elmeg IP290

News and changings from version 3.4 to version 3.61

GUI	WEB	SIP	LID
fixed wrong first line address book lookup match independent from the number	SIP line status appears now at sysinfo page	SDP Offer/Answer changes for improved third party call control	fixed increasing timestamps for out-of-band DTMF
change of backlight setting is recognized immediately	flash plugin usage on by default	SIP stack should not unnecessarily prevent reboot any more	fixed RTP stream wasn't following re-INVITE (MPO)
DST works now for absolute dates (e.g.Iran), too	fixes for danish texts	added missing Expires header in SUBSCRIBE reply	fixed missing audio when entering CMC
fixes for danish texts	added new option to use SIP compact headers	added missing Subscription-Status header to NOTIFY	speaker gains increased for handset and headset
MWI count was wrong according to RFC	added microphone volume control	fixed bad Contact in SUBSCRIBE reply	tone generation improved for easily playing different kind of tones
fixed Ringer1 for SIP lines	hash '#' can now be dialed properly via command.htm?key=	fixed problem when the ACK was missing	added sending media keep alive packets (STUN requests)
fixed edit mode problem for Xfer on AddressBook jump	phone numbers with leading + can be handled properly via web page	fixed problem with stale connections when call was rejected	fixed a problem with infinite DHCP lease time
switch audio devices on SPKR key for snom190	number guessing is off by default	fixed disturbed dialtone when non-RFC3264 devices on-hold	added microphone volume control (mic vol. passed from web interface)
enum input on log-on wizard	dont show user passwords on settings log	fixed missing new SUBSCRIBE when function key destination was changed	tuned sidetone
knocking for priority cwi	fixes for french texts	added un-SUBSCRIBE for the former function key destination	improved SRTP
fixed CALL-INFO answer-after	added DTMF for programmable keys in SIP (#732)	registrations and subscriptions are now cleaned up before reboot	synchronized LEDs
reenabled partial number address book lookup	rejected NOTIFY's will delete the subscription	reboot on check-sync can now only be avoided by adding parameter reboot=false	fixed ptime problem
added some few new french translations	subscription termination via NOTIFY are now processed	by default turn ICE off	fixed mute problem, DTMFs also stop on mute now
partial number lookup is now setting dependent	fixed SDP version number was not increased in 200 OK	ignore Record-Route from PRACK	improved sending keep alive mechanism
CCNR fixed	removed un-necessary DNS lookup for STUN if ICE supports is off	resolve issues with stale connections	first keep alive packet is sent immediately now
fixed return from blind transfer	added support for 3xx responses on REGISTER requests	fixed STUN wasn't working any more after re-registration	audio problems in transfers, conferencing, one way audio etc. fixed
fixed call acceptance on transfer	subscription dialog state will be periodically verified	fixed STUN results where not used in SDP	codecs and packet sizes tuned
exact match for display name when dialing out	enhancements of the french translation	fixed missing IP address in SDP (o and c line)	echo cancellation added
lookup addressbook entry for user if uri match is not found		keepalive is now independent from the STUN server	audio sub system improved, headset volume increased
lookup number match in uri of call list entry		fixed potential endless loop in dialplan	added multiple key generation set by programmable keys
partial number matching looks for complete match at the end		fixed reboot=false in check-sync prevents reboot	corrected RTP playout time
added minimum length of number guessing setting when dialing out		added support for Proxy Require header	added support for receiving 10ms and 30ms RTP packets
speed dial support for Xfer		wasn't respecting DNS NAPTR records with sips content	fixed voice lag problem
Ringer device for headset is only used for ringing and not CWI		fixed problem with multiple authentications of INVITEs	
fixes for non peer to peer pickup		refresh to the registrar, not to the STUN server	
added auto exit on volume change		fixed subscription deletion which leads to freezing on sw update	
fixed irish timezone DST		fixed wrong domain names in REGISTER after 3xx message	
look for partial number match in Address Book lookup		fixed a wrong on-hold state with MPO	
added Auto Answer per SIP line (#692)		Expires header in SUBSCRIBE and its replies will be evaluated	
fixed offhook call jump problem on multiple incoming calls			
fixed led for incoming call on terminated line set to a function keyo SIP stale NOTIFY's will be rejected now			
enhancements of the french translation			
fixed repeated park/pickup freeze			
volume up/down keys in idle state play static ring melody			
fixed to long timeout for terminated screen after canceled transfer with AOC			
enhancements of french translation			