Mitel 6800 Series SIP Phones

58015548 REV 00

4.5.0 RELEASE NOTES

Mitel 6800 Series SIP Phones 4.5.0 Release Notes

58015548 REV 00 - November 2017

ABOUT THIS DOCUMENT

This document provides details on the new features, enhancements to existing features, and/or issues resolved for the Mitel 6800 series (6863i, 6865i, 6867i, 6869i, and 6873i) SIP phones for Release 4.5.0.

Note: This release applies to the phone models mentioned above only.

For more detailed information about features associated with each phone, and for information on how to use the phones, see your model-specific *SIP Phone Installation Guide* and the *SIP Phone User Guide.* For detailed information about more advanced features, see the *6800 Series SIP Phone Administrator Guide* and/or the *Development Guide XML API For Mitel SIP Phones.*

RELEASE NOTES TOPICS

Topics in these release notes include:

- General Information
- New Features in Release 4.5.0
- Additional Information
- Issues Resolved in Release 4.5.0
- Contacting Mitel Support

GENERAL INFORMATION

RELEASE CONTENT INFORMATION

This document provides release content information on the Mitel 6800 series SIP phone firmware.

MODEL	RELEASE NAME	RELEASE VERSION	RELEASE FILENAME	RELEASE DATE
6863i	Generic SIP	4.5.0	58015543-REV 00	November 2017
6865i	Generic SIP	4.5.0	58015544-REV 00	November 2017
6867i	Generic SIP	4.5.0	58015545-REV 00	November 2017
6869i	Generic SIP	4.5.0	58015546-REV 00	November 2017
6873i	Generic SIP	4.5.0	58015547-REV 00	November 2017

HARDWARE SUPPORTED

This release of firmware is compatible with the following Mitel SIP portfolio products:

- 6863i
- 6865i
- 6867i
- 6869i
- 6873i

BOOTLOADER REQUIREMENTS

This release of firmware is compatible with the following Mitel SIP portfolio product bootloader versions:

- 6863i: Boot2 1.0.0.0 or higher
- 6865i: Boot2 1.0.0.0 or higher
- 6867i: Boot2 1.0.0.6 or higher
- 6869i: Boot2 1.0.0.6 or higher
- 6873i: Boot2 1.0.1.9 or higher

IMPORTANT 6800 SERIES SIP PHONE FIRMWARE UPGRADE INFORMATION

WARNING: DO NOT ATTEMPT TO UPGRADE YOUR PHONE TO RELEASE 4.5.0 FROM A RELEASE LOWER THAN 4.3.0 WHEN IN WEB RECOVERY MODE. DOING SO WILL CAUSE YOUR PHONE TO BECOME NON-OPERATIONAL.

IMPORTANT M685I EXPANSION MODULE FIRMWARE DOWNGRADE INFORMATION

If you upgrade your phone from a release lower than 4.3.0 to Release 4.5.0 and an M685i Expansion Module is attached, the M685i Expansion Module will also upgrade to align itself with the new UI changes.

If downgrading the phone with an M685i Expansion Module from Release 4.5.0 to a firmware version of Release 4.1.0 (or below), it is required to first downgrade to a Release 4.1.0 Hot Fix or 4.1.0 Service Pack and then to the Release 4.1.0 (or below) firmware version in such scenarios.

This will ensure the UI of the M685i Expansion Module is aligned with the UI at all times.

NEW FEATURES IN RELEASE 4.5.0

This section provides the new features in SIP Phone Release 4.5.0. The following table summarizes each new feature and provides a link to more information within this release note. Each feature also specifies whether it affects the Administrator, the User, or the XML Developer.

