Software Release Note

1.1 Scope of the Delivery

This is a release of STD-SIP v3.1 firmware for DT730G terminals. This release contains defect fix.

1.2 Software Release Details

Date	January 30, 2019	Release Version	Type of Release (Initial / Incremental / Defect fix)
		3.1.49.30	Defect fix

1.3 Details

S.N.	Deliverable (File Name)	Description	Version	Change Description (Defect ID / MR ID etc.)	Remarks
1	Lynx Firmware	Lynx DG Firmware for ITL-12/24/32 DG Lynx CG Firmware for ITL-12/24/32 CG	3.1.49.30	Release firmware contains following • Configuration files of STD-SIPv3.1.49.30 is same as STD-SIPv4.3.13.17 • Defect Fixes.	

1.4 Notice

None

1.5 List of Fixed Issues

1.5.1 List of Fixed Issue in STD-SIP v3.1.49.30

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T), Jira Issue ID(STDSIP)	Defect/Enhancement	Defect Description	Remarks
1.	FR-903180006(SSPV-627)	Defect	Message waiting LED cannot be lit	
2.	FR-903180028(SSPV-416)	Defect	CRP transfer to DT700	
3.	FR-903180050(SSPV-602)	Defect	DT730DG Standard SIP Firmware TFTP time-out	
4.	FR-001180122(SSPV-496)	Defect	Incorrect DTMF digits are send on Speaker	
5.	FR-903160069(SSPV-596)	Enhancement	DT7xx: Display Diversion destination i.c.o. activated diversion	
6.	P2490	Defect	Localization for Saudi Arabia Dterm	This is merge from v2.3 ,and is completed as per latest firmware 2.3.82.21

1.5.2 List of Known Issues in STD-SIP v3.1.49.30

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T), Jira Issue ID(STDSIP)	Defect/Enhancement	Defect Description	Remarks
1.	B -758	DSS detection failed when disconnect/connect side -2 connecter	To reproduce steps following 1) Configure DSS-60 and reboot terminal 2) Make sure DSS-60 keys are functioning 3) Disconnect and reconnect, DSS-60 side 2 connecter Result: DSS detection failed after second or third attempt	
2.	T:27491	If “username” and password” is fetched from DHCP (option 66) then similar issue as bug 770 gets reproduced on v2.3.44.24.	To reproduce configure DHCP option 66 with ‘:’ and ‘@’ in user name and password. Example: “ftp://Sp!()Ch&%:India@1234:@10.112.94.82” Result: Terminals expecting FTP connection with “host=1234, user=Sp! () Ch&%, password=India”, and it’s incorrect.	
3	FR-903160051	DT7xx/DT8xx Should not accept digits after cancel of second call	Call scenario: DT700 (1100) dials 1808. 1808 Answers. DT700 presses Transfer and dials 1107. 1107 starts ringing. DT700 presses End Call and offers RESUME or allows pressing new Transfer. However DT700 also accepts new digit information e.g. 1106 for the second call. Successive Transfer isn't possible now. In this situation a Polycom doesn't accept digit information. Only RESUME or New Transfer is allowed. To keep a consistent user interaction I propose that the DT7xx/DT8xx also doesn't allow new digit information in this state. It is also possible to reproduce this behavior on a 3C configuration	

4	FR-927170001	Boot Server setting is not saved properly after FW conversion.	Boot Server setting is not saved properly after FW conversion.	
5	FR-001170131	"IF user has changed ringtone and gets direct call phone plays that ringtone, however when it gets call from AA then it plays default ring tone. Customer is not using distinctive ringtone."	IF user has changed ringtone and gets direct call phone plays that ringtone, however when it gets call from AA then it plays default ring tone. Customer is not using distinctive ringtone.	
6	FR-903180032	Consultation Call Fails.	This issue is common in STD-SIPv4.3	

1.5.3 List of Fixed Issue in STD-SIP v3.1.45.14

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T), Jira Issue ID(STDSIP)	Defect/Enhancement	Defect Description	Remarks
1	FR-903170058(SSPV-402)	Defect	SRTP negotiation wrong DT820 returns 2 times AVP instead of SAVP and AVP	
2	FR-001170150(SSPV-403)	Defect	DT700 and DT820 stopped registering	This is merge from v2.3.

3	T-30684(SITD-504/STSI-1127)	Defect	DHCP Option 120 is not supported DT820, DT730CG/DG Terminal	This is merge from v2.3.
4	FR-001180012(SSPV-342)	Defect	DT820 stuck in DNS query forever if boot server is DN	
5	FR-927180001(SSPV-401)	Defect	No Speech in Case of Forwarding to external	
6	AR-001180023(SSPV-345)	Defect	Sometimes RTP Event keeps repeating even though digit only momentarily pressed a single time.	
7	AR-001180018(SSPV-344)	Defect	" place a all" should be " place a call" c is missing from word call	
8	FR-001180081(SSPV-400)	Defect	Phone fails to failback on certain SIP response	
9	FR-001180025(SSPV-341)	Defect	1) one way speech path with operator 2) Hold reminder enable and switching from handset to headset 3) Hold reminder enable and switching from handset to speaker 4) Auto Answer on Headset 5) Call Drop when doing multiple times shuttle and Hold (SSPV-404). 6) Terminal hang on continuous ringing and ringtone set is other than default ringtone	
10	STSI-1279	Defect	Unable to update Secondary FQDN address	
11	T30927	Defect	DT700 sent EPK web service page http requests to port 80 regardless of the specified port in the service URL	This is merge from v2.3.

12	FR-001170156-T30921	Defect	Univerge Blue - Error message when user tries to access his call history	This is merge from v2.3.
13	FR-001160045-T27945	Defect	It takes 2 reboots before new log.level.xx setting takes affect	This is merge from v2.3.
14	T-31620/ BRQ-2806	Defect	Delete On-hook dialing buffer after timeout	This is merge from v2.3.
15	FR-903170037/T-30476	Defect	dt-dnr-local.cfg is corrupted in case of desk sharing	This is merge from v2.3.
16	T-31952	Defect	All the lights switch off then on after a failed call on DT	This is merge from v2.3.
17	T-31207	Defect	Help text for Listconf and supervise EPK has to be updated	This is merge from v2.3.
18	T-31179	Defect	EBLF key with function ParkExt doesn't work if Monitor is not set	This is merge from v2.3.
19	FR-001180108 (SSPV-440)	Defect	DT820 on above firmware drops call when speaker and headset is switched fast. Number of switches is not fixed but call gets dropped	
20	FR-001180106 (SSPV-441)	Defect	When DT820 receives 480 Response to REGISTER request instead of sending re-REGISTER to same UCM it starts sending to members of SRV record.	
21	FR-001180114 (SSPV-460)	Defect	No ringtone on the first call when it is set to the default. The test case steps and videos were provided to the Dev team	
22	AR-001180025	Defect	When DT820 using SPKR does Blind Transfer, after hang up pressing SPKR the DT820 SPKR key lights by itself, user hears ROT, then key go back out and phone becomes idle	
23	AR-001180026	Defect	When DT820 using SPKR does Call Park, after hang up pressing SPKR the DT820 SPKR key lights by itself, user hears ROT, then key go back out and phone becomes idle	

24	FR-001170151	Defect	"On Univerge Blue deployment under certain unknown conditions a test DT820 phone stopped Registering"	This is merge from v2.3.
25	FR-903180028	Defect	CRP transfer to DT700	
26	ITE_65612/T_31952 (DT700 v2.3 code changes)	Defect	It seems that after a failed call , when the phone gets a 408 Request Timeout, it switches off all the lights on BLF keys and turns them back on	This is merge from v2.3.
27	SI3C77A / FR-001180099	Defect	DUT is getting echo while receiving a call by using speaker	
28	SSD-438	Defect	Error while firmware download when FTP server is not reachable.	
29	SSPV-415	Defect	Terminal crashes during Soft ringtone and Internal ringtone toggle case	
30	SSPV-417	Defect	Incorrect character is included at accessing to incorrect XML server	
31	SSD-287	Defect	Crash when terminal accepts new call during conference call.	
32	T:31708	Defect	One way speech which involved BCT operator using headset	This is merge from v2.3.
33	T:31523	Defect	When Jabra Headset mode is enabled and hold reminder enabled sometimes no audio	This is merge from v2.3.

1.5.4 List of Known Issues in STD-SIP v3.1.45.14

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T),	Defect/Enhancement	Defect Description	Remarks

	Jira Issue ID(STDSIP)			
1.	B -758	DSS detection failed when disconnect/connect side -2 connecter	To reproduce steps following 1) Configure DSS-60 and reboot terminal 2) Make sure DSS-60 keys are functioning 3) Disconnect and reconnect, DSS-60 side 2 connecter Result: DSS detection failed after second or third attempt	
2.	T:27491	If “username” and password” is fetched from DHCP (option 66) then similar issue as bug 770 gets reproduced on v2.3.44.24.	To reproduce configure DHCP option 66 with ‘:’ and ‘@’ in user name and password. Example: “ftp://Sp!()Ch&%:India@1234:@10.112.94.82” Result: Terminals expecting FTP connection with “host=1234, user=Sp! () Ch&%, password=India”, and it’s incorrect.	
3	FR-903160051	DT7xx/DT8xx Should not accept digits after cancel of second call	Call scenario: DT700 (1100) dials 1808. 1808 Answers. DT700 presses Transfer and dials 1107. 1107 starts ringing. DT700 presses End Call and offers RESUME or allows pressing new Transfer. However DT700 also accepts new digit information e.g. 1106 for the second call. Successive Transfer isn't possible now. In this situation a Polycom doesn't accept digit information. Only RESUME or New Transfer is allowed. To keep a consistent user interaction I propose that the DT7xx/DT8xx also doesn't allow new digit information in this state. It is also possible to reproduce this behavior on a 3C configuration	
4	FR-903160069	DT7xx: Display Diversion destination i.c.o. activated diversion	Maintenance Enhancement Request	This is a feature request.
5	FR-927170001	Boot Server setting is not saved properly after FW conversion.	Boot Server setting is not saved properly after FW conversion.	
6	FR-001170131	"IF user has changed ringtone and gets direct call phone plays that ringtone, however when it gets call from AA then it plays default ring tone. Customer is not using distinctive ringtone."	IF user has changed ringtone and gets direct call phone plays that ringtone, however when it gets call from AA then it plays default ring tone. Customer is not using distinctive ringtone.	

1.5.5 List of Fixed Issue in STD-SIP v3.1.35.3

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T), Jira Issue ID(STDSIP)	Defect/Enhancement	Defect Description	Remarks
1	FR-903170047 (STDSIP-117)	Defect	DT820 phone with configured 802.1x restarts after unplugging PC connected to PC port.	
2	FR-001170129 (STDSIP-118)	Defect	Speakerphone muting a call from AT&T meeting solution is not effective until they press "1" to join a conference	
3	FR-903160068 (STDSIP-119)	Defect	Customer is asking how to disable call history feature.	
4	STDSIP-120	Defect	Time Stamp information in SIP PUBLISH message is not UTC.	JIRA Id for v4.1 is STSI-1160
5	STDSIP-121	Defect	Event header is not included in SIP PUBLISH message from DT terminal. SIP PUBLISH message must contain “Event: vq-rtcpxr” header in order to identify it as RTCP-XR packet.	JIRA Id for v4.1 is STSI-1161
6	STDSIP-122	Defect	When RTP Payload types (PT) is “dynamic” (96~), SIP PUBLISH doesn’t include the payload description (PD) in SessionDesc.	JIRA Id for v4.1 is STSI-1164
7	T-30450,T-30451, T-30452 (STDSIP-123)	Defect	Modifying Telephony Area PSTN Numbers for Austria, Belgium and Switzerland	This is merge from v2.3.

8	T-30470 (STDSIP-124)	Defect	Change the language tags for some country entries	This is merge from v2.3.
9	ECR-2742 (STDSIP-125)	Defect	New Call History Logoff Persistence setting to allow persistent in flash only	This is merge from v2.3.

1.5.6 List of Known Issues in STD-SIP v3.1.35.3

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T), Jira Issue ID(STDSIP)	Defect/Enhancement	Defect Description	Remarks
1.	B -758	DSS detection failed when disconnect/connect side -2 connecter	To reproduce steps following 1) Configure DSS-60 and reboot terminal 2) Make sure DSS-60 keys are functioning 3) Disconnect and reconnect, DSS-60 side 2 connecter Result: DSS detection failed after second or third attempt	
2.	T:27491	If "username" and password" is fetched from DHCP (option 66) then similar issue as bug 770 gets reproduced on v2.3.44.24.	To reproduce configure DHCP option 66 with ':' and '@' in user name and password. Example: "ftp://Sp!()Ch&%:India@1234:@10.112.94.82" Result: Terminals expecting FTP connection with "host=1234, user=Sp! () Ch&%, password=India", and it's incorrect.	
3	FR-903160051	DT7xx/DT8xx Should not accept digits after cancel of second call	Call scenario: DT700 (1100) dials 1808. 1808 Answers. DT700 presses Transfer and dials 1107. 1107 starts ringing. DT700 presses End Call and offers RESUME or allows pressing new Transfer. However DT700 also accepts new digit information e.g. 1106 for the second call. Successive Transfer isn't possible now. In this situation a Polycom doesn't accept digit	

			information. Only RESUME or New Transfer is allowed. To keep a consistent user interaction I propose that the DT7xx/DT8xx also doesn't allow new digit information in this state. It is also possible to reproduce this behavior on a 3C configuration	
4	FR-903160069	DT7xx: Display Diversion destination i.c.o. activated diversion	Maintenance Enhancement Request	This is a feature request.
5	STDSIP-115	DHCP Option 120 is not supported DT820, DT730CG/DG Terminal	DT terminal does not use SIP server option provided through DHCP option 120.	
6	FR-903170058	SRTP negotiation wrong DT820 returns 2 times AVP instead of SAVP	SRTP negotiation wrong DT820 returns 2 times AVP instead of SAVP and AVP. Further there is one way speech. But this is most likely caused by the incorrect SDP generated by the DT820.	
7	FR-927170001	Boot Server setting is not saved properly after FW conversion.	Boot Server setting is not saved properly after FW conversion.	
8	FR-001170151	On Univerge Blue deployment under certain unknown conditions a test DT820 phone stopped registering.	There is similar defect (FR-001170150) reported against DT700 and it might be affecting phones based on Lynx firmware as well.	
9	FR-001170131	"IF user has changed ringtone and gets direct call phone plays that ringtone, however when it gets call from AA then it plays default ring tone. Customer is not using distinctive ringtone."	IF user has changed ringtone and gets direct call phone plays that ringtone, however when it gets call from AA then it plays default ring tone. Customer is not using distinctive ringtone.	

1.5.7 List of Fixed Issue in STD-SIP v3.1.32.25

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T), Jira Issue ID(STDSIP)	Defect/Enhancement	Defect Description	Remarks
1	AR-927170003 (STDSIP-113)	Defect	DT phone will stop sending RTP when negotiating srtp.	
2	T-30238 (STDSIP-112)	Defect	The dial tone played for German setting is incorrect	This is merge from v2.3.
3	FR-927170003 (STDSIP-114)	Defect	if DSS loses power , the unit will stop updating the buttons LEDs	

1.5.8 List of Known Issues in STD-SIP v3.1.32.25

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T), Jira Issue ID(STDSIP)	Defect/Enhancement	Defect Description	Remarks
1.	B -758	DSS detection failed when disconnect/connect side -2 connecter	To reproduce steps following 1) Configure DSS-60 and reboot terminal 2) Make sure DSS-60 keys are functioning 3) Disconnect and reconnect, DSS-60 side 2 connecter Result: DSS detection failed after second or third attempt	

2.	T:27491	If “username” and password” is fetched from DHCP (option 66) then similar issue as bug 770 gets reproduced on v2.3.44.24.	To reproduce configure DHCP option 66 with ‘:’ and ‘@’ in user name and password. Example: “ftp://Sp!()Ch&%:India@1234:@10.112.94.82” Result: Terminals expecting FTP connection with “host=1234, user=Sp! () Ch&%, password=India”, and it’s incorrect.	
3	FR-903160051	DT7xx/DT8xx Should not accept digits after cancel of second call	Call scenario: DT700 (1100) dials 1808. 1808 Answers. DT700 presses Transfer and dials 1107. 1107 starts ringing. DT700 presses End Call and offers RESUME or allows pressing new Transfer. However DT700 also accepts new digit information e.g. 1106 for the second call. Successive Transfer isn't possible now. In this situation a Polycom doesn't accept digit information. Only RESUME or New Transfer is allowed. To keep a consistent user interaction I propose that the DT7xx/DT8xx also doesn't allow new digit information in this state. It is also possible to reproduce this behavior on a 3C configuration	
4	FR-903160068	Customer is asking how to disable call history feature	Maintenance Enhancement Request	This is a feature request.
5	FR-903160069	DT7xx: Display Diversion destination i.c.o. activated diversion	Maintenance Enhancement Request	This is a feature request.
6	STDSIP-115	DHCP Option 120 is not supported DT820, DT730CG/DG Terminal	DT terminal does not use SIP server option provided through DHCP option 120.	

1.5.9 List of Fixed Issue in STD-SIP v3.1.30.30

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T), Jira Issue ID(STDSIP)	Defect/Enhancement	Defect Description	Remarks
1	FR-001160172 (STDSIP-97)	Defect	The DT820 will ring itself and show its extension as callerID after the user terminates a parked call scenario	
2	FR-903170014 (STDSIP-98)	Defect	VLAN tagging when set manually on the DT820 does not seem to work	This issue is fixed with the fix applied for FR-903170023.
3	FR-903170023 (STDSIP-99)	Defect	Problems during the installation of DT820 (STD SIP) on CISCO The issue seems to be related to VLAN tagging. The trace file made on the same port with a DT820 does not show any traffic for the DT820. For the DT730 there is proper trace information. So the question is why is that? Will a wrong VLAN tag result in an empty trace file when a monitor port is applied?	
4	FR-001170036 (STDSIP-100)	Defect	DT730G and DT820 firmware does not append MAC address to syslog making it impossible to segregate logs	
5	FR-903170018 (STDSIP-101)	Defect	DT730G stops doing TFTP after 2 minutes when TF TP responses are slow	
6	T-29705 (STDSIP-102)	Defect	Add “Silent Ring” option in DT700	Merge from v2.3

7	STDSIP-103 (P:2711)	Defect	Addition of the three new country and language templates (Austria, Belgium, and Switzerland)	Merge from v2.3
8	T-29348 (STDSIP-104)	Defect	Request to update the default list of languages on the DTERM (English-Australia is removed from the default list and replaced by French-France.)	Merge from v2.3
9	STDSIP-105	Defect	DNS information is added to Syslog. (STD-SIP-v4.1:Univerge Blue: Two phones in Sanko USA are getting into RS_NEVER_REGISTER state)	The following DNS server error/information logs is added to the existing "log.level.cc" for SIP. 1. List all DNS servers learnt by phone from DHCP and/or static. 2. Query timeouts reported per learnt DNS IP address. 3. Actual DNS response as seen in Wireshark.(at log level 7)
10	FR-001170027 T-29596 (STDSIP-106)	Defect	On Univerge Blue cloud hosting we are seeing that random phones start swinging registration between two entries in SRV records	Merge from v2.3. (Only one part of this issue is fixed) This fix is that, terminal will re-subscribe once the terminal will identify that source port is changed in the transport.
11	STDSIP-107	Defect	Inconsistency in language (Checked for Austrian vs Italian and Austrian vs German)	
12	FR-927170002 (STDSIP-108)	Defect	Phone no sending or playing RTP	
13	FR-001170099 (STDSIP-109)	Defect	In UCAAS deployment customer phone ends up sending INVITE to private IP address of MGC instead of sending it to IP addressed where it sends general INVITE as a result of Refer-To header having private IP address.	
14	FR-607170001 (STDSIP-110)	Defect	The Led of the Speaker + Mic + line is ON + dial tone is is give, The caller starts dialing and tone disappear as if the call is processing but in reality the terminal is still STD.BY and it stays like this (going	

			NOWHERE) as if you are really making a call, when pushing speakerphone key. The caller at the end has the perception that call is processing but the terminal is still on STD and at the end looks like the call fail. Steps are: 1. do not press speaker. 2. Dial any number (internal or external) 3. The speaker and mic keys are lit but the call never progresses.	
15	FR-607170002 (STDSIP-111)	Defect	Generate Dial tone when the Trunk Dial key is used on a DT-terminal	
16	FR-903160060 (STDSIP-110)	Defect	NEC's DT-terminals, using the Standard SIP protocol, behave incorrectly when an end-user starts dialing digits in an idle on-hook situation.	This issue is fixed with the fix applied for FR-607170001.

1.5.10 List of Known Issues in STD-SIP v3.1.30.30

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T), Jira Issue ID(STDSIP)	Defect/Enhancement	Defect Description	Remarks
1.	B -758	DSS detection failed when disconnect/connect side -2 connecter	To reproduce steps following 1) Configure DSS-60 and reboot terminal 2) Make sure DSS-60 keys are functioning 3) Disconnect and reconnect, DSS-60 side 2 connecter Result: DSS detection failed after second or third attempt	
2.	T:27491	If "username" and password" is fetched from DHCP (option 66) then similar issue as bug 770 gets reproduced on v2.3.44.24.	To reproduce configure DHCP option 66 with ':' and '@' in user name and password. Example: "ftp://Sp!()Ch&%:India@1234:@10.112.94.82"	

			Result: Terminals expecting FTP connection with “host=1234, user=Sp!()Ch&%, password=India”, and its incorrect.	
3	FR-903160051	DT7xx/DT8xx Should not accept digits after cancel of second call	Call scenario: DT700 (1100) dials 1808. 1808 Answers. DT700 presses Transfer and dials 1107. 1107 starts ringing. DT700 presses End Call and offers RESUME or allows pressing new Transfer. However DT700 also accepts new digit information e.g. 1106 for the second call. Successive Transfer isn't possible now. In this situation a Polycom doesn't accept digit information. Only RESUME or New Transfer is allowed. To keep a consistent user interaction I propose that the DT7xx/DT8xx also doesn't allow new digit information in this state. It is also possible to reproduce this behavior on a 3C configuration	
4	FR-903160068	Customer is asking how to disable call history feature	Maintenance Enhancement Request	This is a feature request.
5	FR-903160069	DT7xx: Display Diversion destination i.c.o. activated diversion	Maintenance Enhancement Request	This is a feature request.
6	FR-927170003	if DSS loses power , the unit will stop updating the buttons LEDs	The problem can be reproduced on following way: -Configure 30 DSS keys (trunk dial) -In normal working conditions the light is ON -Power off DSS console and power ON -All light indications are gone - To re-establish the correct working you must restart completely the phone (from 3C Administration). This issue appears from the day one, but only it is reported from 3 highest ranking officer. They tried to use PoE for DSS separated power supply- but doesn't help after few weeks LAMP suddenly went OFF. NEC Italy doubts on the phone SW and some refreshment.	

