RS485 statistics
- If payload type is not negotiated in a SIP call (it really should be), the KWS use the user-selected value of payload type.
- Corrected error that caused over the air not being able to work on analogue handsets.
- Back plane temperature below zero was previously shown wrong.
- When using Q2-release for handsets, handset sharing was not working, due to some new implemented security features in the KWS and handset (BGR 139). DECTESC-463.
- General improvements of statistic module, data protection and stability.
- Corrected rarely randomly occurring error that caused cross talk issues when a DECT-to-DECT handsets was involved. DECTESC-453

Version PCS13A_, 2013 - Q2

For this and future software releases, use only the WEB-GUI to configure your system. If you have to use the OAM Administration Program, use Software version 0.4.0.88. Please note if you create a new user or edit an existing user with the OAM Administration Program, the user database will be corrupted

If the WEB-GUI cannot be accessed, please use TinyOAM 3.0.0 for configuration settings.

Added or Changed Features

- Over the air update has been implemented. The function can be accessed from WEB-GUI. The WEB-GUI is built as a 5-step wizard where you can go back and forth, until you eventually start the Over the air update process. Among other things, you will be asked whether you want to upgrade an individual handset or all handsets, whether you want to upgrade now or later (up to 24 hours later), and finally how you want to load the system (many or few simultaneous updates). The number of possible simultaneous updates is based on the number of base stations and number of media resources available. Using Over the air update requires a media resource card present in the server.

So far, it’s only the Butterfly handset that supports Over the air update (from PCS13A). (Butterfly handsets have a hardware platform that supports dual flash partition to ensure that the new firmware can be CRC16 verified before the handset start up with the new firmware on the new partition). A Butterfly handset is updated in less than 5 minutes.

- The CPU cards will reconfigure after 90 seconds to the default IP address (192.168.0.1) after a power reboot if the network connector (RJ45 connector) is missing but only if the network configuration on the KWS is configured to receive the IP address from DHCP server.
- When the KWS is booting, the “internal switching setting” is checked. If it is set to “all local calls”, then it is changed to only switch internally when a “DECT-2-DECT handset” is involved.
- Base station ports can be enabled / disabled from the WEB GUI. Ports are default enabled. If a base station port is disabled, the power (-48V) is removed and the LED on the base station interface card is switched off.

Known issues

Shared lines (Handset Sharing) and internal switching cannot be combined.

Corrections and Improvements

- Previously, when setting analogue CLIP gain for FSK or DTMF receiver, settings were not actual set in analogue interface card.
- If the XML-RPC password was different from the default password and at the same time a value was changed in the WEB-GUI configuration page (and the button “Save all settings” is pressed), the XML-RPC password was changed to “*****”. This error has been corrected.
- The number of reported sync errors (on the wire between base station and KWS) has been further aligned compared with other KIRK servers.
- Service report has been adapted to new features. A bug in one of the tables with handover between base stations has been corrected.
- An error in users.xml file is corrected (which is included in the service report) regarding disable & AC code when a handset was set to block outgoing.
- Replacing a SIP handset via a master handset now works.
- A bug in HeartBeat MSF functions has been fixed along with an issue via XML_RPC.
- In PCS13__ the following problem was addressed on the new generation CPU (PowerPC) card firmware: “*Changed XML-RPC buffer handling from static to dynamically. The change was implemented as result of a costumer inquiry. The 3rd Party application was in some situations polling with a very low frequency that caused multiple headers in a large segmented message.”*
  Now it is also implemented on the CPU (ARM) card firmware.
- After a new layout in file structures, the SIO password was not deleted when pushing the default button. The error has been corrected.

Version PCS13__, 2013 - Q1

For this and future software releases, use only the WEB-GUI to configure your system. If you have to use the OAM Administration Program, use Software version 0.4.0.88. Please note if you create a new user or edit an existing user with the OAM Administration Program the user database will be corrupted!

If the WEB-GUI cannot be accessed, please use TinyOAM 3.0.0 for configuration settings.

Added or Changed Features

- The KIRK Wireless Server now supports reception and handling of a user ID (Local Number/DN) during a DECT subscription. Some KIRK handsets will now potentially send a user ID during subscription. This user ID will be used to find a matching user template in the user database if the IPEI of the handset is not in the database.
- Master handset feature: A handset configured a Master Handset has via MSF function “1234” access to a menu for replacing an existing handset. It will only ask for local number to replace and the new handsets IPEI (Handset unique serial number).
- VLAN –priority of voice - 802.1p Class-of-Service added.

Known issues

- Shared lines (Handset Sharing) and internal switching cannot be combined.

Corrections and Improvements

- WEB-GUI stability performance and interoperability with Firefox, IE8, IE9 and Chrome has been improved.
- Most double vertical scrollbars have been removed.
- Sort functions have been improved.
- Mouse over texts & help texts has been improved.
- When changing the user to a shared line the corresponding phone will automatically be de-subscribed and the IPEI removed.
- When a MSF is send to a shared line and the shared line has no assigned handsets the release reason is change from “unknown PP” to “User Not in Range”.
- Handset sharing parameter is added to provisioning. Only mandatory parameter for provisioning is now <localnumber></localnumber>.
- If a user has been deleted and users with higher PPID still exist, then sometimes the KWS could not find handsets with the higher PPID.
- In some situations the universally unique identifier (UUID) was regenerated every time the KWS was rebooted.
- *MasterRS232MaxTraceLevel* is reduced to 2 after restart to prevent overload of RS232 on Master shelf.

