Service Manual
T3 IP Telephones
Connected to IP Office

Supplement to
Service Manual T3 Telephones Connected to IP Office

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Copyright

Declaration of conformity (T3 IP)

We, Avaya-Tenovis GmbH & Co. KG, declare that the telephones T3 IP Compact, T3 IP Classic and T3 IP Comfort comply with the basic requirements and other relevant regulations in accordance with directive 1999/5/EU (Radio and Telecommunication Terminal Equipment, R&TTE).

These telephones are intended for connection to Ethernet/LAN interfaces of IP-capable Avaya telecommunication systems and fulfill the basic requirements of directive 1999/5/EU (R&TTE).

Avaya cannot guarantee successful operation of the telephones in LAN or Intranet environments in which no Avaya telecommunications system is present.

When using links and DSS modules:

This link / DSS module is designed for the connection to T3 IP telephones and complies with the basic requirements as per EU guideline 1999/5 (radio and telecommunication terminal equipment R&TTE).

This link / DSS module was designed only for use with T3 IP telephones.

If any problems should occur during operation, please initially contact your specialist dealer or your system administrator.

The complete Declaration of Conformity can be requested at the following Internet address:

www.avaya.de/gcm/emea/de/includedcontent/conformity.htm

or using the keyword "conformity" in the index.

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Contents

Introduction

This document is intended as a supplement to "Service Manual T3 connected to IP Office".

It describes the processes necessary to commission the IP telephone for operation in a data network.

When in operation, the IP telephone has the same functionality as a T3 ISDN telephone connected to IP Office.

Configurations, except via the IP telephone menus, can be carried out via the Web terminal, the "IP Phone Admin" tool or the IP Office manager.

General explanations

Basic information about IP telephones can be found in the Service Manual T3 IP connected to Integral 5, order number 4.999.105.974. That manual provides basic information about DHCP, MAC addresses, IP addresses, Subnet masks and FTP.

How does the IP telephone communicate with the IP Office?

Communication between an ISDN terminal and the IP Office

- The configuration data, for example to establish and disconnect a connection (e.g. setup...), are transferred to/from the system in the D channel of layer 3.
- The voice data are sent via the B channel.

Communication between an IP telephone and the IP Office

The D and B channel information typical of ISDN applications also exists here. However, because there is no ISDN connection, the ISDN data must be sent to the recipient in a different way.

Basically, the IP phone and the IP Office communicate via the TCP/IP protocol.

- **Layer 3** of the OSI layer model is the **switching layer**. This layer is used to establish the connection between the end systems (IP telephone-IP Office). Information regarding the transmitting and receiving station is included in the **message packets.** The **data packages** are addressed in this layer. The **IP protocol** (TCP / **IP**) applies here. This layer is also responsible for routing in complex networks, in other words establishing the route for getting the data packages through to the end station. At the appropriate point in the IP header, it is indicated which protocol of higher-level layers will be prefixed to the data portion next, e.g. **06 = TCP** or **17 = UDP**.

- **Layer 4** of the OSI layer model is the **transport layer**. This layer is responsible for establishing and controlling connections between communication partners. Communication partners are communication applications in this context. These applications must be addressed and data loss in a connection detected and resolved. The transport layer is the link between the transport and application system. It doesn't matter to the functions of the application system what type of network the messages come from or are forwarded to. The protocols **TCP** (**T**ransmission **C**ontrol **P**rotocol) or **UDP** (**U**ser **D**atagram **P**rotocol) apply here.

- **TCP protocol**

  The main task of this protocol is to transfer data securely through the network.

- **UDP protocol**

  Unlike TCP, this protocol does not confirm transfer or implement other security measures. This means that UDP is, however, more efficient and therefore suitable for high-speed applications.

IP Telephone ↔ IP Office

The ISDN configuration information is packed into a data packet (payload/user data). It is noted in the **IP** header that the **TCP** protocol is to be used. The data packet is now **securely** sent to the recipient. **Security** is crucial when configuration information is being transmitted.

The voice information is also packaged into a data packet. It is noted in the **IP** header that the **UDP** protocol is to be used. The data packet is now sent to the recipient **quickly and unsecured**. **Speed** is crucial when voice information that must arrive in good quality is being transmitted.

AND

All IP Office-specific codings of a system telephone remain intact.

This is also known as **Tunneling**

Codecs, bandwidth, delay, jitter and jitter buffer

Codec

Both the IP telephone and IP Office must have an audio Codec. This encodes the voice signals from the microphone for transfer via the transmitting terminal and decodes the received audio signals sent to the loudspeaker/earphone of the terminal.

G.711 and G.729 are supported in the IP telephone.

Process	Nominal data rate Codec generates its output data with…	Packetization delay Pass through the Codec	Short description
G.711A	64 kBit/s	1 ms	No compression, voice quality comparable with ISDN, European process for audio digitization
G.711 µ	64 kBit/s	1 ms	As for G.711A, except a US process for audio digitization
G.729	8 kBit/s	25 ms	Best voice quality of the **compressing** process

Bandwidth

- Data flow depends on the Codec and the configured delay.
- Voice data are not continuously transferred via the packet-oriented TCP/IP protocol.
- The Codec combines the voice data into a packet during a specific period of time, 10ms for the IP telephone.
- IP packets consist of a variable portion of user data and a fixed portion of management information.
- For small packets the ratio between user and management data is very unfavorable for the required bandwidth but there is only a short voice delay.

RTP-Datagram			
IP Header 20 bytes	UDP Header 8 bytes	RTP Header 12 bytes	RTP Payload (User data) 8 bytes
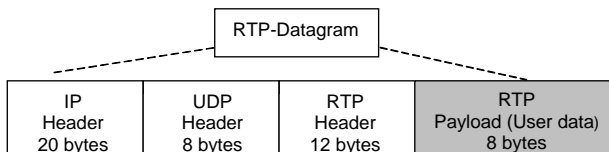
Section of data transfer

The header before the user data is 40 bytes long.

If sent via a G.729 Codec every 20 ms, the user data are 20 bytes per datagram plus additional layer 2 bytes. (Ethernet plus 18 bytes).

Delay

(delay between packet or speech packet length)

Delay times can be set using the IP telephone. These times indicate the time delay before the transmitter dispatches the data into the network. The Codec can of course continue to collect data during this delay time.

> With a 64 kb/s Codec and a delay of 20 ms, the transmitter generates of 160 bytes of user data.

$$\frac{\text{Nominal Data Rate (bits/s) x Delay (ms)}}{\text{8 (bits/byte) x 1000 (ms/s)}} = \frac{64000 \times 20}{8 \times 1000} = 160$$

> 240 bytes of user data are sent every 30 ms at a delay time of 30 ms.

Increasing the delay reduces the required bandwidth because the ratio of management data to user data is less. However, the voice quality is reduced because the recipient obtains the information later.

Jitter

There should be a constant time interval between the arrival of each speech packet at the receiver. Network topology could cause the packets to arrive later however. This variation is known as inter-arrival time or jitter.

Jitter buffer

Using the management data contained in the RTP header, the jitter buffer

- corrects the time interval between received packets
- corrects the sequence of received packets
- detects packets received twice
- initiates algorithms in the event of packet loss

The packet loss rate in any given time interval should be lower than 1%. (This requires network specialists)

The transfer process selected for the IP telephone on the Ethernet is an asynchronous data transfer process. The voice data transferred via the network should be heard synchronously however. A **jitter buffer** is installed as the link between asynchronous data transfer and synchronous listening. The jitter buffer temporarily stores the arriving data packets so that they can be transferred to end-users at constant time intervals. **Jitter buffers** can usually offset runtime variations of up to 100 ms. Larger variations cause the buffer to empty (buffer underflow). This can adversely affect the service to the

same extent as a packet loss. As a result, the network is required to limit packet losses in the data network and runtime variation greater than 100 ms to considerably less than 1 %.

The jitter buffer can be set to

	Default	
• short		If speech quality is adequate after the IP telephone has been commissioned, the default setting should not be changed.
• normal		
• long		
• auto	x	

TOS/DiffServ parameters

If you wish to modify the quality of service (QoS) of the voice packets you can configure the TOS (Type of Service) / DiffServ (Differentiated Services Architecture) field in the IP header of the RTP voice packets. This makes prioritization possible.

There are 7 bits to be set in the TOS field (bit 0 to bit 6). Bits 0 to 5 represent the 6 DIffServ bits. Bits 3 to 6 represent the TOS parameters bits. Bit 7 of the TOS field cannot be set since it is not used by DiffServ nor as a TOS parameter bit.

Gateway

A gateway is a node in the LAN that communicates with terminals in other networks. The gateway transfers the data between the various networks. If components in other networks are addressed through a gateway, the IP Office system must know the gateway address.

In the IP Office

Separate network

If an IP Office together with the associated equipment is connected to a separate network, no gateway address is required. Only the gatekeeper address of the IP Office must be noted in the IP telephone.

(Example: 16 subscribers)
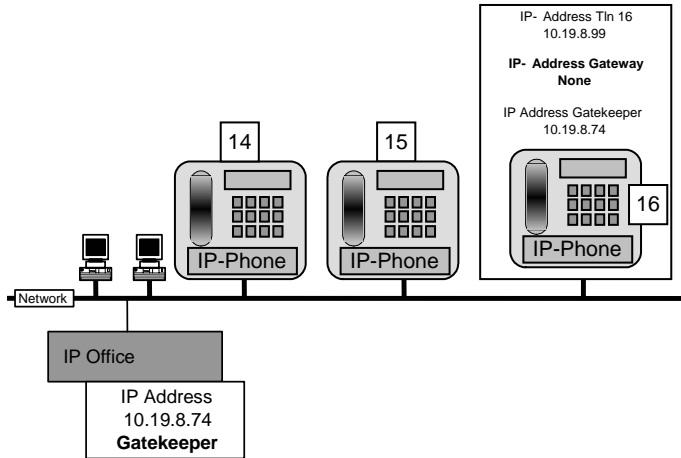
2 networks, IP telephones of one IP Office distributed on both networks

IP telephone 14 in Building B:

The following must be noted:

- **IP telephone 14 cannot be logged in via the IP Office DHCP**. Login has to be carried out manually (see "Commissioning an IP telephone"). **Alternatively:** The IP telephone can be logged in with DHCP with certain router configurations. To achieve this, a DHCP relay agent has to run on the router. This DHCP relay agent has to react to the DHCP broadcast of the IP telephone, collect the IP address from the DHCP server and forward it to the IP-telephone.
- **IP telephone 14** must have entered the gateway address of **its** network (Network Building B).
- **IP telephone 14** must have entered the gatekeeper address of **its** IP Office (IP Office located in Network Building A).

