

# TECHNICAL MANUAL Turbine IP Intercoms

TCIS / TCIV+ / TEIV+/TFIE / TFIX / TMIS / ECPIR-3P

A100K11194

zenitel.com

# About this document

This document describes the setup procedure and configuration of the various models of the Turbine Compact and Extended IP Intercom series. The configuration procedures described here are for the general functions of the Turbine range. For information on more advanced features and settings, see <u>Zenitel Wiki</u>.

Firmware Version: VSF-Turbine 5.1 (software package vsft-5.1.3.0 or later)

# **Related documents**

Doc. Number	Mounting Guides	
A100K11254	Turbine Compact IP Stations Mounting Guide	
A100K12125	TMIV-1 Turbine Mini Mounting Guide	
A100K12140	Exigo-Turbine Ex Mounting Manual	
A100K11575	Turbine Industrial Intercoms Mounting Manual	
A100K11589	Turbine Mini Mounting Guide	
A100K12201	TEIV+ Mounting and Installation Manual	
	Connection Guides	
A100K11526	TAX-3 Handset for Turbine Ex Connection Guide	
A100K11567	TA-23 Handset for Turbine Industrial Connection Guide	
A100K11528	EMMAX-1H Compact Microphone for Turbine Ex Connection Guide	
A100K11566	EMMAI-2H Compact Microphone for Turbine Industrial Connection Guide	
A100K11525	TAX-2b Plugbox for Turbine Ex Connection Guide	
A100K11564	TA-22b Plugbox for Turbine Industrial Connection Guide	
	General Manuals	
A100K12133	Exigo Ex Access Panels Turbine Ex Intercoms Installation & Maintenance Procedures	
A100K10805	AlphaCom XE Installation & Configuration	
A100K12031	IC-EDGE Quick Configuration	

# **Publication log**

Rev.	Date	Author	Status
1.0	21.9.2012	HKL	Published
2.0	24.9.2014	HKL	Turbine Extended intercoms
2.1	5.2.2015	HKL	TFIX-4 intercom
2.5	3.11.2017	HKL	TFIE-6, VS-IS 4.7
3.0	June 2020	HKL	IC-EDGE
3.1	22.2.2021	HKL	TCIV+, Zenitel branding
4.0	April 2022	ES	New Template, TFIX-X-V2, TMIS-4, TMIV-1
4.1	Sept 23	ES	TEIV+
4.2	April 24	ES	Removed Mobile app

# Contents

1 Turbine Com	pact Intercoms	6
1.1	Turbine Compact Intercoms	6
1.1.1	Turbine Compact Station Keys & Functions	8
1.1.2	Turbine Compact Kits	8
1.2	Connectors on Turbine Compact Intercom	9
1.2.1	Power Supply	10
1.2.2	Network Connection	10
1.2.3	Input/Output Connectors	11
2 Turbine Exter	nded Intercoms	11
2.1	Turbine Extended Industrial & Ex Intercoms	11
2.1.1	Turbine Extended Station Keys & Functions	12
2.1.2	ECPIR-3P Indoor Intercom - Keys & Functions	13
2.1.3	TKIE-2 Turbine Extended Kit	13
2.2	Connectors on Turbine Industrial	14
2.2.1	Power Supply	14
2.2.2	Network Connection	14
2.2.3	Input/Output Connectors	15
2.2.4	Connect Accessories	15
2.2.4.1	TA-23 Handset	15
2.2.4.2	AK5850HS Headset TA-22b Plugbox	16
2.2.4.3	EMMAI-2H Handheld Compact Microphone	17
2.3	Connectors on Turbine Ex	17
2.3.1	Connect Accessories	18
2.3.1.1	TAX-3 Handset	18
2.3.1.2	AK5850HS Headset TAX-2b Plugbox	19
2.3.1.3	EMMAX-1H Handheld Compact Microphone	20
2.4	Connect ExternalLoadspeaker	20
2.5	Connect Signal Beacon	21
2.6	EBMDR-8 Expansion Module for ECPIR-3P	21
2.6.1	Connect EBMDR-8 to ECPIR-3P	22
3 Starting Up th	ne Station	22
4 ICX-AlphaCo	m Configuration	23
4.1	Logging into the Station	23
4.2	MainSettings	23
4.3	Audio Settings	25
4.4	I/O Settings	25
4.5	Address Book	28
4.6	OLED Labels	29
4.7	Sound Detection	30
4.8	Time Settings	30

5 IC-EDGE Co	onfiguration	31
5.1	Logging into the Station	31
5.2	Main Settings	32
5.3	Connect other Intercom Stations	33
5.4	Configure Edge Controller Directory	33
5.5	Verify System Setup	34
5.6	Configure Call and Audio Settings	34
5.7	Modify IC-EDGE Device Profiles	35
5.8	Group Call	36
5.9	Configure 3rd-party SIP Devices	37
5.9.1	Install Licence	37
5.9.2	Create SIP Account	38
5.9.3	Configure SIP Phone	38
5.9.4	Verify Operation	38
5.10	Advanced Configuration Mode	39
5.11	Direct Access Key & Ringlist Settings	40
5.11.1	Ringlist Settings	42
5.11.1.1	Forward Unattached Call	43
5.11.1.2	Forward Unattached Call with Loopback	44
5.11.1.3	Parallel Ringing - ForwardCall if Unattended	45
5.12	Script Settings	45
5.12.1	Script Upload	46
5.12.2	Script Configuration	46
5.12.3	Script Events	47
5.13	Audio Messaging	47
5.13.1	Updating Audio Files	47
5.13.2	Play Message on DTMF Event	48
5.13.3	Play Message on Call Event	48
5.13.4	Play Message on Relay Event	49
5.13.5	Routing of Audio Call	49

6 SIP Configuration		
6.1	Logging into the Station	
6.2	MainSettings	
6.3	Account/Call Settings	
6.4	Audio Settings	
6.5	Direct Settings	
6.6	Relay/Output Settings	
6.7	Time Settings	
6.8	Audio Messages	
6.9	Advanced Configuration Mode	
6.10	I/O Settings	
7 Common Adva	nced Network Settings	
7.1	SNMP Settings	
7.2	Network Access Control	
7.3	Firewall Settings	
8 Station Softwa	re Upgrade	
8.1	Prerequisites	
8.2	Upgrade via Station Web Interface	
9 Station Indicati	ion LEDs	
9.1	LEDs on Front Plate - Compact Stations	
9.1.1	ECPIR-3P, Industrial and EX Stations	
9.2	Status LEDs on PCB	
9.3	Ethernet Activity & Speed LEDs	
10 Restoring Fac	tory Default	
10.1	Reset to Factory Default Settings with Ad	
10.2	Reset to Factory Default Settings with Sta	
A: Compact Board	d Connectors	
A.1	PCB - Front	
A.2	Input Connectors	
A.3	Output Connectors +1Relay	
A.4	Output Connectors + MRBD Relay Board	
A.5	PCB-Rear	
A.6	Front Board - Front	
A.7	Front Board - Rear	
B: Extended Boar	d Connectors	
B.1	PCB-Front	
B.2	Input Connectors	

	50
	50
	50
	52
	54
	55
	56
	57
	57
	59
	59
	60
	60
	62
	64
	65
	65
	65
	66
	66
	67
	67
	67
	68
lvanced DHCP	68
atic IP	68
	69
	69
	70
	70
d	71
	71
	72
	72
	73
	73
	73

# **1** Turbine Compact Intercoms

# **1.1 Turbine Compact Intercoms**

Model	Product Description	Item No.	Accessories	
TMIS-1	Turbine Mini IP Intercom	1008116010		
TMIS-2	Turbine Mini IP Intercom	1008116020	TA-2 Flush Mount 2-Gang Double-Depth Back Box - 1008140020	
TMIS-4	Turbine Mini IP Intercom	1008116040	TA-3 Deep Flush mount Backbox - 1008140030	
TMIV-1+	Turbine Mini Video Intercom	1008117010		
TCIS-1	Turbine Compact IP Intercom	1008111010		
TCIS-2 TCIV-2+	Turbine Compact IP Intercom Turbine Compact IP Video Intercom	1008111020 1008315020		
TCIS-3 TCIV-3+	Turbine Compact IP Intercom Turbine Compact IP Video Intercom	1008111030 1008315030		
TCIS-4	Turbine Compact IP Intercom	mpact IP Intercom1008111040TA-1 Surface Mount Back Box - 1008140010mpact IP Intercom1008111050TA-2 Flush Mount 2-Gang Double-Depth Back Box - 10mpact IP Video Intercom1008315050TA-5 Flush Mount Bracket - 1008140050	TA-1 Surface Mount Back Box - 1008140010	
TCIS-5 TCIV-5+	Turbine Compact IP Intercom Turbine Compact IP Video Intercom		TA-5 Flush Mount Bracket - 1008140050	
TCIS-6 TCIV-6+	Turbine Compact IP Intercom Turbine Compact IP Video Intercom	1008111060 1008315060		
TCIS-C1	Turbine Compact IP Intercom	1008111901		
TCIA-2	Turbine Compact IP Intercom	1008113020		
TEIV-1+	Turbine Extended IP Video Intercom with Keypad and Display	1008317110	TA-33 Turbine Extended On-wall Backbox - 1008140330	
TEIV-4+	Turbine Extended IP Video with RFID card reader mount	1008318240	TA-34 Turbine Extended Flushmount Backbox-1008140340	

## TMIS-1/TMIV-1+ Turbine Mini

TMIS-2 Turbine Mini

# TMIS-4 Turbine Mini

Thermoplastic Frontplate with Single Call Button and PMOLED Display



S • (-) • STENTOFON

Stainless Steel Frontplate with Single Call Button

## TCIS-1

Thermoplastic Frontplate with Single Call Button plus Push-To-Talk & Cancel



TCIS-2/TCIV-2+/TCIA-2 Stainless Steel Frontplate with Single Call Button



# Thermoplastic Frontplate with Single Call Button

TCIS-3/TCIV-3+

Thermoplastic Frontplate with Two Call Buttons and PMOLED Display

6

 $\odot$ 



**Turbine Stations Technical Manual** 

## TCIS-4

Thermoplastic Frontplate with Single Call Button and PMOLED Display

# Thermoplastic Frontplate with Two Call Buttons and PMOLED Display

TCIS-5/TCIV-5+





# TEIV-1+

Turbine Extended IP Video Intercom with Keypad and Display



## TCIS-6/TCIV-6+



Thermoplastic Frontplate with Call & Scrolling Buttons and PMOLED Display



## TEIV-4+

Turbine Extended IP Video with RFID card reader mount



# 1.1.1 Turbine Compact Station Keys & Functions



# Speaker 0 PMOLED Display Scroll to select **x**+ Scroll Down - Cancel Call Button Scroll Up 63 Microphone

# **1.1.2 Turbine Compact Kit**

## TKIV+

Turbine Video Kit VoIP Intercom Module



# TKIS-2

Turbine Kit VoIP Intercom Module



# **1.2 Connectors on Turbine Compact Intercom**

This Turbine Compact Board is used by the following stations: TCIS-1, TCIS-2, TCIS-3, TCIS-4, TCIS-5, TCIS-6, TKIS-2, TMIS-1, TMIS-4.



Note! The TCIA-2 has a different set of connectors.

See Appendix A: Compact Board Connectors for further information.



Item	Description	Comment
1	Relay	Double Throw relay contact with 60W switch 250VAC/220VDC, 2A, 60W.
2	I/O interface	6 general purpose I/Os. Each I/O can be conf
3	Line Output	Balanced 600 ohm audio line out. Suitable for
4	Power Input	Optional 24VDC power input from external po
5	LAN Port	10/100/1000Mbps RJ-45 port connecting to



5

ing power. COM, NO, NC contacts are provided. Max:

- figured either as an input or 12mA LED driver output
- r e.g. external induction loop amplifier
- ower supply
- Ethernet. PoE/PoE+ is supported.

