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.

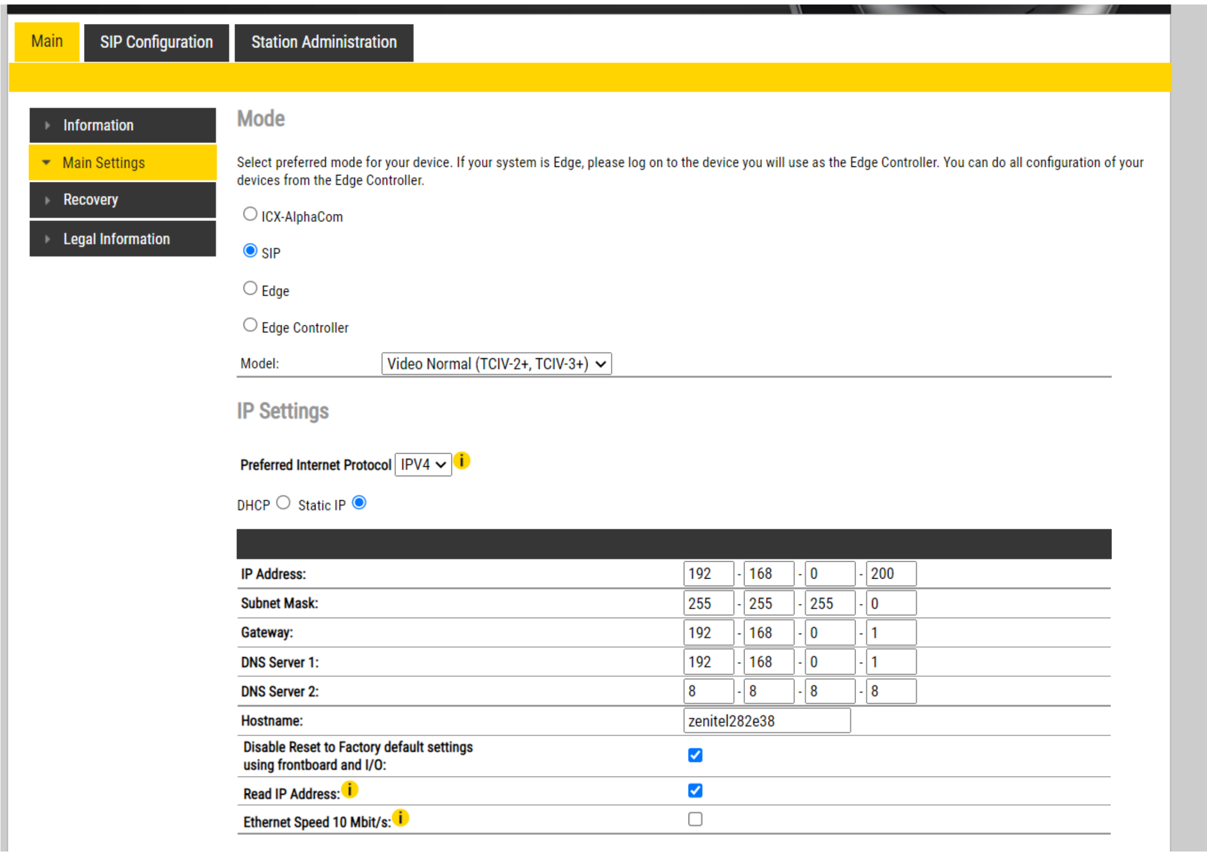
To log into the station:

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



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)

