



Configuration Manual

Turbine Compact IP Video Stations

TCIV-2

TCIV-3

TCIV-6

About this Document

Document Scope

This document describes the setup procedure and configuration of the various models of the Turbine Compact IP Video station series.

Station Firmware Version: VSF-Turbine 4.7

Product	Item Number
Turbine Compact IP Video Station - TCIV-2	1008115020
Turbine Compact IP Video Station - TCIV-3	1008115030
Turbine Compact IP Video Station - TCIV-6	1008115060

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1.1	9.5.2016	HKL	Published	Same IP for camera & station
1.2	20.6.2016	HKL	Published	New screenshots IP Desktop
1.3	22.8.2016	HKL	Published	Pulse and SIP for IP Desktop
1.4	27.9.2017	HKL	Published	ITSV-1, SW 4.7

Related Documentation

For further information, refer to the following documentation:

Doc. number	Documentation
A100K11194	Turbine IP Stations Technical Manual
A100K11625	Turbine Compact IP Video Station Mounting Guide
A100K11293	Turbine Compact IP Station Getting Started for Alphacom
A100K11335	Turbine Compact IP Station Getting Started for SIP
A100K11336	Turbine Compact IP Station Getting Started for Pulse
A100K11664	IP Desktop Station with Video Display Manual
A100K11619	VS-IMT User Manual
A100K11705	ITSV-1 Video Phone Quick Installation Guide

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1 Turbine Compact IP Video Stations

All IP Video stations in the Turbine Compact series offers audio features such as: HD voice quality, Open Duplex, Active Noise Cancellation, MEMS microphone, a 10W Class D amplifier and our unique speaker grille design. The Video camera features wide FoV HD Video, Digital PTZ, and support for H.264 or MJPEG.

There are three station models in the Turbine Compact IP Video series:



Figure 1 Turbine TCIV-2 / TCIV-3 Station Keys & Functions



Figure 2 Turbine TCIV-6 Station Keys & Functions

2 Station Connections

2.1 External Connectors on IP Video Station

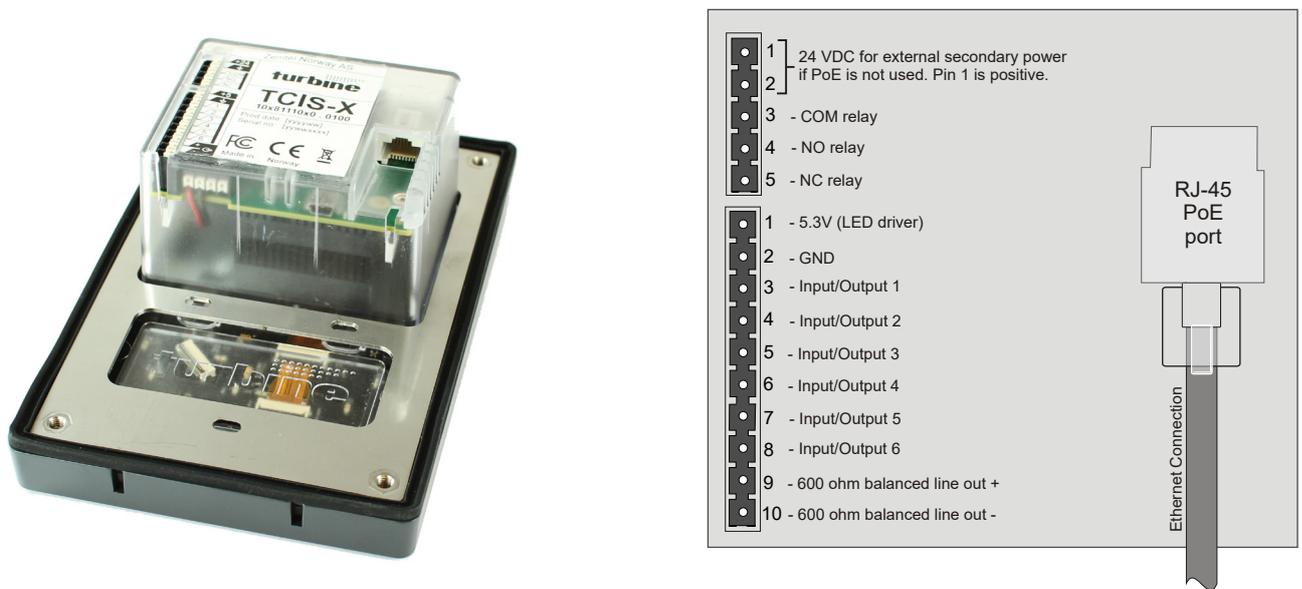


Figure 3 External Connectors on IP Station

The following table is an overview of the main connectors involved when installing the Turbine IP Stations.

Ethernet / Power	10/100 Mbps Ethernet RJ-45 port for LAN (uplink) connection. Supports PoE (802.3af). Draws power from either spare line or signal line.
Secondary Power	24 VDC (16 – 48 V) secondary power is provided from an external adapter.
Relays	There is one Double Throw relay contact with 60W switching power. COM, NO, NC contacts are provided. Max: 250VAC/220VDC, 2A, 60W.
Input/Output	6 general purpose I/Os are available. Each I/O can be configured as either button input or LED driver.
Audio Line Out	A balanced 600 ohm audio line out with induction loop signal

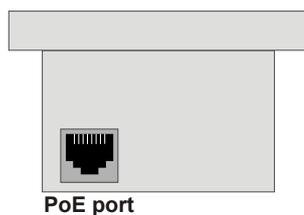
2.2 Power Supply

The Turbine Station supports Power over Ethernet (PoE, IEEE 802.3 a-f) where power can be drawn from either the spare line or signal line.

If PoE is not available, the Turbine Station can be connected to a 24 VDC local power supply.

2.3 Network Connection

There is one RJ-45 port located on the Turbine station that is used as the PoE/LAN port.



2.4 Input/Output Connections

There are 6 I/O connection options for the Turbine Station.

These connections are used as relay contacts for door lock control and external I/O devices.

3 Starting Up the Video Station

The Turbine Video Station features an embedded web interface, which allows users to log in via a standard web browser.

- ① **Software upgrade procedure for the video station is the same as for the audio stations in the Turbine series. For further details, see A100K11194 Turbine IP Stations Technical Manual.**

To start up the station, your PC and the IP station have to be connected together via a PoE switch using network cables:

1. Connect the PC to the PoE switch
2. Connect the PoE port on the IP station to the PoE switch

When the Turbine Video Station is connected to the network, the **IP address** of the station is automatically obtained in one of two ways:

1. An IP address is obtained from a **DHCP server** if there is one.
2. If there is no DHCP server, an IP address in the range **169.254.x.x** will be assigned.

To make the station speak its IP address:

- Press the **call button** on the station
- when the station is not registered yet



At commissioning, the IP Video station needs to be configured to enable it to be used as:

- Station subscribed to an AlphaCom server
- SIP station
- Pulse station

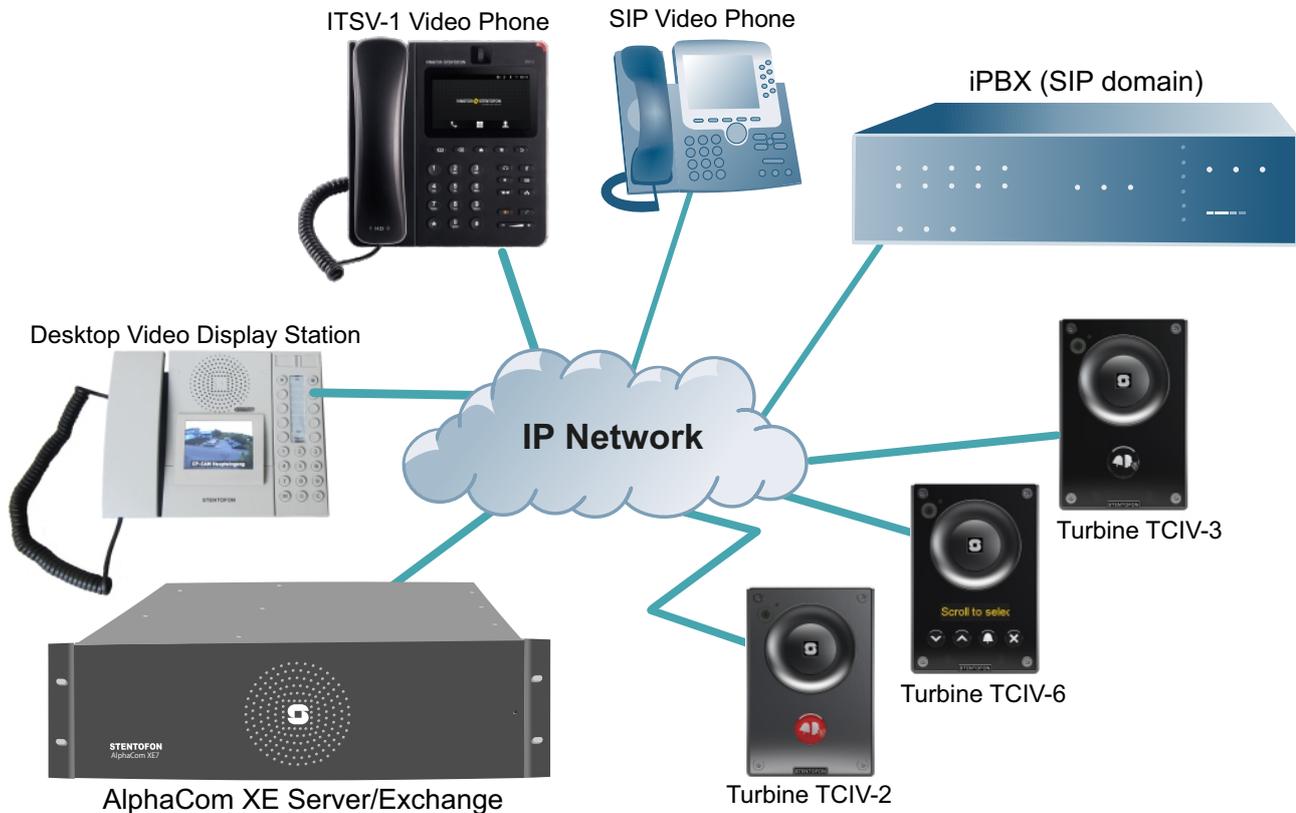


Figure 4 Turbine Video AlphaCom/Pulse/SIP System

4 Turbine Video Settings

After logging into the station via the web interface:

- Select menu option **Video Settings**

The parameters for video settings are as shown below:

Video Settings	
Description	Configuration
Video mode:	<input type="radio"/> H264 RTP <input checked="" type="radio"/> MJPG HTTP
Enable Video:	<input checked="" type="checkbox"/>
Resolution:	240P
Frames per second	15 fps
Camera IP address and port:	10.5.101.46:8090
Enable HTTP basic authentication:	<input type="checkbox"/>
Video setup mode:	Default

Advanced Settings	
Description	Configuration
Lens distortion correction:	<input checked="" type="checkbox"/>
Night mode:	<input type="checkbox"/>
Zoom [1.00 2.50]x	1
Color saturation [0 255]:	128
Contrast [0 255]:	128
Brightness [0 255]:	128
Backlight compensation:	2

Video Mode

This parameter defines whether the video stream will be in either of the two modes: **MJPG** in http or **H.264** in RTP.

Enable Video

This parameter defines whether calls made from the station will be video calls.

Camera IP address and port

This is the IP address and port number of the camera streaming the video to a web browser or video display station. **The video camera and the Turbine station have the same IP address.**

Lens distortion correction

Enabling this parameter will correct the “fish eye” effect that can occur on the edges of the video image.

Night Mode

Enabling this parameter will make it possible for the camera to record in low-light conditions.

Zoom

This parameter sets the digital zoom of the video image. The zoom level range is 1.0 to 2.5. Once it is set higher than 1.0 it allows for offsetting the view horizontally and vertically. The offset range is -100 to 100.

Color saturation

This parameter sets the color saturation of the video image. The range is 0 to 255. Default value is 128.

Contrast

This parameter sets the contrast of the video image. The range is 0 to 255. Default value is 128.

Brightness

This parameter sets the brightness of the video image. The range is 0 to 255. Default value is 128.

Backlight compensation

This parameter sets the backlight compensation for the video image. The range is from 0 to 5. The default value is 2.

5 AlphaCom Configuration

The Turbine Video Stations are connected to the AlphaCom XE server/exchange. The AlphaCom XE server/exchange includes all main service configurations for the IP stations and only a minimum configuration is needed to be carried out on the actual station.