- When toggling between static and dhcp assigned IP settings, the static settings are preserved.
- TAP interface: Decoding of call back number has been improved.
- Error in endpoint_nv_data (XML-RPC) has been fixed.
- Error in MSF subevent "ExtenHwInd/ endpoint_hardware_extension (XML-RPC) has been fixed. This sometimes caused the KWS to restart.
- Changed XML-RPC buffer handling from static to dynamically. The change was implemented as result of a costumer inquiry. The 3rd Party application was in some situations polling with a very low frequency that caused multiple headers in a large segmented message.

Version PCS12M_, 2012 - Q4

WEB interface: We recommend using the latest version of Firefox or Chrome.

For this and future software releases, only use the WEB-GUI to configure your system. If you have to use the OAM Administration Program, use Software version 0.4.0.88, and note that you will not be able to create new users.

Added or Changed Features

- Handset sharing. (License required)
  When a shared handset is lifted from a charger, the user has to assign the handset to a line. This is done by entering the line DN/extension and the line password. The shared handset and the line (Analogue or SIP) must be in the same group.
  The shared handset is created as a special handset with no assignment to a line. A shared handset with no line assignment cannot be used for making call.
  The shared line is created as a special "handset" with no IPEI. A shared line with no shared handset assigned will act as busy towards the line.
- IPv6 support and new network configuration implemented. (IPv6 is only available on next generation CPU cards and Media Resource cards.)
- VLAN support (IEEE 802.1Q).
- Implemented support for BGR139 events via XML-RPC and EMD, so it is ready when it is implemented in 7040 & 6040 handsets.
- Prepared for longer texts and call back numbers in MSF (EMD protocol), so it is ready when it is implemented in handsets.

Known issues

- The provisioning interface does not understand the handset sharing parameters and values.

Corrections and Improvements

- When an MR card starts up without a LAN connection to the Master CPU, a warning message will be displayed in trace window.
- Previously, it was not always possible to “Cancel Service Report”.
- Improved support of a specific attended transfer scenario for a Tadiran IPx800. (The scenario is marked as “not recommended” in the RFC. Latest bug fixed Coral PBX firmware is required.)
- When sending a broadcast (MSFClmsFixedReq) with more than 19 characters via XML-RPC, the KWS was resetting; now the text is truncated.
- The corporate phonebook has been updated based upon customer feedback.
  The Corporate phonebook now supports special characters (ISO 8859-1 ISO Latin-1).
  This means that it is now possible to search for e.g., Müller in the corporate phonebook.
  The Corporate phonebook now accesses the ldap server asynchronously for better performance and a better user experience.
- In service report, value of the disable tag is changed from “allow”/”disabled” to “false”/”true”.
- When Broadcast (MSFClmsFixed-Req & -Cfm) is received and executed, a trace message has been added on trace level 3.

Version PCS12K_, 2012 - August

WEB interface: We recommend using the latest version of Firefox or Chrome.

For this and future software releases, only use the WEB-GUI to configure your system. If you have to use the OAM Administration Program, use Software version 0.4.0.88, and note that you will not be able to create new users.

Added or Changed Features

None

Corrections and Improvements

- New help text for TAP in web GUI added.
- If a SIP user dials the CFU enable or disable command (typical *21* or #21#), the system would reboot
- In a specific transfer scenario the system was rebooting when the handsets involved was released in a certain order. Scenario:
  - A call B and B answer the call.
  - A press R key and dial a not existing or busy number.
  - When A hears a busy tone B goes on hook.
  - When A goes on hook the system reboots.

- KWS 8000 has never supported the special synchronizing method for RFP2 generation of base stations. Now if KWS detects a RFP2 base station, the base station will be rebooted.
- It´s now possible to use "*" in callback number in Apps. Demo.

Version PCS12G_, 2012 - Q3

WEB interface: We recommend using the latest version of Firefox or Chrome.

For this and future software releases, only use the WEB-GUI to configure your system. If you have to use the OAM Administration Program, use Software version 0.4.0.88, and note that you will not be able to create new users.

Added or Changed Features

- TAP (Telocator Alphanumeric Protocol) implemented. TAP is a widely used protocol in the healthcare sector in the US. The TAP "server" sends messages through an RS232 interface and connects directly to the KIRK Wireless Server via its RS232 interface. The KIRK Wireless Server receives and interprets the TAP messages and sends corresponding messages to the handsets (with or without callback number). Configuration is done via the web GUI. In order to run TAP, a license is required in the KIRK Wireless Server.
- The interface can handle up to 2800 messages per hour (and up to 700 messages to the same handset per hour).
- New embedded Media resource firmware containing DSP corrections that improve random noise problems. (Rarely occurring and only seen with MR-card placed in slot 1).
- Added help buttons in WEB-GUI for TAP, phonebook and analogue settings.

Corrections and Improvements

- RSSI (Radio Signal Strength Indication) level detected by base stations adjusted to follow *99989* levels from handset.
- Fixed problem with parsing of port number in proxy URI (web GUI issue).
- Added check for very old RFP2 base stations (produced before middle of 1998), and writing a warning that RFP2 will not work with KWS 8000 / KWS 2500.
- Fixed issues with decoding of UTF8 / Latin-1 characters in SIP.
- Improved service report:
- Added users.xml to service report, change recommendation limit for updating firmware in KIRK 50-, 60- , and 70-Handset Series from PCS09A to PCS12P.
- Improved handling of the not recommended method of Semi-Attended Transfer. For further detail see description in RFC 5589 (figure 12). (http://tools.ietf.org/html/rfc5589#section-7.6)

Version PCS12F_, 2012 - May

WEB interface: We recommend using the latest version of Firefox or Chrome.

