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Release of the Beta – SW 7.30

Herewith the Beta Software 7.30 for the ICT Systems 46 / 88 / 880 is released.

The Beta SW 7.30 is a feature – but also a Bug-fix Release, please see the contents in the following table.

Firmware ICT 46

File: ICT46_V7.30 RC 11

Firmware ICT 88

File: ICT88_V7.30 RC 11

Firmware ICT 880 (rack)

File: ICT880_V7.30 RC 11

Firmware VoVPN-Gateway

File: ictgw_v7.30 RC 18

WIN-Tools

File: 7.30 Build

List of bug fixing and changes from Release 6.16 to Release 7.30

Nr.	Changes / Bugfixing
	SIP adaptations
1.	**Extended VoIP & SIP Provider settings** The following prameters are now configurable by use of the WIN-Tools: - Registrar IP port - Outbound Proxy - Codec selection / List - G.726 coding method - Registration Timer - Prefix (calling number replacement) - hold in the PBX - replacement of 00 by + - Max. number of simultaneous connections
2.	**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 pasword and it is necessary to configure them n-times with every single thelephone number
	System
3.	**Emergency call routing** The call will only be dialled via ISDN or POTS - VoIP is disabled **Attention: An automatic disconnection is only possible on ISDN!**
4.	**Call forwarding** The configuration of call forwarding is now possible by keypad procedure of Standard IP-Telephones
5.	**Room monitoring** This function is now possible with IP-SysTel's as well as Standard IP-Telephones
6.	**MSN - related trunk access** The keypad procedure fort he MSN related trunk access is now possible with Standard IP- Telephones
7.	**Bugfix:** The CCBS announcement of the public switch was not connected to the analogue ports
8.	**DTMF inband / outband** Now the full DTMF signalling (in – and outband) is possible over VoIP
9.	**Fax-Signalling over VoIP** In case of (outgoing) fax-transmission the codec G 7.11 for transmission is always used
10.	**2. trunk port with 4- S0- Modul** The S0 3 Port of the new Moduls 4 S0 (2 S0), V2.0 can now be configured in the WIN-Tools to the functionality of a trunk. The max. number of trunk lines is not increased. Further on the configuration of max. 8 S0 externa for the ICT 88/880 and 4 S0 external for the ICT 46 is possible.

11.	**TAPI with IP-system telephones** It is now possible to use TAPI with IP-System telephones **Attention:** The use of TAPI is only possible for system telephones – blocked for Standard SIP- Telephones!
12.	**New function: Call Through** Externe inward dialing with the possibility to use the external trunk lines for further dialing. Also support of SIP-providers. With active LCR also routing via internal connected analogue GSM-gateways possible. **Attention:** This new function doesn't realise the function "local break out" in coupled ICT systems!
13.	**Number of trunks at ICT 46:** The number of trunks are expanded from 4 to 5 – only with ICT46
14.	**Open inquiry** This feature is now also possible with multiple connections in hold status.
15.	**Bugfix** Voice Applications: Transfer with DISA to another announcement is now possible
16.	**Bugfix:** The LCR call-by-call Providername will not longer shown in the display of the system-telephones in case of calls via SIP-Provider
17.	**Bugfix:** A transferred call (via UbA) could'nd refused from the SysTel by softkey, the call came back immediately
18.	**Bugfix:** After transferring a call to a team (by UbA) the callers ID was not shown in the display of the team members.
19.	**Bugfix:** A call forwarding from a IP-SysTel to a SIP-provider was not established
20.	**Bugfix:** 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
21.	**Bugfix:** 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 assignement", 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".
22.	**Bugfix:** During fully load-tests (Calls via SIP Provider) with our ICT-Test environment, we faced sporadic restarts of the ICT system. This behaviour could fixed by performance improvments.
23.	**Bugfix:** The analogue terminals: Panasonic KX-TSC 11 EXB + Siemens 5020 didn't reacte on call. The signalling voltage internal was increased – the terminals are signalling

