F1000 Release Letter

Revision 0.1

Release Number <V4.50st >

Revision History

Revision #	Author	Reviewed By	A. Description Of Change B. Summary of Review	Issued By / Issue Date
0.1	Channu Sannappanavar		V4.50st Release Notes for F1000 handset	2006-7-06

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1. OVERVIEW

This is external release letter for UTStarcom F1000 release package V4.50st.

This release has a fix to prevent downloading of F1000G firmware into F1000 handset. As a result once the F1000 handset is upgraded to 4.50st release, it can not be downgraded back to previous release using local or remote firmware upgrade procedures.

2. RELEASE CONTENT

2.1 NEW FEATURES AND FUNCTIONS

2.1.1 F1000 Release V4.50st

- **Incoming RTP port:** F1000 didn't accept RTP message from other end if its originating RTP port did not match negotiated RTP port. This restriction is removed in this version and handset doesn’t check if originating port of incoming RTP messages matches negotiated port in SDP.
- **WPA enabled AP connectivity:** Sometimes handset could not connect to the AP which had WPA security turned ON and SSID broadcast disabled. This issue is fixed in this version.
- **WPA Key hiding:** Access to WPA key in SSID profile configuration is enabled only after entering user code. This restriction is added to prevent WPA key access by unauthorized person thro’ handset menu, which could be used to access Enterprise or user WLAN network.
- **Dynamic payload type** for telephone event in SDP: Apart from 100 and 101, F1000 should accept any payload type selected by SIP server.
- **SIP 302:** SIP 302 Redirect/Moved temporarily message is supported.
- **Removing eZi mode:** eZi predictive text input option is removed from Address book and other configuration menus since it is not used.
- **NULL Encryption key:** Default encryption key in F1000 is NULL and in previous firmware version NULL key was used to decode provisioning file received from provisioning server. In version 4.50st NULL key is treated as no key or no encryption. So whenever F1000 has NULL key (or no key) as an encryption key, F1000 will look for ut<MAC>.ini file on the provisioning server. When encryption key exists (other than NULL) then handset will look for encrypted ut<MAC>.aes provisioning file and decode it using configured encryption key on the receipt of the file.
- **Block F1000G firmware download:** This version blocks downloading F1000G firmware into F1000 handset. Due to this modification, F1000 handset can not be downgraded using remote or local firmware upload methods below 4.50st firmware version. Even in the future firmware versions, handset can be downgraded to firmware version up to 4.50st version but not below 4.50st version.
- **Close the telnet port:** Access to telnet port is disabled.
- **Display Sip Username on main display:** Add a menu in "Settings->Scheme" level for users to enable or disable the main display to show SIP username.
- **Version No. in SIP UA:** F1000 version number is added to SIP User agent field in all the SIP messages from F1000.
- **“Allow” field in SIP messages:** “Allow” field is added in SIP INVITE and other messages.

2.1.2 F1000 Release V3.80st

- Admin password length is increased to the maximum length of 20 characters instead of 8.
- 40X challenge for Call transfer call is supported.
- CSeq would not be increased in BYE message on call transfer.

2.1.3 F1000 Release V3.70st

- **Refer/Replace method** is implemented for Consult Transfer calls.
- **RTCP BYE** message is implemented according to RFC 3550.
- "**SIP Terminal Use NULL Packet**" parameter (in Web Interface and in configuration file) is implemented to enable or disable SIP NULL packets sent to the SIP server every 6 seconds.
- Port information is removed from request-uri of Register message.
- Calling Number sends escape sequence "%23" instead of "#" sign when called number is dialed starting with # sign.

2.1.4 F1000 Release V3.68st

- Hot spot login script can be uploaded thro' web interface. The script should begin with "##<!@UTSTARCOM WIFI PHONE". Please note that Hot spot login script has limited capability and presently it can support HTTP based hot spots.
- User can put "+" sign by holding * key for a longer time in phone number in address book.