For this and future software releases, only use the web GUI to configure your system. If you have to use the OAM Administration Program, use Software version 0.4.0.88, and note that you will not be able to create new users.

Added or Changed Features

- None

Corrections and Improvements

- New embedded RFP5 code (pcs12__), which fixes bug that caused base stations to restarts if heavily loaded
- Fixed bug that caused the server to restart when dialing an empty number from a Butterfly handset
- Fixed bug that caused the server to restart when making SIP calls without having a media resource card present
- Fixed bug that randomly caused analog and DECT-to-DECT users to have the text "(CFU)" added to their standby text
- Fixed problem with Nortel re-directing SIP server
- Fixed problem with special characters in phonebook
- Fixed issue with broadcast in the applications demo tab (Apps Demo) in the web GUI. Did send discriminator 0x86 and not 0x87 for silent&vibrate - resulting in an audible alarm instead of a silent alarm
- Improved and updated Service report
- Network configuration improved. Sometimes it was not possible to see and configure a MR-card on a secondary shelf

Version PCS12E_, 2012 - April

WEB interface: We recommend using the latest version of Firefox or Chrome when working with the web GUI.

From this and future firmware releases, please only use the web GUI for configuration of your system. However, if you need to use the OAM Administration Program, please make sure to use software version 0.4.0.88, and be aware that creation of new users will not function.

Corrections and Improvements

- Some Siemens handsets could cause the server to restart when initiating a call. This has been corrected.

- An error could cause analog users to not be able to receive calls when an analog user subscribed to the system was updated (new standby text, new local number, etc.) or trace filters had been used. This has now been corrected.
- The management of base stations via the web GUI has been improved. Now, there is no limitation on the number of base stations that can be viewed in the web GUI. Please note that this is only relevant for systems with more than 21 base stations.

Version PCS12C_, 2012 - Q2

WEB interface:

We recommend using the latest version of Firefox or Chrome.

For this and future software releases, please only use the web GUI for configuration of your system. However, if you have to use the OAM Administration Program, please use Software version 0.4.0.88, and note that creation of new users will not function.

Added or Changed Features

- Added number of SIP registrations to statistics in service report. Added reset of counted cable delay values for base stations when statics are reset.
- In case of poor signaling condition (to many bit errors) between server and base station, the base station will be powered down for a period, starting with 10 minutes and ends up to 50 minutes if the poor signaling conditions remain. If a base station is powered down, the LED on the corresponding interface card is flashing red/orange.
- For network diagnostics, a ping, traceroute and nslookup feature has been added. It is accessible through the Network menu in web GUI.
- Starting with PCS12C_ the firmware is prepared for the Security Package license. If a Security Package license is installed, various security enhancing features become available.
  Encryption of media (stream between the KIRK Wireless Server/media resource and the external endpoint/PBX) according to RFC 3711 (Secure RTP or SRTP) is possible.
  Encryption of media
  SRTP handling is supported in optional as well as required mode.
  Configuration of SRTP is done via web GUI in SIP configuration menu - Media.
  If ‘enabled’, SRTP is supported and optional, and it must be negotiated with the remote endpoint. If ‘enabled and required’, the use of SRTP is mandatory, and if negotiation of SRTP with the other end is unsuccessful, call establishment will be aborted.
  Handling of RFC 4568 SRTP lifetime key parameter and Master Key Index parameter in SDP offers are configurable.
  If SRTP is enabled, the number of available voice channels on a KIRK Wireless Server/media resource will NOT be reduced – all 32 channels are available.

- Added support for certificate handling. A certificate is required to be able to use SIP over TLS. The KIRK Wireless Server is delivered with a Certificate Authority (CA) bundle with common Certificate Authorities. This means that the KIRK Wireless Server will accept certificates issued by, for example Verisign out-of-the-box. In addition to the CA-bundle, the GUI allows for installing a local CA certificate bundle if a certificate is generated by a local authority (e.g. a service provider or the local IT department). A certificate bundle in PEM-format may be imported.
- Embedded new firmware for base station RFP5 type (PCS03H_). New firmware improves performance on sites with a lot of other DECT systems or problematic deployment.
- Possible to configure TCP server port (default is 10000) and HTTP port (default is 80).
- If an Automatic Alarm Call License is loaded Automatic Alarm Call application is enabled in the KIRK Handset (KIRK 60- and 70-Handsets only).
- User licenses are now required in KIRK Wireless Server 2500 and KIRK Wireless Server 8000. See product Bulletin 1500.

Known issues

Some systems will not have all RFP’s displayed correctly in the web GUI. Corrections are in progress.

Corrections and Improvements

- Previously “old” pcap trace was not deleted once it was downloaded in a service report. Old pcap trace was still part of a service report.
- Service report warnings refined – more warnings if: Added advice to update RFP5s if there are many other DECT systems and older RFP5 firmware. Added advice to make sure that 50-Handset Series, 60-Handset Series and 70-Handset Series are updated to 2012-Q2 release, if there are many alien DECT systems.
- SIP stack updated. See configuration parameters.
- Disabled level 4 trace - SIP signaling trace. If enabled and many SIP calls are ongoing, the KIRK Wireless Server could be loaded heavily. An alternative to level 4 trace is to run a service report with PCAP enabled.