1.5.11 List of Fixed Issue in STD-SIP v3.1.27.21

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T), Jira Issue ID(STDSIP)	Defect/Enhancement	Defect Description	Remarks
1	FR-903170016 (STDSIP-95)	Defect	DT730 phones work with DT-macaddress.cfg instead of DT-ext.cfg after update of phone.	External Defect

1.5.12 List of Known Issues in STD-SIP v3.1.27.21

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T), Jira Issue ID(STDSIP)	Defect/Enhancement	Defect Description	Remarks
1.	B -758	DSS detection failed when disconnect/connect side -2 connecter	To reproduce steps following 1) Configure DSS-60 and reboot terminal 2) Make sure DSS-60 keys are functioning 3) Disconnect and reconnect, DSS-60 side 2 connecter Result: DSS detection failed after second or third attempt	
2.	T:27491	If “username” and password” is fetched from DHCP (option 66) then similar issue as bug 770 gets reproduced on v2.3.44.24.	To reproduce configure DHCP option 66 with ‘:’ and ‘@’ in user name and password. Example: “ftp://Sp!()Ch&%:India@1234:@10.112.94.82”	

			Result: Terminals expecting FTP connection with “host=1234, user=Sp!()Ch&%, password=India”, and its incorrect.	
3	FR-903160051	DT7xx/DT8xx Should not accept digits after cancel of second call	Call scenario: DT700 (1100) dials 1808. 1808 Answers. DT700 presses Transfer and dials 1107. 1107 starts ringing. DT700 presses End Call and offers RESUME or allows pressing new Transfer. However DT700 also accepts new digit information e.g. 1106 for the second call. Successive Transfer isn't possible now. In this situation a Polycom doesn't accept digit information. Only RESUME or New Transfer is allowed. To keep a consistent user interaction I propose that the DT7xx/DT8xx also doesn't allow new digit information in this state. It is also possible to reproduce this behavior on a 3C configuration	
4	FR-903160060	NEC's DT-terminals, using the Standard SIP protocol, behave incorrectly when an end-user starts dialling digits in an idle on-hook situation.	NEC's DT-terminals, using the Standard SIP protocol, behave incorrectly when an end-user starts dialling digits in an idle on-hook situation. Furthermore : the on-hook dialling in idle mode is showing an inconsistent behaviour comparing this with dialling in loud-speaking mode. Current behaviour: On-hook/idle dialling: 1. Terminal is on-hook/idle 2. End-user presses a digit 3. Line-Key, Mic-Key and Speaker-Key are all lit and the digit is on the display. 4. This is causing the end-user to expect the terminal to be in the same situation as if the end-user would have started the call by means of pressing the Speaker Key and subsequently pressing a digit. 5. So in both situations (i.e. on-hook/idle dialling and loud- speaking dialling) exactly the same keys are lit and the same information is on the display 6. There is however an important difference; i.e. the behaviour of the terminal: a. In loud-speaking dialling mode the terminal takes the digit map into account. So a Number-Complete is being recognized and the terminal automatically starts dialling if the number is complete. b. In the on-hook/idle dialling situation the digit map is NOT taken into account. That's the confusing part. Required behaviour: 1) Maintain the current behaviour of the DT-std-SIP phones (i.e. the default behaviour of the Standard SIP DT-terminals 2) Introduce an option to change the behaviour when you press a digit. The new behaviour will be Speaker LED off, MIC led off, Line key LED off. With the new behaviour pressing the Loudspeaker key or Lifting the handset or Pressing the Dial key, when digits are	

			present, will initiate dialling (=Going Off hook). This situation can be cancelled only by pressing the "EndCall" soft key, returning the phone to the idle state. 3) Introduce an option to allow the phone to check the digits pressed while on hook against the digit map and initiate dialing when a match is found.	
5	FR-903160068	Customer is asking how to disable call history feature	Maintenance Enhancement Request	This is a feature request.
6	FR-903160069	DT7xx: Display Diversion destination i.c.o. activated diversion	Maintenance Enhancement Request	This is a feature request.
7	FR-001160172	The DT820 will ring itself and show its extension as caller ID after the user terminates a parked call scenario	Using the DT820 phone The customer can place a call from an extension (1890) to an outside number (763–512–1111). Then from the calling extension (1890) they will place the call on park. From that same extension (1890) they will take the call off of Park and continue the call with the called extension (763–512– 1111). Then from the calling extension (1890) the customer will hang up the call with the called extension (763–512– 1111). In just a few seconds after the parked call has ended the calling extension (1890) will ring and show its extension (1890) as caller ID. The customer will answer at 1890 and there will be dead air. Next they will hang 1890 up of course. The description above is just one call that the customer shared for this issue. They can reproduce this issue from all the DT820 extensions making outbound calls at the new location listed above.	
8	FR-001170027	On Univerge Blue cloud hosting we are seeing that random phones start swinging registration between two entries in SRV records.	The exact cause of issue is unknown but we feel it stars due to temporary network outage and then never recovers until phone is rebooted. We have managed to capture syslog from one of such phone.	This FR is handled in v2.3 by NECECT. After fixing in v2.3, the fix and release of v3.1/v4.0/v4.1 will be executed.
9	FR-903170018	DT730G stops doing TFTP after 2 minutes when TFTP responses are slow	When 2 different TFTP servers are used (we don't have the TFTP server). But based on the traces you can see that in the failing case the DT stops after exactly 2 minutes. The question	

			is why is there a time limit and shouldn't we extend that limit as a successful TFTP session takes a minute 23 seconds.	

1.5.13 List of Fixed Issues in STD-SIP v3.1.26.23

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T), Jira Issue ID(STDSIP)	Defect/Enhancement	Defect Description	Remarks
1	FR-927160002 (STDSIP-77)	Defect	In the current implementation, only 503 and 580 were pulled out to play a tone (busy/reorder) in 5xx codes by phone	External Defect This bug was fixed in STD-SIPv3.1.24.20 also. However in this release this fix has been modified. 1) As per the applied fix, Reorder tone will not be played for response code 500. 2) Reorder tone will only be played if there is a single call on terminal for which terminal receives 4XX to 6XX response.
2	FR-001160199 (STDSIP-91)	Defect	After an extension places two incoming calls on hold the next calls to the ext will not have voice	External Defect **Merge from v2.3**
3	FR-903170001 (STDSIP-88)	Defect	Customer requests improvement on the Dutch translations	External Defect nl_NL.mo file is updated for fixing this issue

4	STDSIP-94	Defect	Terminal goes from RTCP-XR to DHCP Mode if exit key is pressed	Internal Defect
5	STDSIP-90	Defect	CRL validation not working for HTTPS	Internal Defect

1.5.14 List of Known Issues in STD-SIP v3.1.26.23

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T), Jira Issue ID(STDSIP)	Defect/Enhancement	Defect Description	Remarks
1.	B -758	DSS detection failed when disconnect/connect side -2 connecter	To reproduce steps following 1) Configure DSS-60 and reboot terminal 2) Make sure DSS-60 keys are functioning 3) Disconnect and reconnect, DSS-60 side 2 connecter Result: DSS detection failed after second or third attempt	
2.	T:27491	If “username” and password” is fetched from DHCP (option 66) then similar issue as bug 770 gets reproduced on v2.3.44.24.	To reproduce configure DHCP option 66 with ‘:’ and ‘@’ in user name and password. Example: “ftp://Sp!()Ch&%:India@1234:@10.112.94.82” Result: Terminals expecting FTP connection with “host=1234, user=Sp!()Ch&%, password=India”, and its incorrect.	
3	FR-903160051	DT7xx/DT8xx Should not accept digits after cancel of second call	Call scenario: DT700 (1100) dials 1808. 1808 Answers. DT700 presses Transfer and dials 1107. 1107 starts ringing. DT700 presses End Call and offers RESUME or allows pressing new Transfer. However DT700 also accepts new digit information e.g. 1106 for the second call. Successive Transfer isn't possible now. In this situation a Polycom doesn't accept digit information. Only RESUME or New Transfer is allowed. To	

			keep a consistent user interaction I propose that the DT7xx/DT8xx also doesn't allow new digit information in this state. It is also possible to reproduce this behavior on a 3C configuration	
4	FR-903160060	NEC's DT-terminals, using the Standard SIP protocol, behave incorrectly when an end-user starts dialling digits in an idle on-hook situation.	NEC's DT-terminals, using the Standard SIP protocol, behave incorrectly when an end-user starts dialling digits in an idle on-hook situation. Furthermore : the on-hook dialling in idle mode is showing an inconsistent behaviour comparing this with dialling in loud-speaking mode. Current behaviour: On-hook/idle dialling: 1. Terminal is on-hook/idle 2. End-user presses a digit 3. Line-Key, Mic-Key and Speaker-Key are all lit and the digit is on the display. 4. This is causing the end-user to expect the terminal to be in the same situation as if the end-user would have started the call by means of pressing the Speaker Key and subsequently pressing a digit. 5. So in both situations (i.e. on-hook/idle dialling and loud- speaking dialling) exactly the same keys are lit and the same information is on the display 6. There is however an important difference; i.e. the behaviour of the terminal: a. In loud-speaking dialling mode the terminal takes the digit map into account. So a Number-Complete is being recognized and the terminal automatically starts dialling if the number is complete. b. In the on-hook/idle dialling situation the digit map is NOT taken into account. That's the confusing part. Required behaviour: 1) Maintain the current behaviour of the DT-std-SIP phones (i.e. the default behaviour of the Standard SIP DT-terminals 2) Introduce an option to change the behaviour when you press a digit. The new behaviour will be Speaker LED off, MIC led off, Line key LED off. With the new behaviour pressing the Loudspeaker key or Lifting the handset or Pressing the Dial key, when digits are present, will initiate dialling (=Going Off hook). This situation can be cancelled only by pressing the "EndCall" soft key, returning the phone to the idle state. 3) Introduce an option to allow the phone to check the digits pressed while on hook against the digit map and initiate dialing when a match is found.	
5	FR-903160068	Customer is asking how to disable call history feature	Maintenance Enhancement Request	This is a feature request.

6	FR-903160069	DT7xx: Display Diversion destination i.c.o. activated diversion	Maintenance Enhancement Request	This is a feature request.

1.5.15 List of Fixed Issues in STD-SIP v3.1.24.20

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T), Jira Issue ID(STDSIP)	Defect/Enhancement	Defect Description	Remarks
1	FR-001160132 (STDSIP-84)	Defect	Univerge Blue customer's phones are constantly ringing due to rouge INVITE attack from internet. DT700 responds to any INVITE received. This is causing Univerge Blue customer's phones ring constantly.	**External Defect** **(Merge from v2.3)**
2	FR-927160002 (STDSIP-77)	Defect	In the current implementation, only 503 and 580 were pulled out to play a tone (busy/reorder) in 5xx codes by phone. A customer in Italy has all DT730G phones. They are having a problem with certain calls that fail. The system topology consists of a 3C system and an IS3000 system connected via SIP trunk. The outside trunks are on the IS3000 side. The problem is as follows: 1- User on 3C making an outside call . 2- If the IS3000 returns a 504 Gateway Time-out, the MGC send this response to the phone and the phone terminates the call without playing any sort of tone 3- If the IS3000 returns a 502 Bad Gateway, the MGC send this response to the phone and the phone terminates the call without playing any sort of tone 4- If the IS3000 returns a 503 , the MGC sends this	**External Defect**

			response to the phone and the phone plays an error tone for a second before terminating the call giving the user a chance to understand that the call errored We would like to know if this behavior can be reproduced on the DT700 phones before we escalate this to Japan. Knowing if this is consistent between all the NEC phone models is important before we escalate this issue.	
3	FR-927160003 (STDSIP-81)	Defect	If you make a call between two DT 730G phones , then answer the call, then mute both sides of the call, you will still hear low level background noise coming to originating phone. Reproduction in NECPF I tried the reproduction with DT700, DT730G and DT820. [Operation] The following steps are different from Sam-san's informed. However you will be able to confirm low noise easily. Step 1: TEL A calls TEL B with On Hook status (not use handset). Step 2: TEL B answers with handset. (With using handset, you can confirm low noise via handset.) Step 3: TEL A goes on Mute (press Mic button). And tap TEL A's mic that is located in front of TEL A with your finger. TEL B hears small noise via handset. [Result] Terminal FW version Result DT710 and DT730 2.3.57.1 Not reproduced. When I executed the above operation, I couldn't hear the noise. Also I executed same operation that Sam-san has informed. However I couldn't hear low noise in v2.3.57.1. DT730G 3.1.22.21 *Note Reproduced. I could hear low noise. DT820 4.0.22.21 *Note Same as above *Note These FW are the temporary version for FR-001160154. However the basic phone behavior is no changing from the previous official release version.	**External Defect**
4	FR-927160004 (STDSIP-86)	Defect	If a key is configured as a TrunkDial and is pressed the display only shows the number and not the name.	**External Defect**

			Essentially the General is complaining that in the time between pressing a function key on his DSS console (DT700CG Rel 3.1.20.12) and being connected, he sees trunk & line info in his display BUT NOT the Name label that is associated with that pressed function key. He wishes to see Name field at all times from pressing key, during call set-up, alerting and after connect, for the entire duration of the call. See also attached document. Let's take this as being the minimal requirement. What I mean to say is that as long as the Name field is displayed in the same display location thrgoughout all phases of the call as described above, then I assume it is OK if there is also additional trunk and or number information as well.	
5	ntpd 4.2.8p9 Upgrade	Enhancement	To fix following vulnerability, ntpd 4.2.8p9 has been upgraded. CERT Vulnerability Note http://www.kb.cert.org/vuls/id/633847 Network Time Protocol Project http://support.ntp.org/bin/view/Main/SecurityNotice#November_2016_ntp_4_2_8p9_NTP_Se	**External Defect**
6	FR-903160014	Defect	After transfer with a DT730 (2.3) with SRTP there's no media played. From a SIP trunk (192.168.160.20) a call is made towards 2106 (192.168.164.2). Then from the trunk the call is transferred blind to 2105 (192.168.164.26). There's one way speech after the transfer. At 192.168.164.2 does not play the received media. There's no ICMP. There are some error messages after the transfer, that could very well be related. The scenario works fine if 2 DT730G phones are used. Also in case I make the first calls towards 2105 and then transfer blind over the trunk towards 2106 it works fine too.	**External Defect** This issue is not reproducible on STD-SIPv3.1. Hence no fix is applied for this issue in this firmware.

7	FR-903160039	Defect	Speech problems on DT-710 Sometimes there are speech problems during a connection. Its just as if the speech is not played. I took the speech from the Wireshark, then its OK. There's also no large delays so it also can't be a networks related issue. Attached a file that has been recorded. Note: Yesterday (= 8/31/2016) the our Polish Partner - on behalf of the end- customer - requested to upgrade the ITE56047 ticket to 'Very Urgent'; for that reason the priority of the FR is upgraded to 'E'. Gerben Wennink.	**External Defect** This issue is not reproducible on STD-SIPv3.1. Hence no fix is applied for this issue in this firmware

1.5.16 List of Known Issues in STD-SIP v3.1.24.20

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T), Jira Issue ID(STDSIP)	Defect/Enhancement	Defect Description	Remarks
1.	B -758	DSS detection failed when disconnect/connect side -2 connecter	To reproduce steps following 1) Configure DSS-60 and reboot terminal 2) Make sure DSS-60 keys are functioning 3) Disconnect and reconnect, DSS-60 side 2 connecter Result: DSS detection failed after second or third attempt	
2.	T:27491	If “username” and password” is fetched from DHCP (option 66) then similar issue as bug 770 gets reproduced on v2.3.44.24.	To reproduce configure DHCP option 66 with ‘:’ and ‘@’ in user name and password. Example: “ftp://Sp!()Ch&%:India@1234:@10.112.94.82” Result: Terminals expecting FTP connection with “host=1234, user=Sp!()Ch&%, password=India”, and its incorrect.	

3	FR-903160051	DT7xx/DT8xx Should not accept digits after cancel of second call	Call scenario: DT700 (1100) dials 1808. 1808 Answers. DT700 presses Transfer and dials 1107. 1107 starts ringing. DT700 presses End Call and offers RESUME or allows pressing new Transfer. However DT700 also accepts new digit information e.g. 1106 for the second call. Successive Transfer isn't possible now. In this situation a Polycom doesn't accept digit information. Only RESUME or New Transfer is allowed. To keep a consistent user interaction I propose that the DT7xx/DT8xx also doesn't allow new digit information in this state. It is also possible to reproduce this behavior on a 3C configuration	
4	FR-903160060	NEC's DT-terminals, using the Standard SIP protocol, behave incorrectly when an end-user starts dialling digits in an idle on-hook situation.	NEC's DT-terminals, using the Standard SIP protocol, behave incorrectly when an end-user starts dialling digits in an idle on-hook situation. Furthermore : the on-hook dialling in idle mode is showing an inconsistent behaviour comparing this with dialling in loud-speaking mode. Current behaviour: On-hook/idle dialling: 1. Terminal is on-hook/idle 2. End-user presses a digit 3. Line-Key, Mic-Key and Speaker-Key are all lit and the digit is on the display. 4. This is causing the end-user to expect the terminal to be in the same situation as if the end-user would have started the call by means of pressing the Speaker Key and subsequently pressing a digit. 5. So in both situations (i.e. on-hook/idle dialling and loud-speaking dialling) exactly the same keys are lit and the same information is on the display 6. There is however an important difference; i.e. the behaviour of the terminal: a. In loud-speaking dialling mode the terminal takes the digit map into account. So a Number-Complete is being recognized and the terminal automatically starts dialling if the number is complete. b. In the on-hook/idle dialling situation the digit map is NOT taken into account. That's the confusing part. Required behaviour: 1) Maintain the current behaviour of the DT-std-SIP phones (i.e. the default behaviour of the Standard SIP DT-terminals 2) Introduce an option to change the behaviour when you press a digit. The new behaviour will be Speaker LED off, MIC led off, Line key LED off. With the new behaviour pressing the Loudspeaker key or Lifting the handset or Pressing the Dial key, when digits are present, will initiate dialling (=Going Off hook). This situation can be cancelled only by pressing the "EndCall" soft key, returning the phone to the idle state. 3) Introduce an option to allow the phone to check the digits pressed while on hook against the digit map and initiate dialing when a match is found.	

1.5.17 List of Fixed Issues in STD-SIP v3.1.23.10

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T), Jira Issue ID(STDSIP)	Defect/Enhancement	Defect Description	Remarks
1	FR-001160154 (STDSIP-76)	Enhancement	RTCP-XR cannot be recorded in cloud environment. The RTCP-XR is normally sent to another address than the registration address, yet when proxy is configured, it seems to only be able to use the same proxy as the registrations.	**External Defect** A new parameter "**phone.rtcpxr.proxy.routing**" is introduced for this enhancement. This parameter is only used for the case when Outbound Proxy parameter is set. When Outbound Proxy parameter is not set, this parameter is not used. If "0- Disable" is set, all RTCP-XR SIP messages like PUBLISH are sent to the metrics collector that is specified by RTCP-XR VoIP Metrics Collector Address and Port of current parameter. If "1- Enable" are set, all RTCP-XR SIP messages like PUBLISH is sent to Outbound Proxy that is the specified by "line.1.outboundproxy.address" and "line.1.outboundproxy.port".

				Please note that earlier behavior of STD-SIP firmware will be altered with default value "0" of this parameter. If proxy support for RTCP-XR is required then please change the value of "phone.rtcpxr.proxy.routing" to 1.
2	FR-903160061 (STDSIP-78)	Defect	Terminal does spontaneous reboots, both during and outside calls. Terminal does spontaneous reboots, both during and outside calls know that the customer is not allowed to use 3.1.15.15, but they did install this package due to the fact that we waited with releasing 3.1.20.12 that has a formal fix for ITE-55372 (FR-903160029). 3.1.15.15 was prepared for testing the fix for FR-903160029 and was found as a fix. Therefore the field engineer decided to apply this package also for this customer. This issue was previously issued as FR-903160055	**External Defect** The fix of STDSIP-73 fixes this issue also.
3	STDSIP-73	Defect	DUT rebooted during the FR-001160092 testing with sipp when terminal needed to failback from primary to secondary server.	**Internal Defect**

1.5.18 List of Known Issues in STD-SIP v3.1.23.10

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T), Jira Issue ID(STDSIP)	Defect/Enhancement	Defect Description	Remarks

1.	STDSIP - 73	DUT rebooted during testing with sipp	DUT rebooted during the FR-001160092 testing with sipp when terminal needed to failback to primary from secondary server	
2.	B -758	DSS detection failed when disconnect/connect side -2 connecter	To reproduce steps following 1) Configure DSS-60 and reboot terminal 2) Make sure DSS-60 keys are functioning 3) Disconnect and reconnect, DSS-60 side 2 connecter Result: DSS detection failed after second or third attempt	
3.	T:27491	If "username" and password" is fetched from DHCP (option 66) then similar issue as bug 770 gets reproduced on v2.3.44.24.	To reproduce configure DHCP option 66 with ':' and '@' in user name and password. Example: "ftp://Sp!()Ch&%:India@1234:@10.112.94.82" Result: Terminals expecting FTP connection with "host=1234, user=Sp!()Ch&%, password=India", and its incorrect.	
4	FR-927160002 (STDSIP-77)	In the current implementation, only 503 and 580 were pulled out to play a tone (busy/reorder) in 5xx codes by phone	A customer in Italy has all DT730G phones. They are having a problem with certain calls that fail. The system topology consists of a 3C system and an IS3000 system connected via SIP trunk. The outside trunks are on the IS3000 side. The problem is as follows: 1- User on 3C making an outside call . 2- If the IS3000 returns a 504 Gateway Time-out, the MGC send this response to the phone and the phone terminates the call without playing any sort of tone 3- If the IS3000 returns a 502 Bad Gateway, the MGC send this response to the phone and the phone terminates the call without playing any sort of tone 4- If the IS3000 returns a 503 , the MGC sends this response to the phone and the phone plays an error tone for a second before terminating the call giving the user a chance to understand that the call errored We would like to know if this behavior can be reproduced on	

			the DT700 phones before we escalate this to Japan. Knowing if this is consistent between all the NEC phone models is important before we escalate this issue	
5	FR-927160003 (STDSIP-81)	If you make a call between two DT 730G phones , then answer the call, then mute both sides of the call, you will still hear low level background noise coming to originating phone	Reproduction in NECPF I tried the reproduction with DT700, DT730G and DT820. [Operation] The following steps are different from Sam-san's informed. However you will be able to confirm low noise easily. Step 1: TEL A calls TEL B with On Hook status (not use handset). Step 2: TEL B answers with handset. (With using handset, you can confirm low noise via handset.) Step 3: TEL A goes on Mute (press Mic button). And tap TEL A's mic that is located in front of TEL A with your finger. TEL B hears small noise via handset. [Result] Terminal FW version Result DT710 and DT730 2.3.57.1 Not reproduced. When I executed the above operation, I couldn't hear the noise. Also I executed same operation that Sam-san has informed. However I couldn't hear low noise in v2.3.57.1. DT730G 3.1.22.21 *Note Reproduced. I could hear low noise. DT820 4.0.22.21 *Note Same as above *Note These FW are the temporary version for FR-001160154. However the basic phone behavior is no changing from the previous official release version	

1.5.19 List of Fixed Issues in STD-SIP v3.1.22.3

S No.	Bugzilla ID (B), NEC Issue ID (N),	Defect/Enhancement	Defect Description	External/Internal	Remarks

	GM Issue ID , Tracker (T), Jira Issue ID(STDSIP)				
1	FR-903160038 (STDSIP-70)	Enhancement	**Don't offer the EndCall key if that results in clearing of a held call** **Trouble Symptom :** **Never offer the EndCall softkey if that results in clearing of a held call** **1) In SIP@Net whenever a user has a call on hold and clears the call while there's still a call on hold SIP@Net will offer the held call to the Operator for assistence.** **2) On the DT700 it is possible when you put a call on hold to clear the held call by pressing the EndCall softkey. This results in 1) causing a lot of confusion, to the held caller and the Operator.** **3) The issue also seems to be caused by ingorance of the users.** **4) To prevent the situation above we propose NOT to offer the EndCall softkey in case this will result in clearing of a held call. To prevent that the current behavior can still be kept we propsoe to introduce an option for that in the config file.** **5) By means of the above we prevent that the user can press the EndCall softkey in such situations, in the hold situation now the following Softkeys are offerred: EndCall Conference Resume NewCall It should be possible to configure the phone such that the EndCall softkey is no longer offered, so the Menua looks like this Conference Resume NewCall In this way the user can no longer select the EndCall softkey and needs to press**	External	• This change is waiting the customer's confirmation. This change doesn't impact current behavior in default. Therefore this change has been included in this release. • "phone.disable.endcall.softkey ="0"" has been added for this FR in dt-000000000000-sip.cfg.