2.1.5 F1000 Release V3.60st

- **Web Interface:** HTTP based web interface is added to provision F1000 parameters. There are two levels of access; user menu and advanced menu. "user" login has user menu access and default password is nothing but user security code. Default user security code is "**888888**". "admin" login has both advanced and user menu access and its default password is "**psw**".
- **Web Interface Capabilities:** Depending on access level (user and advanced) web interface has different provisioning and reset capabilities. These are as listed below:
  - user menu has Handset LAN configuration, Wireless AP related configuration, user settings and Call feature configuration, SIP and RTP configuration, Codec and DTMF configuration and STUN configuration
  - User menu also has generic information menu.
  - Through Advanced menu all the handset parameters, except remote TFTP provisioning parameters, could be RESET to default values.
  - Advanced menu can also modify "admin" password and configure remote TFTP provisioning server
  - Handset can be rebooted via web interface.
- **WPA-PSK Support:** WPA-PSK TKIP based encryption key is supported.
- **SSID Profile Menu Modifications:** Since WPA-PSK support is added, SSID profile configuration is modified in handset menu. Also WEP Authentication method (open or shared key) is added for each AP profile instead of global configuration for all the APs.
- **Spanish Version:** Separate load with Spanish/English language preferences is released as part of this release.
- RFC2833 DTMF functionality is improved with dynamic payload type.
- Modify the auto scan process to cut down the power consumption.
- **Configurable SIP timers:** SIP timer configuration is supported.
- **Increased RTP port range:** RTP port range is increased from 5K to 65535.
- Called number supports leading "+" sign.
- User can enter "+" sign by holding * key for a longer period of time.
- **Improvements of Call Features in STUN Mode:** Call hold/unhold, Call waiting and 3-way call features are improved when handset is operating in stun mode

2.1.6 F1000 Release V3.10st

- **Increased Volume at earpiece**: DSP receiver gain is slightly increased to increase the volume at earpiece.
- **SNMP and rlogin is disabled**: As this could have been security loophole, these ports on F1000 are disabled to stop unauthorized access.
- **Support for WWW-Authenticate:** WWW-Authenticate tag in Register and INVITE message is supported in this release.

2.1.7 F1000 Release V2.90st

- **Remote HTTP Configuration and firmware upgrade**: This works similar to remote TFTP configuration and upgrade. Operators can select either TFTP or HTTP for remotely managing provisioning and firmware on F1000.
- **"qop" tag:** "qop" tag for Proxy-authentication is supported in SIP INVITE message also. Previously it was supported in only SIP Register message.

2.1.8 F1000 Release V2.80st

- **STUN:** Implementation for STUN is added. In WiFi-Settings->Signal Protocol menu, there is new menu "NAT" under which STUN menu is introduced. STUN Server Name, Server Port and STUN On/Off options are provided in this menu.

  STUN implementation has a known issue due to which if phone is idle for some time, RTP port mapping becomes invalid thereby causing no audio during the call.

- **WiFi Driver Upgrade:** Driver is upgraded to the the latest version from Agere.
- **Local TFTP Update:** Local TFTP update under MISC menu is "Access Code" protected. Default access code is "888888".
- **Remote TFTP:** "Remote TFTP Server" menu is added in MISC menu to configure remote TFTP server FQDN from which handset can upgrade the software and configuration.
- **DNS CNAME:** This release supports DNS CNAME.
- **DNS SRV:** Support for DNS SRV.
- **Web Authentication and Hot spot login:** There is a partial implementation of menus and software for this feature. It still doesn't work.