This Turbine Compact Board is used by the following video stations: TMIV-1+,TCIV-2+, TCIV-3+, TCIV-5+, TCIV-6+, TEIV-1+ and TEIV-4+



Item	Description	Comment
1	Relay	Double Throw relay contact with 60W switching power. COM, NO, NC contacts are provided. Max: 250VAC/220VDC, 2A, 60W.
2	I/O interface	6 general purpose I/Os Each I/O can be configured as either input, output, or LED driver.
3	Line Output	A balanced 600 ohm audio line out with induction loop signal.
4	Speaker Input	Connect an external speaker rated for 8 Ohm
5	Microphone input	Electret mic input
6	Power Input	Optional 24VDC power input from external power supply
7	LAN Port	10/100/1000Mbps RJ-45 port connecting to Ethernet. PoE/PoE+ is supported.
8	SD-card Slot	
9	USB-C Port	Used during production process and for recovery

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 V DC (16 – 48 V) secondary power is provided from an external adapter.
Input/Output	6 general purpose I/Os are available. Each I/O can be configured as either button input or LED driver.
Relays	There is one Double Throw relay contact with 60W switching power. COM, NO, NC contacts are provided Max: 250 V AC/220 V DC, 2A, 60W.
Audio Line Out	A balanced 600 ohm audio line out with induction loop signal

# 1.2.1 Power Supply

The Turbine Compact 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 Compact Station can be connected to a 24 V DC local power supply.

# **1.2.2 Network Connection**

There is one RJ-45 port located on the Turbine Compact station that is used for PoE/LAN Ethernet connection. **Turbine Stations Technical Manual** 

# 1.2.3 Input/Output Connections

There are 6 I/O connection options for the Turbine Compact Station. These connections can be used to interface with external I/O devices. The connections are typically used as digital input (eg. form a closing contact), digital output (5V), or LED driving (20mA max) for connecting sensors, indicators or integration to other systems. See Appendix A for further information.

# 2 Turbine Extended Intercoms

# 2.1 Turbine Extended Industrial & Ex Intercoms

Model	Product Description	Item No.	Accessories	
TFIE-1	Turbine Extended Industrial IP Intercom	1008122010	TA-23 Handset for Industrial Stations - 1008140230	
TFIE-2	Turbine Extended Industrial IP Intercom	1008122020	EMMAI-2H Handheld Compact Mic for Industrial Stations - 1023533312	
TFIE-6	Turbine Extended Industrial IP Intercom	1008122060	TA-22D Cable and Flugbox for Ak3630HS Headset - 1006140225 TA-10 Connection Board with Relays for Turbine Industrial - 1008140100	
TFIX-1-V2	Turbine Extended Ex IP Intercom	1008123110	Headset for Ex & Industrial Stations - AK5850HS	
TFIX-2-V2	Turbine Extended Ex IP Intercom	1008123120	TAX-2b Cable and Plugbox for AK5850HS Headset - 1008150025	
TFIX-3-V2	Turbine Extended Ex IP Intercom	1008123130	EMMAX-1H Handheld Microphone for Ex Stations - 1023533511	
ECPIR-3P	Turbine Extended Indoor Intercom	1023200033	PAM1H Handheld Microphone - 1023533012 EBMDR-8 8-Button Expansion Module - 1023253008	
	* · · · · · · · · · · · · · · · · · · ·			

# TFIE-1

### Industrial Intercom with Full Keypad

# TFIE-2

# Industrial Intercom with 6 Programmable Keys





TFIX-1-V2 TurEX Intercom with Full Keypad

TFIX-2-V2 Turbine EX with 6 Programmable Keys





**Turbine Stations Technical Manual** 

## TFIE-6

Industrial Intercom with Scrolling Unit & PMOLED Display



## TFIX-3-V2

Turbine Ex Intercom Station, 3 buttons



# 2.1.1 Turbine Extended Station Keys & Functions



# 2.1.2 ECPIR-3P Indoor Intercom - Keys & Functions

## **ECPIR-3P**

Indoor Intercom with 3 Programmable Keys



PTT Key\_

# 2.1.3 TKIE-2 Turbine Extended Kit

# TKIE-2

Turbine Extended IP Kit





# 2.2 Connectors on Turbine Industrial

The Turbine Extended board is used in the following stations: TFIE-1, TFIE-2, TFIE-6, ECPIR-3P, TKIE-1.

See Appendix B: Extended Board Connectors for further information.



Note! All connections are made on the main board inside the enclosure of the Turbine Industrial station.

# 2.2.1 Power Supply

The Turbine Industrial 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 in not available, the Turbine Industrial Station can be powered from a 24-48 V DC local power supply.

Note! If used together with PoE the station can be powered by 24 V DC as a local power backup.

Note! If connected to two PoE-capable switches, only one of the ports will negotiate PoE; hence this is not a good solution for power redundancy (switch failure might lead to a reboot). Instead a "dumb" PoE injector (spare-pair power) should be used for the two Ethernet ports or a local 24 V DC local power backup.

# 2.2.2 Network Connection

There are two RJ-45 ports located on the Turbine Industrial station, either of which can be used for a single PoE/LAN Ethernet connection.

# 2.2.3 Input/Output Connections

There are 6 I/O connection options for the Turbine Industrial Station. These connections are used as digital input (eg. form a closing contact), digital output (5 V), or LED driving (20mA max.) for connecting sensors, indicators or integration to other systems. See Appendix B: Extended Board Connectors for further information.

# 2.2.4 Connect Accessories

Accessories such as the handset, plugbox for the headset, and handheld microphones are connected to the connection terminal block J8 on the main board inside the station enclosure.



# 2.2.4.1 TA-23 Handset

## TA-23

Industrial Handset for Turbine with PTT



Plug the IDC connectors on the Handset cable into the J8 terminal block according to the pin configuration below.



## 2.2.4.2 AK5850HS Headset TA-22b Plugbox

#### **AK5850HS**

Headset with boom microphone and curled cord

### TA-22b

 $Plugbox\,\&\,Cable$  (10 m.) for Headset with PTT



# 2.2.4.3 EMMAI-2H Handheld Compact Microphone

EMMAI-2H

Handheld Compact Industrial Microphone



## 2.3 Connectors on Turbine Ex





Note! All connections are made on the main board inside the enclosure of the Turbine Ex station.

Note! Turbine Ex stations must be powered by a two-wire Flowire link.

Procedures for connecting power supply, LAN network and Inputs/Outputs are described in the manual A100K12133 Ex Turbine Intercoms & Exigo Access Panels Installation & Maintenance Procedures.

For information on configuration of the Flowire interface see A100K11958 Flowire Converter - Configuration Manual.

**Turbine Stations Technical Manual** 

Plug the IDC connectors on the Compact Microphone cable into the J8 terminal block according to the pin configuration below.



rnal Mic +	14	GND External Push Button
rnal Mic -	15	GPI2ExternalPushButton
nal GND	16	GND External Push Button
lio Accessory Speaker +	17	External Loudspeaker +
lio Accessory Speaker -	18	External Loudspeaker -
lio Accessory Microphone Mic +	19	Signal GND
lio Accessory Microphone Mic -	20	Relay COM
lio Accessory Push To Talk (PTT)	21	Relay NO
lio Accessory Hook	22	Flowire PLC1+
lio Accessory GND PTT/Hook	23	Flowire PLC1 -
rnal Loudspeaker +	24	Flowire PLC2 +
rnal Loudspeaker -	25	Flowire PLC2 -
1 External Push Button		

# **2.3.1 Connect Accessories**

Accessories are connected to the top 16-screw connection terminal block in the Turbine Ex enclosure.



# 2.3.1.1 TAX-3 Handset

## TAX-3

Industrial Handset for Ex Intercom



Connect the wire ferrules on the Handset cable to the terminal block according to the pin configuration below.



# 2.3.1.2 AK5850HS Headset TAX-2b Plugbox

### **AK5850HS**

Headset with boom microphone and curled cord



Connect the wire ferrules on the Plugbox cable to the terminal block according to the pin configuration below.

Note ! The black (shield) wire is labeled MIC- and the black (main shield) wire is labeled GND.



# TAX-2b

Plugbox & Cable (10 m.) for Headset with PTT



8	CHASSIS GROUND (MAIN SHIELD)   BLACK
Q	SPEAKER +   GREEN
ę	SPEAKER -   YELLOW
Q	MIC+   PINK
	MIC - (SHIELD)   BLACK
9	PTT+   WHITE

PTT- | BROWN

# 2.3.1.3 EMMAX-1H Handheld Compact Microphone

## EMMAX-1H

Handheld Compact Industrial Microphone

Connect the wire ferrules on the Compact Microphone cable to the terminal block according to the pin configuration below.

MIC + | YELLOW

PTT+ | BLUE

💻 PTT- | RED

MIC - (SHIELD) | BLACK

3

5 6

7 8

9 10 V

 $\bigcirc$ 

N

 $\sim$ 

# 2.5 Connect External Signaling Device

A signal beacon can be connected to the Relay terminals located on the bottom connection terminal block in the Turbine Ex enclosure.



The signaling device needs an external power source, 24-48 VDC SELV, max 320mA

0.0.0

# 2.4 Connect External Loudpeaker

The loudspeaker connector is located on the bottom connection terminal block in the Turbine Ex enclosure.



Note! Only Ex-certified loudspeakers which comply with the minimum power ratings listed above may be used

Note! Maximum load of external loudspeaker: 25 W for maximum 30 minutes

Note! Maximum average load of external loudspeaker in service: 4 W

**Turbine Stations Technical Manual** 

# 2.6 EBMDR-8 Expansion Module for ECPIR-3P

The EBMDR-8 Expansion Module is a slave unit to the ECPIR-3P. The EBMDR-8 hence receives power from and communicates through its master, the ECPIR-3P.

Note! Up to four EBMDR-8 modules may be daisy-chained to one ECPIR-3P

20



### EBMDR-8

Expansion Module with 8 Buttons

0		EBMDR-8	0
1			E
1		:0	E
1		:0	E
		: 0	E
•	VINGTOR-STENTOFON		•

# 2.6.1 Connect EBMDR-8 to ECPIR-3P

- Using the small connection cable supplied, connect the EBMDR-8 expansion module to the ECPIR-3P or 1. preceding module as shown in the figure. If more than one EBMDR-8 is connected to the same ECPIR-3P, these are daisy-chained together.
- 2. Connect the white connector on the cable to the corresponding white connector on the ECPIR-3P or preceding EBMDR-8.
- 3. Connect the black connector on the cable to the corresponding black connector on the EBMDR-8.



# **3** Starting up the Station

The Turbine Station features an embedded web interface, which allows users to log in via a standard web browser.

To start up the station, your PC and the IP station have to be connected together via a PoE switch using network cables:

- 1. Connect the PC to the PoE switch
- Connect the PoE port on the IP station to the PoE switch 2.

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

- 1. An IP address is obtained from a DHCP server if there is one.
- 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 key, number keys or DAKs on the station when the station is not yet registered.

At commissioning, the IP station needs to be configured to enable it to be used as:

- Station subscribed to an ICX-500 ٠ Intelligent Communication Gateway or AlphaCom XE server
- SIP station •
- IC-EDGE station



ICX-500 Intelligent Communication Gateway

# 4 ICX-AlphaCom Configuration

The Turbine stations are connected to the ICX-AlphaCom platform comprising the ICX-500 Intelligent Communication Gateway or AlphaCom XE server/exchange lying at the heart of our security and communication system. The communication between the ICX-AlphaCom platform and the Turbine Stations utilize the CCoIP® protocols. The ICX-Alpha-Com platform includes all main service configurations for the IP stations and only a minimum configuration is needed to be carried out on the actual station.

For more information on ICX-AlphaCom configuration see wiki.zenitel.com/wiki/Turbine\_Configuration\_\_AlphaCom\_ mode

The following sections describe configuration procedures using the web interface of the station.

# 4.1 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 User name: admin
- Enter the default Password: alphaadmin З.

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

Note! The user interface and parameters displayed in the following sections are dependent on the Turbine station type (Compact, Industrial or Ex) selected.