			Resume to reActive the held call. This prevents that the situations as mentioned in 2) happen.		
2	FR-903160020/ T-27654 (STDSIP-58)	Defect	**DT700 Registration Icon changes when REGISTER transaction fails**	External	Merge from v2.3
3	FR-001160092/ T-28488 (STDSIP-71)	Defect	**DT700 has failed over to a survivable gateway, once it recovers back to 3C it fails to subscribe for message-summary**	External	Merge from v2.3
4	FR-001160093/ T-28489 (STDSIP-72)	Defect	**DT700 has failed over to a survivable gateway, once it recovers back to 3C park and pickup fails until phone is restarted.**	External	Merge from v2.3
5	FR-001160077/ T-28335	Defect	**Two phones entered into state where it was registering every 2 mins and stops responding to NOTIFY from MGC register**	External	Already merged with FR-001160092 and FR-001160093
6	FR-001160082/ T-28336	Defect	**DT700 stops subscribing for DPA keys**	External	Already merged with FR-001160092 and FR-001160093

1.5.20 List of Known Issues in STD-SIP v3.1.22.3

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T), Jira Issue ID(STDSIP)	Defect/Enhancement	Defect Description	Remarks
1.	STDSIP - 73	DUT rebooted during testing with sipp	DUT rebooted during the FR-001160092 testing with sipp when terminal needed to failback to primary from secondary server	

2.	B -758	DSS detection failed when disconnect/connect side -2 connecter	To reproduce steps following 1) Configure DSS-60 and reboot terminal 2) Make sure DSS-60 keys are functioning 3) Disconnect and reconnect, DSS-60 side 2 connecter Result: DSS detection failed after second or third attempt	
3.	T:27491	If “username” and password” is fetched from DHCP (option 66) then similar issue as bug 770 gets reproduced on v2.3.44.24.	To reproduce configure DHCP option 66 with ‘:’ and ‘@’ in user name and password. Example: “ftp://Sp!()Ch&%:India@1234:@10.112.94.82” Result: Terminals expecting FTP connection with “host=1234, user=Sp!()Ch&%, password=India”, and its incorrect.	

1.5.21 List of Fixed Issues in STD-SIP v3.1.20.12

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T), Jira Issue ID(STDSIP)	Defect/Enhancement	Defect Description	Remarks
1	FR-903160028 (STDSIP-62)	**Defect**	**With an EHS connected the DT730G does several reboots before getting operational**	**External Defect**
2	FR-903160029 (STDSIP-63)	**Defect**	**DT730G reboots several times when Headset connected**	**External Defect**

3	FR-903160035 (STDSIP-67)	**Defect**	**Wrong translation for Resume softkey in Dutch (nl_NL)**	**External Defect**
4	FR-903160036 T: 28592 (STDSIP-68)	**Defect**	**Parameter security.ipphn.mgr.mode set to 0 prevents sidecar to work**	**External Defect** **(This was discovered in v2.3.)**
5	FR-903160044 T:28721 (STDSIP- 69)	**Defect**	**DT730CG/DG don't send SUBSCRIBES for the DCL-60 keys**	**External Defect**
6	FR-903160016 T:27890 (STDSIP-60)	**Defect**	**DT700 gets operational very slow in Dual server environment**	**External Defect (Merge from v2.3)**
7	FR-903160020 T:27654 (STDSIP-58)	**Defect**	**DT700 shows not registered, while it is still registered**	**External Defect (Merge from v2.3)**
8	B-768 (STDSIP-50)	**Defect**	**[HCL]STD-SIPv3.1 HTTPS SSL Session Resumption Issue** Please consider following scenario 1) HTTPS protocol is set as boot protocol at DHCP. 2) Terminals boots up and perform TLS Client hello (before downloading configuration file using HTTPS) with session length 0. This is same in both v2.3 and v3.1 3) A Session Id is returned by HTTPS server in “Server Hello”. This is same in v2.3 and v3.0 4) All the configuration files are downloaded. After downloading of all files TLS connection is closed .If in downloaded master file (dt-<MAC>.cfg), Different firmware version is present then firmware download is triggered. For this TLS Client Hello is again sent by terminals. This is same in v2.3 and v3.0	**Internal Defect**

			5) In TLS client hello, v2.3 reuses Session Id sent by server in step 3. While v3.1 sent Client Hello is same as in step 2. To fix this defect, Curl Library was upgraded to new version 7.49.1.	
9	None	**Enhancement**	**Vulnerability of ntpd (NTP is upgraded to 4.2.8p8.)** 4.2.8p8 has fixed the following vulnerabilities. Therefore NTP was upgraded. The detail of 4.2.8p8 is written in the following URL. http://support.ntp.org/bin/view/Main/SoftwareDownloads#Stable_Release_NEWS [CVSS Severity: High] CVE-2016-4957 / VU#321640: Crypto-NAK crash [CVSS Severity: Medium] CVE-2016-4953 / VU#321640: Bad authentication demobilizes ephemeral associationsSec CVE-2016-4954 / VU#321640: Processing spoofed server packets CVE-2016-4955 / VU#321640: Autokey association reset CVE-2016-4956 / VU#321640: Broadcast interleave	**Fix the vulnerability**
10	None	**Defect**	**glibc vulnerability "GHOST" patch** This vulnerability has been fixed in v3.0.40.23. However we found that this patch wasn't included in v3.1. Therefore this release included this patch.	**Internal Defect**

1.5.22 List of Known Issues in STD-SIP v3.1.20.12

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T), Jira Issue ID(STDSIP)	Defect/Enhancement	Defect Description	Remarks
1.	B -758	DSS detection failed when disconnect/connect side -2 connecter	To reproduce steps following 1) Configure DSS-60 and reboot terminal	

			2) Make sure DSS-60 keys are functioning 3) Disconnect and reconnect, DSS-60 side 2 connecter Result: DSS detection failed after second or third attempt	
2.	T:27491	If "username" and password" is fetched from DHCP (option 66) then similar issue as bug 770 gets reproduced on v2.3.44.24.	To reproduce configure DHCP option 66 with ':' and '@' in user name and password. Example: "ftp://Sp!()Ch&%:India@1234:@10.112.94.82" Result: Terminals expecting FTP connection with "host=1234, user=Sp!()Ch&%, password=India", and its incorrect.	

1.5.23 List of Fixed Issues in STD-SIP v3.1.18.7

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T), Jira Issue ID(STDSIP)	Defect/Enhancement	Defect Description	Remarks
1	FR-903160018 / STDSIP-61	**Defect**	**[NECU]_FR-903160018 DT820 does not send a SUBSCRIBE**	**This issue is reported on DT820. However since source code which is causing this issue is common in DT730G and DT820 terminals therefore fix of**

				this issue is also applied in STD-SIPv3.1.
2	T:27890 / FR-903160016 / STDSIP-60	**Defect**	**DT700 gets operational very slow in Dual server environment**	
3	T:27654 / FR-903160020 / STDSIP-58	**Defect**	**DT700 Registration Icon changes when REGISTER transaction fails**	

1.5.24 List of Known Issues in STD-SIP v3.1.18.7

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T), Jira Issue ID(STDSIP)	Defect/Enhancement	Defect Description	Remarks
1.	B -758	DSS detection failed when disconnect/connect side -2 connecter	To reproduce steps following 1) Configure DSS-60 and reboot terminal 2) Make sure DSS-60 keys are functioning 3) Disconnect and reconnect, DSS-60 side 2 connecter Result: DSS detection failed after second or third attempt	
2.	B-768	STD-SIPv3.1 HTTPS SSL Session Resumption and session signature reuse.	Please consider following scenario 1) HTTPS protocol is set as boot protocol at DHCP. 2) Terminals boots up and perform TLS Client hello (before downloading configuration file using HTTPS) with session length 0. This is same in both	

			v2.3 and v3.1 3) A Session Id is returned by HTTPS server in "Server Hello". This is same in v2.3 and v3.0 4) All the configuration files are downloaded. After downloading of all files TLS connection is closed .If in downloaded master file (dt-<MAC>.cfg), Different firmware version is present then firmware download is triggered. For this TLS Client Hello is again sent by terminals. This is same in v2.3 and v3.0 5) In TLS client hello, v2.3 reuses Session Id sent by server in step 3. While v3.1 sent Client Hello is same as in step 2.	
3.	T:27491	If "username" and password" is fetched from DHCP (option 66) then similar issue as bug 770 gets reproduced on v2.3.44.24.	To reproduce configure DHCP option 66 with ':' and '@' in user name and password. Example: "ftp://Sp!()Ch&%:India@1234:@10.112.94.82" Result: Terminals expecting FTP connection with "host=1234, user=Sp!()Ch&%, password=India", and its incorrect.	

1.5.25 List of Fixed Issues in STD-SIP v3.1.17.10

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T), Jira Issue ID(STDSIP)	Defect/Enhancement	Defect Description	Remarks
1.	STDSIP-59	[NECPF]MW LED blink by XML's PushLEDItem	Use Push client tool When "6 (Light blue)" or "8 (Gradation)" is set in color of PushLED Item, MW LED blinks with red.	

				External Defect

1.5.26 List of Known Issues in STD-SIP v3.1.17.10

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T), Jira Issue ID(STDSIP)	Defect/Enhancement	Defect Description	Remarks
1.	B -758	DSS detection failed when disconnect/connect side -2 connecter	To reproduce steps following 1) Configure DSS-60 and reboot terminal 2) Make sure DSS-60 keys are functioning 3) Disconnect and reconnect, DSS-60 side 2 connecter Result: DSS detection failed after second or third attempt	
2.	B-768	STD-SIPv3.1 HTTPS SSL Session Resumption and session signature reuse.	Please consider following scenario 1) HTTPS protocol is set as boot protocol at DHCP. 2) Terminals boots up and perform TLS Client hello (before downloading configuration file using HTTPS) with session length 0. This is same in both v2.3 and v3.1 3) A Session Id is returned by HTTPS server in “Server Hello”. This is same in v2.3 and v3.0 4) All the configuration files are downloaded. After downloading of all files TLS connection is closed .If in downloaded master file (dt-<MAC>.cfg),	

			Different firmware version is present then firmware download is triggered. For this TLS Client Hello is again sent by terminals. This is same in v2.3 and v3.0 5) In TLS client hello, v2.3 reuses Session Id sent by server in step 3. While v3.1 sent Client Hello is same as in step 2.	
3.	T:27491	If “username” and password” is fetched from DHCP (option 66) then similar issue as bug 770 gets reproduced on v2.3.44.24.	To reproduce configure DHCP option 66 with ‘:’ and ‘@’ in user name and password. Example: “ftp://Sp!()Ch&%:India@1234:@10.112.94.82” Result: Terminals expecting FTP connection with “host=1234, user=Sp!()Ch&%, password=India”, and its incorrect.	

1.5.27 List of Fixed Issues in STD-SIP v3.1.17.6

None

1.5.28 List of Known Issues in STD-SIP v3.1.17.6

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T)	Defect/Enhancement	Defect Description	Remarks
1.	B -758	DSS detection failed when disconnect/connect side -2 connecter	To reproduce steps following 1) Configure DSS-60 and reboot terminal 2) Make sure DSS-60 keys are functioning 3) Disconnect and reconnect, DSS-60 side 2 connecter Result: DSS detection failed after second or third attempt	

2.	B-768	STD-SIPv3.1 HTTPS SSL Session Resumption and session signature reuse.	Please consider following scenario 1) HTTPS protocol is set as boot protocol at DHCP. 2) Terminals boots up and perform TLS Client hello (before downloading configuration file using HTTPS) with session length 0. This is same in both v2.3 and v3.1 3) A Session Id is returned by HTTPS server in “Server Hello”. This is same in v2.3 and v3.0 4) All the configuration files are downloaded. After downloading of all files TLS connection is closed .If in downloaded master file (dt-<MAC>.cfg), Different firmware version is present then firmware download is triggered. For this TLS Client Hello is again sent by terminals. This is same in v2.3 and v3.0 5) In TLS client hello, v2.3 reuses Session Id sent by server in step 3. While v3.1 sent Client Hello is same as in step 2.	
3.	T:27491	If “username” and password” is fetched from DHCP (option 66) then similar issue as bug 770 gets reproduced on v2.3.44.24.	To reproduce configure DHCP option 66 with ‘:’ and ‘@’ in user name and password. Example: “ftp://Sp!()Ch&%:India@1234:@10.112.94.82” Result: Terminals expecting FTP connection with “host=1234, user=Sp!()Ch&%, password=India”, and its incorrect.	

1.5.29 List of Fixed Issues in STD-SIP v3.1.15.15

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID ,	Defect/Enhancement	Defect Description	Remarks

	Tracker (T)			
1	STDSIP-20	Enhancement	**v3.1- [NECECT]DT730G should check that the firmware version is 3.1.x.x before downloading.**	External
2	STDSIP-18	Defect	**v3.1 - [NECPF]018_Incorrect "Please Wait..." indication in French (France and Belgium).**	External
3	T:27335 / FR-001150238	Defect	**Memory Leak Issue**	External
4	T:27653	Defect	**V2.3 - DT700 crashes due to freeing an invalid memory block** **[Note]** **Similar issue has already been fixed in v3.1.**	External
5	STDSIP-4	Defect	**[NECGB29] speed mismatch than that of set in HP PoE switch when LAN port vlan is enabled**	Internal
6	STDSIP-5	Defect	**[NECGB033] speed mismatch than that of set in PC NIC when PC port vlan is enabled.**	Internal
7	STDSIP-6	Defect	**[NECGB008] Autonegotiation and speed are mismatch in wireshark that of set in terminal in v4.0**	Internal
8	STDSIP-7	Defect	**LLDP_8/NECLLDP008 - there is no description for 1000M Half Duplax in LLDP packet**	Internal
9	STDSIP-12	Defect	**v3.1 - In LLDP packets under the capabilities if we see the capabilities value its 0x0024(Bridge: capable , Telephone: capable) and enabled capabilities is 0x0020 (Bridge: Not capable, Telephone: capable**	Internal

1.5.30 List of Known Issues in STD-SIP v3.1.15.15

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T)	Defect/Enhancement	Defect Description	Remarks

1.	B -758	DSS detection failed when disconnect/connect side -2 connecter	To reproduce steps following 1) Configure DSS-60 and reboot terminal 2) Make sure DSS-60 keys are functioning 3) Disconnect and reconnect, DSS-60 side 2 connecter Result: DSS detection failed after second or third attempt	
2.	B-768	STD-SIPv3.1 HTTPS SSL Session Resumption and session signature reuse.	Please consider following scenario 1) HTTPS protocol is set as boot protocol at DHCP. 2) Terminals boots up and perform TLS Client hello (before downloading configuration file using HTTPS) with session length 0. This is same in both v2.3 and v3.1 3) A Session Id is returned by HTTPS server in “Server Hello”. This is same in v2.3 and v3.0 4) All the configuration files are downloaded. After downloading of all files TLS connection is closed .If in downloaded master file (dt-<MAC>.cfg), Different firmware version is present then firmware download is triggered. For this TLS Client Hello is again sent by terminals. This is same in v2.3 and v3.0 5) In TLS client hello, v2.3 reuses Session Id sent by server in step 3. While v3.1 sent Client Hello is same as in step 2.	
3.	T:27491	If “username” and password” is fetched from DHCP (option 66) then similar issue as bug 770 gets reproduced on v2.3.44.24.	To reproduce configure DHCP option 66 with ‘:’ and ‘@’ in user name and password. Example: “ftp://Sp!()Ch&%:India@1234:@10.112.94.82” Result: Terminals expecting FTP connection with “host=1234, user=Sp!()Ch&%, password=India”, and its incorrect.	

1.5.31 List of Fixed Issues in STD-SIP v3.1.14.22

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T)	Defect/Enhancement	Defect Description	Remarks
1	STDSIP-19	Enhancement	[NECAM](AR-903160003)One Touch Deployment	

1.5.32 List of Known Issues in STD-SIP v3.1.14.22

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T)	Defect/Enhancement	Defect Description	Remarks
1.	B -758	DSS detection failed when disconnect/connect side -2 connecter	To reproduce steps following 1) Configure DSS-60 and reboot terminal 2) Make sure DSS-60 keys are functioning 3) Disconnect and reconnect, DSS-60 side 2 connecter Result: DSS detection failed after second or third attempt	
2.	B-768	STD-SIPv3.1 HTTPS SSL Session Resumption and session signature reuse.	Please consider following scenario 1) HTTPS protocol is set as boot protocol at DHCP. 2) Terminals boots up and perform TLS Client hello (before downloading configuration file using HTTPS) with session length 0. This is same in both v2.3 and v3.1	

			3) A Session Id is returned by HTTPS server in “Server Hello”. This is same in v2.3 and v3.0 4) All the configuration files are downloaded. After downloading of all files TLS connection is closed .If in downloaded master file (dt-<MAC>.cfg), Different firmware version is present then firmware download is triggered. For this TLS Client Hello is again sent by terminals. This is same in v2.3 and v3.0 5) In TLS client hello, v2.3 reuses Session Id sent by server in step 3. While v3.1 sent Client Hello is same as in step 2.	
3.	T:27491	If “username” and password” is fetched from DHCP (option 66) then similar issue as bug 770 gets reproduced on v2.3.44.24.	To reproduce configure DHCP option 66 with ‘:’ and ‘@’ in user name and password. Example: “ftp://Sp!()Ch&%:India@1234:@10.112.94.82” Result: Terminals expecting FTP connection with “host=1234, user=Sp!()Ch&%, password=India”, and its incorrect.	

1.5.33 List of Fixed Issues in STD-SIP v3.1.14.9

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T)	Defect/Enhancement	Defect Description	Remarks
1	AR-001160013 (SITD-304)	Defect	NECAM (AR-001160013) - User gets an error when "List Conferences" is executed on DT820 when conference address contains Japanese characters.	Merged from v4.0
2	BR-001160002 (SITD-365)	Defect	[NECAM][BR-001160002]with two DNS terminal fails to REGISTER	Merged from v4.0

1.5.34 List of Known Issues in STD-SIP v3.1.14.9

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T)	Defect/Enhancement	Defect Description	Remarks
1.	B -758	DSS detection failed when disconnect/connect side -2 connecter	To reproduce steps following 1) Configure DSS-60 and reboot terminal 2) Make sure DSS-60 keys are functioning 3) Disconnect and reconnect, DSS-60 side 2 connecter Result: DSS detection failed after second or third attempt	
2.	B-768	STD-SIPv3.1 HTTPS SSL Session Resumption and session signature reuse.	Please consider following scenario 1) HTTPS protocol is set as boot protocol at DHCP. 2) Terminals boots up and perform TLS Client hello (before downloading configuration file using HTTPS) with session length 0. This is same in both v2.3 and v3.1 3) A Session Id is returned by HTTPS server in “Server Hello”. This is same in v2.3 and v3.0 4) All the configuration files are downloaded. After downloading of all files TLS connection is closed .If in downloaded master file (dt-<MAC>.cfg), Different firmware version is present then firmware download is triggered. For this TLS Client Hello is again sent by terminals. This is same in v2.3 and v3.0 5) In TLS client hello, v2.3 reuses Session Id sent by server in step 3. While v3.1 sent Client Hello is same as in step 2.	

3.	T:27491	If “username” and password” is fetched from DHCP (option 66) then similar issue as bug 770 gets reproduced on v2.3.44.24.	To reproduce configure DHCP option 66 with ‘:’ and ‘@’ in user name and password. Example: “ftp://Sp!()Ch&%:India@1234:@10.112.94.82” Result: Terminals expecting FTP connection with “host=1234, user=Sp!()Ch&%, password=India”, and its incorrect.	

1.5.35 List of Fixed Issues in STD-SIP v3.1.14.4

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T)	Defect/Enhancement	Defect Description	Remarks
1	Tracker 26389	Call to PSTN via Adtran MG fails with Optional SRTP calls from Dterm	When SDP Answer's SAVP media line, ** does not have a Crypto attribute **, DTERM terminates the outbound call. However	
2	Tracker 26390	DT(Optional SRTP with CR) upholding an RTP only call disconnects the call	During an SDP update the DTERM recreates the SDP based on the originally configured SRTP mode (MANDATORY, OPTIONAL or DISABLED) ending up in sending both media lines (SAVP and AVP), irrespective of the originally established session mode.	
3	Tracker 26400	DT700 Doesn't upload local overrides when configured for SIP@Net User Portability and an Admin setting is changed locally	The MAC override file (System Test is testing the MAC file upload versus user file upload) is not getting uploaded to the boot server; as the RESET event (on Admin settings change) that triggers the upload does not reach the XML config task; before the phone reboots.	
4	Tracker 26588	Some phones do not automatically failback to the Primary Registar after a	Failback does not work, as the the certificate validation fails during PING exchange. The phone (logic) tries to compare the current registrar address (Secondary) to the	

		failover has been experienced when using TLS as the SIP transport	CN/SAN in the certificate, and as it does not match, the validation fails.	
5	Tracker 26947	HTTPS connection fails with error: "TLS Unknown CA" after downloading a custom CA with the same name as the default CA file name.	On downloading a custom CA with the same name as the default CA (TLSrootCA.pem) file name, HTTPS connection fails. The issue was because, the startup logic copies the CA file, thus overwriting the custom CA file, without checking whether the file already exists.	
6	Tracker 27193	Race condition with SNTP	Race with SNTP response delays firmware download by HTTPS	
7	Tracker 27357	EPK Web service	The DT700 micro browser sends HTTP requests to the base URL configured in the 3C Admin (phone.home.url). It is a feature request,Currently, the EPK web service request is sent to the registered UCM	

1.5.36 List of Known Issues in STD-SIP v3.1.14.4

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T)	Defect/Enhancement	Defect Description	Remarks
1.	B -758	DSS detection failed when disconnect/connect side -2 connecter	To reproduce steps following 1) Configure DSS-60 and reboot terminal 2) Make sure DSS-60 keys are functioning 3) Disconnect and reconnect, DSS-60 side 2 connecter Result: DSS detection failed after second or third attempt	

2.	B-768	STD-SIPv3.1 HTTPS SSL Session Resumption and session signature reuse.	Please consider following scenario 1) HTTPS protocol is set as boot protocol at DHCP. 2) Terminals boots up and perform TLS Client hello (before downloading configuration file using HTTPS) with session length 0. This is same in both v2.3 and v3.1 3) A Session Id is returned by HTTPS server in “Server Hello”. This is same in v2.3 and v3.0 4) All the configuration files are downloaded. After downloading of all files TLS connection is closed .If in downloaded master file (dt-<MAC>.cfg), Different firmware version is present then firmware download is triggered. For this TLS Client Hello is again sent by terminals. This is same in v2.3 and v3.0 5) In TLS client hello, v2.3 reuses Session Id sent by server in step 3. While v3.1 sent Client Hello is same as in step 2.	
3.	T:27491	If “username” and password” is fetched from DHCP (option 66) then similar issue as bug 770 gets reproduced on v2.3.44.24.	To reproduce configure DHCP option 66 with ‘:’ and ‘@’ in user name and password. Example: “ftp://Sp!()Ch&%:India@1234:@10.112.94.82” Result: Terminals expecting FTP connection with “host=1234, user=Sp!()Ch&%, password=India”, and its incorrect.	

1.5.37 List of Fixed Issues in STD-SIP v3.1.12.16

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID ,	Defect/Enhancement	Defect Description	Remarks

	Tracker (T)			
1.	B-771	STD-SIPv3.1 Issue T21340 - DT700: User portability fails if dt-dnr-local.cfg is not present on IIS	To reproduce steps following 1) Reboot DUT and before boot up remove dt-xxxx-local.cfg from IIS server 2) Go to Menu-> user setting do some modification Result: If a dt-xxxx-local.cfg is not present on IIS the DT700 does not generate dt-xxxx-local.cfg	
2.	B-772	OpenSSL 1.0.1.2e upgrade	Upgrade OpenSSL library to version 1.0.1.2e	

1.5.38 List of Known Issues in STD-SIP v3.1.12.16

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T)	Defect/Enhancement	Defect Description	Remarks
1.	B -758	DSS detection failed when disconnect/connect side -2 connecter	To reproduce steps following 1) Configure DSS-60 and reboot terminal 2) Make sure DSS-60 keys are functioning 3) Disconnect and reconnect, DSS-60 side 2 connecter Result: DSS detection failed after second or third attempt	
2.	B-768	STD-SIPv3.1 HTTPS SSL Session Resumption and session signature reuse.	Please consider following scenario 1) HTTPS protocol is set as boot protocol at DHCP. 2) Terminals boots up and perform TLS Client hello (before downloading configuration file using HTTPS) with session length 0. This is same in both	

			v2.3 and v3.1 3) A Session Id is returned by HTTPS server in “Server Hello”. This is same in v2.3 and v3.0 4) All the configuration files are downloaded. After downloading of all files TLS connection is closed .If in downloaded master file (dt-<MAC>.cfg), Different firmware version is present then firmware download is triggered. For this TLS Client Hello is again sent by terminals. This is same in v2.3 and v3.0 5) In TLS client hello, v2.3 reuses Session Id sent by server in step 3. While v3.1 sent Client Hello is same as in step 2.	
3.	T:27491	If “username” and password” is fetched from DHCP (option 66) then similar issue as bug 770 gets reproduced on v2.3.44.24.	To reproduce configure DHCP option 66 with ‘:’ and ‘@’ in user name and password. Example: “ftp://Sp!()Ch&%:India@1234:@10.112.94.82” Result: Terminals expecting FTP connection with “host=1234, user=Sp!()Ch&%, password=India”, and its incorrect.	

