

# Logging into the Station

Ensure that the IP address of your PC is in the same range as that of the station IP address. Access the station by logging into the web interface using a standard web browser on your PC:

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 Username: **admin**
3. Enter the default Password: **alphaadmin**



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

## Main Settings

- Click Main > Main Settings

**Mode**

Select preferred mode for your device. If your system is Edge, please log on to the device you will use as the Edge Controller. You can do all configuration of your devices from the Edge Controller.

ICX-AlphaCom

SIP

Edge

Edge Controller

Model:

**IP Settings**

Preferred Internet Protocol:

DHCP  Static IP

IP Address:	192	-	168	-	0	-	200
Subnet Mask:	255	-	255	-	255	-	0
Gateway:	192	-	168	-	0	-	1
DNS Server 1:	192	-	168	-	0	-	1
DNS Server 2:	8	-	8	-	8	-	8
Hostname:	<input type="text" value="zenitel282e38"/>						
Disable Reset to Factory default settings using frontboard and I/O:	<input checked="" type="checkbox"/>						
Read IP Address:	<input checked="" type="checkbox"/>						
Ethernet Speed 10 Mbit/s:	<input type="checkbox"/>						

## IP Settings:

- DHCP – Select this option if the IP station shall receive IP Settings from a DHCP server.
- Static IP – Select this option if the IP station shall use a static IP address. Enter values for:
  - IP-address
  - Subnet-mask
  - Gateway
  - DNS Server 1 (option for network administration)
  - DNS Server 2 (option for network administration)
  - Hostname (option for network administration)

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

- Check the box to disable factory reset using front board and I/O.

Read IP Address:

- Read IP Address enables an unregistered station to speak the IP address when the call button is pressed.
- Read IP Address box is checked, i.e. enabled, by default.

Enable RSTP: (for Industrial & Ex)

- Check the Enable RSTP box to enable RSTP. RSTP is only required when using redundant networking.
- Click SAVE followed by APPLY

## Account/Call Settings:

- Select SIP Configuration > Account / Call

Main SIP Configuration Station Administration

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Account / Call

- Audio
- Direct Access Keys
- Relays / Outputs
- Time
- Video
- Audio Messages
- Certificates

### Account Settings

Description	Configuration
Name:	Türstation
Number (SIP ID):	901
Server Domain (SIP):	192.168.0.250
Backup Domain (SIP):	
Backup Domain 2 (SIP):	
Registration Method:	Parallel
Authentication User Name:	901
Authentication Password:	...
Register Interval:	100 (min. 30 seconds)
Register Failure Interval:	60 (min. 5 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
Verify TLS hostname:	<input type="checkbox"/>
TLS Private Key:	turbine_server_sha256.key

### Call Settings

Description	Configuration
Enable Auto Answer:	<input checked="" type="checkbox"/>

SAVE

# Direct Access Key & Ring list Settings

There are two ways of configuring DAKs, Inputs and Ring List functions.

1. Via the Edge Controller whereby all stations in the system can be configured:
  - Select Edge Controller > System Configuration > Direct Access Keys
2. Via each individual station as describe below:
  - Select SIP Configuration > Direct Access Keys

zenitel
WEB CONFIGURATION

Main
SIP Configuration
Station Administration

- ▶ Account / Call
- ▶ Audio
- ▼ Direct Access Keys
- ▶ Relays / Outputs
- ▶ Time
- ▶ Video
- ▶ Audio Messages
- ▶ Certificates

### Account Settings

	Function				
Button 1	Idle:	<input type="text" value="Do Nothing"/>			
	Call:	<input type="text" value="Do Nothing"/>			
Input 1	Idle:	<input type="text" value="Call To"/>	<input type="text"/>	<input type="text" value="Ringlist 1"/>	<input type="text"/>
	Call:	<input type="text" value="Answer Call"/>	<input type="text" value="Filter Dir. No."/>	<input checked="" type="checkbox"/>	Answer Group Call
Input 2	Idle:	<input type="text" value="Call To"/>	<input type="text"/>	<input type="text" value="No Ringlist"/>	<input type="text"/>
	Call:	<input type="text" value="Do Nothing"/>			
Input 3	Idle:	<input type="text" value="Call To"/>	<input type="text"/>	<input type="text" value="No Ringlist"/>	<input type="text"/>
	Call:	<input type="text" value="Do Nothing"/>			
Input 4	Idle:	<input type="text" value="Call To"/>	<input type="text"/>	<input type="text" value="No Ringlist"/>	<input type="text"/>
	Call:	<input type="text" value="Do Nothing"/>			
Input 5	Idle:	<input type="text" value="Call To"/>	<input type="text"/>	<input type="text" value="No Ringlist"/>	<input type="text"/>
	Call:	<input type="text" value="Do Nothing"/>			
Input 6	Idle:	<input type="text" value="Call To"/>	<input type="text"/>	<input type="text" value="No Ringlist"/>	<input type="text"/>
	Call:	<input type="text" value="Do Nothing"/>			