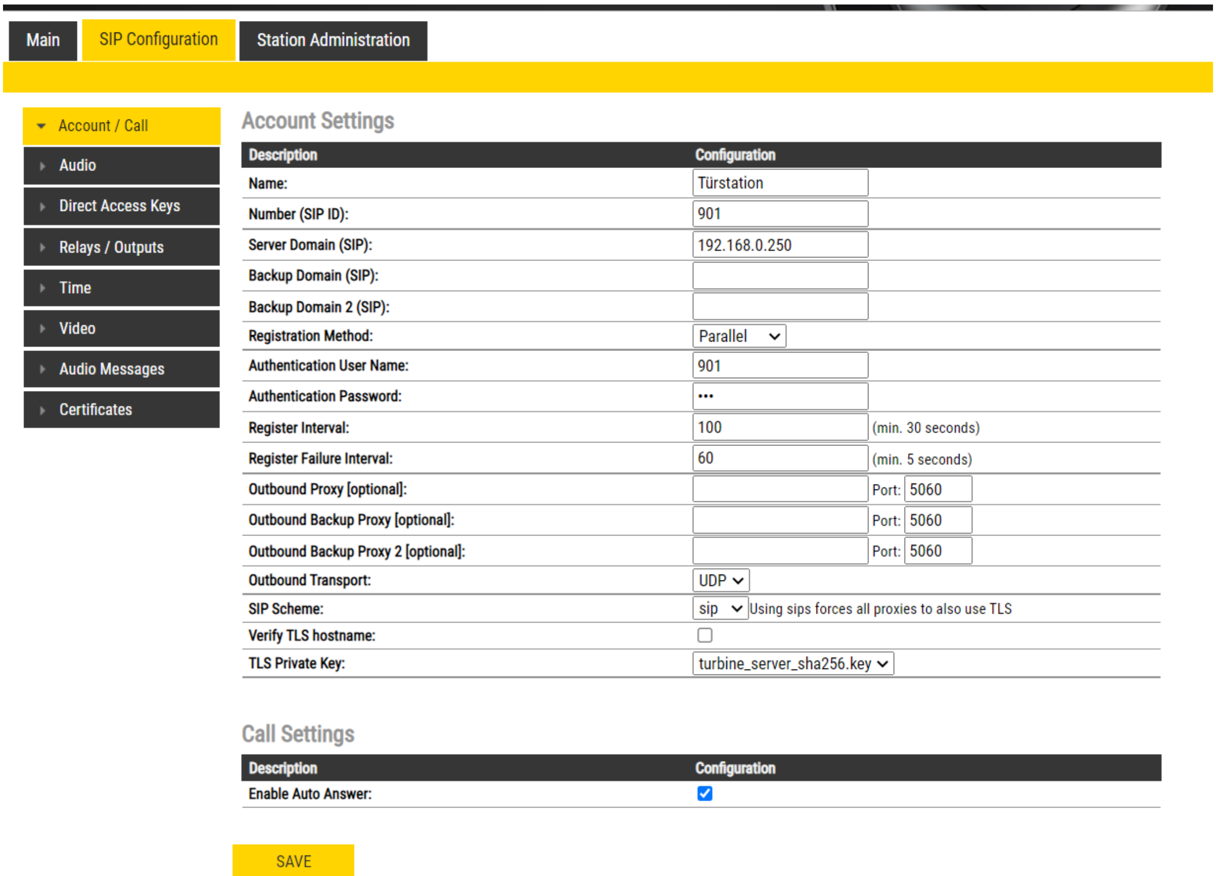
Disable Reset to Factory default settings using front board and I/O:
- Check the box to disable factory reset using front board and I/O.

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



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:	☐
TLS Private Key:	turbine_server_sha256.key