1.5.39 List of Fixed Issues in STD-SIP v3.1.12.1

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T)	Defect/Enhancement	Defect Description	Remarks
3.	B-770	Handling of special character like ‘@:%....’ etc. in username/password	If protocol is FTP/HTTP and “username/password” contains ‘: / @ ’ then curl library failing to parse correct username or password because syntax of curl library is “username:password@hostname:port/file” and ‘: / @’	

			are part of syntax so firmware download always failed if “username/password” have special character.	
4.	B-767	V3.1 needs to support MOS-LQ and CQ for G.711 and the other CODECs	V3.1 needs to support MOS-LQ and CQ for G.711 and the other CODECs. This will be done once the Broadcom patch that supports MOS-LQ and CQ for G.711 and the other CODECs is provided by Broadcom.	

1.5.40 List of Known Issues in STD-SIP v3.1.12.1

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T)	Defect/Enhancement	Defect Description	Remarks
1.	B -758	DSS detection failed when disconnect/connect side -2 connecter	To reproduce steps following 1) Configure DSS-60 and reboot terminal 2) Make sure DSS-60 keys are functioning 3) Disconnect and reconnect, DSS-60 side 2 connecter Result: DSS detection failed after second or third attempt	
2.	B-768	STD-SIPv3.1 HTTPS SSL Session Resumption and session signature reuse.	Please consider following scenario 1) HTTPS protocol is set as boot protocol at DHCP. 2) Terminals boots up and perform TLS Client hello (before downloading configuration file using HTTPS) with session length 0. This is same in both v2.3 and v3.1 3) A Session Id is returned by HTTPS server in “Server Hello”. This is same in v2.3 and v3.0	

			4) All the configuration files are downloaded. After downloading of all files TLS connection is closed .If in downloaded master file (dt-<MAC>.cfg), Different firmware version is present then firmware download is triggered. For this TLS Client Hello is again sent by terminals. This is same in v2.3 and v3.0 5) In TLS client hello, v2.3 reuses Session Id sent by server in step 3. While v3.1 sent Client Hello is same as in step 2.	
3.	B-771	STD-SIPv3.1 Issue T21340 - DT700: User portability fails if dt-dnr-local.cfg is not present on IIS	To reproduce steps following 1. Reboot DUT and before boot up remove dt-xxxx-local.cfg from IIS server 2. Go to Menu-> user setting do some modification Result: If a dt-xxxx-local.cfg is not present on IIS the DT700 does not generate dt-xxxx-local.cfg	

1.5.41 List of Fixed Issues in STD-SIP v3.1.10.29

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T)	Defect/Enhancement	Defect Description	Remarks
1.	B-769	STD-SIPv3.1 Code Signing Certificate Expiry	STD-SIPv3.1 Code Signing Certificate has expired on October 9 as a result signed firmware feature is not working.	

1.5.42 List of Known Issues in STD-SIP v3.1.10.29

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T)	Defect/Enhancement	Defect Description	Remarks
1.	B -758	DSS detection failed when disconnect/connect side -2 connecter	To reproduce steps following 1) Configure DSS-60 and reboot terminal 2) Make sure DSS-60 keys are functioning 3) Disconnect and reconnect, DSS-60 side 2 connecter Result: DSS detection failed after second or third attempt	
2.	B-767	V3.1 needs to support MOS-LQ and CQ for G.711 and the other CODECs	V3.1 needs to support MOS-LQ and CQ for G.711 and the other CODECs. This will be done once the Broadcom patch that supports MOS-LQ and CQ for G.711 and the other CODECs is provided by Broadcom.	
3.	B-768	STD-SIPv3.1 HTTPS SSL Session Resumption and session signature reuse.	Please consider following scenario 1) HTTPS protocol is set as boot protocol at DHCP. 2) Terminals boots up and perform TLS Client hello (before downloading configuration file using HTTPS) with session length 0. This is same in both v2.3 and v3.1 3) A Session Id is returned by HTTPS server in “Server Hello”. This is same in v2.3 and v3.0 4) All the configuration files are downloaded. After downloading of all files TLS connection is closed .If in downloaded master file (dt-<MAC>.cfg),	

			Different firmware version is present then firmware download is triggered. For this TLS Client Hello is again sent by terminals. This is same in v2.3 and v3.0 5) In TLS client hello,v2.3 reuses Session Id sent by server in step 3. While v3.1 sent Client Hello is same as in step 2.	

1.5.43 List of Fixed Issues in STD-SIP v3.1.10.26

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T)	Defect/Enhancement	Defect Description	Remarks
5.	B-766 T:26369	T:26369 DT700: Feature request to support Best Effort SRTP	When MC SMP srtp mode is set to Optional, it sends media as RTP (media line with AVP), but includes crypto attribute(s). DTERM ends up answering with RTP, as it reads the offer media as RTP ignoring the crypto attribute(s).	
6.	B-765	Openssl 1.0.1.p upgrade	Upgrade OpenSSL library to version 1.0.1p	

1.5.44 List of Known Issues in STD-SIP v3.1.10.26

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID ,	Defect/Enhancement	Defect Description	Remarks

	Tracker (T)			
1.	B -758	DSS detection failed when disconnect/connect side -2 connecter	To reproduce steps following 1) Configure DSS-60 and reboot terminal 2) Make sure DSS-60 keys are functioning 3) Disconnect and reconnect, DSS-60 side 2 connecter Result: DSS detection failed after second or third attempt	
2.	B-767	V3.1 needs to support MOS-LQ and CQ for G.711 and the other CODECs	V3.1 needs to support MOS-LQ and CQ for G.711 and the other CODECs. This will be done once the Broadcom patch that supports MOS-LQ and CQ for G.711 and the other CODECs is provided by Broadcom.	
3.	B-768	STD-SIPv3.1 HTTPS SSL Session Resumption and session signature resuse.	Please consider following scenario 1) HTTPS protocol is set as boot protocol at DHCP. 2)Terminals boots up and perform TLS Client hello(before downloading configuration file using HTTPS) with session length 0. This is same in both v2.3 and v3.1 3) A Session Id is returned by HTTPS server in “Server Hello”. This is same in v2.3 and v3.0 4) All the configuration files are downloaded. After downloading of all files TLS connection is closed .If in downloaded master file (dt-<MAC>.cfg), Different firmware version is present then firmware download is triggered. For this TLS Client Hello is again sent by terminals. This is same in v2.3 and v3.0 5) In TLS client hello,v2.3 reuses Session Id sent by server in step 3. While v3.1 sent Client Hello is same as in step 2.	

1.5.45 List of Fixed Issues in STD-SIP v3.1.10.14

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T)	Defect/Enhancement	Defect Description	Remarks
7.	B -764	DSS-60	Add DSS-60 support in v3.1	
8.	B-759	Signed Firmware Support	Add Signed Firmware support in v3.1	
9.	B-760	RTCP-XR	Add RTCP-XR support in v3.1	
10.	B-762	Port hardening	Add Port hardening support in v3.1	
11.	B-763	Hotline	Add HOTLINE support in v3.1	
12.	B-755	HTTPS Protocol	Add HTTPS protocol Support, including mutual TLS for config/firmware file download/Upload, in v3.1	
13.	B-761	802.1X Ehancements	P:0186,T:14552,SIP Phone Main - DT700: Setting to factory defaults keeps 802.1x username and password	
14.	B-757	DSS-60 is not detected sometimes	DSS-60 is not detected sometimes when phone is rebooted. The occurrence was 1 out of 10 times. If we unplug power supply then DSS detection failed every time.	

1.5.46 List of Known Issues in STD-SIP v3.1.10.14

S No.	Bugzilla ID (B), NEC Issue ID (N),	Defect/Enhancement	Defect Description	Remarks

	GM Issue ID , Tracker (T)			
1.	B -758	DSS detection failed when disconnect/connect side -2 connecter	To reproduce steps following 1) Configure DSS-60 and reboot terminal 2) Make sure DSS-60 keys are functioning 3) Disconnect and reconnect, DSS-60 side 2 connecter Result: DSS detection failed after second or third attempt	
2.	B-765	Openssl 1.0.1.p upgrade	Upgrade OpenSSL library to version 1.0.1p	
3.	B-766 T:26369	T:26369 DT700: Feature request to support Best Effort SRTP	When MC SMP srtp mode is set to Optional, it sends media as RTP (media line with AVP), but includes crypto attribute(s). DTERM ends up answering with RTP, as it reads the offer media as RTP ignoring the crypto attribute(s).	
4.	B-767	V3.1 needs to support MOS-LQ and CQ for G.711 and the other CODECs	V3.1 needs to support MOS-LQ and CQ for G.711 and the other CODECs. This will be done once the Broadcom patch that supports MOS-LQ and CQ for G.711 and the other CODECs is provided by Broadcom.	
5.	B-768	STD-SIPv3.1 HTTPS SSL Session Resumption and session signature resuse.	Please consider following scenario 1) HTTPS protocol is set as boot protocol at DHCP. 2)Terminals boots up and perform TLS Client hello(before downloading configuration file using HTTPS) with session length 0. This is same in both v2.3 and v3.1	

			3) A Session Id is returned by HTTPS server in “Server Hello”. This is same in v2.3 and v3.0 4) All the configuration files are downloaded. After downloading of all files TLS connection is closed .If in downloaded master file (dt-<MAC>.cfg), Different firmware version is present then firmware download is triggered. For this TLS Client Hello is again sent by terminals. This is same in v2.3 and v3.0 5) In TLS client hello,v2.3 reuses Session Id sent by server in step 3. While v3.1 sent Client Hello is same as in step 2.	

1.5.47 List of Fixed Issues in STD-SIP v3.0.43.9

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T)	Defect/Enhancement	Defect Description	Remarks
1.	B -754	SIP@NET dt-<MAC-local File upload issue for DT730G Color terminal	There is an issue with upload of dt-<Mac>-local.cfg in STD-SIP v3.0.43.5 color terminal with SIP@NET. This issue doesn't occur on gray terminal. This issue also doesn't occur on 3C for both gray and color terminal(STD-SIP v3.0.43.5). Following is the scenario 1) User portability is enabled on SIP@NET. 2) Change any admin pararmeter(like "telenet mode"). And reboot. 3) dt-<MAC>-local.cfg file is not uploaded on SIP@NET for color terminals.	

1.5.48 List of Fixed Issues in STD-SIP v3.0.43.5

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T)	Defect/Enhancement	Defect Description	Remarks
1.	B -745 T-25553	DT700 Registers and Unregisters unnecessarily	When the DNS SRV record ages out of the DT700's DNS cache, the DT700 unregisters and re-registers unnecessarily. The visible symptom of this problem is the registration icon goes from solid to hollow and back to solid for roughly 30 seconds. This occurs because the MGC responds 480 Temporarily Unavailable to the re-register request. The DT700 retries the registration 30 seconds later, and receives a 200 OK.	
2.	B-747 FR-001140385 T-25418	DT700: Factory default doesn't clear speed dials	The phone will clear the dt-<MAC>-directory.cfg and dt-<MAC>-history.cfg files on factory default.	
3.	B-748 FR-001150090 T-25914	Dialing an outside number while on-hook, and an incoming call receives, all digits entered prior to reviving the call is wiped out and outgoing call is interrupted	If user is dialing an outside number while on-hook, and an incoming call receives, all digits entered prior to reviving the call is wiped out and outgoing call is interrupted. The site was upgraded from 8.5.3.11-P5 to P8. Within last couple of days the customer experiencing the following: 1- While the phone is on hook the user dial an outbound number. 2- While the user enters the outbound digit numbers (example: 9-1-214-6) receives a call. 3- All digits that were already dialed (9-1-214-6), cleared from the screen and outbound call is never made after user finish entering the rest of the outbound call digits.	

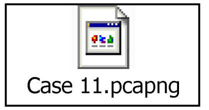
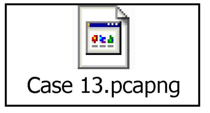
4.	B-749 FR-001150078 T-25830	DT700: Feature Request to support wildcard certificates	The DT700 does not supports wildcard certificates for SIP over TLS. This functionality would help NEC Canada, since they intend to have multiple customers' phones pointed to the same SBC, but via different domain names. For example, customer1.neccanada.ca, customer2.neccanada.ca, etc. NEC Canada can add these domain names to the SANs, but this becomes cumbersome after a few names. It also requires reloading the certificate on the SBC.	
5.	B-750 T-26095	Display names longer than 31 characters do not appear.	line.1.displayname is limited to 31 characters. If more than 31 characters are specified, nothing is displayed. It would make more sense to display a truncated string. Suggest truncating from the left. Note: The 3C Admin writes "FirstnameLastname" to line.1.displayname. The addition of the space limits the combination of first name and last name to 30 characters.	
6.	B-751	“Call Back On Idle” Feature	STD-SIP v3.0 did not supported “Call Back On Idle” feature which is available on STD-SIP v2.3 firmware. With this release support of this feature is added to STD-SIP v3.0.	
7.	B-752 FR-001150100	When a call is placed into Park, you cannot pick up the call until the your telephone is idle.	Big City Auto would like a change in behavior to the DT700 Enhanced Programmable Keys (EPKs). Big City Auto has a number of call agents that are members of a group address. These agents receive many calls. It’s common for each agent to have multiple calls ringing on their phone at any given time. The first available agent	

<table>
<tr><td></td><td></td><td></td><td>answers and processes a call. They frequently park and pickup calls in their
daily work flow.

The DT700 only allows the user to pick up a parked call when the user’s phone
is idle. Big City Auto would like the ability to pick up a parked call when
the user’s phone has one or more ringing calls.</td><td></td></tr>
<tr><td>8.</td><td>B-753
T -25869</td><td>Media not setup with 3C SMP when SRTP is optional</td><td>During SRTP interop testing with the 3C SMP, two cases fail as shown below. The same works well between 3CSMP and Polycom.

<table><tr><td>CallFrom(SRTP mode)</td><td>CallTo(SRTP mode)</td><td>Result</td></tr><tr><td>DT700(optional)</td><td>3CSMP(disabled)</td><td>Media not encrypted.</td></tr><tr><td>DT700(optional)</td><td>3CSMP(optional)</td><td>Media is encrypted.</td></tr></table>The problem seems to be that DT700 does not process the SDP returned from 3CSMP correctly. Below is a snapshot that shows two media (audio) lines in the SDP i.e., RTP/SAVP and RTP/AVP; one with an active port and one with an inactive (0) port. In both the cases, DT700 successfully establishes the call, but not the speech path, probably because of misrouting the audio path (using wrong RTP port/contact address).</td><td>Please see Note below.</td></tr>
</table>

Note: The following two issues were discovered during the testing of T:25869, and were 3C's issue.

A SRTP Mode	B SRTP Mode	Steps And Expected Results	Actual Result
RTP Only(0) Auto Call recording On 3C	Optional(1)	1) Establish call between A party and B party. Verify that voice is exchanged on RTP through 3C. 2) Put call on hold from A. Verify that MOH(on RTP) is played on B. 3) Unhold call from A. Verify that voice is exchanged on RTP through 3C. 4) Put call on hold from B. Verify that MOH(on RTP) is played on A. 5) Unhold call from B. Verify that voice is exchanged on RTP through 3C. 6) Disconnect Call	**After Step 5, B party is sending Bye.** 
Optional(1) Auto Call recording On 3C	RTP Only(0)	1) Establish call between A party and B party. Verify that voice is exchanged on RTP through 3C.. 2) Put call on hold from A. Verify that MOH(on RTP) is played on B. 3) Unhold call from A. Verify that voice is exchanged on	**No Voice path established after step 1.** 

		RTP through 3C. 4) Put call on hold from B. Verify that MOH(on RTP) is played on A. 5) Unhold call from B. Verify that voice is exchanged on RTP through 3C.. 6) Disconnect Call	

1.5.49 List of Fixed Issues in STD-SIP v3.0.41.4

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T)	Defect/Enhancement	Defect Description	Remarks
1.	B - 744 FR-903150013	The DT700 does no longer respond to INVITES.	After installing firmware on DT730G, they lose connection after 7-8 minutes and they stop answering to ARPrequests. If the Network consultant is setting the timer for clearing the ARP table to less than 5 min, the phones are runningsmoothly	
2.	B - 743 T-25862	DT700: Upgrade OpenSSL libraries to version 1.0.1m	Security flaws have been found in OpenSSL 1.0j. The DTERM should be upgraded to OpenSSL 1.0.1m.	
3.	B – 742 T: 25985	DT700: 802.1X Connecting message not displayed correctly in languages other than English	The first issue is the string “802.1X: Connecting(x)” appearing during 802.1X authentication isn’t translated for any country which does not use English.	

			The second issue is the string “Press Exit to Reset.” appearing during 802.1X authentication isn't translated when country is set to Argentina, Chile, Colombia or France. This string gets displayed below line “802.1X: Connecting(x)”. For the remaining countries this string is translated correctly.	
4.	B-746 T: 25564	DT700: Stops transmitting and receiving RTP midway through call	The DT700 reuses the socket that was used for the FTP data connection for the RTP stream. Note that the socket is an internal representation of the network port maintained by the OS. The socket is allocated by the OS, so our code isn’t necessarily at fault here. When the TCP RST arrives for the FTP data connection, the DT700 mistakenly closes the RTP stream. I’m speculating that because there are unacknowledged packets on the FTP data connection when the TCP RST is received from the UCM, the OS is cleaning up the socket without realizing that it (the OS) has reused the socket for a completely different purpose.	This issue was reproducible on STD-SIP v2.3.33.11. **This issue is not reproducible on STD-SIP v3.0.38.20. No fix has been applied in firmware version 3.0.41.4.** This issue is being marked as closed in this firmware version.

1.5.50 List of Fixed Issues in STD-SIP v3.0.40.23

S No.	Bugzilla ID (B), NEC Issue ID (N),	Defect/Enhancement	Defect Description	Remarks

	GM Issue ID , Tracker (T)			
1.	B - 735 T - 24859	DT700: Feature Request to disable upload of call history files	Please look at attached document for detail. T-24859.docx	
2.	B - 722 FR - 903140049 T - 25063	DT700: Call Waiting Tone cannot be switched off for the DT7x0(G) HANDset	Currently the call waiting tone can be controlled for the headset only. It cannot be used to disable the call waiting tone on either the speaker or the handset. There is no configurable to control this. The customer wants to keep the visual indication, but is complaining about the call waiting tone in the HANDset. The proposal is to use the current configuration parameter for the headset also for the HANDset/loudspeaker in order to modify the waiting tone volume through the Menu > User Settings > Headset > Tone Volumes > Call Waiting Volume	
3.	B – 734 FR - 903140056 T - 25163	DT700: Use the TOP LED to indicate missed calls	We got a customer request. I think if the LED could be synched with the missed calls icon. I can also imagine that there will be customers that don't want that feature and only want to use the LED for message waiting, so it needs to be switchable via a setting in the configuration file at least. Incident Description: On the top of the DT 700 Phone is a led, in the SV 8100 this led flashes when there is a missed call. On the IS 3000 this does not work. Is there a way to make this possible	
4.	B - 731	GNU C Library (glibc) has vulnerability that name is GHOST	➢ Reference URL CVE-2015-0235 https://cve.mitre.org/cgi-bin/cvename.cgi?name=CVE-2015-0235	This issue was marked as fixed in release

			NIST : Vulnerability Summary for CVE-2015-0235 http://web.nvd.nist.gov/view/vuln/detail?vulnId=CVE-2015-0235 CERT : Vulnerability Note VU#967332 http://www.kb.cert.org/vuls/id/967332 [Detail Information] Qualys Security Advisory CVE-2015-0235 https://www.qualys.com/research/security-advisories/GHOST-CVE-2015-0235.txt	v3.0.38.20 however this issue was not fixed in this release. This issue is fixed in current release i.e STD-SIP v3.0.40.23.
5.	B-736 FR-001140369 T - 25676	Change the text from "802.1x" to "802.1X"	FR-001140369 reports "802.1x" should be "802.1X".	A related issue T: 25985 “DT700: 802.1X Connecting message not displayed correctly in languages other than English” is not fixed in this release.
6.	B-739 T-23648	DT User Settings menu shows English - Australia as the language selection when phone is configured in the	If you set a DTERM to use UK as the Template in the Admin GUI under localization, this automatically sets the language as "en_uk" The default dt-000000000000-sip.cfg file contains entries for English (Australia), English (UK) and then English (US). Because the phone knows his language value has been set to "en-uk",	

		Admin for UK Localization	he picks the first entry in the list that matches and therefore shows the Language selection as English (Australia). We did send the files for translation but they came back the same for Australia as UK, so we never made a language selection for en_au. I think the easiest fix is to copy the en_uk language files and make an entry for en_au and then change the mapping in the DTERM files to use en_au instead of en_uk. This should solve the issue in the field on an upgrade since all the phones currently have a language value en_uk which will no longer match the English (Australia) selection and the phone menu will show the correct language selected. his problem is exacerbated by the addition of: New Zealand, Hong Kong, Malaysia, Singapore, India, Thailand (collide with UK and Australia on en_UK) Taiwan, Philipines (collide with US on en_US) Chile, Columbia (collide with Mexico on es_MX) Argentina (collides with Spain on es_ES)	
7.	B-738 FR-001150064 T-25776	Rebooting phones after attempt to pick up parked call	They are reporting that phone are rebooting when attempting to pickup Notepad++ search filter d9171d2bb372a8e3|17815222|17815216|2ca54cdaf5ee60d63b2506009fb3bb|1c598d83-6022-47ca-ae2d14efaaf2dae4 will	

			show both the original parked call as well as the call attempting to retrieve it. Seems the he original sphere call id 17815216 is in a bad state before the attempt to unpark it – UnimplementedFeature on line 33998 in Univerge3C1_mgc0_03_05_15_45_Thu_file#1980.log: The SIPp script is used to recreate this problem. This script sends an INVITE to the DT700. Once the DT700 answers by responding with 180 Ringing and 200 OK, the script sends an ACK, waits 4 seconds, and sends the INFO request that causes the crash.	
8.	B-740 T - 24949	DT700 must support SHA2 certificate signatures for Mutual TLS	The industry is deprecating SHA1 certificate signatures and moving to SHA2 (a family of signatures that includes SHA224, SHA256, SHA384 and SHA512). The DTERM must support SHA2 signatures. The attached Wireshark shows a handshake failure between 2.3.33.11 and an 8.5.4.11 3C server using a certificate with a SHA256 signature. Note that the certificate of the CA that issued the server certificate also uses SHA256. Also note the DT730G (3.0.34.20) supports SHA2. The DT700 supports SHA256 for validating the server's certificate, but it does not support SHA256 for its own client certificate for mutual TLS.	

9.	B-741 T - 25303	When the client certificate Private key is password protected (Delimited with Begin and End Encrypted Private key tags), the DT700 enters into an endless loop.	CertPrecheck / 3C Admin creates certificates with "-----BEGIN PRIVATE KEY-----" and "-----END PRIVATE KEY-----" if the private key is NOT password protected. CertPrecheck / 3C Admin creates certificates with "-----BEGIN ENCRYPTED PRIVATE KEY-----" and "-----END ENCRYPTED PRIVATE KEY-----" if the private key is password protected. CertPrecheck generates server certificates whose private key is encoded with PKCS#8. The private key is delimited by "-----BEGIN PRIVATE KEY-----" and "-----END PRIVATE KEY-----". For SIP over TLS, the DT700 only handles certificates encoded with PKCS#1. Keys encoded with PKCS#1 are delimited by "-----BEGIN RSA PRIVATE KEY-----" and "-----END RSA PRIVATE KEY-----". The DT700 will recognize "-----BEGIN PRIVATE KEY-----", "-----BEGIN RSA PRIVATE KEY-----" and "-----BEGIN ENCRYPTED PRIVATE KEY-----" as delimiters for the private key. However, since the DT700 does not have a mechanism to obtain the private key password, it will be unable to decrypt encrypted private keys and therefore will not support certificates with "-----BEGIN ENCRYPTED PRIVATE KEY-----".	Issue T-26006 has been raised where STD-SIP v3.0 as well as STD-SIP v2.3.39.19 reboots continuously. This issue need to be confirmed and is not fixed in current release. i.e 3.0.40.23

1.5.51 List of Fixed Issues in STD-SIP v3.0.38.20

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T)	Defect/Enhancement	Defect Description	Remarks

1.	B-718 FR-001140327 T-25173	DT700: Request for "tertiary" registrar	The DT700 cannot handle NAPTR domain for line.1.primary.address and IP address for line.1.secondary.address of “DNSNAPTR” protocol	
2.	B-723	Outgoing Call Not Initiated when terminal failed over to Secondary server and primary server is not resolved.	This issue has 2 sub-issues related to it. These are as follows: 1. Outgoing call not working in this scenario: Consider the following scenario: A. Ensure that DNS server configured at DHCP server is either not ping-able or DNS server is not running on that host. B. Set protocol for DT730/DT730G terminal to “UDP” in configuration file(either manually or through 3C Admin) C. Set primary registrar to either DOMAIN NAME/FQDN in configuration file(either manually or through 3C Admin) D. Set secondary registrar to valid 3C server “IP ADDRESS” in terminal’s configuration file(manually) E. Start DT730/DT730G terminal. Ensure that DT730/DT730G terminal registers over “UDP” with configured secondary server IP.	