24.	**Bad quality of converted wave-files** The quality of the wave-files became very bad during conversion. Due to this reason, the already knownBEYERTONE conversion tool was implemented again into the Win-Tools(as former). This tool delivers a very good quality of converted wave-files. **Attention:** The volume of the files is not adjustable any longer! For volume adjustments we recommend the use of freeware tools. Additional informations are given in the FAQ area.
	VoVPN – Gateway
25.	**Expanded location settings:** The following parameters are now configurable by WIN-Tools: - Codec selection / list - G.726 coding method
26.	**Expanded SIP-extension settings:** The following parameters are now configurable by WIN-Tools: - G.726 coding method
27.	**Expanded VoIP settings**: 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
28.	**Automatic disconnection of the WAN-connection:** Configurable by time – to avoid the disconnection of the Provider
29.	**VPN-Tunnel** Increase of simultaneous VPN connections with IPsec from 5 to 10. The limitation is only valid for the number of simultaneous active tunnels. The configuration of VPN tunnels is not limited.
30.	**GW-Data send - flag** For a better support of the configuration by telnet, a new configuration flag was introduced, which the sending / and the override possibility of the gateway configuration generally on- or off switches. The flag can be found under the registercard „general". **Attention:** The flag is disabled in the default configuration!
31.	**Special RAS dial-in to the gateway module for support purposes:** A new Telefone procedure was introduced zu enable / disable the GW Serviceaccess for Debug, Trace, Diagnose etc. The feature code is: *52.../#52.... **Attention: Please make sure, that the IP addresses** 192.168.1.254 and 192.168.1.253 are not already used, because these addresses are the default addresses for the RAS access.
32.	**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) Type of the used DSP modules, max. number of channels b) SIP Provider registration - status. c) SIP extensions and IP-SysTel registration - status d) VPN connection status

33.	**Bugfix:** Was a Voip-connection transferred without announcement to a team and picked-up by a Voip-phone, no voice connection was etablished.
34.	**Bugfix:** The SIP-provider-data where not deleted in case of default setting of the whole system – only in an ICT system equipped with POTS- and a GW-module.
35.	**Bugfix:** A SIP telephone, which is already in hold state cannot establish a another call hold yet.
36.	**Bugfix:** In case of a simultaneous change of the configuration in the ICT as well as in the gateway module, the ICT did a restart too fast, even though the writing of the GW-data was not finished. Due to this, GW-data became lost. **Solution:** The ICT is waiting now for ca. 30 sec. (only in case if a gateway module is inserted and the data-tranfer-flag is set) before the reset takes place.
37.	The proposals of the VPN tunnel protocols are changed to the predecessor version. **Attention:** Update or already coupled systems – **all** systems have to be updated to the new SW Release! The VPN / IPSec support of the gateway is now given overlapping with all actual FEC products.
	System - telephones
38.	**Codec support:** A special protocol between system and the IP-SysTels is introduced to drive the compromised codecs. **Attention: This new function is only available after the SW Update to version 4.3x**
39.	**SysTel line keys** It is now possible to configure line keys for SIP-provider access.
40.	**Bugfix:** In the display of the IP-SysTel the umlaut couldn't shown correctly in the display. A conversion as with the So-system telephones where missing
41.	**Bugfix:** If an incoming call to an IP-S phone was not answered, but in this ringing status a line-key was used – this behauviour causes first a loss of the phones registration, then a reset of the gateway module and later to a reset of the whole system.
	Adaptations of analogue trunks (POTS) for export
43.	**New POTS (FXO) functions:** - **busy tone detection** is now implemented and is switch able, the tones / intervals are listed in country specific tables, - **Busy tone**: the length of the busy tone is now configurable, - **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 and CLIP FSK:** this new function is now possible. FSK or DTMF is for every analogue POTS trunk line configurable.

