Tadiran Telecom

Release Note **June 2013**
RN1306-010
UCx Software Version 2.07.01

Dear Business Partner,

We are happy to announce the release of UCx Software Version 2.07.01.

This replaces previous release version 2.06.04.

This document describes the new capabilities and improvements of the UCx since this release. Tadiran periodically sends update notifications for available new builds for the relevant products.

A number of problems were addressed and fixed in these versions, as described in this release note. The following tables list the main fixes that have been added in UCx Versions 2.07.01.

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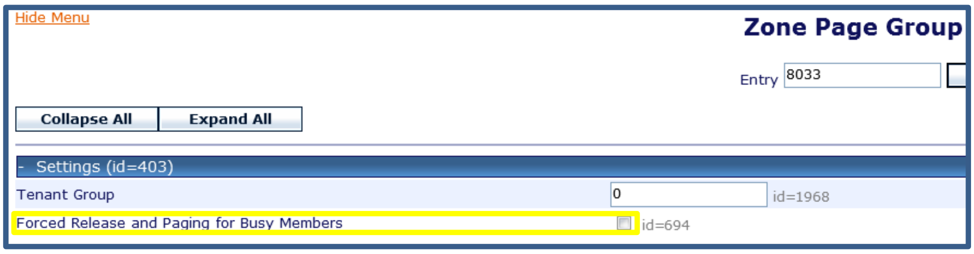
1 Improvements

The following tables list the main enhancements that have been added in UCx Main Software Version 02.07.01.

Ref #	F#23480	
Title	BRIA SIP phone will be recognized as Tadiran SIP Phone	
Configuration	• UCx system • 'A' is BRIA SIP phone	
Scenario	'A' is registered to UCx.	
Problem	One 3rd party SIP Terminal License was allocated.	
Solution	One Tadiran SIP Terminal License is now allocated.	

Ref #	F#24436	
Title	Add option in Zone Page Group to allow Zone Page to a busy SIP Terminal	
Configuration	• UCx system • A new parameter was added to VPZ named: FORCED_RELEASE_AND_PAGING_FOR_BUSY_MEMBERS(Y/N)- Y [Default N]  • 'A' is a member of Zone Page Group #1.	
Scenario	'A' is on a call and another caller dialed to Group #1.	
Problem	The Voice Page Zone call was rejected on ‘A’.	
Solution	A special support was added to force Voice Page Zone Group on a busy member. Previous call will be disconnected.	

2 Fixes

The following tables list the main fixes that have been added in UCx Main Software Version 02.07.01.

Ref #	F#24671	Ticket# 20120815018450
Title	Getting main greeting instead of personal greeting connecting to SIP Voice Mail (LxCMC)	
Configuration	• UCx system with PRI Trunks, SIP Voice Mail and station (A) • Remote Coral/ UCx system with station (B)	
Scenario	PRI incoming call is ringing at station (A), being answered and transferred to station (B). Station (B) does not answer and the call is transferred to the Voice Mail.	
Problem	The caller party does not hear station (B) personal greeting and cannot leave a message.	
Solution	A fix was added and the caller party can leave a message to station (B) mailbox.	

Ref #	F#24797	Ticket#20120829018838
Title	Disconnect problem on Group Call	
Configuration	• UCx system • 'GroupCall1' is a Group Call defined with the following: - Library as a member. The Library includes Loop Start trunk as *OUT-TK*. - *AUTO_DISCONNECT BY(Operator/Initiator/None) – Initiator* • No Night Service destinations are configured • SFE Timer →*CONF_SUPV_RECALL* timer	
Scenario	Initiator dialed to 'Group Call1' and the TK of 'Group Call1' answered the call. Initiator disconnected this call, dialed to 'GroupCall1' again and waited for CONF_SUPV_RECALL timer.	
Problem	The call is dropped.	
Solution	A fix was added and the group call stays connected until the initiator disconnects the call.	

