Known Issues

Please contact Samsung Technical Support with any questions or comments concerning this list of Known Issues.
1(800) 737 - 7008 or bcs.support@samsung.com

September 2010

Ref # Issue Name	Issue Type	Issue Description	Workaround Description
P100719004 - 1282 **ITP 5112 phones 2nd Call follows Ring Volume**		On the ITP 5112 phones if a user is on a call, the call-waiting notification comes in, the call volume of the existing calls is lowered or raised according to the volume level of the ringer.	New software itp T3.55 Software fixes this issue. Contact Technical Support until official release
TS08172010 - 1295 **Ringing stops on SMT IP phones**		When making a call from a SMT ip phone and another call comes to the station while making a call to a internal station the ringing indication stops. 7400 v4.46d SMT-i5210 & SMT-i5230 v1.07 SMT-i5243 v1.64 Ext 3201 calls 201. Ext 202 calls 3201 during the time ext 3201 is hearing ringback from ext 201. Ext 3201 will hear the beep for a second call. Ext 202 decides to hang up and not complete the call to ext 3201. Ext 3201 should continue to hear ringing if 201 hasnt answered the phone or forwarded. At this point there is no ringing.	New software for all SMT-iPhones posted on GSBN website corrects issue
TS08242010 - 1297 **SMT display-MMC 103 affects whats displayed**		All SMT phones except the 5243, when put into either Voice Announce or Auto Answer (MMC 103) when idle, the display shows that mode not the station name. Does not happen in ring mode or on TDM or ITP phones. Ex: Ext 2001 has the name Joel Kin the top line, day, date, month and time on the bottom line. Turn on a condition other than ring in MMC 103 Ext 2001 has = Auto Answer= top line, day, date, month and time on the bottom line.	No work around
TS08241001 - 1298 **NND feature while doing auto record cause lost calls**		Tested on all 3-7000 series phone system Tested on sw v4.46d (released) and v4.51(beta) MMC 701 COS1 useable feature VM REC and VM AREC set to YES MMC 743 ext 201 set to auto record to mailbox 201; both inbound and outbound traffic. MMC 722 assign NND to ext 201. Note: softkey can be used too. MMC 414 assign called id analog trunks, using PRI nothing to turn on fails using them too. Ext 201 answers a incoming call, auto answer feature is activated. Ext 201 user presses the NND no problem they can see CID info. When the screen times out from them pressing the NND it goes back to normal (conf page mute). If ext 201 user press the NND button again they have audio until the display returns to normal. At this point the inside and outside parties hears silence. The call isn't put on hold because there is no MOH heard. Ext 201 can press the call button or DT and they can't retrieve the call. Ext 201 will have to press the hold button on their phone and then go back off hook to get their call back.	No work around

Ref # Issue Name	Issue Type	Issue Description	Workaround Description
TS08261001 - 1299 **SMT forwarding feature appending digits**		SMT-5210 & 5230 Duplicated it by doing the following: 1) Set foward all to 918007377008 number using SMT fowarding Feature --- 2) call your fowarding station to make sure fowarding is working --- 3) then cancel fowarding by dialing 600 feature code --- 4) Set foward all to 91234567 --- 5) Press save. Now the phone will Display fwd 912345677008. If you don't do these step by step you will not be able to duplicate.	New SMT-iPhone software posted on GSBN corrects this issue
P100324002 - 1267 **The PRI will not synch up to DMS100 and some 5ESS**		OS7100,OS 7200, OS7200-S You can do the follwoing steps too have Telco recover the PRI circuit if it goes down. 1. The Telco Technician will need to access their GUI interface to the DMS100 SYSTEM. 2. They will need to access the carrier level with the following commands. MAPCI MTC TRKS CARRIER POST DTCI 10 6 BSY 0 RTS 0 FORCE 3. THIS COMMANDS WILL bring the PRI circuit back into service.	OS7100,OS 7200, OS7200-S There is a new TEPRIa software load v4.27 or use the older Tepri v1.07 card You can also do the follwoing steps too have Telco recover the PRI circuit if it goes down. 1. The Telco Technician will need to access their GUI interface to the DMS100 SYSTEM. 2. They will need to access the carrier level with the following commands. MAPCI MTC TRKS CARRIER POST DTCI 10 6 BSY 0 RTS 0 FORCE 3. THIS COMMANDS WILL bring the PRI circuit back into service.
TS71320102 - 1283 **Page group with 36 or more phone causes system reset.**		If you have 36 or more phones in a page group and do a page all it will reset the system. I have tested it on 4.45a and 4.46d, in default. To duplicate, assigned 36 phones to all page group using MMC 604. Start doing "all page" using 55*. This is happening on the 7200 (MP20S) & 7100 (MP10A).	Either lower to 35 the number of sets paged at one time or request T4.46e for 7100 and v4.51 MP20S software from Tech Support
p100724001 - 1284 **Conference squeal on 3 party conference call from a analog (SLT) phone.**		Initiate a conference from a single line phone using 3 analiog trunks. Caller initiates or recieves the first trunk call. The SLT party then initiate the first conference call. At this point theres 1 internal and 2 outside trunk calls in progress, no squeal. They then initiate the 3rd outside trunk into the conference. Once all 3 (trunks) are brought together the squeal occurs.	Contact Technical Support for v4.51 software until official release
TS71320104 - 1285 **7200 expansion cabinet reseting**		In a 2 cabinet configuration, if the expansion cabinet contains 34 or more DLI ports (phones plugged in) and all page is perfomed the second cabinet will reset. It may take more than one all page to duplicate this. 7200 w/MP20.	Contact Technical Support for v4.18 LCP until official release