		Attention: CLIP DTMF is necessary for the countries NL and DK. This function was implemented as ETSI specification, but it was no possible to qualify very deeply.
		**General details**
44.		**Attention: Important informations to the SMC adapter with 512 xD card:** In the ICT 512 MB (but no 256 MB) memory cards can be read – but not formatted! The qualification of those cards is not finished yet! To use those memory cards, the following details have to be attended: • The XD card types must be formatted to FAT, via a external card reader, who supports this format. • The formatting of cards with this size by ICT / WINTOOLS is generally not possible, the ICT doesn't support this size of memory cards. • Generally it is not allowed to insert a card into an active system – this will cause a restart! **Important detail:** To insert a SMC adapter into an ICT – it is necesary to follow this instructions: 1.) Switch off the ICT wall-system 2.) Insert the SM/XD Adapter with XD card into the module slot 3.) When the SM/XD adapter touches the housing, the adapter is **not** yet in the correct contacted position. Move the upper part of the adapter carefully backwards and then it is possible to insert the adapter appr. 1 mm lower. Then a resistance can be encountered and the adapter has reached the final position. 4.) Switch on the ICT 5.) Read out of the configuration - WINTOOLS must then show the XD card. 6.) In RACK – systems the described details are not valid.
		**Open – but known problems**
		It is generally possible to configure a static IP address via the WinTools, but some concept related spezial quality has to attend. If the static IP address is configured via Telnet, she is not shown in the WinTools. This could be dangerous, because the gateway data will be overwritten – in case of remote access or directly from the customer and the whole Telnet Konfiguration gets lost. Mor informations can be found in the FAQ area.
		At present we have a registration problem with the SIP provider freenet. This problem was announced out of the field – but we couldn't follow up on this in Rel. 7.30 due to time reasons.
		No DTMF detection by using the auto attendant **only** on POTS line – no reaction. The SW was improved – but due to less time, we couldn't qualify deeply.

	Problems with CLIP on POTS: Spain, Denmark and Holland beta testers reports that no or sporadic wrong digits are presented on incoming calls via POTS. We improved the SW – but due to less time, we couldn't qualify deeply.
Hints for additional informations in the FAQ area of the FEC homepage	
	A list of tested and qualified SIP Providers can be found on the FEC homepage in the FAQ area.
	There are informations available regarding the different handling of codecs at different providers e.g.: 1&1, Toplink, Sipgate

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 moduls 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.
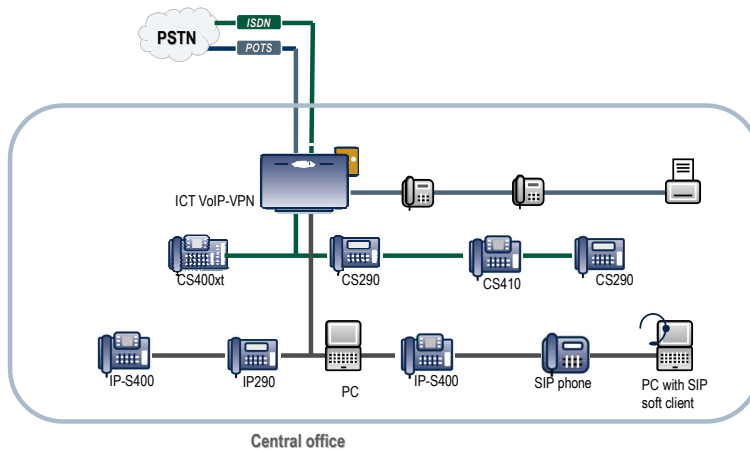
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IP telephony within the LAN

- Start
- IP telephony within LAN
- Off-premise
- SIP provider
- SIP/VPN coupling I
- SIP/VPN coupling II
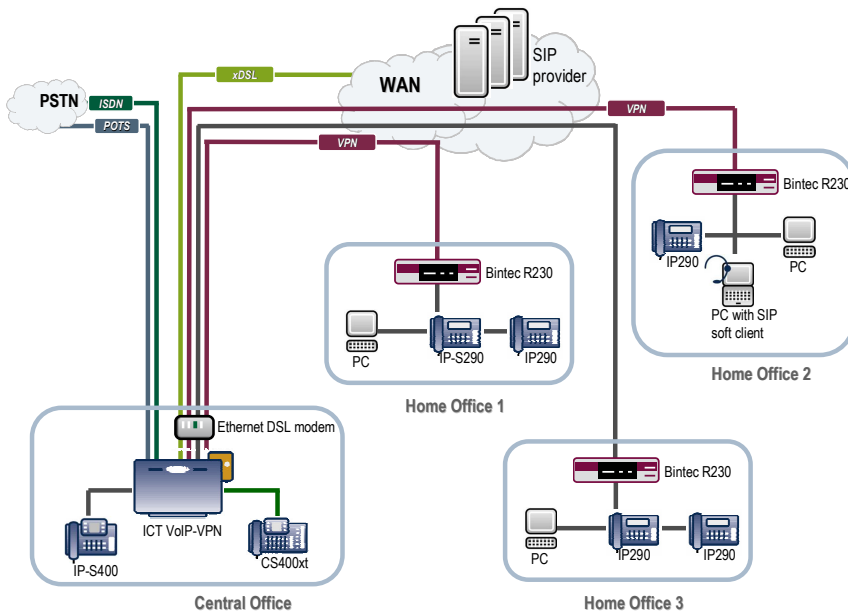
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IP telephony within the LAN