2.1.9 F1000 Release V2.45st

- **Remote TFTP:**Complete support for TFTP provisioning and TFTP firmware upgrade. Please refer to "**F1000 Remote TFTP Process Guide**" for detailed procedure.
- **G726:** Added support for codec g726
- **3-way Call:** Added dual channel support (3-way call) for codecs g726 and g729.
- **Volume:** Increased the volume on F1000 receiver.
- Implemented SDP negotiation even after start audio session according to 200Ok(offer)/ACK(answer) model.
- "**Time Format**" is moved from "Settings" menu to "Tools" menu.
- New parameter "**Incoming call timer**" and corresponding functionality is added. Default value of this parameter is set to 60 sec. Refer TFTP user manual for description of this parameter.
- New parameter "**Outgoing call timer**" and corresponding functionality is added. Default value of this parameter is set to 60 sec. Refer TFTP user manual for description of this parameter.
- Repeat dial on busy has two new parameters: "**Set Repeat Dial on busy interval**" and "**Set Repeat Dial on busy count**". Please refer to "**Generic Configuration Parameters**" for description of these parameters.
- Support for entering **'+' character** by holding "*" key for brief interval.
- **Time zone** screen is altered to add more information and additional time zones.
- Wepkey index is modified to 1~4 (similar to popular APs and wireless routers) instead of 0~3.
- Registering screen shows "Registering via <AP name>" instead of "Registering to <AP name>".
- "Call Transfer" in "Network Service" menu is added new option to support **both Blind and Consult call transfer**
- Telnet user/password is modified from standard target/password.
- Parameter "**Terminal use NAT**" can be set to "0" to **disable SIP NULL packets**. ("SIP terminal use NULL packet" is not used for disabling NULL packets)
- Codec order could be set to "0", to prevent F1000 from publishing it during SDP negotiation.
- **DNS Failover:** Complete DNS fail over function is added.
- Added protection to local log feature, which was leading to f1000 crash in previous version.

- **In band DTMF:** Added parameter "DTMF transfer mode" and support for Inband (PCM) DTMF digit transfer along with existing DTMF transfer mode based on RFC2833. Refer to TFTP user manual for description.
- Partial support for **RTP statistics** function in syslog.
- **MWI:** Modified the header in NOTIFY message from "voicemail" to "Voice-Message" to support MWI functionality according to RFC specification.
- Display message on screen in case F1000 can't find AP in roaming state

2.1.10 F1000 Release V2.20st

- **Remote TFTP for firmware upgrade and provisioning**
- **DST :** Daylight Saving Setting can be turned ON or turned OFF in "Tools" menu
- **Local Log:** "MISC" menu has a local log. Local log shows last twenty Registration and SIP messages that were sent and received on the handset.
- **Sys Log:** This release implements logging of the debugging messages from handset to the carrier's server configured in the handset. Carrier network is required to support this feature in order to work.
- **MISC menu:** New main menu for miscellaneous item is added. This menu contains "Vendor Information", "PC Config", "Remote TFTP", "Local TFTP" and "Local Log"
- **SIP Failover:** F1000 supports SIP server failover to secondary SIP server in case of no response and some of the 400/500/600 error messages. This feature needs to be supported by carrier's network for proper operation.
- **Repeat Dial on Busy**
- **SO Writer (Provisioning) software s1.3** to support additional parameters like SIP Authentication, Call features and Codec selection
- **Time Format:** "Setting" menu has a "Time Format" selection to select either '12 Hour' or '24 Hour' format to display the time on handset.
- **Configurable SIP Timers:** Timers T1, T2, T4, B, F, H, D and Timer J can be configured thro' *.ini file and SO Writer utility and also using remote TFTP provisioning process.

2.1.11 F1000 Release Vs2.1

- **SIP Authentication:** Some of the SIP Proxies use separate "SIP Authentication" string other than "SIP user name" for user authentication. In this release new field called "SIP Authentication" is added in WiFI Settings->Signal Protocol -> SIP menu. The string value configured in this field is used for user SIP authentication. ***Note: Please configure "SIP User Name" in "SIP Authentication" field if separate "SIP Authentication" is not required.***
- **Security Code Fix:** Security code padding issue, which added 8's on the RHS of security code in case it was smaller than 6 digits is fixed
- **Beep on Connecting to AP:** If in the "Profile" menu "Alert Tone" is turned ON then every time handset connects to AP and registers with SIP server, handset will beep.
- **Layer 2 Handover:** In an enterprise network where Access Points (APs) are just configured as APs with same SSID and do not have either DHCP or routing functionality, in such network F1000 can move from one AP to another AP without dropping the call.
- **Roaming to new AP:** As soon as handset goes out of range from currently connected AP, it will start 'AP connection' process again and if handset has Auto Scan enabled, then it goes thro' search for open AP after trying SSIDs in the profile.
- **Call Tranfser Consultation**
- **FQDN support for SNTP server domain**
- **WEP Key Index Support:** Now all 4 WEP Key Index can be selected and configured.
- **RTP Port Negotiation fixed:** This issue observed in Vs1.8 is fixed