Call Settings

Description	Configuration
Enable Auto Answer:	☑

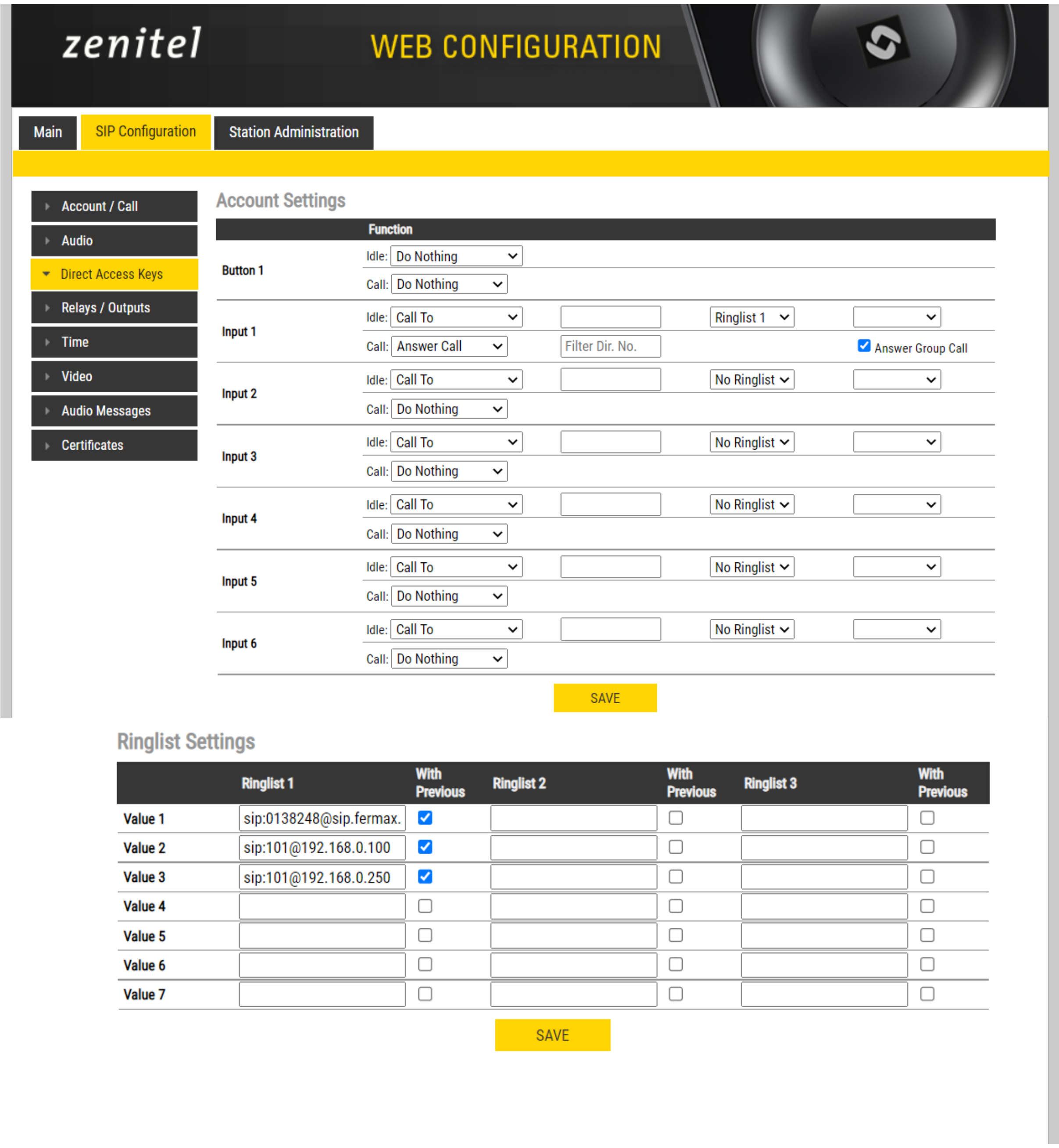
SAVE

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



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)