Configuration File Parameter Changes

File	Action	Parameter	Description
config.xml	Added	sip.media.srtp.enable	If enabled, SRTP is supported and optional. It must be negotiated with the remote endpoint. If external SRTP is enabled, the number of available voice channels on a media resource will remain the same (32 channels) Values: true/false Default: false.
config.xml	Added	sip.media.srtp.required	If enabled, the use of SRTP is required. If negotiation of SRTP with the other end is unsuccessful, call establishment is aborted. Values: true/false Default: false.
config.xml	Added	sip.media.srtp.lifetime	Handling of RFC 4568 SRTP lifetime key parameter in SDP offers. Values: true/false Default: false.
config.xml	Added	sip.media.srtp.mki	Handling of RFC 4568 SRTP Master Key Index parameter in SDP offers. Values: true/false Default: false.
config.xml	Added	sip.use_sips_uri	Normally SIP communication on a TLS connection is using the SIPS: URI scheme. Disabling this option causes the KIRK Wireless Server to use the SIP: URI scheme with a transport=tls parameter for TLS connections. Values: true/false

File	Action	Parameter	Description
			Default: true.
config.xml	Added	sip. tls_allow_insecure	By default UDP and TCP transports are disabled when TLS transport is the default. If this setting is true, UDP and TCP are allowed as fallback if TLS fails. Values: true, false. Default: false
config.xml	Deprecated	sip.proxy.transport	Deprecated. In release PCS12C_, this setting is replaced by sip.transport & sip.dnsmethod. The KIRK Wireless Server still understands this setting, but the new settings should be used.
config.xml	Added	sip.dnsmethod	Specifies the DNS method used to resolve host names for SIP requests. Values: arecord/ dnssrv. arecord: Use simple DNS A records to resolve host names. Basically, A records are used to translate a hostname to an IP-address. dnssrv: Use DNS SRV records to determine host addresses. Refer to RFC3263. DNS SRV records can be used to specify multiple servers with different priorities and/or multiple servers for load-balancing. Default: arecord.
config.xml	Added	sip.transport	Specifies the transport mechanism used for SIP requests. Values: UDP, TCP, TLS. Default: UDP.

File	Action	Parameter	Description
config.xml	Added	sip.gruu	Specifies the use of Globally Routable UA URI (GRUU) which is an URI that routes to a specific UA instance. If enabled, a GRUU will be obtained from a server and communicated to a peer within a SIP dialog. Values: true/false Default: true.

Version PCS12A_, 2012 - Q1-bug fix

This software version is only relevant if SIP interface is used.

Corrections and Improvements

- An incoming SIP call with empty display name sometimes results in a reboot of the server.

Version PCS12__, 2012 - Q1

WEB interface tested on:

Internet Explorer 8.0

Firefox 6.0

For this and future software releases, please only use the web GUI for configuration of your system. However, if you have to use the OAM Administration Program, please use Software version 0.4.0.88, and note that creation of new users will not function.

Added or Changed Features

- Added handling of licenses.
- Added provisioning. Possible to centralize configuration and maintenance (see latest version of Provisioning Guide – 1418 4650). Provisioning is operated and working the same way as the KIRK Wireless Server 6000 and KIRK Wireless Server 300.
- UPNP support also supports broadcast announcements. The server publishes itself as KWS8000 [ARI] (eg. KWS8000 10034725164). If Company Name field in Company Info section (under Email Report) is configured, the name will be used instead of the ARI.
- Faster firmware downloads via web GUI (especially on single shelf systems).

- Now possible to get PCAP (network Packet CAPture) traces in service report. When starting “Capture Scenario” a new window appears. In this window the operator can choose to add SIP, RTP or Everything to the PCAP trace.
- New settings and configuration are added to service report.
- Added the following events to the XML-RPC interface (see latest version of XML-RPC SDK 1412 4618) :
    - Temporary standbytext to the handset
    - Call Control events
    - Detach and attach
- Added Reboot button for all base stations in web GUI.
- Added ARI to trace window in web GUI. Also, automatically enable trace monitor window when trace level is changed from disabled to any other level.

Known issues

- Web interface does NOT and will never work correctly on Windows Internet Explorer 7 or older versions.
- Even though the web servers chance to use the browsers cache has been reduced, it can still be necessary to clear the browsers cache. If you experience some strange behavior with the web server it might be possible to solve the problem by clearing the browser cache.
- When using the OAM Administration program for system configuration the creation of new users does not function. Instead use the web GUI.

Corrections and Improvements

- Fixed the settings bug for analogue interface cards.
- Listing of Base Stations in web GUI could sometimes fail and not show all Base Stations. This has now been corrected.
- When setting date and time manually, the Real Time Clock was not update properly. The consequence was a KIRK Wireless Server time incorrectly adjusted after a reboot of the KIRK Wireless Server.
- Corrected error in SIP registration during startup procedure of the KIRK Wireless Server. It could be a problem on installations with more than 100 SIP registrations. A typically user experience would be handsets not making proper SIP registrations.
- Corrected error when receiving more than 16 digits from the handset.
- Improved cache handling in web GUI.
- No longer sending empty display text to handsets. This improves interoperability against other DECT GAP handsets.
- Reduced CPU load while producing the service/e-mail report.

- Configuration of RS232 and messaging protocol settings made more flexible, but still only available in trace window.
- Phonebook from LDAP server has not been working in latest GA SW.
- Muting handset transmit voice path during sending of DTMF digits on an analogue line. We have experienced that sometimes under very noisy conditions some or all digits were lost.
- Fixed error in web GUI when reading IP address on last CPU card in a system with more than 3 shelves. Also fixed issue with IP address in web GUI. The IP address could not contain 4 *3 digits (like 123.111.222.333).
- Fixed issues in Apps Demo tab in web GUI. A problem with “Always vibrate” flag was not passed correctly in a MSF Format III message. Empty callback number in Normal MSF and MSF Format III was previously incorrectly filled with a ‘-‘.

Configuration File Parameter Changes

None.