- Connection of IP telephones , soft clients or system phones
  - IP telephones or soft clients with SIP standard (e.g. IP290)
  - IP system phones (IP-S290, IP-S290plus, IP-S400)

- System feature with system phone only
  - e.g.: busy lamps, function keys/led's, phonebook access, system menu

- Internal communication between all connected phones with up to 16 DSP channels for media crossing (Analogue/ ISDN ←→IP, IP-S)
  - Up to 16 IP system phones or 30 SIP standard IP phones supported

- Start
- IP telephony within LAN
- Off-premise
- SIP provider
- SIP/VPN coupling I
- SIP/VPN coupling II

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Off-premises with IP / IP system phones

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Off-premises with IP / IP system phones

- Connection of IP telephones, soft clients or system phones through the WAN interface
    - IP telephones or soft clients with SIP standard (e.g. IP290), with or without VPN
    - IP system phones (IP-S290, IP-S290plus, IP-S400) with VPN only
- Secure connections using VPN / IPSec
    - Terminating of VPN within the Gateway, Up to 10 simultaneous VPN connection supported
- System feature with system phone only
    - e.g.: busy lamps, function keys/led's, phonebook access, system menu
- Communication between two off-premise phones always through the VoIP-VPN Gateway

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External SIP in addition to ISDN, POTS
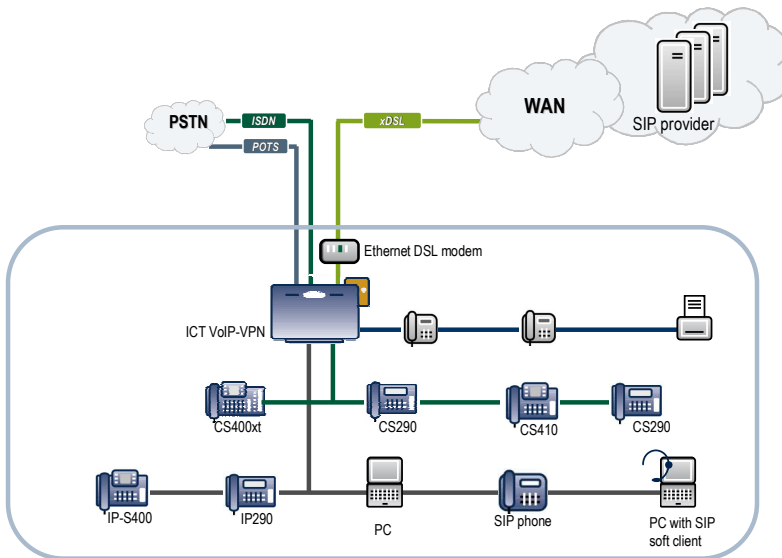
- Start
- IP telephony within LAN
- Off-premise
- SIP provider
- SIP/VPN coupling I
- SIP/VPN coupling II

6

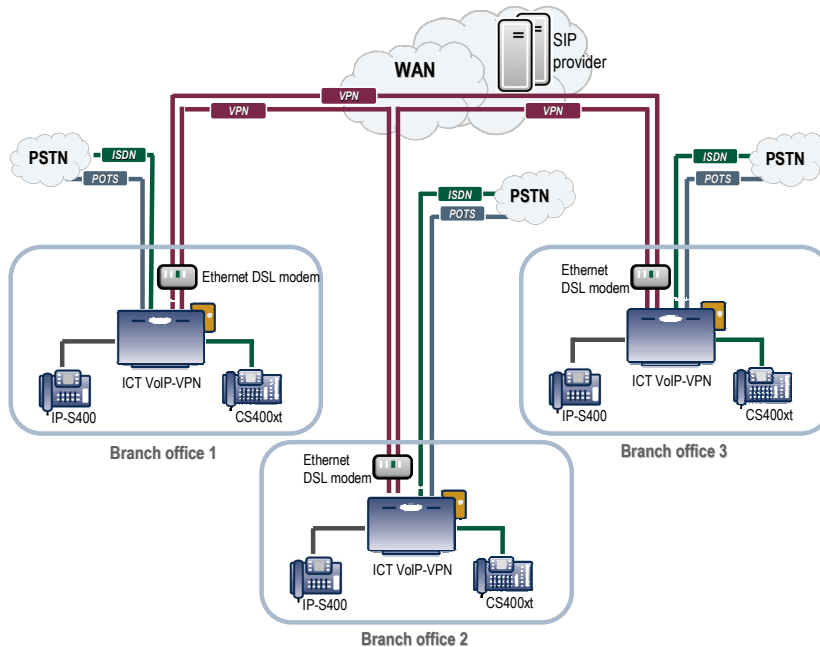## External SIP in addition to ISDN, POTS