IP telephones 11, 12, 13 in Building A:

The following must be noted:

- **IP telephones 11, 12, 13** can be logged in via DHCP if the DHCP server on the IP Office is not active. Login has to be carried out manually (see "Commissioning an IP telephone").
- **IP telephones 11, 12 and 13** must have entered the gateway address of **their** network (Network Building A).
- **IP telephones 11, 12 and 13** must have entered the gatekeeper address of **their** IP Office. (IP Office is located in Network Building A)

IP-Address Tln 14
10.10.9.99

**IP-Address Gateway
10.10.9.77**

IP-Address Gatekeeper
10.10.8.74

14

IP-Phone

Router/Gateway

Network
Building B

IP-Address Gateway
10.10.9.77

Network
connection

IP-Address Tln 11
10.10.8.96

**IP-Address Gateway
10.10.8.88**

IP-Address Gatekeeper
10.10.8.74

11

IP-Phone

IP-Address Tln 12
10.10.8.97

**IP-Address Gateway
10.10.8.88**

IP-Address Gatekeeper
10.10.8.74

12

IP-Phone

IP-Address Tln 13
10.10.8.98

**IP-Address Gateway
10.10.8.88**

IP-Address Gatekeeper
10.10.8.74

13

IP-Phone

Router/Gateway

Network
Building A

IP-Address Gateway
10.10.8.88

IP Office

IP-Address Gateway
10.10.8.88
IP-Address Gatekeeper
10.10.8.74

14

Gatekeeper

The IP Office has gatekeeper functionality. The IP telephones are logged in at the gatekeeper.

If there are several gatekeepers in an IP network, ensure that the associated address is entered in the default gatekeeper entry of the IP telephone.

Media streaming / VCM

The entire telephone data traffic via the gatekeeper (central media streaming) occurs at the IP Office 4.0. A direct data connection between two IP telephones (decentralized media streaming) to provide relief for the gatekeeper is not supported.

The internal link will be looped through via a RTP relay function in the IP Office without using VCM channels. When making a connection, the VCM channels of the IP Office are always needed however, even if both IP telephones use the same Codec.

External connections, connections with different Codecs or connections to non-IP telephones always need VCM channels.

T3 IP telephone connections
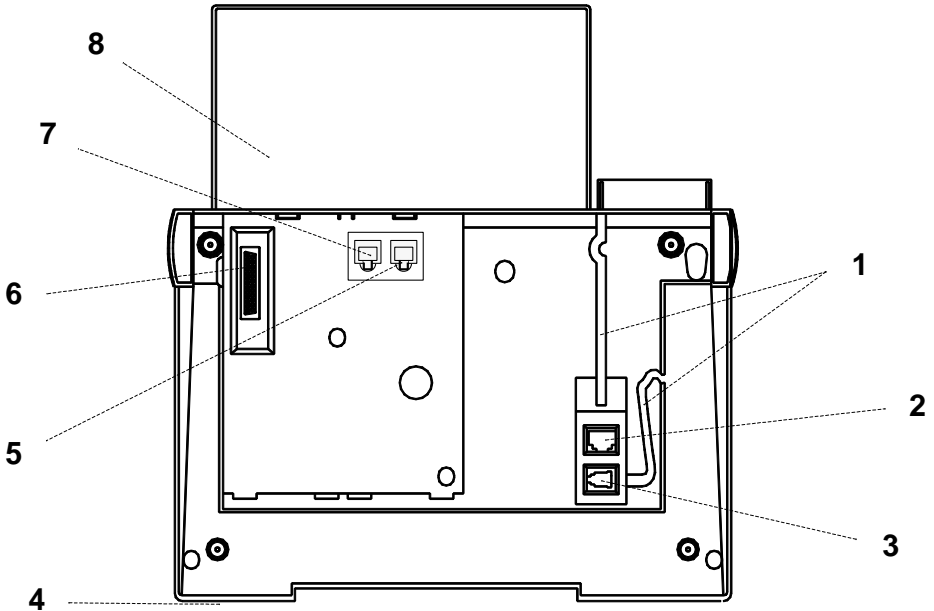
1	Cable routing	5	PC
2	Connector socket for power supply	6	Socket for connection of links (not T3 IP Compact)
3	Handset connecting socket	7	Network connection
4	Microphone opening	8	Display

The T3 IP Compact has a socket for direct connection of a headset (below sockets 5 and 7). It is not equipped with a link connecting socket.

T3 IP telephone cabling
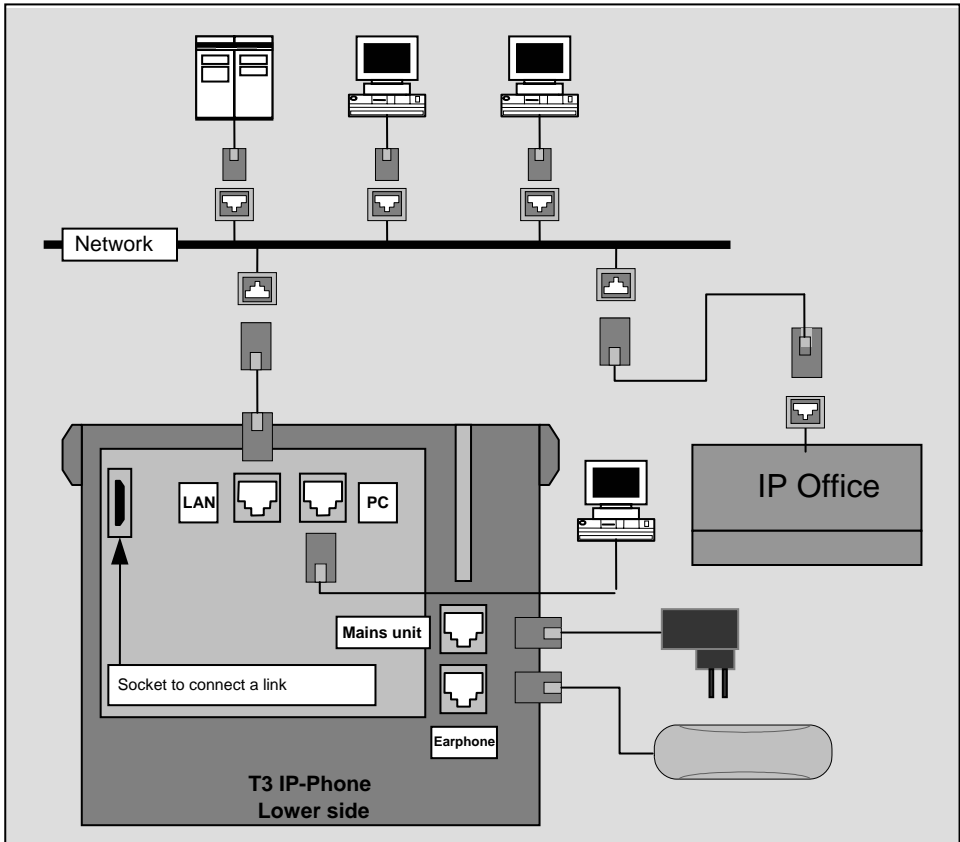
The IP telephone is connected to the **network** via its LAN socket.

A **PC** can be connected using the PC socket. The PC then behaves as if it were directly connected with the data network.

The plug-in mains unit is connected to the **mains unit** socket.
Note: The plug-in mains unit is not supplied with the IP telephone and must be ordered separately. Only the plug-in power supply unit with reference number 4.999.086.800 may be used.

The handset is attached via in the **handset socket**.

Information about connecting **Links** can be found in the chapter "AEI / Headset Link".

Interfaces

Network interfaces

8-pin Western jack	**1: TD+** **Transmit Data +** **2: TD-** **Transmit Data –** **3: RD+** **Receive Data +** 4: NC 5: NC **6: RD-** **Receive Data-** 7: NC 8: NC

Handset interface

4-pin Western jack	**1 – SK-** Speech circuit **2 – HK+** Listening circuit **3 – HK-** Listening circuit **4 – SK+** Speech circuit **Make sure the polarity (+. -) is correct.**
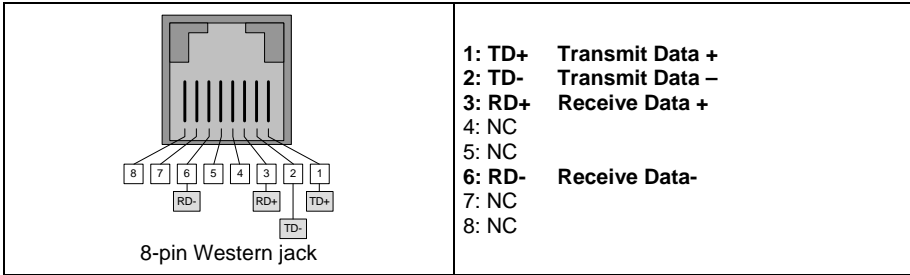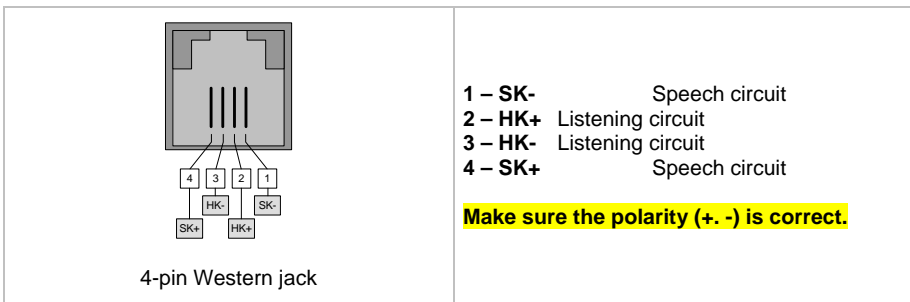
Plug-in mains unit interface

	40V DC/max.150 mA Reference number 4.999.086.800 Supplies the power to the handset and possibly also one link module **7: GND** **8: +40 V**
8-pin Western jack	**Do NOT insert the Western jack of the plug-in mains unit into the LAN or PC sockets!**

18
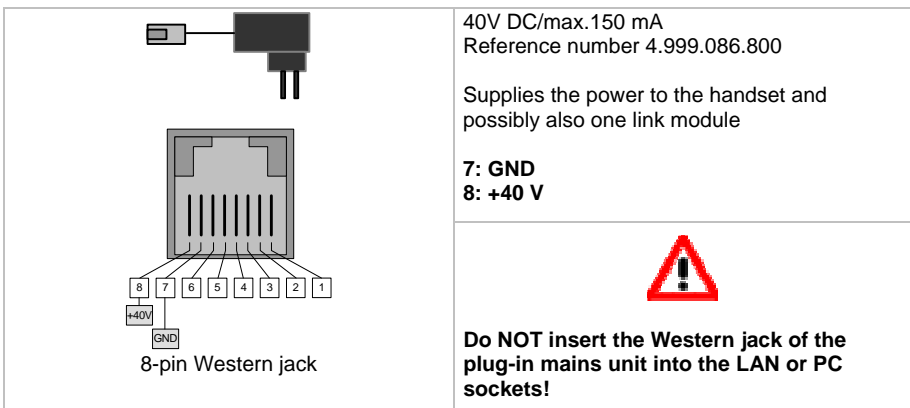
Power supply

The T3 IP telephones can be powered in various ways. The power supply also depends on the network components being used.

