Release of the Software 7.50

Herewith the Beta Software 7.50 for the compact PBX systems elmeg T240, elmeg T444 and elmeg T484 is released for external field test. The naming of this release corresponds now with other PBX systems. Content of the release based on the release 7.30 for the ICT systems.

The Beta SW 7.50 is a feature – but also a Bug-fix Release, please see the contents in the following table.

Firmware elmeg T240

File: T240_v750_rc14.fwr

Firmware elmeg T444

File: T444_v750_rc14.fwr

Firmware elmeg T484

File: T484_v750_rc14.fwr

WIN-Tools

File: WIN-Tools-Txxx-V750-INT_080304.exe

List of new features and bug fixing from Release 1.14 RC02 to Release 7.50

Nr.	Changes / Bug fixing
	System
	Call Through External inward dialling (via ISDN or VoIP) with the possibility to use the external trunk lines for further dialling (via ISDN, VoIP, FXO). With active LCR also routing via internal connected analogue GSM-gateways is possible. Authentication of the inward dialling is done using Pin's, CLIP information or combination of both. For authentication with CLIP information PBX phone book entries are marked with the Call Through option. **Attention:** This new function doesn't realise the function "local break out" in coupled scenarios with ICT systems!
	Public holidays for integrated calendar A list of public holidays (50 entries at maximum) can be created. For these days additional switching times are configurable.
	Message Waiting Indication for internal FXS and ISDN ports (T484 only) MWI is now supported for internal analogue extensions as well as for internal ISDN extensions. Following MWI variants are supported: 1) MWI without ringing and 2) MWI with ringing. MWI is only used and qualified for signalisation of new voice box messages (refer to expanded voice box functionalities).
	Exchange line authorization with PIN Using an individually PIN a user is able to make a call from any PBX extension, also from extensions with restricted line access. To get an exchange line the user have to dial a new code number (*5*) and a user individual PIN (4 digits). For this call extension settings of the PIN-user will be used: e.g. MSN signalisation, exchange line authorization, feature settings. **Attention:** Enquiry calls and transferring calls are possible, but automatic recalls are not supported. Automatic call back on busy (CCBS) or on no reply (CCNR) is not possible. The line authorization PIN can not be dialled during an inquiry call.
	CLIP No Screening Is now supported for the external ISDN trunk lines with P-P protocol.
	Expanded Voice box functionalities (T484 only) - The activation status of a voice box is visible and changeable via WIN tools. - New activation types "on busy" and "on busy or on no reply" are realized in addition to the existing activation "immediately" and "on no reply". - For team voice boxes is now possible to signal new messages at all team extensions. - At ISDN and analogue phones new messages can be signalled with the special off hook tone and future also using the MWI function. The kind of showing MWI depends on the phones. - For voice box access the authentication with the voice box PIN is needed. This authentication can be disabled for the owner of the voice box. - Call transfer to extensions with an assigned voice box is now supported. After transferring the caller hears the welcome announcement of the voice box. - Standard welcome and final announcements are loadable via WIN tools.