FEATURE	DESCRIPTION
SRTP AES_256_CM Encryption Support (For Administrators and Users)	Release 4.5.0 provides SRTP AES_256_CM encryption support to SIP phones which offers capability with current AES 128 implementation from the application level. ***New for 6867i, 6869i and 6873i SIP Phones.**
Support TLS v1.2 For 802.1x EAP (For Administrators)	With Release 4.5.0, the 6800 series SIP phones support TLS 1.2 for 802.1x EAP. ***New for all 6800 series SIP phones.**
DNS Parameter update in DHCP Lease Renewal-DHCP ACK (For Administrators)	In release 4.5.0, SIP phones update the DNS parameters in DHCP ACK through DHCP release renewal when the user changes DNS settings on the DHCP server. ***New for all 6800 series SIP phones.**
DNS-SRV handling for different 5xx error conditions (For Administrators)	In Release 4.5.0, 6800 series SIP phones performs DNS-SRV failover for all SIP messages when a 5xx is received. A new DNS SRV failover mode named DNS-SRV failover-follow-registration is added to the SIP engine. ***New for all 6800 series SIP phones.**
XSI Application Update For Call ID Behavior With Multiple Accounts (For Administrators)	In release 4.5.0, an enhancement is made to the existing XSI application (directory, call log and speeddial8) to consider screen focus while dialing out as opposed to using the first idle line when using multiple accounts or XSI directories. ***New for all 6800 series SIP phones.**
SIP Unregister on boot or reboot (For Administrators)	In Release 4.5.0, a new boolean configuration parameter "**sip unregister on boot**" is introduced to enable or disable de-registration of SIP phones at boot or reboot (manual, remote or electrical). ***New for all 6800 series SIP phones.**
Call Forward Busy or Call Forward No Answer option to turn off MWI LED (For Users)	With Release 4.5.0, when Call Forward is set to either “Call Forward Busy” or “Call Forward No Answer” options, the MWI LED on the phone is turned off. ***New for all 6800 series SIP phones.**

FEATURE	DESCRIPTION
Hash Password Support For SIP Authentication (For Administrators)	Release 4.5.0 supports configuration of digest authentication a1 hash as opposed to the clear text password. A new configuration parameter "**sip lineN hash**" is introduced in this release to support provisioning of digest authentication (a1 hash). ***New for all 6800 series SIP phones.**
LED Behavior In a Failover Scenario (For Users)	In release 4.5.0, an enhancement is made wherein during a failover from the current to an alternative registrar, the red LED in the upper right corner of a 6800 series phone is switched off unless the registration to an alternative server fails. ***New for all 6800 series SIP phones.**
BLF Behavior In a Failover Scenario (For Users)	In release 4.5.0, an enhancement is made wherein during a failover from the current to an alternative registrar, the "???" symbol does not display next to the corresponding BLF key unless the registration to the alternative server fails. ***New for 6865i, 6867i, 6869i, and 6873i series SIP phones.**
Call Transfer Through DHSG Headset Release Key (For Users)	In release 4.5.0, an enhancement is made to the DHSG headset hook button to facilitate a call transfer on pressing the hook switch button. ***New for 6865i, 6867i and 6869i series phones.**
Telia CA Certificate Addition (For Administrators)	In release 4.5.0, Telia CA certificate is added to the Mitel repository for 6800 series SIP phones. ***New for all 6800 series SIP phones.**
Additional Parameters In VDP Filter List (For Administrators)	In release 4.5.0, additional configuration parameters are added to the VDP filtering feature for 6800 series SIP phones. ***New for all 6800 series SIP phones.**
Handle 5xx Response During VDP Login (For Administrators and Users)	In Release 4.5.0, an enhancement is made wherein during a VDP auto login procedure, when the phone receives a 500 or 503 response, the VDP user retries to auto login after every 300 seconds. ***New for all 6800 series SIP phones.**

ADDITIONAL INFORMATION

SRTP AES_256_CM ENCRYPTION SUPPORT

Release 4.5.0 provides SRTP AES_256_CM encryption support to the 6867i, 6869i and 6873i SIP phones which offers capability with current AES 128 implementation from the application level.