# 4.2 Main Settings



**Turbine Stations Technical Manual** 

itel	WEB CONFIGURATION	0
	Secure Login (HTTPS)	
	Unsecure Login (HTTP)	

og on t	to the device	you will u	se as the	Edg	e Contr	oller. Y	ou can	do ell	configu	uration of y
umbe	r 101									
unice	101									
	10	. 0	- 5	ŀ	101	1				
	10 255	• 0 • 255	- 5 - 255	•	101					
	10 255 10	- 9 - 255 - 9	- 5 - 255 - 5	•	101 0 1					
	10 255 10 200	• 9 • 255 • 9 • 200	- 5 - 255 - 5 - 200	•	101 0 1 200					
	10 255 10 200 200	- 9 - 255 - 9 - 200 - 200	- 5 - 255 - 5 - 200 - 200	•	101 0 1 200 200					
	10 255 10 200 200 zenite	- 0 - 255 - 9 - 200 - 200 :1064508	- 5 - 255 - 5 - 200 - 200	•	101 0 1 200 200					
	10 255 10 200 200 zenite	- 0 - 255 - 9 - 200 - 200 - 200	- 5 - 255 - 5 - 200 - 200	•	101 0 1 200 200					
	10 255 10 200 200 200	- 0 - 255 - 9 - 200 - 200 - 200 - 200	- 5 - 255 - 5 - 200 - 200	-	101 0 1 200 200					

To access the page for configuring station mode and IP parameters:

Select Main > Main Settings •

#### Mode:

Select the ICX-AlphaCom radio-button

#### Product Model and Accessory:

The options presented will depend on the Turbine model (Compact, Industrial, Ex).

- Select one of the options from the Model drop-down box:
- Model: (for Compact)
  - Kit(TKIS-2)
  - TKIE-1 (Kit) • Normal (TCIS-1, TCIS-2, • TFIE-1
  - TCIS-3)OLED
    - TFIE-2 TFIE-6
  - Labels(TCIS-4,TCIS-5)
  - ScrollingStation(TCIS-6)
  - Mini (TMIS-1, TMIS-2, TMIS-4)
- Model: (for Extended)
  - TFIX-3-V2

  - ECPIR-3P

- Handheld
- Headset
- Headset w/ Auto

#### **Registration Settings:**

#### ICX-AlphaCom IP-Address

Enter the IP address of the ICX-500 or AlphaCom server in which the IP station is to be a subscriber in the • field.

#### Number

Enter the directory number of the station in the field. - If a directory number is not entered, the station will register • with its MAC address. The MAC address is found on the Station Information page and needs to be entered into AlphaPro.

### **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)x
  - DNS Server 2 (option for network administration)
  - Hostname (option for network administration)
- Disable Reset to Factory default settings using frontboard and I/O Check the box to disable factory reset using frontboard 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.
- Ethernet Speed 10 Mbits/s Check the box if the switch is configured to 10 Mbit/s. Default Ethernet speed is 100 Mbit/s.
- 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 •

# 4.3 Audio Settings

Main Stati

 Audio DAVC Setti

> Offline Rela I/O Setting

> Sound Dete

**RTSP Setti** 

Time Settin

Audio Me

Select Advanced ICX-AlphaCom > Audio

Au	dio Settings	
Des	scription	Configuration
ys Vol	ume Override Level:	8 🔻
s Mic	ronhone Sensitivity:	2 •
	rophone ochonny.	2 1
ttings Dig	ital/Analog Mic Switch:	Always Digital
tion		
js Di	nital/Analog Mic Switch Pin	None V
	gital Analog Mic Ownen Pint	Hone Y
M	ute speaker while analog mic:	
iges Voi	ce Engine Mode:	Voice 🔻
Volu	ime Control Ch1:	0
Volu	me Control Ch2:	0
Con	versation Mode:	Full Open Duplex
Audi	io profile:	Normal
Hard	dware AGC:	Disabled 🔻
Har	dware DRC:	8
Aut	omatic Gain Control (AGC):	
Audi	io Out Source:	Voice Audio 🔻
Line	Out Source:	Audio Ch2
Adr	number of "Mixer Channels"	0 🔻
Aut	omatic Volume Control (AVC):	
AVC	C Debua:	0
AVC	Advanced	0
Far	End Audio Squelch:	Disabled <b>V</b>
Sq	uelch Threshold:	-60
Sq	uelch Activate Delay:	100

Select or set values for the parameters:

### Audio Settings:

- Volume Override Level
- Select the Volume Override Level in the range 0 to 8 from the drop-down box. The default setting is 7.
- Microphone Sensitivity • Select the sensitivity level in the range 0 to 7 from the drop-down box. The default setting is 4.

Note! If used as a local PA panel, a setting in the 1 to 3 range will reduce the chance of acoustic feedback (howling).

- Digital/Analog Mic Switch
  - Switches from Digital to Analog Mic based on settings.
  - Options: Always Digital (Default), Always Analog, Analog I/O Pin, Analog I/O Pin Call End
  - Mute speaker while analog mic
- Voice Engine Mode
  - Voice: Regular audio quality (default).
  - HD-Audio: Play audio in PCM L16/48kHz format. Use HD-Audio only on devices that handle high quality music streaming/playback, typically IP loudspeakers and Turbine kits used as PA interface.
- Volume Control Ch1/Ch2
  - Offset Gain (default routed to Amplifier/Speaker)
- Valid range: -62 to +24 dB **Turbine Stations Technical Manual**

- · Accessory: (for Extended & Ex)
  - Handset

- - (Normally Closed)

  - detect

# Microphone

- Model: (for Ex) • TFIX-1-V2 • TFIX-2-V2

#### Handset w/ Offhook Handset w/ Offhook

	Sets the volume during volume override. Volume and handset
	override happens during Emergency Group calls.
	Default value 4. 0 = very low sensitivity
	Default is Always Digital. Switch from Digital to Analog Mic based on settings.
•	Analog I/O Pin: When I/O Pin is triggered, Analog Mic is used, when not Digital Mic is used. Analog I/O Pin Call End: Analog Mic is switched On (Digital Off) by first I/O Pin trigger in call and used until call end.
	Input/Output Pin for Digital/Analog Mic switching. This doesn't affect other I/O pin configured functionality. NOTE: Pin must be configured as Input.
	Mute speaker while Analog Mic is used.
	Default is Voice. Use HD-Audio only on devices that handle high quality music streaming/playback NOTE: Changing this will reboot main application
	Audio Channel 1 Offset Gain (default routed to Amplifier/Speaker) Valid range: [-62+24] dB
	Audio Channel 2 Offset Gain. (default routed to Line Out Amplifier) Note! In Standard Turbine ch2 is same signal as ch1 Valid range: [-62+24] dB
<b>T</b>	
	Select Manual Control to enter own values
	Dynamic Range Compression
	Automatic Gain Control. If speech level and environmental noise are very unstable it may be turned on.
	Main Audio Out (Speaker) Sources
•	Line out can play audio either from VoIP signal or direct from microphone
	ICX-AlphaCom Multi-Conference Mixer function 🕕
	Volume depends on noise level
	Shows current volume level on OLED display
	Check to open advanced settings
	Audio Squelch on Far-End Signal (suppress audio on low signal levels)
	Threshold level for suppressing audio signal
	Valid range: [-920] dBm0



Note! This feature must be used with caution as incorrect settings may severely degrade the performance of the echo-cancelling algorithm.

- **Conversation Mode**, For this parameter, there are five options:
- 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. (Only applicable to stations with M-key)
- Open: Full Open Duplex without echo cancellation

#### Audio Profile

- Normal: Standard Acoustic Echo Cancelling (AEC) Profile with a few extra filters added to try to minimize the effects on the voice-signal when operating in a standard environment.
- Noisy Environment: Environment with high noise levels but maximum speaker level is not required.
- Very Noisy Environment: Extreme environment with very high noise level and maximum speaker level is required.

#### Hardware DRC, Enables Dynamic Range Compression

- Hardware AGCE nables Automatic Gain Control. Options are Disabled, Quit Area, Noisy Area, Manual Control. When selecting Manual Control, there will be more advanced AGC parameters available.
- Automatic Gain Control (AGC), If speech level and environmental noise is very unstable, it may be turned on.
  - AGC Speed: 0..3 selects different attack/release times. Lesser values mean faster attack and slower release time
  - AGC Volume: 0..7 corresponds to -20..0 dBm
- Audio Out Source, Main audio out (speaker) sources:
  - Voice Audio: Plays audio from VoIP signal
  - LineIn Idle: Plays audio from the Line IN input when the station is in idle. During a call, plays audio from the VoIP signal.
  - LineIn Idle + GPI0-6: If input 6 is activated while the station is in idle, plays audio from the Line IN input. Otherwise, plays audio from the VoIP signal.
- Audio Input Source (Turbine Extended only)
  - Normal Microphone (default): Select this option when using an electetret microphone as audio source. On the TKIE kit, the microphone is connected to connector P6, pins 3 & 4.
  - Line In: Select this option when using an external audio source with line level. On the TKIE kit, the line Input is on connector J8, pins 8 & 9.
- Line Out Source, Line Out can play audio either from VoIP signal or directly from microphone.
  - Audio Ch2: Plays audio from VoIP signal.
  - Microphone before AEC: The analog and digital (only 4.7 software) microphone signal is sent to Line Out. The loudspeaker signal is picked up by the microphone through the air, and also sent to the Line Out. There is no signal processing done.
  - Microphone after AEC:
  - If the station is in Open Mode: The microphone signal is sent to Line Out. The speaker signal is picked up by the microphone through the air, and also sent to the Line Out. Echo Canceling won't work, but Noise Reduction works. Audio from mic should be heard all the time on Line Out, while audio from other party should be heard only when PTT is pressed by said party.
  - If the station is in Full Duplex Mode: The microphone signal is sent to Line Out.
- Add number of "Mixer Channels"\_ICX-AlphaCom Multi-Conference Mixer function.
- Automatic Volume Control (AVC), Automatically adjusts the volume according to background noise level.
- AVC Debug, Shows current volume level on OLED display.

- AVC Advanced
  - AVC Lower Threshold: Threshold level for AVC starts working. Valid range: [-92..0] dBm0
  - AVC Upper Threshold: Threshold level for AVC stops working. Valid range: [-92..0] dBm0
  - AVC Attack Rate: How fast gain is raised when ambient noise level is increasing. Valid range: [0.100] 1/10th db/ sec ([0.1..10] dB/sec)
  - AVC Decay Rate: How fast gain is reduced when ambient noise level is falling. Valid range: [0..100] 1/10th db/sec ([0.1..10] dB/sec)
  - AVC Hysteresis: Required mic signal change level before new gain is adjusted. This is applied when there is a change in gain directions. Valid range: [1..10] db
  - AVC Lockout Time: Time for successive frames after AVC lock, due to far-end active or optional high near-end signal, before commencing AVC adjustments. A negative sign locks AVC if signal is above upper threshold. Valid range: [1..10] db
- Far-End Audio Squelch, Enables Audio Squelch on Far-End Signal (suppresses audio on low signal levels).
- Squelch Threshold, Signal level below this theshold will be suppressed to silence. Range is 0dB to -92dB. Default is-60dB.
- Squelch Activate Delay, Delays time with signal below threshold level before squelch is activated. Squelch is turned off again on first audio frame with level above threshold. This will implement a simple hysteresis hindering too rapid on/off with a signal changing around threshold.

# 4.4 I/O Settings

The I/Os can either be configured as an Input or as an Output. By default all I/Os are set as Inputs.

### Select Advanced ICX-AlphaCom > I/O Settings

Main	n Station Administration		Advanced ICX-AlphaCom	Advanced Networ	
► Au	dio	1/0	Settings		
	VC Cattings	Des	cription	Configur	
► DA	vo settings	Fast	t Blink Pattern	111000	
⊧ Off	line Relays	Slow	w Blink Pattern	111111	
▼ 1/0	Settings	1/0 P	Pin 1:	Input 1	
		1/0 P	Pin 2:	Input 2	
⊦ Key	board Settings	I/O P	Pin 3:	Input 3	
> Sou	und Detection	1/0 P	Pin 4:	Input 4	
		1/0 P	Pin 5:	Input 5	
► RIS	SP Settings	1/0 P	Pin 6:	Input 6	
) Tim	ne Settings	Relay	y:	Pin Num	
► Aut	dio Messages			s	

I/O Settings:

Fast Blink Pattern

Set the fast blink pattern for the LEDs

- Slow Blink Pattern
  - Set the slow blink pattern for the LEDs
- Select either Input or Output options from the drop-down box for I/O Pins 1 to 6

ation		
111000111	000111000	1 = on, 0 = off, 100 ms interval
111111000	00000000	1 = on, 0 = off, 100 ms interval
•		
T		
•		
•		
•		
•		
ber 7		
AVE	REBOOT	

# 4.5 Address Book

Note! Only the Turbine Compact station configured as a Scrolling Station (TCIS-6, TCIV-6+, TFIE-6) under Main Settings will have this menu option.