	Welcome announcements of each individual voice box are also loadable via WIN tools. **Remarks:** If not at all announcements are loaded into the PBX the firmware-integrated German announcements (standard welcome and final) will be used. **Attention:** Duration of all announcements is not limited. The users have to be aware of The available memory size on Compact Flash Card. One half of memory size is reserved for voice applications (e.g. auto attendant, music on hold). The other half is used for voice boxes.
	2 GB Compact Flash cards (T484 only) Future also 2 GB Compact Flash Cards are supported for voice boxes and voice applications.
	ECT function for external connections The handover handling of two external connections can be configured. If activated two external connections can be transferred (coupled) by putting down the handset.
	Emergency call routing (T444, T484 only) The call will only be dialled via ISDN or POTS - VoIP is disabled **Attention:** An automatic disconnection is only possible on ISDN!
	Call forwarding The configuration of call forwarding is now possible by keypad procedure of Standard IP-Telephones
	Room monitoring This function is now possible with Standard IP-Telephones
	Music on Hold for connections over SIP provider (T444, T484 only) Connections over SIP provider putting on hold can get also the internal music on hold. Therefore a new configuration flag for SIP provider is available.
	MSN - related trunk access The keypad procedure fort he MSN related trunk access is now possible with Standard IP- Telephones
	Public dial tone – Intern generated tone simulation While using some features (e.g. LCR / ARS, Call through, SIP provider access) an intern generated special dial tone is used. Future simulated external dial tone of the country specific settings will be provided to the user.
	Open inquiry This feature is now also possible with multiple connections in hold status.
	Ringing Voltage for internal analogue ports (FXS) (T484 only) The ringing voltage for internal analogue ports of the T484 is increased to 42V. **Bug fixing:** The analogue terminals: Panasonic KX-TSC 11 EXB + Siemens 5020 didn't react on call. The signalling voltage internal was increased – the terminals are signalling
	Bug fixing: The CCBS announcement of the public switch was not connected to the analogue ports
	Bug fixing: Voice Applications: Transfer with DISA to another announcement is now possible
	Bug fixing: The LCR call-by-call provider name will not longer shown in the display of the system-telephones in case of calls via SIP-Provider.

	Bug fixing: A transferred call (via UbA) couldn't refused from the SysTel by softkey, the call came back immediately
	Bug fixing: After transferring a call to a team (by UbA) the callers ID was not shown in the display of the team members.
	Bug fixing: An incoming call with the CLIP-Info "anonymous" (Calling party length =0) **and** a configuration of international prefixes "00" and "49", causes in this combination a reset of the system.
	Bug fixing: At incoming calls to a team with **external** call variant, the subscriber authorizations like (LCR, trunk-configuration...etc.) are now established via the configured "charge assignment", so the routing as well as LCR are considered and maybe the trunk assignment will be done according to the setting of the "charge-subscriber".
	Bug fixing: (T484 only) Internal ringing timer increased from 3 minutes to 10 minutes.
	Bug fixing: Outgoing MSN's and DDI's are now sending in "National Number" format. (French country settings only).
	Bug fixing: (T484 only) Sometimes wave files could not be sending to the PBX. This error is now solved.
	Bug fixing: (T240 only) CNIP is now enabled and supported in the same way as with T444 and T484.
	Bugfixing: At TAPI extensions external calls via SIP trunk lines are now signalled with the complete phone number.
Routing adaptations (T444, T484 only)	
	Automatic disconnection of the WAN connection The WAN connection can now be disconnected Configurable by time – to avoid the disconnection of the Provider
	Integration into existing LAN structures The PBX can be connected into existing LAN's using the LAN interface. Into the PBX IP address and net mask of the default gateway have to be configured. **Attention:** In such scenarios following interfaces or features are disabled automatically: WAN interface, internet connections and fallback settings, packet filter firewall, dynamic DNS settings)
VoIP functionalities (T444, T484 only)	
	Expanded location settings The following parameters are now configurable by WIN-Tools: - Codec selection list (It is possible to define allowed codec's for connections from or to the selected location) - G.726 coding method (I366 or RFC 3551 / X.420)