A new configuration parameter "**srtp aes256"** is introduced to enable the SRTP AES_256_CM support. Administrators can enable this functionality by defining the "**srtp aes256**" parameter as "**1**" in the configuration files. By default, this feature is disabled.

Enable or Disable SRTP AES_256_CM Support Using the Configuration Files

Use the following parameter to enable or disable the SRTP AES_256_CM support on the 6867i, 6869i and 6873i SIP phones.

PARAMETER – *srtp aes256*	**CONFIGURATION FILES** startup.cfg, <model>.cfg, <mac>.cfg
DESCRIPTION	Enables or disables the SRTP AES_256_CM support on the SIP phones.
FORMAT	Boolean
DEFAULT VALUE	0 (Disabled)
RANGE	0 - 1 0 (Disabled) 1 (Enabled)
EXAMPLE	srtp aes256: 1

Note:

1. The configuration parameter is not dynamic. Reboot the phone to reflect the parameter update.
2. The configuration parameter is to be only used by servers that support SRTP AES_256_CM.

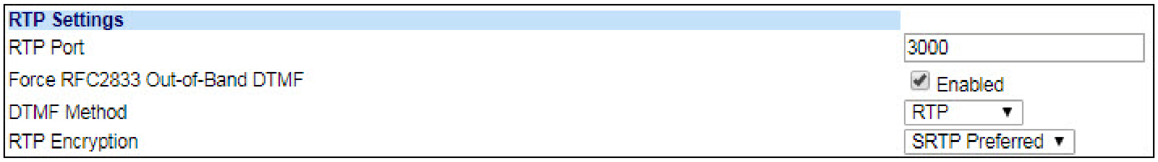
On the SIP phones, the RTP encryption should be set as SRTP Preferred.

Enable SRTP Preferred Option USING The Mitel Web UI

Use the following procedure to enable the RTP encryption to SRTP Preferred on the phone using Mitel Web UI:

1. Click on **Advanced Settings > Global SIP**

2. In the **RTP Encryption** field under **RTP Settings**, select **SRTP Preferred** from the drop down list.



3. Click **Save Settings**.

SUPPORT TLS V1.2 FOR 802.1X EAP

With Release 4.5.0, the 6800 series SIP phones support TLS 1.2 for 802.1x EAP.

At reboot, the phone processes 802.1xEAP and sends client hello with TLS v1.2. The Radius server sends back server hello and the phone processes it with TLS v1.2.

Note:

1. Install Radius server with 802.1x TLS v1.2 support on the user network.
2. If the radius server does not support TLS v1.2, the phone automatically uses the TLS version (802.1x EAP TLS v1.1 or 802.1x EAP TLS v1.0) supported by the server.

The following TLS 1.2 cipher suites are supported for this feature:

- Cipher Suite: TLS_DHE_RSA_WITH_AES_256_CBC_SHA256 (0x006b)
- Cipher Suite: TLS_RSA_WITH_AES_256_CBC_SHA256 (0x003d)
- Cipher Suite: TLS_DHE_RSA_WITH_AES_256_CBC_SHA (0x0039)
- Cipher Suite: TLS_RSA_WITH_AES_256_CBC_SHA (0x0035)
- Cipher Suite: TLS_DHE_RSA_WITH_AES_128_CBC_SHA256 (0x0067)
- Cipher Suite: TLS_RSA_WITH_AES_128_CBC_SHA256 (0x003c)
- Cipher Suite: TLS_DHE_RSA_WITH_AES_128_CBC_SHA (0x0033)
- Cipher Suite: TLS_RSA_WITH_AES_128_CBC_SHA (0x002f)
- Cipher Suite: TLS_DHE_RSA_WITH_3DES_EDE_CBC_SHA (0x0016)
- Cipher Suite: TLS_RSA_WITH_3DES_EDE_CBC_SHA (0x000a)
- Cipher Suite: TLS_RSA_WITH_RC4_128_SHA (0x0005)
- Cipher Suite: TLS_RSA_WITH_RC4128MD5 (0x0004)

DNS PARAMETER UPDATE IN DHCP LEASE RENEWAL-DHCP ACK

In release 4.5.0, the SIP phone updates the DNS parameters in DHCP ACK through DHCP release renewal when the user changes DNS settings on the DHCP server.