Note

- **All** new T3 IP telephones from T3 IP II.2 and higher support power over LAN (PoL) according to standard 802.3a/f
- Phantom power supply according to Cisco is **not** supported.

a)   Local power supply via power supply unit:

The power supply unit is connected to the IP telephone. It then supplies power to the telephone and also to the link, if one is present.
If you are using a switch or another power supply component in your network which supports Power over LAN, the power supply unit is not required.

Plug-in mains unit T3 IP phone lower side	8 7 6 5 4 3 2 1 +40V GND	40 V DC/ max. 150 mA Reference number 4.999.086.800 Supplies telephone and possibly 1 link 7: GND 8: +40 V
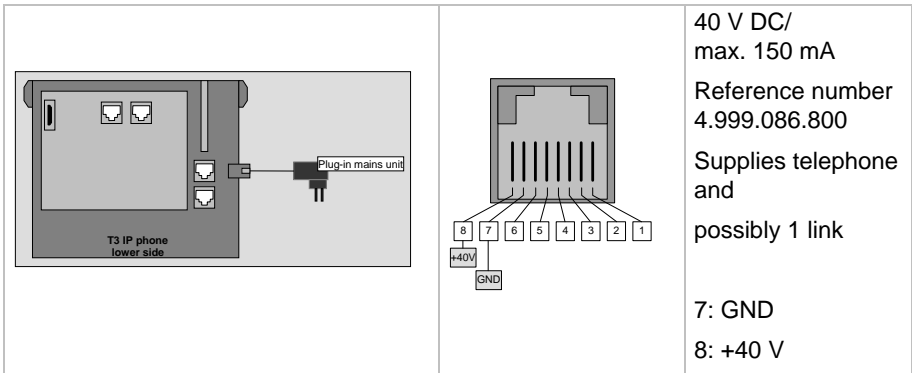
b)   Remote Power over LAN (PoL)
The telephone is supplied with power via two unused wires of the LAN cable (this is only possible if these wires run without interruptions throughout the installation).

Detection and power supply in accordance with current standard 802.3a/f (midspan). Max. range: 100 m with Cat.5 cable 26AWG (same as transmission range acc. to IEEE802.3u)

1: TD+   Transmit Data +
2: TD-   Transmit Data –
3: RD+   Receive Data +
6: RD-   Receive Data –

4 and 5: GND
7 and 8: – 48 V DC
            (35...57 V)
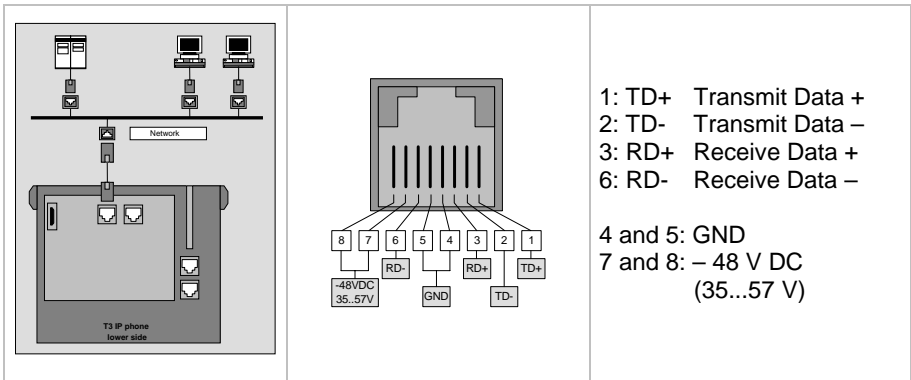
c) Remote power supply via the LAN (PoL with phantom power supply):

The telephone is supplied with power using the signal wires in the LAN cable. This is also possible with 2-pair cables.

Max. range: 100 m with Cat. 5 cable 26 AWG (same as transmission range acc. to IEEE802.3u).

Detection and power supply in accordance with current standard 802.3a/f (endspan solution).

1: TD+   Transmit Data +

2: TD-   Transmit Data –

3: RD+   Receive Data +

6: RD-   Receive Data –

Phantom power supply
1 and 2: – 48 V DC (35...57 V),
3 and 6: GND
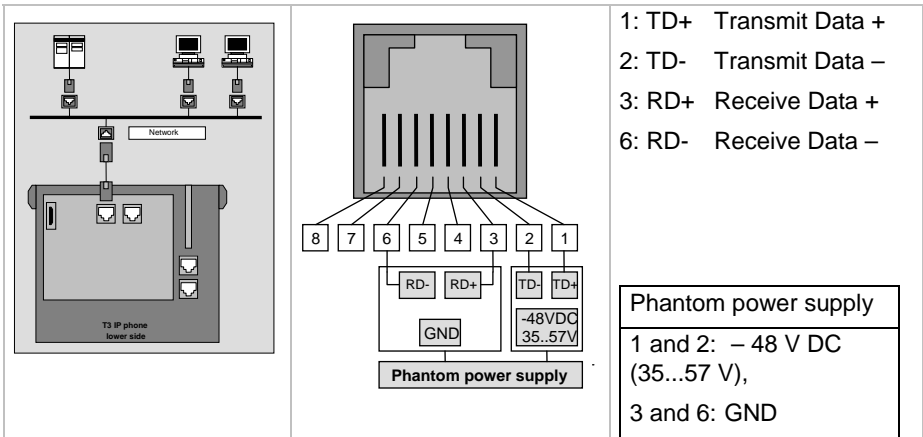
Power class

All T3 IP telephones have "Power class detection". This enables a suitable switch or another power supply component to detect how much power each IP telephone requires. The classes are defined as follows:

- Class 0: undefined (max. 15.4 W per port)
- Class 1: up to 4 W per port
- Class 2: up to 7 W per port
- Class 3: up to 15.4 W per port

The telephones T3 IP Classic and T3 IP Comfort are allocated to power class 2 and the IP Compact to power class 1.

Displays during commissioning

The T3 IP telephone shows various messages on the display during start-up. These let you know what is happening and may also notify you about errors.

For information on how to proceed if the start-up fails, please refer to the chapter "What to do if start-up fails

Message text	Meaning
bootloader B02.xx	Start message with Boot Loader version information (hardware level II)
DHCP...	DHCP procedure has been started
Booting from flash: T2xx_0DE.x4i	Downloading an image from flash memory
Download activated: T2xx_0DE.x4i	Downloading image from the FTP server

Links (from IP Office 3.2 and higher)

Links can be connected to T3 IP Classic and T3 IP Comfort. The T3 IP Compact does not have a link interface. But it does have a headset socket. It is possible to connect the same headsets there as at the links.

AEI/Headset link

View of the AEI/Headset link

AEI pin allocation

1: XTEout (Audio Output Telephone)
2: XTEin (Audio Input Telephone)
3: XGndD (Audio Ground)

4: YTEin (Digital Input Telephone)
5: YGndD (Digital Ground)
6: YTEout (Digital Output Telephone)
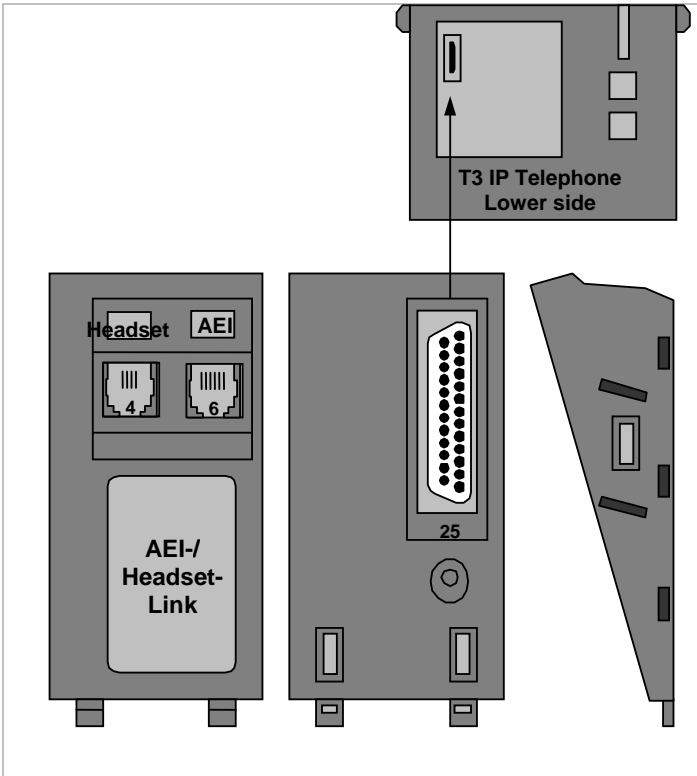
AEI interface
6-pin Western jack
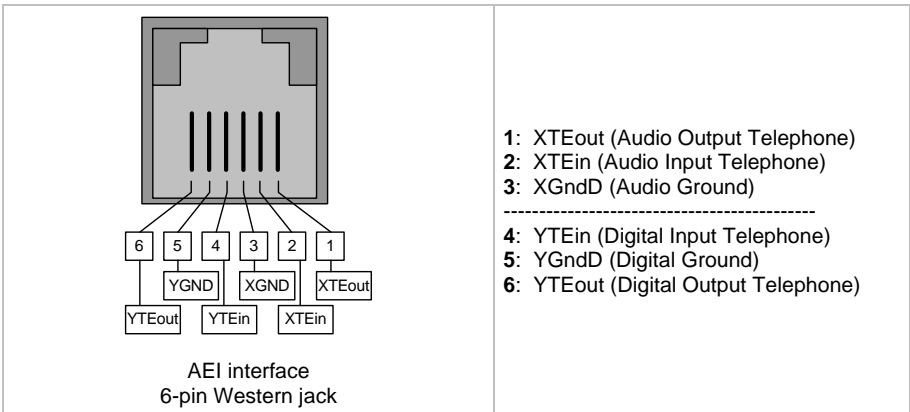
Headset Interface pin allocation

Allocation is identical to the one with the headset link (see below).