Ref # Issue Name	Issue Type	Issue Description	Workaround Description
P08092300 - 1164 **Disabled CID still shows on 2nd Call as CW: ###-###-####**	500 R2	Caller ID information of a second call at a station with CID disabled in MMC 312 will still display "CW" and the incoming phone number even though caller ID is disabled for that station.	Upgrade to 4.22c to resolve this issue for 7000 series. PIL is still open against 500 system.
P081202002 - 1183 **Usinga PIC code for international dialing is failing to connect**	500 R2	Using a PIC code and then dialing international phone numbers will not connect on current software. The digit string is too long. This fails on 500R2, 100R2, 7100, 7200, 7400 but has been fixed in the software on the 7000 series.	There is no workaround at this time. You cannot use PIC codes for international calling on our 100 and 500 systems, but is fixed with 4.42 software on our 7200/7400.
TS20090303 - 1200 **VT Transfers will intermittently contain extra digits when sent to voicemail.**	500 R2	This problem has been seen across all platforms that support the E series VM. We have not been able to duplicate the issue in the lab, but we get enough calls on the issue that we know it is a real issue. We are opening the PIL to request assistance from product development on getting a handle on this issue and to track the issue. Intermittently we will see extra digits being inserted in front of the extension number when a VT transfer is made from the phone system to a subscribers mailbox. Most commonly we see the digits "#" or "2". In the event of a # it will prompt the outside caller for their password. In the case of a 2 it will route the call to the main menu under an error condition. In call cases the workaround below has corrected the issue. However this is only a workaround.	On page 2 of the menu input processor of both routing menus (forward trunk and forward station) place the following entry (assuming that your extensions are 3 digit; if you have 4 digit extensions this will not work). #??? TRANS ???. 4.30i and 5.3.3.5 resolve this issue for the 7000 series. We are awaiting a software fix for the 100/500 series.
TS0908092 - 1222 **7100 - MP10a - 4.30i - AMIS Protocol Error**	7100	Set up two systems from default and programmed minimum programming needed for SPNet and network mailboxes. I left a sample message in system A network mailbox. system A calls system b and begins transmission. Just after the message transmission on system B you get: Invalid end of message command-	There is no workaround at this time.
p100523002 - 1274 **MP10 voicemail when playback with garbling - rewind, pause, and forward.**	7100	7100 - MP10 - sw V4.46c setup: press #, # to record/leave a message to your mailbox then playback by verifying from the Rewind (press key 7), pause (press key 8), and Forward (press key 9) audio is garbled	There is test software vT4.46F available, contact Technical Support
TS0720101 - 1287 **7100 MP10a default/initialization process**	7100	Proper initialization of MP10a, 7030, MP20 & MP20-S The procedure for all 3 processors should match whats in the documentation MP20 and MP20s.They state "after 3 minutes of RUN LED and SM LEd activity, press and hold RESET SWITCH for 10 seconds".	Wait 3 minutes before pushing reset switch on MP10a card.

