Release of the SW 6.16

Herewith the Software Version 6.16 for the ICT systems 46 / 88 / 880 is released.

The SW 6.16 is mainly a bug-fix software, the content is described in the following table.

Firmware ICT 46

File: ICT46_V2.16_rc002.fwr

Firmware ICT 88

File: ICT88_V2.16_rc002.fwr

Firmware ICT 880 (rack)

File: ICT880_V2.16_rc002.fwr

Firmware VoVPN-Gateway

File: ictgw_v116_rc8.fwr

WIN-Tools

File: 7.16

Table of errors and changes from Release 6.10 to Release 6.16

Nr.	Changes / Bugfixing
	SIP improvements
1.	**Support of the STUN Protocol** Per SIP provider the ICT supports 2 individual STUN – server, with automatic fallback function.
2.	**SIP provider registration fails** Different registrations to the same IP-address are now supported by the ICT-gateway.
3.	**Two SIP provider configured - selection fails** It is now possible to configure two different service provider configured with the same login-name.
5.	**SIP-Provider call number length: 24 digits**
6.	**Several problems solved with some SIP-Providers** Toplink, 1&1, QSC
7.	**CLIP to a SIP Provider in case of call forwarding** The complete number will now be transferred correctly
	System
8.	**LCR – improvements regarding Call forwarding in conjunction with SIP Providers / trunks**
9.	**Speech Quality getting worse after a couple of days – problem solved**
10.	**Echo improvements with IP-S Phones**
	VoVPN Gateway
11.	**Stabilisation of extensions in Off Premises connected via IP**
12.	**Stabilisation of system – coupling via VPN**
13.	**Functionality improvement of the 5 possible VPN channels**
14.	**Correct allocation of the busy tone in case of occupied DSP's** The 17th extension, requiring a DSP channel, gets the busy tone
	Systemtelefone
15.	**MCID is now possible with IP-S phones**
16.	**Support of the new System – telephone IP-S 400**
17.	**Correct signalisation of the Voicemail function key (VMS 350) at IP-SysTels.**
18.	**Support of IP-phones / Soft Clients / or IP- System telephones** 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.

	Remarks All systems support only VoIP and media interworking between TDM and IP interfaces while using the DSP modules only. The available DSP channels will be managed from the pbx system and used dynamically for internal or external connections. It is the responsibility of the dealer to establish solutions with a balance between available DSP channels and simultaneous needed internal and external connections.