Version PCS06A_, 2011 - Q4

WEB interface tested on:
Internet Explorer 8.0

Firefox 6.0

For this and future software releases, please only use the web GUI for configuration of your system. However, if you have to use the OAM Administration Program, please use Software version 0.4.0.88, and note that creation of new users will not function.

Added or Changed Features

- The Service Report page has now become the start page. The purpose is to advertise on the advantage or sometimes necessity to run a service report. The installer/user is encouraged to run the service report with the four following points:
  - Run a Service Report as the first step in any service session in order to document the start configuration.
  - The Service Report includes text files with an overview of system configuration, statistics, detected errors and problems.
  - Run a Service Report as the last step in any service session in order to document the configuration.
  - A Service Report is always required if you need any support on this product from the Polycom Support team (Please include the description of observed and expected behavior).
- Added start and stop of “capture scenario”. Function automatically sets all trace levels to high, when started. When stopped, the trace levels are set back to previous settings.

The function is used when a certain scenario (most likely an error situation) has to be captured in the service report. The first step would be to start the “capture scenario”, then reproduce the scenario/error situation. Finally, press the “stop scenario” button or just run the service report (starting the service report will automatically stop the “capture scenario”).

- Added option for 9600 bps on RS232 on CPU card in Primary shelf.
- Previously, SIP calls have automatically been disconnected when ended on far end call release (e.g. If a call between 2 handsets is terminated from one handset, the other handset will automatically disconnect). This behavior is not always suitable for all the users. Also, if the far end release is not intended – the user needs some kind of audible feedback about the terminated call. In this software release a 2500 ms delayed release timer has been added - When a call is normally or abnormally released from far end, the handset plays internal busy tone in 2500 ms before being released automatically. The user can also choose to go on hook by himself.
- Added simple handling of 305 Use Proxy response (this was required by Coral UGW).
- Added SIP user domain. Every SIP user can now subscribe to own proxy.
- Reset of chosen parameters to factory defaults.
- Files with “Erlang statistics” and “Traffic distribution” have been added to email- and service report. The Erlang statistics is calculated on RFP level and includes both voice and messaging traffic. Statistical counter values for each 15 minutes of a week are written in the “Traffic distribution” file.
- Busy hour is calculated if there has been any traffic.

Known issues

- WEB interface does NOT and will never work correctly on Windows Internet Explorer 7 or older versions.
- Remember to clear the browser cache.
- When using the OAM Administration Program for system configuration the creation of new users does not function. Instead use the web GUI.

Corrections and Improvements

- XML_RPC: Supporting persistent http version 1.1.
- The text length in an MSFdisplayReq is changed from 36 characters to 72.
- Default blocking trace level for Master RS232 port is reduced from 5 to 3. This is done to prevent flooding of the RS232 port.
- Now it is possible to have a more flexible dial plan (previously it was not possible to have local number 200 and 2100 at the same time. It is still not possible to have local number 200 and 2000 at the same time).
- Longer trace messages were previously truncated; today they are split into two separated trace messages.

- The problems with calls between handsets subscribed as “DECT to DECT” and handsets subscribed as Analogue or SIP have been fixed. Call was not properly released in case of abnormal releases from the far end handsets. Furthermore the “DECT to DECT” handsets could not make proper connection handover in some situations.
- When the KIRK Wireless Server indicates DTMF support in outgoing SDP offers, DTMF events 1-15 are indicated. Previously 1-11 were indicated. The KIRK Wireless Server will actually send only 1-11 events. However CUCM 8.6.x apparently needs the offer to indicate 1-15 to be able to handle DTMF correctly.
- The “Daylight saving” error when changing time from NTP to manual time setting has now been corrected.
- WEB interface: DTMF Tx “level offset”, Gain FSK and DTMF in Analogue settings was previously not shown as signed values.
- A risk of getting a negative SIP Subscription Expire timer values if IPBX responded with short re-registration time value is now prevented (Registration could be lost).
- Some third party KIRK handsets sometimes send DETACH when subscribing causing previous release to reboot.
- The problem when using automatic DNS has now been fixed. Previously, only the static DNS settings were working.

Configuration File Parameter Changes

None.

Version PCS06__, 2011 - Q3

Please use OAM SW 0.3.0.88 or newer.

Added or Changed Features

- Central supervision of dummy-bearers to ensure better handover possibilities in places with several DECT systems has been implemented.
- XML-RPC application interface has been added.
- The new XML-RPC based application interface uses open standards and it’s easy to
- use. This interface has almost the same functionality as the existing MSF interface. The existing MSF interface will not be affected. The interface is now (probably in the future, as well) working the same way as KWS300/KWS6000. See how to interface in the document "KIRK Wireless Server 300 and 6000/XML-RPC SDK 1.5.1” or newer.
- From web GUI, XML_RPC can be enabled and user name and password can be entered.

- "click to dial" has been implemented (see the latest version of EMD PA-1411 0629). It enables a 3rd party application to control voice calls (establish, dial, connect and release calls).
- SetUserDetachStatus and GetPPStatus have been implemented (see the latest version of EMD PA-1411 0629). Depending on the handset implementation, it is possible to see if the handset has been turned off or on.
- Added the possibility to handle HTTPS (Built in certificate), both web and XML-RPC.

Known issues

- It is not possible to change timeslot (odd, even or follow RFP) on a RFP4 from the web GUI.
- MSF format 3 does not have the correct format on the web GUI "MSF demo" tab. Message ID is only using the low byte in 16 bit value. The callback number can only be 4 digits long. Furthermore in the "PP hardware extension" tab, the action field can be set to "Activate" and "Deactivate". These settings are not supported by any handsets.
- The IPEI of a user cannot be changed in the web GUI.