Ref #	F# 23078	
Title	SIP trunk call to ANSWER=N Wait_Q received 200 OK (LYNC)	
Configuration	• UCx system • 'A' SIP trunk • 'B' is a Wait_Q defined as *ANSWER=N*	
Scenario	'A' calls 'B'	
Problem	UCx sent OK message back to SIP trunk.	
Solution	A fix was added and 'B' still ringing (SIP 200 OK message is not sent to 'A')	

Ref #	F# 24442	Ticket#20120711017326
Title	No audio after Call diversion (LYNC)	
Configuration	• UCx system • 'A' SIP trunk • SIP Dial Service configuration: *SUPPORT_SESSION_PROGRESS_183 – Y* • 'B' and 'C' are FlexSet-IP configured in different ZONE than SIP trunk.	
Scenario	Incoming 'A' SIP trunk is ringing in 'B' and diverted to 'C'. 'C' answered the call.	
Problem	The call had no audio	
Solution	A fix was added and the call is established with 2-way audio.	

Ref #	F# 23752	Ticket#20120905019034, Ticket#2012011010311
Title	No audio on Forwarded No Answer SIP trunk call (LYNC)	
Configuration	• UCx system • 'A' SIP trunk • SIP Dial Service configuration includes: *SUPPORT_SESSION_PROGRESS_183 – Y* • 'B' and 'C' are FlexSet-IP configured in different ZONE than SIP trunk. • 'B' is defined with forward no answer to 'C'.	
Scenario	Incoming call 'A' SIP trunk is ringing to 'B' and was forwarded no answer to 'C'. 'C' answered the call.	
Problem	The call has no audio. SDP in OK response was changed after Forward No Answer after 183 progress message.	
Solution	A fix was added and the call now is successfully established.	

Ref #	F#25494	
Title	No audio on Zone Page call to SIP terminal from SIP terminal	
Configuration	• UCx system • 'A' and 'B' are SIP terminals	
Scenario	'A' generated a Voice Page call to 'B'.	
Problem	The call was established without audio.	
Solution	A fix was added and the call was established successfully.	

Ref #	F#25405	Ticket#201301140112619
Title	Yealink SIP terminal disconnects from 3-way call	
Configuration	• UCx system • 'A' is T300 series SIP terminal • 'B' and 'C' are any other type of phones	
Scenario	'A', 'B' and 'C' established a 3-way call. 'A' disconnected his phone and left the call.	
Problem	The other two parties are also disconnected from the call.	
Solution	A fix was added and the two party's call is continued.	

Ref #	F# 23101	Ticket#20120229013809
Title	Cannot keep audio path set to headset FlexSet-IP 280S	
Configuration	• UCx system • 'A' is FlexSet-IP phone on which 1302 is set (Audio Path to Headset)	
Scenario	Restart 'A' or Restart Coral system	
Problem	Audio path returned to handset/box	
Solution	A fix was added and audio path is kept to headset.	

Ref #	F# 23354	Ticket#20120429015485
Title	OUTGOING ONLY SIP trunk rejects incoming calls on all SIP trunks	
Configuration	• UCx system • 'A' is the first SIP trunk in the Trunk group. • 'A' is configured as *FEATURES CONTROL* → *OUTGOING ONLY* • There are also available SIP trunks in this trunk groups	
Scenario	Incoming call on SIP trunk group.	
Problem	Call was rejected	
Solution	A fix was added. The system skips the "OUTGOING ONLY" SIP trunk and establishes incoming call on one of the available SIP trunks.	

Ref #	F# 24235	Ticket#20120711017326
Title	SIP terminal transfers to Undefined Destination and gets Ring Back Tone	
Configuration	• UCx system • Two SIP terminals 'A' and 'B'. • 'C is undefined destination in UCx	
Scenario	'A' was on a call with 'B'. 'A' Blind transferred the call to 'C'.	
Problem	'A' was removed from the call and 'B' heard Ring Back tone.	
Solution	A fix was added and transfer is rejected now.	