			E. After registration is successful, initiate an outgoing call from DT730/DT730G terminal. ISSUE: The problem is that no outgoing call is initiated from terminal and no INVITE is received by 3C server from terminal. 2. Second is fail-back to primary when primary server is resolved: Consider the following scenario: A. Ensure that DNS server configured at DHCP server and DNS server is ping-able and has DNS SRV entries for SIP over UDP, but services are disabled on the host server. B. Set protocol for DT730/DT730G terminal to “UDP” in configuration file(either manually or through 3C Admin). C. Set primary registrar to either DOMAIN NAME/FQDN in configuration file(either manually or through 3C Admin). D. Set secondary registrar to valid 3C server “IP ADDRESS” in terminal’s configuration file(manually). E. Start DT730/DT730G terminal. Ensure that DT730/DT730G terminal sends a DNS query to resolve primary	

			server but as services are down, it isn't resolved and the terminal registers over "UDP" with configured secondary server IP. F. Now enable DNS service on host server. ISSUE: The problem is that with the DNS server now available, the terminal should query to DNS server to resolve primary registrar address. DNS should resolve the primary registrar address and terminal should use it to fail back to primary registrar from secondary registrar. However, the terminal remains registered with secondary registrar and never fails back to primary.	
3.	B-724	Outgoing INVITE sent to Primary Server when Phone is Registered with Secondary Server in Configuration File	Scenario is following 1) DNS server is up. 2) In terminal's configuration file a) protocol "UDP" is selected , b) "domain name" is provided as "primary address" in configuration file.At DNS server, SRV query for "domain name" resolves to 3 servers where higher priority server "Server	

			1”,”Server 2” are unreachable while lower priority “Server 3” is reachable c) IP address of “Server 4” is given as “secondary address” in configuration file 3) When terminal boots up, then terminal registers with “Server 3” as “Server 1”,”Server 2” are unreachable and “Server 4”’s priority is lower than “Server 3”. 4) Stop 3C services on “Server 3”, Phone fails over to lower priority server “Server 4” and registers with it 5) Initiate outgoing call. Problem starts here, outgoing “INVITE” is sent to “Server 1” instead of “Server 4”. Following is the server configurations 1) win-gb020873dh5.sphere.com/10.112.94.50 is the “Server 3” where 3C server as well as DNS server is running. It is also “Server 1” and “Server 2” although with invalid/incorrect SIP ports.	

			2) 10.112.94.82 is the "Server 4" which is configured as 3C "secondary address" in configuration files 3) 10.112.94.6 is the IP address of the terminal with firmware version STD-SIP v3.0.36.19. This firmware doesn't contain code changes for Bugzilla bug 723.	
4.	725	With DNS server having only TLS SRV configuration, DT730G doesn't failback to primary server from secondary server when service of primary server is restored.	Steps of reproduction: With DNS server running observer "Fail-over" in following way : 1. DNS should have DNS NAPTR & SRV entries configured for SIP over TLS with a primary and a secondary server. 2. DUT should send a DNS NAPTR,SRV,AAA query sequentially to DNS server for identifying the services and hosts offering SIP over TLS as services where primary registrar is DOMAIN NAME and secondary registrar is not set. 3. DNS server responds to the NAPTR request with a list of servers(Server A, Server B) offering TLS services. 4. DUT should register with TLS as SIP transport on Server A as registrar.	

			5. After registration with server A, make an outgoing call as well as an incoming call from DUT. 6. Observer that SIP "INVITE" message over TLS is sent out from DUT and reaches server A and call gets connected with voice path established in each case. 7. After confirmation of calls, stop Sphericall services on Server A. 8. Verify that DUT should fail over to Server B with TLS as SIP transport on registration expiry. 9. Once it registers with Server B, make an outgoing call as well as an incoming call from DUT. 10. Verify that SIP "INVITE" message over TLS is sent out from DUT and reaches server B and call gets connected with voice path established in each case 11. After confirmation of outgoing call, start Sphericall services on Server A. EXPECTED BEHAVIOUR: With Sphericall services on a started, DUT should fail back to Server A with TLS as SIP transport after 5 minutes from the completion of last call.	

			OBSERVED BEHAVIOUR: DUT does not fail back to Server A and remains registered on Server B with TLS as SIP transport.	
5.	B-727	Current Registrar under 'SIP Information' status is not updated when DT730G fails back to higher priority server from a lower priority server	Steps of reproduction: With DNS server running observer "Fail-over" in following way : 1. DNS should have DNS NAPTR & SRV entries configured for SIP with a primary and a secondary server. 2. DUT should send a DNS query sequentially to DNS server for identifying the services and hosts offering SIP services where primary registrar is DOMAIN NAME and secondary registrar is set to an IP address of Server C. 3. DNS server responds to the NAPTR request with a list of servers(Server A, Server B) offering SIP services. 4. DUT should register with Server A as registrar. 5. Stop Sphericall services on Server A. 6. Verify that DUT should fail over to Server B on registration expiry. 7. Stop Sphericall services on Server B. 8. Verify that DUT should fail over to Server C on registration expiry.	

			8. Start Sphericall services on any of the higher priority servers i.e. Server A and/or Server B. 9. Wait for the DUT to fail-back to any of the higher priority servers. 10. View 'Current Registrar' under 'SIP Information' status. (Menu->Status->Current Registrar) EXPECTED BEHAVIOUR: Once the DUT fails back to any of the higher priority servers, Server A or Server B, the 'Current Registrar' should be updated with appropriate value under 'SIP Information' status. OBSERVED BEHAVIOUR: DUT fails back to any of the higher priority servers Server A or Server B, but the 'Current Registrar' is not updated. Its value is set to Server C even when the DUT fails back to a higher priority, Server A or Server B.	
6.	B-728 T-17186	T-17186: DT700 Bulk Unsubscriptions are retransmitted during failback.	Scenario is as follows: - DT700 is configured for EPK with DCL-60 module and subscribes for presence notifications for all 92 keys.	

			- DT700 Registers and Subscribes to primary MGC for presence notifications - Primary MGC is brought down - After some time DT700 Registers and subscribes with Secondary MGC - Primary MGC comes back up again - DT700 UnSubscribes for all the presence notifications with the Secondary MGC. Issue: While performing bulk UnSubscriptions even though Secondary MGC responds with 200 OK for UnSubscribe requests, the DT700 is unable to process some of the 200 OK responses and ends up retransmitting them.	
7.	B-729	DT730G immediately un-registers and crashes after failing back to a higher priority secondary server from a lower priority server	Steps of Reproduction: With DNS server running observer "Fail-over" in following way : 1. DNS should have DNS NAPTR & SRV entries configured for SIP with a primary and a secondary server.	

			2. DUT should send a DNS query sequentially to DNS server for identifying the services and hosts offering SIP services where primary registrar is DOMAIN NAME and secondary registrar is set to an IP address of Server C. 3. DNS server responds to the NAPTR request with a list of servers(Server A, Server B) offering SIP services. 4. Explicitly disable Sphericall Services on Server A. 5. DUT should register with Server B as registrar. 5. Stop Sphericall services on Server B. 8. Verify that DUT should fail over to Server C onregistration expiry. 8. Start Sphericall services on Server B only. 9. Wait for the DUT to fail-back to higher priority server Server B only.	

			10. View 'Current Registrar' under 'SIP Information' status. (Menu->Status->Current Registrar) EXPECTED BEHAVIOUR: Once the DUT fails back to a higher priority Server B, the DUT should successfully register itself with Server B and the 'Current Registrar' should be updated with appropriate value under 'SIP Information' status. OBSERVED BEHAVIOUR: DUT fails back to higher priority Server B, but immediately get un-registered by sending a un-register request after sending a registration request on Server B. The DUT then remains unregistered. Also while accessing the 'Current Registrar' under 'SIP Information' status, the DUT crashes.	
8.	B-730	For tertiary server support, after DNS expiry and DNS server down , terminal doesn't failover to "line1.secondary.server" server	The scenario is as follows: 1. line.1.primary.address is configured with an SRV record. line.1.secondary.address is configured with an IP address. 2. The DT700 is registered with one of the registrars specified in the SRV	

			record line.1.primary.address is configured with. 3. Stop DNS service. The registrar the DT700 is registered with is stillreachable. 4. Wait until the DNS records expire from the DT700's internal DNS cache due to the DNS Time To Live (TTL). Note: the DNS TTL is typically one hour. 5. DT700 cannot register with the registrar since the M5T SIP stack cannot resolve the SRV record or host names. 6. DT700 does not failover to the IP address specified by line.1.secondary.address.	
9.	B-726 FR-001140363	FR-001140363: removed the "+T" delay at the end of the four digit dialing rule in the country config file	The text of the FR is as follows: The default digitmap rule in dt-us.cfg and dt-000000000000-sip.cfg is missing arule for 11-digit dialing. I	Configuration files of dt-us.cfg and dt-000000000000-sip.cfg are updated for this issue. In dt-000000000000-sip.cfg file , value of parameter "line.1.digitmap" is changed

			always have customers and students add that in, i.e. 9,1[2-9]xx[2-9]xxxxxx Also, in my unofficial survey of American techs in my 3C classes, the bulk of new 3C installs and many migrations to 3C use 4-digit extensions. So they were in agreement with the rule in dt-000000000000-sip.cfg file which was [1-8]xxx. However, the rule in dt-us.cfg which over rides dt-000000000000-sip.cfg has an open ended timer after the 4-digit extension rule, i.e., [1-8]xxx+T. So I get lots of questions from the field and in training class why there is a delay in dialing another station. I actually like my students asking this question since I get to discuss a great reason for creating a dt-customer.cfg file and how configuration files work in general. However, for the field who may find the default digitmap good enough, we can avoid support calls by simply removing the "+T" in the dt-us.cfg file and adding an 11-	from "[1-8]xxx|9,[2-9]xx[2-9]xxxxxx|9,011x+T|*xx+T|911|9,911" to "[1-8]xxx|9,1[2-9]xx[2-9]xxxxxx|9,[2-9]xx[2-9]xxxxxx|9,011x+T|*xx+T|911|9,911" In dt-us.cfg file, value of parameter "line.1.digitmap" is changed from [1-8]XXX+T|9,[2-9]XX[2-9]XXXXXX|9,011X+T|*XX+T|911|9,911" to "[1-8]XXX|9,1[2-9]xx[2-9]xxxxxx|9,[2-9]XX[2-9]XXXXXX|9,011X+T|*XX+T|911|9,911".

			digit rule to both dt-us.cfg and dt-000000000000-sip.cfg files.	
~~10.~~	~~B-731~~	~~GNU C Library (glibc) has vulnerability that name is GHOST~~	~~➢ Reference URL~~ ~~CVE-2015-0235~~ ~~https://cve.mitre.org/cgi-bin/cvename.cgi?name=CVE-2015-0235~~ ~~NIST : Vulnerability Summary for CVE-2015-0235~~ ~~http://web.nvd.nist.gov/view/vuln/detail?vulnId=CVE-2015-0235~~ ~~CERT : Vulnerability Note VU#967332~~ ~~http://www.kb.cert.org/vuls/id/967332~~ ~~[Detail Information] Qualys Security Advisory CVE-2015-0235~~ ~~https://www.qualys.com/research/security-advisories/GHOST-CVE-2015-0235.txt~~	~~A patch of ghost vulnerability is applied in this firmware.~~
11.	B-732	Vulnerability of NTP	➢ Reference CVE https://cve.mitre.org/cgi-bin/cvename.cgi?name=CVE-2014-9293	SNTP is upgraded from 4.2.4p0 to 4.2.8 in this firmware.

			https://cve.mitre.org/cgi-bin/cvename.cgi?name=CVE-2014-9294 https://cve.mitre.org/cgi-bin/cvename.cgi?name=CVE-2014-9295 https://cve.mitre.org/cgi-bin/cvename.cgi?name=CVE-2014-9296	

1.5.52 List of Fixed Issues in STD-SIP v3.0.36.19

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T)	Defect/Enhancement	Defect Description	Remarks
1.	B-720 T-25035	DT700 Registers with two UCMs at Bakery Craft	The MGC logs at Bakery Craft show at least 4 DT700s registering with both the primary and secondaryUCMs.	
2.	B-721 T-18314	When a TFTP failure occurs with TFTP error code 0, the DT700 does not attempt any subsequent TFTP downloads.	When the DT700 attempts a TFTP download and a TFTP failure occurs with TFTP error code 0 (Undefined), the DT700does not attempt any subsequent TFTP downloads. Since the TFTP connection uses UDP (connectionless), every	This issue occurred for TFTP error code 2 and 6 on STD-SIP 3.0. This issue doesn’t occur for TFTP error code 0.This difference is because STD-SIP v3.0 uses CURL library for TFTP purpose.

			TFTPtransaction should be attempted regardless of the previous result.	
3.	B-719 FR-903140055 T-25135	DT700 disconnects seems caused by session guarding	This is breakage caused by the repair of T:23631. It appears that when you have session guarding active (in this case UPDATE is used because the DT700 supports it) that the DT700 clearsat 2/3 of the session guarding time (which is set to 90 seconds).	
4.	B-657 T-22540	STD-SIP 2.3.23.26 is sending 487(very rarely) to Refer during Race	DT700 transferee sends 487 when the REFER and BYE cross each other. This case (as below) is experienced,when the transferee hangs up the call at the same time as the transferor completes the Blind transfer.	
5.	B-711 FR-903140009 T-23014	DT700:Does not respond with an ACK to a repeated 200 OK	Based on RFC section 13.2.2.4, the UAC core must generate an ACK request received for each 2xx received fromthe transaction layer, but it does not. The reason is that the transaction layer (within Broadcom's M5T stack) dropsthis packet instead of passing it to the core.	
6.	B-712 FR-903140010 T-23015	DT700:Does not respond to a repeated 200 OK after reINVITE	Based on RFC section 13.2.2.4, the UAC core must generate an ACK request received for each 2xx received fromthe transaction layer, but it does not. The reason is that the transaction layer (within Broadcom's M5T stack) dropsthis packet instead of passing it to the core.	

7.	B-714 T-23632	DT700: Doesn't always respect session expiry timeout	When acting as a UAS, the DT700 should use the session refresh interval specified in the Session Expires header of the INVITE. The DT700 only seems to do this if the UAC also includes a Min-SEheader. The Min-SE header is optional. If it is not present, the DT700 should use the session refresh interval specified in the Session-Expires header if it meets the default minimum session interval	
8.	B-715 T-23569	DT700 Phone crash Task: "CMGR" Injection Point: excArchLib.c:1050	Phone Crashes. This is same issue as T-23643 and T-23725.	This issue doesn't get reproduced on STD-SIP v3.0, however fix of this issue is merged. Please refer attached document. T-23569, T-23643 & T-23725 Defect Analy
9.	B-716 T-23643	NEC DT700 Phone lock up / Crash Task: "RegHelper" Injection Point: excArchLib.c:1050	Phone Crashes. This is same issue as T-23569and T-23725.	This issue doesn't get reproduced on STD-SIP v3.0, however fix of this issue is merged. Please refer attached document. T-23569, T-23643 & T-23725 Defect Analy

10.	B-717 T-23725	DT700 Phone: 0x82885f60 mmuVaddrShift+0xe66b3c: 0x801edc50 ()	Phone Crashes. This is same issue as T-23569 and T-23569.	This issue doesn’t get reproduced on STD-SIP v3.0, however fix of this issue is merged. Please refer attached document. T-23569, T-23643 & T-23725 Defect Analy
11.	B-703 T-15457	Sphericall doesn't parse underscore "_" in the request URI header of the SIP Register.	A customer has created a DNS SRV record under the _sites container in their Microsoft DNS Server. This results in the following (using an example from my lab): 1.The DT700 AOR is 671@moore.lake.illinois.spherecom.com 2.The DT700 registrar is _sites.moore.lake.illinois.spherecom.com 3. The DT700 transport is DNS NAPTR 4. The DT700 performs an SRV query for _sip._udp._sites.moore.lake.illinois.spherecom.com, which resolves to an MGC	This issue is already fixed in v3.0.35.12. In this version, small piece of unused and redundant code is removed.

			5. The DT700 REGISTER Request-URI is _sites.moore.lake.illinois.spherecom.com, the To and From headers contain the AOR, 671@moore.lake.illinois.spherecom.com 6. The MGC cannot parse the Request-URI because it contains "_" 7. MGC responds "400 Missing Request Line" to the REGISTER request. The MGC's behavior in step 6 is correct per RFC 3261 section 25. The Request-URI can only containalphanumeric characters and "-"	

1.5.53 List of FixedIssues in STD-SIP v3.0.35.12

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T)	Defect/Enhancement	Defect Description	Remarks
1.	B-706 T-24768	DT700: Deprecate SSL V3.0	SSL 3.0 contains a number of weaknesses including POODLE (CVE-	

			2014-3566). SSL V3.0 is now considered insecure. Support for SSL V3.0 should be removed from the DT700.	
2.	B-705 T-24747	DT700: Upgrade OpenSSL libraries to version 1.0.1j	Security flaws have been found in OpenSSL 1.0.1i (see attached security advisory). The DTERM should be upgraded to OpenSSL 1.0.1j.	
3.	B-708 FR-903140042 T-24506	DT700: Make the SUBSCRIBE timer for speed dial keys adjustable	At the courts justice (10.000 SIP ext) we see that the server is spending lot of time on speed dial keys refreshing of the SUBSCRIBER. Please make it programmable via the config file (Default timer=3600). Max value 24*3600=86400	
4.	B-709 FR-001140257 T-24371	Remove line.1.maxcalls from dt-000000000000-phone-sip.cfg	line.1.maxcalls is set to 4 in dt-000000000000-phone-sip.cfg. Customers who want to change maxcalls cannot do so since dt-<MAC>-phone-sip.cfg appears first in the list of config files (and therefore has higher priority than dt-customer.cfg) and the 3C Admin does not write maxcalls. The entry for line.1.maxcalls must be removed from or set to "" in dt-000000000000-phone-sip.cfg. The default value should remain at 4 in the code.	
5.	B-707 T-24340	The UK is requesting a change be made to the DT-uk digit map to fit their new dialing needs	Based on the feedback from their resellers and experience from last	

			sales, they feel that these setting (dt-uk.cfg) fits their model best.	
6.	B-710 T-23583	User is unable to Barge from DT700 if Address Group contains special characters.	User is unable to Barge from DT700 if Address Group contains special characters. 1) Add Address Group with name ->Barge----,-----,??????,"Monitor" 2) Initiate Barge-Monitor from DT phone. Issue : User gets an error "Parse Error Line :1 not well-formed (invalid token)" though user is able to Barge the call from Polycom phone without any issue.	No code changes /configuration file changes has been done for this defect .Only verification of this bug has been done on DT730G with 3Cv v8.5.4.11.
7.	B-704 T-15530	DT700 does not handle RFC4028 properly in case T.38 is present	DT700 ends the call with a BYE after it sends re-Invite for session timer refresh when T.38 is present	
8.	B-703 T-15457	Sphericall doesn't parse underscore "_" in the request URI header of the SIP Register	A customer has created a DNS SRV record under the _sites container in their Microsoft DNS Server. This results in the following (using an example from my lab): 1.The DT700 AOR is 671@moore.lake.illinois.spherecom.com	

			2.The DT700 registrar is _sites.moore.lake.illinois.spherecom.com 3. The DT700 transport is DNS NAPTR 4. The DT700 performs an SRV query for _sip._udp._sites.moore.lake.illinois.spherecom.com, which resolves to an MGC 5. The DT700 REGISTER Request-URI is _sites.moore.lake.illinois.spherecom.com, the To and From headers contain the AOR, 671@moore.lake.illinois.spherecom.com 6. The MGC cannot parse the Request-URI because it contains "_" 7. MGC responds "400 Missing Request Line" to the REGISTER request. The MGC's behavior in step 6 is correct per RFC 3261 section 25. The Request-URI can only containalphanumeric characters and "-"	

1.5.54 List of Fixed/Irreproducible Issues in STD-SIP v3.0.34.20

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T)	Defect/Enhancement	Defect Description	Remarks
1.	B-91 N-3 T- 23623	DT730CG When the digit key is pressed for a first time, the phone offers a tone which also has a slight noise	DT730CG - When the digit key is pressed for a first time, the phone offers a tone which also has a slight noise.The issue was seen by Narata-san in his desk phone in Irving office. I did a quick check in standard sip VxWorksverison and found no noise.	**This issue is not fixed in this release**.A fix for this bug was applied in Release 3.0.34.15, however this issue was not completely fixed therefore fix applied in release 3.0.34.15 is reverted back.
2.	**This release contains all the bug fixes which are present in release v3.0.34.15 except for T-23623. Please refer 1.5.55 for the bugs.**			

1.5.55 List of Fixed/Irreproducible Issues in STD-SIP v3.0.34.15

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T)	Defect/Enhancement	Defect Description	Remarks
1.	B-698 T-24075	DT700: Upgrade OpenSSL libraries to version 1.0.1i	Security flaws have been found in OpenSSL 1.0.1h. The DT700 should be upgraded to OpenSSL 1.0.1i	STD-SIP v3.0.34.15 has upgraded OpenSSL version to 1.0.1i

2.	B-697 T-23631 [Note: This issue was fixed completely in 3.0.36.19.]	DT700 doesn’t fully respect sessiontimer.mode	Setting sessiontimer.mode to 0 should disable Session Timers in the DT700. The DT700 should notadd the timer tag to the Supported or Require headers, and it should not react to any Session-Expiresheader in incoming requests. When the DT700 receives an INVITE with Session-Expires: 14400 from NECAM's Ingate, it respondswith a 200 OK that includes a Session-Expires: 1800;refresher=uas header, even thoughsessiontimer.mode=0. This is incorrect. The DT700 should ignore the Session-Expires header fromtheIngate	This issue was reproducible in STD-SIP v3.0. With v3.0.34.15 release,v3.0behavior will be same as v2.3.33.11 firmware.
3.	B-700 T-23360	DT700 should check certificate SAN before CN for SIP	The following statement appears in RFC 2818: 3. Endpoint Identification 3.1. Server Identity ... If a subjectAltName extension of type dNSName is present, that MUST be used as the identity. Otherwise, the (most specific) Common Name field in the Subject field of the certificate MUST be used. Although the use of	This issue was reproducible in STD-SIP v3.0. STD-SIP v3.0 certificate checking logic was same as v2.3. With v3.0.34.15 release, v3.0 certificate checking logic will be same as v2.3.33.11 firmware.

			the Common Name is existing practice, it is deprecated and Certification Authorities are encouraged to use the dNSName instead. Matching is performed using the matching rules specified by [RFC2459]. If more than one identity of a given type is present in the certificate (e.g., more than one dNSName name, a match in any one of the set is considered acceptab le.) Names may contain the wildcard character * which is considered to match any single domain name component or component fragment. E.g., *.a.com matches foo.a.com but not bar.foo.a.com. f*.com matches foo.com but not bar.com. This implies we should look at the SAN first, if available, then the CN. We spec'd the DT700 to look at the CN first	
4.	B-696 FR-903130049 T-21183	DT700 does not ring when an incoming call arrives and 1 digit pressed	When a user (cleaning lady) incidently touches the dialpad, lets say presses the `1` key but is not going off-hook. The phone stays in this way. Incoming call can`t be heard only seen as an LED blinking. The phone user needs to and this onhookprepaired dialing to become	This issue was reproducible in STD-SIP v3.0. With v3.0.34.15 release, v3.0 behavior will be same as v2.3.33.11 firmware.

