Turbine Compact IP Stations

Getting Started for SIP

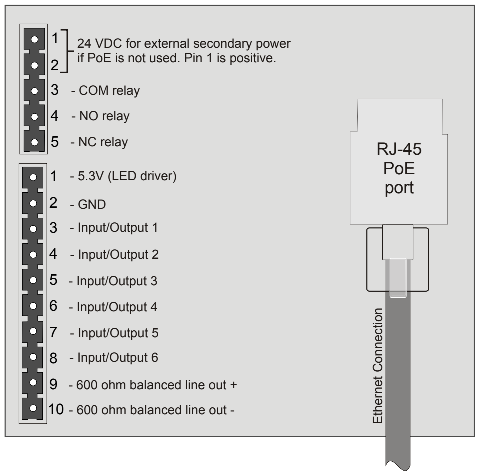
The information in this document pertains to the Turbine Compact IP Stations TCIS-1/TCIS-1-V, TCIS-2, TCIS-3, TCIS-4, TCIS-5, TCIS-6, TKIS-1.

Turbine TCIS-1/TCIS-1-V Station Keys & Functions

Turbine TCIS-6 Station Keys & Functions

1 Station Connections

1.1 External Connectors



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

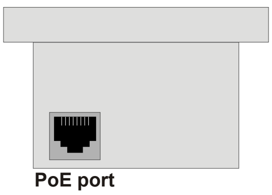
Ethernet/Power	10/100 Mbps Ethernet RJ-45 port for LAN (uplink) connection. Supports PoE (802.3af). Draws power from either spare line or signal line.
Secondary Power	24 VDC (16 – 48 V) secondary power is provided from an external adapter.
Relays	There is one Double Throw relay contact with 60W switching 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

1.2 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.

1.3 Network Connection



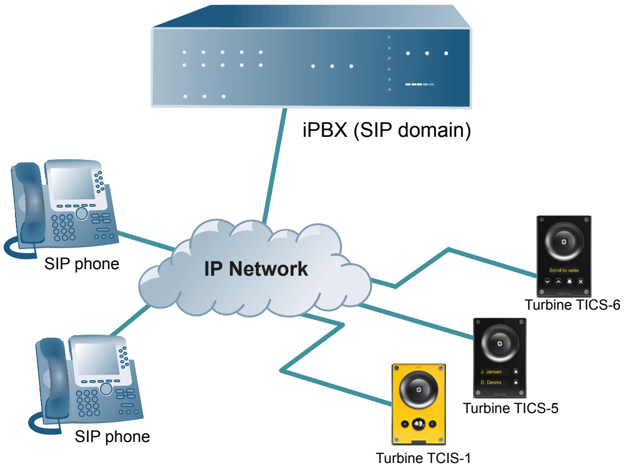
There is one RJ-45 port located on the Turbine station that is used as the PoE/LAN port.

1.4 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.

2 Station Configuration

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



Logging into the Turbine Station

The Turbine Station features an embedded web interface which allows users to log in via a standard web browser. To do this, your PC and the Turbine 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 station to the PoE switch

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

1. IP address obtained from a **DHCP server**
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 yet registered

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

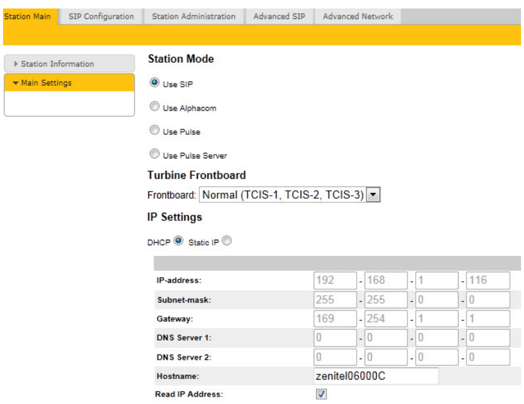
1. Open a web browser
2. In the browser’s address bar, type the station IP address and press the ENTER key
   - The station login page will be displayed.

To log into the station:

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

2.1 Station Main Settings

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



Station Mode

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

Turbine Frontboard

Depending on the type of Turbine Compact station, select one of the options from the drop-down box:

- **Kit**
- **Normal (TCIS-1, TCIS-2, TCIS-3)**
- **OLED Labels (TCIS-4, TCIS-5)**
- **Scrolling Station (TCIS-6)**

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** (optional - for network administration)
  - **DNS Server 2** (optional - for network administration)
  - **Hostname** (optional - for network administration)

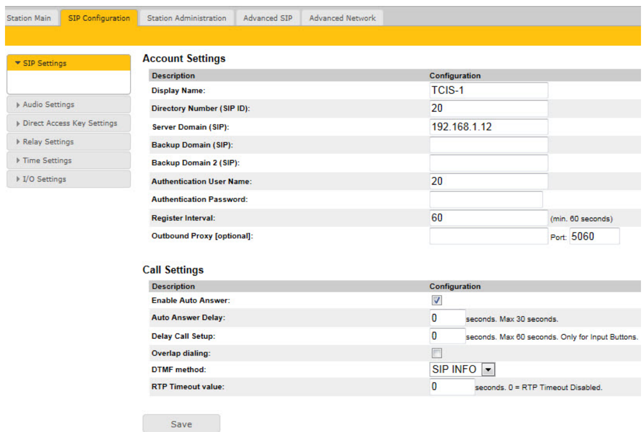
Read IP Address

- 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.

