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

- Secure Login (HTTPS)

 Unsecure Login (HTTPS)
- 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.

tion Main	SIP Configuration	Station Administration	Advanced SIP	Advar	iced N	letwork				
Station Ir	formation	Station Mode								
Main Sett	ings	Use SIP								
		O Use Alphacom								
		O Use Pulse								
		O Use Pulse Server								
		Turbine Frontboard	i i							
		Frontboard: Normal (TCIS-1, TCIS-	2, TCI	S-3)	-				
		IP Settings								
		DHCP Static IP								
		IP-address:		192	-	168	-	1	-	116
		Subnet-mask:		255	1.	255	٦.	0	٦.	0
		Gateway:		169		254].	1	٦.	1
		DNS Server 1:		0	-	0	-	0		0
		DNS Server 2:		0		0]-	0	•	0
		Hostname:		zenite	1060	00C				
		Read IP Address:		1						

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.

Station Main	SIP Configuration	Station Administration	Advanced SIP	Advanced Network				
▼ SIP Settin	05	Account Settings						
		Description			Config	uration		
		Display Name:			TCIS	-1		
Audio Sett	tings	Directory Number (SIP	ID):		20			
Direct Acc	ess Key Settings	Server Domain (SIP):			192.1	168.1.12		
▶ Relay Sett	lings	Backup Domain (SIP):						
▶ Time Setti	ings	Backup Domain 2 (SIP):					
I/O Setting	gs	Authentication User N	ame:		20			
		Authentication Passw	ord:					
		Register Interval:			60		(min. 60 seconds)	
		Outbound Proxy [optic	onal]:				Port: 5060	
		Call Settings						
		Description			Config	uration		
		Enable Auto Answer:			V			
		Auto Answer Delay:			0	seconds. Max 30 se	conds.	
		Delay Call Setup:			0	seconds. Max 60 se	conds. Only for Input E	Buttons.
		Overlap dialing:						
		DTMF method:			SIP I	NFO 💌		
		RTP Timeout value:			0	seconds. 0 = RT	P Timeout Disabled.	

Account Settings

Display Name

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

Directory Number (SIP ID)

Save

- 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.

Call Settings

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

Station Administration	Advanced SIP	Advanced Network	
Audio Settings			
Description		Configuration	
Speaker Volume:		5 💌	
Noise Reduction Level:		0 💌	0 = disabled.
Microphone Sensitivity:		5 💌	Default value 5
Remote Controlled Volu Mode:	ime Override		(DTMF * to talk, DTMF # to listen, DTMF 0 for open duplex)
Message Controlled Vol Mode:	ume Override		(SIP MESSAGE controls audio direction)
Automatic Volume Cont	rol:		Volume depends on noise level
Debug Automatic Volum	e Control:		Shows current volume level on OLED display
Conversation Mode:		Full Open Duplex	
Audio Profile:		Normal 💌	
			Save

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.

Conversation Mode

- 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. on Administration Advanced SIP Advanced Network

	Function (idle)	Value	Option
Direct Access Key 1	Call To 💌	11	Unused 💌
Direct Access Key 2	Call To 💌	12	Unused 💌
nput 1	Call To		Ringlist 1 💌
nput 2	Call To		Ringlist 2
nput 3	Call To		Ringlist 3 💌
Input 4	Call To		Unused 💌
Input 5	Call To		Unused 💌
	0.07		Upused -
Input 6	Call 10	Save	Ullused
nput 6 rect Access Key Set	tings (In Call)	Save	
rect Access Key Set	tings (In Call) Function (in call)	Save	Deactivated
rect Access Key Set Direct Access Key 1 Direct Access Key 2	tings (In Call) Function (in call) Do Nothing v	Save	Deactivated
nput 6 rect Access Key Set Direct Access Key 1 Direct Access Key 2 nput 1	tings (In Call) Function (in call) Do Nothing • End Call •	Save	Deactivated
nput 6 rect Access Key Set Direct Access Key 1 Direct Access Key 2 nput 1 nput 2	tings (In Call) Function (in call) Do Nothing • End Call •	Save Activated	Deactivated
rect Access Key Set Virect Access Key 1 Virect Access Key 2 Nove 1 Nove 1 Nove 1 Nove 1 Nove 1 Nove 1	tings (In Call) Function (In call) Do Nothing w End Call w End Call w	Save Activated	Deactivated
ect Access Key Set irect Access Key 1 irect Access Key 2 iput 1 iput 2 iput 3 iput 4	Lan 10 Ton Nothing * Do Nothing * End Call * End Call * End Call *	Save Activated	Deactivated
rect Access Key Set Direct Access Key 1 Hirect Access Key 2 uput 1 aput 2 aput 3 aput 4	tings (In Call) Function (in call) Do Nothing w End Call w End Call w Do Nothing w Do Nothing w	Save	Deactivated

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.

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

Restoring Default Settings

4.1 Reset to Default with Activated DHCP



- While pressing any button, power up the station by connecting to a PoE switch.
- Hold the button until the station audio starts 2. counting, and release the button on count 1.
- 3. Press and hold the button on count 5 and release on count 0.
- Press the call button to make the station 4 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



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

Default values

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

Station Software Upgrade 5

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

Reboot	Enter the	follo	wing p	aramet	ers:
▶ Logging	TFTP-server IP:	10	- 5	- 2	- 183
Licensing	Image file	tsi-3.	0.2.2		
Change Password	Save setti	ngs			
Backup and Restore					

- 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.



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