Ref #	F# 24620	Ticket#20120711017326
Title	No Ring Back tone and no audio when SIP calls ACD with Mandatory ANN and MUSIC_WITH_ANSWER=Y	
Configuration	• UCx system • ACD group is defined with: - Mandatory Announcer is configured - *MUSIC_WITH_ANSWER=Y* • 'A' is SIP terminal or SIP trunk • ACD includes available ACD members	
Scenario	'A' established a call to ACD group, mandatory announcer was played until the end of the message and one of the available ACD member's phone rang.	
Problem	The caller did not hear Ring Back Tone. When ACD member answered this call there was no audio.	
Solution	A fix was added. Caller hears a ring back tone and the call has 2-way audio.	

Ref #	F# 23034	
Title	ELAPSE_TIME definition causes one way voice	
Configuration	• UCx system • PUGW card 10.109 or higher • 'A' is a digital FlexSet defined with *ELAPSE_TIME=Y* • 'B' is SIP terminal	
Scenario	'A' established a call with 'B'. 'A' put the call on Hold and retrieved it.	
Problem	The call was established with one way audio. 'A' heard 'B' but 'B' did not hear 'A'.	
Solution	A fix was added. The call retrieved includes 2-way audio.	

Ref #	F#24110	Ticket#20120612016629
Title	No audio when SIP trunk calls Net_IP and transferred back to the hub	
Configuration	• UCx system1 is configured with SIP trunk and Net_IP trunks to Coral system #2. • UCx system2 is configured with: - Net_IP to UCx system #1 - *SFE→ NETWORK ALLOW_TRANSFER_BY_REROUTING_VIA_NET_IP- Y* • 'A' is TDM phone in UCx system #1 • 'B' is TDM phone in UCx system #2	
Scenario	Incoming SIP trunk is routed to 'B' over Net_IP. 'B' answered the call, transferred it back to 'A' and the call was answered by 'A'.	
Problem	Re-route was performed in Coral system #2. Net_IP channels were released but the call had no audio.	
Solution	A fix was added and the call now is successfully established with 2-way audio patch.	

Ref #	F#23533	Ticket#20120308014107 Ticket#20120505015693 Ticket#201212120111726
Title	Blind transfer of SIP trunk to SIP terminal recalls after a short time	
Configuration	• UCx system • 'A' is P-Series SIP terminal • 'B' is SIP trunk • 'C' is SIP terminal	
Scenario	'B' is dialing to 'A'. 'A' transferred the call by pressing **Xfer**, dialed the destination number and pressed **Send**.	
Problem	'B' rang at 'C' for 5-6 seconds and recalled back to 'A'.	
Solution	A fix was added and the call rings according to 'C' configuration, as follows: *If call forward no answer feature is set - C_FWD_NO timer value. *Else if 'C' defined as Multi Appearance - Multi Appearance timer value *Else Ring timer value	

Ref #	F#23381	Ticket#20120503015622
Title	No Music when SIP phone is being put on HOLD	
Configuration	• UCx system • 'A' is digital FlexSet • 'B' is SIP terminal that is defined with *HOLD_SUPPORT – LATE*	
Scenario	'A' established a call with 'B' and put the call on Hold.	
Problem	'B' did not hear the music (no audio).	
Solution	A fix was added and now 'B' hears the MOH.	

Ref #	F#23614	
Title	Help feature for SIP terminal	
Configuration	• UCx system • 'A' is SIP terminal or SIP trunk • 'Hunt1' is hunt Group that includes 'A' as a member • 'B' is any Keyset that includes a programmed Help button to 'Hunt1'	
Scenario	While 'B' was on a call he pressed Help key (1443+'C') and 'A' rejected the request.	
Problem	The Coral port of terminal 'A' stuck in RING state although the display of 'A' showed Idle display.	
Solution	A fix was added and Coral port of terminal 'A' returns to Idle state.	