Connection options

DSS modules can be connected to the *AEI socket*. These use the Y interface. Further additional devices cannot be connected. The X interface is not supported.

The headsets HSG 21...24 from Plantronics can be connected to the *headset socket*.

For more information on links and accessories, please refer to the service manual "T3 connected to IP Office".

Download

Since the AEI/headset link does not contain any software, no download is necessary.

Headset link

This link can be used to connect the headsets HSG 21…24 from Plantronics.

The setting options for a headset via the headset link are summarized in a menu.

Front and rear view of the headset link

The headset link has one interface: the headset interface.
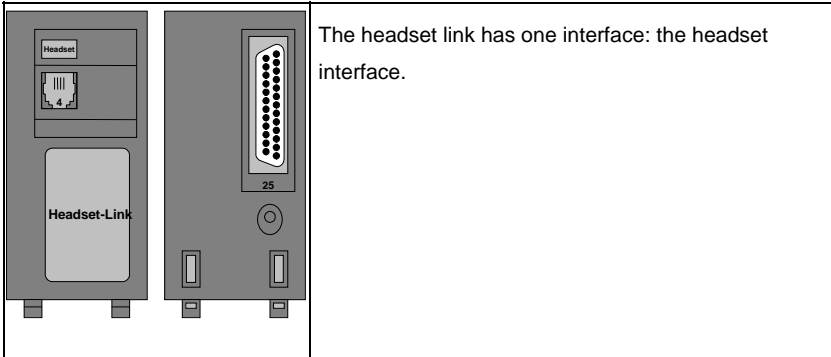
The headset interface of the headset link

Headset jack

1 HK-
2 MIC-
3 MIC+
4 HK+
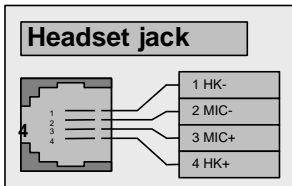
Connect the headset to the headset link or the AEI/headset link
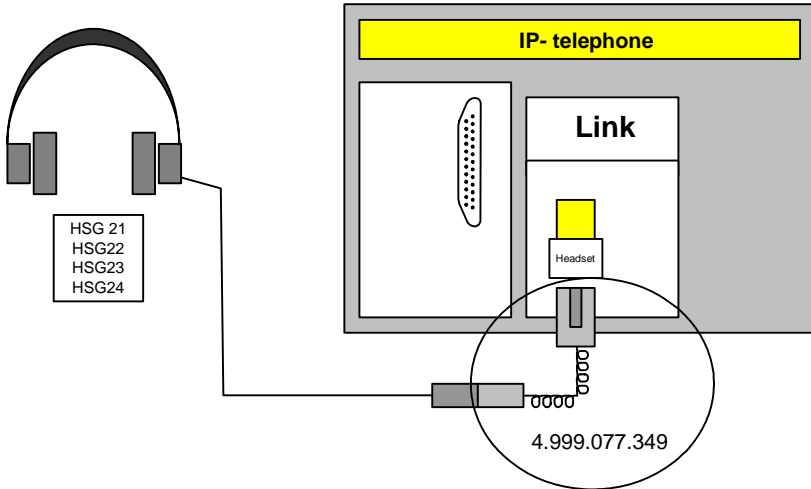
Only headsets HSG 21, 22, 23 and 24 may be connected!

Connection to the headset socket of the headset link is only possible via a spiral cable WE4-on-QD (Quick Disconnect) (reference number: 4.999.077.349).
The cable is supplied with the HSG module 2 and can be used if required.

Using a headset

A headset can be connected by means of the handset connector, at the headset interface of a link or directly to the headset socket of the T3 IP Compact.

The telephone cannot determine whether a headset is plugged in or not. Therefore, you have to log it in at the telephone (see user manual T3 IP).

Note:

- After unplugging the headset, the user has to log it out from the telephone if the telephone is to be used without the headset.
- The headset interfaces in the links and in the T3 IP Compact have the same functions.

Menu structure with headset logged in

If a headset link is connected, the menu item "Headset" is displayed in the Main menu. This is irrespective of whether a headset is connected or not.

After selecting this menu item via the "OK" key, the display changes to the menu "Settings > Display/Tones > Acoustics/Contrast".

The new menu items "Headset (off)" and "Headset vol." will be found here.

The headset can be activated via the menu item "Headset (off)". The headset volume is set via the menu item "Headset vol.".

The easiest way to activate and deactivate the headset is to assign a function key to "Headset".

Download

Since the headset link does not contain any software, no download is necessary.

Installation of the links

Link installation is described in the installation instructions
"Links for Integral T3 / T3 IP".

Safety information for link installation

When a link is inserted, the IP telephone must be **completely** isolated from the power supply.

1. Unplug all existing connecting cables (power supply, LAN connecting cable).

2. Insert link and connect any auxiliary equipment (e.g. DSS module).

3. Connect the power supply units.

4. Plug the connecting cable into the T3 IP telephone.

Technical data

T3 IP links

	AEI/Headset	Headset
Temperature range for operation	0 ºC to 45 ºC	
Temperature range for storage / transport	-25 ºC to +70 ºC	
Power consumption	0.5 W (only AEI interface); Approx. 0.1 W via headset interface	Approx. 0.1 W
Power supply	Via T3 IP telephone	
Connection hardware for links	D-Sub, 25 pin	
Analogue interface	Plantronics headset	Plantronics headset
Digital interface	AEI-Y	
Connection system	*AEI:* 6-pin Western Electric connector *Plantronics headset:* 4-pin Western Electric connector	*Plantronics headset:* 4-pin Western Electric connector
Width	62 mm	
Depth	143 mm	
Height	42 mm	
Weight	94 g	

T3 IP Comfort

Temperature range for operation	0 ºC to +45 ºC
Temperature range for storage / transport	–20 ºC to +60 ºC
Operating voltage	35 to 57 V polarity-dependent
Power consumption	3.4 W to 4 W depending on the operating mode without link max. 6 W with plugged link
Power supply	Power supply via LAN cable (phantom power supply or via spare wire), or via T3 mains unit
Connecting cable	LAN cable twisted 4-pair Cat 5, length 3 m (accessory pack)
Connection system	Western-Electric-plug connector RJ 45 on both sides, shielded
D-channel protocol	TN1R6 (tunneling)
Audio frequency range	300 Hz to 3400 Hz
Interfaces for links	1
Connection hardware for links	D-Sub, 25 pin
Line interfaces (2)	Ethernet 10 Base-T or 100 Base-Tx to IEEE 802.3, built-in switch
Range	100 m with 100 Base-Tx with Cat. 5 cable with 4 connectors (IEEE 802.3)
Width	287 mm
Depth	207–275 mm (depending on display setting)
Height (without display) (including display)	82 mm 89–207 mm (depending on display setting)
Weight	1364 g

T3 IP Classic

Temperature range for operation	0 ºC to +45 ºC
Temperature range for storage / transport	−20 ºC to +60 ºC
Operating voltage	35 to 57 V polarity-dependent
Power consumption	3.4 W to 4 W depending on the operating mode without link max. 6 W with plugged link
Power supply	Power supply via LAN cable (phantom power supply or via spare wire), or via T3 mains unit
Connecting cable	LAN cable twisted 4-pair Cat 5, length 3 m (accessory pack)
Connection system	Western-Electric-plug connector RJ 45 on both sides, shielded
D-channel protocol	TN1R6 (tunneling)
Audio frequency range	300 Hz to 3400 Hz
Interfaces for links	1
Connection hardware for links	D-Sub, 25 pin
Line interfaces (2)	Ethernet 10 Base-T or 100 Base-Tx to IEEE 802.3, built-in switch
Range	100 m with 100 Base-Tx with Cat. 5 cable with 4 connectors (IEEE 802.3)
Width	238 mm
Depth	207 mm (depending on display setting)
Height (without display) (including display)	82 mm 120 mm (depending on display setting)
Weight	980 g

T3 IP Compact

Temperature range for operation	0 ºC to +45 ºC
Temperature range for storage / transport	–20 ºC to +60 ºC
Operating voltage	35 to 57 V polarity-dependent
Power consumption	Max. 3 W
Power supply	Power supply via LAN cable (phantom power supply or via spare wire), or via T3 mains unit
Connecting cable	LAN cable twisted 4-pair Cat 5, length 3 m (accessory pack)
Connection system	Western-Electric-plug connector RJ 45 on both sides, shielded
D-channel protocol	TN1R6 (tunneling)
Audio frequency range	300 Hz to 3400 Hz
Interfaces for links	none
Connection hardware for headset	4 pin WE for headset HSG 21-23
Line interfaces (2)	Ethernet 10 Base-T or 100 Base-Tx to IEEE 802.3, built-in switch
Range	100 m with 100 Base-Tx with Cat. 5 cable with 4 connectors (IEEE 802.3)
Width	200 mm
Depth	207 mm (depending on display setting)
Height (without display) (including display)	82 mm 120 mm (depending on display setting)
Weight	820 g

Mixed operation with other telephones

T3 IP telephones can be combined with any other Avaya T3 ISDN telephones
(T3 Comfort, T3 Classic and T3 Compact) depending on the capacity limits of the
IP Office.

Combination of T3 IP telephones with other Avaya IP telephones is **not** permitted.

Menus of the IP telephone

Three menus can be opened on the IP telephone:

- The Standard menu. It can be opened while the telephone is registered. Menu control occurs via the IP Office. The manual only deals with the new feature "Hot Desking". The Standard menu is described in the service manual T3 connected to IP Office.

- The User menu, which can only be accessed when the telephone is not registered.

- The Audit menu which is accessible via a PIN number from the user menu.

User menu

Gatekeeper list			
	Gatekeeper 1 to 9	Mirror of entries in the Audit menu **Cannot be changed**	
Network settings			
	IP address	Shows a list of the network settings entered in the Audit menu **Cannot be changed**	
	Subnet		
	Gateway		
	Def. Gatekeeper		
	Gatekeeper		
	MAC address		
Security			
	Web browser		
		free	If the phone is set to "free", Web browser can be viewed
		blocked	
		Do not change	
	Change PIN		
		Old PIN	Phone identification
		New PIN	
		Re-enter PIN	
	CTI application		
		Login mode	
		Delete login	
Switch language			
	Deutsch	User interface language	
	English		
	Français		
	Italiano		
	Español		
	Nederlands		
	Magyar		
	Slovensko		
	Polski		
	Cesky		
	Russian		
	Do not change		

Send PING		
	IP address	Checks whether the IP address can be reached
	Def. Gatekeeper	
	FTP server	
User administration		
	Number	Enter new user
	PIN	

The User menu is a local menu. It can only be accessed when the telephone is **not** registered. It is **not** sufficient to just log out the telephone via Hot Desking.