This feature is supported on the 6863i 6865i, 6867i, 6869i, and 6873i SIP phones.

DNS-SRV HANDLING FOR DIFFERENT 5XX ERROR CONDITIONS

Previously SIP phone models allowed any SIP request to trigger DNS SRV failover and treated only 503 status code in 5xx class of response as service unavailable.

In Release 4.5.0, 6800 series SIP phones perform DNS-SRV failover for all SIP messages when a 5xx is received. A new DNS SRV failover mode named DNS-SRV failover-follow-registration is added to the SIP engine. Also the service unavailable response fail over rule is now configurable in the SIP stack.

The key point of the feature is DNS SRV failover follows registration. Ensure the following configurations are specified on the SIP phone to ensure DNS SRV failover follows registration:

- “sip outbound proxy” is set to hostname
- “sip outbound proxy port” is set to 0
- no backup proxy or registrar is configured
- no backup outbound is configured

SERVICE UNAVAILABLE STATUS CODES

Service unavailable status codes indicate that the server service is unavailable. The SIP phone switches server when a SIP response with one of the status codes is received.

Administrators can define the service unavailable status codes rule by defining a new parameter “**sip service unavailable status codes**” in the configuration files.

Define Service Unavailable Status Codes Using Configuration Files

PARAMETER – *sip service unavailable status codes*	**CONFIGURATION FILES** startup.cfg, <model>.cfg, <mac>.cfg
DESCRIPTION	Configure service unavailable status codes on the SIP phone.
FORMAT	Integer
DEFAULT VALUE	N/A
RANGE	300 - 699
EXAMPLE	sip service unavailable status codes: 500, 501, 502, 503, 504, 505, 513

Note: 408 and 503 are not included in the parameter list but yet treated as service unavailable status codes.

SERVICE UNAVAILABLE RESPONSE FAILOVER RULE

A service unavailable response is a SIP response with one of the defined service unavailable status codes. The following are the two failover rule options:

- option1: failover only if the service unavailable response is the first response received.

- option2: failover when the service unavailable response is received and no response other than 100 trying responses are received for the SIP request.

Administrators can configure the service unavailable response failover rule by defining a new parameter "**sip service unavailable failover rule**" as "**1**" in the configuration files. By default this feature is disabled.

Define Service Unavailable Response Failover Rule Using Configuration Files

PARAMETER – *sip service unavailable failover rule*	**CONFIGURATION FILES** startup.cfg, <model>.cfg, <mac>.cfg
DESCRIPTION	Configure service unavailable response failover rule on the SIP phones.
FORMAT	Boolean
DEFAULT VALUE	0
RANGE	0 - 1 0 (failover only if the service unavailable response is the first response received) 1 (failover when the service unavailable response is received and no response other than 100 trying responses were received for the SIP request)
EXAMPLE	sip service unavailable failover rule: 0

DNS SRV FAILOVER MODE

Administrators can configure the DNS SRV failover mode by defining the parameter "**sip srv failover enabled**" as "**2**" in the configuration files.

Define DNS SRV Failover Mode Using Configuration Files

PARAMETER – *sip srv failover enabled*	**CONFIGURATION FILES** startup.cfg, <model>.cfg, <mac>.cfg
DESCRIPTION	Configure DNS SRV failover mode on the SIP phones.
FORMAT	Integer
DEFAULT VALUE	0 (Disabled)
RANGE	0 - 2 0 (DNS SRV failover is disabled) 1 (DNS SRV failover is enabled, current behavior) 2 (DNS SRV failover following registration, new behavior)
EXAMPLE	sip srv failover enabled: 2

XSI APPLICATION UPDATE FOR CALL ID BEHAVIOR WITH MULTIPLE ACCOUNTS

In release 4.5.0, an enhancement is made to the existing XSI application (directory, call log and speeddial8) to consider screen focus while dialing out as opposed to using the first idle line when using multiple accounts or XSI directories.

