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Release Notes

SV9100 CP10 A10.60.55

Product	-	Univerge SV9100 CP10
Version	-	Release A10.60.55
Date	-	08-Sep-2020
Document Reference	-	FCO 2020.071
Category	-	Maintenance
Issued by	-	NEC Enterprise Solutions EMEA

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Contents

1. INTRODUCTION

This FCO provides information about the Maintenance Release of Univerge SV9100 CP10 Main Software.

- SV9100 Main Software A.10.60.55

2. IDENTIFICATION

This release is SV9100 Main Software A10.60.55 is a maintenance release containing A10 features and including fixes. The Prefix A is used to distinguish from the SV9100 CP20 Software. All new SV9100 CP10 versions will have this prefix.

3. COMPATIBILITY

Any UNIVERGE SV9100 CP10 can be upgraded with this system software.

4. UPGRADE INSTRUCTIONS

It is always advisable to save the system configuration prior to any upgrade.

WARNING: Powering off while card firmware is occurring, can cause corruption of cards. Please ensure all cards are running (up to 10 minutes after upgrade dependent on number and type of cards (LCF upgrade is longest) before performing any reset. See further explanation later in this document.

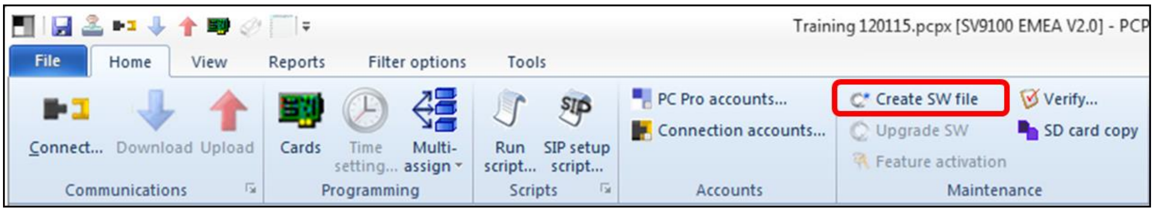
To perform a system software and firmware upgrade:

1. Turn the system power off.
2. Once the system has powered down, insert the USB Memory containing the software upgrade into the USB port on the GCD-CP10.
3. Push in and hold the **Load** button.
4. Turn the system power on.
5. Continue holding the **Load** button for approximately 10 seconds or until Status LED5 begins flashing red.
6. Release the **Load** button.
7. Wait until the Status LEDs on the GCD-CP10 have the following indications (approximately two minutes):
   LED 2: Flashing Red
   LED 3: Flashing Red
   LED 4: Flashing Red
   LED 5: Steady Red
8. Turn the system power off and un-install the USB Memory.
9. Turn the system power back on.
10. When the system has completed reloading the software, the Status LED begins flashing on the GCD-CP10. The remaining four LEDs are off.
    - To confirm the new software version has been installed, the system version number can be viewed by pressing the FEATURE + 3 keys on any display multiline terminal.
    - The existing system software in the flash memory is replaced, but the customer data (stored in the RAM) is saved.

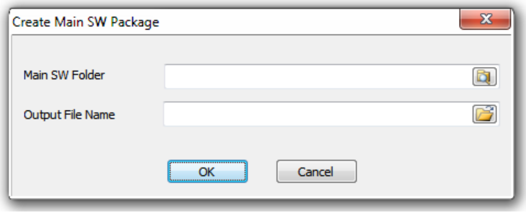
Or via Remote

First create a remote upgrade file from the same software you would add to the USB stick.

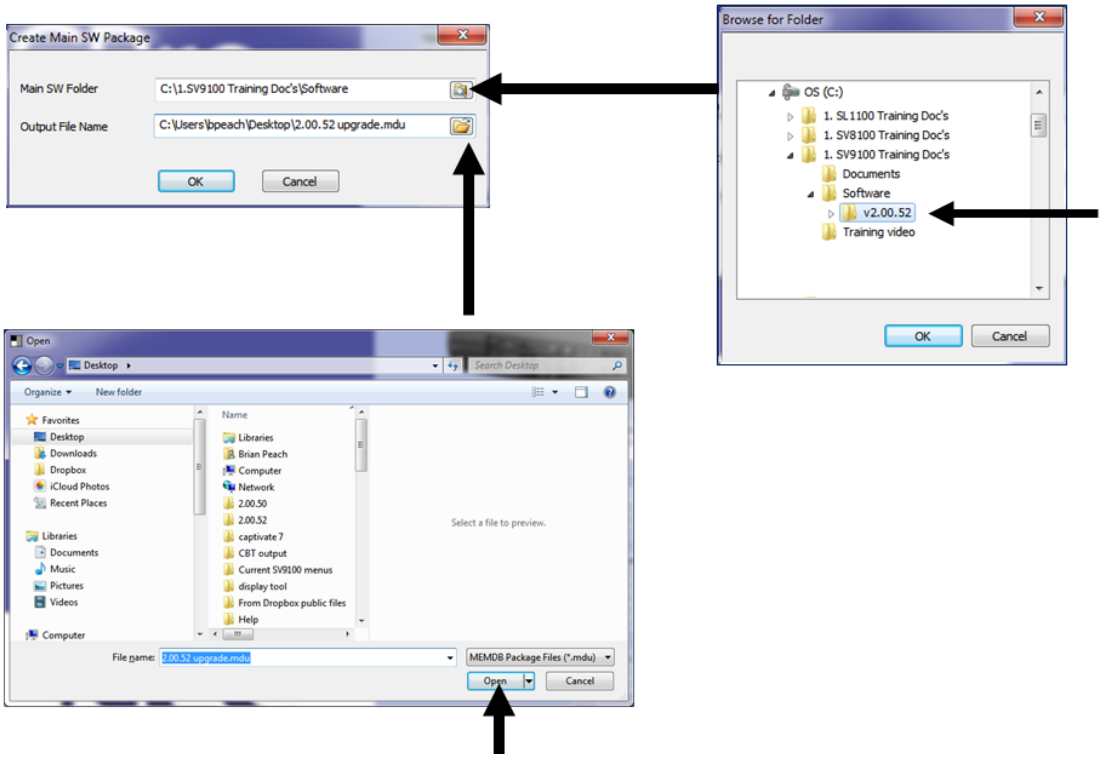
From the ribbon bar, select Create SW file:



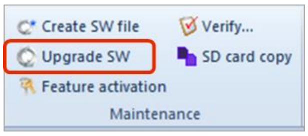
This will pop a window:



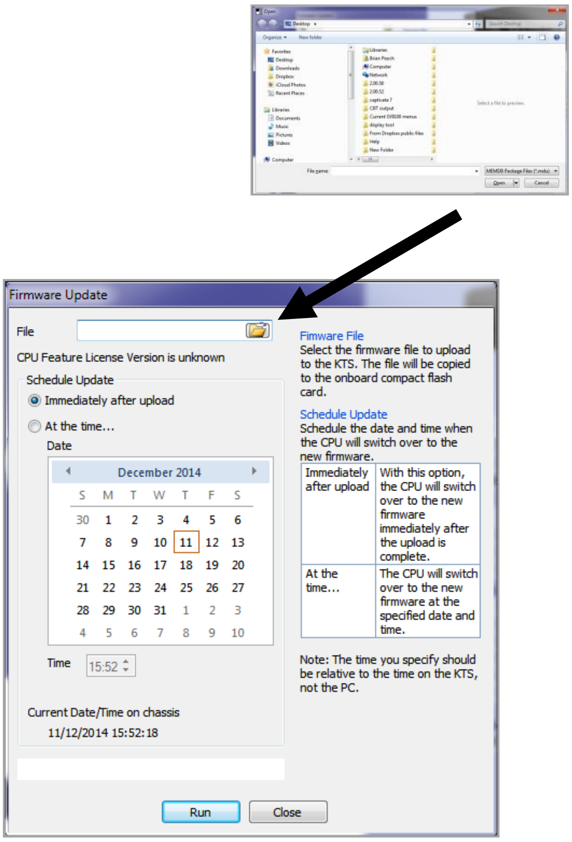
Select the area of the Main SW and where to save the output of the remote upgrade file:



Connect to the system via PCPro in the normal manner, and from the ribbon bar select Upgrade SW



In the File box, select the mdu file you created for the upgrade, then select when the upgrade should occur (immediately or the date specified). At this point the system will perform its normal upgrade cycle and reboot.



Main Software Upgrade and Option Card Firmware

After a main software upgrade the SV9100 reboots (either manually if via USB or automatically if via PcPro/Webpro).

After reboot the main software will then push out to the option cards (including IPLE) any firmware updates.

Firmware updates are not always required, it is dependent on version upgrading from and to.

It is important that during this firmware upgrade the system is not reset, as doing so interrupts the firmware upgrade and can cause corruption of the cards performing upgrade at the time.

Upgrade in a Netlink Environment

1.Access Primary system via PCPro or WebPro in NetLink network.
2.Set PRG51-16-01 to Disable.
Note: This change prevents replication error from occurring between systems.
3.Access Secondary systems via PCPro or WebPro in NetLink network.
Note: Upgrade should start with Secondary systems to update properly.
As such NetLink network is kept up, and you do not need to worry about
the presence of substitute Primary system.
4.Upgrade main system software of Secondary systems.
Note: Check it has upgraded the main system software of all properly.
5.Upgrade main system software of Primary system.
Note: Check it has upgraded the main system software properly.
6.Restore PRG51-16-01 to previous setting data.

5. FUNCTIONAL CHANGES

This version provides:

- Security enhancements

6. SOLVED PROBLEMS

6.1. List of Solved Problems

The following items are fixed in this version:

F191219002	**System doesn’t play fixed message for inbound external calls if CLI is enabled**
	System doesn’t play fixed message for inbound external calls if CLI is enabled on the trunk.
F200714001	**CPU stops responding to SIP Trunks connection**
	SV9100 CPU card not keeps locking up with connection with SIP Trunk that receives TLS Packet from the Carrier. System reboot clears.

6.2. List of Solved Problems in Previous Releases

A10.00.53 Main Software

Reference	Description
F200108001	**Intermediate Certificate not supported**
	SV9100 platform does not support use of intermediate certificates in the chain. Now corrected.

A10.00.53 Main Software

Reference	Description
F191014002	**SL2100 sends a MIX of UDP and TCP in registration request**
	The SL2100 switches between UDP and TCP for outgoing call invites following the DNS Query. If the Transport layer is UDP the call routes correctly but if TCP is used then the call is rejected. When this happens Incoming and externals calls cannot be made. Corrected via use of CMD 84-39-35 set to 1.

A10.00.50 Main Software

Reference	Description
F181210001	**Hotel Wake Up display languages**
	Setting Wake Up Call on a checked in hotel extension does not follow handset language for the following
F181212002	**Outbound IP PC Pro Connection - wrong user/password when using external connection**
	When using the Outgoing IP PC pro connection remotely, The PC pro tries to connect then reports wrong user name/password. If you set 90-69-02 to be an internal address instead of an external public IP address it works fine.