#### Select Advanced ICX-AlphaCom > Address Book •

Audio	Address Book		
I/O Settings	Description	Configuration	
RTSP Settings	Display Text:	Scroll to Select	
Address Book	Font Size	12 V	Under Landsland and ALED Ministra
Sound Detection	Start Scrolling After:	5 minutes	0 is off. Scrolling increases OLED lifetime
Time Settings	Menu Text Color	Dark 🔻	
Audio Mossagos	Sort Address Book	OFF T	
Audio Messages	Show Number.	Nu 🔻	
	Download or View Address Book	DOWNLOAD VIEW	
		SAVF	
	Upload Address Book		
	Description	Configuration	
	Upload Address Book:	Choose File No file chosen	Must be .csv file with format: number; te:

SAVE

Warning: Uploading new address book will delete existing address book entries.



#### Address Book:

- **Display Text** ٠
  - The idle text shown in the display may be changed. The default text is: Scroll to Select
- Font Size •
  - The font size can be either 12 or 16
- **OLED Brightness** •
  - Select Brightness levels: Extra Dim, Dim, Default, Bright, Extra Bright, Max Brightness
- Start Scrolling After •
  - Set the time in minutes, after which horizontal scrolling of the display text should start.
- Menu Text Color •
  - The text color can be either Dark or Light
- Sort Address Book • Sort according to: Name Ascending/Descending, DrNo Ascending/Descending
- Show Number •
  - Set: No or Yes

- Download or View Address Book ٠ • DOWNLOAD or VIEW the address book
- Upload Address Book, A CSV file consisting of directory numbers and display text with the semi-colon character (;) as delimiter may be uploaded.
  - Click Choose File to upload a CSV file

#### Address Book Entries

Fill in the Number and Name for each entry and click ADD

Up to 50 entries may be added.

## 4.6 OLED Labels

- Note! Only Turbine Compact stations configured with OLED Labels (TCIS-4, TCIS-5) under Main Settings will have this menu option.
- Select Advanced ICX-AlphaCom > OLED Labels •

Main	Station Administ	ration Advanced ICX-AlphaCom	Advanced Network		
	Audio	OLED Labels			
- F - I	/O Settings	Description	Configuration		
		OLED 1 Display Text:			
• (	OLED Labels	OLED 1 Font Size	12 🔻		
Þ.	RTSP Settings	OLED 2 Display text:			
	Deve d Detection	OLED 2 Font Size	12 •		
	Sound Detection	OLED Brightness	Default 🔻		Higher brightness reduces OLED lifetime
× 1	Time Settings	Start Scrolling After:	5	minutes	0 is off. Scrolling increases OLED lifetime
+ 1	Audio Messages	Menu Text Color	Dark 🔻		
			SAVE		

- **OLED Display Text** ٠
  - Enter display text in the relevant fields for OLED1 and OLED2.
- OLED Font Size •
- The font size can be either 12 or 16 •
- **OLED Brightness** ٠
- Select Brightness levels: Extra Dim, Dim, Default, Bright, Extra Bright, Max Brightness
- Start Scrolling After ٠
  - Set the time in minutes, after which horizontal scrolling of the display text should start.
- Menu Text Color ٠
  - The text color can be either Dark or Light

1	A
1	118;Guardroom
2	121;Logisitics
3	201;Stockroom
4	202;Meeting Room
5	203;Accounting
6	204;Garage
7	205;Reception
8	206:Canteen

# **4.7 Sound Detection**

Select Advanced ICX-AlphaCom > Sound Detection from the menu ٠

Audio	Sound Detection Settings		
DAVC Sattinge	Description	Configuration	
DAVE Settings	Sound Detection status:	DISABLED	
Offline Relays	Minimum amplitude (dBA):	57 🕶	
1/0 Settings	Minimum duration of audio (ms):	100	
y o octainigo	DAK key to activate:	5	
Keyboard Settings	Minimum time before reactivation (ms):	2000	
Sound Detection	Report DAK key off after (ms):	3000	
RTSP Settings		CANE	

#### Sound Detection Settings:

Here you can set the minimum amplitude and duration of the audio, the DAK to activate, etc.

# 4.8 Time Settings

Select Advanced ICX-AlphaCom > Time Settings from the menu •

Main Station Adminis	tration Advanced ICX-AlphaCom Adva	nced Network					
▶ Audio	Apr 23 2020 15:11						
DAVC Settings	Time Settings						
- orte octainingo	Description	Configuration	Configuration				
<ul> <li>Offline Relays</li> </ul>	Enable Network Time Protocol:		8				
I/O Settings	NTP server:	pool.ntp.org					
	Select Region:	Europe	•				
<ul> <li>Keyboard Settings</li> </ul>	Select Zone:	London	•				
Sound Detection	Enter Manual Date & Time:	2015 01 0	1 01	00	yyyy-MM-dd-hh-mi		
RTSP Settings		SAVE					
Time Cattings	Note: It may take a couple of minutes before	the NTD elient undetee time at a	tortup	ar of	tor hostnome char		

#### **Time Settings:**

You can enable Network Time Protocol, and select the region and time zone. You can also enter the date and time manually.

# **5 ICX-EDGE Configuration**

IC-EDGE is an IP-based intercom system for up to 32 intercom stations, expandable to 64 stations. The IC-EDGE system is easy to install and configure via a standard web browser.

For more information on IC-EDGE configuration, see wiki.zenitel.com/wiki/Category:IC-EDGE\_Configuration.

One of the stations must be set as the Edge Controller. The Edge Controller acts as a server for the other intercom stations in the system. Any Turbine device can serve as an Edge Controller.



# 5.1 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:

- Open a web browser 1.
- 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

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

Note! The user interface and parameters displayed in the following sections are dependent on the Turbine station type (Compact, Industrial or Ex) selected.



# 5.2 Main Settings

Select Main > Main Settings to access the page for configuring station mode and IP parameters. •



#### Mode:

Select the Edge Controller radio-button

Edge Controller must be selected to be able to carry out the configurations described in the following sections.

Product Model and Accessory: The options presented will depend on the Turbine model (Compact, Industrial or Ex).

- Select one of the options from the Model drop-down box:
- Model: (for Compact)
  - Kit(TKIS-2)
  - Normal (TCIS-1, TCIS-2, TCIS-3)OLED
  - Labels(TCIS-4,TCIS-5)
  - TFIE-6 • ECPIR-3P ScrollingStation(TCIS-6)

TFIE-1

• TFIE-2

- Mini (TMIS-1.TMIS-2. TMIS-4)
- Model: (for Extended) TKIE-1 (Kit)
  - TFIX-1-V2 • TFIX-2-V2
    - TFIX-3-V2

Model: (for Ex)

- Accessory: (for Extended & Ex)
  - Handset
  - Handset w/ Offhook
  - Handset w/ Offhook
  - (Normally Closed)
  - Handheld
  - Microphone Headset
  - Headset w/ Auto
  - detect

#### **IP Settings:**

- DHCP DO NOT select this option for the Edge Controller.
- Static IP The Edge Controller should always 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 frontboard and I/O · Check the box to disable factory reset using frontboard 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

# 5.3 Connect to other Intercom Stations

- Connect all other IP intercom stations to the network.
- Note! All other IP intercom stations have to be on the same LAN (IP subnet) as the Edge Controller for them to be auto-discovered by the Edge Controller. Stations on a different subnet have to be manually added under Manually Added Devices

Wait for the stations to boot up (approximately 60 seconds) before proceeding to the next step.

# 5.4 Configure Edge Controller Directory

- Log into the Edge Controller station with the new IP address that you have just set under Station Main > Main Settings. After login, you will find a new Edge Controller tab.
- Select Edge Controller > System Configuration > Directory

The Edge Controller station will auto-discover all the other stations on the LAN. To identify the individual stations on the LAN:

Click Play Tone. You should now hear a tone from the station you selected. •

System Overview	Directo	ry .						
System Configuration	Auto Disc	overed Devices						REFRESH
	Number	Name	SIP Password	Proble		DHCP / Static IP	Read IP Address	Play Tone
Directory	10	Main door		Default	~			-40
Call and Audio Direct Access Keys Special Settings System Time Credentials	11	Back door		Default	Ŷ	10.9.8.17		-40
	12	Deliveries		Default	~	10.9.8.38		-40
	13	Reception		Default	~	10.9.8.34		-40
	14	Gate A2		Default	~	169.254.1.100		-40

You can now set Numbers, Names, IP addresses, Profiles, etc. for all the stations in the network. In our example, the Edge Controller is designated as Reception with directory number 13 while the two substations are the Main Door and Back Door with directory numbers 10 and 11 respectively.

Click SAVE. The Edge Controller will now push the directory settings to all the IP intercom stations.

The devices should now have the status Registered.



Note! SIP Password is an optional field you can set to increase the security level. The password is used by other intercom stations to register with the Edge Controller.

# 5.5 Verify System Setup

To verify that the devices are successfully registered to the Edge Controller:

Select Edge Controller > System Overview •

an coye constant							
<ul> <li>System Overview</li> </ul>	Directory						
System Configuration	Static report of	all configured device	s in your system.				
> Device Profiles							Download CSV
	Number	Name	Status	IP Address	Profile	Туре	
Group Calls	10	Main Door	Registered	10.9.8.7	Default	Vingtor-Stentofon Device	
	11	Back Door	Registered	10.9.8.117	Default	Vingtor-Stentofon Device	
Software Upgrade	12	Deliveries	Recistered	10.9.8.127	Default	Vingtor-Stentofon Device	
	15	Bacastina	Desistand	10 0 0 124	Default	Visator Clastadas Baulca	
	1.5	PRECE PLACES	Partitioner and	10.7.0.1.34	LUTC'S BLUE	A BURNER ORGINATION DEALTS	

All stations that have been configured should be displayed in the table. Stations that are up and running will have the • status Registered.

# 5.6 Configure Call and Audio Settings

It is optional to configure the call and audio settings. Default settings will be used if they are not configured.

Select Edge Controller > System Configuration > Call and Audio ٠

Edge Controller									
System Overview	Call and	Audio							
System Configuration	Number	Name	Auto Answer	Speaker Volume	Volume Override Lever	ANC	AGC	AVC	Accessory
Directory	10	Main door	Ŋ	5 ~	8 ~	N			
Call and Audio	11	Back door		5 ~	8 ~				
Urect Access Keys Special Settings	12	Deliveries		3 ~	8 ~				
Operation Time	12	Reception		4.4	and an				

The Call and Audio menu include the following parameters:

- Auto Answer. When not enabled, incoming calls will ring (Private mode) and has to be accepted by the user. When enabled, the call will connect straight through (Open mode). Auto Answer is typically enabled for substations and not for master stations.
- Speaker Volume. This parameter sets the speaker volume for the station. •
- Volume Override Level. This sets the loudspeaker volume during volume override. Volume Override is used by Group Calls with Emergency priority.
- ANC. Active Noise Cancellation: When enabled, most of the background noise will be filtered out of the microphone signal.
- AGC. Automatic Gain Control: When enabled, the microphone signal will automatically be amplified when speaking ٠ with a low voice or if you are far away from the station.
- AVC. Automatic Volume Control: When enabled, the loudspeaker volume is adjusted according to the ambient noise level. AVC will perform better if the Speaker Volume is set to a fairly low level, as this will give AVC a more dynamic range for adjustments. Note that AVC adjustment is disabled while the speaker is playing audio. Adjustments are made in silent periods (min. 100 ms silence required).
- Accessory. If the station model supports accessories (headset, handset, etc.), you can choose the type of accessory here.

# 5.7 Modify IC-EDGE Device Profiles

The Station Profile defines a set of service features and parameters that are available for a group of devices.