Corrections and Improvements

- System, configuration, trace file and the email report have been improved/extended.
- Handle incoming SIP INFO, rfc2976 has been implemented (Astra 5000 uses SIP INFO for keep-alive).
- SystemRestartReq was improved with the following 4 options:

1. Now.
2. when idle. (no MSF & no voice calls).
3. when no voice calls.
4. block new voice calls and wait until all voice calls are released.

- Ensuring that unused DN in broadcast (to All PP) request (MSFClmsFixedReq, SMSSetupReq and ExtenHwReq) is not inserted in MSFClmsFixedCfm.
- The possibility to set separate master RS232 TraceMessageLevel has been implemented. This is used to prevent the overflow on the RS232 port.
- SW has been made more robust to faulty MSF events (such as missing local number).
- If SIP username is marked in update or create user data event and the string is empty, then the use of local number as SIP user name is insured.
- Statistics for page time on incoming voice calls and for some types of Supplementary Services have been added (they can only be seen in service report).
- Age to detection of each alien RFPI has been added (it can only be seen in service report).
- A wildcard subscribed handset can now use the servicekeys ***999*00-03 (ex. to see the IP address or MAC of the system).
- The Call Forward Unconditional (CFU) issues have been fixed. Now it can also be changed or added from the web GUI.

Configuration File Parameter Changes

None.

Version PCS05A_, 2011 - Q2

Please use OAM SW 0.3.0.88 or newer.

Added or Changed Features

- A web server has been implemented. The web GUI allows the user to maintain and operate the same functions and features as the OAM tool.
- Service codes for users on all interfaces are retrieved by typing the following codes and then press the hook key:
- IP_addr: ***999*00
- MAC_addr: ***999*01
- FW_version: ***999*02
- MSF events to write directly to PP EEPROM have been reinvented.
- A phone book application has been implemented. This feature offers a centralized phone book. The formats supported for the phone book are csv-file and LDAP. It should be noted, that while retrieving phone book data from a remote LDAP-server, the phone book will be inaccessible. This means that the refresh interval (the interval in which the central phone book data is being copied from the LDAP to the KWS) should be chosen with care. The combination of a slow LDAP-server/slow LDAP-server connection and a large number of entries in the corporate phone book (> 10,000) should be configured with a long refresh interval, e.g. once a day.
- Local call forward (unconditional) is now supported. The number to forward to is configurable through the web GUI (or OAM program) as well as directly from the handset. Using the GUI, the local call forward number can be viewed/edited directly from the user entry of the user in question. The feature code for enabling/disabling local call forward from the handset can be configured through the GUI menu. The default code is "*21*$#" where "$" denotes the number to forward to. If has call forward is enabled in a handset, the standby text will be pre-ended (with CFU) to give the user an indication that the handset is forwarded.
- It is now possible to receive the time from an NTP server.
- In SW PCS04F_ "internal routing" was presented and only accessible from trace mode. Today it can be reached from the GUI and is called DECT Call Configuration.
- New embedded RFP5 firmware, PCS03G_, has been introduced to be used for RFP5 update.
- "SIP username" has been added. To create a SIP user both "local number" and "SIP username" must be programmed. In many installations, "local number" and "SIP username" will be the same.

Known issues

None.

Corrections and Improvements

- Reset of web server password when pushing the default button has been added.
- An error that caused MSF_REL_REQ with forced release reason NOT to release msf session/instance has been corrected.
- Default value of SendMailAllowed has been changed from true to false. New default: Sending e-mail report is disabled.
- After restore of backup of user data, the KWS will automatically reboot.
- Trace mode has undergone a small makeover.
- A trace level 5 containing “all trace messages + debug messages” has been introduced. The old level 4 is now “level 3 + sip messages”.
- System, configuration and trace file (and of cause the e-mail report) have been improved/extended. This function has also had its name changed to “Service report” in the web GUI.
- Network status has been improved.
- A bug regarding analogue users has been fixed. If dial tone timer expires (i.e. if the analogue interface card could not recognize the dial tone,) a faulty text would sometimes be sent to the handset "ABxx card in slot x is not present".
- A bug regarding systems configured with MR interface has been fixed. Each time a handset made a location registration, a SIP endpoint was created (sending SIP REGISTER towards SIP PBX). This is only a potential problem if default domain or proxies are configured.

Configuration File Parameter Changes

None.

Version PCS04F_, 2011 - Q1

Please use OAM SW 0.3.0.81 or newer.

Added or Changed Features

- If a SIP PBX sends a CANCEL with the reason header “Call Completed Elsewhere”, the message “Missed call” will no longer be displayed in the handset (50-, 60-, or 70-Handset series). This feature requires a recent firmware for the handsets.
- An auto-answer feature which can be used for intercom and loudspeaker calls has been implemented. If an INVITE with an Alert-Info header, a Call-Info header or an Answer-

Mode header is received, it is possible to make a Polycom handset automatically answer the call, mute the microphone and turn on the speakerphone.

- The reason why several headers are needed to handle this feature is that different SIP-PBX's have different default implementations. The following list of headers will activate auto answer:
  - Alert-Info: Auto Answer
  - Alert-Info: info=alert-autoanswer
  - Alert-Info: Ring Answer
  - Alert-Info: info=RingAnswer
  - Alert-Info: Intercom (*This is the default setting on Trixbox*)
  - Alert-Info: info=intercom
  - Call-Info: =\;answer-after=0
  - Call-Info: ;answer-after=0
  - Answer-Mode: Auto (*This is according to RFC 5373*)
- The feature is implemented in the 50-, 60- and 70-Handset series (except in 7010 which does not have speakerphone). The feature requires a recent firmware for the handsets.
- It is now possible to update the RFP software via the KWS8000/KWS2500 server. RFP update is only supported on RFP5 and RFP4 (Infineon RF, SW PCS >= PCS09_). The latest RFP code is embedded in the server software.
- When a handset is subscribed as a SIP user, it is now possible to add RX gain (receiver gain for the handset).