In AlphaCom mode, the Turbine Video stations are used together with:

- IP Desktop Station with Video Display (Item Number: 1408001635)
- ITSV-1 Video Phone (Item Number: 1490001010)

① Configuration of the non-video part of the station such as audio and I/O settings is described in the manual: **A100K11293 Turbine Compact IP Station Getting Started for AlphaCom.**

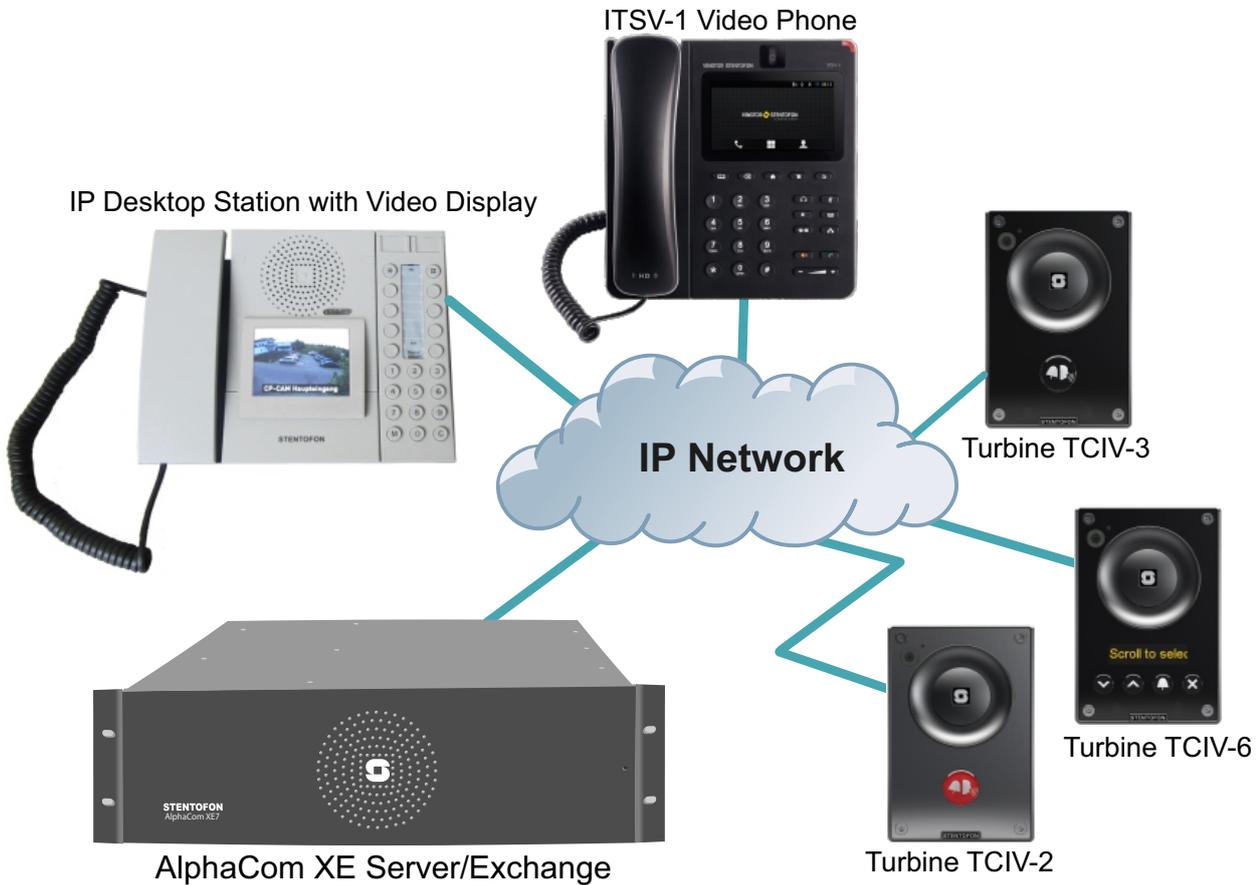


Figure 5 AlphaCom Video Intercom System

5.1 Logging into the Station

Access the station by logging into the web interface using a standard web browser:

1. Open a web browser
2. In the browser's address bar, type the station IP address and press the ENTER key
- The station login page will be displayed.

To log into the station:

1. Click **Login**
2. Enter the default User name: **admin**
3. Enter the default password: **alphaadmin**



The **Station Information** page will now be displayed, showing the IP station configuration and status.

5.2 Station Main Settings

- Click **Station Main > Main Settings** to access the page for configuring station mode and IP parameters.

The screenshot shows the 'Station Main Settings' configuration page. The navigation tabs at the top are 'Station Main', 'Station Administration', 'Advanced Alphascom', and 'Advanced Network'. The left sidebar shows 'Station Information' and 'Main Settings'. The main content area is divided into sections:

- Station Mode:** Radio buttons for 'Use Alphascom' (selected), 'Use Exigo', 'Use SIP', 'Use Pulse', and 'Use Pulse Server'.
- Product Model And Accessory:** A dropdown menu for 'Model' set to 'Video Normal (TCIV-2, TCIV-3)'.
- Registration Settings:** Fields for 'AlphaCom IP-address' (10 - 5 - 101 - 40) and 'Directory Number' (2222).
- IP Settings:** Radio buttons for 'DHCP' and 'Static IP' (selected). Fields for 'IP-address' (10 - 5 - 101 - 46), 'Subnet-mask' (255 - 255 - 255 - 0), and 'Gateway' (10 - 5 - 101 - 1). Other fields include 'DNS Server 1' (10 - 5 - 2 - 19), 'DNS Server 2' (0 - 0 - 0 - 0), 'Hostname' (zenitel063a41), and checkboxes for 'Disable Reset to Factory default settings using frontboard and I/O', 'Read IP Address', and 'Ethernet Speed 10 Mbit/s'.

A 'Save' button is located at the bottom of the form.

Station Mode

- Select the **Use Alphascom** radio-button

Product Model And Accessory

- Model**
Select one of the options from the drop-down box :
 - **Video Normal (TCIV-2, TCIV-3)**
 - **Video Scrolling Station (TCIV-6)**

Registration Settings

- AlphaCom IP-address**
 - Enter IP address of AlphaCom in which TCIV is to be registered as a subscriber
- Directory Number**
 - Enter the directory number of TCIV (e.g. 2222)

IP Settings

- Static IP** – Select this option if the IP station shall use a static IP address. Enter values for:
 - **IP-address: IP address of TCIV** (e.g. 10.5.101.46)
 - **Subnet-mask: Enter subnet mask**
 - **Gateway: Enter Gateway IP address**
 - **DNS Server 1** (option for network administration)
 - **DNS Server 2** (option for network administration)
 - **Hostname** (option for network administration)

Read IP Address

- Check the **Read IP Address** box to enable an unregistered station to speak its IP address when the call button is pressed.
- Click **Save** followed by **Apply** to apply the new configuration settings.

5.3 AlphaCom Configuration for ITSV-1 Video Phone

- Vingtör-Stentofon ITSV-1 Video Phone - item no. 1490001010

5.3.1 Video Settings for ITSV-1

To configure video settings:

- Select **Advanced Alphacom > Video**

The screenshot shows the configuration interface for the AlphaCom system. The top navigation bar includes 'Station Main', 'Station Administration', 'Advanced Alphacom' (highlighted), and 'Advanced Network'. On the left, a sidebar menu lists 'Audio', 'I/O Settings', 'Sound Detection', 'Time Settings', and 'Video' (highlighted). The main content area is titled 'Video Settings' and contains a table with the following parameters:

Description	Configuration
Enable Video:	<input checked="" type="checkbox"/>
Resolution:	480P ▼
Frames per second	15 fps ▼
Camera IP address and port:	10.5.101.46 8090
Enable HTTP basic authentication:	<input type="checkbox"/>
Video setup mode:	Default ▼

Below the 'Video Settings' table is the 'Advanced Settings' section, which contains another table:

Description	Configuration
Lens distortion correction:	<input checked="" type="checkbox"/>
Night mode:	<input type="checkbox"/>
Zoom [1.00 2.50]x	1
Color saturation [0 255]:	128
Contrast [0 255]:	128
Brightness [0 255]:	128
Backlight compensation:	2 ▼

At the bottom of the configuration area is a 'Save' button.

- Select or set values for the parameters:

Enable Video: Check box to enable video calls

Resolution: Select **480P**

Frames per second: Select **15fps**

Camera IP address and port: Enter the **port number** - default is **8090**

Video setup mode: Select **Default**

① **The video camera and the Turbine station have the same IP address.**

- The video stream from the camera can be viewed by entering the IP address and port number in a web browser, e.g. **10.5.101.46:8090**

① **Video calls in the AlphaCom system are made in MJPG mode only.**

① The same **IP address** (e.g. **10.5.101.46**) and **port number** (e.g. **8090**) set here must be entered into the **Camera Settings** of the ITSV-1.

- Click **Save**
- Click **Back to config page**

5.3.2 ITSV-1 Phone Settings

- Log into the ITSV-1 phone interface by entering its IP address in a browser on your PC



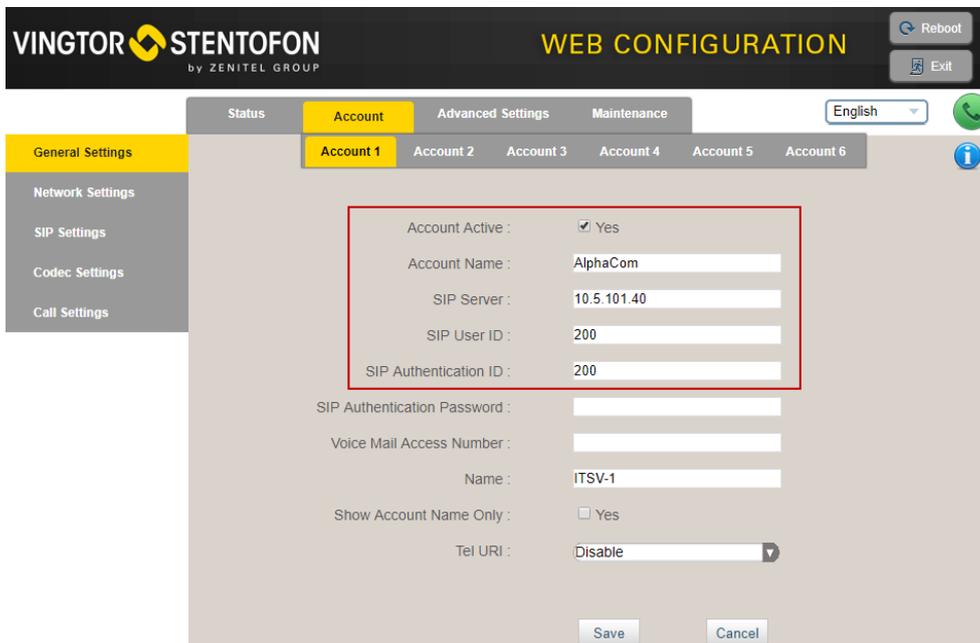
Login Credentials

Username: **admin**

Password: **alphaadmin**

5.3.3 ITSV-1 Account Setup

- Select **Account > Account 1 > General Settings**



- Enter the values shown above for the parameters

Account Active: Check **Yes** box

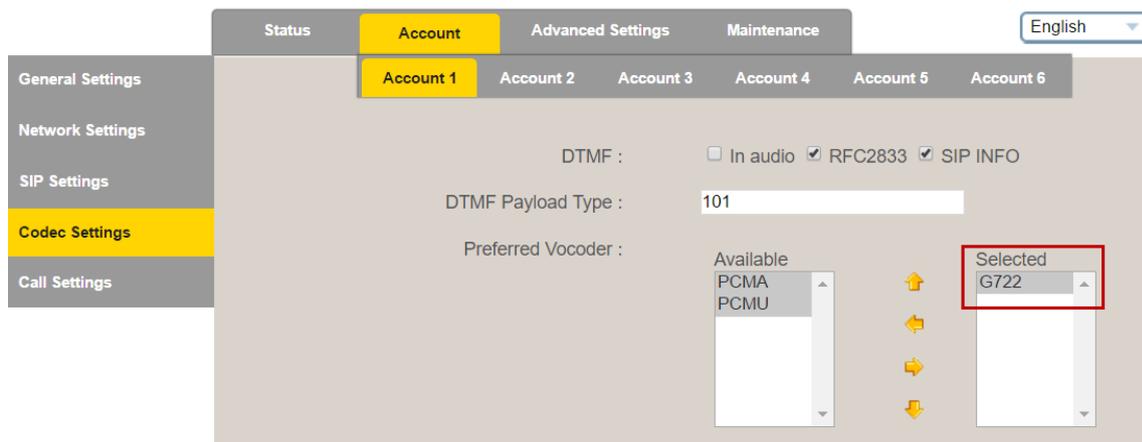
SIP Server: IP address of **AlphaCom** server (see **Main Settings** in TCIV)

SIP User ID: Directory Number of **ITSV-1** phone

SIP Authentication ID: Same as SIP User ID

5.3.4 ITSV-1 Audio Codec Settings

- Check in AlphaPro under **Users & Stations** the codec that has been selected for the SIP phone (normally **G722**)
- Select **Account 1 > Codec Settings**

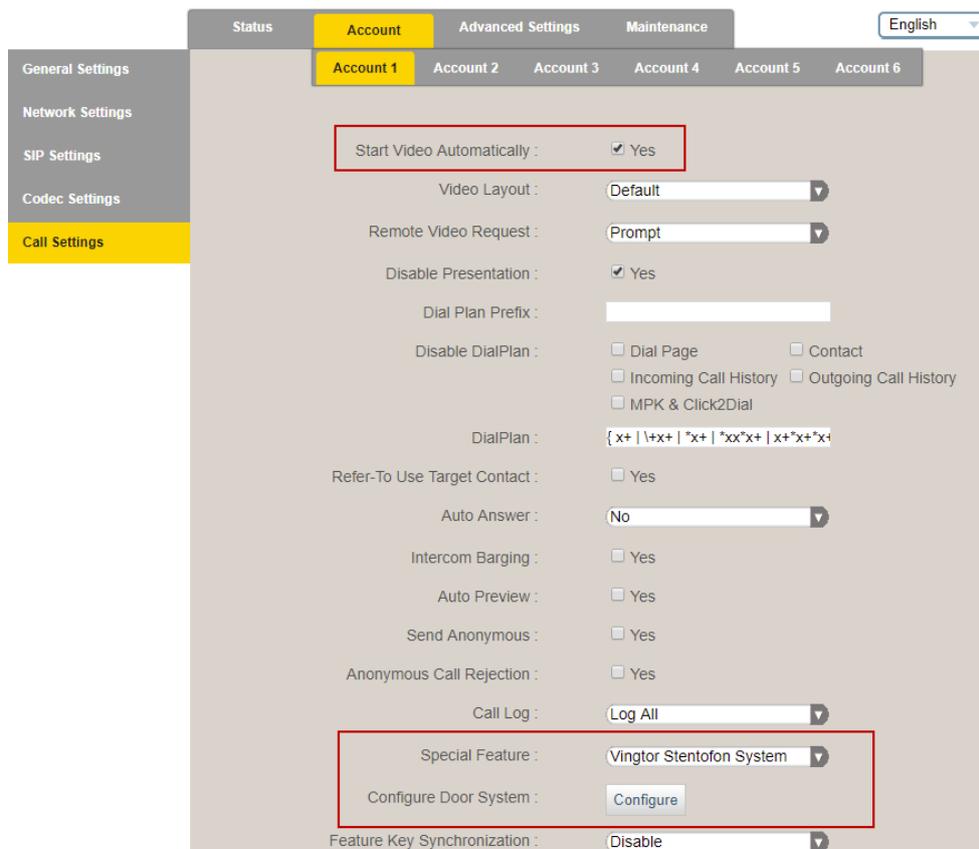


- Remove all codecs from the **Selected** list except the one defined in AlphaPro, i.e. **G722**.

