

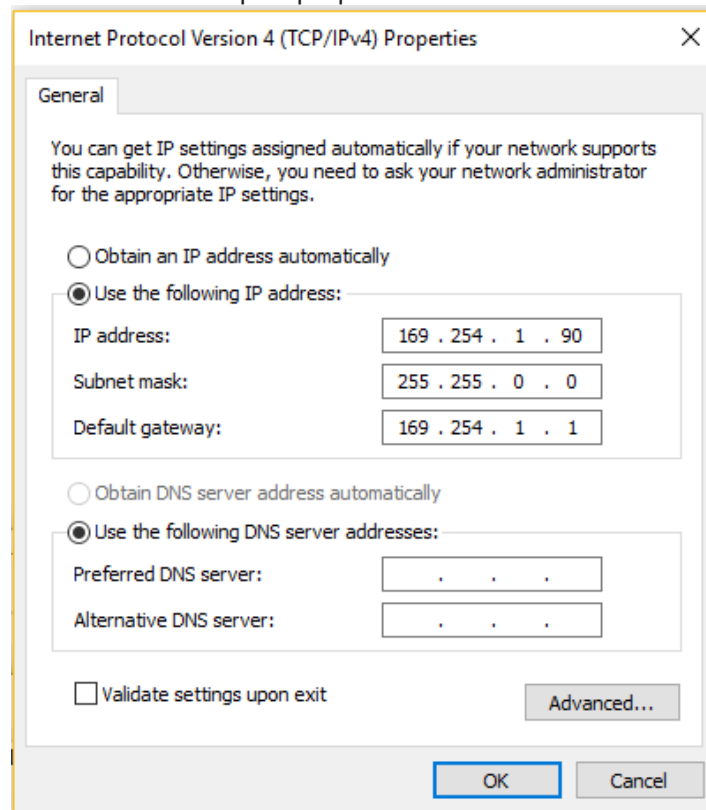
# Account Settings for SIP Mode

Below are settings required to use a TCIS-2 (or TCIV-2) as a SIP device. **Note that by using a 3<sup>rd</sup>-party SIP device with the TCIS-2 you must purchase the ZT1015 SIP-Client Extension license and apply it to the TCIS-2.** Otherwise your SIP device will fail after 60 days.

## Set a static IP Address on your Windows 10 computer

To access the web interface of the station, your computer must be in the same subnet range as the station.

1. Click **Start**, and type **Control** in the search field, then select **Control Panel**.
2. Click on **Network and Internet**.
3. Click on **Network and Sharing Center**.
4. On the left pane, click the **Change adapter settings** link.
5. Right-click the network adapter and select **Properties**.
6. Select the **Internet Protocol Version 4 (TCP/IPv4)** option.
7. Click the **Properties** button.
8. Select the **Use the following IP address** option
9. Set the **IP address** to **169.254.1.90**
10. Set the **Subnet mask** to **255.255.0.0**
11. Set the **Default gateway** to **169.254.1.1**
12. Click **OK**.
13. Click **Close** again to close the network adapter properties.



Set the TCIS-2 to SIP mode under Main and Main Settings and press the Save button.

Main Edge Controller

Information

Main Settings

Recovery

Help

### 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

### Product Model And Accessory

Model:

### IP Settings

Set the below settings to match your SIP Server requirements.

Main SIP Configuration Station Administration

Account / Call

Audio

Direct Access Keys

Relays / Outputs

Time

Audio Messages

Certificates

### Account Settings

Description	Configuration
Name:	<input type="text" value="Scale 1"/>
Number (SIP ID):	<input type="text" value="300"/>
Server Domain (SIP):	<input type="text" value="169.254.1.100"/>
Backup Domain (SIP):	<input type="text"/>
Backup Domain 2 (SIP):	<input type="text"/>
Registration Method:	<input type="text" value="Parallel"/>
Authentication User Name:	<input type="text" value="300"/>
Authentication Password:	<input type="text"/>
Register Interval:	<input type="text" value="100"/> (min. 30 seconds)
Register Failure Interval:	<input type="text" value="60"/> (min. 5 seconds)
Outbound Proxy [optional]:	<input type="text"/> Port: <input type="text" value="5060"/>
Outbound Backup Proxy [optional]:	<input type="text"/> Port: <input type="text" value="5060"/>
Outbound Backup Proxy 2 [optional]:	<input type="text"/> Port: <input type="text" value="5060"/>
Outbound Transport:	<input type="text" value="UDP"/>
SIP Scheme:	<input type="text" value="sip"/> Using sips forces all proxies to also use TLS

- **Name:** The name that will be shown on the display, and used as Caller ID.
- **Number (SIP ID):** This is the phone number for the station.
- **Server Domain (SIP):** The IP address of your SIP server.
- **Backup Domain (SIP):** Not in use
- **Backup Domain 2 (SIP):** Not in use

- **Authentication User Name:** This is the authentication user name used to register the station to the SIP Server. This is required only if the server requires authentication and is normally the same as the Number (SIP ID).
- **Authentication Password:** This is the authentication user password used to register the station to the SIP Server. This is required only if the server requires authentication.
- **Register Interval:** This parameter specifies how often the station will register, and reregister at the server. This parameter will affect the time it takes to detect that a connection to the server is lost. Enter the values in number of seconds from 60 to 999999. The default interval is 600 seconds.
- **Register Failure Interval:** In case Primary and both Backup servers are failing with SIP INVITEs, the device will go into failure mode, and immediately start sending REGISTER requests to all SIP servers, and start sending INVITEs to the first server where registration succeeds.