SAVE

### Ringlist Settings

	Ringlist 1	With Previous	Ringlist 2	With Previous	Ringlist 3	With Previous
Value 1	<input type="text" value="sip:0138248@sip.fermax."/>	<input checked="" type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 2	<input type="text" value="sip:101@192.168.0.100"/>	<input checked="" type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 3	<input type="text" value="sip:101@192.168.0.250"/>	<input checked="" type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 4	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 5	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 6	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>
Value 7	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>	<input type="text"/>	<input type="checkbox"/>

SAVE

**Note!** The availability of these parameters and the number of keys depend on the Turbine station type selected under Main Settings.

**Note!** Some telephone gateways accept a call immediately on incoming calls – even before the call has been accepted by the user (telephone). This terminates the ringlist sequence and any subsequent numbers in the ringlist will never be called. This is solved by placing the gateway number at the end of the ringlist and disabling the “With Previous” option. Ringlists that include an auto-accepting gateway will never loop back.

DAKs and Inputs have two states:

Idle: There is no active call on the station.

Call: There is an incoming, outgoing or active call on the station.

Idle: In Idle state the following options are available (available options will depend on the station type):

- **Call To:** Enter the directory number to call when the DAK/Input is pressed. Extended call options (parallel ringing, call escalation, etc.) are available if the call is routed via the Ring list.
- **Forward Call:** Forward calls to the configured destination. Forwarding will be toggled On / Off every time the DAK key is pressed. Forwarding status is shown on DAK keys LEDs (ECPIR panel). Red LED indicates that Forwarding is turned on. The Forwarding function is not available on INCA stations, Substations and Kits.
- **Group Call:** Enter the directory number of the Group Call to call when the DAK/Input is pressed.

Two modes of operation are available:

- **Open (No M-key):** When the DAK key is activated, the microphone is open, and the user can talk handsfree
- **PTT (M-key):** When the DAK key is activated, the user must activate the microphone by pressing and holding the M-key (PTT key)

- **Conversation Mode:** The station will toggle between Simplex mode (PTT - Press to Talk) and duplex mode (hands-free) every time the DAK is pressed. The selected mode is permanently stored. This function works in idle as well as during a call.

- **Volume Control:** Change speaker volume by 1 point (+/- 4 dB). This function works in idle as well as during a call.

There are two options:

- Down: decrease speaker volume by 1 point
- Up: increase speaker volume by 1 point

- **Call:**

During an active Call the following options are available (available options depend on station type):

- **Do Nothing:** No action executed if the DAK/Input is activated during a call (Default)
- **Answer/End Call:** Answers an incoming call and ends an active call. Action can be executed either On Key Press or On Key Release. Has an option to also Answer Group Call.
- **End Call:** Ends an active call. Action can be executed either On Key Press or On Key Release
- **Answer Call:** Answers an incoming call. Has an option to also Answer Group Call.
- **Transfer Call:** The “Transfer Call” function can work in two modes:
  - Transfer immediately on key press when destination call number is configured

- **Select transfer destination manually.** When a destination call number is not configured, a key press will set the station in "Transfer Mode", waiting for the user to dial the transfer destination, or to transfer to a predefined destination by pressing a key with "Call" action.

The active call is immediately hung-up if the transfer has progressed. If transfer fails (e.g. destination not found), the transfer will be aborted.

- **Park Call:** Places a call on hold into a specific parking location (a fictional number) such that the call can then be picked up by another number.