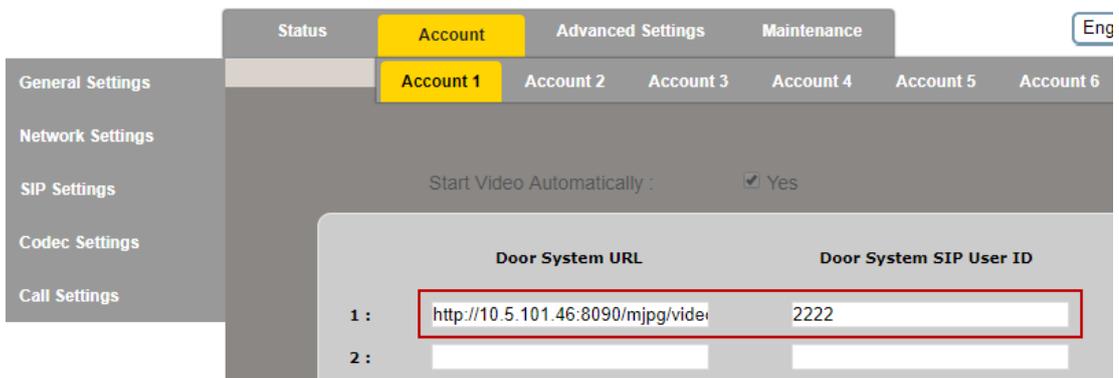
5.3.5 ITSV-1 Video Configuration

The video is streamed in MJPEG format directly from the TCIV camera to the ITSV-1. The AlphaCom server is not involved in the video stream. The TCIV camera must have a static IP address.

- Select **Account 1 > Call Settings**



- Click the **Yes** box for **Start Video Automatically**
- Select **Vingtor Stentofon System** from **Special Feature** dropdown box
- Click **Configure** to open the camera list



- Enter the camera URL and the directory number of the TCIV station

Door System URL : http://< TCIV camera IP address>:<port no./>mjpg/video.mjpg

Example: **http://10.5.101.46:8090/mjpg/video.mjpg**

Door System SIP User ID : Directory Number of TCIV station

5.4 AlphaCom Configuration for VS Desktop Video Display Station

- Vingtor-Stentofon IP Desktop Video Station - item no. 1408001635

5.4.1 Video Settings for Desktop Station

To configure video settings:

- Select **Advanced Alphacom > Video**

The screenshot shows the 'Advanced Alphacom' configuration page. The top navigation bar includes 'Station Main', 'Station Administration', 'Advanced Alphacom', and 'Advanced Network'. On the left, there is a sidebar menu with 'Audio', 'I/O Settings', 'Sound Detection', 'Time Settings', and 'Video' (selected). The main content area is titled 'Video Settings' and contains a table with the following parameters:

Description	Configuration
Enable Video:	<input checked="" type="checkbox"/>
Resolution:	240P
Frames per second	15 fps
Camera IP address and port:	10.5.101.46 8090
Enable HTTP basic authentication:	<input type="checkbox"/>
Video setup mode:	Default

Below the 'Video Settings' table is the 'Advanced Settings' section, which contains another table with the following parameters:

Description	Configuration
Lens distortion correction:	<input checked="" type="checkbox"/>
Night mode:	<input type="checkbox"/>
Zoom [1.00 2.50]x	1
Color saturation [0 255]:	128
Contrast [0 255]:	128
Brightness [0 255]:	128
Backlight compensation:	2

At the bottom of the page, there is a 'Save' button.

- Select or set values for the parameters:

Enable Video: Check box to enable video calls

Resolution: Select **240P**

Frames per second: Select **15fps**

Camera IP address and port: Enter the **port number** - default is **8090**

Video setup mode: Select **Default**

- ① **The video camera and the Turbine station have the same IP address.**
 - The video stream from the camera can be viewed by entering the IP address and port number in a web browser, e.g. **10.5.101.46:8090**
- ① **Video calls in the AlphaCom system are made in MJPG mode only.**
- ① The same **IP address** (e.g. **10.5.101.46**) and **port number** (e.g. **8090**) set here must be entered into the **Camera Settings** described in section "5.4.2 VS Desktop Station Video Configuration".
- Click **Save**
- Click **Back to config page**

5.4.2 VS Desktop Station Video Configuration

- Vingtör-Stentofon IP Desktop Video Station - item no. 1408001635

The camera of the Turbine Video station has to be set in the video display part of the Vingtör-Stentofon desktop station. This is done by logging into video part of the desktop station interface.

- Tapping anywhere on the LCD touchscreen will show the IP address of the video display part.



- Enter the **Video-IP** address as shown above (e.g. 192.168.45.45) in a web browser to log into the video part of the desktop station.

To log into the video part of the station:

1. Enter the default User name: **admin**
2. Enter the default password: **alphaadmin**



- Click **Cameras**

Camera Settings

Passcode

Passcode (Numbers only)

Camera Types

Baudisch	:80/mjpg/video.mjpg
AXIS	:80/axis-cgi/mjpg/video.cgi
TCIV	:8090/mjpg/video.mjpg
<input type="text"/>	<input type="text"/>
<input type="text"/>	<input type="text"/>

Camera Allocation

<< < 1 / 80 > >>

Name	FrontDoor
AlphaCom Node Number	3
AlphaCom Directory Number	2222
SIP ID	<input type="text"/>
Camera IP	10.5.101.46
Camera Type	TCIV
Camera User	<input type="text"/>
Camera Password	<input type="text"/>
Passcode required	<input type="checkbox"/>
Allocation active?	<input checked="" type="checkbox"/>

Submit settings

- Enter values for the parameters as shown

Camera Types

- Camera Type is **TCIV** with URL **:8090/mjpg/video.mjpg**
- ① '8090' is the default port number for the camera set in section "5.4.1 Video Settings for Desktop Station"

Camera Allocation

AlphaCom Node Number: Node number of network (e.g. 3)

AlphaCom Directory Number: Directory number of TCIV (e.g. 2222 as set in *Main Settings*)

Camera IP: IP address of TCIV (e.g. 10.5.101.46)

Camera Type: TCIV

- Click **Submit settings**

6 SIP Configuration

The STENTOFON SIP Stations are custom-made IP intercom stations that can integrate with any iPBX system.

In SIP mode, the Turbine Video stations have been tested for use with the following video display phones:

- IP Desktop Station with Video Display (Item Number: 1408001635)
- ITSV-1 Video Phone (Item Number: 1490001010)
- Cisco CP-9971 Video Phone
- Bria Softphone

① Configuration of the non-video part of the station such as SIP and DAK settings is described in the manual: *A100K11335 Turbine Compact IP Station Getting Started for SIP.*



Figure 6 SIP Video Intercom System

6.1 Logging into the Station

Access the station by logging into the web interface using a standard web browser:

1. Open a web browser
2. In the browser's address bar, type the station IP address and press the ENTER key
- The station login page will be displayed.

To log into the station:

1. Click **Login**
2. Enter the default User name: **admin**
3. Enter the default password: **alphaadmin**



The **Station Information** page will now be displayed, showing the station settings and status.

6.2 Station Main Settings

- Click **Station Main > Main Settings** to access the page for configuring station mode and IP parameters.

Station Main | Station Administration | Advanced SIP | Advanced Network

Station Information | Main Settings

Station Mode

Use Alphacom
 Use Exigo
 Use SIP
 Use Pulse
 Use Pulse Server

Product Model And Accessory

Model: Video Normal (TCIV-2, TCIV-3)

IP Settings

DHCP Static IP

IP-address:	10	-	5	-	101	-	46
Subnet-mask:	255	-	255	-	255	-	0
Gateway:	10	-	5	-	101	-	1
DNS Server 1:	10	-	5	-	2	-	19
DNS Server 2:	0	-	0	-	0	-	0

Hostname: zenitel063a41

Disable Reset to Factory default settings using frontboard and I/O:

Read IP Address:

Ethernet Speed 10 Mbit/s:

Save

Station Mode

- Select the **Use SIP** radio-button

Product Model And Accessory

- **Model**
Select one of the options from the drop-down box :
 - **Video Normal (TCIV-2, TCIV-3)**
 - **Video Scrolling Station (TCIV-6)**

IP Settings

- **Static IP** – Select this option if the IP station shall use a static IP address. Enter values for:
 - **IP-address: IP address of TCIV** (e.g. 10.5.101.46)
 - **Subnet-mask: Enter subnet mask**
 - **Gateway: Enter Gateway IP address**
 - **DNS Server 1** (option for network administration)
 - **DNS Server 2** (option for network administration)
 - **Hostname** (option for network administration)

Read IP Address

- Check the **Read IP Address** box to enable an unregistered station to speak the IP address when the call button is pressed.
- Click **Save** followed by **Apply** to apply the new configuration settings.

6.3 SIP Configuration for ITSV-1 Video Phone

- Vingtor-Stentofon ITSV-1 Video Phone - item no. 1490001010

6.3.1 SIP Settings for ITSV-1

- Select **SIP Configuration > SIP Settings**

Station Main		SIP Configuration		Station Administration		Advanced SIP		Advanced Network																																																																																							
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Account Settings

Directory Number (SIP ID): Directory number of Turbine Video station

Server Domain (SIP): IP address of the SIP Server

- ① The values for both these parameters are determined by the system administrator in the SIP server domain.
- Enter values for the other parameters under **Account Settings** and **Call Settings**
 - Click **Save**

6.3.2 Video Settings for ITSV-1

To configure video settings:

- Select **SIP Configuration > Video Settings**

The screenshot shows the configuration interface for the ITSV-1 device. The top navigation bar includes 'Station Main', 'SIP Configuration', 'Station Administration', 'Advanced SIP', and 'Advanced Network'. The left sidebar contains various settings categories, with 'Video Settings' selected. The main content area is divided into two sections: 'Video Settings' and 'Advanced Settings'. The 'Video Settings' section includes a table with the following parameters:

Description	Configuration
Video mode:	<input checked="" type="radio"/> H264 RTP <input type="radio"/> MJPG HTTP
Enable Video:	<input checked="" type="checkbox"/>
Resolution:	480P
Frames per second:	15 fps
Bitrate:	1000 kb/s
Video setup mode:	Default

The 'Advanced Settings' section includes a table with the following parameters:

Description	Configuration
Lens distortion correction:	<input checked="" type="checkbox"/>
Night mode:	<input type="checkbox"/>
Zoom [1.00 2.50]x:	1
Color saturation [0 255]:	128
Contrast [0 255]:	128
Brightness [0 255]:	128
Backlight compensation:	2

A 'Save' button is located at the bottom of the configuration area.

- Select or set values for the parameters:

Video Mode: Set to **H264 RTP**

Enable Video: Check box to enable video calls

Resolution: Select **480P**

Frames per second: Select **15fps**

Bitrate: Select **1000 kb/s**

Video setup mode: Select **Default**

- Click **Save**
- Click **Back to config page**

6.3.3 ITSV-1 Phone Settings

- Log into the ITSV-1 phone interface by entering its IP address in a browser on your PC

The screenshot shows the login interface for the ITSV-1 device. The title is 'ITSV-1 IP Touch Station with Video'. The interface includes the following fields:

- Username: admin
- Password: [Redacted]
- Language: English

A yellow 'Login' button is located at the bottom right of the form.

Login Credentials

Username: **admin**

Password: **alphaadmin**

6.3.4 ITSV-1 Account Setup

- Select **Account > Account 1 > General Settings**

The screenshot shows the 'General Settings' page for 'Account 1'. The 'Account Active' checkbox is checked. The 'Account Name' is 'SIP', 'SIP Server' is '10.5.11.55', 'SIP User ID' is '105', and 'SIP Authentication ID' is '105'. The 'SIP Authentication Password' is masked with dots. The 'Name' is 'ITSV-1'. The 'Show Account Name Only' checkbox is unchecked. The 'Tel URI' is set to 'Disable'. There are 'Save' and 'Cancel' buttons at the bottom.