Note:

You can view the IP settings on the Web terminal via the Boot Line menu item without having to log out your telephone first.

Delete registration of the telephone in the IP Office

1st Option:

1. Unplug the power supply of the IP telephone (mains plug or, for Power over LAN, the LAN plug)

2. Press and hold the star key (*).

3. Reconnect the telephone via the mains or LAN plug.

4. Release the star key when the idle display appears.

2nd Option:

1. Press the menu key in the idle display.

2. Enter 1590.

The IP telephone is no longer registered. You can call the User menu (menu key) or the Audit menu (press the menu key and then enter the audit PIN number).

Audit menu

Boot list				
	on	Download via number keys possible		
	off (def.)	Download via number keys not possible		
	Do not change			
Use DHCP Server				
	on	Log in via DHCP		
	off (def.)	Log in manually		
	Do not change			
IP audio settings				
See "General explanations" and "IP audio settings"	Jitter buffer			
		short	Physical size of the jitter buffer	
		normal		
		long		
		automatic		
		Do not change		
	Audio Codec priority list			
		Prio1 .. Prio 3		
		Priority 1 must be set on G.711 Codec	Codec	G.711A
				G.711µ
				G.729A
			Delay	10ms
				..
				60ms
			Delete priority	
	TOS/DiffServ parameters			
		Bit 0 (off)		
		Bit 1 (off)		
		Bit 2 (off)		
		Bit 3 (on)		
		Bit 4 (off)		
		Bit 5 (off)		
		Bit 6 (off)		
Gatekeeper list				
	Create new entry (max. 1 - 9)			
		Name	"Company 1"	
		IP address	10.19.9.66	
		Save entry		
Network settings				
	IP address	192.168.0.1		
	Subnet	255.255.255.000		
	Gateway	0.0.0.0		

	Def. Gatekeeper	0.0.0.0
	Gatekeeper	10.19.9.66
	MAC address	00-07-3b-00-0c-4e.
Bit rate LAN PC (auto)		
	auto	
	10 Mbit	
	Do not change	
Test mode		
		For laboratory, development
Delete all entries		
	Deletion takes place without any further security prompts	Gatekeeper list, user defaults, Web browser - reset to the default: Web browser blocked
Telephone ID		
	Hardware type	
	Reference no.	
	Boot Loader	
	Software	
	Date loaded	
	Protocol	
	BZT approval	
	Serial number	
	MAC address	
	Supplier	
	Service class	
	Extras	
	Color	
	Memory	
	Languages	
	Service	
	Hardware revision	
	1st Installation	
Server port numbers		
	CTI	
	HTTP	
	RAS registration	
	RTP	
RTCP		
	RTCP	(on)
	FTP server	(off)

Port settings			
	VLAN settings		
		VLAN 1	
		…	
		VLAN 4	
	LAN port		
		Duplex mode	(Auto)
		Bit rate	(Auto)
	PC port		
		Duplex mode	(Auto)
		Bit rate	(Auto)
AEC handset			
	on		
	off		
	Do not change		
Sidetone level			
	Only for T3 IP Compact or if a link has been plugged in.		
Delete Bootline entries			
		Deletion takes place without any further security prompts: Telephone name, IP address, subnet mask, Gateway IP address, FTP settings, boot file name	
Software reset			
		Software reset takes place without any further security prompts	
DSP reset			
		A DSP reset takes place without any further security prompts	
Blocked features list			
	CTI application		
		free	
		blocked	
		Do not change	
	Web browser		
		free	
		blocked	
		Do not change	
	SNMP		
		free	
		blocked	
		Not changed	

Note: The call processing status of the IP telephone is unknown in the Web terminal (Stimulus telephone). This means that during a reset, it is possible that a call is inadvertently disrupted, for example due to data transfer.

Sending ping
It is possible to send a ping to the IP address of various network components from the menu. This allows you to check whether the gatekeeper, gateway, FTP server etc. can be reached from the telephone.
You can also enter an IP address manually. Enter all 12 digits of the IP address with zeros.

Menu (local menu for logged out telephone):

Menu > Send ping > Network components

Menu (local menu for logged out telephone):

Menu > Send ping > IP addr.

When you confirm your selection, an information message will inform you whether the selected network component was reached.

Using RTCP in the IP telephone

The RTCP protocol supplies current information about the quality of the voice data in the network, doing so on the basis of the RTP packets sent and received. This feedback allows the system administrator to manually adapt to network conditions the data stream generated by them (for example, by reducing the data rate in the case of a poor QoS) and to locate faults. To do so, RTCP periodically transmits so-called sender reports (SR) and receiver reports (RR). The reports received are saved for statistical purposes. (For more detailed information about RTCP: RTP - A Transport Protocol for Real Time Applications, RFC 1889, January 1996)

Activation of RTCP and the associated logging can be set in the Audit menu. Logging occurs on an FTP server. The data is filed there in csv format.

Note: An FTP server address for logging must *previously* be entered in the Web terminal.

Evaluating the csv log file
Every RTCP report received or sent is documented on a separate line. The values from the reports are shown in hexadecimal format.

The individual data are separated by semicolons in the file.

Version	Logging version
Time stamp	Time stamp DD.MM.YYYY SS
Direction	Received / sent RTCP report (RX/TX)
Report type	Packet type, type of report according to RFC 1889
SSRC source	Synchronization source identifier of the sender
NTP time sec	Sender information
NTP time frac	NTP time stamp, 64-bit time indication, when the report was sent
RTP time stamp	Time stamp, when the packet which corresponds to the RTP time stamp was sent
Packets count	Number of RTP packets which were sent
Octets count	Quantity in bytes of the voice data which was sent
SSRC report block	Synchronization source identifier of the source for which the report is determined
Fraction lost	Fraction of RTP packets not received
Cum. no. of packets lost:	The number of RTP packets lost during the connection
Highest seq. No. Rx:	Report block info
Interarrival jitter	Variance in the arrival times of the RTP packets
Max. interarrival time:	Max. time in ms of intervals at which RTP packets arrive at the IP telephone
Round trip delay	Time in ms which an IP frame requires to be transmitted to the called station and back
Cname	Textual presentation of the name of the sender of the RTCP report for the IP telephone. For example, Tenovis_IPPhone_4711@192.168.123.15

Note:

- Individual data can be missing depending on the controller chip being used.

- RTCP data for a call are only recorded if at least three RTCP reports have been sent.

- A date and time are entered in the timestamp field of the RTCP trace. These are reset to 01.01.70 00:00.00 every time the terminal is reset.

- The call number of the subscriber is not shown in the RTCP journal. The IP address of the IP Office is specified in the name field.

- The FTP server parameters can be configured and switched in the phone.

- The calculation of the time-dependent RTCP log data in the IP telephone is triggered after 10 ms.
  For operation in an IP Office, the value 20 ms is strongly recommended because of the RTP relay function (IP audio settings, page 40).

Portoverview

The following table shows all servers operating in the T3 IP telephone and the ports they use.

Server	Assigned port number (default)
DHCP client	8
CTI	32000
HTTP (TCP port)	80
RAS registration	5050
RTP	1722
SNMP (UDP port)	161
RTCP	1723
Maintenance server	35000

Settable port numbers

It is possible to configure the port numbers using

- The user interface (Audit menu) in functional mode
- The Web terminal
- The Admin tool "IP Phone Admin"

.

The ports of the following servers can be configured:

- CTI
- HTTP
- RAS
- RTP

Switching functionality and VLAN-configuration

It is possible to configure the switch using:

- The user interface (Audit menu) in functional mode
- The Web terminal
- The Admin tool

The following switch parameters can be configured:

- Speed for LAN and PC port (10/100 Mbit and automatic)
- Duplex for LAN and PC port (half, full and automatic)
- Interport VLAN configuration

Up to four VLANs can be configured. Configuration is possible using

- The user interface (Audit menu) in functional mode
- The Web terminal
- The Admin tool

The following VLAN parameters can be configured:

- VLAN ID for max. of four VLANs
- VLAN assignment to port
- Specify or change priority of frames (IEEE 802.1p) to the LAN and PC port.

The integrated switch is part of the controller chip being used. It is not possible to reset the IP telephone and switch separately. Inter-switch protocols, such as CISCO's proprietary protocol, are not supported.

To ensure that the VLAN settings in the IP telephone network are functioning, all the connected network components have to support VLAN functionality and be configured accordingly. The connected PC does **not** require a specific configuration.

Note:

- Cascading (i.e., connecting one IP telephone to the PC port of another IP telephone) is not supported!

- An IP telephone reset also disrupts the existing IP connections of a connected PC.

- If you press and hold the "OK" key when connecting the IP telephone to the power supply, the telephone immediately jumps to the configuration settings of the VLAN (after requesting the audit password).

Commissioning options for an IP telephone

Commissioning options without DHCP

An IP telephone can also be commissioned without DHCP server. Then you have to manually allocate an IP address and subnet mask for the telephone.

Basic conditions

The Web browser can be enabled or blocked.

The boot list and DHCP server are switched off.

Carry out IP setting

Without a DHCP server the IP addresses and subnet mask have to be allocated manually. The settings are carried out in the following menus:

Audit menu > Network settings > IP and

Audit menu > Network settings > Subnet

The IP and subnet address have to correspond to the network settings of the IP Office. The IP address must not appear twice in the entire network.

If there are several gatekeepers in your network or GRQ messages do not reach the gatekeeper automatically, you must also enter a default gatekeeper:

Audit menu > Network settings > Def. gatekeeper

If necessary, you have to adjust the IP audio settings in the Audit menu.

Log in user

Once the IP settings have been made, a (new) user can be logged in.

User menu > User administration > Log in new user / PIN

The telephone number and the PIN are usually defined by the configuration tool of the IP Office, the manager, and the user is notified. However, it is also possible to enter a new telephone number (a user telephone number is automatically created if the functions "Create device list automatically" and "Create user automatically" are activated in the manager under "System > H323 Gatekeeper"). A PIN is not mandatory.

Commissioning an IP telephone with DHCP

If a DHCP server already exists in the company network, it should be used. If this is the case, the DHCP server of the IP Office has to be switched off.

Configuring the IP Office using Manager Tool/PC

The PC on which the manager runs should have a fixed IP address, similarly the IP Office itself. The DHCP server in the IP Office must be deactivated or set to "Dial-in".

Settings at the IP telephone

In the IP telephone, use of the DHCP server has to be activated in the Audit menu.

Audit menu > Use DHCP server

If you have set this set to "On", the IP telephone performs out an automatic reset.
A new software is loaded if necessary, depending on the DHCP statements.