- Connection to external SIP provider (25 max.)

- Support of several provider in Germany (single numbers and DDI numbers)
  - Support of further international SIP provider on progress

- Start
- IP telephony within LAN
- Off-premise
- SIP provider
- SIP/VPN coupling I
- SIP/VPN coupling II
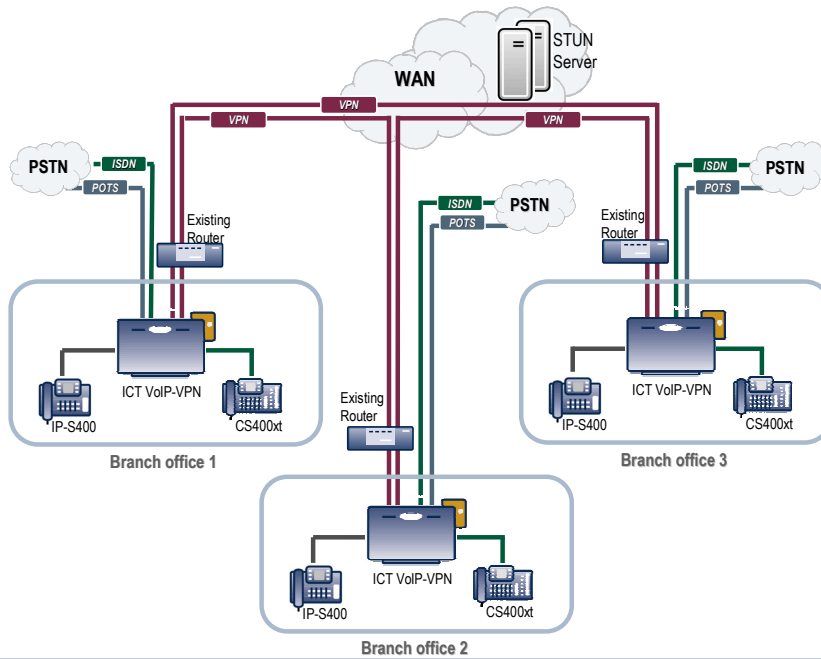
7

Coupling via SIP & VPN (I)

- Start
- IP telephony within LAN
- Off-premise
- SIP provider
- SIP/VPN coupling I
- SIP/VPN coupling II

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Coupling via SIP & VPN (I)

- Coupling of several ICT systems with VoIP-VPN Gateway

- Secure connections using VPN / IPSec

- Internal connections between the ICT systems through the VPN tunnel

- Routing of voice data using the bundle and ARS / LCR functions

- Restrictions:

  - System features (e.g. busy lamp) are not supported between several ICT systems
    (only supported inside one system and with off-premise subscribers in the same system)

  - No common configuration and number plan
    Each system has to be configured separately and has it's own number plan

  - Customer have to dial the full number of branch offices (e.g. 0911-1234-5678) but no additional cost due tot the use of existing VPN

- Start
- IP telephony within LAN
- Off-premise
- SIP provider
- SIP/VPN coupling I
- SIP/VPN coupling II

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Coupling via SIP & VPN (II)

- Start
- IP telephony within LAN
- Off-premise
- SIP provider
- SIP/VPN coupling I
- SIP/VPN coupling II

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Coupling via SIP & VPN (II)

- STUN support for VoIP-VPN Gateway available in the next release

- Using the ICT with VoIP-VPN Gateway behind an existing NAT device
  - Existing routers (own FEC brand or other brands) responsible for IP access and routing
  - ICT responsible for VoIP functions only

- Integration in customer established LAN scenarios possible

- Start
- IP telephony within LAN
- Off-premise
- SIP provider
- SIP/VPN coupling I
- SIP/VPN coupling II

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