- Enter the values shown above for the parameters

Account Active: Check **Yes** box

SIP Server: IP address of **SIP Server**

SIP User ID: Directory Number of the **ITSV-1** phone

6.3.5 ITSV-1 Video Configuration

The video is streamed directly from the TCIV camera to the ITSV-1. The TCIV camera must have a static IP address.

- Select **Account 1 > Call Settings**

The screenshot shows the 'Call Settings' page for 'Account 1'. The 'Start Video Automatically' checkbox is checked. The 'Video Layout' is set to 'Default' and the 'Remote Video Request' is set to 'Prompt'. There are 'Save' and 'Cancel' buttons at the bottom.

- Click the **Yes** box for **Start Video Automatically**

6.4 SIP Configuration for Cisco Video Phone

6.4.1 SIP Settings

- Select **SIP Configuration > SIP Settings**

Station Main	SIP Configuration	Station Administration	Advanced SIP	Advanced Network
SIP Settings				
Account Settings				
Description				
Configuration				
Display Name: Door 4 floor				
Directory Number (SIP ID): 2353				
Server Domain (SIP): 10.5.101.120				
Backup Domain (SIP):				
Backup Domain 2 (SIP):				
Registration Method: Parallell				
Authentication User Name: 2353				
Authentication Password:				
Register Interval: 600 (min. 60 seconds)				
Outbound Proxy [optional]: Port: 5060				
Outbound Backup Proxy [optional]: Port: 5060				
Outbound Backup Proxy 2 [optional]: Port: 5060				
Outbound Transport: UDP				
SIP Scheme: sip Using sips forces all proxies to also use TLS				
RTP Encryption: disabled				
SRTP Crypto Type: AES_CM_128_HMAC_SHA1_80				
Use Unencrypted SRTP: <input type="checkbox"/>				
TLS Private Key: turbine_server_sha256.key				
Call Settings				
Description				
Configuration				
Enable Auto Answer: <input checked="" type="checkbox"/>				
Auto Answer Delay: 0 seconds. Max 30 seconds.				
Press and Hold Time: 0 seconds. Max 60 seconds. Defines how long a DAK key/Input must be pressed before the call is established.				
Max Ringing Time: 120 How long a call can be ringing before hanging up.				
Max Conversation Time: 3600 How long a call can be in conversation before hanging up.				
Max Queued Time: 20 How long a call can be queued before hanging up.				
Max Queued Calls: 5 How many incoming calls can be queued. Max 5.				
Dialing Method: Enbloc Dialing				
Enbloc Dialing Timeout: No Timeout				
DTMF method: SIP INFO				
Conversation Mode: Full Open Duplex				
PTT Mode: Mic and speaker is controlled by PTT button				
Remote Controlled Audio Direction: <input type="checkbox"/> (Received DTMF * to listen, DTMF # to talk, DTMF 0 for open duplex)				
SIP Message Controlled Audio Direction: <input type="checkbox"/> (SIP MESSAGE controls audio direction)				
Boost Volume on Push To Talk: <input checked="" type="checkbox"/>				
Override Remote Push To Talk: <input type="checkbox"/>				
Force Open Duplex Using DTMF: -				
Send DTMF */# with M key: <input checked="" type="checkbox"/>				
RTP Timeout value: 0 seconds. 0 = RTP Timeout Disabled.				
Codec g729: Medium Priority				
Codec g722: High Priority				
Codec g711a: Medium Priority				
Codec g711u: Low Priority				

Account Settings

Directory Number (SIP ID): Directory number of Turbine Video station

Server Domain (SIP): IP address of the Cisco Unified Communications Manager (CallManager)

- ① The values for both these parameters are determined by the settings in Cisco Unified Communications Manager (CallManager).

- Enter values for the other parameters under **Account Settings** and **Call Settings**
- Click **Save**

6.4.2 Video Settings for Cisco Phone

H.264 bitrate and resolution combinations for Cisco video phone

	128 kb/s	300 kb/s	500 kb/s	1 Mb/s	2.5 Mb/s
320x240 (240P)	Yes	Yes	Yes	Yes	No
640x480 (480P)	Not recommended	Yes	Yes	Yes	No
1280x720 (720P)	No	No	No	No	No

① 'Not recommended' means that this combination should not be used when bandwidth is limited.

- Select **SIP Configuration > Video Settings**

The screenshot shows the 'SIP Configuration' page with the 'Video Settings' tab selected. The 'Video Settings' section is highlighted with a red box and contains the following parameters:

Description	Configuration
Video mode:	<input checked="" type="radio"/> H264 RTP <input type="radio"/> MJPG HTTP
Enable Video:	<input checked="" type="checkbox"/>
Resolution:	240P
Frames per second	15 fps
Bitrate:	1000 kb/s
Video setup mode:	Cisco

The 'Advanced Settings' section contains the following parameters:

Description	Configuration
Lens distortion correction:	<input checked="" type="checkbox"/>
Night mode:	<input type="checkbox"/>
Zoom [1.00 2.50]x	1
Color saturation [0 255]:	128
Contrast [0 255]:	128
Brightness [0 255]:	128
Backlight compensation:	2

A 'Save' button is located at the bottom of the settings area.

- Select or set values for the parameters according to the example above:

Video Mode: Set to **H264 RTP**

Enable Video: Check box to enable video calls

Resolution: Select **240P**

Frames per second: Select **15fps**

Bitrate: Select **1000 kb/s**

Video setup mode: Select **Cisco**

- Click **Save**
- Click **Back to config page**

6.4.3 Direct Access Key Settings

- Select **SIP Configuration > Direct Access Key Settings**

	Function	Idle	Call	No Ringlist
DAK 1		Call To	2323	No Ringlist
Input 1		Call To		No Ringlist
Input 2		Call To		No Ringlist
Input 3		Call To		No Ringlist
Input 4		Call To		No Ringlist
Input 5		Call To		No Ringlist
Input 6		Call To		No Ringlist

To set up the call key on the Turbine station to call the Cisco phone directly:

- Enter the directory number of the Cisco phone in the **Value** field for **DAK 1**
- ① **This parameter is valid for TCIV-2 and TCIV-3 only**
- ① **See *A100K11194 Turbine IP Stations Technical Manual* for the configuration and import of an Address Book for TCIV-6.**

6.5 SIP Configuration for Bria Softphone

- ① Exceptions must be made for Bria in Windows Firewall to be able to receive video. During installation Bria adds rules to Windows Firewall by default, but in some cases this is not sufficient and exceptions must be added manually. If in doubt, consult your system administrator on how to add exceptions for Bria in Windows Firewall.

- Select **SIP Configuration > SIP Settings**

Station Main	SIP Configuration	Station Administration	Advanced SIP	Advanced Network																																																
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Account Settings

Directory Number (SIP ID): Directory number of Turbine Video station

Server Domain (SIP): IP address of the SIP server

- ① The values for both these parameters are determined by the system administrator in the SIP server domain.
- Enter values for the other parameters under **Account Settings** and **Call Settings**
 - Click **Save**

6.5.1 Video Settings for Bria Softphone

H.264 bitrate and resolution combinations for Bria softphone

	128 kb/s	300 kb/s	500 kb/s	1 Mb/s	2.5 Mb/s
320x240 (240P)	Yes	Yes	Yes	Yes	Yes
640x480 (480P)	Not recommended	Yes	Yes	Yes	Yes
1280x720 (720P)	Not recommended	Not recommended	Yes	Yes	Yes

① 'Not recommended' means that this combination should not be used when bandwidth is limited.

- Select **SIP Configuration > Video Settings**

The screenshot shows the 'SIP Configuration' page with the 'Video Settings' tab selected. The 'Video Settings' section is highlighted with a red box and contains the following configuration:

Description	Configuration
Video mode:	<input checked="" type="radio"/> H264 RTP <input type="radio"/> MJPG HTTP
Enable Video:	<input checked="" type="checkbox"/>
Resolution:	480P
Frames per second:	15 fps
Bitrate:	1000 kb/s
Video setup mode:	Default

The 'Advanced Settings' section includes the following configuration:

Description	Configuration
Lens distortion correction:	<input checked="" type="checkbox"/>
Night mode:	<input type="checkbox"/>
Zoom [1.00 2.50]x:	1
Color saturation [0 255]:	128
Contrast [0 255]:	128
Brightness [0 255]:	128
Backlight compensation:	2

A 'Save' button is located at the bottom of the page.

- Enter the values shown above for the parameters

Video Mode: Set to **H264 RTP**

Enable Video: Check box to enable video calls

Resolution: Select **480P**

Frames per second: Select **15fps**

Bitrate: Select **1000 kb/s**

Video setup mode: Select **Default**

- Click **Save**
- Click **Back to config page**

6.5.2 Direct Access Key Settings

- Select **SIP Configuration > Direct Access Key Settings**

Function	Idle	Call	No Ringlist
DAK 1	Call To	9910	No Ringlist
Input 1	Call To		No Ringlist
Input 2	Call To		No Ringlist
Input 3	Call To		No Ringlist
Input 4	Call To		No Ringlist
Input 5	Call To		No Ringlist
Input 6	Call To		No Ringlist

To set up the call key on the Turbine station to call the Bria softphone directly:

- Enter the directory number of the Bria softphone in the **Value** field for **Direct Access Key 1**
- In this example, the directory number of the Bria softphone is 9910

6.5.3 Bria Softphone Settings

- Start the Bria softphone application on your PC

Video Codecs

- Select **Softphone > Preferences > Video Codecs**

Preferences

Video Codecs

Available Codecs: H.263, H.263+ (1998), VP8

Enabled Codecs: H.264

Select a codec from the above lists to view properties

Description: H.264

CPU usage: [Green progress bar]

Quality: [Green progress bar] Low High

Reset to Default OK Cancel

- Enable codec **H.264**

Account Settings

- Select **Softphone > Account Settings**

SIP Account

Account Voicemail Topology Presence Storage Transport Advanced

Account name: Hon Bria

Protocol: SIP

Allow this account for

Call

IM / Presence

User Details

* User ID: 9910

* Domain: 10.5.11.55

Password:

Display name: Hon Bria

Authorization name:

User Details

User ID: Directory number of the Bria softphone

Domain: IP address of SIP Server Domain

6.6 SIP Configuration for VS Desktop Video Display Station

- Vingtör-Stentofon IP Desktop Video Station - item no. 1408001635

6.6.1 SIP Settings

- Select **SIP Configuration** > **SIP Settings**

Station Main	SIP Configuration	Station Administration	Advanced SIP	Advanced Network
SIP Settings				
Account Settings				
Description		Configuration		
Display Name:		Door 4 floor		
Directory Number (SIP ID):		9900		
Server Domain (SIP):		10.5.11.55		
Backup Domain (SIP):				
Backup Domain 2 (SIP):				
Registration Method:		Parallel		
Authentication User Name:		2353		
Authentication Password:				
Register Interval:		600 (min. 60 seconds)		
Outbound Proxy [optional]:		Port: 5060		
Outbound Backup Proxy [optional]:		Port: 5060		
Outbound Backup Proxy 2 [optional]:		Port: 5060		
Outbound Transport:		UDP		
SIP Scheme:		sip Using sips forces all proxies to also use TLS		
RTP Encryption:		disabled		
SRTP Crypto Type:		AES_CM_128_HMAC_SHA1_80		
Use Unencrypted SRTP:		<input type="checkbox"/>		
TLS Private Key:		turbine_server_sha256.key		
Call Settings				
Description		Configuration		
Enable Auto Answer:		<input checked="" type="checkbox"/>		
Auto Answer Delay:		0 seconds. Max 30 seconds.		
Press and Hold Time:		0 seconds. Max 60 seconds. Defines how long a DAK key/input must be pressed before the call is established.		
Max Ringing Time:		120 How long a call can be ringing before hanging up.		
Max Conversation Time:		3600 How long a call can be in conversation before hanging up.		
Max Queued Time:		20 How long a call can be queued before hanging up.		
Max Queued Calls:		5 How many incoming calls can be queued. Max 5.		
Dialing Method:		Enbloc Dialing		
Enbloc Dialing Timeout:		No Timeout		
DTMF method:		SIP INFO		
Conversation Mode:		Full Open Duplex		
PTT Mode:		Mic and speaker is controlled by PTT button		
Remote Controlled Audio Direction:		<input type="checkbox"/> (Received DTMF * to listen, DTMF # to talk, DTMF 0 for open duplex)		
SIP Message Controlled Audio Direction:		<input type="checkbox"/> (SIP MESSAGE controls audio direction)		
Boost Volume on Push To Talk:		<input checked="" type="checkbox"/>		
Override Remote Push To Talk:		<input type="checkbox"/>		
Force Open Duplex Using DTMF:		-		
Send DTMF */# with M key:		<input checked="" type="checkbox"/>		
RTP Timeout value:		0 seconds. 0 = RTP Timeout Disabled.		
Codec g729:		Medium Priority		
Codec g722:		High Priority		
Codec g711a:		Medium Priority		
Codec g711u:		Low Priority		

Account Settings

Directory Number (SIP ID): Directory number of Turbine Video station

Server Domain (SIP): IP address of the SIP Server

- ① The values for both these parameters are determined by the system administrator in the SIP server domain.
- Enter values for the other parameters under **Account Settings** and **Call Settings**
 - Click **Save**

6.6.2 Video Settings for Desktop Video Display Station

- Select **SIP Configuration > Video Settings**

Description	Configuration
Video mode:	<input type="radio"/> H264 RTP <input checked="" type="radio"/> MJPG HTTP
Enable Video:	<input checked="" type="checkbox"/>
Resolution:	240P
Frames per second	15 fps
Camera IP address and port:	10.5.101.46 8090
Enable HTTP basic authentication:	<input type="checkbox"/>
Video setup mode:	Default