The same rules apply for the *default gatekeeper* as for commissioning without DHCP.

IP audio settings

The IP audio settings can be adjusted to local conditions.

This is generally done by the network specialist. He or she must determine the network and transmission quality. This data is then used to decide:

- Which protocol to use (G.711, ...); if necessary, a greater traffic load requires another Codec (G.729)
- Whether you want to receive small packets with little information but faster
- Whether you want to receive bigger packets with more information but more slowly. **A setting of 20ms for the packet size (delay) is strongly recommended**.
- How to determine the jitter buffer depending on these points

IP audio settings				
	Jitter buffer			
		Action	short	
			normal	
			long	
			auto	
			Do not change	
	Audio Codec priority list			
		Prio1 .. Prio n		
			Codec	
			Action	G.711A
				G.711 µ
				G.729
			Delay	
			Action	10-60 ms
			Delete priority	
	TOS/DiffServ parameters			
		Bit 0 (off)		
		Bit 1 (off)		
		Bit 2 (off)		
		Bit 3 (on)		
		Bit 4 (off)		
		Bit 5 (off)		
		Bit 6 (off)		

Before "useful communication" the IP telephone and IP Office "discuss" which shared Codec to use. Setting of the Codec on the IP Office side occurs via the manager.
The Codec is selected according to the following rules:

- The Codec set in the manager is always used, if it appears in the Prio list of the IP telephone.
- The Codec set under Prio 1 in the IP telephone is used for the manager setting "autodetect" or for a Codec unknown to the IP telephone.

Note: It is recommended not to change the Codec default setting in the IP telephone and to set the desired Codec in the manager.

IP telephone logging out / logging in again (Hot Desking)

The telephone remains registered when you log out. Once you are logged out, the user "NoUser" occupies the telephone. Further internal calls can be carried out from this telephone.

If you wish to cancel the connection to the gatekeeper / IP Office, you have to delete the registration.

Log out user from the telephone:
Menu > Logout

Log in user:

Menu > Login

A PIN is required for the login. You can occupy each external telephone which is registered via a login (Hot Desking). The previous user is then automatically logged out. Your login at another telephone is also deleted.

When you log out at the occupied telephone, the previous user is logged in again and you are logged in again at your original telephone.

Note: Automatic login and logout procedures can be modified via the manager.

What to do if start-up fails

There could be several reasons why cannot log-in your telephone:

- No gatekeeper is found.
  Enter a default gatekeeper or choose another gatekeeper from the gatekeeper list.

- The gatekeeper at which the telephone wants to log-in does not accept your old telephone number and does not allocate a new one (deactivated in IP Office Manager).
  Delete your old telephone number using the Audit menu item Delete completely.
  If necessary you must also disconnect the LAN connection and provide your telephone with a local power supply.

- The IP address of the telephone is already allocated in the LAN.
  Change the IP address of the telephone or other device.

Update the telephone software

Options

There are various ways to load new software onto an IP telephone.

1. Via Revisor laptop

   a) Connected to the IP telephone network

   b) Via a direct connection to the IP telephone (crosslink patch cable)

2. Remotely via remote dial-in to the IP telephone network. The option is especially used by the Avaya helpdesk if a dial-in into the company network is permitted.

3. Manually direct from the IP telephone (by boot list)

Prerequisites/Tools

The following conditions have to be fulfilled for a SW update:

- An FTP server has to be working in the IP telephone network. There has to be:

   a. A special configuration file **and**

   b. The new software

- On the Revisor laptop:

   a. The "IP Phone Admin" tool has to be installed **or**

   b. You have to be able to access the Web terminal of each IP telephone via a Web browser.

   Note: When a cross link patch cable is used, a plug-in power supply must be connected to the remotely powered telephone (PoL).

FTP server

The FTP server has to have an IP address which can be reached from the IP telephone network. It can run on the Revisor laptop or another PC in the IP telephone network.

The IP address does **not** have to be made known to the IP telephone if you are using the "IP Phone Admin" tool. In all other cases, the address must be made known to the IP telephone.

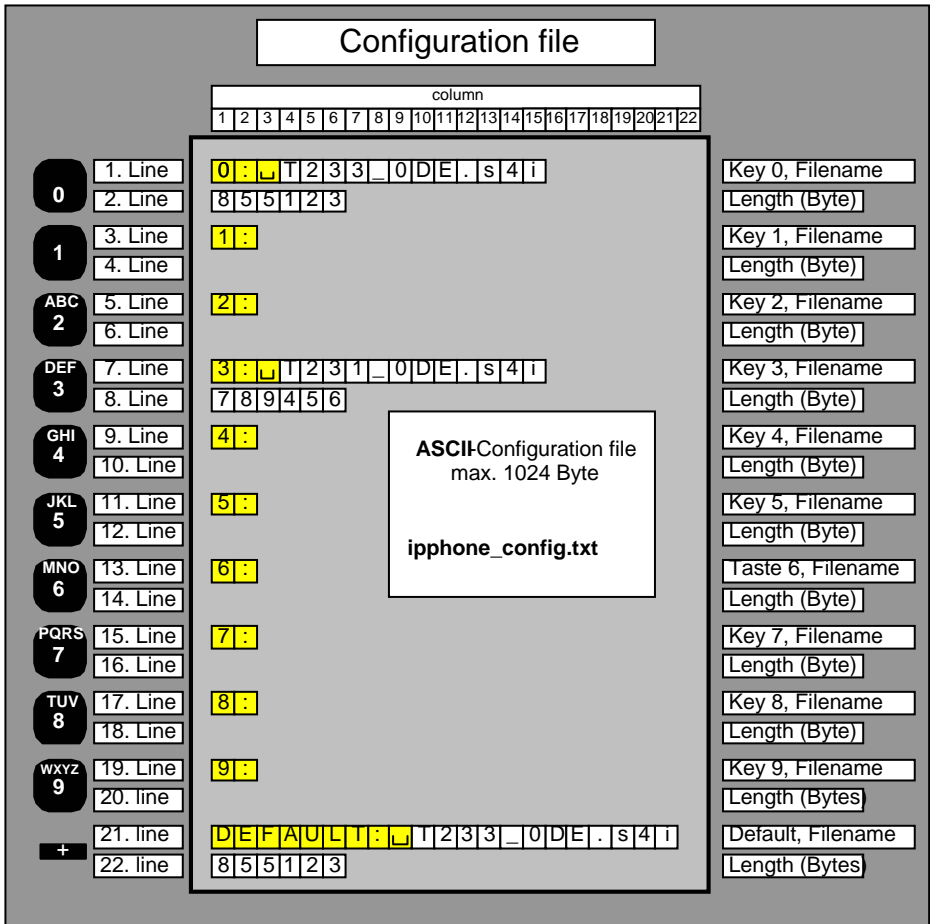**The setup and operation of an FTP server is not part of this document.**

Configuration file on the FTP server

There has to be a suitable configuration file on the FTP server. Via this file it is possible to determine a software which is then loaded into the IP telephone.

Structure of the configuration file

The diagram shows an example structure of the configuration file.

3

Configuration file

column

| | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 | 9 | 10 | 11 | 12 | 13 | 14 | 15 | 16 | 17 | 18 | 19 | 20 | 21 | 22 |

Key	Line	Content	Description
0	1. Line	0 : ␣ T 2 3 3 _ 0 D E . s 4 i	Key 0, Filename
	2. Line	8 5 5 1 2 3	Length (Byte)
1	3. Line	1 :	Key 1, Filename
	4. Line		Length (Byte)
ABC 2	5. Line	2 :	Key 2, Filename
	6. Line		Length (Byte)
DEF 3	7. Line	3 : ␣ I 2 3 1 _ 0 D E . s 4 i	Key 3, Filename
	8. Line	7 8 9 4 5 6	Length (Byte)
GHI 4	9. Line	4 :	Key 4, Filename
	10. Line		Length (Byte)
JKL 5	11. Line	5 :	Key 5, Filename
	12. Line		Length (Byte)
MNO 6	13. Line	6 :	Taste 6, Filename
	14. Line		Length (Byte)
PQRS 7	15. Line	7 :	Key 7, Filename
	16. Line		Length (Byte)
TUV 8	17. Line	8 :	Key 8, Filename
	18. Line		Length (Byte)
WXYZ 9	19. Line	9 :	Key 9, Filename
	20. line		Length (Bytes)
+	21. line	D E F A U L T : ␣ I 2 3 3 _ 0 D E . s 4 i	Default, Filename
	22. line	8 5 5 1 2 3	Length (Bytes)

ASCII-Configuration file
max. 1024 Byte

ipphone_config.txt

Description of the configuration file

- The name can be chosen at will, e.g. "ipphone_config.txt"
- The size is 1024 bytes maximum.
- It is organized into 22 rows and 22 characters
- 2 rows always belong together

Rows 1 to 20:

- Row 1: Filename of the file to be loaded
- Row 2: Length of the file in bytes
- Both rows are allocated a certain number key
- The first name row generally begins with **0:(blank)**
- The second name row generally begins with **1:(blank)**
- "**0**" ,"**1**" .. are the allocated number keys

Rows 21 to 22

- Row 1: Filename of the file to be loaded
- Row 2: Length of the file in bytes
- A default file is entered in rows 21 to 22
- Row 21 must begin with **DEFAULT:(blank)**
  Note: DEFAULT corresponds to the "+" key

Structure of the file name

For example, a software file could be named: **T233_0DE.a4i**

- **T**233_0DE.a4i   it always begins with «**T**» **T**= terminal download file

- T**223**_0DE.a4i   **233** = release ID

- T223_**0**DE.a4i   **0** = project ID

- T223_0**DE**.a4i   **DE** = country code

- T223_0DE.**a4i**   **a4i** = hardware ID

    1st position = "**a**" = T3 Comfort
    1st position = "**b**" = T3 Classic
    1st position = "**c**" = T3 Compact
    1st position = "**x**" = the telephone automatically loads the matching type-specific software.
    This setting is especially practical, since the same configuration file can be used for various telephones!

    2nd position = "**1**" = H.323
    2nd position = "**2**" = SIP
    2nd position = "**3**" = Tunnelphone functional
    2nd position = "**4**" = Tunnelphone stimulus
    2nd position = "**..**" = reserved for other protocols

    3rd position = "**i**" = software for IP telephones

- It is always eight characters long

- There is always a «**.**» between the actual filename and the extension

- The extension is always three characters long

1. Download with a Revisor laptop

The easiest way to carry out a download is via a Revisor laptop connected to the IP telephone network. Together with the "IP Phone Admin" tool, no settings then have to be carried out at the IP telephone which is to receive new software.