2.2 FEATURES

F1000 provides the following features and functions:

- Call Features
  - Basic Call setup
  - Call Waiting
  - Blind Call transfer
  - Consult Call transfer
  - Three Way call
  - Call Hold and unhold
  - Mute capability while in conversation
  - Repeat dial on busy
  - Caller ID Block
  - Reject Anonymous Calls
  - SIM (Simultaneous) Ring
  - Shared Call Appearance
  - Alternate Number

- Call Processing:
  - SIP protocol (RFC3261 and RFC3264)
  - SDP for codec negotiation (RFC 2327)
  - RTP for media transfer (RFC 1889)
  - RTCP for RTP statistics collection (RFC 1890)
  - SNTP
  - DNS failover
  - DNS SRV
  - FQDN support for SIP server and Registration server
  - SIP server fail over
  - Handover between APs with same SSID, Wepkey and same subnet
  - MWI-Message waiting Indicator

- RF and Power Management:
  - Output Power: 20mW
  - 80211b at 2.4GHz
  - Receive Sensitivity: -91dBm at 1mbps, -83 dBm at 11mbps
  - Battery:Li-ion DC 3.6V 1500mAh
  - Battery Life: 4 Hours Talk time, 100 Hours Standby time
  - Battery Charge Time: 2-3 Hours
- Codec and Voice Processing:
  - G711u-law
  - G711a-law
  - G729
  - G726
  - RFC 2833 for DTMF encoding
  - In band (PCM) DTMF
  - Comfort Noise Generation (CNG)
  - Voice Activity Detection (VAD)
  - Adaptive Jitter Buffer
  - Echo Cancellation

- .Various Indicators on first row of the screen:
  - Antenna strength indicator to connected AP
  - Battery Life indicator
  - Screen Lock indicator
  - Ring mode and Vibration mode indicator
  - Message Waiting Indicator
  - Alarm indicator

- Provisioning Capability:
  1. SO writer hardware along with provisioning software st1.5 can be used to provision SIP parameters, Username/Password , SSID, DHCP/static IP, DNS, SIP and RTP port etc on F1000.
  2. All the above-mentioned parameters and more can be provisioned thro' F1000 menu also.
  3. Remote TFTP process could be used for secure provisioning over Internet.
  4. HTTP based Web interface could be used to configure most of the parameters.

- Software Upgrade Capability:
  1. SO writer hardware along with aFlash Utility can be used for upgrading the software.
  2. Local TFTP utility for software upgrade.
  3. Remote TFTP utility

- Supports four set of AP configuration (SSID and WEP Key)
- Support for Auto Scanning for open AP
- Supports NULL, 64 bit and 128bit WEP key encryption, 1 to 4 index keys and open or shared key.
- Support for TKIP based WPA-PSK encryption.
- Support for 12 and 24 hour time format.
- Call Log: Dialed, Missed, Received and Delete call log and Duration of call
- Calculator, Calendar and Alarm in Tools menu
- Phone Book
- Speed Dial
- Ear Piece
- Partial WiFi-Settings parameter can take effect without reboot
- Power on can be enabled to be User Code protected
- Supports various user modifiable profiles and user modifiable setting. Please refer to the user guide for learn about these options.

2.2.1 Hardware

F1000's hardware platform is based on
Agere's T8307BWB(VoIP processor)
CSP8307C1(Analog conversion and power management IC)
WL600411(MAC)
WL1141 (802.11b physical module)

2.3 PROBLEMS RESOLVED

2.3.1 V4.50st

- When ring mode is set to vibration, and incoming call is rejected by pressing "hang up", the vibration is still extended even after pressing "hang up" (red) key. This problem is resolved in 2.70st version.
- After powering on the handset, if entered into the MISC->Local TFTP menu and stayed there, when the handset is registered successfully, then the screen was modified to show 'Parse the config' instead of 'Enter Code' screen.
- Sometimes if call between two F1000 subscribers (A-calling party and B- Called Party) is picked up within one ring on called party (B) handset , then A to B there is no voice transmission, and B to A has very quite volume. This happened in very few scenarios and it is fixed in this version

- Interaction between Mute and hold/unhold: When both the parties in a call are turned "Mute", pressing hold/unhold few times could lead to two way voice transmission despite phone is still in "Mute" state. This interaction is corrected in this version.