Description	Configuration
Lens distortion correction:	<input checked="" type="checkbox"/>
Night mode:	<input type="checkbox"/>
Zoom [1.00 2.50]x	1
Color saturation [0 255]:	128
Contrast [0 255]:	128
Brightness [0 255]:	128
Backlight compensation:	2

Save

- Enter the values shown above for the parameters

Video Mode: Set to **MJPG HTTP**

Enable Video: Check box to enable video calls

Resolution: Select **240P**

Frames per second: Select **15fps**

Camera IP address and port: Enter the **port number** - default is **8090**

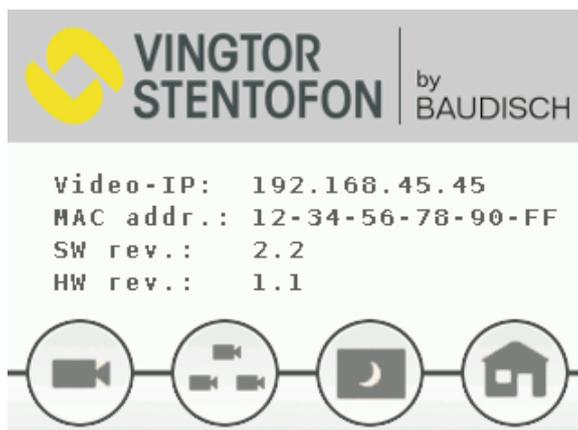
Video setup mode: Select **Default**

- ① **The video camera and the Turbine station have the same IP address.**
 - The video stream from the camera can be viewed by entering the IP address and port number in a web browser, e.g. **10.5.102.61:8090**
- ① The same **IP address** (e.g. **10.5.101.46**) and **port number** (e.g. **8090**) set here must be entered into the settings for the Desktop Station described in section “6.6.3 Desktop Video Display Station Settings”.
- Click **Save**
- Click **Back to config page**

6.6.3 Desktop Video Display Station Settings

The camera of the Turbine Video station has to be set in the video touchscreen of the station. This is done by logging into the video touchscreen of the desktop station interface.

- Tapping anywhere on the LCD touchscreen will show the IP address of the video display part.



- Enter the **Video-IP** address as shown above (e.g. 192.168.45.45) in a web browser to log into the video part of the desktop station.

To log into the video part of the station:

1. Enter the default Username: **admin**
2. Enter the default password: **alphaadmin**



- Click **Cameras**

[<- back](#)

Camera Settings

Passcode

Passcode (Numbers only)

Camera Types

Baudisch	:80/mjpg/video.mjpg
AXIS	:80/axis-cgi/mjpg/video.cgi
TCIV	:8090/mjpg/video.mjpg
<input type="text"/>	<input type="text"/>
<input type="text"/>	<input type="text"/>

Camera Allocation

/ 80

Name	FrontDoor
AlphaCom Node Number	<input type="text" value="0"/>
AlphaCom Directory Number	<input type="text" value="0"/>
SIP ID	<input type="text" value="9900"/>
Camera IP	<input type="text" value="10.5.101.46"/>
Camera Type	TCIV
Camera User	<input type="text"/>
Camera Password	<input type="text"/>
Passcode required	<input type="checkbox"/>
Allocation active?	<input checked="" type="checkbox"/>

- Enter values for Camera Types and Camera Allocation as shown

Camera Types

- Define Camera Type for Turbine Video station TCIV by entering URL :**8090/mjpg/video.mjpg**
- ① **'8090' is the default port number** for the camera set in section "6.6.2 Video Settings for Desktop Video Display Station".

Camera Allocation

AlphaCom Node Number : 0 (Not in use for SIP system)

AlphaCom Directory Number : 0 (Not in use for SIP system)

SIP ID : Directory number of TCIV as specified in section 6.6.1 (e.g. 9900)

Camera IP : IP address of TCIV (e.g. 10.5.101.46)

Camera Type : TCIV

Allocation active? : Check the box to enable video streaming from the camera

- Click **Submit settings**

7 Pulse Configuration

STENTOFON Pulse is an IP-based intercom system for up to 16 intercom stations. The system works with all STENTOFON IP intercom stations. In Pulse mode, the Turbine Video stations have been tested for use with the following video display phones:

- IP Desktop Station with Video Display (Item Number: 1408001635)
- ITSV-1 Video Phone (Item Number: 1490001010)
- Snom 760 / Snom 821 Video Phone
- Bria Softphone

① Configuration of the non-video part of the station such as directory, call and audio settings is described in the manual: A100K11336 Turbine Compact IP Station Getting Started for Pulse.



Figure 7 Pulse Video Intercom System

① It is recommended to **NOT** use the Turbine Video station as the Pulse Server. In the Pulse system example above, the Desktop Video Display Station is used as the Pulse Server. In order for SIP stations to be registered in the system, the Pulse Server must first install SIP station licenses.

7.1 Logging into the Station

The Turbine Video Station features an embedded web interface, which allows users to log in via a standard web browser.

Access the station by logging into the web interface using a standard web browser:

1. Open a web browser
2. In the browser's address bar, type the station IP address and press the ENTER key
- The station login page will be displayed.

To log into the station:

1. Click **Login**
2. Enter the default User name: **admin**
3. Enter the default password: **alphaadmin**



The **Station Information** page will now be displayed, showing the station settings and status.

7.2 Station Main Settings

- Click **Station Main > Main Settings** to access the page for configuring station mode and IP parameters.

Station Main | SIP Configuration | Station Administration | Advanced Network

Station Information

Main Settings

Station Mode

Use Alphacom

Use Exigo

Use SIP

Use Pulse

Use Pulse Server

Product Model And Accessory

Model: Video Normal (TCIV-2, TCIV-3)

IP Settings

DHCP Static IP

IP-address:	10	-	5	-	101	-	46
Subnet-mask:	255	-	255	-	255	-	0
Gateway:	10	-	5	-	101	-	1
DNS Server 1:	10	-	5	-	2	-	19
DNS Server 2:	0	-	0	-	0	-	0
Hostname:	zenitel063a41						

Disable Reset to Factory default settings using frontboard and I/O:

Read IP Address:

Ethernet Speed 10 Mbit/s:

Save

Station Mode

- Select the **Use Pulse** radio-button

① *For optimal system operation, it is recommended to NOT use TCIV-x as the 'Pulse Server'.*

Product Model And Accessory

Model: Select one of the options from the drop-down box :

- Video Normal (TCIV-2, TCIV-3)
- Video Scrolling Station (TCIV-6)

IP Settings

- **Static IP** – Select this option if the IP station shall use a static IP address. Enter values for:
 - **IP-address:** IP address of TCIV (e.g. 10.5.101.46)
 - **Subnet-mask:** Enter subnet mask
 - **Gateway:** Enter Gateway IP address
 - **DNS Server 1** (option for network administration)
 - **DNS Server 2** (option for network administration)
 - **Hostname** (option for network administration)

Read IP Address

- Check the **Read IP Address** box to enable an unregistered station to speak the IP address when the call button is pressed.
- Click **Save** followed by **Apply** to apply the new configuration settings.

7.3 Pulse Configuration for ITSV-1 Video Phone

- Vingt-Stentofon ITSV-1 Video Phone - item no. 1490001010

7.3.1 SIP Settings

- Select **SIP Configuration** > **SIP Settings**

Station Main	SIP Configuration	Station Administration	Advanced Network																																																						
<div style="display: flex;"> <div style="width: 20%;"> <ul style="list-style-type: none"> ▼ SIP Settings ▶ Audio Settings ▶ Direct Access Key Settings ▶ Relay Settings ▶ Time Settings ▶ I/O Settings ▶ Video Settings ▶ Script Configuration ▶ Script Events ▶ Script Upload ▶ Audio Messages ▶ Certificates </div> <div style="width: 80%;"> <h4>Account Settings</h4> <table border="1"> <thead> <tr> <th>Description</th> <th>Configuration</th> </tr> </thead> <tbody> <tr> <td>Display Name:</td> <td>video-turbine2</td> </tr> <tr> <td>Directory Number (SIP ID):</td> <td>888</td> </tr> <tr> <td>Server Domain (SIP):</td> <td>10.5.2.114</td> </tr> <tr> <td>Backup Domain (SIP):</td> <td></td> </tr> <tr> <td>Backup Domain 2 (SIP):</td> <td></td> </tr> <tr> <td>Authentication User Name:</td> <td>888</td> </tr> <tr> <td>Authentication Password:</td> <td></td> </tr> <tr> <td>Register Interval:</td> <td>600 (min. 60 seconds)</td> </tr> </tbody> </table> <h4>Call Settings</h4> <table border="1"> <thead> <tr> <th>Description</th> <th>Configuration</th> </tr> </thead> <tbody> <tr> <td>Enable Auto Answer:</td> <td><input checked="" type="checkbox"/></td> </tr> <tr> <td>Auto Answer Delay:</td> <td>0 seconds. Max 30 seconds.</td> </tr> <tr> <td>Press and Hold Time:</td> <td>0 seconds. Max 60 seconds. Defines how long a DAK key/Input must be pressed before the call is established.</td> </tr> <tr> <td>Max Ringing Time:</td> <td>120 How long a call can be ringing before hanging up.</td> </tr> <tr> <td>Max Conversation Time:</td> <td>3600 How long a call can be in conversation before hanging up.</td> </tr> <tr> <td>Max Queued Time:</td> <td>20 How long a call can be queued before hanging up.</td> </tr> <tr> <td>Max Queued Calls:</td> <td>5 How many incoming calls can be queued. Max 5.</td> </tr> <tr> <td>Conversation Mode:</td> <td>Full Open Duplex</td> </tr> <tr> <td>PTT Mode:</td> <td>Mic and speaker is controlled by PTT button</td> </tr> <tr> <td>Boost Volume on Push To Talk:</td> <td><input checked="" type="checkbox"/></td> </tr> <tr> <td>Override Remote Push To Talk:</td> <td><input type="checkbox"/></td> </tr> <tr> <td>Force Open Duplex Using DTMF:</td> <td>-</td> </tr> <tr> <td>Send DTMF *# with M key:</td> <td><input checked="" type="checkbox"/></td> </tr> <tr> <td>Codec g729:</td> <td>Medium Priority</td> </tr> <tr> <td>Codec g722:</td> <td>High Priority</td> </tr> <tr> <td>Codec g711a:</td> <td>Medium Priority</td> </tr> <tr> <td>Codec g711u:</td> <td>Low Priority</td> </tr> </tbody> </table> </div> </div>				Description	Configuration	Display Name:	video-turbine2	Directory Number (SIP ID):	888	Server Domain (SIP):	10.5.2.114	Backup Domain (SIP):		Backup Domain 2 (SIP):		Authentication User Name:	888	Authentication Password:		Register Interval:	600 (min. 60 seconds)	Description	Configuration	Enable Auto Answer:	<input checked="" type="checkbox"/>	Auto Answer Delay:	0 seconds. Max 30 seconds.	Press and Hold Time:	0 seconds. Max 60 seconds. Defines how long a DAK key/Input must be pressed before the call is established.	Max Ringing Time:	120 How long a call can be ringing before hanging up.	Max Conversation Time:	3600 How long a call can be in conversation before hanging up.	Max Queued Time:	20 How long a call can be queued before hanging up.	Max Queued Calls:	5 How many incoming calls can be queued. Max 5.	Conversation Mode:	Full Open Duplex	PTT Mode:	Mic and speaker is controlled by PTT button	Boost Volume on Push To Talk:	<input checked="" type="checkbox"/>	Override Remote Push To Talk:	<input type="checkbox"/>	Force Open Duplex Using DTMF:	-	Send DTMF *# with M key:	<input checked="" type="checkbox"/>	Codec g729:	Medium Priority	Codec g722:	High Priority	Codec g711a:	Medium Priority	Codec g711u:	Low Priority
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Account Settings

Directory Number (SIP ID): directory number of Turbine Video station

Server Domain (SIP): IP address of intercom station set up as Pulse Server

- Click **Save**
- Click **Back to config page**

7.3.2 Third Party SIP Terminals

- ① To configure third-party SIP terminals, you need to log into the station that has been set up as the **Pulse Server**.