The conditions given above still apply for the FTP server, the configuration file stored there and the new telephone software.

Information on how to perform a download can be found in the Chapter "Software update with the IP Phone Admin tool".

2. Remote download

For the service PC, you require access to the IP telephone network in which the IP telephones onto which the new software is to be loaded are connected. If this access is available, the service PC behaves exactly the same as a Revisor laptop connected directly to the IP network. Therefore, the same applies as described under "1. Download via a Revisor laptop".

3. Download via boot list

Upon delivery the IP telephone has the settings shown on the left-hand side of the table.

- The settings of the FTP server (right-hand side) have to correspond to these settings.
- The "boot list" has to be switched on in the Audit menu or via the Web terminal.

IP telephone			FTP server, e.g. on the Revisor laptop	
IP phone factory settings				
Own IP address	192.168.0.1			
Subnet mask	255.255.255.0	->	Subnet mask	255.255.255.0
FTP server IP address	192.168.0.2	->	Own IP address	192.168.0.2
FTP username	tenovis	->	FTP username	tenovis
FTP password	ipphone	->	FTP password	ipphone
Boot file name	Ipphone_config.txt	->	Boot file	ipphone_config.txt (File must be on the laptop)

Conditions if a suitable DHCP server is available

The IP telephone is allocated its own IP address by the DHCP server. Some DHCP servers can additionally notify the IP telephone of the IP address of the FTP server and the name of the configuration file on the FTP server.

- The FTP server has to be reachable via this IP address and has to have a corresponding configuration file stored.
- The software versions stated in the configuration file have to be stored from the FTP server.
- The "boot list" has to be switched on in the Audit menu or via the Web terminal.

Carry out download

Use the configuration file to determine which software is to be loaded into the IP telephone. After pressing the corresponding number key, the software located next to this number is loaded into the IP telephone. The "+" key loads the default software.

1. Disconnect the IP telephone from its power supply.
2. Press and hold the "+" key or a number key, e.g. "3".
3. Reconnect the power supply unit or LAN plug.
4. Release the key once "Download activated" appears on the display.

The desired software is loaded into the IP telephone.

Note

- If a software download is locally started on a T3 IP telephone without software or with Stimulus software, then the message "Download Demand" is displayed for about 3 minutes (the telephone is waiting for a message from the PBX system). You can interrupt this wait time if you press the upper right function key.
- After the software has been loaded, the boot list is automatically reset to "off". The next time an individual software program is loaded, the boot list must be set to "on" again.

Software download if the software is faulty or incorrect

Theoretical background: An IP telephone with multi-platform software can retain three different software images in its flash memory at the same time. Each image resides in a separate boot loader area. One of the areas is activated. By default the software is booted from this area.

The factory determines which boot loader area is active at the time of delivery to the customer.

The active boot loader area can be identified using the Admin tool. In exceptional cases, it can also be manually selected at the telephone using the keys (refer to page 55 for the procedure):

"*" Key	Boot loader area 1	Default: Tunnel software for I55
"#" Key	Boot loader area 2	Default: Tunnel software for I5 / IP Office

The third boot loader area is intended for factory software. It is not accessible via the Admin tool.

Commissioning with emergency download (Emergency download) of the software

If the software stored in the active boot loader area is damaged and the IP telephone can no longer be operated ("No software loaded" in display), a software download with default parameters can be activated.

To do this, the download is started while pressing and holding down the "Clear" key (instead of a number or "+" key), as described on page 25. The procedure is as follows:

1. The IP telephone loads the configuration file from the FTP server. All the default settings from production are used. If the format of the configuration file is correct, it will be saved in the flash memory. **The FTP server must have the IP address 192.168.0.2 and the subnet mask 255.255.255.0**.

2. The IP telephone once again sets up connection with the FTP server and loads the software file.

Note

- If the software is damaged, another working image of another boot loader area will **not** be started automatically.

- After downloading the software, the IP telephone accepts again the last values saved for the settings.

- No DHCP is performed with this emergency download, even if the corresponding setting was set to "on". The DHCP will be performed again, if necessary, during the course of the next commissioning.

How to proceed if the software is incorrect or if the emergency download fails

In this case the following procedure may possibly lead to success:

1. At the boot operation, change the boot loader area on the telephone to "Working image" (see above for procedure). This can also be software for another system! Load this software.

2. Using the Admin tool, reset the boot loader area to the area which contains the defective image.

3. Using the Admin tool, activate a software update of the Stimulus software (I5, IP Office).
   This will load the correct software in this area.

Web terminal

Web terminal is a user-friendly administration tool for Avaya T3 IP telephones. It runs on every modern Web browser that supports Java.

The Web terminal allows you to carry out a lot of IP-specific settings, some of which can also be directly reached at the IP telephone via various menus.

A prerequisite for using the Web terminal is that the service PC and the telephone can establish an IP connection.

Releasing and blocking

The entry "Web browser" is only available in the User menu if it has been released in the "Blocked features list" in the Audit menu. Then the user can release and block functionality in the "Security" menu.

Blocking and releasing the Web browser does not affect the blocked features list. In other words, blocking an entry in the "Security" User menu does not block the entry in the blocked features list!

Resetting does not block the Web browser.

You can view the status of the Web browser after loading new software under "Security". If it is not visible, it is possible that the entry was blocked in the blocked features list.

Accessing Web terminal

Using a recent standard browser (Java must be activated) the T3 IP telephone in question can be called from the service PC using its IP address. The IP telephone will then act as a Web server and in the browser will display various input windows, the Web terminal.
The windows are shown schematically here. The following symbols are used:

Meaning within the mask being described	
Deutsch	**E.g.: "Deutsch" input button**
	Input box
▼	**Drop-down menu, cursor**

After the call via the browser, the IP telephone registers with the login window.

Web Terminal Version: Stimulus	
PIN	[ ] Enter **PIN** from the **Security** User menu**.** Enter xxxx or **1590**
Status	E.g.: Not connected
Deutsch **English** **Connect**	**Available options:** • **Deutsch** (web browser language) • **English** (web browser language) • **Connect** (establish connection to IP telephone)

The Main menu appears after login.

Web terminal: Auditor mode Main menu		
E.g.: MAC address: 00-07-3b-00-0c-23 E.g.: Software version: T233_0DE.s4i E.g.: Boot Loader version: B02.01 E.g.: Call number: 51		**Display of** (when connection established)**:** • **MAC address** • **Software version** • **Boot Loader version** • **Number** • **DSP versions**
Bootline		**Status:** e.g.: Data unchanged
Registration & Admission		**Status**: e.g.: Data unchanged
IP Audio Settings		**Status:** e.g.: Data unchanged
VLAN settings		**Status:** e.g.: Data unchanged
Telephone ID		**Status:** e.g.: Data unchanged
Server port number configuration	**Selection**	**Status:** e.g.: Data unchanged
RTCP settings		**Status:** e.g.: Data unchanged
RTCP journal		**Status:** Cannot be changed
LAN statistics		**Status:** Cannot be changed
Send data		
Cancel		

The following windows can be called from the Main menu.

- Bootline
- Registration & Admission
- IP Audio Settings
- VLAN settings
- Telephone ID
- Server port number configuration
- RTCP settings

- RTCP journal
- LAN statistics

On the following pages there are overviews of these windows.

Web terminal: Bootline	
✓ **DHCP** (**Audit menu**: Use DHCP Server)	✓ = DHCP server active
☐ **Boot list**	If activated, older software versions can be loaded from the directory. Default: Not activated.
Telephone name:	E.g.: Tenovis_IPT
IP address (**Audit menu**: Network settings)	E.g.: 192.168.0.1
Subnet mask (**Audit menu**: Network settings)	E.g.: 255.255.255.0
Gateway IP address (optional) (**Audit menu**: Network settings)	0.0.0.0
FTP server IP address	E.g.: 192.168.0.2
FTP user name	E.g.: tenovis
FTP password	E.g.: ipphone
Boot file name	E.g.: ipphone_config.txt
Accept	
Last settings	

Note: In the table above, the rows for FTP server IP address, FTP user name, FTP password and Boot file name are grouped with "Boot/Download area of the IP telephone". The Accept and Last settings buttons are grouped with "Selection".

Tenovis Web terminal: IP Audio Settings		
Jitter buffer size: **(Audit menu**: IP audio settings – Jitter Buffer**)**	automatic short long normal ▼	

	Codec	**Delay (ms)**
Priority 1: **(Audit menu**: IP audio settings - Audio Codec priority list**)**	G.711A G.711µ G.729A ▼ none	10 20 30 ▼ 40 50 60
Priority 2: **(Audit menu**: IP audio settings - Audio Codec priority list**)**	G.711A G.711µ G.729A ▼ none	10 20 30 ▼ 40 50 60
Priority 3: **(Audit menu**: IP audio settings - Audio Codec priority list**)**	G.711A G.711µ G.729A ▼ none	10 20 30 ▼ 40 50 60

TOS/DiffServ parameters	
	Bit 0 (off)
	Bit 1 (off)
	Bit 2 (off)
	Bit 3 (on)
	Bit 4 (off)
	Bit 5 (off)
	Bit 6 (off)
Accept	**Selection**
Last settings	

Web terminal: Registration & Admission	
-- Default Gatekeeper -- (**Audit menu**: Network settings)	
IP address	**143.1.56.111**
-- Gatekeeper List -- (**Audit menu**: Gatekeeper list)	
Name	
IP address	
Adding a word Delete	E.g.: GK, 10.19.8.74
Accept	**Selection**	Data is accepted in the web terminal
Last settings		The original data is applied

Web terminal: Telephone ID
Display of data as in the Audit menu

Web terminal: Server port numbers	
HTTP	E.g.: 80
RTP	E.g.: 1722
RAS registration	E.g.: 5050
CTI	E.g.: 32000
Accept	**Selection**	
Last settings		

Web terminal: RTCP settings		
		RTCP on
		Save on FTP server
Save the RTCP data		
FTP server IP address	E.g.:0.0.0.0	
Username	E.g.:	
Password		
Default file name		
Accept	**Selection**	
Last settings		

Web terminal: RTCP journal
List of calls made
Associated RTCP data

Web terminal: LAN statistics
Display of messages

Log Off

You log out of the Web terminal by clicking on "Cancel" in the main window.

If you have not carried out an entry in the Web terminal for 5 minutes, you will receive a message that the Web terminal will automatically shut down after one minute. Each entry resets both timers.

Sending data to the IP telephone

Data entered into the Web terminal is sent to the IP telephone after actuating

the "**Send data**" button in the Main menu.

The IP telephone accepts this data and carries out a subsequent reset. Changed default data is reset to the default setting.