R9.00.60 Main Software

Reference	Description
F160513002	**Calls to VRS Attendant getting Ringback Tone**
	Intermittently callers are ringing inbound and would normally be answered by a VRS attendant.
F160520001	**V Mail dial options don`t work on supervised transfer**
	voicemail does not allow a breakout from the mailbox if the call is done using a supervised transfer to the mailbox
F160527003	**NetLink Sip Trunk redundancy problem**
	Attempts to use primary SIP trunks from secondary when Primary unavailable..
F160822001	**Inmail - Incorrect Voice Prompt in Greek Language**
	A Greek language prompt incorrect.
F161011003	**Caller List does not follow F-Routing**
	If a caller list number is called back, it doe not follow the F-Route configuration.
F171116001	**TAPI Redirect/Blindtransfer to an off hook standard SIP device**
	If perform a redirect or blindtransfer to a standard SIP device that is off hook with no call, the TAPI tries to route the call then disappears from TAPI.
F180315001	**SV9100 Does not send CANCEL Request when call ended**
	If clear the call within 1 Second (immediately on-hook after off-hook) the system does not send a SIP CANCEL. The carrier proceeds the call and it will ring at the target, disregarding that the calling party is already on-hook
F180413001	**re-INVITE fails when SDP version exceeds max value of 32-bit integer**
	If the 'version' parameter in the owner/creator field of the SDP body exceeds the max value of a 32 bit integer the SV9100 will reject the re-INVITE when call is placed on hold.
F180430002	**No CLI in Outgoing and Incoming Calls on analogue trunks**
	No Caller ID if an unknown parameter included into “Reason for absence of Calling Party Name”
F180430003	**Wrap-Up on TRF calls to another ACD Group**
	After receiving an ACD call and being Answered by an Agent that when they TRF the call to another ACD Group they complete the TRF they were not always presented with the Wrap-up in MyCalls.

Reference	Description
F180510001	**Registration fails on 1 profile causes other profiles to de-register.**
	If a registration fail occurs on Profile 1, the other profiles will de-register. This should be independant.
F180522001	**Call forward set to Trunk access code allows external calls can be made**
	A call forward set to 9 will give dialtone to the caller and allow them to dial externally.
F180601001	**Some Dutch Translations incorrect in InUC**
	Dutch Translations incorrect in InUC v8.00.59
F181102001	**SIP Trunks are occupied and in the state \`DISCONNECT INDICATION\`. After a few days**
	After a certain time all SIP Trunks are occupied and in the state \`DISCONNECT INDICATION\`.
F181207001	**Multiple Registrations on single port (SV9100)**
	Possible to register and call from multiple devices on one port.
F190102001	**SIP station ports locked**
	If you connect with PCPro application, configure a free station port with SIP password (15-05-16) and upload the data. With PCPro still connected register the SIP terminal, it will then operate. Make another change to 15-05 command, for example set SIP password for another extension, then upload modified programming changes. Disconnect PCPro. The first SIP device will not be able to re-register or make any calls.
F190308002	**Fully disable the ability to change the IP Address at Admin2 level in WebPro**
	IP Address of 10-12 should follow setting in 90-26 for Sys B level.

R8.50.52 Main Software

Reference	Description
F170703001	**Not possible to modify Type of number using 26-12**
	60080 Problem reported that it is not possible to change the Type of number for Called Party Number for outgoing ISDN calls using 26-12.
F171012002	**GCD-8LCF Analogue Extension Cards on SV9100 are down after version upgrade**
	GCD-8LCF cards are down after version upgrade to V7 The red and green LEDs are always on. Downgrading the main software does not restore the card.
F171116001	**TAPI Redirect/Blindtransfer to an off hook standard SIP device**
	If perform a redirect or blindtransfer to a standard SIP device that is off hook with no call, the TAPI tries to route the call then disappears from TAPI.
F180504001	**Using 2048 Bit Private Key Certficate & loading Webpages** **・#F181011001: SV9100 sends 'requir**
	Unable to load a 2048 bit key for certificate usage for TLS/Webpro/InUC
F180803002	**ISDN Trunks do not send the full subscriber number but only the DDI.**
	ISDN Trunks do not send the full subscriber number but only the DDI. Resolution is to copy the Number / Numbering Plan Byte from the incoming ISDN Message to the outgoing ISDN Message.
F181011001	**SV9100 sends 'require: timer' as answer to an Invite where 'timer' is not mentioned as supported**.
	If the SIP carrier sends a re-invite where \`Timer\` is not mentioned as supported (Packet 344) then the SV9100 sends in the 200 OK (Packet 358) a Header \`Require: Timer\` which causes the call to be disconnected by the carrier.
-	**F-Route Re-routing**
	Correction of the problem that part of the dial string is missing when re-routing of trunk (stepover) while using F-route.

R8.00.63 Main Software

Reference	Description
F170609001	**SV9100 sends Busy tone when SIP Message 404 not found is sent by the SIP Carrier for outbound call.**
	When an outbound call is made to an number that is no longer in service or incorrect the SIP carrier sends 404 not found, the SV9100 then presents Busy tone. It is now possible to send the SIP Out Of Range tone instead or Busy Tone via command 84-39-39 in PCPro (set to Mode 1)
F180530002	**ST500 TLS Outbound Calls Fail**
	If using a self signed certificate and register a ST500 client using TLS on iOS to the PBX, when making outbound calls from the ST500 client the ST500 Cancels the call after receiving the SDP payload from the PBX to negotiate the setup of the sRTP
F180606001	**Call Cut Off on SIP Trunks**
	SIP Trunk connected to an SV9100 is disconnecting by itself.