Screen focus is only considered when xsi is configured per line. This feature is supported only when “xsi allow sip authentication” is enabled.

In case of speeddial8, user can specify the line number as part of the speeddial8 softkey configuration. If the line number is configured as 'global', then the screen focus is used as the line for dialing out.

SIP UNREGISTER ON BOOT OR REBOOT

The SIP phones already support unregister on soft reboot. Enhancements are made in this release to support SIP unregister at electrical reboot.

A new configuration parameter “**sip unregister on boot**” is introduced to enable or disable unregistering of SIP phones at boot or reboot (manual, remote or electrical).

Administrators can enable this functionality by defining the “**sip unregister on boot**” parameter as 1 in the configuration file. By default, this feature is disabled.

Enable or Disable SIP Unregister on Boot Functionality Using the Configuration Files

Use the following parameter to enable or disable unregister on boot functionality on the 6800 series SIP phones.

PARAMETER – *sip unregister on boot*	**CONFIGURATION FILES** startup.cfg, <model>.cfg, <mac>.cfg
DESCRIPTION	Enables or disables unregister on boot or reboot functionality on SIP phones.
FORMAT	Boolean

DEFAULT VALUE	0 (Disabled)
RANGE	0 - 1 0 (Disabled) 1 (Enabled)
EXAMPLE	sip unregister on boot: 1

CALL FORWARD BUSY OR CALL FORWARD NO ANSWER OPTION TO TURN OFF MWI LED

With Release 4.5.0, when Call Forward is set to either “Call Forward Busy” or “Call Forward No Answer” options, the MWI LED on the SIP phone turns off.

The LED turns on only when Call Forward is set to “Call Forward All”.

HASH PASSWORD SUPPORT FOR SIP AUTHENTICATION

Release 4.5.0 supports configuration of digest authentication a1 hash as opposed to the clear text password.

A new configuration parameter “**sip lineN hash**” is introduced in this release to support provisioning of digest authentication (a1 hash). When this parameter is configured, sip lineN hash will override the existing parameter “sip lineN password”.

Configure Digest Authentication a1 Hash Using the Configuration Files

Administrators can configure the hash password support for SIP authentication by defining the following parameter in the configuration files.

PARAMETER – *sip lineN hash*	**CONFIGURATION FILES** startup.cfg, <model>.cfg, <mac>.cfg
DESCRIPTION	Password supported for SIP authentication in hashed format
FORMAT	Text
DEFAULT VALUE	N/A
RANGE	Alpha numeric characters
EXAMPLE	sip lineN hash: *************

The following lists the limitations to the new parameter:

- Applies to lineN and global line
- Works with the 401 response and 407 response
- If the configuration parameter “mask sip password” is enabled, when local.cfg and server.cfg are downloaded from the troubleshooting page in Web UI, the hash value is masked.

LED BEHAVIOR IN A FAILOVER SCENARIO

Previously, the red LED in the upper right corner of a 6800 series phone was switched on when the phone was not registered in a failover scenario. The LED is on for approximately 0.5 seconds during a failover from the current to an alternative registrar.

In release 4.5.0, an enhancement is made wherein the LED is switched off unless the registration to an alternative server fails.

If no backup or alternate server (through DNS SRV) is configured, phone will attempt to register to Registrar. If the registrar fails to respond, the phone will display “No Service” and lights up the MWI LED.

If the backup server or alternate server (through DNS SRV) is configured and the registration attempt to primary server fails (due to server maintenance etc), the phone attempts to register with the alternate server. Only if the alternate server registration attempt also fails, the phone displays “No Service” and lights up the MWI LED.