- **Hold Call:** When a "Call" button is configured to do "Hold Call" during the call, the button has different functionalities depending on the current state:

Incoming Call: Accept call

Established call: Put call on hold (disable audio)

On hold: Resumes a call (enable audio)

- **Send DTMF:** Sends configured DTMF for key press and key release.

- **Send Text:** Sends configured text for key press and key release. Enter text in the text field.

- **Push To Talk:** Configure key to behave like a PTT key during active call.

- **Standby:** Standby will reduce the volume of local speaker by a selected dB value, for the duration of the call or until pressed again.

- **Defer:** Enables you to save the details of a call as it has been logged or updated, and to put off any action until a later stage.

Depending on the type of station and configured accessories, DAK Settings may also show additional options

- Module keys 1 .... n (e.g. ECPIR-3P expansion module buttons)

- PTT / M-key

- Off hook

- On hook

# Video Settings

1. Via each individual station as describe below:
  - Select SIP Configuration > Video Settings

Main
SIP Configuration
Station Administration

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- ▶ Account / Call
- ▶ Audio
- ▶ Direct Access Keys
- ▶ Relays / Outputs
- ▶ Time
- ▶ Video
- ▶ Audio Messages
- ▶ Certificates

## Video Settings

### SIP H264 Call Settings

Description	Configuration
Resolution	720P
Quality	High
Frames per second	25 fps
Bitrate	1500 kb/s

### RTSP H264 Settings

Description	Configuration
Resolution	1080P
Quality	High
Frames per second	25 fps
Bitrate	2500 kb/s
RTSP URL	rtsp://192.168.0.200:554/media?encoder=H264

### HTTP/MJPEG Settings

Description	Configuration (fixed)
Resolution	480P
Frames per second	15
HTTP/MJPEG URL	http://192.168.0.200/mjpg/video.mjpg

Description	Configuration		Set to Default
Rotation	None	None - Default value	Default
Brightness		128 - Default value	Default
Saturation		128 - Default value	Default
Contrast		128 - Default value	Default
Zoom		1 - Default value	Default
Pan/Tilt			
Live preview HTTP/MJPEG			

SAVE

# Advanced Configuration Mode

Settings for I/O, Scripts, etc. described in the following sections are only available in the Advanced Configuration Mode.

To enter Advanced Configuration Mode:

- Select Main > Recovery

Main SIP Configuration Station Administration

Information  
Main Settings  
Recovery

**Commands**

Description	Action
Full reboot	REBOOT
Partial reboot	REBOOT
Factory reset	FACTORY RESET
Factory reset with DHCP	FACTORY RESET

**Preferences**

Description	Configuration
Advanced configuration mode	Type offline password to unlock advanced configuration mode

Under Preferences:

- Enter the offline password 1851 in the Advanced configuration mode field

Main SIP Configuration Station Administration

Information  
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**Commands**

Description	Action
Full reboot	REBOOT
Partial reboot	REBOOT
Factory reset	FACTORY RESET
Factory reset with DHCP	FACTORY RESET

**Preferences**

Description	Configuration
Advanced configuration mode	<input checked="" type="checkbox"/> Type offline password to unlock advanced configuration mode

SAVE

- Check the Configuration box

- Click SAVE

Main SIP Configuration Station Administration

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Advanced configuration mode activated. Reload page to continue.

RELOAD

- Click RELOAD

New menu items such as I/O, Script under SIP Configuration and a new tab called Advanced SIP will now appear:

## Script Settings

Script (Virtual I/O) is a feature for activating scripts on station events. These scripts can be uploaded and configured via the menu options:

- Script Upload
- Script
- Script Events

## Script Upload

- Select Edge Configuration > Script Upload

# ADVANCED WEB CONFIGURATION

VINGTOR STENTOFON

Main SIP Configuration Station Administration Advanced SIP Advanced Network

- > Account / Call
- > Audio
- > DAVC
- > Direct Access Keys
- > Relays / Outputs
- > Time
- > I/O
- > Frontboard Mapping
- > Video
- > Advanced Video
- ▼ Script Upload
- > Script Configuration
- > Script Events