## Put the TCIS-2 in Advanced Programming Mode

1. Use a Firefox or Microsoft Edge browser to log in to the TCIS-2 using the IP address you set.
2. Click Login and enter the default User name: **admin**
3. Enter the default Password: **alphaadmin**
4. **Under Main > Recovery** click on Advanced configuration mode and type in the password 1851
5. Put a checkmark in the box and click **Save**.

## Direct Access Key & Ringlist Settings

The TCIS-2 is equipped with one DAK key (**D**irect **A**ccess **K**ey, also known as speed-dial buttons). The DAK keys is used to call other phone numbers or to trigger a dedicated feature by a single button press.

### Direct Access Key Settings

To configure the DAK key, Inputs and Ring List functions, select **SIP Configuration > Direct Access Key Settings**.

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

DAK keys and Inputs have three states:

- **Idle:** There is no active call on the station.

- **Call:** There is an incoming, outgoing or active call on the station.
- **Hold:** Resumes a call (This state is available as from firmware 6.1.3.0 in [Advanced Configuration Mode](#))

## Idle state

From **IDLE** state the following options are available (note that the available options will depend on the station type):

- **Call To:** Enter the directory number to call when the DAK key/Input is pressed. Extended call options (parallel ringing, call escalation, etc.) are available if the call is routed via the Ring List. See [#Ringlist Settings](#) for details.
- **Forward Call:** Forward calls to the configured destination. Forwarding will be toggled On/Off every time the DAK key is pressed.
- **Group Call:** Enter the directory number of the [Group Call](#) to call when the DAK key/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)
  - This option is available on [Turbine stations](#) only.
- **Conversation Mode:** The station will toggle between Simplex mode (PTT - Press-To-Talk) and duplex mode (handsfree) every time the DAK key is pressed. The selected mode is permanently stored. This function works from idle as well as during a call.
- **Volume Control:** Change speaker volume by 1 point (+/- 4 dB). This function works from idle as well as during a call.
  - **Up:** increase speaker volume by 1 point
  - **Down:** decrease speaker volume by 1 point
  - This option is available on [Turbine stations](#) only.
- **Select Line:** "Select Line" is used on DAK keys to choose from Active or Incoming Calls. This option is available on the [IP Flush Master station](#) only. For details, see [IP Flush Master - Call Queuing](#).

## Call state

During an active **CALL** the following options are available (note that available options will depend on the station type):

- **Do Nothing:** No action executed if the DAK key/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 DAK key press or key release. Has an option to also answer [Edge Group call](#).
- **End Call:** Ends an active call. Action can be executed either on DAK key press or key release.
- **Answer Call:** Answers an incoming call. Has an option to also answer [Edge Group call](#).
- **Transfer Call:** The "Transfer Call" function can work in two modes:
  1. Transfer immediately on key press when destination call number is configured.
  2. 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 To" action.

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

- **Hold Call:** When a "Call To" button is configured to do "Hold Call" during the call, the button has different functionalities depending on the current state:
  1. *Incoming Call:* Accept a call
  2. *Established Call:* Put a call on hold (disable audio)
  3. *On Hold:* Resume a call (enable audio)

Notes:

- The "Hold Call" option is available on [Turbine stations](#) only.
- There can be multiple calls put On Hold, for every DAK key with "Call To" destination configured with an "Hold Call" action. There can be only one active or resumed call.
- If one or more DAKs are configured with "Call To" destination and "Hold Call" action, then incoming calls from some other destinations are only possible if there is an additional DAK key with Call To destination set to "\*", and a Hold action defined.
- A DAK key with "Hold Call" action cannot terminate a call. An additional DAK key needs to be configured to terminate a call.
- **Send DTMF:** Sends configured DTMF for key press and key release.
- **Send Text:** Sends configured text for key press and key release.
- **Push-To-Talk:** Configure key to behave like PTT key during active call.
- **Standby:** Standby will reduce the volume of the local speaker by a selected dB value, for the duration of the call or until the key is pressed again. The "Standby" feature is available on [Turbine stations](#) only.

## Hold state

This functionality is available from [Advanced Configuration Mode](#) only. In Basic Configuration it is hidden.

Functions are executed only when call is put on Hold.

- **Do Nothing:** No action executed when selected.
- **End Call:** Ends current active call.
- **Resume Call:** The call currently on hold will be resumed.