Ref #	F# 23494	Ticket# 20120605016423
Title	Name from incoming SIP trunk is not encoded correctly in outgoing QSIG SETUP	
Configuration	• UCx system • Incoming SIP trunk is installed • Outgoing QSIG PRI is installed	
Scenario	An incoming call on SIP trunk with a caller name routed out on QSIG.	
Problem	The calling name encoded with length of 1 digit caused a failure in integration tests with Cisco.	
Solution	The problem was fixed and the calling name is now sent out correctly on QSIG.	

Ref #	F#24440	Ticket# 20120702017128
Title	Calling names are not displayed out on QSIG PRI.	
Configuration	UCx system	
Scenario	The name on QSIG outgoing calls were not displayed at the called party.	
Problem	The option to display the name was Off in the UCx database.	
Solution	The default of displaying name in QSIG was changed to be On.	

Ref #	F# 22603	Ticket# 2011111510000091
Title	The Soft Key FWD loses its mark (the black dot) after you perform Call Forward All and cancel it	
Configuration	• UCx system • 'A' is a FlexSet extension on which CFA (Call Forward All) and CNA (Call Forward No Answer) features were set.	
Scenario	'A' removed the CFA feature by dialing 141 + cancellation code.	
Problem	The black dot Call forward indication disappeared (although CNA feature was set).	
Solution	The problem was fixed and the black dot stays.	

Ref #	F#24212	Ticket#20120726017725
Title	480 replaced by 503, when the Coral cannot handle SIP SUBSCRIBE message	
Configuration	• UCx system • SIP terminal is defined on PUGW	
Scenario	SIP phone sent SUBSCRIBE SIP message before the phone was registered to the UCx.	
Problem	UCx sent '480' "Temporary Failure".	
Solution	UCx now sends '503' "Server temporary Failure".	

Ref #	F#25339	Ticket#201212270112195
Title	ACD call delivered to agent after queuing	
Configuration	• UCx system • ACD group is defined	
Scenario	An incoming call arrives to a busy ACD group and is queued. When an ACD agent becomes Idle, UCx diverts the call to the Idle agent.	
Problem	UCx sends a wrong CSTA diverted message, reporting the new state of the call. The wrong event corrupted the Real Time monitoring for the ACD group.	
Solution	UCx now sends a correct diverted distributed CSTA event to a 3rd party application that reports the correct call state of the call.	

Ref #	F#22747	Ticket#2012011510341
Title	LCR Numbering plan problem in case of XML backup	
Configuration	• UCx system • ACD group is defined	
Scenario	Administrator user activated XML backup through UCx web page	
Problem	LCR Number plan was not backed up	
Solution	XML backup of UCx includes the LCR Number plan	

Ref #	F#22811	Ticket#**2012011510340, 20120131012852**
Title	No display name of Directory upon incoming call over PRI (ETSI & QSIG)	
Configuration	• UCx system • Directory SIZE & AUTH is configured • Long Name of LIB A is configured in UCx • Dial number of LIB A is 03-9262000	
Scenario	Caller 03-9262000 dialed to any UCx keyset extension.	
Problem	The Long Name of LIB A was not displayed.	
Solution	The Long Name is displayed now.	

Ref #	F#15737	SR#4413
Title	UGW restarts –DNS missing address problem	
Configuration	• UCx system • SIP Trunk installed • UCx uses DNS for resolving ITSP IP address • DIAL SERVICE → OUTBOUND_PROXY_NAME/ADDRESS is configured with ITSP name • SIP TRUNK→ PROXY NAME/ADDRESS is not configured	
Scenario	Due to the fact that SIP TRUNK→ PROXY NAME/ADDRESS is not configured UCx was restarted every 10 minutes	
Problem	Every 10 minutes UCx is restarted	
Solution	A protection was added and UCx is no longer restarted when DNS IP address is missing	

Ref #	F#23673	Ticket#20120504015671
Title	No audio when SIP Trunk call is being transferred to SIP Trunk FlexiCall call.	
Configuration	UCx system with: • SIP Trunk • Station (A) • Station (B) has a FlexiCall out via SIP Trunk	
Scenario	An incoming SIP Trunk call is being answered by station (A) and blind transfers the call to station (B).	
Problem	There is no audio when FlexiCall destination answers the call.	
Solution	A fix was added and there is a 2-way audio when FlexiCall destination answers the call.	