R8.00.59 Main Software

Reference	Description
F160329002	**Unanswered trunk transferred from IP terminal using VE set as DSS causes one way speech**
	IP terminal (210) has *03 (virtual extension) key for extension 200 on key. 210 has PRG15-02-07 set as 0 (hold) 210 has PRG15-02-21 set as 0 (DSS) Command 20-02-12 must be set to signal (1) 1. 210 answers trunk call. 2. 210 presses virtual extension key for 200 3. 200 does not answer 4. 210 retrieves unanswered call by pressing trunk key. 5. No speech id sent from extension 210
F161109001	**Translation error in the display by erasing missed calls**
	Wake up prompts P87, 88, 89, relating to wake up calls in Russian language missing.
F170313001	**Space added to number in Display Header**
	We have an issue seen on IP DECT devices that is being caused by the SV9100. Normally the IP DECT handset can store local contact information directly on the handset. Previously (SV8100) if a call came in to the handset and the number matched an entry on the phone the name information would be displayed. For the SV9100: the From header, the display part shows the number but there is a space added as a prefix. FROM: " 01159695700" Because of this space no numbers are matching in the IP DECT device and that is why it is failing.
F170609001	**SV9100 sends Busy tone when SIP Message 404 not found is sent by the SIP Carrier for outbound call.**
	When an outbound call is made to an number that is no longer in service or incorrect the SIP carrier sends 404 not found, the SV9100 then presents Busy tone, This is misleading. Can we send the SIP Out Of Range tone instead or Busy Tone. This has been raised for SIP Carriers that don't send their own announcements for number not in service.
F171031002	**Missed Calls log not showing in UT880**
	When using v1.1.27, on either the SV9100 or the SL1x00, if you get a missed call, it doesn't show in the logs page. Inbound and Outbound show correctly. This only appears to be an issue on this version. It is a requirement that the following commands, on a per UT880 extension basis, are set on the platform in order to store the missed call data. PRG15-02-13 LND-Outgoing mode – Set to Extension/Trunk Mode PRG15-02-34 CID List-Call Register Mode – set to Extension/Trunk Mode. Also requires MLC version 1.1.28 (video) or 1.1.38 (non video) above to resolve the issue.
F180212001	**UT880 MLC slowdown with numerous calls to virtual extensions**
	Site is configured with 8 x UT880 terminals. 5x UT880 terminals are configured in IRG 1. IRG 1 receives incoming calls for system. All UT880 terminals contain virtual extension keys for all other UT880 terminals. After numerous incoming calls the MLC performs slowly.
F180309001	**Hotel Wake Up display incorrect in German Language**
	Set hotel to German in 15-02 or with InHotel Set wake up call - display shows Wechruf ein this is correct. When the wake up call is made the display shows Good Morning This is in correct as its in English and should be the language that is set.
F180412001	**84-39-09 is not working as TAG should remain the same in both Invites**

Reference	Description
	Setting 84-39-09 to mode 1:Both should result in the sip trunk invites having the same sip tag in the from information. However, its been reported from site that when making an outbound call on sip trunk using a DT800 or DT700 keyset, the sip tags of the 2nd invite to carrier is taking the sip tag from the DT800 invite, and is not the same as the initial invite to carrier, resulting in the 1st and 2nd invite to trunk having difference tag information. This is incorrect and results in the call disconnecting.
F180511001	**Real Project - ISDN to ISDN / SIP to ISDN GW**
	Allow exceeding of 12 digits CDPN for SIP Proxy
F170113002	**Increase Virtual Slots from 128 to 240 in SV9100 system software**
	PC Pro not able to add any further cards when 128 virtual slots the maximum should be 240. PC Pro/Software for maximum card BRI Cards also needs to be increased from 25 to 46 combined CD BRI and PZ BRI.

R8.00.59 Main Software

Reference	Description
F180308001	**InUC screen sharing does Not work**
	During a Web Chat with users if you attempt to press the screen share button it automatically in Chrome loads the Chrome Extension page to install the NEC Extension, we can see the Extension is already installed as it shows the NEC logo in Chrome in the top right corner even before pressing for a screen share session. For some reason it always seems to take us the Chrome Store to re-install the App even though it is already installed.
F180312001	**Russian Wake up setting has no voice prompt.**
	Wake up prompts P87, 88, 89, relating to wake up calls in Russian language missing.

R8.00.57 Main Software

Reference	Description
F180116001	**BCT - consult-blind-transfer on Version 8 SV problem**
	If using a version 8 SV9100 8.00.51 in BCT mode (CMD 20-23-09 set to 2) the behavior of a consult transfer converted into a blind transfer has changed.
F170418001	**OAI locked status following Busy/No Anwer to Voicemail**
	When a trunk call is transferred to an extension that is call forwarded busy no answer to a UM8000 voicemail, the OAI status of the call doesn't complete when the call follows the call forward. This locks the visibility of the trunk and extension on BCT, as the OAI state is locked to previous call state.
F180104001	**The OAI status of the trunks used in an external conference lock, preventing visibility or use of the trunks in BCT**
	If a conference is performed using external trunks, then the trunks no longer give events leading to non-routing of calls.

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R8.00.55 Main Software

Reference	Description
-	**Improve memory management mechanism to prevent memory fragmentation**
	If using an InHotel with large database (generated from many customers/many stays) the memory area for running InHotel would run out of memory causing InHotel to stop.