Ref # / Issue Name	Issue Type	Issue Description	Workaround Description
p100910003 - 1300 **IT Tool MMC 831 public rtp port update**	7100	OS7100 with OAS card. IT Tool v1.46d Configuration 7100 MP10a, UNI card 4TRM & 4DLM, OAS card Start system up and do a proper initialization Log into the system using KMMC. MMC 831 shows the public rtp port as 30,000 for embedded MGI channels and OAS card channels. Log out of KMMC and use IT Tool to log in. Search for MMC 831 (2.2.2) the public rtp port show as 30,000 for embedded MGI channels and 65535 for the OAS card channels. Now log in thru KMMC and change the public to something other than 30,000 (ex: 30,333), reset card for it to take effect. Log out and log in using IT Tool. The public rtp port show as 65535 instead of 30,333. It doesn't matter if you change it thru IT Tool it will show the same. The changes thru IT Tool does take effect on the system but if you refresh it will still display 65535 in IT Tool.	Log into system using KMMC instead of IT Tool
TS03231001 - 1266 **Cracking and Popping VM**	7200	OS7200-S, MP20S, sw V4.45a Received customer DB files and load to the 7200-S system by default. Setup w/ 1 MP20S, 1 16-DLI2 (slot1), 8-TRK (slot5) - leave message to x2002 VM from x2001 and listen to the cracking noise. - leave message to x2001 (itself) VM can still hear some cracking noise. - Called from outside through the trunk will be able to captured the cracking noise as well when left the message to the VM of either x2001 or x2002. note: using MMC 102 to set the No Answer for call forward between the station) This cracking and popping sound is very very light and almost un-noticable in the background.	Contact Technical Support for V4.46e software
p100523001 - 1271 **Call Centrex transfer failed on trunk 5-X on v4.46s for MP20 - 8TRK2 and 16TRK**	7200	OS7200 - V4.46a vs OS500 default the system before the setup 1. on OS7200, setup slot1 for 16 DLI2, slot2 for 16 TRK, slot3 for 8 TRK, slot4 for 8 TRK2. 2. on OS7200, connected a 18D keyset with x201 (incl X2006). 3. From OS500 (sw V2.76a), connect a iDCS 28 keyset to 16DLI w/X2018, another 18D keyset to 16SLI w/X2001 4. Operation: X2018 call to X2006 w/audio then press FLASH key on X2006 to transfer the call to X2001 (X2006 can hang up after X2018 and X2001 communicate each other) with audio when the SLI trunk cable was connected to the port 1 – 4 of the OS7200, TRK2 on slot4 or port 1 – 4 of 16 TRK slot2. BUT there was no AUDIO at all from the port 5 – 8 on 8 TRK2 slot 4, and port 5 – 16 on 16 TRK slot2. Note: This issue was appeared on V4.42 and found on V4.46a as well.	The problem fixed in V4.51 software contact Techniocal Support for software until release
- 1289 **UCD supv. dial tone on SP Key after page to group**	7200	Default,(MP20, 16 DLI), ver 4.46d; 3 phones o MMC 601–grp 501 set as UCD (201,202,203). o MMC 604-Page grp 0 – (201,202,203). o MMC 722 Station 201 button 3 (SP501). Using station 202 make a int. page (55*), hang up Press Button 3 (SP key) on station 201 Internal dial tone can be heard from station 201	Remove the Supervisor with the SP key from the page group. Soon to be released s/w 4.51 corrects the issue, contact Technical Support.

Ref # Issue Name	Issue Type	Issue Description	Workaround Description
p100910001 - 1241 **Non Verified Account Codes with Toll Overide can prevent from using Conf Calling**	7400	7400 on 4.30i software - If you turn on non verified account codes and program two phone numbers for toll overide, then you can dial one number, but if you try to conference in the other, it will fail.	Make the first call, put the call on hold, make a second call, press conference button or softkey, press the call button for the first call you put on hold and press the conference button to bring the two calls together.
P100105003 - 1247 **4.42 software doesnt allow Voice Announce - VA will have 2 way audio like AA**	7400	On 4.42 and 4.42 software, if you set a telephone set for Voice Announce and call it, it will have two way audio whereas it should only be 1way until you pick up the call.	There is no workaround at this time.
P100217001 - 1256 **Executive Mobex user disconnected after placing a call on hold.**	7400	Executive Mobex user disconnected after placing a call on hold.	There is no workaround at this time.
P100522004 - 1279 **No Audio on VM AME on IP phone**	7400	OS7400 V4.46d Setup: MMC102 Call Forward station NA to SVMi go to MMC301 assign COS to this station go to MMC 701select 'YES' for VMAME for that COS go to MMC 722 and assign a VMAME button Enable VMAME feature with button Call the station and let it forward to VM. You will see the station display showing that VMAME is working, but no audio from speaker. This same feature works fine on a TDM phone with AME enabled.	There is no workaround at this time.
P100813001 - 1292 **MMC 119 Group Name option display**	7400	Tested on current 4.46d software MMC 119- Ring Line 1 is set group name first MMC 601- Set group XXX or XXXX to have members: Group 5001 members 201,202 MMC 602-Assign group name MMC 714-Assign DID XXXX to ring group: 1234 RP1-5001 RP2-5001 etc. MMC 406-Assign trunk XXXX to ring group: 7001 RP1-5001 RP2-5001 Make a call in on an analog trunk, the group name will display on the phone and it stays there. Make a call in on a PRI trunk, the group name will display on the phone for a split second and then the caller id information is displayed.	No work around