Known issues

None.

Corrections and Improvements

- Now valid system information is available when no interface card has been inserted.
- Internal routing between "DECT to DECT and Analog" and "DECT to DECT and SIP" has been made possible. The feature is set from "Trace mode".
- When a handset is turned off/on, a SIP registration is sent to the server (if user is SIP and user is not registered on an IPBX).
- An error in SIP configuration events (codec priority list) has been corrected. Also a problem when typing a URI in proxy domain like sip:example.com:5555 has been solved. The port number 5555 was not used/stored correctly.
- System, configuration and trace files (and of course the email report) have been improved/extended. P-ID (KIRK-ID) for backplane, (AB16 and AB08) card, BIF08 card and MR32 card are all stored in files.
- Timestamp and priority have been added to MSF3.

Configuration File Parameter Changes

None.

Version PCS04D_, 2010 October

Added or Changed Features

- Supporting MWI for KIRK produced handsets.
- Built in a RS232 log facility on Master shelf (controlled from trace mode).

Known issues

None.

Corrections and Improvements

- In the previous firmware version PCS04C_, it was not possible to make connection handover when a user was subscribed as analog user. This has been corrected.
- Call waiting can now be turned off.
- Problems with noise at the beginning and end of a SIP call have been solved.
- Setting of DHCP in server and MR32 card is now working.
- System, configuration and trace files (and of cause the email report) have been improved/extended; including decoding of more KIRK handset types, added production serial number for interface cards, CPU and backplane and SIP username in system log.

Configuration File Parameter Changes

None.

Version PCS04C_, 2010 September

Added or Changed Features

- Support for MR32 interface card.
    - When MR32 card is inserted in the KWS8000 backplane the SW version in the MR32 card is verified by the KWS8000 server. If SW version on MR32 card is not matched by KWS8000 server the MR32 card will by down- or upgraded to correct version.
    - MR32 and Analog interface card can coexist in a KWS8000 system.
    - SIP stack is inherited from the KWS6000 project.

        - Note: Provisioning is not a part of SIP! – it is a configuration tool for KWS6000. KWS8000 still need the OAM PC software for configuration.
        - XML-RPC is not a part of SIP, but a messaging protocol for KWS6000. KWS8000 still support the EMD specification on both RS232 and LAN.
    - Added possibility to block and unblock all calls.
    - Added more information when receiving none supported FSK clip.
    - Implemented setting of ringing mode for analog calls (Exchange ringing or System ringing).

Known issues

- MWI envelope will not be shown in handset.

Corrections and Improvements

- System, configuration and trace file (and of course the email report) have been improved/extended.
- Error in handling old KWS1500/500 protocol format has been fixed. (Still only supporting basic MSF).

Configuration File Parameter Changes

None.

Version PCS03N_, 2010 - Q3

Added or Changed Features

- Implemented exchange/system ringing for analog trunks. (Until OAM support is possible, the function can be controlled in Command mode.) Exchange ringing is selected by default. When set to exchange ringing, the ringing of the handset follows the ringing cadence of the analog line. When set to system ringing, the handset follows its own ringing pattern.
- Every time a handset is subscribed or switched on (location registration), an EMD message is sent to the serial or IP interface.

Removed Features

None.

Corrections and Improvements

- System, configuration and trace file (and of course the email report) have been improved/extended.

- Handling of EMD events with wrong length has been improved. The EMD developer will now get more and better feedback.

Configuration File Parameter Changes

None.

Version PCS03M_, 2010 - Q2

Added or Changed Features

- Driver support for BIF02 cards.
- Support for depopulated backplane (KWS8000-light backplane).

Removed Features

None.

Corrections and Improvements

- Increased stability for some EMD events.
- Added content to log file generation from OAM program / email report.

Configuration File Parameter Changes

None.

Version PCS03K_, 2010 - Q2

Added or Changed Features

- Added email report content.
- From Command mode it is possible (via the T command) to see the last 3000 trace messages.

Implemented EMD access to the following features: (This will be available from OAM program in next version 0.1.0.5x)

- Enable or disable “MSF between PP” functionality.
- Change temporary standby text via MSF.
- Alien DECT systems seen by RFP5. This is used to indicate how many other DECT systems (ARI code and RSSI) the RFP can see.

- Get system report (like email report - or an extended version of “Save configuration statistics on PC”.
- Get Cable Delay Values (CDV) statistics (every 64 sec. the CDV is read on every connected RFP). If too many different values are read, it can indicate (together with sync errors from the RFP) that the cable or connector needs some extra attention.
- Added more debug info when receiving an “unsupported clip / message type”.
- Supporting clip in POLYCOM 2010 DECT handset. When clip is received with only DN and no NAME, the content of DN is copy to NAME field.
- Improved RFP interface robustness.
- Removed not relevant functions from Trace mode.
- New messaging and hardware extension facilities for use with 60- and 70-Handsets.

Removed Features

None.

Corrections and Improvements

Fixed bug introduced in PCS03G_ regarding mails in single shelf system.

Configuration File Parameter Changes

None.