- Syslog Messages: Some messages were not transmitted to syslog server and some RTP flow related messages were not transmitted fully to the syslog server. This is fixed in 2.70st.

2.3.2 V3.80st

- While downloading the firmware via remote TFTP or HTTP, display on the handset's screen was not complete. It was 'downloading f...' and in V3.80st it is corrected to display 'downloading firmware....'
- While setting up WiFi Parameters on Web interface, sometimes handset would crash. This is fixed in this release.
- Web interface would not allow Terminal Subnet Mask anything other than 255.255.255.0, even though it is valid subnet mask.

2.3.3 V3.70st

- During remote TFTP/HTTP configuration sometimes invalid parameters were not causing the parameter to be defaulted. This issue is fixed in this release.
- During remote configuration some invalid parameters would cause handset to reboot. Now if parameter is invalid then its value is defaulted.
- Handset menu can accept RTP port number from 10000 to 50000.

2.3.4 V3.68st

- In case AP/router looses it's WAN/Internet connection, F1000 will identify it during next Registration cycle and will go into registering mode. Handset will display "Registering via SSID" for some time while it continues trying to register via same SSID. After that it will display "Couldn't connect Try another AP"
- Extended RTP port range (5000 to 64K) can be entered thro' Web interface. Handset menu still can not accept extended RTP port range.
- Fixed spelling mistakes in SIP menu in the handset.
- Parameter names in "SIP and RTP Config" menu in web interface are made consistent with handset menu.
- STUN query is renewed in case STUN is enabled when handset looses the registration.
- "TFTP Config" in "Advanced Menu" section of Web Interface can accept http.
- "Local Log" menu in the handset can use both "Next" (soft key) and left/right rocker key to navigate thro' the menu items.
- In handset Menu->Settings->Display->On/Off Animation show the correct animation pictures

2.3.5 V3.60st

- Fixed a bug that will stop ACK retransmission when doing swap operation during Call Waiting.
- Re-Registration interval would follow interval specified in 200 OK but "Expire time" in Register message will not be altered.
- Fixed a bug which occasionally used to lead to crash while releasing the call.
- Modified G726 payload type to 2 for rtpmap to G726-32
- Fixed a bug that will erroneously stop retransmission of 200 OK for invite.
- Fixed a bug to renew DHCP lease before it expires
- Fixed a bug to make sure that wifi driver is in wakeup state before net search.
- Fixed bugs in Session Audit functionality.
- Improvement in failover operation.
- During auto scan, if user activates connection to any AP, the handset will try to connect to that AP immediately regardless of current state of handset (connecting, registering, registered)
- It is ensured that the ARP request is sent out when doing ARP resolve in roaming state,
- If AP is not found in net search state, then WiFi module is reset to make sure that AP unavailability is not due to WiFi module error.

2.3.6 V3.10st

MR Number	Problem Severity	Problem Description	Notes (including impact and potential work around)	Status
		Remote HTTP provisioning does not work.		Fixed
		Voice quality is not good on certain networks		Fixed.

2.3.7 V2.90st

MR Number	Problem Severity	Problem Description	Notes (including impact and potential work around)	Status
		When STUN is enabled in the handset, if phone is kept idle for some time before making calls, there will be no audio, since the RTP port mapping is no more valid		Fixed.
		Call hold does not work on certain networks.		Fixed.
		WiFi setting menu sometimes displays incorrect information		Fixed
		Local log display does not work properly		Fixed
		In case call waiting is enabled and during the call on F1000 if call waiting indication is received for another incoming call, even though 'call waiting' call is not picked up, after indication (beep) stops there is one way audio in existing call.		Fixed

2.3.8 V2.80st

MR Number	Problem Severity	Problem Description	Notes (including impact and potential work around)	Status
		Delayed cut thro' when SIM ring is enabled		Fixed