2.2 SIP Settings

- Select **SIP Configuration** > **SIP Settings** to access the page for configuring SIP parameters.



Account Settings

Display Name

- Enter name that will be shown on display at remote end.

Directory Number (SIP ID)

- This is the identification of the station in the SIP domain, i.e. the phone number for the station. This parameter is mandatory. Enter the SIP ID in integers according to the SIP account on the SIP domain server.

Server Domain (SIP)

- This parameter is mandatory and specifies the primary domain for the station and is the IP address for the SIP server (e.g. Asterisk or Cisco Call Manager). Enter the IP address in regular dot notation, e.g. 10.5.2.138.

Backup Domain (SIP) / Backup Domain 2 (SIP)

- This is the secondary (or fallback) and tertiary backup domain. If the station loses connection to the primary SIP domain, it will switch over to the secondary one. Enter the IP address in regular dot notation.

Authentication User Name

- Authentication user name used to register the station to the SIP server. Only required if the SIP server requires authentication and is normally the same as the SIP ID.

Authentication Password

- The authentication user password used to register the station to the SIP server. This is required only if the SIP server requires authentication

Register interval

- Specifies how often the station will register, and reregister in the SIP domain. This parameter will affect the time it takes to detect that a connection to a SIP domain is lost.
- Enter the values in number of seconds from 60 to 999999. The default interval is 600 seconds.

Outbound Proxy [optional]

- Enter the IP address of the outbound proxy server in regular dot notation, e.g. 10.5.2.100

Port

- Port number used for SIP on the outbound proxy server. The default port number is 5060.

<u>**Call Settings**</u>

Enable Auto Answer

- This is not required. Enables automatic answer after a set number of seconds.
- Check the box to enable this function and enter the delay in seconds in the field for **Auto Answer Delay**. The default delay setting is 0 and the maximum is 30 seconds.

Delay Call Setup

- This only applies to input buttons and DAKs. The default delay setting is 0 and the maximum is 60 seconds.

Overlap dialing

- This will lead to the phone starting to dial each time a digit is entered and the SIP proxy replying with ‘Number incomplete’ until such time as the number has been entered and the call can be initiated successfully without the enter key having to be pressed.

DTMF method

- Choose between SIP INFO or RFC 2833 to select DTMF signalling method.

RTP Timeout value

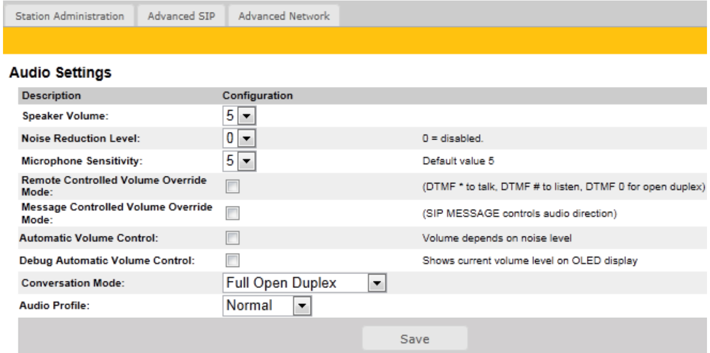
- This cancels a call if the station does not receive RTP packets from the remote party. Enter values in the range 0-9999 seconds. The default setting is 0 which means RTP timeout is disabled.

- After entering all the desired values, click **Save** and then click **Reboot** to enable the SIP settings.

2.3 Audio Settings

To configure audio settings:

- Select **SIP Configuration** > **Audio Settings** from the menu



Speaker Volume

- Select volume level in range 0-7. Default setting is 5

Noise Reduction Level

- The higher the noise reduction level the more deterioration there is in audio quality.
- Default setting is 0 (i.e. the function is disabled)

Microphone Sensitivity

- Select sensitivity level in range 0-7. Default setting is 5

Remote Controlled Volume Override Mode

- This acts as simplex mode. This feature is activated after the first DTMF * or # is received from the remote station. Send DTMF * to talk and DTMF # to listen.