### Scripts

Name	
Script 1	Webrelay1.lua <span>DELETE</span>

Space used: 0.001 mb of 20 mb

### Upload Script

Keine Datei ausgewählt

- Click UPLOAD to upload the desired script

# ADVANCED WEB CONFIGURATION

VINGTOR STENTOFON

Main SIP Configuration Station Administration Advanced SIP Advanced Network

- > Account / Call
- > Audio
- > DAVC
- > Direct Access Keys
- > Relays / Outputs
- > Time
- > I/O
- > Frontboard Mapping
- > Video
- > Advanced Video
- ▼ Script Upload
- > Script Configuration
- > Script Events

```
#!/usr/bin/env lua
-- Script creator: dietmar.schlichtherle@siblik.com
-- Script to activate a WebRelay™ | Single Relay and Input Module >> https://www.controlbyweb.com/webrelay/
-- Modify the IP address in the line "local url" for adaptation to your network environment.
-- Support: xenios.maroudas@scanvest.de

local io = require("io")
local http = require("socket.http")
local ltn12 = require("ltn12")

-- Set http timeout to 10 seconds (if the system is too slow then this timeout must be higher)
http.TIMEOUT = 10

-- URL to trigger the WebRelay
local url = "http://192.168.0.50/state.xml?relay1State=2&pulseTime=5"

-- local url = "http://192.168.0.50/state.xml?relay2State=2&pulseTime=5"
-- local url = "http://192.168.0.50/state.xml?relay2State=2&pulseTime=5"
-- local url = "http://192.168.0.50/state.xml?relay4State=2&pulseTime=5"

print ("Requesting relay activation")

local r, c, h = http.request{
    url= url,
    method = "GET"
}
```

## Script Configuration

- Select Edge Configuration > Script

### Digital Outputs - Scripts:

- Select the script Slot 1 - Slot 10 to configure
- Assign a logical name to the script in the Assign a Label field
- Enter the script to activate and add parameters in the text field under the Enter script in selected slot field

The screenshot shows the 'ADVANCED WEB CONFIGURATION' interface for 'DIGITAL OUTPUTS - SCRIPTS'. The top navigation bar includes 'Main', 'SIP Configuration', 'Station Administration', 'Advanced SIP', and 'Advanced Network'. A sidebar on the left lists various configuration categories, with 'Script Configuration' highlighted. The main content area is titled 'Digital Outputs - Scripts' and features a dropdown menu for 'Choose script slot to configure:' set to 'Slot 1'. Below this, there are two input fields: 'Assign a Label' containing 'Webrelay1' and 'Enter script in selected slot' containing 'lua Webrelay1.lua'. A yellow 'SAVE' button is positioned at the bottom right of the configuration area.

## Script Events

- Select Edge Configuration > Script Events

For more information on Script settings, see [wiki.zenitel.com/wiki/Virtual I/O \(SIP\)](http://wiki.zenitel.com/wiki/Virtual_I/O_(SIP))

- › Account / Call
- › Audio
- › DAVC
- › Direct Access Keys
- › Relays / Outputs
- › Time
- › I/O
- › Frontboard Mapping
- › Video
- › Advanced Video
- › Script Upload
- › Script Configuration
- ▼ Script Events
- › Audio Messages
- › Multicast Paging
- › Certificates

## Digital Outputs - Events

Event	Execute		
Remote Digit (DTMF):	Digit # ▼	Webrelay1 ▼	after a duration of 0 ▼
Incoming Call:	- ▼	after a duration of	0 ▼
Outgoing Call:	- ▼	after a duration of	0 ▼
Incoming Ringing:	- ▼	after a duration of	0 ▼
Outgoing Ringing:	- ▼	after a duration of	0 ▼
Idle:	- ▼	after a duration of	0 ▼
Error (Not Registered):	- ▼	after a duration of	0 ▼
Input 1 Pressed	- ▼	after a duration of	0 ▼
Input 1 Released	- ▼	after a duration of	0 ▼
Input 2 Pressed	- ▼	after a duration of	0 ▼
Input 2 Released	- ▼	after a duration of	0 ▼
Input 3 Pressed	- ▼	after a duration of	0 ▼
Input 3 Released	- ▼	after a duration of	0 ▼
Input 4 Pressed	- ▼	after a duration of	0 ▼
Input 4 Released	- ▼	after a duration of	0 ▼
Input 5 Pressed	- ▼	after a duration of	0 ▼
Input 5 Released	- ▼	after a duration of	0 ▼
Input 6 Pressed	- ▼	after a duration of	0 ▼
Input 6 Released	- ▼	after a duration of	0 ▼

SAVE