- Select **Server Management** > **Server Configuration** > **Directory Settings**

Station Main	SIP Configuration	Station Administration	Server Management																						
<div style="display: flex;"> <div style="width: 20%;"> <ul style="list-style-type: none"> ▶ Server Monitoring ▼ Server Configuration <ul style="list-style-type: none"> • Directory Settings • Call and Audio Settings • Direct Access Key Settings • System Settings ▶ Ringlist ▶ Station Profiles ▶ Group Call ▶ Software Upgrade </div> <div style="width: 80%;"> <h4>Directory Settings</h4> <h5>STENTOFON Stations</h5> <table border="1"> <thead> <tr> <th>Directory Number</th> <th>Name</th> <th>Password</th> <th>DHCP / Static IP</th> <th>Station Profile</th> </tr> </thead> <tbody> <tr> <td>888</td> <td>video-turbine2</td> <td></td> <td><input checked="" type="checkbox"/> 169.254.1.100</td> <td>Default</td> </tr> </tbody> </table> <p style="text-align: center;">Refresh Save Apply</p> <p><small>Note! Subnet-mask and gateway for all STENTOFON Stations are set to be the same as this station's configuration.</small></p> <h5>Third Party SIP Terminals</h5> <table border="1"> <thead> <tr> <th>Directory Number</th> <th>Name</th> <th>Profile</th> <th>Password</th> </tr> </thead> <tbody> <tr> <td></td> <td></td> <td>Default</td> <td></td> </tr> <tr> <td>999</td> <td>ITSV-1</td> <td>Default</td> <td></td> </tr> </tbody> </table> <p style="text-align: center;">Add Delete Save</p> </div> </div>				Directory Number	Name	Password	DHCP / Static IP	Station Profile	888	video-turbine2		<input checked="" type="checkbox"/> 169.254.1.100	Default	Directory Number	Name	Profile	Password			Default		999	ITSV-1	Default	
Directory Number	Name	Password	DHCP / Static IP	Station Profile																					
888	video-turbine2		<input checked="" type="checkbox"/> 169.254.1.100	Default																					
Directory Number	Name	Profile	Password																						
		Default																							
999	ITSV-1	Default																							

Under Third Party SIP Terminals:

- Enter the **Directory Number** and **Name** of the ITSV-1 Video Phone
- Click **Add** and **Save**

Under **STENTOFON Stations**:

- Click **Apply**
 - This will reboot all the stations in the Pulse system

7.3.3 Video Settings for ITSV-1

- Select **SIP Configuration > Video Settings**

The screenshot shows the 'Video Settings' configuration page. The left sidebar contains a navigation menu with 'Video Settings' selected. The main content area is divided into two sections: 'Video Settings' and 'Advanced Settings'. The 'Video Settings' section has a table with the following configuration:

Description	Configuration
Video mode:	<input checked="" type="radio"/> H264 RTP <input type="radio"/> MJPG HTTP
Enable Video:	<input checked="" type="checkbox"/>
Resolution:	480P
Frames per second:	15 fps
Bitrate:	1000 kb/s
Video setup mode:	Default

The 'Advanced Settings' section has a table with the following configuration:

Description	Configuration
Lens distortion correction:	<input checked="" type="checkbox"/>
Night mode:	<input type="checkbox"/>
Zoom [1.00 2.50]x:	1
Color saturation [0 255]:	128
Contrast [0 255]:	128
Brightness [0 255]:	128
Backlight compensation:	2

A 'Save' button is located at the bottom center of the page.

- Enter the values shown above for the parameters

Video Mode: Set to **H264 RTP**

Enable Video: Check box to enable video calls

Resolution: Select **480P**

Frames per second: Select **15fps**

Bitrate: Select **1000 kb/s**

Video setup mode: Select **Default**

- Click **Save**
- Click **Back to config page**

7.3.4 Direct Access Key Settings

- Select **SIP Configuration > Direct Access Key Settings**

The screenshot shows the 'Direct Access Key Settings' configuration page. The left sidebar contains a navigation menu with 'Direct Access Key Settings' selected. The main content area is a table with the following configuration:

	Function	Idle	Call	No Ringlist
DAK 1		Call To	999	No Ringlist
Input 1		Call To		No Ringlist
Input 2		Call To		No Ringlist
Input 3		Call To		No Ringlist
Input 4		Call To		No Ringlist

ⓘ This feature applies to TCIV-2 and TCIV-3 only

- ① See *A100K11194 Turbine IP Stations Technical Manual* for the configuration and import of an Address Book for TCIV-6.

To set up the call key on the Turbine station to call the ITSV-1 Video Phone directly:

- Enter the directory number of the ITSV-1 Video Phone in the **Value** field for **Direct Access Key 1**
 - In this example, the directory number of the ITSV-1 Video Phone is **999**
- Click **Save**

7.3.5 ITSV-1 Phone Settings

- Log into the ITSV-1 phone interface by entering its IP address in a browser on your PC

Login Credentials

Username: **admin**

Password: **alphaadmin**

7.3.5.1 ITSV-1 Account Setup

- Select **Account > Account 1 > General Settings**

- Enter the values shown above for the parameters

Account Active: Check **Yes** box

SIP Server: IP address of intercom station set as **Pulse Server**

SIP User ID: Directory Number of the **ITSV-1** phone

SIP Authentication ID: Same as SIP User ID

7.3.5.2 ITSV-1 Video Configuration

The video is streamed directly from the TCIV camera to the ITSV-1. The TCIV camera must have a static IP address.

- Select **Account 1 > Call Settings**

The screenshot shows a configuration page with a sidebar on the left containing 'General Settings', 'Network Settings', 'SIP Settings', 'Codec Settings', and 'Call Settings'. The top navigation bar includes 'Status', 'Account', 'Advanced Settings', and 'Maintenance'. Under 'Account', 'Account 1' is selected. The 'Call Settings' section for 'Account 1' includes a checkbox for 'Start Video Automatically' which is checked and highlighted with a red box. Below it are dropdown menus for 'Video Layout' (set to 'Default') and 'Remote Video Request' (set to 'Prompt').

- Click the **Yes** box for **Start Video Automatically**

7.3.6 Verifying Registration of ITSV-1

When the configuration for both the Turbine Video station and the ITSV-1 phone has been done, verify that they're both registered in the Pulse system.

- ① **To verify station registration, you need to log into the station that has been set up as the Pulse Server.**

- Select **Server Management > Server Monitoring**

The screenshot shows the 'Server Management' section with a 'Station Directory' table. The table has columns for 'Directory Number', 'Name', 'Status', 'IP Address', 'Station Profile', and 'Terminal Type'. Two entries are listed: '888' for 'video-turbine2' (Registered, IP 192.16.1.20, Default Station Profile, STENTOFON Station Terminal Type) and '999' for 'ITSV-1' (Registered, IP 192.16.1.21, Default Station Profile, 3rd Party SIP Terminal Type).

Directory Number	Name	Status	IP Address	Station Profile	Terminal Type
888	video-turbine2	Registered	192.16.1.20	Default	STENTOFON Station
999	ITSV-1	Registered	192.16.1.21	Default	3rd Party SIP Terminal

Now you should be able to:

- Call the ITSV-1 phone directly by pressing the call key on the Turbine Video station
- Call the Turbine Video station by dialing its number (e.g. 888) on the ITSV-1 phone

7.4 Pulse Configuration for Bria Softphone

7.4.1 SIP Settings

- Select **SIP Configuration > SIP Settings**

Station Main	SIP Configuration	Station Administration	Advanced Network																																				
SIP Settings																																							
Account Settings	<table border="1"><thead><tr><th>Description</th><th>Configuration</th></tr></thead><tbody><tr><td>Display Name:</td><td>video-turbine2</td></tr><tr><td>Directory Number (SIP ID):</td><td>800</td></tr><tr><td>Server Domain (SIP):</td><td>10.5.2.114</td></tr><tr><td>Backup Domain (SIP):</td><td></td></tr><tr><td>Backup Domain 2 (SIP):</td><td></td></tr><tr><td>Authentication User Name:</td><td>800</td></tr><tr><td>Authentication Password:</td><td></td></tr><tr><td>Register Interval:</td><td>600 (min. 60 seconds)</td></tr></tbody></table>			Description	Configuration	Display Name:	video-turbine2	Directory Number (SIP ID):	800	Server Domain (SIP):	10.5.2.114	Backup Domain (SIP):		Backup Domain 2 (SIP):		Authentication User Name:	800	Authentication Password:		Register Interval:	600 (min. 60 seconds)																		
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Authentication Password:																																							
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Send DTMF *# with M key:	<input checked="" type="checkbox"/>																																						
Codec g729:	Medium Priority																																						
Codec g722:	High Priority																																						
Codec g711a:	Medium Priority																																						
Codec g711u:	Low Priority																																						

Account Settings

Directory Number (SIP ID): directory number of Turbine Video station

Server Domain (SIP): IP address of station set up as Pulse Server

- Click **Save**
- Click **Back to config page**

7.4.2 Video Settings for Bria Softphone

- Select **SIP Configuration > Video Settings**

Station Main	SIP Configuration	Station Administration	Advanced Network																
Video Settings																			
Video mode:	<input checked="" type="radio"/> H264 RTP <input type="radio"/> MJPG HTTP																		
Enable Video:	<input checked="" type="checkbox"/>																		
Resolution:	480P																		
Frames per second	15 fps																		
Bitrate:	1000 kb/s																		
Video setup mode:	Default																		
Advanced Settings																			
<table border="1"><thead><tr><th>Description</th><th>Configuration</th></tr></thead><tbody><tr><td>Lens distortion correction:</td><td><input checked="" type="checkbox"/></td></tr><tr><td>Night mode:</td><td><input type="checkbox"/></td></tr><tr><td>Zoom [1.00 2.50]x</td><td>1</td></tr><tr><td>Color saturation [0 255]:</td><td>128</td></tr><tr><td>Contrast [0 255]:</td><td>128</td></tr><tr><td>Brightness [0 255]:</td><td>128</td></tr><tr><td>Backlight compensation:</td><td>2</td></tr></tbody></table>				Description	Configuration	Lens distortion correction:	<input checked="" type="checkbox"/>	Night mode:	<input type="checkbox"/>	Zoom [1.00 2.50]x	1	Color saturation [0 255]:	128	Contrast [0 255]:	128	Brightness [0 255]:	128	Backlight compensation:	2
Description	Configuration																		
Lens distortion correction:	<input checked="" type="checkbox"/>																		
Night mode:	<input type="checkbox"/>																		
Zoom [1.00 2.50]x	1																		
Color saturation [0 255]:	128																		
Contrast [0 255]:	128																		
Brightness [0 255]:	128																		
Backlight compensation:	2																		

- Enter the values shown above for the parameters

Video Mode: Set to **H264 RTP**

Enable Video: Check box to enable video calls

Resolution: Select **480P**

Frames per second: Select **15fps**

Bitrate: Select **1000 kb/s**

Video setup mode: Select **Default**

Video Mode: Set to **H.264 RTP**

- Click **Save**
- Click **Back to config page**

7.4.3 Direct Access Key Settings

- Select **SIP Configuration > Direct Access Key Settings**

Function	Idle	Call	No Ringlist
DAK 1	Call To	801	No Ringlist
Input 1	Call To		No Ringlist
Input 2	Call To		No Ringlist
Input 3	Call To		No Ringlist
Input 4	Call To		No Ringlist

① This feature applies to TCIV-2 and TCIV-3 only

① See Turbine Configuration Manual for the configuration and import of an Address Book for TCIV-6.

To set up the call key on the Turbine station to call the Bria softphone directly:

- Enter the directory number of the Bria softphone in the **Value** field for **Direct Access Key 1**
- In this example, the directory number of the Bria softphone is **801**
- Click **Save**

7.4.4 Third-Party SIP Terminals

① To configure third-party SIP terminals, you need to log into the station that has been set up as the Pulse Server.

- Select **Server Management > Server Configuration > Directory Settings**

Under **Third Party SIP Terminals**:

Directory Number	Name	Profile	Password
		Default	
801	Hon Bria	Default	

- Enter the **Directory Number** and **Name** of the Bria softphone
- Click **Add** and **Save**

Under **STENTOFON Stations**:

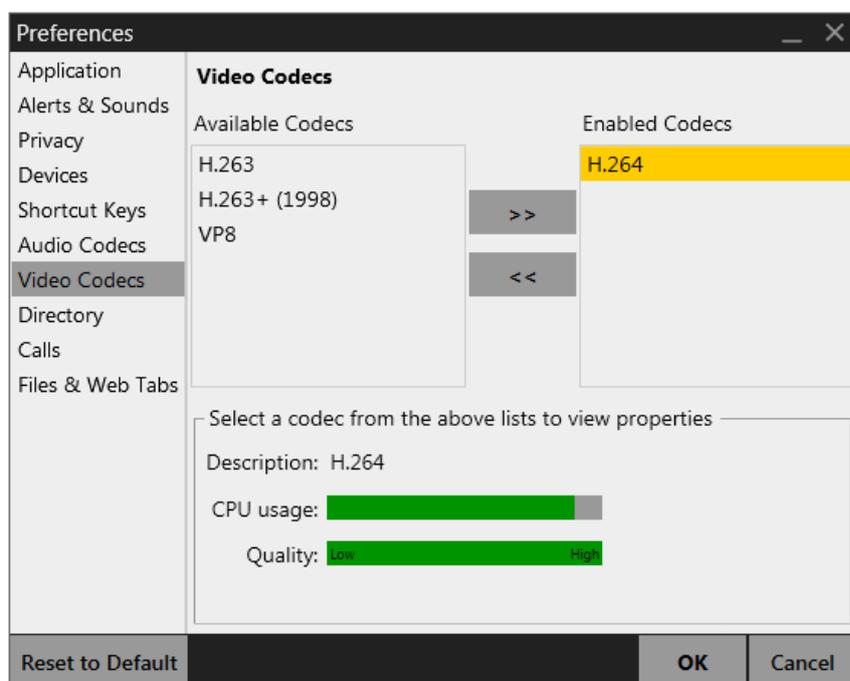
- Click **Apply**
 - This will reboot all the stations in the Pulse system

7.4.5 Bria Softphone Configuration

- Start the Bria softphone application on your PC

Video Codecs

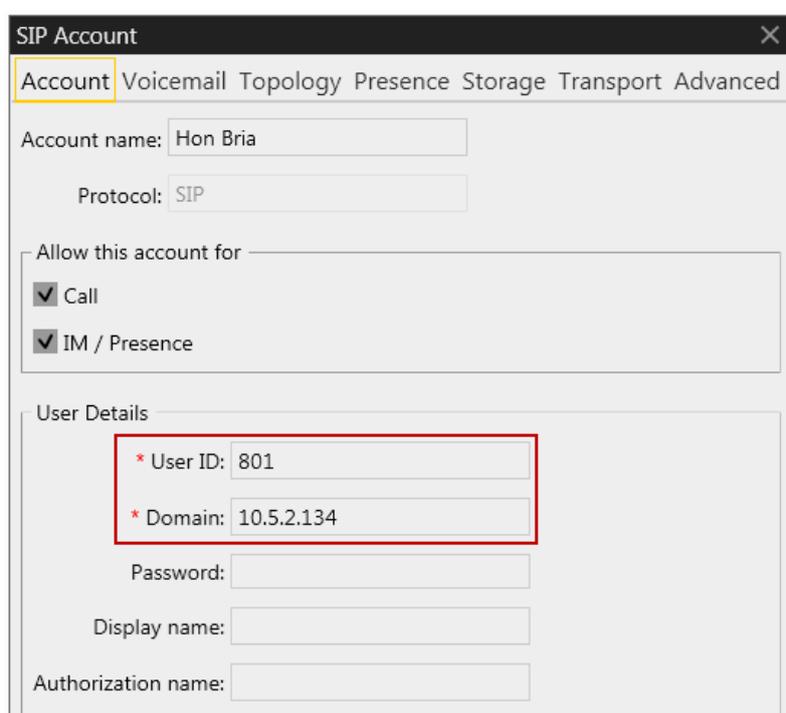
- Select **Softphone > Preferences > Video Codecs**



- Enable codec **H.264**

Account Settings

- Select **Softphone > Account Settings**



User Details

User ID: Directory number of Bria softphone

Domain: IP address of intercom station set as Pulse Server

7.4.6 Verifying Registration of Bria Softphone

When the configuration for both the Turbine Video station and the Bria softphone has been done, verify that they're both registered in the Pulse system.