Message Controlled Volume Override Mode

Check the box to enable the following messages:

- SIP MESSAGE “Audio_receive_only”: Turns the microphone off and loudspeaker on
- SIP MESSAGE “Audio_send_only”: Turns microphone on and loudspeaker off
- SIP MESSAGE “Audio_send_receive”: Turns both microphone and loudspeaker on.

Automatic Volume Control

- Check box to enable automatic volume control that is adjusted according to the noise level.

Debug Automatic Volume Control

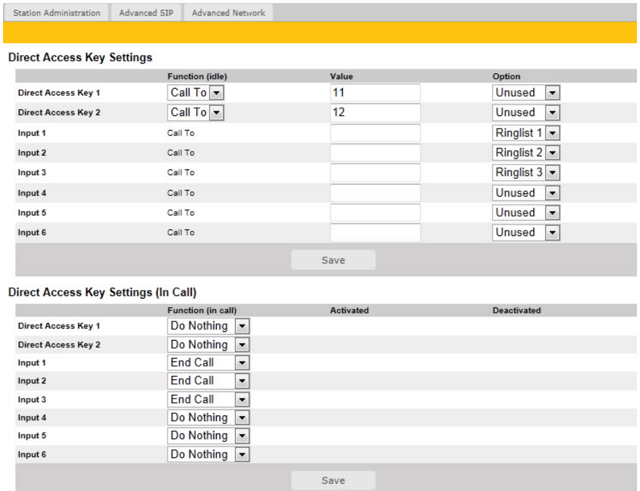
- Check box to show current volume level on OLED display.

<u>**Conversation Mode**</u>

- **Full Open Duplex**: Normal mode with echo cancellation
- **Robust Duplex**: Option used when open duplex fails due to excessive speaker loudness, microphone overload or very high nonlinear distortions.
- **Half Duplex Switching**: Switches speech direction depending on who speaks the loudest
- **Push-To-Talk**: Half-duplex communication. Initially the microphone is shut off. Push the M-button to open the microphone, and release to listen. (TCIS-1 station only)
- **Open**: Full Open Duplex without echo cancellation

2.4 Direct Access Key Settings

- Select **SIP Configuration** > **Direct Access Key Settings** to access the page for configuring DAKs.



__Direct Access Key Settings__

Direct Access Key 1 - Direct Access Key 2

- Enter the number to call in the **Value** field.

Input Buttons 1 to 3

- These are the SIP IDs for the extensions to be called when call buttons no. 1 to 3 are pressed.

__Direct Access Key Settings (In Call)__

- Input buttons 1-6 for direct calls while in conversation.
- Options: End Call, Do Nothing, Send Text, Send DTMF

2.5 Relay Settings

- Select **SIP Configuration** > **Relay Settings** to access the page for configuring relays.

__Relay Settings__

- Select Relay 1, Ouput 1, Ouput 2, or Ouput 3

__Timed Relay Duration__

This parameter determines how long the relay should stay ON in seconds.

3 LEDs on Station 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

4 Restoring Default Settings

4.1 Reset to Default with Activated DHCP

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**.
4. Press the **call button** to make the station speak its IP address.

__Default values__

- Station IP address: (determined by DHCP server)
- Username: **admin**
- Password: **alphaadmin**

4.2 Reset to Default with Static IP

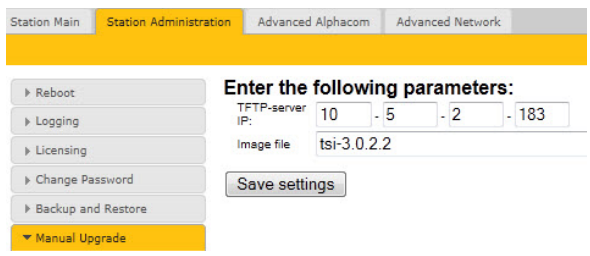
1. While **pressing any button**, power up the station by connecting to a PoE switch.
2. Hold the button until the station audio starts counting, and release the button on **count 1**.
3. Press and hold the button on **count 3** and release on **count 0**.
4. Press the **call button** to make the station speak its IP address.

__Default values__

- Station IP address: **169.254.1.100**
- Username: **admin**
- Password: **alphaadmin**

5 Station Software Upgrade

1. Start the TFTP server program and click **Browse** to select the folder containing the software image files
2. Log on to the IP station web interface
3. Select **Station Administration** > **Manual Upgrade**



4. Enter the IP address of the **TFTP server** (your PC's IP address)
5. Enter the prefix (e.g. **tsi-3.x.x.x)** to the software image files in the **Image file** field
6. Click **Save settings** to store the data
   - The station will now try to contact the TFTP server. If the response is *TFTP_CONN_OK* the settings are saved, and the **Upgrade** button will appear.
7. Click the **Upgrade** button to upgrade the software on the IP station.
   - The upgrade procedure takes about 3 minutes.

support@stentofon.com
support@vingtor.com