			reachable. The incoming call is indicated by the LED and can be answered in 2 ways: - press the "speaker" button once (terminal starts ringing) and on the second press the call is answered. - press the "answer" button and the call is directly answered. In both cases the "dialled" digit is gone when the call is ended. On the Polycom the terminal starts ringing. The "dialled" digit (e.g. 1) remains in the display, also when the call is ended.	
5.	B-701 T-21622	DT700: TLS connection fails if Server accepts TLS 1.2 version	When DT700 offers TLS 1.0 and TLS 1.2 to the Server , DT700 closes the TCP connection on receiving Server Hello willing to connect using TLS 1.2 version	This issue is **not reproducibleinSTD-SIP 3.0**because, for SIP over TLS, STD-SIP 3.0 offered TLS v1.0 only. STD-SIP v3.0 doesn't support HTTPS. Modification has been done in v3.0 for offering TLSv1.0 as well as TLS v1.2 for SIP over TLS. With v3.0.34.15 release, v3.0 behavior will be same as v2.3.33.11 firmware's SIPoverTLS behavior.
6.	B-699 FR-903140012 T-23028	Received exception log files for a DT710		This issue was **not reproducedin STD-SIP3.0**however code changes has been done for preventing its possible occurrence. This

				issue occurs because of assert failure which was present in STD-SIP 3.0 also. Therefore there are chances that this issue might get reproduced. To prevent this issue from occurring in v3.0.34.15, assert statement has been replaced with warning and error logs with proper boundary checks
7.	T-23503	Feature Request for DT700 to ignore HTTPS Certificates when fresh from the box	The DT700 ships with root certificates from the NECS CA and NECU CA built into the firmware. The DT700 will not trust the HTTPS server if it is not using certificates issued by the NECS CA or NECU CA. Therefore, for customers using certificates issued by CAs other than NECS or NECU, the root CA certificates must be downloaded to the DT700 by FTP or TFTP. They cannot be downloaded by HTTPS. This is inconvenient for customers, who don't want to instantiate an FTP or TFTP server just to download root CA certificates.	This issue is **not reproducible in STD-SIP 3.0** because STD-SIP 3.0 doesn't support HTTPS. No fix has been applied in this release for this bug except for addition of configuration parameter "security.ssl.server.validate" in"dt-000000000000-sip.cfg"file. This has been done for keeping common configuration file of v2.3 and v3.0 in sync.
8.	B-702 FR-903140033 T-23755	FTP fails caused by CWD / command	Customer can't set the proper default language on the phone. Phone starts up for the first time , CDW / command fails in FTP. Second boot works fine.	This issue is **not reproducible in STD-SIP 3.0** because STD-SIP usescurl library for FTP.This issue occurred in STD-SIP v2.3 because of v2.3 specific code. These code changes are not

				present in v3.0therefore this issue doesn't occur in v3.0. No fix has been applied in this release for this bug.
9.	B-686 T-18301	DT700 does not attempt to download the upper case version of the data configuration file	When downloading a data configuration file, if the download fails, the DT700 is supposed to attempt to download the lower case version followed by the upper case version. The upper case version is no longer attempted.	This issue was reproducible in STD-SIP v3.0. With v3.0.34.15 release, v3.0 behavior will be same as v2.3 firmware in which this issue was fixed.
10.	B-693 FR-903130008 T-18917	Restart DT710 when median is in SAS	The scenario is a sipserver with two median 1k (ip 192.168.0.203 and 192.168.0.204) with SAS enabled and DT710 v2.2.6.28 ext 3640 registered in the sipserver and as a second registration the median1 ip 192.168.0.203 Now, the sipserver is stopped so DT710 get the registration from median1 .203.I setup an outgoing call from ext 3640 to E1 (Aurora tester) in median2 192.168.0.204 (dialed number 2345). The call is sent to median1 but this one has a forward on busy to median2.The E1 in median2 goes to alert and I answer the call but the DT710 still send ring back tone and the call is not connected.I release the call in the DT710 and the terminal stop ringback and stop the call. After	This issue was reproducible in STD-SIP v3.0. With v3.0.34.15 release, v3.0 behavior will be same as v2.3 firmware in which this issue was fixed.

			5 to 10 seconds the DT710 restart with any external action.I understand the call could fail but the terminal must remain working an never do a restart. For me there are two problems: Call is not connected and terminal restarted.I've activated syslog in the terminal and also wireshark, config files on DT710 and median	
11.	B-689 T- 19001	Hot-desking enabled station shows station name on screen instead of extension name	For the DT700 terminals for which hot desking is enabled shows Station name on display instead of exension name. This is a regression defect from 8.1.3.6 Steps 1. Select a DT700 terminal which has no hot desking enabled. 2. Verify display name is name of extension. 3. Enable hot desking on the terminal. Observed Result: Extension name gets changed to station name.	This issue was reproducible in STD-SIP v3.0. With v3.0.34.15 release, v3.0 behavior will be same as v2.3 firmware in which this issue was fixed.

			Expected Result: Irrespective of hot desking status display name should be always maintained as extension. This is regression defect from 8.1.3.6	
12.	B-685 T-19104	DT700 directory file upload	If user has modifies some Speed dial keys then phone(DT700 V2.3) takes almost 1 minute before it uploads that to FTP server. By that time if using 3C admin some one restarts that phone then phone never uploads that file.On the other hand if user restarts phone using phone menu then it forces phone to upload that file immediately. On the same line if a NOTIFY withevent check-sync is received then phone needs to upload file before booting.	This issue was reproducible in STD-SIP v3.0. With v3.0.34.15 release, v3.0 behavior will be same as v2.3 firmware in which this issue was fixed.
13.	B- 687 FR-903120050 T-16463	Off hook ringing situation counter-intuitive	MRDB-2525 Users (Switzerland) are reporting the following unexpected scenario when using DT700`s with a SIP server:Call in progress on DT710 & someone camps-on (call waiting) the MW light flashes & you get Call Waiting tone. If the person you are in a conversation with clears first, your phone immediately rings with the handset off hook. The user is now confused with this unexpected behavior. The MW & `my line` light flashes but they can`t press this key to	This issue is **not reproduciblein STD-SIP 3.0** because v2.3 fix for this bug was already present in v3.0. No fix has been applied in this release for this bug.

			pick up. The only way to get the call is to hang-up, which is counter-intuitive with a ringing phone or press the answer key (non flashing and unlit therefore anonymous). This is causing confusion with users and so the customer has requested to have an `answer` soft key in the screen beside the reject & new call and the ability to pick up from the my line or a flashing answer key (or both). This would be quite useful in an idle state too.	
14.	B- 688 T-18122	DT700 Conference Join Fails when one call is already held.	Scenario 1. A calls B, B answers the call 2. B places A on Hold 3. B calls C, C answers the call 4. B presses Conference key and initiates conference with D 5. D answers the call 6. B presses the conference key again to complete conference between B-C-D Step 6 Fails due to the following error:	This issue is **not reproducible in STD-SIP 3.0** because v2.3 fix for this bug was already present in v3.0. No fix has been applied in this release for this bug.

			[CMGR] (CC) ERROR: JoinStreams, eptTopoloy failed to start, cid primary=22, consult=23, cnxId primary=1, consult=2, streamId primary=1, consult=3	
15.	B- 695 T-19075	DT700 registered over UDP fails to pickup call if MCG uses TCP for INVITE due to large payload	1. Register 3 DT700 terminals on UDP transport. 2. Ext C has EBLF for Ext B. 3. A calls B. 4. While B is ringing Ext C hits EBLF for picking up call. Observed Behavior: 1. Extn C gets and inbound INVITE on TCP transport from MGC and phone never responds and pickup fails 2. Extn C display shows "anonymous". 2. The original A to B calls also gets dropped.	This issue is **not reproduciblein STD-SIP 3.0** as it closes SIP’s TCP connection before rebooting**.** No fix has been applied in this release for this bug. Please refer attached document for more details. T19075 Analysis.docx
16.	B-91 N-3 T- 23623	DT730CG When the digit key is pressed for a first time, the phone offers a tone which also has a slight noise	DT730CG - When the digit key is pressed for a first time, the phone offers a tone which also has a slight noise.The issue was seen by Narata-san in his desk phone in Irving office. I did a quick check in standard sip VxWorksverison and found no noise.	This is v3.0 specific bug which doesn’t occur in v2.3. This issue was fixed earlier while fixing “N-3” but a better fix has been applied for “T-23623”.

17.	B-691	Syslog to be added to detect HWThread lock	If Hwthread lock will be detected an error syslog "HWThread Lock Detected" will be generated.	FR-903140002 was not reproduced however it was closed because it occurred rarely and only on few terminals. In this enhancement, a syslog has been added in the code which will indicate whether HWTHREAD is locked or not which is assumed to trigger FR-903140002.

1.5.56 List of enhancement in STD-SIP v3.0.32.8

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T)	Defect/Enhancement	Defect/Enhancement Description	Remarks
1.		Provide Localization Support for three Latin American Countries- Argentina,Chile and Colombia.	Please refer attached document for details of Localization for Latin America. Latin_America_Localization.xlsx	
2.		Date format in "dt-mexico.cfg" is changed from value "1" to value "2"	Please refer attached configuration file of Mexico.	

			dt-mexico.cfg	

1.5.57 List of Fixed Issues in STD-SIP v3.0.31.9

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T)	Defect/Enhancement	Defect Description	Remarks
1.	B-680 T:23247	DT700 should accept encrypted Digest Authentication Password	The DT700 can accept an encrypted 802.1x password (security.8021x.password). The same encryption should be applied to the digest authentication password (line.1.authentication.password). Notes: security.8021x.password may be encrypted or in clear text. For backward compatibility, line.1.authentication.password should also accept encrypted or clear text forms.The corresponding change must be made in the 3C Admin.	
2.	B-681 T:23414	DTERM should encrypt Digest Authentication Password before uploading	If a SIP Digest Authentication Password is entered via the phoneUser Interface, it appears in clear text when the phone uploads the dt-<MAC>-local.cfg. Related	

			defect T:23247 requests that the the phone accept an encrypted password in the config files. The main reasons for this is so the password is not left in clear text on server disk and so that it does not appear in clear text in unsecure file transfers (e.g. TFTP and FTP). The password should also be encrypted when it is uploaded in the dt-<MAC>-local.cfg.	
3.	B-684 T:23639	DT700 should respect the registration interval in the 200 OK	A SIP registrar sends a 200 OK response with Expires:90 to the DT700's REGISTER request. This instructs the DT700 to adjust its registration refresh interval to 90 seconds. However, the DT700 will only go as low as 180 seconds (minimum value of sip.reg.expiry). Section 10.3 of RFC 3261 states "The registrar MAY choose an expiration less than the requested expiration interval." Therefore, the DT700 should respect the requested 90 seconds.	

1.5.58 List of Fixed Issues in STD-SIP v 3.0.30.30

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID , Tracker (T)	Defect/Enhancement	Defect Description	Remarks
1.	B-682 T-23562	Upgrade OpenSSL libraries to version 1.0.1h	Security flaws have been found in OpenSSL 1.0.1g. The DT700 should be upgraded to OpenSSL1.0.1h.	
2.	B-683 T-23531	DT700 Should respect send-only in ACK	Steps for Reproduction : 1. Configure a DT700 as a 3C group paging destination (we can provide specific instructions for that) 2. Configure the 3C MediaServer to use late media (via a setting in the 3C Admin) 3. Make a group page call to the DT700 4. With Wireshark: a. Observe 3C's INVITE does not contain SDP (Use 3C v8.5.3.11_SP1 for this). b. Observe the DT700 responds with sendrecv media in the 200 OK c. Observe 3C's ACK changes the media to 'sendonly'.	DT700 should not send any RTP packets after it receives the ACK.

			5. DT700 sends RTP packets after it receives the ACK.	

1.5.59 List of Fixed Issues in STD-SIPv3.0.30.11

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID	Defect/Enhancement	Defect Description	Remarks
1.	B-669 Issue ID-136	Incorrect display on DT730CG	Steps for Reproduction : 1. Enter the missed calls list by selecting Menu –> Call History –> Missed Calls. 2. Exit the missed calls list by pressing Exit three times in quick succession. 3. Immediately press the Feature key. 4. Verify that 3C's XML contents are displayed. 5. Return to idle screen by pressing Exit. 6. DT730CG displays incorrect screen.	Fix has been confirmed by NECi-san.

1.5.60 List of Fixed Issues in STD-SIPv3.0.29.8

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID	Defect	Defect Description	Remarks

1.	B-667 FR-927140001	DT730CG Reboots after specific key sequence	Steps for Reproduction 1. Make a copy of the history file from the FTP server 2. Delete all calls at the phone 3. Allow the phone to upload the history file to the FTP server 4. Reboot the phone 5. Place a call to the phone but don't answer 6. Clear the missed calls, exit the missed calls menu and press the Feature key 7. Phone does NOT crash 8. Reset the phone to factory defaults 9. While the phone is rebooting, copy the attached history file to the FTP  dt-0060B98BC508-history.cfg server 10. After the phone reboots, observe that it has downloaded the attached history file 11. Clear the missed calls, exit the missed calls menu and press the Feature key 12. Phone crashes	
2.	B-677 FR-903140020	Company directory has numbers (incorrect if accessed via Directory key)	BCT directory access via the on the DT700 3.0 is shown incorrect if I access the company directory via 1) The directory key (little phone book).	

			2) The Directory. 3) Company directory. 4) Enter a part of a name in the search screen. The above works fine if I access the company directory via menu, Home.	

1.5.61 List of Fixed Issues in STD-SIP v3.0.28.24

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID	Defect	Defect Description	Remarks
1.	B-678	Heartbleed: Serious OpenSSL zero day vulnerability revealed (Upgrade OpenSSLlibrary to version 1.0.1g)	FW v3.0.23.1 – v3.0.28.4 has used OpenSSL library 1.0.1e that had security issue as described in below link http://www.openssl.org/news/secadv_20140407.txt To fix this issue, OpenSSL library isupgraded to openssl 1.0.1g	
2.	B-679	Close SSH port 22	NECU asked NECi to close SSH Port 22 to reduce security risk.Firmware v3.0 closes this port from this version(3.0.28.24).	
3.	B-676 FR-903140024	RFC2833 not repeated properly (only Last once)	Whenever an RFC2833 digits is sent this results in several RFC2833 "messages" for the same digit. The	

			last digit is marked explicitly with a last digit indication. This last frame is only sent once. This is incorrect as RFC2833 digits are sent using UDP, therefore the last message with the last indication also needs to be repeated a couple of times. In release 2.3 this is done 3 times, 3.0 only sends the last frame once which is incorrect. Refer to the attached trace.	
4.	N-136 B-669	Incorrect display on DT730CG	[Steps] 1. Enter the missed calls list by selecting Menu – Call History – Missed Calls. 2. Exit the missed calls list by pressing Exit three times in quick succession. 3. Immediately press the Feature key. 3C's XML content is displayed. 4. Return to idle screen by pressing Exit. Repeats steps from No.1 to No.4. DT730CG displays incorrect screen.	Two screens are displayed incorrectly 1. Home Window where all the content of the screen disappear below the date and time line. 2. ScreenSaver appears while accessing "Menu" similar to Call history screens Both the screens are fixed in this release.

1.5.62 List of Fixed Issues in STD-SIP v3.0.28.4

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID	Enhancement	Enhancement Description	Remarks

1.	None	Localization For Asia	This firmware supports localization for Asia.	This firmware contains localization for Asia along with all the bug fixes done in v3.0.27.18.

1.5.63 List of Fixed Issues in STD-SIP v3.0.27.19

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID	Defect	Defect Description	Remarks
1.	None	Localization For Asia – Change Malaysia's Congestion tone cadence from "500 MS ON 250 MS OFF" to "500 MS ON 500 MS OFF"	Cadence of Malaysia's congestion tone is changed from "500 ms-on, 250 ms-off" to "500 ms-on, 500 ms-off" in this firmware The following is the reason that was informed from NEC Corporation of Malaysia on 14th March. [Reason] If follow Malaysia MCMC PABX Type Approval specification congestion tone spec as below, 1) 0.5 ON, 0.5 OFF Or 2) 0.25 ON, 0.25 OFF. Hence 'Congestion tone - I', I guess we don't acceptable here.	This firmware supports localization for Asia similar to firmware version 3.0.27.5 except for the Malaysia's congestion tone. Bug fixes done in release v3.0.27.18 is not available in this release

1.5.64 List of Fixed Issues in STD-SIP v3.0.27.18

- This release doesn't support "Localization For Asia" support which was present in v3.0.27.5
- Following is the explanation of bug number in below table
  - The number in ( ) means FW version that bug was discovered.
  - V2.3's bug number is FR-xxxxxxxxx (v2.3) or T:xxxxx (v2.3).
  - V3.0's bug number is FR-xxxxxxxxx (v3.0) or T:xxxxx (v3.0).
  - If there isn't ( ), this bug is v3.0.

<table>
<tr><th>S No.</th><th>Bugzilla ID (B),
NEC Issue ID (N),
GM Issue ID ,
Tracker Issue ID(T)</th><th>Defect</th><th>Defect Description</th><th>Remarks</th></tr>
<tr><td>1.</td><td>B-674
FR-903130054 (v2.3)
T: 21414 (v2.3)</td><td>Config download via HTTPS on DT700 terminals is not secure.</td><td>Incident Description: When someone opens Internet explorer on his desktop and browse to HTTPS:{servername} he sees and downloads all config files from the HTTPS server after inserting the default credentials. Changing these credentials is not useful because they are sent to the DT700 by DHCP responce which is plain text. The default credential are shown in the administrator guide of the DT700 . This guide is available on a ftp-server wich belongs to NEC. The adres is ftp://ftp.neci.co.uk/UNIVERGE%203C%20and%20Collaboration/Manuals/UNIVERGE%203C%20DT700%20Manuals/. When</td><td>HTTPS is unsupported in STD-SIP 3.0.STD-SIP 3.0 has implemented Mutual TLS for SIP. When Mutual TLS for SIP is enabled, DT700/730G FW that don't support Mutual TLS for SIP fail the connection.
To connect, DT730G FW supports Mutual for SIP.
Combination of 3C and DT700/730G FW<table><tr><td></td><td>New 3C has Mutual TLS for SIP</td><td>Old 3C doesn't have Mutual TLS for SIP</td></tr><tr><td>DT700 and DT730G FW support Mutual TLS for SIP.
(From this version)</td><td>Mutual TLS for SIP</td><td>Only TLS for SIP</td></tr><tr><td>DT700 and DT730G FW don't support Mutual TLS for SIP.
(Old version)</td><td><u>Connection fails</u> if Mutual TLS for SIP is enabled on 3C</td><td>Only TLS for SIP</td></tr></table></td></tr>
</table>

			the settings of IIS for the SSL certificate is set to `require` for desktopcliënts is it impossible to see or download the config- files with internet explorer but the DT700 terminals can`t download the files either. Is it possible to change this in the dt700 terminals?	
2.	B-665 T:22343(v2.3)	Missed call count does not get cleared on DT phones after you view the missed calls from menu--> call history-->missed calls	This was seen in-house (Andrew saw this too on his phone) If you have 1 or more missed call it will show on the display of the DT phone (I have Desi 8 key) once you go to Menu --> Call History --> Missed Calls and view them and you exit out the missed call message (count) should go away from Phone display.	
3.	B-663 T:22118(v2.3)	Request for secondary DNS entry on DT700 phone.	A new configuration screen that allows entry of the second DNS server address will be added to the DT700.	
4.	B-666 FR-903130072(v2.3) T:22342 (v2.3)	DT700 phones generate high load on SIP@Net server (DDOS)	DT700 phones generate high load on SIP@Net server (DDOS) caused by subscribes. We already advised to move to the latest DT700 release and use UDP. That's what has been done. Now we are analyzing the issue.	
5.	B-670 T:21678 (v2.3)	FTP time out	Assign 20 phones that need to download FW,the FTP time out happens. Put some HUB (20 ports with 10MB capacity).	

6.	B-672 FR-903130079(v2.3) T: 22384(v2.3)	Exception log present on the TFTP server	Exception log present on the TFTP server	
7.	B-673 FR-001140038(v2.3) T:22814(v2.3)	DT700: Phone crash Task: "CMGR"	Two phones rebooted by themselves. See their dt-mac.log files. The two dt-mac.log files showed the two phones crashed. d-0060B98FA0B4.log shows: Assertion failed: status_ == BOS_STATUS_OK, file Q:/phonex/mcu/rtpt/rtpt.c, line 502 dt-0060B98FA10C.log shows: Assertion failed: status_ == BOS_STATUS_OK, file Q:/phonex/mcu/rtpt/rtpt.c, line 502	
8.	B-675 T:22914(v2.3)	Digit map length must be extended	Digit map of Germany as specified in "dt-germany.cfg" file is [1-8]XXX+T|118XX+T|[1-9]X+T|10+T|070[01]XXXXXXXX|090[1-5]XXXXXXX|090[6-9]XXXXX+T|0[5-9][1-9]XXXXX+T|0[568]0[01]XXXXXXX|0[5-8]0[2-9]XXXXX+T|0[2-4]XXXXX+T|014XXXXX+T|013[012345689]XXXXX+T|0137XXXX|019[1-9]+T|0190XXXXX|018[1-9]XXXXX+T|0180XXXXXXX|01[67]XXXXXXXX|01[25]XXXXXXXXX|011XX+T|00X+T|010XX+T|11[02]. The maximum length of "line.1.digitmap" parameter is	

			255 , however above map's length is greater than 255. As a result of this length, if line.1.digitmap.impossiblematch is set to "0" then the digits dialled as per pattern "01[25]XXXXXXXXX|011XX+T|00X+T|010XX+T|11[02]" are not dialled out(for e.g 001) and congestion tone is played.	
9.	B-668 N-135(v3.0)	FW version isn't displayed after FW conversion	1. Prepare DT730G terminal that FW is NEC-SIP v1.0.3.0. 2. This DT730G's FW is converted to STD-SIP v3.0.24.23 with IP Phone Manager. 3. After FW conversion, Press Menu. 4. Press 6 (Status). 5. Press 1 (Terminal Information). 6. Change to the screen of "Firmware Version". However FW version is not displayed.	

1.5.65 List of Fixed Issues in STD-SIP v3.0.27.5

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID	Enhancement	Enhancement Description	Remarks

1.	None	Localization For Asia	This firmware supported localization for Asia. The purpose of this firmware is ONLY for 3C’s integration testing and confirmation in Asian country.	

1.5.66 List of Fixed Issues in STD-SIP v3.0.24.23

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID	Defect	Defect Description	Remarks
1.	B-661 N-133 FR-001130296	DT700G Firmware has issue when you do BOOT SERVER ", "SOURCE PREFERENCE", "MANUAL CONFIGURATION" and set PROTOCOL to HTTPS. Phone uses IP port 443 while UCM expects connection on IP Port 9443	DT700 phones boot server port can be set for port 9443 via DHCP This setting is not accessible via the TUI config menu. The issue is with remote users / home office users - ex sales persons. There are a handfulnow, but several hundred are planned. DHCP is not an option for these. Short term new certificates are required with discrete IP addr. This is tedious to configure. Steps : Go to Menu, Network Administration, Boot Server, Server Address. You will see that the only option is to enter IP Address	Current firmware(3.0.24.23)doesn't support HTTPS. "Server Port" setting is used for TFTP and FTP only.
2.	None	Boot Server Port can be configured via DHCP.	This functionality has been added in STD-SIP v2.3 NEC IP Phone Network	

			Features Functional Spec.doc (Revision 7.11). This functionality was added in this release.	
3.	B-660 FR-927130007	DT700: Customer find items under menu status a security risk	The items under the main menu number 6 (Status) hold 3 items, 1) terminal information, 2) Network information, 3) SIP information. Items 2 and 3 are seen as a security risk.	A configuration parameter "security.secure.status.enabled" is introduced. When it is enabled, the status screen (all three, therefore, terminal, network, sip information) will be displayed under Admin settings screen only. When it is disabled the status screen (all three, therefore, terminal, network, sip) will be displayed only under Menu screen.