- ① To verify station registration, you need to log into the station that has been set up as the Pulse Server.
- Select **Server Management > Server Monitoring**

The screenshot shows the 'Server Management' section of the Pulse system interface. On the left, there is a sidebar with 'Server Monitoring' selected. The main area displays the 'Station Directory' table. The table has columns for Directory Number, Name, Status, IP Address, Station Profile, and Terminal Type. The row for Directory Number 800 is highlighted with a red box, showing it is registered with the name 'video-turbine2' and IP address 10.5.2.134.

Directory Number	Name	Status	IP Address	Station Profile	Terminal Type
10		Registered	10.5.2.114	Default	STENTOFON Station
11	wadaw	Registered	10.5.2.171	Default	STENTOFON Station
800	video-turbine2	Registered	10.5.2.134	Default	STENTOFON Station
801	Hon Bria	Registered	10.5.2.160	Default	3rd Party SIP Terminal

Now you should be able to:

- Call the Bria softphone directly by pressing the call key on the Turbine Video station
- Call the Turbine Video station by dialing its number (e.g. 800) on the Bria softphone

To stream the video, you have to activate the video display on the Bria softphone.

7.5 Pulse Configuration for Snom Video Phone

- Snom Video Phone models: Snom 760 / Snom 821

7.5.1 SIP Settings

- Select **SIP Configuration > SIP Settings**

The screenshot shows the 'SIP Configuration' section of the Pulse system interface. The 'SIP Settings' tab is selected. The page is divided into 'Account Settings' and 'Call Settings' sections. In the 'Account Settings' section, the 'Directory Number (SIP ID)' is set to 800 and the 'Server Domain (SIP)' is set to 10.5.2.114, both fields are highlighted with a red box. The 'Call Settings' section includes options for 'Enable Auto Answer' (checked), 'Auto Answer Delay' (0 seconds), 'Press and Hold Time' (0 seconds), 'Max Ringing Time' (120 seconds), 'Max Conversation Time' (3600 seconds), 'Max Queued Time' (20 seconds), 'Max Queued Calls' (5), and 'Conversation Mode' (Full Open Duplex).

Description	Configuration
Display Name:	video-turbine2
Directory Number (SIP ID):	800
Server Domain (SIP):	10.5.2.114
Backup Domain (SIP):	
Backup Domain 2 (SIP):	
Authentication User Name:	800
Authentication Password:	
Register Interval:	600 (min. 60 seconds)

Description	Configuration
Enable Auto Answer:	<input checked="" type="checkbox"/>
Auto Answer Delay:	0 seconds. Max 30 seconds.
Press and Hold Time:	0 seconds. Max 60 seconds. Defines how long a DAK key/Input must be pressed before the call is established.
Max Ringing Time:	120 How long a call can be ringing before hanging up.
Max Conversation Time:	3600 How long a call can be in conversation before hanging up.
Max Queued Time:	20 How long a call can be queued before hanging up.
Max Queued Calls:	5 How many incoming calls can be queued. Max 5.
Conversation Mode:	Full Open Duplex

Account Settings

Directory Number (SIP ID): directory number of Turbine Video station

Server Domain (SIP): IP address of station set up as Pulse Server

- Click **Save**
- Click **Back to config page**

7.5.2 Video Settings for Snom Phone

- Select **SIP Configuration > Video Settings**

Description	Configuration
Video mode:	<input type="radio"/> H264 RTP <input checked="" type="radio"/> MJPG HTTP
Enable Video:	<input checked="" type="checkbox"/>
Resolution:	240P
Frames per second	15 fps
Camera IP address and port:	10.5.101.46:8090
Enable HTTP basic authentication:	<input type="checkbox"/>
Video setup mode:	Default

Description	Configuration
Lens distortion correction:	<input checked="" type="checkbox"/>
Night mode:	<input type="checkbox"/>
Zoom [1.00 2.50]x	1
Color saturation [0 255]:	128
Contrast [0 255]:	128
Brightness [0 255]:	128
Backlight compensation:	2

- Enter the values shown above for the parameters

Video Mode: Set to **MJPG HTTP**

Enable Video: Check box to enable video calls

Resolution: Select **240P**

Frames per second: Select **15fps**

Camera IP address and port: Enter the **port number** - default is **8090**

Video setup mode: Select **Default**

Camera IP address and port: Enter the **port number** - default is **8090**

① **The video camera and the Turbine station have the same IP address.**

- The video stream from the camera can be viewed by entering the IP address and port number in a web browser, e.g. **10.5.101.46:8090**

① The same **IP address** and **port number** set here must be entered into the settings for the Snom Phone described in section "7.5.5 Snom Phone Settings".

- Click **Save**
- Click **Back to config page**

7.5.3 Direct Access Key Settings

- Select **SIP Configuration > Direct Access Key Settings**

Function	Idle	Call	No Ringlist
DAK 1	Call To	802	No Ringlist
Input 1	Call To		No Ringlist
Input 2	Call To		No Ringlist
Input 3	Call To		No Ringlist
Input 4	Call To		No Ringlist

① This feature applies to TCIV-2 and TCIV-3 only

① See Turbine Configuration Manual for the configuration and import of an Address Book for TCIV-6.

To set up the call key on the Turbine station to call the Snom Video Phone directly:

- Enter the directory number of the Snom Video Phone in the **Value** field for **Direct Access Key 1**
 - In this example, the directory number of the Snom Video Phone is 802
- Click **Save**

7.5.4 Third Party SIP Terminals

① To configure third-party SIP terminals, you need to log into the station that has been set up as the Pulse Server.

- Select **Server Management > Server Configuration > Directory Settings**

Under **Third Party SIP Terminals**:

Directory Number	Name	Profile	Password	
		Default		Add
802	Hon Snom	Default		Delete

Save

- Enter the **Directory Number** and **Name** of the Snom Video Phone
- Click **Add** and **Save**

Under **STENTOFON Stations**:

- Click **Apply**
 - This will reboot all the stations in the Pulse system

7.5.5 Snom Phone Settings

- Log into the Snom phone interface by entering its IP address in a browser on your PC

Directory Setup

- Select **Operation > Directory**

Directory

Operation

- Home
- Directory**

Setup

- Preferences
- Speed Dial
- Function Keys
- Identity 1
- Identity 2
- Identity 3
- Identity 4
- Identity 5
- Identity 6
- Identity 7
- Identity 8
- Identity 9
- Identity 10
- Identity 11
- Identity 12
- Action URL Settings
- Advanced
- Certificates
- Software Update

Status

- System Information
- Log
- SIP Trace
- DNS Cache
- Subscriptions

Name:	Number:	Contact Type:	Outgoing
800 <- active identity	800	None	Active

Add or Edit Entry:

Number: 800

Number Type: sip

Contact Type: None

Outgoing Identity: Active

Group: None

Title:

Organization:

Email:

Note:

Photo:

Action-Url: http://10.5.2.134:8090/snoma.cgi

Nickname:

First Name:

Family Name:

Birthday:

- Enter the values shown above for the parameters

Number: Directory number of the Turbine Video station

Action-Url: http://<IP address of Turbine Video station>:<port no.>/snoma.cgi

- in this example: **http://10.5.2.134:8090/snoma.cgi**

Configuration Identity

- Select **Setup > Identity 1 > Login**

Configuration Identity 1

Operation

- Home
- Directory

Setup

- Preferences
- Speed Dial
- Function Keys
- Identity 1**
- Identity 2
- Identity 3
- Identity 4
- Identity 5
- Identity 6
- Identity 7
- Identity 8
- Identity 9
- Identity 10
- Identity 11
- Identity 12
- Action URL Settings
- Advanced
- Certificates

Login Information:

Identity active: on off

Displayname: Snom

Account: 802

Password:

Registrar: 10.5.2.134

Outbound Proxy:

Failover Identity: None

Authentication Username:

Mailbox:

Ringtone: Ringer 1

Custom Melody URL:

Display text for idle screen:

XML Idle Screen URL:

Ring After Delay (sec):

- Enter the values shown above for the parameters

Account: Directory number of the Snom phone, e.g. 802

Registrar: IP address of intercom station set as Pulse Server, e.g. 10.5.2.134

7.5.6 Verifying Registration of Snom Video Phone

When the configuration for both the Turbine Video station and the Bria softphone has been done, verify that they're both registered in the Pulse system.

① **To verify station registration, you need to log into the station that has been set up as the Pulse Server.**

- Select **Server Management > Server Monitoring**

Station Main	SIP Configuration	Station Administration	Server Management																														
<div style="display: flex;"> <div style="width: 20%;"> <p>▼ Server Monitoring</p> <p>▶ Server Configuration</p> <p>▶ Ringlist</p> <p>▶ Station Profiles</p> </div> <div style="width: 80%;"> <h3>Station Directory</h3> <table border="1"> <thead> <tr> <th>Directory Number</th> <th>Name</th> <th>Status</th> <th>IP Address</th> <th>Station Profile</th> <th>Terminal Type</th> </tr> </thead> <tbody> <tr> <td>10</td> <td></td> <td>Registered</td> <td>10.5.2.114</td> <td>Default</td> <td>STENTOFON Station</td> </tr> <tr> <td>11</td> <td>wadsw</td> <td>Registered</td> <td>10.5.2.171</td> <td>Default</td> <td>STENTOFON Station</td> </tr> <tr style="border: 2px solid red;"> <td>800</td> <td>video-turbine2</td> <td>Registered</td> <td>10.5.2.134</td> <td>Default</td> <td>STENTOFON Station</td> </tr> <tr style="border: 2px solid red;"> <td>802</td> <td>Hon Snom</td> <td>Registered</td> <td>10.5.2.158</td> <td>Default</td> <td>3rd Party SIP Terminal</td> </tr> </tbody> </table> </div> </div>				Directory Number	Name	Status	IP Address	Station Profile	Terminal Type	10		Registered	10.5.2.114	Default	STENTOFON Station	11	wadsw	Registered	10.5.2.171	Default	STENTOFON Station	800	video-turbine2	Registered	10.5.2.134	Default	STENTOFON Station	802	Hon Snom	Registered	10.5.2.158	Default	3rd Party SIP Terminal
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802	Hon Snom	Registered	10.5.2.158	Default	3rd Party SIP Terminal																												

Now you should be able to:

- Call the Snom phone directly by pressing the call key on the Turbine Video station
- Call the Turbine Video station by dialing its number (e.g. 800) on the Snom phone