The IC-EDGE system can have five device profiles:

- Profile 1 Default
- Profile 2 Substation
- Profile 3 Display station
- Profile 4 Operator
- Profile 5 Gateway
- The following service features and parameters are included in the device profile:
  - Outgoing Call restriction
  - Relay Activation (i.e. Door opening)
  - Group Call initiation
  - Busy Override

To modify device profiles:

Select Edge Controller > Device Profile



The following parameters can be set for the station profile:

- Profile nickname
  - The name of the profile. Any text can be used.
- Outgoing calls allowed
  - Stations with this profile can call stations with the checked profiles. Forbidden will be displayed on the station if this is denied.
- **Relay activation allowed** 
  - Stations with this profile can activate the relay (e.g. Door Opening) on stations with the checked profiles.
- Group calls allowed
- · Stations with this profile are allowed to dispatch the checked group calls
- Busy override allowed
  - When calling a busy station, stations with this profile can force a connection by pressing 5.

Co	nfiguration					
Det	ault					
1	Default 🗹	Substatio	on 🕑	Display s	tation 🗹	Operat
	Default	Substatio	on 🔲	Display s	tation 🔲	Operat
1	All call 🕑	Conf 🗹	Group	call 2 🗹	Group ca	all 3
	All call 🗹	Conf 🗹	Group	call 2 🗹	Group ca	all 3

# 5.8 Group Call

The IC-EDGE system supports 4 Group Calls.

A Group Call is activated by dialing the appropriate code (e.g. 84). A ding-dong chime will be heard on all member stations. Press the M-key to speak, and C-key to disconnect. Alternatively, Group Call can be used in handsfree mode, i.e. no need for pressing the M-key. Stations without any M-key will use handsfree mode by default. Group Call can be answered from any station by dialing the answer code 99.

Group Call audio can be sent and received by all Zenitel IP stations and can be sent from most modern SIP IP phones. Group Call audio can be received by SIP IP phones that have support for IP Multicast paging.

- Operation. A Group Call is activated by dialing the appropriate code (e.g. 84). A ding-dong chime is heard in all member stations.
  - Press the M-key to speak.
  - · Press the C-key to disconnect.
- Default Group Call Settings. The factory default settings for group calls are as follows:

Nickname (Display Text)	Directory Number	Priority	Group Members	Chime	Answer
All Call	84	EMERGENCY	All Stations	Ding Dong	Yes
Group Call 1	85	HIGH	None	Ding Dong	Yes
Group Call 2	86	NORMAL	None	Ding Dong	Yes
Group Call 3	87	LOW	None	Ding Dong	Yes

When adding a new station to an existing IC-EDGE system, the station will automatically become a member of the All Call group.

- Configuration of Group Calls. The Group Call properties are configured via the Edge Controller.
  - Select Edge Controller > Group Calls

Main Edge Controller		
▶ System Overview	Group Call 1 - All call	
Custom Configuration	Description	Configuration
<ul> <li>System Conliguration</li> </ul>	Nickname	All call
Device Profiles	Number	84
Group Calls	Answer Group Call	
	Answer Timeout	30
Software Upgrade	Priority	EMERGENCY V
Licensing	Chime	dingdong.wav 🔻
	Add all devices	
	Devices in group	101 TCIS-2 103 TCIV-6 104 TMIS-1 105 IP DeskMaster
	Group Audio Address	239.195.40.64:61060

Change the parameters for each group as required:

- Nickname. This is the display text shown in the initiating station when the group call is activated. Can be any text.
- Number. The number to dial to activate the group call. If changed, a reboot of the Edge Controller is required for the changes to take effect.
- Answer Group Call. Enable Group call answer by pressing "99" and setting the Answer Timeout. Default timeout value is 30 seconds to answer after the Group Call has been terminated. Timeout value of 0 means answering is only possible during the Group Call.

- Priority. Priorities can be set for one Group Call at a time. A Group Call with higher priority will override a lower priority Group Call.
  - EMERGENCY: Volume and handset override. The Group Call audio is mixed with conversation audio
  - HIGH: The Group Call audio is mixed with conversation audio
  - NORMAL: The Group Call audio is not played if there is an ongoing conversation
  - LOW: The Group Call audio is not played if there is an ongoing conversation
- Chime. Enables the playing of a ding-dong chime when a device receives a Group Call.
- Devices in group. Select the devices that should receive Group Call audio. You can select all devices in one operation be checking the Add all devices box.
- Group Audio Address. Multicast IP address and port used to send audio during Group Call. This information can be used to set up a IC-EDGE Group Call (multicast page) on 3rd-party SIP stations. There is a SAVE button for each Group Call.
  - · Remember to click SAVE before you make changes to the next Group Call.

# 5.9 Configure 3rd-party SIP Devices

Up to 10 3rd-party SIP devices such as SIP phones, SIP Speakers, Soft Phones can be registered to an IC-EDGE System. Each SIP device requires that a SIP User account is defined on the Edge Controller.

# 5.9.1 Install Licence

Before adding a 3rd-party SIP account to the IC-EDGE system, you need to obtain a license key. For each SIP device used in the system, there must be a valid SIP license installed. To install a license:

Select Edge Controller > Licensing



Enter the license key in the New License field and click ACTIVATE LICENSE

The Licensing table should now show all the licenses that are available

ses	Currently Used	Available
	0	0
	0	4
	0	0
SX6k3kYFA3hw)	(V85pJ9qj3mq4R@yF@R	
license key		

# 5.9.2 Create SIP Account

Select Edge Controller > System Configuration > Directory

▼ System Configuration         Name         SIP Password         Profile         DHCP / Static IP         Real           Directory Call and Audio Direct Access Keys Special Settings System Time Credentials         Interference         Interference	ad IP Address Play To (1) Play To (1) P
Directory Call and Audio     10     Main Door     Default     ✓       Direct Access Keys Special Settings     11     Back Door     Default     ✓     10.9.8.17     ✓       12     Deliveries     Default     ✓     10.9.8.38     ✓       System Time     13     Reception     Default     ✓     10.9.8.34	<b>4</b> 3) <b>4</b> 3)
Call and Audio     11     Back Door     Default     ✓     10.9.8.17     ✓       Direct Access Keys     12     Deliveries     Default     ✓     10.9.8.38     ✓       System Time     13     Reception     Default     ✓     10.9.8.34	<b>4</b> 10)
Direct Access Keys     12     Deliveries     Default     V     10.9.8.38       System Time     13     Reception     Default     V     10.9.8.34	
System Time 13 Reception Default V 10.9.8.34	<b>4</b> 1))
Credentials	40
14 Mr. Alis Default ~ 169.254.1.100	
Device Profiles  Group Calls  Manually Added Devices  Number Name SIP Password Profile Type	+
Software Upgrade 300 SIP Device Default V 3rd party:	Vingtor-Stentofon Devi
Licensing Total 1 devices manually added to the system.	Vingtor-Stentofon Clie
Linked Edge Controllers	3rd party S
Number Start Number Stop Name Profile Host	one party o

- Under Manually Added Devices select 3rd party SIP from the dropdown list •
- Modify Number and Name as required (a default Number and Name will be inserted which you can modify) •
- Optionally one can add a SIP Password. If a password is entered, one has to also manually set the same password • in the SIP device it self.
- Click SAVE to store the settings of the new SIP device in the Edge Controller. •

# 5.9.3 Configure SIP Phone

You now have to log into the 3rd-party SIP phone to configure the SIP account to register it with the Edge Controller station. The Directory Number and Password (SIP Account) created in the previous section is used to register the 3rd-party station with the Edge Controller.

# 5.9.4 Verify Operation

On the Edge Controller device:

Select Edge Controller > System Overview

Main	Edge Controller							
- Sys	stem Overview	Directory						
> Sys	stem Configuration	Static report of	all configured device	s in your system.				
Device Profiles								[Download CSV]
1.16	vice Profiles							[
De	vice Profiles	Number	Name	Status	IP Address	Profile	Туре	
+ Gro	oup Calls	Number 10	Name Main Door	Status Registered	IP Address 10.9.8.7	Profile Default	Type Vingtor-Stentofon Device	
<ul> <li>Gro</li> </ul>	oup Calls	Number 10 11	Name Main Door Back Door	Status Registered Registered	IP Address 10.9.8.7 10.9.8.117	Profile Default Default	Type Vingtor-Stentofon Device Vingtor-Stentofon Device	
<ul> <li>Gro</li> <li>Sol</li> </ul>	oup Calls ftware Upgrade	Number 10 11 12	Name Main Door Back Door Deliveries	Status Registered Registered Registered	IP Address 10.9.8.7 10.9.8.117 10.9.8.138	Profile Default Default Default	Type Vingtor-Stentofon Device Vingtor-Stentofon Device Vingtor-Stentofon Device	
<ul> <li>→ Gro</li> <li>→ Sol</li> </ul>	vice Promes oup Calls ftware Upgrade	Number 10 11 12 13	Name Main Door Back Door Deliveries Reception	Status Registered Registered Registered Registered	IP Address 10.9.8.7 10.9.8.117 10.9.8.138 10.9.8.134	Profile Default Default Default Default	Type Vingtor-Stentofon Device Vingtor-Stentofon Device Vingtor-Stentofon Device Vingtor-Stentofon Device	
<ul> <li>Def</li> <li>Gro</li> <li>Sol</li> <li>Lic</li> </ul>	vice Promes oup Calls ftware Upgrade censing	Number 10 11 12 13 14	Name Main Door Back Door Deliveries Reception Mr. Alis	Status Registered Registered Registered Registered Registered	IP Address 10.9.8.7 10.9.8.117 10.9.8.138 10.9.8.134 10.9.8.134	Profile Default Default Default Default Default	Type Vingtor-Stentofon Device Vingtor-Stentofon Device Vingtor-Stentofon Device Vingtor-Stentofon Device Vingtor-Stentofon Device	

Check that the SIP Device has gotten the Status = Registered •

Verify that you can call to and from the SIP Device.

# 5.10 Advanced Configuration Mode

Settings for Direct Access Keys, Scripts, etc. described in the following sections are only available in Advanced Configuration Mode.

To enter Advanced Configuration Mode:

Select Main > Recovery

lain Edg	e Controller		
	Comman	nde	
Main Sott	Description		Action
· Recovery	Full reboot		REBOO
Help	Partial reboo	x	REBOO
	Factory reset	t	FACTO
	Factory reset	t with DHCP	FACTO
	Preference	ces	
	Description		Configura
	Advanced co	onfiguration mode	Type offli

#### Preferences:

Mair

•

-

٠

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

Edge Controller		
Information	Commands	
Main Sottings	Description	Action
main Settings	Full reboot	REBOOT
Recovery	Partial reboot	REBOOT
Help	Factory reset	FACTORY RESET
	Factory reset with DHCP	FACTORY RESET
	Preferences	
	Description	Configuration
	Advanced configuration mode	2

- Check the Configuration box •
- Click SAVE •



т			
т			
RY RESET			
RY RESET			

ne password to unlock advanced configuration mode

	-
SAVE	

Advanced configuration mode activated. Reload page to continue.

#### Click RELOAD •

#### A new tab called Edge Configuration will now appear:

Main Edge Configuratio	n Station Administration Fd	ige Controller	Advanced Network	
✓ Account / Call	Account Settings			
▶ Audio	Description		Configuration	
	Name:		TCIS-2	
Direct Access Keys	Number (SIP ID):		101	
Relays / Outputs	Server Domain (SIP):		10.9.5.105	
▶ Time	Backup Domain (SIP):			
	Backup Domain 2 (SIP):			
► I/O	Authentication User Name:		101	
Frontboard Mapping	Authentication Password:			
▶ RTSP	Register Interval:		600	(min. 60 seconds)
▶ Script	Register Failure Interval:		60	(min. 5 seconds)
▹ Script Events				
Corint Unload	Call Settings			
Script opload	Description		Configuration	
Audio Messages	Enable Auto Answer:		2	
Multicast Paging	Auto Answer Delay:		0 seconds	Max 30 seconds.
▹ Certificates	Press and Hold Time:		0 seconds. M pressed before the	ax 60 seconds. Defines how long a DAK key/Input must be call is established.