R8.00.52 Main Software

Reference	Description
F150828001	**Cannot Forward enough digits from TAPI**
	If you preform a lineForward in TAPI to more than 11 digits it only takes the first 11, so cannot forward internationally. This is possible by manually dialling the service code.
F151008001	**Stacking multiple ETIA Blades to function as one switch**
	2 ETIA blades in a chassis. The 2nd ETIA is configured as Add-on In ETIA web configuration we do not see the 2nd ETIA ports 9-16. Cannot access 2nd ETIA via web either.
F151116001	**Transfer call to Busy SIP extension fails**
	On the system if you enable call waiting you can transfer a call to a busy destination and it will queue to be answered. This works for all devices except for a standard SIP extension. When this is used and a transfer takes place the PBX will send a cancel request to the terminal and the call will recall on the transferring extension. Should it not do a refer or update to transfer the call. This works for all other devices so appears to be a software issue. Note the following conditions apply to this fix: 1) The implementation of the current fix requires 20-13-54 Call Waiting for Standard SIP Terminal to be disabled. This means you have to choose between using call waiting at a standard SIP device or being able to receive transferred calls at a busy SIP device (this fix) 2) Transferred calls do not pass through the CLI as seen on other handsets. MLT displays the transferred CLI but SIP device only receives anonymous from the PBX. PRG99-03-51 must be set to 1 (Enabled) through TelPro for this to work
F160426003	**TAPI issue when transferring to external call (Connected State)**
	When an external call is present which is transferred to an internal party It is incorrect on 2 key aspects. 1) When transfer is made (by placing on hook), the call at 4601 stays in Connected state, it does not change to Ringback (in the same way an internal call does). 2) The ConnectedID shows 4601 rather than the external number (2600) See traces Requires system data (20-23-09) to "Mode 2: for BCT", this function is effective.
F160624001	**TAPI stuck call if transfer a call when already ringing**
	If a call is transferred to an extension that is already ringing from another, this results in a stuck call. Requires system data (20-23-09) to "Mode 2: for BCT", this function is effective.
F170303004	**SIP Trunk Layer 4 protocol setting TCP is ignored**

Reference	Description
	if CMD-10-28-3 is set to TCP, the communication for the SIP trunk is sent via TCP Protocol. Additionally we have to set 84-14-11 to “SIP UA Domain” to generate the correct host part. Unfortunately the SV9100 does not use TCP in combination of the two settings above, but only UDP. We do need a correct TCP handling for the SIP Trunk
F170303005	**SIP Trunk DNS Mode forces TLS if DNS SRV Records are offering SIPs (PRG84-39-35)**
	If I change the DNS Server between one that replies with srv records and one which supports only A records, you have to reboot the system. If once SRV records have been resolved, the system will continu with srv but from the new DNS Server… After a reboot the DNS queries are different. OK, here is the test… I uploaded the attached config and rebooted the system.. * 84-39-13 is set to 1. * 84-14-11 is set to “UA Domain” * Correct DNS Server After the restart the system starts DNS with NAPTR and finally swaps to TLS. I cannot check the content anymore..
F170821001	**Busy Lamp for MDG fault**
	DSS key call to MDG, results in DSS staying lit.
F170922001	**Unable to blind transfer to Mobile Ext from TSP Application**
	When the call is TRF to the Mobile Extension it doesn`t ring Mobile Ext and the call disappears
F171005001	**Increase the session ID on re-invite from SIP carrier when the SDP Header is not included**
	This is resolved using this version 7.00.65 with carrier choice L
F171026010	**Missing RTP on peer to peer calls on SIP Trunks**
	Missing RTP on peer to peer calls on SIP Trunks
-	**Upgrade of Main Software requires re-install of InApps**
	Upgrades on previous version required a re-install of InApps (Config and Data were unaffected). This version does not have this issue
-	**Improved Application Manager**
	If a browser is left open to the Application Manager, it will now stop polling after 10 mins, making the assumption, it is no longer in use. A manual refresh of that page will then update. Updated Logo.

Reference	Description
F170920001	**SV ignores Invite with SDP parameter in the 180 ringing**
	SV9100 does not analyze SDP parameter in the 180 ringing so the system ignores the message. After this, out of range timer expired and then the system disconnected the call

R7.00.52 Main Software

The following externally reported problems were solved with this upgrade:

Reference	Description
F170303002	**E164 format now working with 84-39-01 mode settings with incoming invite**
	When e.164 modification is required for incoming calls where the CLIP needs to be used from the From header (84-39-01 mode 2) or the PAID header (84-39-01 Mode 4) the number is not being modified to include the 00. . E.164 added to all 84-39 options.
F151019001	**Call Forward 302 Return**
	When command 84-14-17 is set to 302 return for SIP Trunks incoming calls to divert to an alternative external destination when service code 848 1 is set we have found that this also effects forwards that are set to internal destinations using the same service code. Internal forwards no longer follow the setting in 84-14-17.
F160519002	**Application manager not accessible if # used in password**
	If '#' is included in the system password then it is not possible to access the manager.cgi

R6.00.67 Main Software

The following externally reported problems were solved with this upgrade:

Reference	Description
F170123002	**Change \`Anonymous@Anonymous.invalid\` in the FROM header to be all in lower case**
	Command 84-39-27 allows the first letter to be set to either A or a. When set to mode 1 send anonymous all in lower case format
No ref	
	Deutsche Telekom: Session number and version ID needs to be updated
F170106003	**No ring back tone for call from SLT over CCIS**
	When a call is originated from an SLT over a CCIS connection between 2 SV9100 systems the caller does not hear ring back tone. The problem does not exist with key telephones.
F160405002	**When daylight saving is set with lock key system reverts to auto night setting**
	Example : System has daylight saving set for last Sunday in March 27th System has auto night service set for Sunday 27th mode 2 System has holiday mode set to mode 5 for Friday 25th and Monday 28th System has #07 key set and is activated on Friday 25th system is now locked in holiday mode At 02:00 Sunday 27th the daylight saving is activated and the system moves 1 hour forward At this point the system goes to mode 2 This is incorrect it should stay in mode 5
F150219001	**Trunks are not busy when keep alive threshold is reached**
	If the failover timer limit is reached, even with a failover timer set, the trunks do not display as busy as expected. With this release CMD 84-39-21 can be used, the smaller the value the more often it registers so the less time for a possible outage
F160919001	**No speech path after transfer with auto record set**
	A DDI call rings an extension. When this extension transfers the call, there is no speech after completing a supervised transfer (neither party hears anything) This problem occurs regardless of whether the call is held with he Hold-key or the transfer key. This problem only happens when transferring from a Digital handset to and IP handset and when voice recording is turned on.
F170316001	**Using 15-12 or key 78 for recording loses speech path after hold**
	How to reproduce ;- DT700 Extension 212 has 15-12 set to voice mail and automatic recording set extension 212 makes outgoing call and the call starts recording Extension 212 puts the call on hold Extension 212 retrieves call with line key Call is ok at this point Extension 212 puts the call on hold again Extension 212 retrieves call with line key No speech path This also happens using Key 78
F160614001	**Bad sound quality when Automatic Recording is enabled**
	Bad sound quality from calling party to DT700/DT800 handsets only if automatic recording is enabled.
F150930002	**Missing soft keys for CID list on DT handset set to Italian**

Reference	Description
	The DT400 display includes some English words although the key telephone has been set to Italian. The DT400 display in Italian is a lack of following characters ;-STORE-DEL and the up and down arrows
F161116001	**Dialling timeout for Mobile extension on SIP trunks**
	When dialling an extension which is call forwarded to a Mobile Extn it can take more than 10 seconds to dial the number. If you dial the speed dial directly it dials out normally
F170315003	**SMDR incorrect for an abandoned IRG call**
	Example : Incoming DDI Call is routed for the Pilot Number 0 at 2015-04-15 10:01 from Caller ID 02022832970 at Trunk 15. The Call is routed to IRG 22 via conversion Table 742 (see attached DIM Trace) After 3 seconds the caller clears the call and the SMDR generates a RING NO ANSWER call log. This SMDR log is as follows: 26 IVIN 10:01 015 0 1163 02022832970 0:03 NO ANSWER The SMDR indicates that Extension 1163 did not answer the call which is incorrect. Extension 1163 is located at port 2 which was neither ringing for the call nor a member of IRG which rung. This incorrect reporting is resolved in this release
F170308001	**No Mobile Ext Progress Tone when calls are forwarded on SIP Trunks**
	No progress tone on mobile extensions when routing out via SIP Trunks. Progress tone works fine on ISDN When a user dials the Mobile extension number internally there is no progress tone while the number is being dialled just silence until Ring Back Tone is heard. When testing with ISDN progress tone is heard immediately. This is issue is resolved in this release

R6.00.60 Main Software

The following externally reported problems were solved with this upgrade:

Reference	Description
F151216001	**Handsfree Misdialling Problem in UAE**
	If dialling handsfree, not all digits would be sent to line.
F160122002	**T.38 problem with fax traffic**
	Large faxes can have errors
F160209001	**SIP Profiles not working correctly when used with Networking Mode**
	When using multiple SIP Profiles with networking mode the outbound call always routes as per the first dial number match as per command 10-23-04. 10-23-04 dial number set as 0 for profile 1 in table 0001 10-23-04 dial number set as 0 for profile 2 in table 0002 Trunks assigned to extensions and either profile will always look at the first dial number match in 10-23 which is incorrect instead of routing to the correct profile. CMD 84-39-23 requires enabling to apply this fix.
F160503001	**Issue when Outgoing calls are placed on hold by the SIP Carrier as the SV9100 is sending a BYE.**
	In SIP Call 1 the SIP carrier sends a re-invite when the call is placed on hold (external side) that we respond with 200ok and then send a BYE
F160629001	**SLT Phone continues ringing when IP DECT uses continue dialing feature**
	When IP DECT dials SLT extension and an extra digit for example 0 for TAC the SLT Phone continues to ring but DECT access Trunk (0). This problem only happens when 15-03-09 is enabled for CLIP detection so the SLT receives a burst of FSK before 0 is recognized on the PABX.
F161109002	**TLS function on HTTPS port causes system reset or stop**
	System restarts once or twice a day. R5 main software, uses inGuard app via HTTPS port

R5.00 Main Software

The following externally reported problems were solved with this upgrade:

Reference	Description
F141104001	**Unable to end a conference in UC Suite**
	When the UC Suite full client is on a conference call, the hangup button does not end the call.
F150907001	**Unable to access system programming 15-01-XX for individual (SLT) extension, connected via APR-LA unit.**
	Unable to access system programming 15-01-XX for individual (SLT) extension, connected via APR-LA unit. Customer connecting an EFTPOS machine and requires to setup “Outgoing Trunk Line Preference” in PRG 15-01-02, but unable to access this command via WebPro or PCPro.
F150910001	**Auto Wrap Time does not kick in when transferring an ACD call to another user**
	If an agent transfers an ACD to another location upon releasing the transfer a new ACD Call is delivered rather than entering auto wrap up.
F151209001	**Ringdown using ARS and F-Route Tables**
	Phones that are assigned 9 as the Hotline Destination in 21-11 do not use Priority Number 2 in F-Route Tables when ARS is enabled and the Service Type in 26-02-02 is defined as F-Route Access with the appropriate Additional Data defined in 26-02-03.
F151209003	**Outside caller accidently transferred to unintended internal party.**
	Outside callers when placed on hold are being transferred to an internal station inadvertently.
F151214002	**TAPI Driver Fails to Initialise**
	Sometimes when the 3rd Party TSP is re-initialised by restarting the PC running the driver, the TSP doesn't re-connect to the SV9100 first time. Occurs more often if Toll Fraud Guard running. If stop this TAPI initialises ok more often. Then can start Toll Fraud Guard OK with no impact on TAPI.
F160309003	**DT820 does not support hold down Exit key to logoff hotdesking**
	When DT700/800 manual logon mode is enabled, the DT700 and DT800 terminals can hold the Exit button as a shortcut to logoff the system. This does not work on the DT820.