Note: In case of no response from the server, the phone will retry sending registration until the transaction timer expires, after which the phone displays “No Service” and lights up the MWI LED.

BLF BEHAVIOR IN A FAILOVER SCENARIO

Previously, the “???” symbol was displayed next to the corresponding BLF key when the status of a monitored SIP extension was unreliable. In a failover scenario, the “???” symbol is displayed during a failover from the current to an alternative registrar.

In release 4.5.0, an enhancement is made wherein the “???” symbol does not display next to the corresponding BLF key during a failover from the current to an alternative registrar, unless the registration to the alternative server fails.

CALL TRANSFER THROUGH DHSG HEADSET RELEASE KEY

With the previous SIP implementation, it is possible to answer and initiate calls using the hook switch button on the DHSG headset. However during a call transfer, pressing the release key on the DHSG headset released the call.

In release 4.5.0, an enhancement is made to the DHSG headset hook button to facilitate a call transfer on pressing the hook switch button.

TELIA CA CERTIFICATE ADDITION

In release 4.5.0, Telia CA certificate is added to the Mitel repository for 6800 series SIP phones.

The SIP phone includes the CA to the root certificates[] in certificates.c file. After addition of the certificate, it is automatically made available for users to use.

Note: Reboot the phone to download all configuration files added to the server.

HANDLE 5XX RESPONSE DURING VDP LOGIN

Previously, when a VDP user attempts to log in and receives a 503 or 500 response from VDP configuration server (HTTP or HTTPS only), the phone fails to send further requests and no longer auto logs in. This scenario is only applicable when high security is disabled.

In Release 4.5.0, an enhancement is made wherein during a VDP auto login procedure, when the phone receives a 500 or 503 response, the VDP user retries to auto login after every 300 seconds. The phone continues to retry until a positive response is received from the server.

ADDITIONAL PARAMETERS IN VDP FILTER LIST

When a hotdesk user logs in, all configuration changes made prior to log out need to be either uploaded to network, or saved to local config. Prior to Release 4.3.0 SP2, all hotdesk user's configurations were synchronized to the server, including the network dependency parameters such as IP address, DHCP setting, TLS certificate settings, and so on.

Since the hotdesk user configuration is required to be independent of the network, a change is implemented in release 4.3.0 SP2, to have a list of network parameters stay on the phone (saved in local.cfg instead of user-local.cfg). This helps in avoiding mal-configurations when the VDP user switches networks.

In release 4.5.0, additional VDP filtering parameters are added to the list of configuration parameters.

The firmware implements this feature automatically in the background and no parameter or configuration is required. This feature is new to all 6800 series SIP phones.

The following are the list of blacklisted configuration parameters stored locally on the phone:

Release 4.5.0

- Advanced network settings
  - lldp
  - lldp interval
  - sip rport
- Troubleshooting parameter page
  - log server ip
  - log server port
  - log mac
  - log module
    - log module linemgr
    - log module user interface
    - log module misc
    - log module sip
    - log module dis
    - log module dstore

    - log module ept
    - log module ind
    - log module kbd
    - log module net
    - log module provis
    - log module rtpt
    - log module snd
    - log module prof
    - log module xml
    - log module stun
    - log module lldp
- Watchdog setting
    - watchdog enable
- Crash file retrieval
    - upload system info server
    - upload system info on crash
    - upload system info manual option
- Configuration server settings page
- Download protocol
- TFTP
    - tftp server
    - tftp path
    - alternate tftp server
    - alternate tftp path
    - use alternate tftp
- FTP
    - ftp server
    - ftp path
    - ftp username
    - ftp password
- HTTP
    - http server
    - http path
    - http port
- HTTPS
    - https server
    - https path
    - https port

