InterTel Release Notes

Build 100.015 November 10,2006

Application Changes:
- Fix for bug 3811.   Assert with invalid icon blink rates.
- Fix for bug 3946.   OAI keys transmitted after a call has been established and audio is flowing are sent to the PBX as AXXESS commands, but not also in the RTP stream. This is needed to deliver additional dial digits to the answering device after the call has been established.

Core Changes: Core 6.7.10.

Known Problems:

Build 100.014
(notes to be copied into this file)

Build 100.013 March 23, 2005

Known Problems:
1. **Handsets have been observed to lock up in the battery charger (Bug 1242)**
2. **Handset may reboot if it tries to make a call and the AP is full (Bug 1379)**
3. **If DHCP server fails, handset may continue to operate with an expired lease (Bug 1428)**
4. **The handset has been observed to Assert ring.c ln 1411 when pressing the MENU key after silencing ringing. (Bug 1419)**

Core Changes
- Support for H340 health care handset
- Support for NetLink2.0 feature set including:
    - Enhanced Push-to-Talk
    - Adjustable transmit power
    - Keypad lock
    - Additional ringing options
    - Diagnostic Mode
    - Syslog Capability
    - Enhanced Site Survey
- Support for WMM basic
- Support for WPA and WPA2 with pre-shared key
- Enhanced performance of echo canceller.

Version 100.012 July 28, 2004

Visible Application Changes:

- Converted to using SVP version 2. This requires the use of SVP Sever 17x firmware. We now use the assigned UDP receive port number (defaults to 5567) as these no longer need to be unique. UDP commands that the PBX was previously directing to the assigned UDP port should now be properly recognized. Phones are now identified by unique "alias" IP addresses as assigned by the SVP server Fixed advertising our RTP IP receive address to be our Alias address instead of the SVP address required for SVP version 1. This fixes Peer-to-peer audio using SVP version 2.
- Added generation of a warning tone in the audio stream before playing the Out-of-service tones. This applies to both midstack-detected "No network found" and "No SVP Response" errors and to application-detected loss of communication with the PBX. This tone is generated only if the phone is known to be in a state in which it is likely to be up to ones ear. This feature depends on protocol notification of link-up and link-down events. The warning tone terminates after 8 seconds. The warning tone also is terminated if the user presses the END key to return to the standby state.
- Convert standby and admin menus to new vertical menu orientation
- Added any-key-answer feature. If a phone is ringing and a headset is present, the phone can now be answered by pressing the START key or any other dial key. (Note this only works from the standby screen, not from the active on-hook screen)
- Added display and functionality while the phone is ringing. As with LINE, FCN, and MENU keys, if a soft key is pressed while the phone is ringing loudly, the ringing will be converted into an in-the-audio-stream tone.
- Added support for Cisco fast secure roaming (FSR).
- Adjusted audio levels. Adjusted tones and beeps volume levels.
- Added 802.1g compatibility.
- Add display of alias IP and current IP to standby menu
- Improvements in battery life

Internal Application Changes:

- Moved application qos state machine to core.
- Fixed assert that could occur if the network was going up and down at initialization time -- allow initialization to occur only once even if we get more than one midstack callback for initialization complete.
- Fixed assert if the push-to-talk is pressed before protocol initialization is complete.
- Fixed potential memory assert if many dial keys are pressed abusively fast after pressing START and dialing your own extension number. This was accomplished by reducing the number of LCD refresh commands that are issued to update icons and update only the text portion of a screen.
- Randomized the TCP receive port number within a range of 255 ports. This is to minimize the chances of attempting to use the same TCP connection that

was previously used if the phone is rebooted and the previous TCP connection has not timed out.

- Converted entire application to use display phrases extracted from the international phrase table (phintl file). This requires an update to the phintl file that will be downloaded to the phone from the TFTP server.
- Converted display menu for FCN key, softkeys, and key handling to use our generic library routines for handling a "vertical" menu.
- Fixed a problem in which OAI can enable the protocol application to control the display even if the protocol application is in standby. In this case, the protocol application will simply repaint the standby display.
- Modified displays for both outgoing and incoming calls when we are waiting for QOS verification (QOS implies available bandwidth on an AP). The new QOS mechanism allows for us to continue waiting indefinitely for QOS verification. New key handling is provided to allow the user to terminate these states by pressing the END key. We now detect explicitly if the SVP server is locked and display a message accordingly. Again the END key must be pressed to terminate the wait for QOS and return to standby.
- Converted to using new core functionality for generating and detecting RFC2833 DTMF commands within the RTP stream.
- Added calls for setting the microphone noise level (microphone sensitivity) at initialization and whenever the setting changes in standby menus.
- Changed all tones generated by the protocol application to be played at "call progress tone" priority. This is done to ensure that alerting priority tones played by other applications (TWR, low battery, etc) do not interfere with or cancel our tones. This change affects the sound of the "howler" tone, as we can now play only two frequencies simultaneously instead of four.
- Fixed erroneous volume settings while alerting.
- Added audio path setting and restoring whenever a keytone is played. This allows keytones to be always played on the correct speaker while other audio paths are restored when the keytone terminates.
- Fixed mapping of desk set lamp numbers to features on the wireless phone. The appropriate feature on the FCN menu display page now properly indicates the on-off status of the corresponding Intertel lamp.
- Converted to using function calls to ARB for controlling the message waiting icon.
- Changed refresh softkeys function to clear softkey labels if we have lcd enabled but do not own the phone (keypad).
- Added explicit clear of up_down arrow icons when we are not showing a vertical menu.
- Fixed a minor memory leak having to do with queried non-volatile parameters at startup time.
- Fixed a potential problem in which an allocated memory pointer was freed just before it was used for the last time.
- Added a call to initialize the standby screen when registration completes in case the phone never got ownership of the LCD resource while registering.

This can happen if the push-to-talk button is held ON for the entire duration of the power-up initialization and PBX registration process.

Core Changes:

- Added support for generating and detecting RFC2833 out-of-band DTMF signaling and application interface.
- Improvements to headset detection.
- Fixed TCP far-end-close assert problem.
- Fixed uninitialized softkeys assert problem.
- Fixed assert discovered at Inter-Tel show in January.
- Added protocol notification of link-up and link-down events.
- Fixed assert in ptb24xx.c due to DCA race condition.
- Added throttling of midstack heap usage at 60% utilization.
- Fixed fallback to 2Mbps data rates.
- Fixed rejection of badly formed Beacon/Probe Response.
- Fixed core interactions of key-tones and alerting tones.
- Added application call to midstack to force an SVP keep-alive sequence. This will allow applications to initiate midstack recovery operations if the application discovers a communication problem before the core does.
- Fixed possible audio DTMF assert that could occur at startup of an audio stream if the user previously rapidly pressed dial keys while in a call and pressed the END key before the dtmf keys were processed.
- Fixed possible assert in standby.c having to do with crypto activity enqued while the phone is entering the low power standby state.
- Fixed Push-to-talk memory leak
- Fixed interaction with protocol audio tones if PTT call is received while protocol application is playing tones.
- Added support for handling the message-waiting icon to arbitrator. This allows shared usage of the message-waiting icon by OAI and protocol.

Downloader Changes:

- Fixed support for APs that support 1 and 2 Mbps only data rates.
- Extended TFTP timeouts to 2 seconds.

Version 100.011 January 6, 2004

- Fixed problem of slow registration across subnets
- Fixed assert caused by interrupt coherency problem
- Fixed audio level bug in I640 phone - audio level is now about 1 volume setting louder (~5db louder)
- Made OAI static address config menu consistent with other address config menus