R4.0.55 Main Software

The following externally reported problems were solved with this upgrade:

Reference	Description
F141023001	**Station keeps ringing after outside party hangs up**
	On an incoming call from a SIP trunk with a specific carrier in Guatemala (Claro), if the caller abandons by hanging up, the called station keeps ringing. When the caller abandons, SIP carrier sends CANCEL message and replies with 481 Call/Leg Transaction does not exist.
F150219001	**Trunks are not busy when keep alive threshold reached**
	When setting the keep alive options that if the failover limit is reached the trunks do not display as busy as expected when making calls. With the carrier registration I would expect this to occur as a new re-registration has not taken place and also with the internetworking if a keep alive is not responded to up to failover limit then trunks should be busy also for system to be able to re-route calls if necessary.
F150901001	**SIP keepalive takes 5 minutes to react**
	When the carrier stops replying to SIP keepalive messages, should re-route outgoing calls to a different trunk group according to programming. This happens, but it takes 5 minutes to happen
F150922001	**Getting one way audio when a call is forwarded off site on SIP trunks**
	1. forward VE 201 to an offsite number 2. Call in on a SIP trunk. 3. blind transfer to VE 201. Notice that the original outside party hears ring back. 4. Answer the outbound forwarded call. notice that the original outside party is still hearing ring back
F150928001	**PRG 20-11-30 not being followed**
	1. In PRG 20-11-30 uncheck for CoS 1 and 15. 2. Call forward B/NA 101 to 102. Notice that it does not show that 101 is FWD BNA>> STA 102.
F151007004	**SIP Carrier Type L only shows Calling Number not Calling Name**
	Calling Name and Calling Number, which we understand isn't supported with Carrier Type L.
F151027003	**Last Redialed number is not shown when redial key is pressed.**
	Pressing the Redial Key the display does not show the last dialed number.
F151119002	**Ringing Virtual Extensions keys for Single Line Stations still ring when DND All is enabled.**
	Phones that have *03 Virtual Extensions keys to cover calls for Single Line Extensions still ring when the phone that has the Virtual Extension key is in Do Not Disturb.
F151104001	**SIP Trunk Option Keep Alive using SIP Interconnect (TIE) not working**

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Reference	Description
	The SIP Trunk Option Keep Alive from command 10-23 does not work. Using SIP Interconnect (10-23). The overflow in 44-05 is set to use ISDN PRI. It does not Overflow.
F151120004	**DSS Consoles connected to IP terminals Page (Feature 95) key does not chgange color on page switch.**
	Page Switching button the pages switch correctly however the button color never changes. Page 1 RED Buttons 1~54 Page 2 Green Buttons 61-114 The Feature button 95 (Page Switching) always stays RED. Note if the console is set to page 2 buttons 61-114, then the console is reset (power cycled) the correct color green will be displayed when the DSS is back on-line
F151214001	**Problem if PRG 84-39-03 is set to 1**
	If PRG 84-39-03 is set to 1, the host part of From Header in Invite is changed to the value of PRG 10-12-09
F160223001	**iSIP TAC issue**
	SV9100 R4.00.50 did not have the Trunk access code for ML440 changes in.
-	**Wrong dialing issue**
	DTMF can be echoed on line with handsfree. 99-03-76 to 1
-	**SIP Date header format issue**
	Outdialling issues on Vodafone DE during 1st to 9th of month
-	**SV9100 send the 488 response when the SV9100 receive the invite for Hold**
	During an invite for Hold, the SV9100 could give a 488 Not Acceptable Here response.
-	**SIP SLT IP Terminal Stops Ringing when targeted by DDI in IRG**
	The problem is when having a DDI pointing to IRG in Target 2 then Target 3. When a SIP extension is a member in both targets it only rings when Target 2 IRG is hit and not when Target 3 hit.

R4.0 Main Software

The following externally reported problems were solved with this upgrade:

Reference	Description
F120302001	**Cmmd 10-02-05 adds trunk access code to so port in TAPI**
	Cmmd 10-02-05 adds trunk access code to so port. If the trunk access code is added to cmmd 10-02-05 this will add trunk access code to so ports .This adds them to loop back ports. This causes a problem when IP DECT and loopbacks are used on the same system
F121009001	**Romanian Language Support - Terminal display +1 day (e.g. Saturday displays as Sunday)**
	Saturday was displayed as Du rather than Si
F141211001	**Alarm 64 showing when 10-12-09 is changed and the IPLE is not fitted**
	Where the IP address in 10-12-09 is changed and the extension is enabled to show alarms in 90-50. If you change 10-12-01 then no alarm is shown. If you change 10-12-09 and come back out of programming the alarm will appear on 200.
F150224001	**VoIP Channels not displayed in Feature Activation correctly for a default system**
	The system comes with 4 IP Trunk and 4 IP Terminal licenses. This means there are 8 VoIP channel licenses. If you Feature + 4 on a terminal it will show 8 resources available. However if you use PC Pro or Web Pro to check this through Feature Activation then it reports there are 0 VoIP Channels. This is misleading.
F150703001	**Mobile EXT`s Engaged after Reboot**
	Issue related to counting of port types for System Capacity. Was counting Virtual Loopbacks and therefore causing issues on boot up when counting from fresh again. No longer counts loopbacks.
F150820004	**ACD Calls not presented to Agents when AIC in use**
	Problem occurring when AIC ACD agent is logged into queue. If call or calls start to queue when ACD extension is busy then the new incoming call is stuck in the ACD queue and not delivered. Problem does not occur when ACD login mode used
F150813001	**Unable to load Wav file from SV8100 to SV9100**
	When loading an SV8100 wav file to the SV9100VIA user pro VRS upload the file will not play This same file plays ok on SV8100
F150930001	**System Lock up caused by no vacant key for Incoming Trunk Call**
	There are 3 extensions in the same Department Group Make Extension A busy (so that an incoming trunk call is not allowed to ring) An incoming call 1 arrives at Extension B. An incoming call 2 arrives at Extension C.