Ref # / Issue Name	Issue Type	Issue Description	Workaround Description
p100824001 - 1296 **Dataview agent idle but status is showing agent in ABW state**	7400	-The agent is in ABW. -The agent makes in intecom call to the supervisor. -While the agent is in conversation with the supervisor, the agent then presses ABW to remove themselves from ABW state. -The agent hangs up the call and is now idle. - Now the agent is out of ABW and able to take calls, but the dataview status screen will show the agent stuck in ABW state. The agent must now log out and back in to the UCD group to clear the ABW on the dataview screen. Tested on 7400 with 4.46d software and Dataview V1.5	Log out and back in to the UCD group.
N090730013 - 1210 **Account Code - Non Verified - From SLT**	7400	Forced Account Code - Non-Verified 100 R2 2.76 500 R2 2.76 7200 4.22 7400 4.22 Currently it is not possible to make a call from an SLT port using a non-verified account code. On a digital set this can be accomplished by assigning a "ACC" key with an extender of "000". On a sing line set if you use the feature code 47 and enter 000 for the bin number you will get an error tone.	There is not a workaround at this time.
P100522003 - 1278 **RP S/W bug on OS7400**	7400	Program a 3 RP schedule in MMC507. Typical setup for Day/Night/Lunch. While the phone system is in RP2 via the automatic scheduling; invoke RP3 early (not in the defined time band for RP3 in MMC507). The RP3 key will light for about 30 seconds, but will then revert back to RP2, even though the phones will still ring to the RP3 destination, and IT reports that the system is in RP3. Therefore, the system did not update the new RP while it's still holding the RP2 schedule!!! Similarly, the test in the lab also showing if the originally set at RP1 (open the IT tool to see the RP1) while you invoke the RP2 manually from the keyset and it will revert back to RP1 in less than 30 seconds	Problem fixed in V4.51 software. Contact Technical Support for software until official release
p100522002 - 1280 **Initialize pswd and Stn Name has been lost in MMC100-101 section 5.15.1 IT tool**	7400	Station Option of INITIALIZE PASSWORD and STATION NAME columns under section 5.15.1 are not there when using IT TOOL V1.46d	New IT Tool version v1.51 is available.
P100526001 - 1281 **Conversation got dropped after ext call in prior roll voicemail with Barge-in.**	7400	Scenario: 7400 – Ver 4.46d TDM 18D/28D/ITP-5121D Steps to reproduce: 1) Default 7400 2) set up X2003 for Barge-In to X2002, while X2001 and X2002 had conversation. 5) while the X2002 receiving a call from external party about to roll in to the voicemail; the conversation was dropped totally between the X2001 and X2002 as well as the external call. 6) Instead, the external party would be able to leave a message in X2002 mailbox.	Problem fixed in V4.51 software. Contact Technical Support for software until official release

Ref # Issue Name	Issue Type	Issue Description	Workaround Description
TS08161001 - 1293 **Dialtone is heard through computer speakers.**	CTI	OS Communicator Desk phone mode 7200S with v4.46Dd software DS 5021 or iDCS 18 Install OS Link3 Install OS Communicator Activate Link Set OS Communicator to use desk phone mode thru configuration option Activate Communicator in desk phone mode (Ext 201) Make test call to phone to confirm OS Communicator is working. (Ext 202 calls 201) You can see screen pop on computer. Ext 201 goes off hook and the dial tone can be heard through the computer's speakers. Customer logs into "keyset user options" MMCs dialtone is heard through the speakers. Ex: Ext 201 press "transfer" button to log in (dial tone is heard) it stays there until user logs out of programming.	No work around
p100807001 - 1290 **Cant change headset button mode duing call.**	Keyset	- Only on 4.46d software load. - With a headset button on the phone, press the button (headset button ON). - Now make an outbound call (CO or Intercom). - While the call is active, presssing the headset button to switch from headset to handset (or vise versa) will not work. the headset button stays on the conversation will not switch to the handset.	Place the call on HOLD and then press the headset button to switch to/from the handset.