1.5.67 List of Fixed Issues inSTD-SIP v3.0.24.9

Please refer to the below table for the defects fixed after November 1st Release:

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID	Defect	Defect Description	Remarks
1.	B-658 FR-903130073	DT730CG uploads the directory unwanted after a reboot.	When I change a directory entry the DT730CG will upload it's directory after about one minute. If I do a reboot via the menu the DT730CG again uploads the directory, this is incorrect. The DT700 2.3 works fine. It	After reboot, upload of directory file was triggered even though it was uploaded earlier.

			prevents the customer from using MA4000 and prepare directories for the phones. They will be overwritten due to the incorrect behavior of the DT730CG.	
2.	B-478	Reboot is slow when triggered by DHCP and LLDP change	1. Make sure phone is up normally without any vlan id setting on phone and switch 2. Set VLAN id = 4094 on switch port where Lynx is connected and enable LLDP on switch port using command "lldp run" Expected Result: Lynx phone should reboot and take the VLAN ID through LLDP Actual Result: Lynx phone does reboot and takes VLAN id as 4094 but it is slow as compared to STD-SIP V 2.x	The reboot was slow only when VLAN ID was changed via LLDP. In case of BOOTP/DHCP, the behaviour is same as that of STD-SIP v 2.x.

1.5.68 List of Fixed Issues inSTD-SIP v3.0.23.1

Please refer to the below table for the defects fixed after September19thRelease:

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID	Defect	Defect Description	Remarks
1.	B-613, B-650(Duplicate 613), FR-927130004	DT730CG TLS not compatible when using OpenSSL 1.0.1e Polycom works OK	DT700 TLS not compatible when using openSSL 1.0.1e Polycom works OK. The issue seems to be related to the version numbers that the DT700 uses. The DT700 refuses to send a client key exchange.	This is duplicate bug of STD-SIP v2.3 FR-903130037. This issue is not reproducible on STD-SIP v3.0. However, OpenSSL in STD-SIP 3.0 firmware has been upgraded to version 1.0.1e.
2.	B-651, FR-927130005	Last number repeat button not cleared when call history is cleared.	When I go via the menu and delete all placed and received calls the repeat button this works. From a privacy point of view this is not correct. When I delete all my placed calls I also want the repeat button to be empty.	This is duplicate bug of STD-SIP 2.3 FR-903130050. This bug is fixed.
3.	B-652, FR-927130006	Error in the FTP protocol.	The DT700 issues FTP commands like CDW /" this should be CDW /.	This is duplicate bug of STD-SIP 2.2 FR-903130051. This issue is not reproducible on STD-SIP v3.0. This is because STP-SIP v3.0 uses curl library for the purpose of FTP. This bug occurs because of STD-SIP 2.2’s specific FTP mechanism which is not applicable for STD-SIP 3.0, hence no changes has been done for this bug.

1.5.69 List of Fixed Issues inSTD-SIP v3.0.21.19

Please refer to the below table for the defects closed after June21st Release:

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID	Defect	Defect Description
1.	B-644, N-130, FR-001130215	DT730G Phone are not appending DNS domain Name to host name.	On the DT730G Phones when using the DHCP to hand out the DNS domain name option the DT730G phones are not excepting this option. Example: In the UNIVERGE 3C system under the Station > NEC tab where we would normally program the Service and Corporate Dir. URL. We have configured the following URL http://HostName/nec/home.xml On the DT730G it does not receive the option from the DHCP scope options and does not append the domain name. If we change the URL to the following the phone will use this feature correctly. http://FQDN/nec/home.xml
2.	B-645, N-131, FR-001130217	LLDP not working correctly on DT730G on Enterasys Ethernet	Take any model DT730G phone. Take any version of DT730G firmware. (Currently DART at latest 3.0.18.21). In the lab LLDP works between DT730G and my Cisco SF300 switch. But at DART site, LLDP is failing between DT730G and Enterasys switch. DT730s work fine both in lab and at DART, so this is only an issue between DT700G and Enterasys. What we want is both the Voice and Voice Signaling Network policies for LLDP-MED to be honored by the DT700G. However what we see in a Wireshark is that DT700G never sets these policies.

3.	B-335 FR-927120001	DT730G TFTP fails with booter.exe	Booter.exe accepts file uploads from the DT700, but does not accept uploads from the DT730G. 1.Into the phone "admin" and "6633222" 2.Input "su" 3.Input password "6633222444" 4.Input "necicli" 5.Input "DBGCAP get filename.bmp" 6.Input "DBGCAP upload filename.bmp bootserverip" If the TFTP server is booter.exe, it aborts the file transfer. If the TFTP server is Solar Winds, the transfer works.For some reason the DT730CG sends the upload command twice.
4.	B-401 N-71	Unable to assign the speed dial key to the keys on 8LK. This issue occurs on both 32D DG and 32D CG	Go to "Admin Settings>3.Maintenance Settings>3.Key Kit>1.value(kit2)" screen on 32D. 2. Press soft key "MaxLines". 3. Select "3.32 Line", and then press soft key "OK". 4. Save and Reboot. 5. Phone starts up. 6. While phone is idle state, press the Feature key then press any line keys on 8LK e.g. Line 32. 7. Problem starts here, screen sometimes does NOT move from "Press programmable key to continue" screen to Speed Dial Key registration screen.

5.	B-495, N-121	Even if the extension of IP address lease succeeds, DT730G doesn't send LLDP packet.	1. Prepare one DG or CG terminal. 2. Prepare switch that supports LLDP, and enable LLDP feature of switch. 3. Set lease time limit to two minutes on DHCP server. By this setting, the extension request of IP address occurs every one minutes. 4. Connect terminal to LLDP supported switch, and turn on the power. 5. When terminal sends DCHP renew, DHCP server sends DHCP ACK to terminal. After this, confirm the sending of LLDP packet from terminal. (V2.x sends LLDP packet at the time. This is specification.)
6.	B-482	The phone does not download the certificate mentioned in cfg file if there is a blank space in the ROOT_CERTIFICATES_PATH	1.Place the certificate on tftproot 2.Put a blank space in ROOT_CERTIFICATES_PATH 3.Put security.ssl.root.certs="SIP driver.crt" The phone tries to download "/ /SIP driver.crt" due to which the download fails.
7.	B-637	LLDP timer gets reset once DHCP lease ACK is sent.	1. Prepare one DG or CG terminal. 2. Prepare switch that supports LLDP, and enable LLDP feature of switch. 3. Set lease time limit to two minutes on DHCP server. By this setting, the DHCP lease renew occurs every one minutes. 4. Connect terminal to LLDP supported switch, and turn on the power. 5. When terminal sends DCHP renew, DHCP server sends DHCP ACK to terminal. 6.Verify the LLDP packets throughout and after above activities:

			Actual Result: LLDP packets are sent from terminal every 30 seconds and once DHCP ACK is sent, then LLDP packets are sent immediately and then onwards LLDP packets are sent in 30 sec interval. LLDP packet sending timer gets reset after sending LLDP packets during lease renewal. Expected Result: LLDP timer shouldn't be reset after sending DHCP ACK packets for lease renewal and LLDP packet should be sent continuously within 30 sends interval irrespective of LLDP packets which are sent after lease renewal.
8.	B - 643	DNS server(Option number 6) addition is not picked after DHCP lease renewal if initially if was not present in DHCP server	1) DHCP scope is configured with lease time of 2 minutes. 2) Unselect the DNS server option(Option number 6) from DHCP server scope 3) Perform factory reset on DT730G 4) After booting up,verify that phone displays DNS server address as "127.0.0.1" in status. 5) Configure DNS server (Option number 6) with valid IP address in DHCP server. 6) Wait for DHCP lease renewal. 7) After lease renewal check DNS server address in status. It still shows "127.0.0.1" instead of configured DNS server address on DHCP server.

9.	B - 646	Phone is not updating DHCP domain name (option15)after lease renewal	Pre-requisite: 1. DHCP domain name (option 15) is configured with valid domain name like "sphere.com" and Phone is up. 2. In the UNIVERGE 3C system under the Station > NEC tab where we would normally program the Service and Corporate Dir. URL. We have configured the following URL http://HostName/nec/home.xml. Steps: 1. Access the corporate directory and verify DNS query in trace. DNS query is made by appending “sphere.com” domain which is successful. 2. Change the option value 15 with invalid domain name like "test.com" and wait for lease renewal. 3. Again access the corporate directory and verify DNS query in trace. DNS query is still made with appended “sphere.com” instead of “test.com” which is incorrect behavior. 4. Reboot and access the corporate directory and verify DNS query again. DNS query is made with appended “test.com” domain 5. Again reconfigure valid domain name "sphere.com" and wait for DHCP lease renewal. 6. Now again access the corporate directory and verify DNS query in trace. DNS query is still made to “test.com” instead of “sphere.com” domain which results in “Page cannot be displayed” error.

10.	B - 647	Router addition (Option Number 3) is not picked after DHCP lease renewal if initially if was not present in DHCP server.	1) DHCP scope is configured with lease time of 2 minutes. 2) Unselect the Router option(Option number 3) from DHCP server scope 3) Perform factory reset on DT730G 4) After booting up, verify that phone displays Default gateway address as "0.0.0.0" in status. 5) Configure router address (Option number 3) with valid IP address in DHCP server. 6) Wait for DHCP lease renewal. 7) After lease renewal check Default gateway address in status. It still shows "0.0.0.0" instead of configured router address in DHCP server.

1.5.70 List of Fixed Issues inSTD-SIP v3.0.18.21

Please refer to the below table for the defects closed after May23rdRelease:

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID	Defect	Defect Description
1.	B-550, N-129, FR-927130003	DT730CG/DG sends the unsubscribe messages too fast when applying TLS. It is the same issue as FR-903130022 but now the SIP driver behavior has been improved. The DT730CG/DG cannot handle the 200 OK messages fast enough, so it should not Send the UNSUBSCRIBE messages that fast.	This scenario occurs when multiple SIP “Subscribe” messages for presence are sent together (without any delay) and responses are not received for some “Subscribe” messages. This leads to resending of “Subscribe” messages again for which responses are not received.

2.	B-464, N-105	EAP packet for a terminal is forwarded to the PC side.	This scenario occurs for 802.1x multi-domain authentication on CISCO switch when phone connected directly to switch forwards EAP requests, addressed to its own MAC, to PC/phone connected to its PC port.
3.	B-507	Sometimes phones (connected to switch & other to PC port of first phone) went in held state when simultaneously reboots	This scenario occurs when client limit is not set on HP switch or multi-host configuration on cisco switch for 802.1x authentication. For the above setting, supplicant's (in our case, phone's) MAC specific filtering is not applied on switch. Switch doesn't send supplicant's MAC specific request, instead request addressed to "Nearest" port (01::80::c2::00::00::03) is sent. These requests are received by both the phones (one connected directly to switch and another one connected to PC port of first phone). Both the phone responds with their own 802.1x credentials which sometimes results in "identity" of one phone coupled with "password" of another phone being processed by switch. This leads to authentication failure as switch doesn't perform any check of the MAC from which it has received response (because of settings applied on switch). After authentication failure, both the phone switches to "HELD" state and retry authentication after the expiry of held period timeout. This cycle continues, till phones come up by using authentication credentials ("identity" and "password" of single phone) which leads to both the phone getting authenticated. This situation is prevented by setting client limit on HP switch and switching to multi-domain authentication on CISCO switch. This leads to phone's MAC specific filtering being applied by switch and each phone getting authenticated by their own 802.1x credentials.

4.	B-529	Extra logoff packet when PC port reboots.	This scenario occurs for 802.1x authentication when a phone connected to PC port of another phone is rebooted. Phone which is connected directly to switch sends proxy logoff on behalf of phone connected to PC port. Phone connected to PC port sends logoff for itself before rebooting which leads to duplicate logoff being sent for MAC address of phone.
5.	B-530	LAN port Responds to PC port specific request.	This scenario occurs for 802.1x multi-domain authentication on CISCO (or client limit 2 on HP switch) switch when a phone is connected to switch directly and another phone is connected to PC port of first phone. In this scenario when switch send EAP request addressed to MAC of PC port phone, the phone connected directly to switch sends responses to these request along with forwarding these request towards PC port connected phone.

1.5.71 List of Fixed Issues inSTD-SIP 3.0.17.23

Please refer to the below table for the defects closed after May 6th Release:

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID	Defect	Defect Description
1.	B-503, N-128, AR-927130001	DT730CG: When the user portability feature is enabled, and if the user logs in, the phone is not creating user Local Configuration file if one does not exist on the boot server.	The user fails to login leaving the phone with the Basic DNR and phone functionalities as it fails to create the user local configuration file. Below is the required functionality: When the user logs in, the phone tries to download this file from the ftp server. If the file is not found (GET error response), the phone creates one (say, dt-1025-local.cfg)

			based on the MAC specific (dt- MAC-local.cfg) file, and uploads the same onto the ftp server. Observation: However in this case, DT730CG does not seem to be creating the user local config file. This is probably because the Linux ftp library may be returning a different response code that needs to be handled in creating the file.
2.	B- 502	PC port terminal show status HELD(x) for LAN port terminal authentication.	1. Connect a LYNX terminal to authentication port of switch. 2. Connect another LYNX terminal to PC port of terminal 1. 3. Reboot both the terminals together. Actual Result: After reboot, LAN port terminal gets authenticated while PC port terminal shows HELD(X). Expected Result: PC port terminal should either get authenticated or should show any of the following state: 1. If EAPOL start has been send by PC port terminal, then CONNECTING(X) should be shown. 2. ACQUIRED (X) should be shown for the duration from EAP IDENTITY request receipt to MD5-CHALLANGE response sent. 3. AUTHENTICATING (X) should be shown from MD5-CHALLANGE response to EAP SUCCESS / FAILURE. 4. HELD (x) if FAILURE is received. 5. AUTHENTICATED if success is received.

3.	B-465, N-109	"802.1x: Connecting(X)" displays on LCD after pulling a LAN cable which is connected to LAN port on Lynx. This issue occurs on both DG and CG.	1. Configure Cisco 3560G as follows; Enable multi-domain authentication to authenticate both PC and Lynx. 2. Enable 802.1x setting on Lynx. Note that set account and password for 802.1x authentication correctly. 3. Connect a PC to PC port on Lynx, then connect Lynx to a port on Cisco 3560G which is configured for 802.1x. 4. Reboot a phone. 5. Ensure a phone comes up after succession 802.1x authentication. 6. Pull a LAN cable from PC. 7. Problem starts here, "802.1x: Connecting(X)" displays on LCD.

1.5.72 List of Fixed Issues inSTD-SIP 3.0.17.6

Please refer to the below table for the defects closed after April9th Release:

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID	Defect	Defect Description
1.	B - 492 FR-927130002	DT700 is not showing P&O when assigned to a phone	DT730G is not showing P&O when assigned to a phone. In SIP@Net I can assign a name to a phone using CHNAME. If I assign as name P&O the string is not displayed, eventhough it's passed in the NOTIFY. The phone does display the text when it is called from an extension that has P&O assigned as name.
2.	B - 499	Phone sending second EAPOL start messages after 1 sec after first EAPOL	Phone sending second EAPOL start messages after 1 sec after first EAPOL messages sent even if the 8021x.startperiod is default 30 sec

3.	B – 500 FR-927130001	DT700 reboots/disconnects with Polycom softphone	If I make a call from a Polycom softphone to a DT700, the DT700 reboots. If I make a call from a DT700 to a Polycom softphone the DT700 disconnects after answering.
4.	B – 501 Tracker - 18086	Change of Softkey label from “Detail” to “Details”	Change of Softkey label from “Detail” to “Details”
5.	B – 459 N - 100	802.1x: Held(X)" doesn't display at the following scenario."	1. Prepare authentication hub which is enabled 802.1x authentication feature. 2. Configure a RADIUS server with correct configuration. 3. Enable 802.1x authentication feature in configuration on Lynx, also setaccount and password for 802.1x authentication. 4. Reboot a phone. 5. Ensure a phone comes up after succession 802.1x authentication. 6. Pull a LAN cable which connected with RADIUS server from authentication hub. 7. Reboot a phone. 8. Ensure error message on LCD. 9. Problem starts here, "802.1x:Conecting(X)" displays on LCD instead ofdisplaying "802.1x:Held(X)". NECi kept this situation for over 48 hours, as aresult, "802.1x:Connecting(26000)" still displayed on LCD.

1.5.73 List of Fixed Issues inSTD-SIP 3.0.16.9

Please refer to the below table for the defects closed after March20th Release:

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID	Defect	Defect Description
1.	B- 463 N- 104	DSCP is not configured correctly from LLDP settings	1. Prepare phone A and B. 2. Configure HP switch, HP ProCurve2910al-24G-PoE+, with as an attached file, HPconfig.txt. Note that DSCP is not default value.

			3. Connect a repeater hub to a port which is configured for LLDP on HP, and then connect a phone and PC for Wireshark to the repeater hub. 4. Start a Wireshark. 5. Power on phones, and then phones come up with LLDP settings. 6. Make a call A and B. 7. Stop a Wireshark, and then check a Wireshark trace. Problem starts here, DSCP of SIP/RTP packets is not expected value.
2.	B- 494 N- 120 Tracker 19706	The DT730G is rebooted by operate "Unhold" and "EndCall" at the same time.	When the "UnHold" operation event and the "EndCall" operation event occur at the same time, the hold terminal will be rebooted.
3.	B- 496 N- 122	When VLAN setting in a LAN port is effective, 2 bytes of extra data "00 00" will be added to the tail end of the packet.	When VLAN setting in a LAN port is effective, 2 bytes of extra data "00 00" will be added to the tail end of the packet.
4.	B- 493 FR-903120051	Restart not handled when DT700 not registered	DT700 firmware V2.2+ accepts a NOTIFY request with Event: x-server- restart as an indication that the SIP server has restarted and subscriptions must be reestablished. Note that x-server-restart is currently implemented by SIP@Net, but not by Sphericall. This feature request documents two observations: a. If the DT700 is not registered when x-server-restart is received, the DT700 could cancel any delay timer and attempt to reregister after a short (< 5 seconds) random interval. The random interval distributes the server processing load for installations with a single server and many phones. b. The DT700 should only reestablish the subscriptions if the x- server-restart is received from the current subscription server. For example, the x-server-restart should be ignored in the following case:

			1. DT700 subscribes to server 1 2. Server 1 crashes 3. The subscriptions expire 4. The DT700 subscribes to server 2 5. A x-server-restart is received from server

1.5.74 List of Fixed Issues inSTD-SIP 3.0.15.20

Please refer to the below table for the defects closed after March 7th Release:

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID	Defect	Defect Description
1.	AR-903130009 B-483 N-118	LLDP behavior on the DT730CG/DG incorrect and not identical to DT700 2.3	LLDP behavior on the DT730CG/DG incorrect and not identical to DT700 2.3
2.	Bug-484	With length as 0 under option 43 causes problem in status	Setup DHCP option 43 on DHCP for boot server like "42 00 31 31 ...." here the second length field is 0. Actual Result: Lynx does not ignore the 0 length value and under status of boot server shows some error string "Parse error in line 9".
3.	Bug-451 N-92 AR-000130020	SIP Local Port setting is not changed correctly.	1. Configure sip.local.port to be 50600 from 5060. 2. Start a Wireshark. 3. Reboot a phone. 4. Ensure call control screen with registered icon. 5. Stop a Wireshark Problem starts here, phone is Busy status if trying to call. As per Wireshark trace, a SIP port no. which is sent from phone is NOT 50600, it seems to be random number.
4.	Bug-453 N-94	A time until "Download Failed" displays is longer than v2.3.	1. Download a firmware by using IP Phone manager with TFTP server or FTP server. 2. Pull a LAN cable on purpose after starting download.

			3. Wait until "Download Failed." Problem starts here, the time until "Download Failed" displays is longer than v2.3.
5.	B-426 N-112	SRTP media icon not displayed during call transfer	Call scenario of attended transfer: 1. A calls B,B answers 2. A calls C,C answers 3. A transfers B to C 4. B and C are in call. 5. A is in SRTP mode 0, B and C are in SRTP mode=1. 6. When call is transferred by A to C, B doesn't show SRTP media icon (lock) while C shows. Only C shows the SRTP media icon however Both B and C should show the SRTP media icon.
6.	B-486	DtermIPInput shows incorrect arrows	Emulator Bug 1.Select the option DtermIPInput on Emulator as well as Lynx-CG 2. Scroll up/down 3. Observe the arrows on top-right corner Emulator shows incorrect arrows
7.	B-488	Image mismatch on DtermIPMulti between phone and Emulator	Emulator Bug Select DtermIPMulti on Lynx-CG,DG and Emulator There is an image mismatch between the phone and Emulator This issue occurs on both Lynx-CG as well as DG
8.	B-489	# has different symbol sequence on phone and Emulator	Emulator Bug 1.Select option DtermIPInputLine on Phone as well as Emulator 2.on pressing # the symbol sequence is different on Phone and Emulator in case of alpha [ABC]for options:1,2,5,6,9 Note: Symbol sequence is same for numeric [1] This issue occurs on both Lynx-CG as well as DG

9.	FR-903130026	DT700 stuck in DHCP request it does not start properly, Polycom does	DT700 stuck in DHCP request it does not start properly, Polycom does – This issue was not occurring on v3.x however to make other scenarios behavior similar to v2.x it is merged.

1.5.75 List of Fixed Issues inSTD-SIP 3.0.15.7

Please refer to the below table for the defects closed after February 13th Release:

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID	Defect	Defect Description
1.	B-363 AR-000120198 N-108	Transfer and transferee button pressed at the same time causes reboot	During complete automatic transmission, in the test where forward destination EndCall and forward button of (forwarder) the person who forwards the call (DT730CG), terminal of the forwarder rebooted.
2.	B-481	Extension number is not displayed at times when phone.sphericall.hotdesk.enabled = 1 and phone.show.registration.name = 0	1. Set phone.sphericall.hotdesk.enabled = 1 2. Set phone.show.registration.name = 0 3. Reboot Extension number is not displayed at times on Lynx-CG,DG and Std. Sip
3.	B-68 AR-000120194	Register Icon does not get updated	Sometimes, register icon is not green after startup.
4.	B-362 N-96 AR-000120197	During transfer, when transfer button of the transferor and End call of transferee (DT730DG) are pressed together, the terminal returns nothing to REFER	1. A and B are talking 2. A calls C 3. Transfer button of A and EndCall of B (Soft Key) are pressed simultaneously. Actual Result: When transfer button of the transferor and End call of

			transferee (DT730DG) are pressed together, the terminal returns nothing to REFER. It's an idle state on the screen of the forwardee terminal.
5.	B-466 N-110 AR-903130009	LLDP packet from HP switch is forwarded to PC side	1. Configure HP switch, HP ProCurve2910al-24G-PoE+, for LLDP. 2. Connect a PC for Wireshark to a PC port on Lynx, then connect a Lynx using LAN port to the HP. 3. Start a Wireshark. 4. Power on phones, and then phones come up with LLDP settings. 5. Stop a Wireshark, and then check a Wireshark trace. 6. Problem starts here, LLDP packet is forwarded to PC.
6.	B-471	Lynx phone crashes sometimes when bootp server is not provided	Phone crashes intermittently in the absence of boot server from DHCP
7.	B-457 AR-001130016	DT730G crashed once while changing the language.	DT730G crashed once while changing the language.
8.	B-479	Phone does not display extension number sometimes even when phone.show.registration.name=1	1.Set phone.show.registration.name="1" in phone-sip.cfg 2.Reboot phone Sometimes the lynx phone does not display the extension number
9.	B-410 N-107 AR-000120217	To pick up the incoming call with pressing Pickup softkey, about four seconds is required	The "Pickup" button doesn't work. 1. Push the DT730G's "Pickup" button 2. Push the DT730G's "Extn" button 3. Dial the extention Number and 4. Push the DT730G's "Pickup" button *But the "Pickup" button doesn't work.
10.	B-449 N-90 AR-000130015	Show Registration Name always displays in spite of configuration in cfg file.	1. Configure phone.show.registration.name to be 0 (Disable). 2. Reboot a phone. 3. Ensure right side on call control screen after phone comes up. 4. Problem starts here, extension no. which is line.1.extension displays on call control screen. If phone.show.registration.name is 0 (Disable), this extension no. does not display.