	Expanded SIP provider settings The following parameters are now configurable by WIN-Tools: - Activate / deactivate of configured SIP providers - Hold into the PBX system (providing internal MoH for external SIP calls) - Substitution of international prefix with "+" - Number of simultaneously connections over the same SIP provider - Free configurable substitution of number prefix - Configuring the registration timer - Configuration of two individually STUN server (IP address, port number, repeat time) (Needed for usage of the PBX system behind another NAT device, router) - Configuration of the outbound proxy server - Codec selection list (define codec's to use with the corresponding SIP provider) - G.726 coding method (I366 or RFC 3551 / X.420) - Optional deleting of existing SIP bindings after restart of the PBX system - Optional up streaming device with NAT support
	Expanded SIP-extension settings The following parameters are now configurable by WIN-Tools: - G.726 coding method (I366 or RFC 3551 / X.420)
	Configure RTP port and ToS (Type of Service) value The following parameters are now configurable by WIN-Tools: - Min. RTP Port for SIP connections - ToS (Type of Service) value for SIP packets - ToS (Type of Service) value for RTP packets
	SIP without SDP Incoming SIP connections using "Invites" without SDP parameters will be also answered / accepted from the PBX.
	DTMF inband / outband Now the full DTMF signalling (in – and outband) is possible using SIP provider (Needed for dialling access code numbers, e.g. for external voice mail systems).
	New SIPGATE Tarif „Plus" To configure this account, please note: - Each number will be added in this tarif by a SIP ID - All numbers are using the same password and it is necessary to configure them n-times with every single telephone number.
	VoIP Statusinfo- display: In the configurator, the control-center and the service tool a status display was added for service purposes. It will be shown: a) SIP Provider registration - status. b) SIP extensions registration - status
	Bugfix: Generally improvements of call handling while transferring calls to mixed teams with analogue, ISDN and IP extensions. Was a Voip-connection transferred without announcement to a team and picked-up by a Voip-phone, no voice connection was established.
	Bugfix: A SIP telephone, which is already in hold state cannot establish another call hold yet.

System - telephones (T444, T484 only)	
	SysTel line keys It is now possible to configure line keys for SIP-provider access.
Adaptations of analogue trunk lines (POTS / FXO)	
	Busy tone detection (T484 only) Busy tone detection is now implemented and switchable. The tones / intervals are listed in country specific tables. Because of hardware restrictions not supported with T240 and T444. - The length of the **busy tone** is configurable. Duration time for busy tone detection is set to 3 seconds by default. - **Announcements**: will be played now unlimited. - Call forwarding / explicit call transfer is now unlimited possible from ext. to ext., if the busy tone detection is enabled.
	K-Break Functionality for UK has been implemented. (Maybe the busy-tone- detection can be disabled when K-break is active.)
	CLIP DTMF For external analogue interfaces now CLIP DTMF is supported as well as CLIP FSK. The used CLIP version is configurable for each analogue trunk line individually. **Attention:** CLIP DTMF is necessary for the countries NL and DK. This function was implemented as ETSI specification, but it was not possible to qualify very deeply.
	Multiple numbers for FXO trunk lines (Portugal country settings only) In Portuguese country settings up to three numbers for FXO trunk lines can be supported. The dialled number will be recognised based on a defined ringing cycle (distinctive ringing).
	Bug fixing: CLIP FSK for incoming calls The detection of CLIP FSK with poorly analogue lines (long distance to public switch, bad or attenuate signals) was instable and is now improved.
	POTS Modules: The sound level of analogue trunk lines was increased for international country versions.
General details	
	Service Center The service tool normally used from FEC partners or resellers for the ICT systems is now also supported from the compact PBX systems. Current Service Center version: 1.72
	WIN Tools The settings of a configured team can be taking over to another team (copy and insert using click on the right mouse key).

Open – but known problems	
	With the current software version following quick tests are already done: Off-premises with SIP phones, PBX coupling with ICT systems (fixed and dynamic IP addresses). PBX coupling between compact systems T4x4 is not supported.
Hints for additional information in the FAQ area of the FEC homepage	

Supporting IP phones / soft clients or IP system phones

Several elmeg pbx systems are supporting IP phones / soft clients with SIP standard or IP system phones as follows:

elmeg T444

Using the module M 4 DSP the elmeg T444 supports up to 4 DSP channels. Based on this we recommend and support at maximum the usage of 4 IP phones or IP soft clients with SIP standard. IP system phones are principally not supported.

elmeg T484

Using the module M 4 DSP the elmeg T484 supports up to 4 DSP channels. Based on this we recommend and support at maximum the usage of 4 IP phones or IP soft clients with SIP standard. IP system phones are principally not supported.

elmeg ICT

Using the modules M 4 DSP and M 8 DSP the ICT systems can be expanded up to 16 DSP channels at maximum.
Following scenarios will be distinguished:
If only using IP system phones we recommend and support the usage of 16 IP system phones at maximum.
If only using IP phones or soft clients with SIP standard we recommend and support the usage of 30 phones at maximum.