Reference	Description
	Extension B seizes another trunk and leaves Call 1 ringing. Extension C answers the call 2, puts it on hold and goes on hook. When the call no answer time for Call 1 expires the system locks up.
F151007001	**MOH does not change until reboot**
	System is set with MOH set to source VMDM and tone 1. Change to tone 2, but VRS message 1 is still played on MOH. After a reboot tone 2 is played on MOH. Reboot should not be needed
F160104001	**Crosstalk on Netlink call via Loopbacks with MOH on VRS**
	Transferring a Netlink call via Loopback causes either silence or the speech from another call to be presented to the caller instead of MOH

R3.00.54 Main Software

The following externally reported problems were solved with this upgrade:

Reference	Description
-	**SIP Profiles 3-6 not shown in Webpro**
	Only Profile 1 and 2 were shown (in the same way displayed as in R2).
-	**Memory leak possibility with Toll Fraud Guard running**
	Memory release has been made more efficient in this version.

R3.00.51 Main Software

The following externally reported problems were solved with this upgrade:

Reference	Description
F150602001	**Loop Back Lock up**
	In the following scenario it was possible to lock up the loopback ports. Loop backs to call each other with the final call going to IRG 1 Call is made to 220 (first loop back port) this goes through all loopback ports and ring IRG 1 Extension 200 is in IRG 1 and rings Call is hung up to 220 but extension 200 still rings you have to go off hook and back on to clear call Call is made again to 220 this no returns busy I have now set the firt 17 loop backs to be used and the 17th goes to IRG 1 220 is called and the call is presented to IRG 1 237 is call 18th loop back and busy is returned
F150409001	**Netlink with DT700/800 registration to the secondary node not supported**
	It was not possible to connect a DT700/800 to a secondary node.
-	**Unsupported Extension Types in PcPro**
	If a config contained an MH240, PCPro would crash on pages relating to extensions

R2.00.60 Main Software

The following externally reported problems were solved with this upgrade:

Reference	Description
N/A	**SSL Poodle vulnerability**
	A fix for a known SSL POODLE vulnerability. The SSL protocol 3.0, as used in OpenSSL through 1.0.1i and other products, uses nondeterministic CBC padding, which makes it easier for man-in-the-middle attackers to obtain cleartext data via a padding-oracle attack, aka the "POODLE" issue.
N/A	**SIP Trunk – Asterisk Pedantic Support**
	TAG Parameter contained in Initial invite and following Invite with authentication info have the same value if 84-39-09 is set to 1.
N/A	**SIP Trunk – E.164 issue**
	Caller ID information in P-Asserted-Identity header contained in the Invite does not follow E.164 when the call is made as 'Private Call'
N/A	**SIP Trunk – Timestamp Header Issue**
	A fix for a problem if a Timestamp header has a large value in Invite sent from a Carrier, the call is dropped because the SV9100 cannot handle it properly. The main software has been changed to ignore the Timestamp header.
N/A	**System crash problem on PMS Interface**
	A fix for a problem that SV9100 system crashes if a Morning Call is set via PM interface.
F141222001	**Unable to connect to SV9100 using ISDN remote method**
	Connection is dropped during "Verifying username and password"
F150203001	**VRS Auto Attendent not routing when using search extension in multi level mode**
	If you configure Auto Attendant to have an option to press # to route to the next auto attendant level the VRS accepts the first dialled digit and routes the call to the matching option in the initial single digits options. Example: An Auto Attendant Level 1 has a single digit option setup to route to a Level 2 announcement. An incoming caller selecting a single digit option within the Level 2 announcement will be routed via the equivalent single digit option from the Level 1 setup; the Level 2 options are not available. Note -This fix was included within the previous release of SV9100 Main Software R2.00.60.

7. KNOWN PROBLEMS

The following are not problems but are listed to for awareness.

None

8. SECURITY

All ICT installations are at risk of unauthorized intrusion and subsequent misuse. Such intrusions may result in significant losses to the company affected, including but not limited to financial liabilities, data privacy breach, intellectual property, material assets and associated labour or legal costs.

NEC products contain a variety of features designed to help prevent and combat such misuse. To assure their effectiveness it is essential that such features are configured, deployed and maintained in an appropriate manner by the installing party in consultation with the user of the equipment.

The ultimate responsibility for assuring the overall security of the ICT installation resides with the using company. The effectiveness of their security measures depends on the quality and rigorousness of implementation of their security policy by ICT administrators and their user community.

Information about the security features in NEC products and how to configure them is contained within the product documentation.

This version contains additions/amendments to security features.

9. MATERIALS

9.1. Physical Distribution

N/A

9.2. On-line Distribution

Any software related to this release can be downloaded from the software database on BusinessNet. http://businessnet.nec-enterprise.com.