11.	B-452 AR-000130005 N-93	Terminal sets Refresher. After connecting MeetingHUB, terminal reboots at refresh timing.	Terminal sets Refresher. After connecting MeetingHUB, terminal reboots at refresh timing. *If there is no refresh, terminal works without problem. *If 3C is refresher, terminal works without problem. *This issue doesn't occur in V2.2.8.8.
12.	B-456 AR-000130011 N-98	When headset is enabled, handset volume cannot be changed	1 .Move to "3.User Settings > 2.Headset > 1. Enable/Disable", then select "2.Enable" 2. Move to "3.User Settings > 2.Headset > 2.Ringing", then select "3.Phone and Headset". 3. Back to Home (call control) screen. 4. Go offhook. 5. Press cursor up or down key. 6. Problem starts here, unable to change handset volume though volume indicator on Home screen displays correctly. The DT should play dial tone in the handset.
13.	B- 475	Message time is not synched correctly for Lynx DG and CG in syslog	1. Set syslog server address for Lynx phone with primary address 10.112.94.9 2. The time of messages of the logs is not in synch with the time displayed on phone(from SNTP)

1.5.76 List of Fixed Issues inSTD-SIP 3.0.14.13

Please refer to the below table for the defects closed after February 7th Release:

S No.	Bugzilla ID (B), NEC Issue ID (N), GM Issue ID	Defect	Defect Description
1.	B-460 N-101	“Hard Reset" command by IP Phone Manager	Hard Reset" command by IP Phone Manager
2.	B-416	LCD backlight functionality goes corrupt If there is No light on waiting for lldp screen(DG).-Occurrence 20%	LCD backlight functionality goes corrupt If there is No light on waiting for lldp screen(DG).-Occurrence 20%
3.	N-86	Phone does not use programmable key setting which is as Enhanced BLF Key in dt-000000000000-directory.cfg. This issue occurs on both DG and CG.	Phone does not use programmable key setting which is as Enhanced BLF Key in dt-000000000000-directory.cfg. This issue occurs on both DG and CG.
4.		DT730G DHCP Vendor Class Identifier is not same as STD-SIP	DT730G DHCP Vendor Class Identifier is NECDT700 instead of NECSDT700.

1.6 Installation Requirements

The Firmware package file contains the Lynx DG and CG Terminal Firmware. Extract the contents from this file and place them in the Boot Server. The Firmware file contains

1. itlissipvg.tgz
2. itlissipvg.sig
3. itlissipvc.tgz
4. itlissipvc.sig
5. MO Files
6. Default Config Files
7. Wav Files
8. necsdt700g.ver file

Note: On 3C place the above files in “necsdt700” folder under “ftproot”

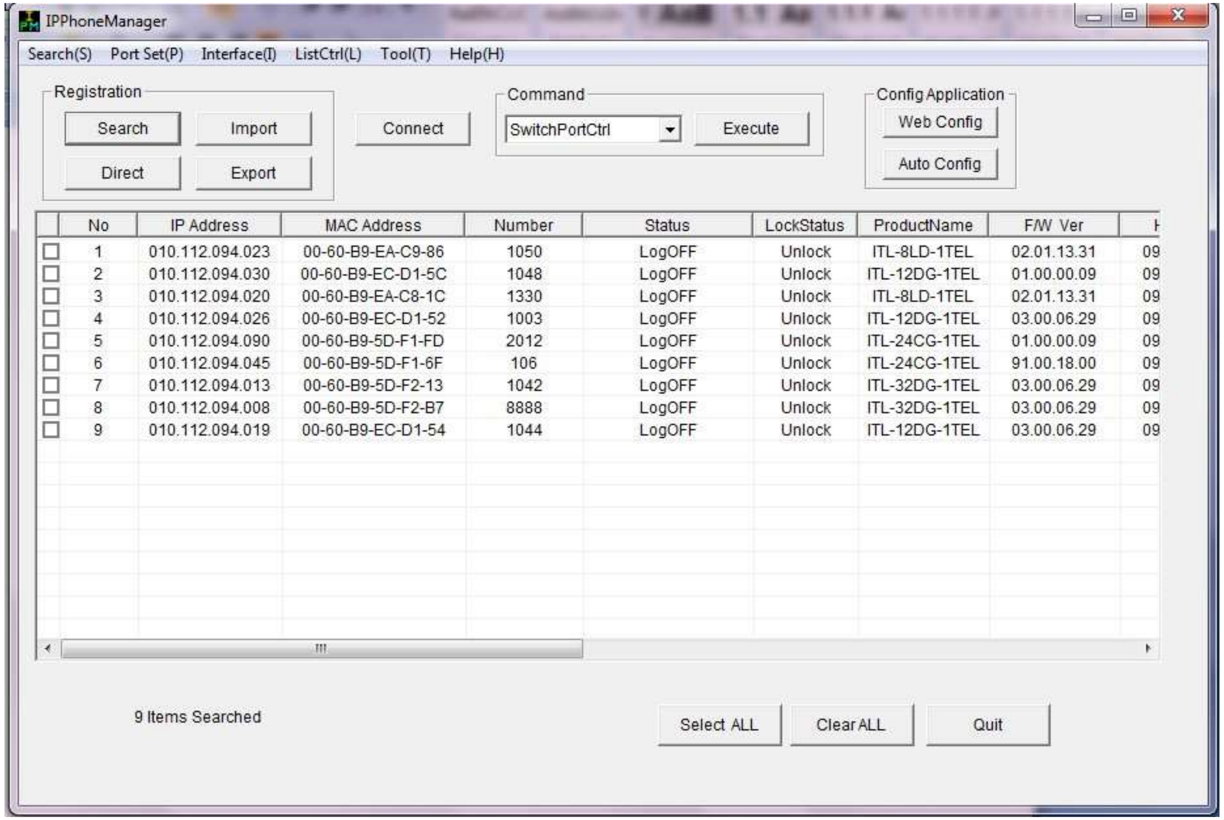
The following are ways to download Lynx Firmware to NEC-SIP R1 Linux DG Terminals:

Using IP Phone Manager

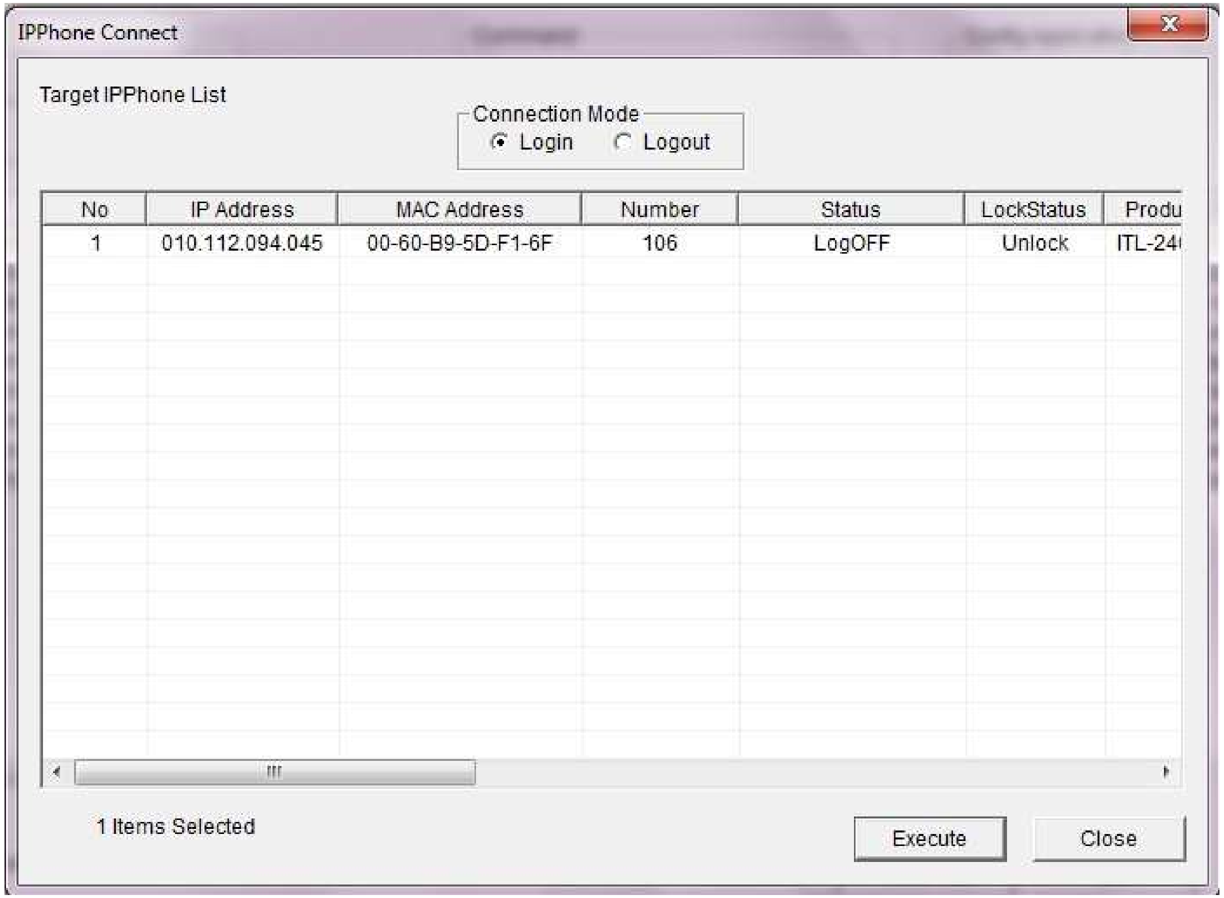
IP Phone Manager Version 6.0.1 supports conversion from NEC-SIP to Lynx firmware and vice-versa. The firmware conversion from NEC-SIP to Lynx on a DT730G Terminal requires a Lynx firmware to be placed on FTP/TFTP Server.

The steps for downloading the firmware through IP Phone Manager are:

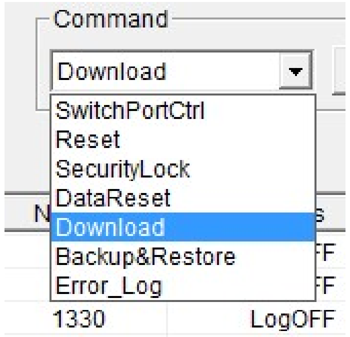
2. Search list of the active Terminals as shown in the figure below.



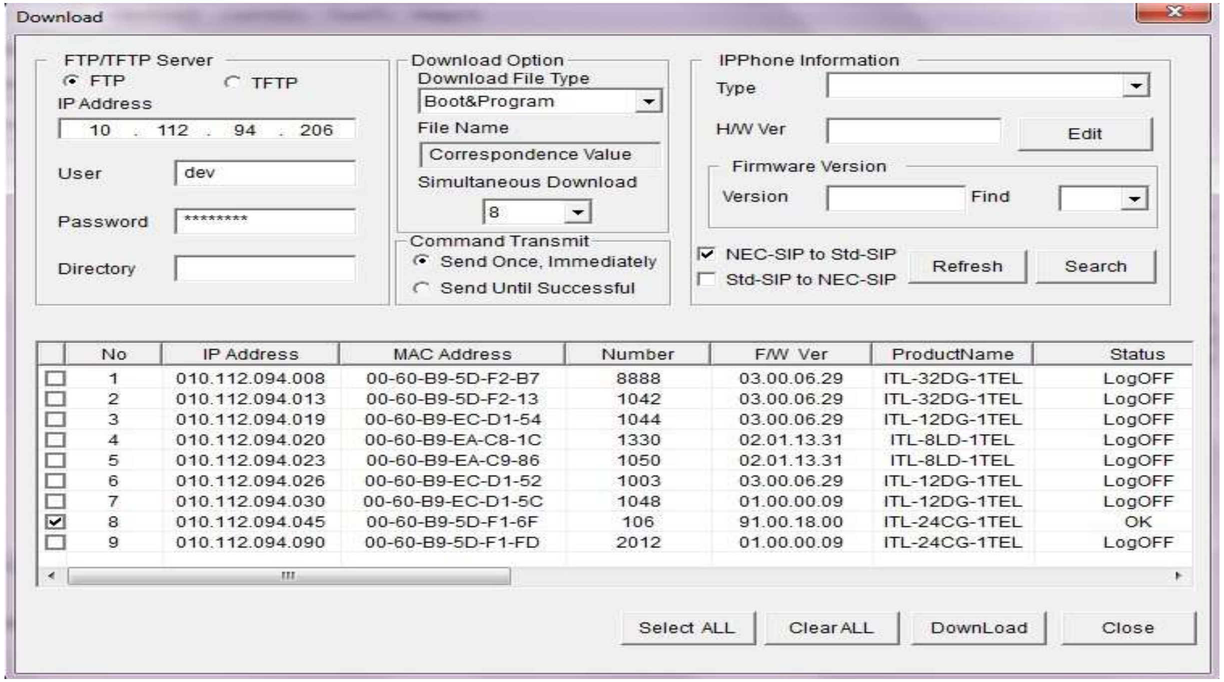
3. Select the Terminal on which the conversion is to be done and press “Connect” to get connected with terminal. After this step terminal will get connected with IP Phone Manager as shown below



4. After the Terminal gets connected, choose the **download** option under the command drop-down menu.



Click on Execute. The following window will pop up



5. Depending upon the Server type FTP or TFTP, the type can be specified in the above screen

    a. FTP/TFTP Server:Select the type of server - FTP or TFTP.
    Define the IP address of the server.
    Enter the User ID (*FTP Only)*.
    Enter the Password (*FTP Only)*.
    Enter the Directory name (*FTP Only)*.

    b. Use the Download Option Field to select the Terminal File type and enter the File type as "**Boot&Program"**.

    c. Under IP Phone Information specify the Type of IP Phone .While converting the firmware from NEC-SIP to Lynx, the type to be specified is:

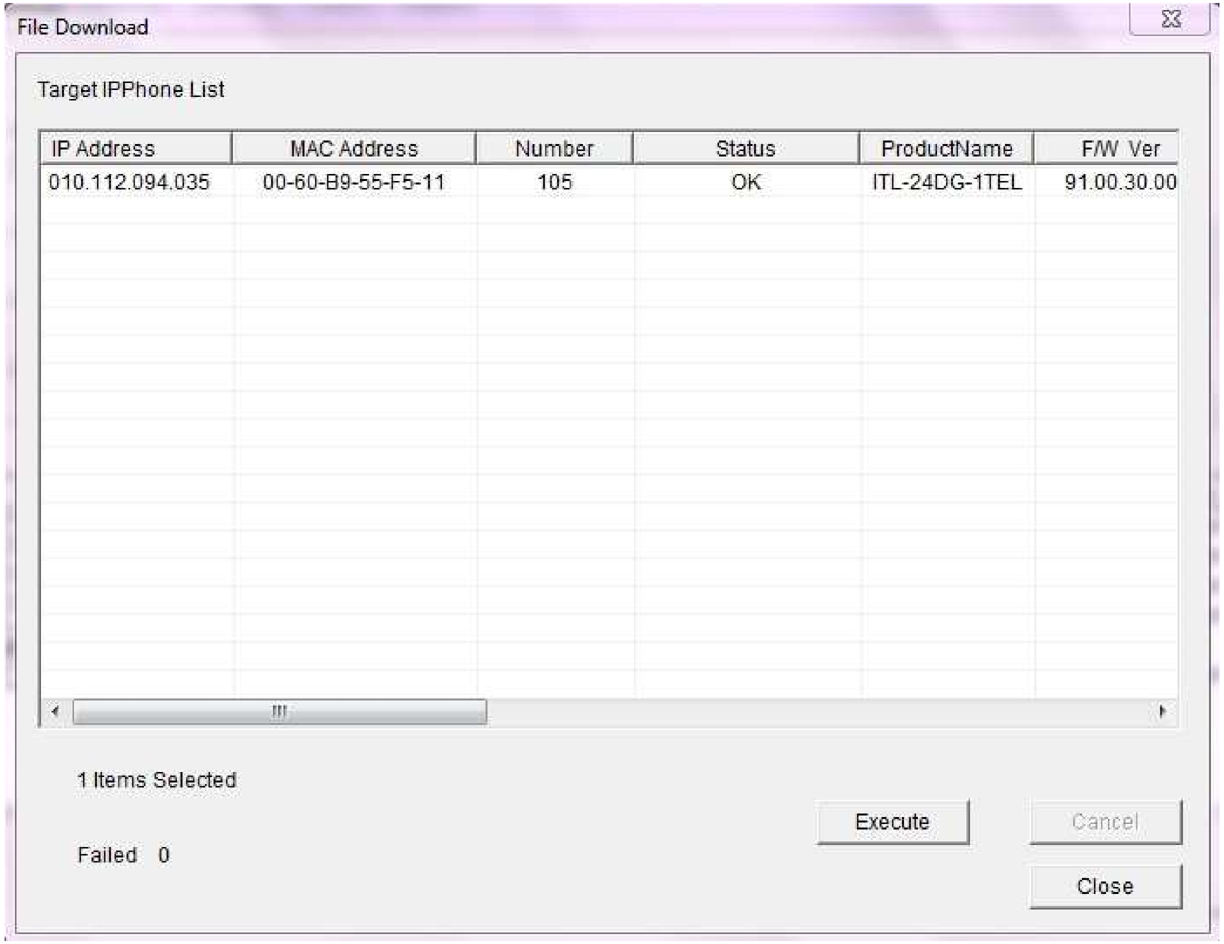
    d. "Std.SIPDT730G" for DG Terminals

    e. Specify the Firmware version in the "Version" field and in the "Find" select "=".

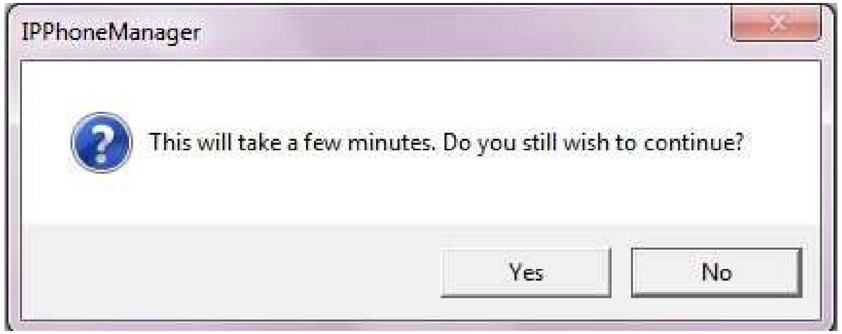
    f. Select the check box for NEC-SIP to Std-SIP for conversion for NEC-SIP to Lynx firmware conversion.

    After the above steps,click on "Download"

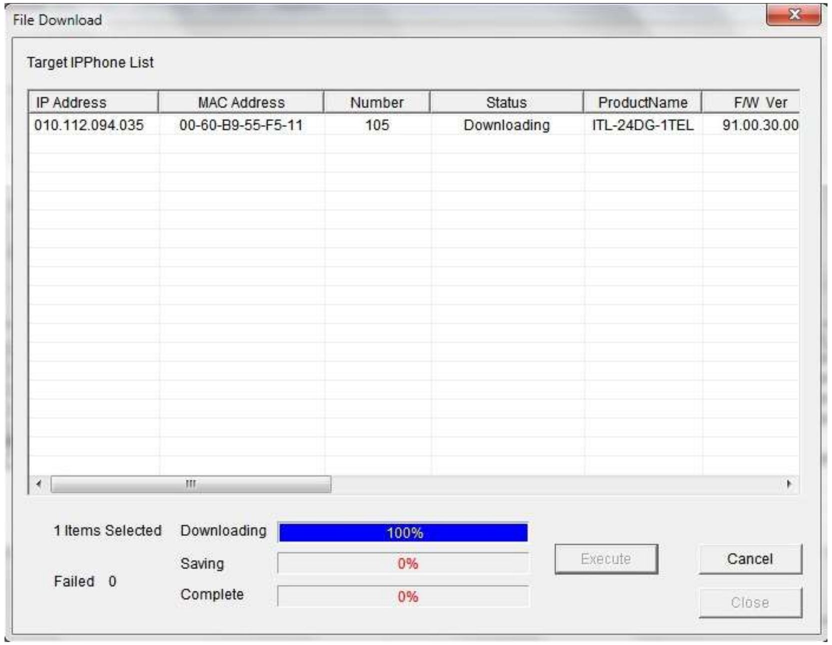
6. The following pop-up window will be displayed .Press Execute to begin the Download process.

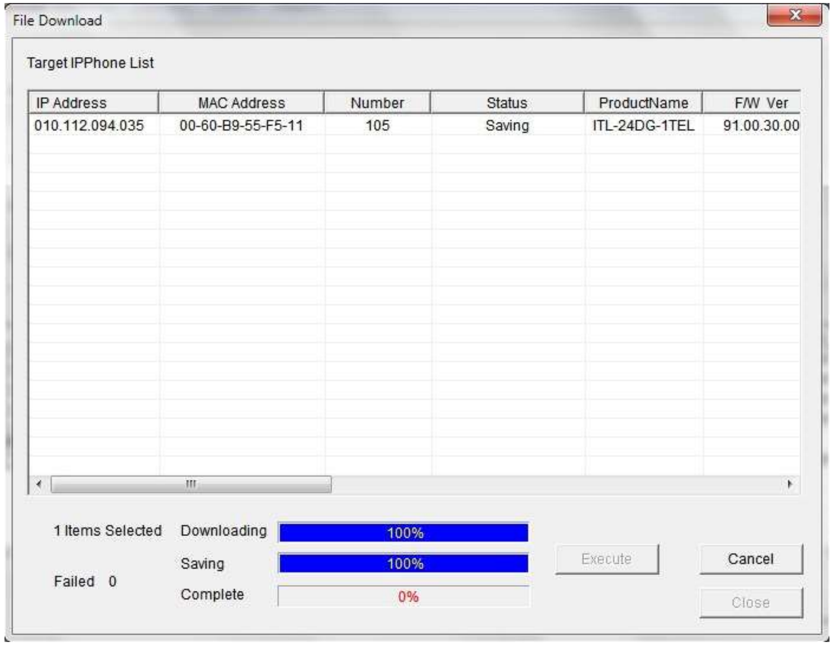


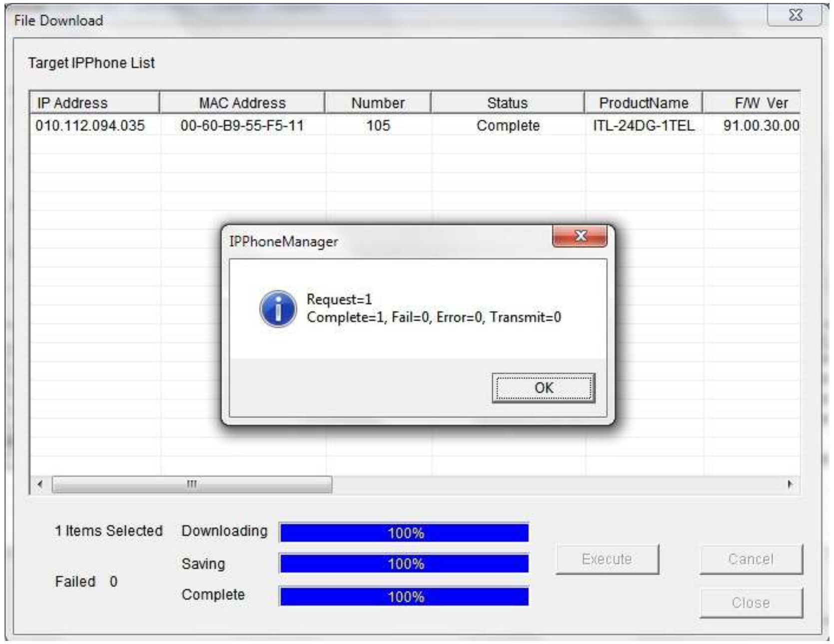
7. On Pressing Execute, the following dialogue will appear...



8. Press “Yes” which triggers the download process. The following screens will come up.




File Download
Target IPPhone List
IP Address
MAC Address
Number
Status
ProductName
F/W Ver
010.112.094.035
00-60-B9-55-F5-11
105
Saving
ITL-24DG-1TEL
91.00.30.00
1 Items Selected
Downloading
100%
Saving
100%
Failed 0
Complete
0%
Execute
Cancel
Close



9. The above screen shows that the Firmware has been Downloaded