# 5.11 Direct Access Key & Ringlist 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 Edge Configuration > Direct Access Keys

Account / Call	Account Sett	ings		
Audio	6	Function		
	Button 1	Idle: Call To	No Ringlist V	Normal •
Direct Access Keys	Button 1	Call: Do Nothing 🔻		
Relays / Outputs		Idle: Call To 🔹	No Ringlist 🔻	
Time	Input 1	Call: Do Nothing 🔻		
/0	Input 2	Idle: Call To	No Ringlist 🔻	T
Frontboard Mapping	input 2	Call: Do Nothing		
RTSP	locut 3	Idle: Call To 🔹	No Ringlist 🔻	•
Sorint	input o	Call: Do Nothing		
penpe		Idle: Call To	No Ringlist 🔻	•
Script Events	Input 4	Call: Do Nothing		
Script Upload		Idle: Call To	No Ringlist 🔻	•
Audio Messages	Input 5	Call: Do Nothing 🔹		
Multicast Paging		Idle: Call To 🔹	No Ringlist V	•
Certificates	input 6	Call: Do Nothing		



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

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 (avaliable 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 Ringlist.
- 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 (handsfree) 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 ٠ 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. ٠
- 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
- Offhook
- Onhook

## 5.11.1 Ringlist Settings

• Log on to the web interface of the station that will be making the call to configure the ringlist of the stations that should be called.

To configure a ringlist:

Select Edge Configuration > Direct Access Keys

There are variants of ringlists that can be set such as ringing time intervals, loopback, and ringing together with previous stations.

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.

# 5.11.1.1 Forward Unattached Call

Main Ed	ge Configuration Statio	n Administration	Fdge Controller	Advanced Network		
> Account,	/ Call Account	t Settings				
> Audio		Func	tion			
Direct de	Rotten 1	Idle:	Call To	10	Ringlist 1 🔻	Normal 🔻
▼ Direct Ac	Cess Keys Button I	Call:	Answer Call 🔹	Filter Dir. No.		Answer Group Call
Relays /	Outputs	Idle:	Call To	•	No Ringlist V	<b>T</b>
⊧ Time	Input 1	Call:	Do Nothing 🔻			
→ I/O		Idia-	Call To	•	No Pinglist ¥	•
Frontboa	rd Mapping Input 2	Call:	Do Nothing V		no mignar v	•
▶ RTSP		Idle:	Call To	•	No Ringlist 🔻	•
<ul> <li>Script</li> </ul>	Input 3	Call:	Do Nothing 🔻			
<ul> <li>Script Ev</li> </ul>	ents	Idle.	Call To	•	No Ringlist 🔻	<b>T</b>
<ul> <li>Script Up</li> </ul>	load	Call:	Do Nothing 🔻			
Audio Me	essages	Idle:	Call To	•	No Ringlist 🔻	•
Multicast	t Paging	Call:	Do Nothing 🔻			
<ul> <li>Certificat</li> </ul>	es locut 6	Idle:	Call To	-	No Ringlist 🔻	•
	input o	Call	Do Nothing V			



Loop Ringlist		(loops the ringlist)
Ringing Time	10	seconds, (0=unlimited)
Max Conversation Time	0	seconds, (0-unlimited)

The above example ringlist setting may be illustrated as follows:



Via the Ringlist, an unattended call can be forwarded to another station after a preset time. The call forwarding can include several stations.

In the example, when the call button is pressed, the call goes to station number 10. If not answered within 10 seconds, the call will be forwarded to station number 12. If not answered, the call will be routed to 11, and finally to station number 17.

#### SAVE

With Previous	Ringlist 3	With Previous
	With Previous	With Previous Ringlist 3



# 5.11.1.2 Forward Unattached Call with Loopback

Account / Call	Account Settings					
Audio		Function				
Direct Access Keys	Button 1	Idle: Call To	۲	10	Ringlist 1 🔻	Normal 🔻
Difect Access regs		Call: Answer Call	•	Filter Dir. No.		Answer Group Ca
Relays / Outputs		Idle: Call To	T		No Ringlist 🔻	
Time	nput 1	Call: Do Nothing	•			
I/O		Idle: Call To	•		No Ringlist V	T
Frontboard Mapping	Input 2	Call: Do Nothing	•		· · ·	
RTSP		Idle: Call To	•		No Ringlist 🔻	•
Script	input 3	Call: Do Nothing	•			
Script Events		Idle: Call To	•		No Ringlist 🔻	•
Script Upload	input 4	Call: Do Nothing	•			
Audio Messages	innut E	Idle: Call To	•		No Ringlist 🔻	<b></b>
Multicast Paging	input 5	Call: Do Nothing	•			
Certificates		Idle: Call To	•		No Ringlist 🔻	•
	input 6	Call: Do Nothing				

**Ringlist Settings** 

	Ring	list 1	With Previous	Ringlist 2	With Previous	Ringlist 3	With Previous
Value 1	12						
Value 2	11						
Value 3	17						
Loop Ringlist	3	loops the ringlist)	wered)				
March and March and	3	(0 = loop until ans	wered)				
Max Loop Number	-	(o roop onth and					
Ringing Time	12	seconds, (0=unlin	nited)				

#### The above example ringlist settings may be illustrated as follows:



Via the Ringlist, an unattended call can be go through a call loop a preset number of times.

In the example, when the call button is pressed, the call goes to station number 10. If not answered within 12 seconds, the call will be forwarded to station number 12. If not answered, the call will be routed to 11, and finally to station number 17. Now the call is looped back to station number 10, repeating the same call pattern 3 times.

# 5.11.1.3 Parallel Ringing - Forward Call if Unattended

Main Edge Co	nfiguration Station Adm	inistration Edge Controller	Advanced N
Account / Call	Account Set	tings	
> Audio		Function	
Direct Access H	eys Button 1	Idle: Call To	10 Filter (
→ Relays / Output ► Time	Input 1	Idle: Call To	
» I/O		Call: Do Nothing 🔻	
Frontboard Map     prep	pping Input 2	Call: Do Nothing 🔻	
FRISP FRISP FRISP	Input 3	Idle: Call To 🔹	
▹ Script Events	Input 4	Idle: Call To 🔻	
Script Upload     Audio Message	25	Call: Do Nothing  Idle: Call To	
<ul> <li>Multicast Pagir</li> </ul>	Input 5	Call: Do Nothing	
Certificates	Input 6	Idle: Call To  Call: Do Nothing	

.

	Ring	plist 1	With Previous	Ringlist :
Value 1	12			
Value 2	11			
Value 3	17			
Loop Ringlist		(loops the ringlist)		
Ringing Time	15	seconds, (0-unlimited)		
Max Conversation	0	seconds, (0=unlimited)	1	

The above example ringlist setting may be illustrated as follows:



By checking the With Previous box in the Ringlist, one can call several stations in parallel. When one of the stations answers, it will stop ringing in the other(s).

In the example, when the call button is pressed, the call goes to station number 10 and 12 in parallel. If not answered within 15 seconds, the call will be forwarded to stations number 11 and 17.

# 5.12 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

The Turbine station should be operating in **Edge** mode. *Turbine Stations Technical Manual* 

	Ringlist 1 V	Normal	٠
2		Answe	r Group
	No Ringlist 🔻		T
	No Ringlist 🔻		•
	No Ringlist 🔻		Ŧ
	No Ringlist 🔻		۲
	No Ringlist 🔻		۲
	No Ringlist 🔻		٠
	With Ringl Previous	ist 3	V P
			_ (

# 5.12.1 Script Upload

Select Edge Configuration > Script Upload

ain Edge Configurati	ion Station Admir	istration Advanced Network	1		
> Account / Call	Scripts				
Audio	Foriat 1	Name		BELETE	
Direct Access Keys	Script 1	activateoutputtica		DELETE	
Relays / Outputs	Upload Script				
Time	Choose File	lo file chosen			
OLED Labels			101010		
⊳ I/O			UPLOAD		
RTSP					
Script					
Script Events					
Script Upload					

#### Upload Script:

- Click Choose File to select the desired script •
- Click UPLOAD to upload the desired script •

## 5.12.2 Script Configuration

Select Edge Configuration > Script

Main Edge Configurat	Station Administration Advanced Network
Account / Call	Digital Outputs - Scripts
> Audio	Choose script slot to configure: Slot 1 V
<ul> <li>Direct Access Keys</li> </ul>	Assign a Label Activate Output
Relays / Outputs	Enter script in selected slot
→ Time	luə activateOutput.luə admin əlphəədmin gpio1 10.9.7.10
▹ OLED Labels	
⊨ I/0	SAVE
► RISP	
- 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 •

# 5.12.3 Script Events

Select Edge Configuration > Script Events

Digital Outputs - Events			
Event	Execute		
Remote Digit (DTMF): Digit 3 •	Activate Output	<ul> <li>after a duration of</li> </ul>	0 •
Incoming Call:	-	<ul> <li>after a duration of</li> </ul>	0 •
Outgoing Call:	-	<ul> <li>after a duration of</li> </ul>	0 •
Incoming Ringing:		<ul> <li>after a duration of</li> </ul>	0 •
Outgoing Ringing:	-	<ul> <li>after a duration of</li> </ul>	0 •
Idle:		<ul> <li>after a duration of</li> </ul>	0 •
Error (Not Registered):		<ul> <li>after a duration of</li> </ul>	0 •
Input 1 Pressed	-	<ul> <li>after a duration of</li> </ul>	0 •
Input 1 Released	-	<ul> <li>after a duration of</li> </ul>	0 -
Input 2 Pressed	-	<ul> <li>after a duration of</li> </ul>	0 -
Input 2 Released	-	after a duration of	0 •
	Event  Remote Digit (DTMF): Digit ③  Incoming Call:  Outgoing Call:  Outgoing Ringing:  Idle:  Error (Not Registered):  Input 1 Pressed Input 1 Released Input 2 Pressed Input 2 Released	Event         Execute           Remote Digit (DTMF):         Digit 3 •         Activate Output           Incoming Call:         -         -           Outgoing Ringing:         -         -           Outgoing Ringing:         -         -           Idle:         -         -           Error (Not Registered):         -         -           Input 1 Pressed         -         -           Input 2 Pressed         -         -	Event         Execute           Remote Digit (DTMF):         Digit 3 ▼         Activate Output ▼         after a duration of           Incoming Call:         -         ▼         after a duration of           Outgoing Call:         -         ▼         after a duration of           Incoming Ringing:         -         ▼         after a duration of           Outgoing Ringing:         -         ▼         after a duration of           Idle:         -         ▼         after a duration of           Error (Not Registered):         -         ▼         after a duration of           Input 1 Pressed         -         ▼         after a duration of           Input 2 Pressed         -         ▼         after a duration of           Input 2 Released         -         ▼         after a duration of

**Digital Outputs - Events:** 

Set the Digit for the Event to Execute

# 5.13 Audio Messaging

Prerecorded audio messages can be uploaded to a Turbine which can be triggered by various events occurring on the station. The audio message playback can be triggered by 3 main events: DTMF, Call, Relay. There are two ways of configuring audio messages.

1. Via the Edge Controller whereby all stations in the system can be configured:

Select Edge Controller > System Configuration > Special Settings

2. Via each individual station as describe below:

Select Edge Configuration > Audio Messages

# 5.13.1 Updating Audio Files

• Select Edge Configuration > Audio Messages

Main	Edge Configuration	Station Administration	Edge Controller	Advanced Net
> Ac	count / Call	Media		
► Au	idio	Nam	e	
→ Dir	rect Access Keys	1 UnattendedBaggage_en	_30_3.wav	
▶ Re	lays / Outputs	Space used: 0.504 mb of 20 Upload Media	mb J	
> Tir	ne			
► I/0	)	Choose File No file chose	n	
> Fro	ontboard Mapping			UPLO
→ RT	SP	Message Settings		
→ SC	ript	Choose Message	Event	Option
> SC	ript Events	Choose Message	▼ Unused ▼	]
► SC	ript Upload			
▼ Au	dio Messages	Choose Message	▼ Unused ▼	]
→ Mu	ulticast Paging	Chanse Message		ĩ
→ Ce	rtificates	choose message	- onused +	1

twork		
	Delete	
LOAD		

#### Under Upload Media:

- Click Choose File to browse to the desired audio file
- Click UPLOAD

The file will now appear in the list of audio files under Media.

After the upload, the amount of space left of the maximum of 20 MB is displayed. It is possible to delete audio files to free up more space by clicking **Delete**.