For example: Security – Web browser – "**free**" is reset to "**blocked**".

Available IP functions

Function		User menu	Audit menu
Bootline	Phone IP address	r/w	
	Subnet mask	r/w	
	Gateway IP	r/w	
	FTP Server IP	r/w	
	FTP Username	r/w	
	FTP Password	r/w	
	Filename	r/w	
	Bit rate on the LAN	r/w	x
	Phone Name	r/w	
	Bootlist on/off	r/w	x
	DHCP on/off	r/w	
Registration & Admission	Default Gatekeeper IP	r/w	
	Gatekeepers list	r/w	
IP Audio Settings	Jitter buffer size	r/w	
	Codec Prio 1- 3	r/w	
	Delay 1-3	r/w	
	DiffServ/TOS Bits	r/w	x
LAN statistics (statistical values of the Ethernet controller)		r	x
Telephone ID	Serial number	r/w	x
	Hardware version	r/w	x
	Reference no.	r/w	x
	Additional device	r/w	x
	Date loaded	r/w	x
	Protocol	r/w	x
	BZT approval	r/w	x
	Supplier	r/w	x
	Service class	r/w	x
	Color	r/w	x
	Memory	r/w	x
	Service date	r/w	x
	Hardware change	r/w	x
	1st Installation	r/w	x
	Software type	r/w	x
	Hardware type	r/w	x
Block Web terminal from further access			
Call number display		r	
MAC address display		r	

Software version display			r	
Boot Loader version display			r	
Integrated switch	Show/Configure Duplex (half/full/automatic)		r/w	x
	Show/Configure speed (10/100/Automatic)		r/w	x
VLAN	Show/Configure ID		r/w	x
	Show/Configure priority		r/w	x
	Show/Configure VLAN/port allocation		r/w	x
	LAN/PC port configuration	Duplex	r/w	x
		Bitrate	r/w	x
DSP	Display of DSP software version		r	
	Display of DSP parameter set version		r	
	DSP API Version		r	
Module	Parameter version module		r	
RTCP settings	FTP server IP address		r/w	x
	User name		r/w	x
	Password		r/w	x
	Default file name		r/w	x
	RTCP on		r/w	x
	Save on FTP server		r/w	x
RTCP Journal	Journal data and associated RTCP data		r	x
Server port number configuration	CTI		r/w	x
	HTTP		r/w	x
	RAS		r/w	x
	RTP		r/w	x

IP Phone Admin

This tool makes it easier to commission several IP telephones simultaneously, makes it possible to monitor statistical values, and allows fast port configuration of the installed switches and the VLAN allocation.

A detailed description of all the functions of this tool can be found in the corresponding operating instructions.

Software update with the "IP Phone Admin" tool

IP Phone Admin can run remotely on a service PC, or on a Revisor laptop in the IP telephone network. The software update can be carried out in various ways:

Software update

This automatically updates the telephone software. The tool requests the address of the FTP server and the access data. Once the telephone has obtained the update trigger, the software is loaded after a short delay. Once the FTP server parameters have been entered, a list of the configuration files found on the FTP server is displayed.

User-confirmed software update

This function is currently not supported.

Scheduled software update

This function is currently not supported.

Updating the Boot Loader

⚠️ The power supply to the IP telephone must not be interrupted while the bootstrap is being updated. A loss of power or a reset during this phase will make the telephone unusable.

While the Boot Loader is being downloaded, the LEDs display status information. This involves all the LEDs.

LED status messages	
Continuously on	Boot Loader file being loaded (normal procedure)
Rapid flashing (50 ms on / 50 ms off)	New Boot Loader being written to flash memory (normal procedure)
Slow flashing (500 ms on / 500 ms off)	*Cause:* Corrupt data in the flash memory after writing -> old Boot Loader is rewritten to flash memory.
	Effect: Updating has not taken place; old Boot Loader remains in the IP telephone; telephone will work after a reset
	Remedy: Internal problem with telephone; start Boot Loader update procedure again
Short flash (200 ms off / 50 ms on)	*Cause:* Something wrong with the Boot Loader file
	Effect: Update not successful, old Boot Loader remains in IP telephone, IP telephone can be operated after reset
	Remedy: Check the Boot Loader file; it might not be the right one for the hardware
Interrupted rapid flashing (500 ms rapid flashing / 500 ms off)	*Cause*: Boot Loader can no longer function properly; corrupted data in the flash memory
	Remedy: Do not interrupt power supply; start Boot Loader update procedure again

Features

The following table describes the functions that are supported when using an IP telephone:

Function	
Automatic software update of several phones	x
Manual software update of several phones	no
Boot Loader update of several phones	x
Display of the software and Boot Loader version numbers of several phones	x
Display and export of the LAN statistics of several phones	x
Display and export of the MAC addresses of several phones	x
DSP Reset	x
Definition of file extension for KuDa and TeBu files	no
Loading IP list from SQL database	x
Configuration of columns to be displayed	x
Selection of the FTP server type to be used	X *1
Sorting of the displayed table	x
Automatic scheduled software update	No No time available on the phone
Automatic reset of several phones	x
Manual reset of individual phones	x
Display/Editing of switch settings	x
Display/Editing of VLAN settings	x
Display/Definition of port numbers	x
RTCP journal displays	x
Convert RTCP journal into cvs Excel format	x
Show status of the last FTP access	no

*1 select Microsoft FTP server for operation on IP Office

Changing the telephone

If you have to replace an IP telephone, follow the same procedure as for logging in a new telephone for the first time.

- You do not have to back up data
- The IP address and gatekeeper address from the old telephone must be used
- You must enter the previous call number when you log in

Delete all bootline entries (expanded Delete all)

The "Delete all entries" function can be configured to reset the Bootline to default values in addition to deleting customer data.

Unprinted keypad (Alphabetic keypad)

The alphabetic keypad in T3 IP Comfort is unprinted. However, various templates which can be stuck onto the keypad are supplied with the telephone.

The desired key assignment can be set via the menu **Settings > Alphabetic keypad**. The options AZERTY, QWERTZ and QWERTY (assignment of the six left-hand keys on the top row of letters) are available.

SNMP (Simple Network Management Protocol)

An SNMP agent is integrated into IP telephones. This allows you to access telephone data via the SNMP protocol. These data are organized using the MIB (Management Information Base) tree structure.

The SNMP agent from WindManage SNMP is integrated into the IP phone. This agent

- Implements the SNMP protocol completely
- Saves and retrieves data from the MIB
- Can signal incoming events to the manager
- Receives packets from an NMS via the UDP level of a network connection
- Accepts a received packet, decodes it and translates it from ASN.1 into a C structure
- This structure is executed and a response generated. This response is coded and packed in ASN.1
- The packet is then returned to the NMS via the UDP connection.

The definitions of the MIB correspond to the SMI standard RFC 1155. MIB-I and MIB-II are defined in RFC 1156 and RFC 1213.

The IP phone does not however support the MIB II standard. Specific objects are defined in a private Tenovis MIB and made available. These objects have read-only access. They are listed in the following table. Limitations are due to chip hardware.

The SNMP agent in the telephone can be accessed using various tools (HP OpenView, for example). Each tool has its own way of displaying the telephone data obtained.

Objects	Description	IP telephone	LAN port	PC port	Phone port	Access
systemUpTime	The time (in minutes) since the network management portion of the system was last re-initialized	x				Read
SystemContact	The textual identification of the contact person for this managed node, together with information on how to contact this person	x				Read
systemName	An administratively-assigned name for this managed node	x				Read
VLAN Settings	**Hardware T8300U only**					
VLAN ID	**Hardware T8300U only**	x				Read
Priority	**Hardware T8300U only**	x				Read
Name	**Hardware T8300U only**	x				Read

Port Settings						
Speed	Current bandwidth in Mbits per second		x	x		Read
Duplex	The current duplex mode of the interface		x	x		Read
VLAN assignment	**Hardware T8300U only**		x	x	x	Read
Connected/not connected	Current Operation status of the interface		x	x		Read
HardwareVersion	Version of the hardware	x				Read
SoftwareVersion	Version number of the software	x				Read
BootloaderVersion	Version number of the bootloader	x				Read
OrderNumber	Order number of the terminal	x				Read
TunnelLost	The Number of lost tunnel connections after last reset	x				Read
RegistrationCounter	The Number of RAS registrations after reset	x				Read

The SNMP versions v1 and v2c supported by VxWorks provide little security. Version v3, which is also supported by VxWorks, has security measures implemented that take the shortcomings of SNMP versions v1 and v2c into consideration. Messages can be logged and therefore third parties have fast access to all features accessible via SNMP.

For this reason, configuring parameters in the phone/switch via SNMP is not supported.

The HP OpenView tool is used as a management console to test the functionality and compatibility of the IP telephone SNMP agent.

Releasing or blocking SNMP feature

To block or release the feature, open the Audit menu and select the "Blocked features list" menu item.

You can make the following settings under "SNMP"

- FREE: Telephone data can be accessed via SNMP.
- BLOCKED: The SNMP agent in the IP telephone is not in operation.

Abbreviations

AEI	**A**dditional **E**quipment **I**nterface
CAPI	**C**omputer **A**ssisted **P**ersonal **I**nterviewing
CODEC	**Co**der/**Dec**oder
CTI	**C**omputer **T**elephony **I**ntegration
DC	**D**irect **C**urrent
DHCP	**D**ynamic **H**ost **C**onfiguration **P**rotocol
FTP	**F**ile **T**ransfer **P**rotocol
GND	**G**rou**ND**
GK	**G**ate**K**eeper
GRQ	**G**atekeeper **req**uest
HK	**L**istening **C**ircuit
HTTP	**H**ypertext **T**ransfer **P**rotocol
IP	**I**nternet **P**rotocol
ISA	**I**ndustry **S**tandard **A**rchitecture
LAN	**L**ocal **A**rea **N**etwork
Ltg	**L**ine
MAC	**M**edium **A**ccess **C**ontrol
NC	**N**ot **C**onnected
PPPoE	**P**oint to **P**oint **P**rotocol **o**ver **E**thernet
RD	**R**eceive **D**ata
Rufnr	Number
RTP	**R**eal-Time **T**ransport **P**rotocol
RTCP	**R**eal**T**ime **C**ontrol **P**rotocol
SK	**S**peech **c**ircuit
SNMP	**S**imple **N**etwork **M**anagement **P**rotocol
TCP	**T**ransmission **C**ontrol **P**rotocol
TD	**T**ransmit **D**ata
UDP	**U**ser **D**atagram **P**rotocol
VCM	**V**oice **C**ontrol **M**odule
WWW	**W**orld **W**ide **W**eb

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