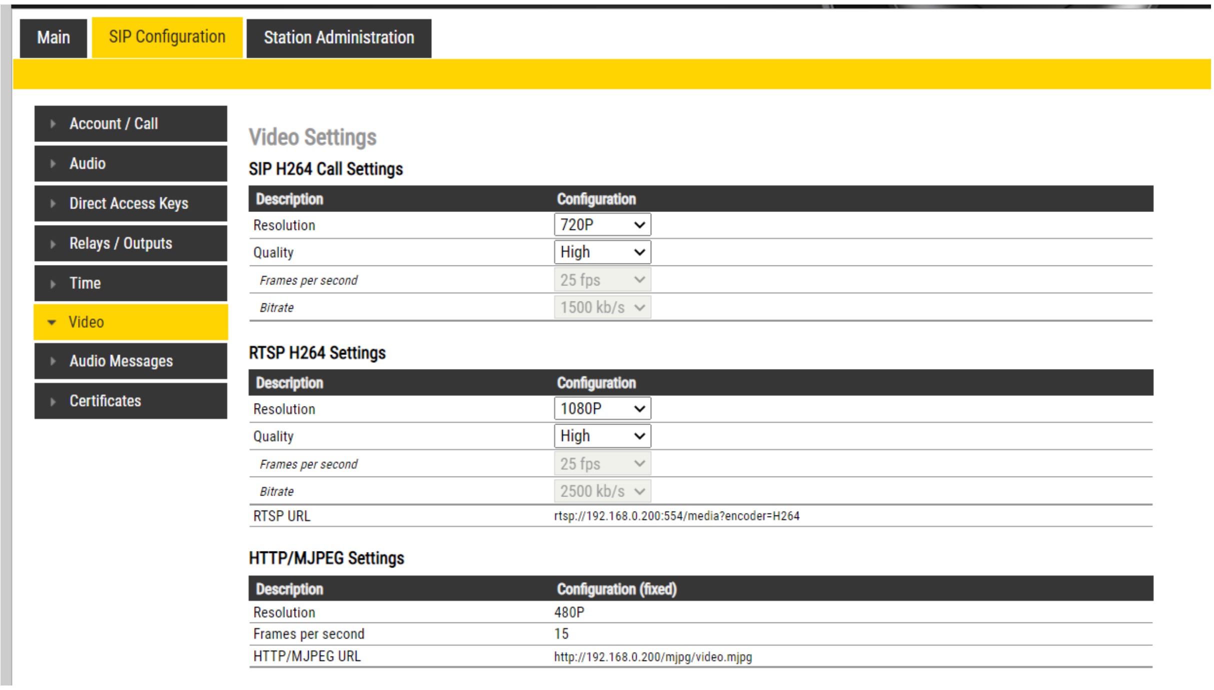
• PTT / M-key

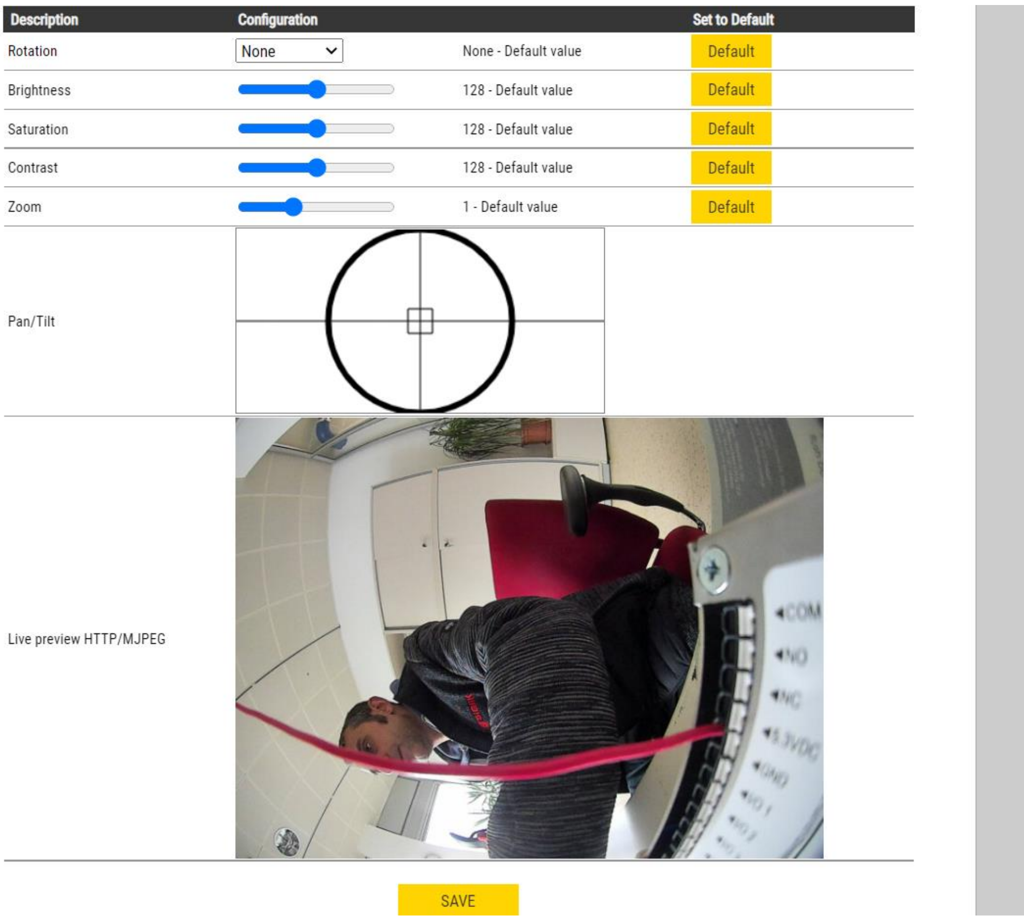
• Off hook

• On hook

Video Settings

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