Version PCS03G_, 2010 February

Added or Changed Features

- Can now read HW_PCS from RFP5. (Available from Command mode – until OAM SW supports the facility). Also available in email report.
- Allows setup of different baud rates on the serial port (default is 115200 baud).

Removed Features

None.

Corrections and Improvements

- Corrected error that caused Clip name not to be shown in PP, when clip info was sent during ringing.
- Updated MAC capabilities to comply with WRFP SW PCS24_ requirements.

Configuration File Parameter Changes

None.

Version PCS03E_, 2009 December

Added or Changed Features

Software support for the following items are only supported from CPU HW PCS05_ (CPU HW PCS05_ is not released yet).

- The power LED morses the IP address. The morse cycle for the power LED is as follows:
  - Steady green 30 seconds.
  - Blinking blue 3 seconds (get ready sequence starts).
  - Green blink represents digits (zero is a long blink). Red means dot between digits.

- Reset of SIO password. To reset the SIO password, press the Reset button on the CPU card.
  - Short press (2-5 seconds) power LED blinks red – reset system.
  - Long press (5-9 seconds) power LED blinks blue – reset SIO password + reset of system.

Pressing longer than 9 seconds (or until power LED is steady green) will leave the system as it is. (No reset of the system.)

Removed Features

None.

Corrections and Improvements

- Mail system not working correctly - could load system heavily and make system hang.

Configuration File Parameter Changes

None.

Version PCS03D_, 2009 November

Added or Changed Features

- New mail client.
- Support for new Analog card types with full wave detector and new AB08 cards.
- Support for CPU cards with no link facilities.

Removed Features

None.

Corrections and Improvements

- Cleanup of Command mode.
- In a linked system, RFP sync and reset errors were stored in statistics only if the RFP card was in the first shelf.
- Added receiver gain for clip module (FSK) and receiver gain for DTMF gain.
- The RFP's could reset due to a few sync errors. The sensitivity has been decreased.
- Improved handling of telnet EMD connection (requires KWS8000 OAM program - 0.1.0.44 or later).
- Corrected error that could cause RFP to hang (power on, but not able to restart).
- More engineering debugging.
- Improved stability and performance.
- Corrected error that caused system to reset. When an incoming call was made on an analog line where no user was created, the system could either reset or go into a non-defined mode (and a number of undefined things could happen).
- New engineering debug file(s) is attached to mail.
- Corrected bug that caused only one ringing when handset was signed up as dect2dect user.
- Changed format on LID in trace messages when using analog lines.

Configuration File Parameter Changes

None.

Version PCS02C_, 2009 August

Added or Changed Features

- Support for AB08 cards.
- Support for Clip - DTMF, FSK Bellcore 202 and FSK_V23
    - The clip type is set from Command mode. (When the next version of the OAM program is available it can be done from the Analog settings tab.)

      Go to Command mode, go to the Trace tab, and then click the Send button at the bottom of the screen (the Main menu appears.)
    - Place the cursor in the Command field and type A to go to the Analog menu (the Analog menu appears). Make sure that internal clip is disabled by typing D in the Command field.

- Type :
  - S1 (to enable FSK Bellcore 202 clip)
  - S2 (to enable DTMF clip)
  - S3 (to enable FSK_V23 clip)

Removed Features

None.

Corrections and Improvements

- Enhanced security. Possible to calculate new encryption key for every new call.
- Error when disconnecting RFP during call.
- Another manufacturer's GAP DECT handsets were not allowed to subscribe.
- When no analog or base station card was inserted in the system, the system configuration was not shown properly and valid.

Configuration File Parameter Changes

None.

Version PCS02__, 2009 - Q3

Added or Changed Features

- Encryption of voice in the air.

Not all RFP's support encryption. If a negative acknowledgement is received on an attempt to enable encryption, it is probably because some or all RFP's do not support encryption.

If a "none encryption supporting" RFP is added while encryption is enabled, the RFP is rebooted and a trace message can be viewed in the OAM program.

Encryption can be set to 3 levels.

- Encryption turned off.
- Encryption enabled – in case of encryption reject (PP or RFP can or will not cipher the call), the call will be allowed to continue.
- Encryption enabled – in case of encryption reject (PP or RFP can or will not cipher the call), the call will be released.

Encryption is turned off by default. We recommend that you use RFP5 (SW part number: 14170201) SW PCS03C_ or newer.

- Authentication of incoming and outgoing calls.

In older SW versions authentication is only done when a PP is turned on (doing a location registration – and receiving standby text). With the current SW release, it is also possible to authenticate a PP when starting or receiving a call.

- Configuration of IP addresses gateway and netmask.

Removed Features

None.

Corrections and Improvements

- In analog configurations, the PP's display is cleared before the calling party number is shown.
- In analog configurations, the PP will - in case of a incoming call - follow the ring voltage on the analog line more precisely than before.
- Automatic configuration of a modem is working again.
- When connecting via telnet, the service would some time hang.
- Faster generation of email reports.
- Setting the DTMF pulse, pause and level had no effect.
- The AB default command is now working as it should.

Configuration File Parameter Changes

None.

Version PCS01N_

Added or Changed Features

None.

Removed Features

None.

Corrections and Improvements

- An error in analog call handling sometimes caused a restart of the system.

Configuration File Parameter Changes

None.

Version PCS01M_

Added or Changed Features

None.

Removed Features

None.

Corrections and Improvements

- Now supports pause digit in MSF (call back number).
- Corrected error regarding manual restart of RFP’s. When a single RFP was reset from OAM program, it did not come up again.
- Automatic restart/reboot on SW crash. If the KWS8000 SW crashed, the system needed manual power cycle.

Configuration File Parameter Changes

None.