# 5.13.2 Play Message on DTMF Event

An audio message can be played on receivingd DTMF tones during a call. Valid digits are **DTMF 0 - 9**, \*, **#**. The event can be filtered by directory number (SIP ID).

Choose Message	Event	Option
RingTone.wav	VDTMF	→ DTMF 3 → Dir. No / SIP ID Speaker →
Choose Message	√ Unused	

In the configuration example above, the audio file "ringcd.wav" will be played when the station receives DTMF 3 during a call (i.e. the other party in the conversation presses digit 3 on his keypad).

Note! The call must be established before the DTMF digit is sent, otherwise the message will not be triggered. The message will be played once, and mixed with the conversation audio.

# 5.13.3 Play Message on Call Event

An audio message can be played on **Incoming/Outgoing**, **Incoming**, **Outgoing** call direction. It can in addition be filtered by directory number (SIP ID). The following call events are supported:

- Call Ringing: Incoming/Outgoing call on the station (no auto-answer)
- Call Queued: Outgoing call from the station has been queued on the master station
- Call Accepted: Incoming/Outgoing call on the station has been accepted/answered
- Group Accepted: Outgoing Group Call from the station has been established
- Call Ended: Incoming/Outgoing call on the station has been ended/hung-up/cancelled
- On Hold: A call from the station has been put on hold by remote master station
- **Transferred**: A call from the station has been transferred by remote master station to a third station, and the station is ringing (no auto-answer). "Your call has been transferred, please wait..." message will be played on the station. The event will not be triggered on the master station that initiated the transfer.

You can choose whether the message should be played once, or be repeated until the Call Event is ended:

- One Time: The audio message will be played once.
- Repeat: The message will be repeated with a time interval set in Pause until the call event is ended.
- File Then Tone: The audio message will be played once, followed by the standard ringing tone.

The example below shows a setting where a reassurance message (e.g. "Your call has been registered, please wait") will be played to the caller while waiting for the receiving station to answer:

Choose Message		Event		Option
RingTone.wav	~	Call	~	Call Ringing         Outgoing         Dir. No / SIP ID           Speaker         File Then Tone
Choose Message	~	Unused	~	

The audio message will be played once, followed by the standard ringing tone.

# 5.13.4 Play Message on Relay Event

An audio message can be played when a Relay or an Output changes state. Available states are: **On**, **Off**, **Slow Blink**, **Fast Blink**.

The example below shows a setting where a message (e.g. "The door is open, please enter") will be played when the relay of the door station is operated:

Play a message when the relay at the door station is operated

Message Settings						
Choose Message		Event		Option		
D		Delaw	_	Relay 1 🗸	On	~
Door Open.wav	~	кегау	~	Speaker	~	
Choose Message	~	Unused	~			

The audio message will be played once and mixed with the conversation audio.

# 5.13.5 Routing of Audio Message

The audio message can be routed to: Speaker or Speaker & Mic

The audio message is always routed to the loudspeaker of the station, but there is an option to send the audio message as a "Microphone" signal as well. This means that the audio message can be transmitted to the other party during an established call. The "Speaker & Mic" option is feasible for the call events **Call Accepted** and **Group Accepted**. When "Speaker & Mic" option is selected, the microphone signal will be disabled while the message is playing. Once the message has stopped playing, the microphone will be enabled again.

The example below shows a setting where a when an elevator alarm is triggered, and the operator answers the call, the location of the alarm will be played as a voice message:

When the alarm call from the elevator intercom is answered, an audio message, e.g. "Alarm Call from Elevator B5", is played in the elevator and to the receiver of the call.

**Turbine Stations Technical Manual** 

# **6 SIP Configuration**

SIP (Session Initiation Protocol) is the de facto standard for IP telephony. The Turbine SIP intercom stations are specially built for easy integration with any iPBX system.

For more information on SIP configuration, see wiki.zenitel.com/wiki/SIP Intercom Configuration



The following sections describe configuration procedures using the web interface of the station.

# 6.1 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:

- Click Login 1.
- 2. Enter the default User name: admin
- З. Enter the default Password: alphaadmin

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



Note! The user interface and parameters displayed in the following sections are dependent on the Turbine station type (Compact, Industrial or Ex) selected.

## 6.2 Main Settings

Click Main > Main Settings •





#### Mode:

Select the SIP radio-button

Product Model And Accessory: The options presented will depe Select one of the options from the Model drop-down box:

• TKIE-1 (Kit)

• TFIE-1

• TFIE-2

• TFIE-6

ECPIR-3P

- Model: (for Compact) Model: (for Extended) ٠
  - Kit(TKIS-2)
  - Normal (TCIS-1, TCIS-2,
  - TCIS-3)OLED
  - Labels(TCIS-4,TCIS-5)
  - ScrollingStation(TCIS-6) • Mini(TMIS-1,TMIS-2,
  - TMIS-4)

# 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 frontboard and I/O:

Check the box to disable factory reset using frontboard and I/O.

• 0         • 5         • 101           • 255         • 255         0           • 0         • 5         1           • 200         • 200         200           • 200         • 200         200	- 0         - 5         - 101           - 255         - 255         - 0           - 255         - 255         - 10           - 200         - 200         - 200           - 620         - 200         - 200									
- 0         - 5         - 101           - 255         - 255         - 0           - 0         - 5         - 1           - 200         - 200         - 200           - 200         - 200         - 200           - 604-508	- 0         - 55         - 101           - 255         - 255         - 0           - 255         - 55         - 1           - 250         - 200         - 200           - 200         - 200         - 200									
•         0         •         5         •         101           •         255         •         255         •         0           •         0         •         5         •         1           •         200         •         200         •         200           •         200         •         200         •         200	- 10 - 15 - 101 - 255 - 255 - 0 - 0 - 5 - 1 - 200 - 200 - 200 - 200 - 200 - 200 - 200 - 200 - 200									
-         0         -         5         -         101           255         255         0         - </th <th>- 19 - 15 - 101 - 255 - 255 - 0 - 9 - 5 - 1 - 200 - 200 - 200 - 200 - 200 - 200</th> <th></th> <th></th> <th></th> <th></th> <th></th> <th></th> <th></th> <th></th> <th></th>	- 19 - 15 - 101 - 255 - 255 - 0 - 9 - 5 - 1 - 200 - 200 - 200 - 200 - 200 - 200									
- 10 - 10 - 101 - 255 - 255 - 0 - 0 - 5 - 1 - 200 - 200 - 200 - 200 - 200 - 200 - 200 - 200 - 200	- 10 - 15 - 101 - 255 - 255 - 0 - 0 - 5 - 1 - 200 - 200 - 200 - 200 - 200 - 200 - 400-4508									
- 10 - 15 - 101 - 255 - 255 - 0 - 0 - 5 - 1 - 200 - 200 - 200 - 200 - 200 - 200 - 200 - 200	- 19 - 15 - 101 - 255 - 255 - 0 - 9 - 5 - 1 - 750 - 750 - 200 - 200 - 200 - 200 tel04-508									
- 0 - 5 - 101 - 255 - 255 - 0 - 0 - 5 - 1 - 200 - 200 - 200 - 200 - 200 - 200 - 200 - 200	-9         -55         -101           -255         -255         -0           -0         -5         -1           -200         -200         -200           -200         -200         -200           H06-4508									
- 10 - 5 - 101 - 255 - 255 - 0 - 0 - 5 - 1 - 200 - 200 - 200 - 200 - 200 - 200 H064508	- 10         - 55         - 101           - 255         - 255         - 0           - 0         - 5         - 1           - 200         - 200         - 200           - 200         - 200         - 200           - 100-4508									
-9         -5         101           -255         -255         -0           -9         -5         -1           -200         -200         -200           +200         -200         -200           +1064508	- 0         - 5         - 101           - 255         - 255         0           - 0         - 5         - 1           - 200         - 200         - 200           - 200         - 200         - 200           - 100         - 200         - 200           - 100         - 200         - 200									
- 10 - 55 - 101 - 555 - 255 - 0 - 00 - 5 - 1 - 200 - 200 - 200 - 200 - 200 - 200	- 10         - 15         - 101           - 255         - 255         - 0           - 0         - 5         - 1           - 700         - 200         - 200           - 200         - 200         - 200           - 200         - 200         - 200           - 200         - 200         - 200									
- 0 - 5 - 101 - 255 - 255 - 0 - 0 - 5 - 1 - 200 - 200 - 200 - 200 - 200 - 200 - 200	P         S         101           255         255         0           0         255         1           200         200         200           200         200         200           200         200         200           200         200         200									
- 255         - 255         - 0           - 0         - 5         - 1           - 200         - 200         - 200           - 200         - 200         - 200           - 200         - 200         - 200	· 255         · 255         · 0           · 0         · 5         · 1           · 200         · 200         · 200           · 200         · 200         · 200           · 106+508         · 10		ŀ	9	ŀ	5	ŀ	101		
- 0 - 5 - 1 - 200 - 200 - 200 - 200 - 200 - 200 + 200 - 200	- 0 - 5 - 1 - 200 - 200 - 200 - 200 - 200 - 200 + 200 - 200 - 200		ŀ	255	ŀ	255	ŀ	0		
- 1200 - 1200 - 1200 - 1200 - 1200 - 1200 Re1064508	- 200 - 200 - 200 200 - 200 - 200 Re/04508		•	0		5	•	1		
- 200 - 200 - 200 ReIO64508	- [200 - [200 - 200 ] HelD64508	)	ŀ	200	ŀ	200	ŀ	200		
ne1064508	161064508			200		200		200		
		10	106	4508		_	1			
			-		-		-		 	 
			-		-		-		 	 
			-		-		-		 	 

and on the Turbing model (Compact Industrial or Ev)	
-10 01 11 11 11 11 11 11 11 10 11 10 11 10 10	

### Model: (for Ex)

• TFIX-1-V2

•

- TFIX-2-V2
- TFIX-3-V2

#### · Accessory: (for Extended & Ex)

- Handset
- Handset w/ Offhook
- Handset w/ Offhook (Normally Closed)
- Handheld Microphone
- Headset
- Headset w/ Auto detect

#### 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

## 6.3 Account/Call Settings:

Select SIP Configuration > Account / Call

ount / Call	Account Settings			
idio	Description	Configuration		
	Name:	TCIS-2		
rect Access Keys	Number (SIP ID):	101		
elays / Outputs	Server Domain (SIP):	10.9.5.101		
	Backup Domain (SIP):			
ime	Backup Domain 2 (SIP):			
Audio Messages	Registration Method:	Parallel V		
ertificates	Authentication User Name:	101		
	Authentication Password:			
	Register Interval:	600	(min. 60 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	
	RTP Encryption:	disabled V		
	SRTP Crypto Type:	AES_CM_128_HMAC_SHA1_80 V		
	Use Unencrypted SRTCP:			
	Verify TLS hostname:			
	TLS Private Key:	turbine_server_sha	256.key V	

Call Settings

SAVE

Description	Configuration	
Enable Auto Answer:	8	

#### Account Settings:

- Name
  - Enter a name that will be shown on the display at the remote party.

#### • 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. 10.6.201.10.

#### Backup Domain (SIP)

• This is the secondary (or fallback) 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.

- Backup Domain 2 (SIP)
  - This is the tertiary SIP domain used as backup in case the primary and secondary domains fail.

#### **Registration Method**

- Parallel: The station will register to the primary, secondary and tertiary SIP server domains at the same time. • Serial: The station will always try to register to the first available SIP server domain, starting with the primary domain, followed by the secondary and the tertiary domains.
- Top-Down: This is the same as for Serial registration except that it will always revert to the SIP domains with the higher priority if available. In decreasing order of priority, the primary SIP domain has the highest priority, followed by the secondary domain and the tertiary domain.
- Cisco: This is for the Cisco environment only with Publisher/Subscriber/SRST setup, where the station is normally registered to Publisher. The Subscriber should be assigned as the secondary SIP domain, and SRST gateway as the tertiary domain.

#### Authentication User Name •

 This is the authentication user name used to register the station to the SIP server. This is required only 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
  - This parameter 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
- Enter the port number used for SIP on the outbound proxy server. The default port number is 5060.