- Auto resync
    - auto resync mode
    - auto resync time
    - auto resync max delay
    - auto resync days
- TLS Support
    - sips root and intermediate certificates
    - sips local certificate
    - sips trusted certificates
    - sips private key
- Action uri
    - action uri poll
    - action uri poll interval
    - action uri startup
    - action uri registered
    - action uri registration event
    - action uri xml sip notify
    - action uri tr69 check sync
    - action uri blf
    - action uri incoming
    - action uri outgoing
    - action uri offhook
    - action uri onhook
    - action uri connected
    - action uri disconnected
- Paging
    - paging group listening
- HTTPS Client Method

Release 4.3.0 SP2

- dhcp
- ip
- subnet mask
- default gateway
- dns1
- dns2
- ethernet port 0
- ethernet port 1

- eap type
- identity
- 802.1x root and intermediate certificates
- 802.1x local certificate
- 802.1x private key
- 802.1x trusted certificates
- tagging enabled
- vlan id
- vlan id port 1
- tos sip
- tos rtp
- tos rtcp
- tos priority map
- priority non-ip
- dhcp option 132 vlan id enabled
- dhcp config option override
- https user certificates

CONFIGURATION FILE PRECEDENCE UPDATE

If the same parameter appears more than once in the configuration files, the last parameter/value read will be used, that is the following precedence rules apply:

- Settings in the <model>.cfg file will overwrite startup.cfg settings
- Settings in the <mac>.cfg file will overwrite <model>.cfg settings
- Settings in local.cfg file will overwrite <mac>.cfg settings
- Settings in user.cfg file will overwrite local settings
- Settings in user-local.cfg file will overwrite user.cfg settings

Note: When a parameter value is changed locally (through WebUI or TUI or XML) to default value (or the same value as server.cfg), it does not display in local.cfg.

ISSUES RESOLVED IN RELEASE 4.5.0

This section describes the enhancements to previously developed features implemented on the 6800 series SIP phones and issues resolved in Release 4.5.0.

The following table provides the enhancement or issue number and a brief description of each enhancement or issue:

ISSUE	DESCRIPTION
DTP-26888/ DEF39824	For the 6867i SIP phone, it was observed that for each IP/PSTN call, a beep was heard when receiving a 183. Until the early media arrives, the phone plays the local ring tone on receiving a 183 and when the early media arrives, the phone stops playing the local ring tone and switches to play the early media. At this cut-off period a beep is heard. This issue is resolved by not playing the local tone when no media is received after 183.
DTP-26886/ DEF40396	For the 6867i SIP phone, it was observed that during an active call, when the user performs a blind transfer and the blind transfer fails, the Pickup softkey disappears when the “Transfer failed” message displays on the phone screen. The Pickup softkey appears back on the screen when the “Transfer failed” message is no more displayed. This issue is resolved.
DTP-27320	With Release 4.5.0, 6800 phones now support the validation of TeliaSonera Root Certificate from Telia.
DTP-28264/ DEF44576	When a SIP extension enabled with crypto 256 calls an InAttend operator, the call is transferred to a digital extension from InAttend. On answering the call, no speech was observed between the SIP extension and the digital extension. This issue is resolved.
DTP-28263/ DEF44577	When a public call is made from a 6800 series SIP phone with crypto 256 enabled, it was observed that a scraping noise was heard on clearing the call. This issue is resolved.
DTP-27584	For Chariot, it is observed that the SIP Phone does not send the requested REGISTER on Line 1 because the phone fails to enable the registration with the primary server when switching from a backup server to the primary server. This issue is resolved.
DEF44533/ CLN44580	For 6873i SIP phone, it is observed that the phone fails to accept the pushed content using a secure socket. This issue is resolved.
DEF44326/ CLN44572	During a VDP login, when the user changes the default language settings defined in the configuration file to local settings and selects the “Save Settings” option, the user is prompted to restart the phone to reflect the new settings. However after a reboot, it is observed that the local language settings are not saved. Instead the settings are set back to values defined in the configuration file. This issue is resolved.
DEF43881/ CLN44565	For a 6800 series SIP phone with 2 pages of top softkeys and one of the bottom softkey configured as the phone’s caller list, an observation was made that on pressing the callers list softkey when the phone was in idle state, the phone displayed the callers list but on pressing "quit" to go back to the idle screen, the display goes back to the second page of top softkeys instead of the first. This issue is resolved.
DEF44263/ CLN44557	For 6800 series SIP phones, it was observed that after a phone crash, the PKM does not recover and displays an empty screen and web page. This issue is resolved.