Ref #	F#24061	Ticket#20120712017349
Title	UCx issue no audio on Analog station to Voicemail	
Configuration	UCx system with: • SIP Trunk • Station (A) is SLT port in UCx system • UCx SeaMail	
Scenario	A user of (A) dialed to his Mailbox to hear his Voicemail messages.	
Problem	User did not hear the greeting message nor can he dial DTMF digits	
Solution	A fix was added and there is 2-way audio on this call now.	

Ref #	F#24310	Ticket# 20120808018110
Title	No audio after ACD first announcer ends	
Configuration	UCx system with: • SIP Terminals (A) and (B) • ACD group with 2 announcers	
Scenario	SIP terminal (A) transfers SIP terminal (B) to a busy ACD group. SIP terminal (B) hears the 1st announcer.	
Problem	When 1st announcer message is ended SIP terminal (B) does not hear music nor 2nd announcer message. When one of the ACD member answers the call, the call has no audio.	
Solution	A fix was added and SIP terminal (B) hears the music, 2nd announcer and the member of ACD.	

Ref #	F#24591	Ticket# 20120627017008
Title	No audio after end of Mandatory announcer message	
Configuration	UCx system with: • ACD with: - Mandatory announcer - 1st announcer - Two or more available members • SIP (A) as a caller (Terminal or Trunk)	
Scenario	Caller (A) calls to ACD and hears the mandatory announcer message. When the message ends (A) hears a ring back tone. The call is not answered and starts ringing at the next available member.	
Problem	From this moment there is no audio or tones on the call.	
Solution	A fix was added and when the call is ringing on the next available member the caller hears a ring back tone. In addition, when next available member answers there is 2-way audio as well.	

Ref #	F#24458	Ticket#20120820018559
Title	Xfer fails due to BYE message from KSW6000/8000 during SUPERVISED REFER	
Configuration	• UCx system • 'A' is SLT Phone • 'B' is SIP KSW6000/8000 handset • Polycom SIP Terminal 'C'	
Scenario	'B' calls 'A'. 'A' answers the call and dials to 'C'. 'A' waits until 'C' answers and disconnects the call to transfer the call to 'B'.	
Problem	'B' is disconnected and the display on 'C' phones shows as if 'B' is connected to the call.	
Solution	A fix was added and 'B' is connected to 'C' with 2-way audio.	

Ref #	F# 24620	Ticket#20120711017326
Title	No Ring Back tone and no audio when SIP calls ACD with Mandatory ANN and MUSIC_WITH_ANSWER=Y	
Configuration	• UCx system • ACD group is defined with: - Mandatory Announcer is configured - *MUSIC_WITH_ANSWER=Y* • 'A' is SIP terminal or SIP trunk • ACD includes available ACD members	
Scenario	'A' established a call to ACD group, mandatory announcer was played until the end of the message, and one of the available ACD member's phone rang.	
Problem	The caller did not hear Ring Back Tone. When the ACD member answered this call there was no audio.	
Solution	A fix was added. Caller hears a ring back tone and the call has 2-way audio.	

Ref #	F#24832	Ticket#201210220110297 Ticket#201210220110283 Ticket#201210290110461
Title	SIP blind transferred causes a G.P. C5D8H in the system	
Configuration	• UCx system • 'A' is a SIP trunk • 'B' is station across Net_IP trunk • 'C' SIP Voicemail AA	
Scenario	A call is established with 'C' via 'A'. 'C' transfers the call to 'B'.	
Problem	A GP is seen in the system and the transfer failed.	
Solution	A fix was added and the call is successfully transferred. No GP is seen in the system.	

Ref #	F#25379	Ticket#20120627017008
Title	PUGW Reset	
Configuration	• UCx system • SIP Trunk	
Scenario	Incoming Un-register SIP trunk	
Problem	PUGW application restarts	
Solution	A fix was added and the card is not restarted	

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