7.6 Pulse Configuration for VS Desktop Video Display Station

- Vingtor-Stentofon IP Desktop Video Station - item no. 1408001635

7.6.1 SIP Settings

- Select **SIP Configuration > SIP Settings**

Station Main	SIP Configuration	Station Administration																																				
<div style="display: flex;"> <div style="width: 20%;"> <p>▼ SIP Settings</p> <p>▶ Audio Settings</p> <p>▶ Direct Access Key Settings</p> <p>▶ Relay Settings</p> <p>▶ Time Settings</p> <p>▶ I/O Settings</p> <p>▶ Video Settings</p> <p>▶ Script Configuration</p> <p>▶ Script Events</p> <p>▶ Script Upload</p> </div> <div style="width: 80%;"> <h3>Account Settings</h3> <table border="1"> <thead> <tr> <th>Description</th> <th>Configuration</th> </tr> </thead> <tbody> <tr> <td>Display Name:</td> <td>video-turbine2</td> </tr> <tr style="border: 2px solid red;"> <td>Directory Number (SIP ID):</td> <td>800</td> </tr> <tr style="border: 2px solid red;"> <td>Server Domain (SIP):</td> <td>10.5.2.114</td> </tr> <tr> <td>Backup Domain (SIP):</td> <td></td> </tr> <tr> <td>Backup Domain 2 (SIP):</td> <td></td> </tr> <tr> <td>Authentication User Name:</td> <td>800</td> </tr> <tr> <td>Authentication Password:</td> <td></td> </tr> <tr> <td>Register Interval:</td> <td>60 (Minimum 60 seconds)</td> </tr> </tbody> </table> <h3>Call Settings</h3> <table border="1"> <thead> <tr> <th>Description</th> <th>Configuration</th> </tr> </thead> <tbody> <tr> <td>Enable Auto Answer:</td> <td><input checked="" type="checkbox"/></td> </tr> <tr> <td>Auto Answer Delay:</td> <td>0 seconds. Max 30 seconds.</td> </tr> <tr> <td>Delay Call Setup:</td> <td>0 seconds. Max 60 seconds. Delays call setup using DAK/Input buttons.</td> </tr> <tr> <td>Max Ringing Time:</td> <td>120 seconds. How long a call can be ringing before hanging up.</td> </tr> <tr> <td>Max Conversation Time:</td> <td>3600 seconds. How long a call can be in conversation before hanging up. (0 = disable timeout)</td> </tr> <tr> <td>Max Queued Time:</td> <td>20 seconds. How long a call can be queued before hanging up.</td> </tr> <tr> <td>Send DTMF *# with M key:</td> <td><input checked="" type="checkbox"/></td> </tr> <tr> <td>Codec g729:</td> <td>Medium Priority ▼</td> </tr> </tbody> </table> </div> </div>			Description	Configuration	Display Name:	video-turbine2	Directory Number (SIP ID):	800	Server Domain (SIP):	10.5.2.114	Backup Domain (SIP):		Backup Domain 2 (SIP):		Authentication User Name:	800	Authentication Password:		Register Interval:	60 (Minimum 60 seconds)	Description	Configuration	Enable Auto Answer:	<input checked="" type="checkbox"/>	Auto Answer Delay:	0 seconds. Max 30 seconds.	Delay Call Setup:	0 seconds. Max 60 seconds. Delays call setup using DAK/Input buttons.	Max Ringing Time:	120 seconds. How long a call can be ringing before hanging up.	Max Conversation Time:	3600 seconds. How long a call can be in conversation before hanging up. (0 = disable timeout)	Max Queued Time:	20 seconds. How long a call can be queued before hanging up.	Send DTMF *# with M key:	<input checked="" type="checkbox"/>	Codec g729:	Medium Priority ▼
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Account Settings

Directory Number (SIP ID): directory number of Turbine Video station

Server Domain (SIP): IP address of station set up as Pulse Server

- Click **Save**
- Click **Back to config page**

7.6.2 Video Settings for Desktop Video Display Station

- Select **SIP Configuration > Video Settings**

Station Main		SIP Configuration	Station Administration	Advanced Network																								
<table border="1"> <tr> <td>▶ SIP Settings</td> <td>Video Settings</td> </tr> <tr> <td>▶ Audio Settings</td> <td></td> </tr> <tr> <td>▶ Direct Access Key Settings</td> <td></td> </tr> <tr> <td>▶ Relay Settings</td> <td></td> </tr> <tr> <td>▶ Time Settings</td> <td></td> </tr> <tr> <td>▶ I/O Settings</td> <td></td> </tr> <tr> <td>▼ Video Settings</td> <td></td> </tr> <tr> <td>▶ Script Configuration</td> <td></td> </tr> <tr> <td>▶ Script Events</td> <td></td> </tr> <tr> <td>▶ Script Upload</td> <td></td> </tr> <tr> <td>▶ Audio Messages</td> <td></td> </tr> <tr> <td>▶ Certificates</td> <td></td> </tr> </table>					▶ SIP Settings	Video Settings	▶ Audio Settings		▶ Direct Access Key Settings		▶ Relay Settings		▶ Time Settings		▶ I/O Settings		▼ Video Settings		▶ Script Configuration		▶ Script Events		▶ Script Upload		▶ Audio Messages		▶ Certificates	
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- Enter the values shown above for the parameters

Video Mode: Set to **MJPG HTTP**

Enable Video: Check box to enable video calls

Resolution: Select **240P**

Frames per second: Select **15fps**

Camera IP address and port: Enter the **port number** - default is **8090**

Video setup mode: Select **Default**

Camera IP address and port: Enter the **port number** - default is **8090**

① **The video camera and the Turbine station have the same IP address.**

- The video stream from the camera can be viewed by entering the IP address and port number in a web browser, e.g. **10.5.101.46:8090**

① The same **IP address** (e.g. **10.5.101.46**) and **port number** (e.g. **8090**) set here must be entered into the settings for the Desktop Video Display Station described in section "7.6.4 Desktop Video Display Station Settings".

- Click **Save**
- Click **Back to config page**

7.6.3 Direct Access Key Settings

- Select **SIP Configuration > Direct Access Key Settings**

	Function	Idle	Call	No Ringlist
DAK 1		Call To	802	No Ringlist
Input 1		Call To		No Ringlist
Input 2		Call To		No Ringlist
Input 3		Call To		No Ringlist
Input 4		Call To		No Ringlist

① This feature applies to TCIV-2 and TCIV-3 only

① See Turbine Configuration Manual for the configuration and import of an Address Book for TCIV-6.

To set up the call key on the Turbine station to call the Desktop Video Display Station directly:

- Enter the directory number of the Desktop Video Display Station in the **Value** field for **Direct Access Key 1**
 - In this example, the directory number of the Desktop Video Display Station is **802**
- Click **Save**

7.6.4 Desktop Video Display Station Settings

The camera of the Turbine Video station has to be set in the video touchscreen of the station. This is done by logging into the video touchscreen of the desktop station interface.

- Tapping anywhere on the LCD touchscreen will show the IP address of the video display part.



- Enter the **Video-IP** address as shown above (e.g. 192.168.45.45) in a web browser to log into the video part of the desktop station.

To log into the video part of the station:

1. Enter the default Username: **admin**
2. Enter the default password: **alphaadmin**



- User
- User Interface
- Cameras
- Network
- System

Firmware-Version: v2.3
MAC-Address: 74-19-F8-80-09-E0

Requirements: JavaScript

- Click **Cameras**

[<- back](#)

Camera Settings

Passcode

Passcode (Numbers only)

Camera Types

Baudisch	:80/mjpg/video.mjpg
AXIS	:80/axis-cgi/mjpg/video.cgi
<input type="text" value="TCIV"/>	<input type="text" value=":8090/mjpg/video.mjpg"/>
<input type="text"/>	<input type="text"/>
<input type="text"/>	<input type="text"/>

Camera Allocation

/ 80

Name	<input type="text" value="FrontDoor"/>
AlphaCom Node Number	<input type="text" value="0"/>
AlphaCom Directory Number	<input type="text" value="0"/>
SIP ID	<input type="text" value="800"/>
Camera IP	<input type="text" value="10.5.101.46"/>
Camera Type	<input type="text" value="TCIV"/>
Camera User	<input type="text"/>
Camera Password	<input type="text"/>
Passcode required	<input type="checkbox"/>
Allocation active?	<input checked="" type="checkbox"/>

- Enter values for Camera Types and Camera Allocation as shown

Camera Types

- Define Camera Type for Turbine Video station TCIV by entering URL :**8090/mjpg/video.mjpg**
- ① **'8090' is the default port number** for the camera set in section “7.6.2 Video Settings for Desktop Video Display Station”.

Camera Allocation

AlphaCom Node Number : 0 (Not in use for Pulse system)

AlphaCom Directory Number : 0 (Not in use for Pulse system)

SIP ID : Directory number of TCIV as specified in section 7.6.1 (e.g. 800)

Camera IP : IP address of TCIV (e.g. 10.5.101.46)

Camera Type : TCIV

Allocation active? : Check the box to enable video streaming from the camera

- Click **Submit settings**

8 Station Indication LEDs

8.1 LEDs on Front Plate

Status LEDs

- **Bell icon** lights **yellow** when a call is placed and ringing
- **Talk icon** lights **green** when a call is active and in conversation
- **Door icon** lights **red** when the door is unlocked or relay is active



Talk Icon: Flashing at 1 second intervals

- Station has no connection to the AlphaCom server/exchange.
Possible reasons:
 - No connection to Ethernet
 - Wrong AlphaCom XE IP address configured
 - Invalid IP address
 - No gateway or wrong gateway to the AlphaCom server/exchange



Talk Icon: Flashing at 5 second intervals



- Station connected but NOT registered in the AlphaCom server/exchange.
Reason:
 - Station has not been programmed in AlphaPro

9 Restoring Factory Defaults

A Turbine IP Station may have to be reset to its original factory default settings if, for instance, the password to the station web interface is forgotten. The defaults can either be set to Activated DHCP or Static IP.

9.1 Reset to Factory Default Settings with Activated DHCP

To reset:



1. While **pressing any button**, power up the station by connecting to a PoE switch.
2. Hold the button until the station audio starts counting, and release the button on **count 1**.
3. Press and hold the button on **count 5** and release on **count 0**.
- if there is no 0 count, the procedure has failed and you have to start again
4. Press the **call button** to make the station speak its IP address.

Factory default values

Station IP address: (determined by DHCP server)

Username: **admin**

Password: **alphaadmin**

9.2 Reset to Factory Default Settings with Static IP

To reset:



1. While **pressing any button**, power up the station by connecting to a PoE switch.
2. Hold the button until the station audio starts counting, and release the button on **count 1**.
3. Press and hold the button on **count 3** and release on **count 0**.
- if there is no 0 count, the procedure has failed and you have to start again
4. Press the **call button** to make the station speak its IP address.

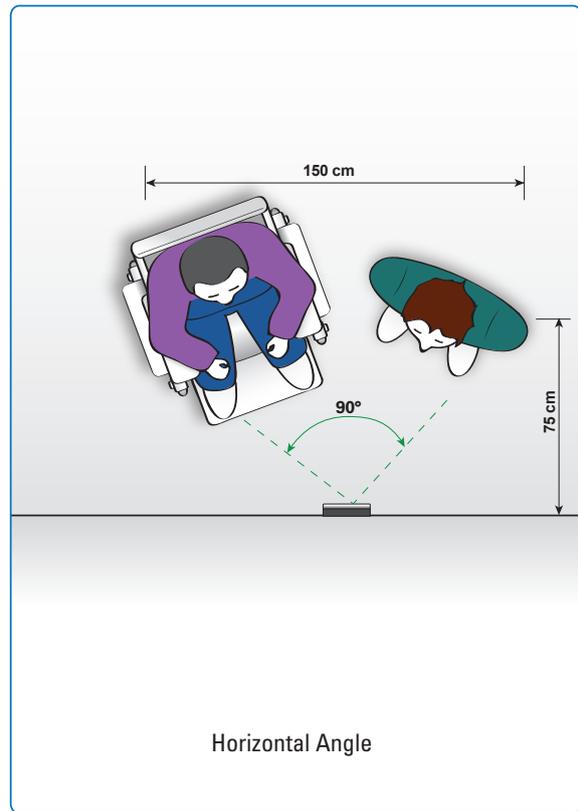
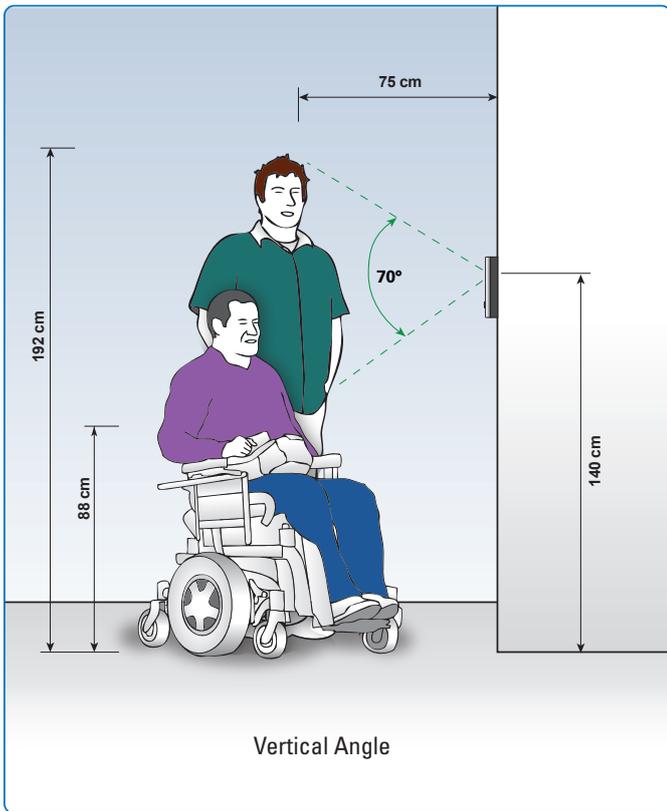
Factory default values

Station IP address: **169.254.1.100**

Username: **admin**

Password: **alphaadmin**

10 Camera Field of View





The WEEE Directive does not legislate that Zenitel, as a 'producer', shall collect 'end of life' WEEE.

This 'end of life' WEEE should be recycled appropriately by the owner who should use proper treatment and recycling measures. It should not be disposed to landfill.

Many electrical items that we throw away can be repaired or recycled. Recycling items helps to save our natural finite resources and also reduces the environmental and health risks associated with sending electrical goods to landfill.



Under the WEEE Regulations, all new electrical goods should now be marked with the crossed-out wheeled bin symbol shown.

Goods are marked with this symbol to show that they were produced after 13th August 2005, and should be disposed of separately from normal household waste so that they can be recycled.