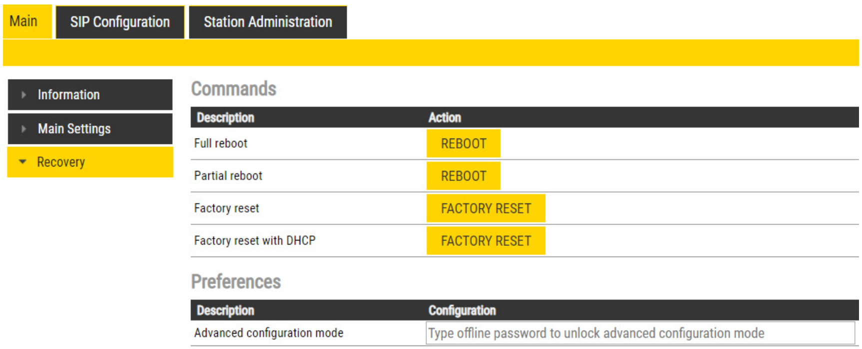


Advanced Configuration Mode

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

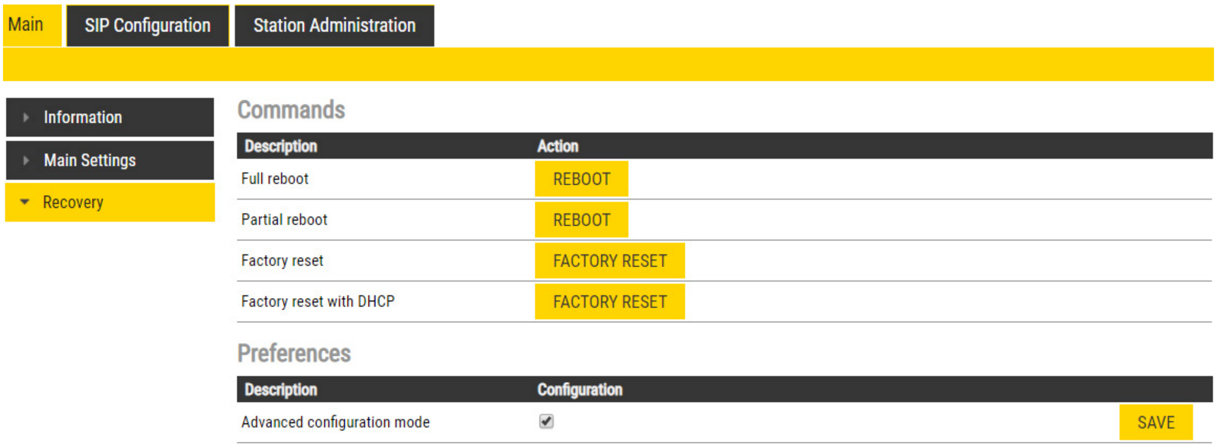
To enter Advanced Configuration Mode:

- Select Main > Recovery

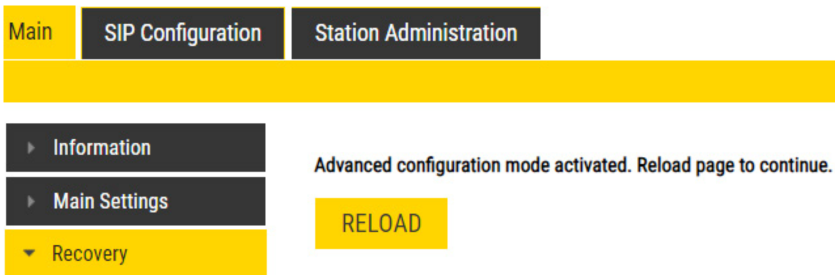


Under Preferences:

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



- Check the Configuration box
- Click SAVE



- Click RELOAD

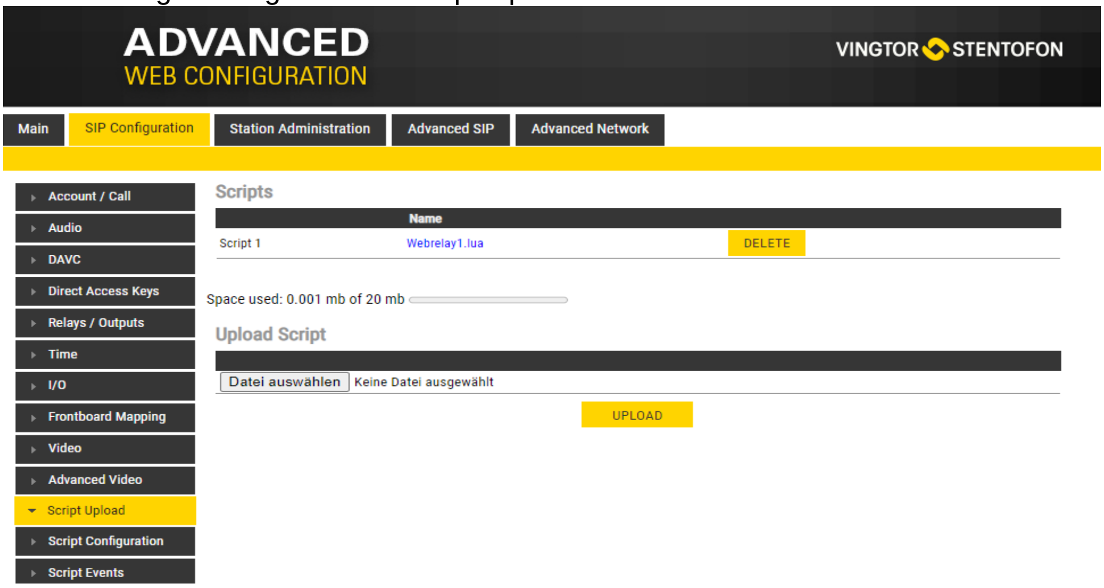
New menu items such as I/O, Script under SIP Configuration and a new tab called Advanced SIP will now appear:

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:

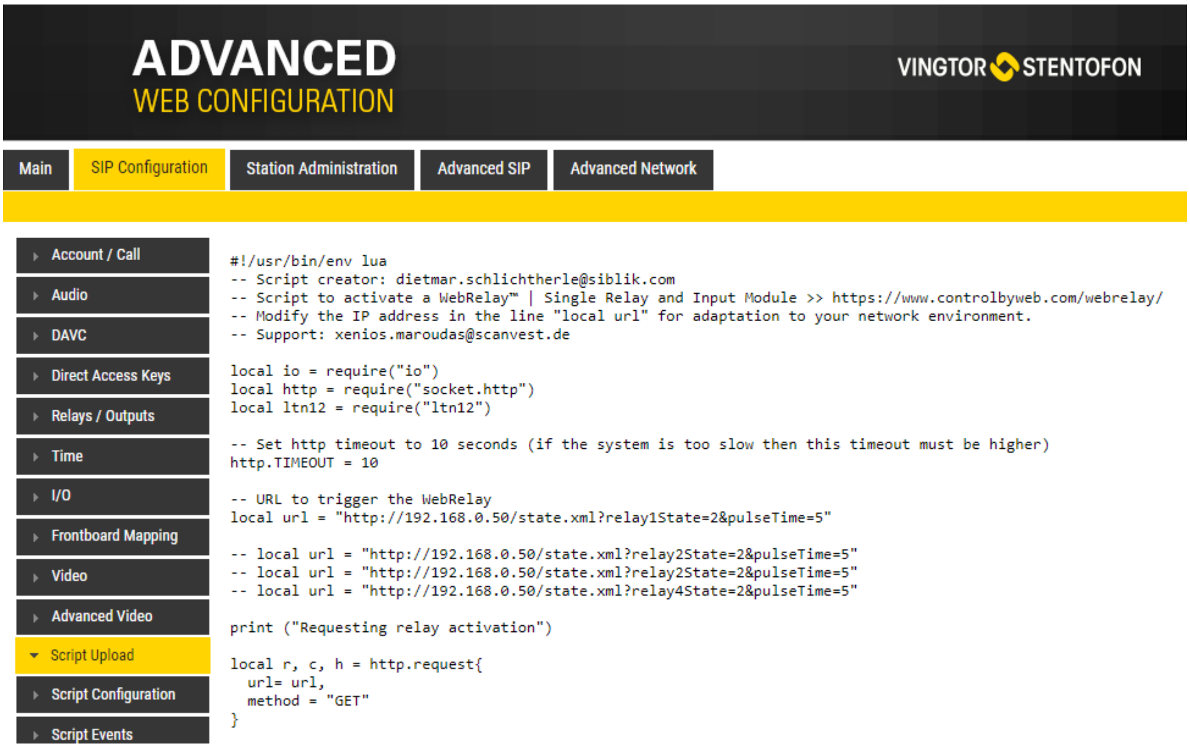
- Script Upload
- Script
- Script Events

Script Upload

- Select Edge Configuration > Script Upload



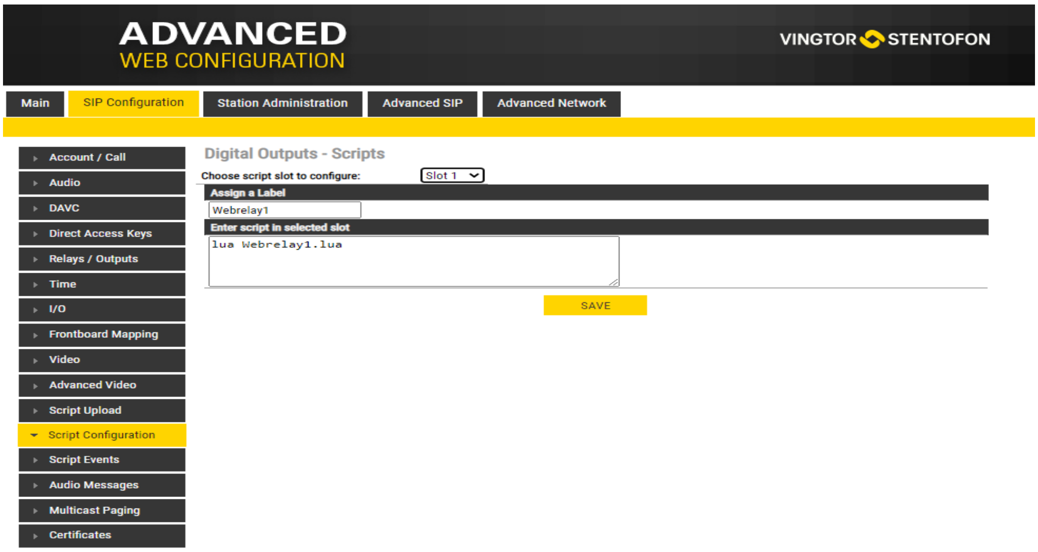
- Click UPLOAD to upload the desired script



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



Script Events

• Select Edge Configuration > Script Events

For more information on Script settings, see wiki.zenitel.com/wiki/Virtual I/O_(SIP)

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
- 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