ISSUE	DESCRIPTION
DEF44509/ CLN44547	For M685i expansion module, it was observed that when there was no delay between the NOTIFY of defining key and NOTIFY of the changing color, the pseudo LED icon color on the phone was randomly updated. However this behavior of the expansion module was not observed when there was even 50 ms delay between the two NOTIFY messages. This issue is resolved.
DEF43553/ CLN44558	An issue was observed with the 6800 series SIP phones wherein the G.722 Overload Point Handling was sometimes broken in transmit direction for analog and USB headsets. This issue is resolved.
DEF43426/ CLN44554	For 6873i SIP phones, it was observed that the phone restarts repeatedly, approximately 3-4 times an hour.However the same behavior was not observed on reducing the number of MNS-keys from 20 to 11.
DEF44526	A reboot on 6867i SIP phone was reported by Chariot which was caused due to usage of a null pointer while validating a SIP through header. This issue is resolved.
DEF44539	For 6800 series SIP phones, it was reported when a factory default reset was initiated by TR-069 server, the phone's internal directory was not deleted. This issue is resolved.
DEF44264/ CLN44516	An issue was observed with the 6800 series SIP phones wherein the BLF list was working incorrectly due to the presence of a double dash "--" in the Calling Line ID Last Name (CLID) on Broadsoft. This issue is resolved.
DEF44469/ CLN44487	For 6800 series SIP phones, it was observed that if the phone receives a183 with SDP and immediately after that (with less than 0.001 ms) receives "486 BUSY HERE", the phone does not play the local fast busy tone. If 486 is received about 0.015 ms after 183+SDP then the phone plays the local fast busy tone. If 183 does not have SDP or 180 was sent instead then the local fast busy tone is successfully played all the times regardless of the delay between messages.
DEF44375/ CLN44480	For 6800 series SIP phones, it was reported that the phone reboots with a "SipEngine Crashed" error placing calls on Hold, Transfer and Park functions.This issue is resolved.
DEF44458/ CLN44459	After a second failover scenario in 6800 series SIP phones, it was observed that no GRUU header was included in the SUBSCRIBE messages and REGISTER message was not sent to the backup IP. This issue is resolved.
DEF43635	For 6800 series SIP phones, it was reported that during a 802.1x authentication if a PC was plugged into the PC port, the phone responds to 802.1x requests intended to the PC and displays a 802.1x error message on the phone. This issue is resolved.
DEF44227/ CLN44446	In release 4.5.0, a security issue was reported wherein Cross Site Scripting vulnerabilities were observed on the 6865i phones web interface. This issue is resolved.
ENH43496	In release 4.5.0, an enhancement is made wherein if the DHCP server does not answer for a certain length of time, the client retransmits the DHCP Discover request by backing off exponentially.
DTP-28592	The minimum supported value for "user config upload" parameter is changed to 5 due to some performance issues seen with lower values.
DTP-28594	Intermittently parameter 'web interface blacklist duration' would not block IPs immediately. A fix was made to block the requests at starting of the processing instead of at the end.

CONTACTING MITEL SUPPORT

If you have read this release note, and consulted the Troubleshooting section of your phone model's manual and still have problems, please contact Mitel Support through one of these methods:

North America

- Toll Free at 1-800-574-1611
- Online at http://www.mitel.com/content/mitel-technical-support

South America

Please contact your regional Mitel Technical Support.

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