2.3.9 V2.40st

MR Number	Problem Severity	Problem Description	Notes (including impact and potential work around)	Status
		SIP timers are not accurate because of higher clock resolution.		Fixed. Clock resolution is increased from 1 second to 100ms.
		Display Name is removed from menu tree.		Fixed. “Personal Information” menu is added in “Network Services” menu
		F1000 does not respond with ACK for 200 OK message in case F1000 hangs up (sends CANCEL message) while Called party is ringing.		Fixed
		SIP messages are retransmitted inaccurately and at incorrect interval.		Fixed
		Hold/unhold fails between two F1000s when using 2 different codec preferences on each of the handset.		Fixed
		Sometimes incoming **OPTION** message will lead to f1000 crash		Fixed
		F1000 reboots while receiving the compound RTCP packets.		Fixed

2.3.10 V2.20st

MR Number	Problem Severity	Problem Description	Notes (including impact and potential work around)	Status
		CNG packets (payload type 19) generated loud noise on F1000 side		Fixed. Since DSP handles CNG internally, CNG packets from network are drpped without processing.
		Order of the codec selection could be modified but none of the codec could be disabled using *.ini provisioning file and SO Writer utility		Fixed

2.3.11 Vs2.10

MR Number	Problem Severity	Problem Description	Notes (including impact and potential work around)	Status
		RTP port switching after voice path is established does not work reliably		Fixed
		“Set Security” in Settings menu does not work properly. By default security code is “888888”. If security code is modified to less than 6 digits (eg. 1234) and PowerOn Code is turned ON, next time when handset is turned ON, it does not take 4 digit security code.		Fixed
		DTMF digit dialing does not work on some carrier’s network.		Fixed

3. SUMMARY OF KNOWN PROBLEMS AND WORKAROUNDS

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- After handset is fully charged, it reboots.
  Work Around: This problem is resolved once handset is “reset to default” thro’ advanced section of web interface.

- In this version F1000 supports dynamic payload type for EFC2833 DTMF event. But for outgoing call, during SDP negotiation for DTMF digits based on RFC2833, if server chooses (receive capability of) DTMF payload type anything other than what F1000 proposed as it’s receive capability in outgoing INVITE message, then F1000 does not transmit DTMF digits at all during the call.
  Work Around: This issue shall be fixed in later release.

- When handset is upgraded from older release like 3.10st, it doesn’t behave consistently. For example WPA doesn’t work properly or it doesn’t connect to AP consistently
  Work Around: This happens due to database mismatch of recent firmwares with older than 3.68st releases. This could be corrected by resetting the handset to default values thro’ “Reset to default” option in Admin section of Web Interface. This requires “Admin password”. Please note that upon resetting the default values all the configurations except user setting would be defaulted. So user should note down all the settings before resetting to default.

4. RELEASE PACKAGE

4.1 SOFTWARE

aflash utility for software upgrade
TFTP upgrade utility

4.2 FIRMWARE

Module	File Name	Version	File Size (MBytes)
Firmware for serial-port Download	f1000_software_4.50st.sre	V4.50st (English/ Chinese version)	14.2
Firmware for TFTP Download	F1000_Firmware.bin.Z	V4.50st (English/ Spanish version)	1.81

4.3 DOCUMENTATION

4.3.1 ENGLISH MANUAL

"F1000 User Guide" is added along with each handset in a box. It is also available at http://www.utstar.com/Document_Library/F1000_User_Manual.pdf

4.3.2 Release Package Distribution

The release package is being made available through the following two methods:

Internally, released through Release Tool.
Externally, released through Email Distribution.

5. INSTALLATION / UPGRADING INSTRUCTIONS

Software Upgrade is supported through following methods:

- Single handset, software upgrade using TFTP download via 802.11b. Please refer to the User manual.
- Single handset, software upgrade using SO writer and "aflash" utility. Please refer to the User manual.
- Upgrading multiple handsets using special 8 serial ports PCI board and SO writer. Please refer to the user manual for setting up and upgrading procedure.
- Upgrading multiple handsets using remote TFTP/HTTP method. Please refer to "F1000 Remote Management" document.

6. REFERENCE DOCUMENTS

1 TFTP upgrade manual
2 "aflash" programming utility user guide
3 F1000 User Guide
4 F1000 Quick Start Guide
5 F1000 Remote Management Guide
6 Generic Configuration Parameters
7 WiFi Handset forum at http://userforums.utstar.com/index.cfm?frmid=27 (registration using email address is required)