#### Outbound Backup Proxy 1&2 [optional]

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

#### Outbound Transport

Possible configuration options: UDP, TCP, TLS. For TCP and TLS options, it is necessary to set Outbound Proxy to have the same value as Server Domain.

- SIP Scheme. Possible configuration options: sip, sips. Using sips forces all proxies to also use TLS. RTP Encryption. Possible configuration options: disabled, srtp encryption
- SRTP Crypto Type. Possible configuration options: AES CM 128 HMAC SHA1 80 / AES CM 128 HMAC SHA1 32
- Use Unencrypted SRTCP. Check the box to enable unecrypted SRTCP
- TLS Private Key, Possible configuration options: turbine server sha256.key / turbine server sha1.key

#### Call Settings:

- Enable Auto Answer. If enabled, the station will answer automatically, and if not, the station will ring.
- After entering the desired values, click SAVE followed by REBOOT to enable the SIP settings.

# 6.4 Audio Settings

Select SIP Configuration > Audio

Main SIP Configuration	Station Administration		
▹ Account / Call	Audio Settings		
- Audio	Description	Configuration	
<ul> <li>Audio</li> </ul>	Speaker Volume:	2 -	
Direct Access Keys	Volume Override Level:	9 -	Sets the volume during volume override. Volume and handset override
		0.	happens during Emergency Group calls. 🕕
Relays / Outputs	Automatic Gain Control (AGC):		Automatic Gain Control. If speech level and environmental noise are very unstable it may be turned on.
⇒ Time	Automatic Volume Control (AVC):		Volume depends on noise level
Audio Messages	Active Noise Cancellation (ANC)	Ø	
▹ Certificates			SAVE

- Speaker Volume •
  - Set speaker volume between 0 and 5
- Volume Override Level •
  - Select the volume override level in the range 0 to 8 from the drop-down menu
- Automatic Gain Control (AGC) •
  - If speech level and environmental noise are very unstable, it may be enabled
- Automatic Volume Control (AVC) •

• When enabled volume control is adjusted according to the noise level. Disabled by default.

- Active Noise Cancellation (ANC) ٠ • When enabled, most of the background noise will be filtered out of the microphone signal. Enabled by default.
- After entering all the desired values, click SAVE •

# 6.5 Direct Access Key & Ringlist Settings

To configure DAK keys, Inputs and Ring List functions:

Select SIP Configuration > Direct Access Keys

SIP Con

Account / Call Audio Direct Access Relays / Output Time Audio Messag Certificates

Main

Function
Idle: Call To
Call: Do Nothing 🔻
Idle: Call To
Call: Do Nothing 🔻
Idle: Call To
Call: Do Nothing 🔻
Idle: Call To
Call: Do Nothing 🔹
Idle: Call To
Call: Do Nothing 🔻
Idle: Call To
Call: Do Nothing

#### **Ringlist Settings**

	Ringlist 1	With Previous	Ringlist 2	With Previous	Ringlist 3	With Previous
Value 1	102					
Value 2	103					
Value 3						
Value 4						
Value 5						
Value 6						
Value 7					3	



under Main Settings.

For more information on the configuration of DAK keys, Inputs and Ring List functions, see wiki.zenitel.com/wiki/Direct Access\_Key\_%26\_Ringlist\_Settings\_(SIP)

	No Ringlist V	Normal V
	No Kinghot V	Norma
	No Ringlist 🔻	T
_		
	No Ringlist V	•
	No Ringlist 🔻	<b>T</b>
_		
	No Ringlist V	T
	No Ringlist V	<b>T</b>
-		
	and the second	· · · · · · · · · · · · · · · · · · ·

#### SAVE

### Note! The availability of these parameters and the number of keys depend on the Turbine station type selected

# 6.6 Relay/Output Settings

Main

Select SIP Configuration > Relays / Outputs

ount / Call	Relay Settings					
io	Choose Relay To Configure:	Relay 1	•			
ect Access Keys	Relav 1 Settings					
ays / Outputs	Description			Config	uration	
ne	Remote Digit For Relay On:		[	-		•
d:- M	Remote Digit For Relay Off:			-		•
lio Messages	Remote Digit For Relay Slow Flash :			-		•
rtificates	Remote Digit For Relay Fast Flash:			-		•
	Remote Digit For Relay Toggle:			-		•
	Remote Digit For Timed Relay On:			6		•
	Timed Relay Duration:			3	seconds	
	Outgoing Ringing:			-	۲	
	Incoming Ringing:			-	۲	
	Outgoing Call:			-	•	
	Incoming Call:			-	•	
	Group Call (Edge mode only):				۲	
	Idle:		[	Off	۲	
	Error (Not Registered):				•	

#### **Relay Settings:**

Select Relay 1, Relay 2, Relay 3 (Relay Board) or Relay 4 (Relay Board) from the drop-down box. •

### There are 4 different states: On, Off, Slow Flash, Fast Flash

- Select the remote digit (DTMF signal) under Configuration. Selecting "-" symbol means that the behaviour does not change when the event occurs. The relay state can change either with a remote digit (DTMF signal) during a call, or if the station state changes.
- Remote Digit For Relay On / Off / Slow Flash / Fast Flash • Select the digit (DTMF signal) that should change the relay to the On/Off/Slow Flash/Fast Flash state
- Remote Digit For Relay Toggle · Select the digit to toggle the relay on/off every time the specified DTMF digit is received
- **Remote Digit For Relay Timed Relay On** 
  - Select the digit that should change the relay to the on state for a number of seconds set in Timed Relay Duration. This state is normally used for the Door Opening function.
- ٠ Outgoing Ringing / Incoming Ringing. Operate the relay while the station is ringing due to an incoming or outgoing call. The ringing state occurs from the start of the call until the call is accepted.
- Outgoing Call / Incoming Call. Operate the relay while the station is in conversation. The call state occurs from ٠ acceptance of the call until the call ends.
  - **Turbine Stations Technical Manual**

- Group Call (IC-Edge mode only). Operate the relay when the station receives a Group Call ٠
- Idle
- Select the state the relay should change to when nothing else is occuring on the station.
- Error (Not Registered)
  - · Select the state the relay should change to when the station is off-line.

## 6.7 Time Settings

Select SIP Configuration > Time

Main SIP Configuration	n Station Administration
Account / Call	Apr 28 2020 13:33
▶ Audio	Time Settings
	Description
Polove / Outpute	NTP server:
F Relays / Outputs	Select Region:
✓ Time	Select Zone:
Audio Messages	Enter Manual Date & Time:
<ul> <li>Certificates</li> </ul>	

Note: It may take a couple of minutes before the NTP client updates time at startup or after hostname change.

#### **Time Settings:**

You can enable NTP (Network Time Protocol), select the time zone and enter the date and time manually.

# 6.8 Audio Messages

Prerecorded audio messages can be uploaded to a Turbine which can be triggered by various events occurring on the station. The audio message playback can be triggered by 3 main events: DTMF, Call, Relay.

Select SIP Configuration > Audio Messages



Under Upload Media:

- Click Choose File to browse to the desired audio file
- Click UPLOAD

The file will now appear in the list of audio files under Media. After the upload, the amount of space left of the maximum of 20 MB is displayed. Turbine Stations Technical Manual

Coning	urat	ion			
•					
pool.n	ntp.c	org			
Europ	e	۲			
Lond	on	,			
2015	01	01	01	00	yyyy-MM-dd-hh-mm

LOAD			

# 6.9 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	
⊢ Infe	ormation	Commands	
► Ma	in Settings	Description	Action
P INIG	in octungs	Full reboot	REBOOT
▼ Rec	overy	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 •

ain SIP Configuration	Station Administration	
Information	Commands	
▶ Main Settings	Description	Action
in an acturing a	Full reboot	REBOOT
<ul> <li>Recovery</li> </ul>	Partial reboot	REBOOT
	Factory reset	FACTORY RESET
	Factory reset with DHCP	FACTORY RESET
	Preferences	
	Description	Configuration
	Advanced configuration mode	SAVE SAVE

- 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:

Main	SIP Configuration	Station Administration	A			
► Acc	count / Call	Digital Outputs - Scrip	ots			
→ Au	dio	Choose script slot to configure:				
► DA	VC	Assign a Laber				
→ Dir	ect Access Keys	Enter script in selected slot				
▶ Rel	ays / Outputs					
→ Tin	ne					
► I/O						
→ Fro	ntboard Mapping					
► RTS	SP					
👻 Scr	ipt					
→ Scr	ript Events					
→ Scr	ript Upload					
→ Au	dio Messages					
→ Mu	lticast Paging					
→ Cer	rtificates					

For more information on Script settings, see <a href="wiki.zenitel.com/wiki/Virtual\_I/O\_(SIP">wiki.zenitel.com/wiki/Virtual\_I/O\_(SIP)</a>

# 6.10 I/O Settings

• Select SIP Configuration > I/O

Main	SIP Configuration	Station Administration	Advanced SIP	Advanced Network	κ.	
→ Ac	count / Call	I/O Settings				
. A.,	dio	Description	t.	Configuration		
► Au	010	Use Inputs As Key Matrix		Off	•	
► DA	VC	Fast Blink Pattern		111000111000111000	0111000	1 = on, 0 = off, 100 ms Interval
→ Dir	ect Access Keys	Slow Blink Pattern		1111111111111000000	000000	1 = on, 0 = off, 100 ms interval
		I/O Pin 1:		Input 1 🔻		
▶ Rel	ays / Outputs	I/O Pin 2:		Input 2 🔻		
⊢ Tin	ne	I/O Pin 3:		Input 3 🔻		
		I/O Pin 4:		Input 4 🔻		
▼ 1/0		I/O Pin 5:		Input 5 🔻		
→ Fro	ntboard Mapping	I/O Pin 6:		Input 6 🔻		
→ RT	SP			SAVE	REBOOT	

dvanced SIP	Advanced Network
Slot 1 🔻	
	/
	SAVE

59

- Use Inputs As Key Matrix. If there is a need for more than the 6 inputs available, it is possible to combine inputs as in a keyboard matrix, and in this way create more DAKs. The options are:
  - 2 Inputs (input 1-2, DAK 1-3)
  - 3 Inputs (input 1-3, DAK 1-7)
  - 4 Inputs (input 1-4, DAK 1-15)
- Fast Blink Pattern
  - Set the fast blink pattern for the LEDs
- Slow Blink Pattern
  - Set the slow blink pattern for the LEDs
  - Select either Input or Output options from the drop-down box for I/O Pins 1 to 6

For further information, see wiki.zenitel.com/wiki/I/O Settings (SIP)

# 7 Common Advanced Network Settings

Note! The configuration settings described in this section are not mandatory.

Advanced Network settings are available in ICX-AlphaCom, Edge Controller and SIP mode.

# 7.1 SNMP Settings

SNMP (Simple Network Management Protocol) is a protocol for centralizing the management of devices in IP networks.

Select Advanced Network > SNMP •

> · 802 → Fire

SNMP Settings		
Description	Configuration	Information
Access	Disabled	Access mode
SIMP v1		Enable SNMP v1
SIMP v2c	<u></u>	Enable SNMP v2c
Community string:	public	For v1 and v2c only
Allowed Network:	0.0.0.0 / 0	Example: 192.168.1.0/24
Name:		
Location:		
Contact:		
SNMP Trap and Inf	forms Settings	
Description	Configuration	Information
Trap receiver		Disable Traps by setting this field empty
Inform receiver:		Disable informs by setting this field empty
Enable SNMP Trap	15	
Enable SNMP Trap	IS Endded	Information
Enable SNMP Trap Description	is Ended	Information
Enable SNMP Trap Description P-Station Statistic Registration Serverschift	S Enddel IIIIIIIIIIIIIIIIIIIIIIIIIIIIIIIIIII	Information The device has started Device has successfully registered with a SIP
Enable SNMP Trap Description P-Station Started: Registration Started	s	Information The device has started Device has successfully registered with a SIP server Provide Informations with a OR assess
Enable SNMP Trap Decention IP-Station Statest Registration Statestuff Registration Failed Call Consector	s	Information The device has started Device has successfully registered with a SIP server Device failed to register with a SIP server Cell has reconcerted excessfully
Enable SNMP Trap Decentor P-Sarion Sarret: Registration Sarreschift Registration Fallet: Call Connection		Information The device has started Device has successfully registered with a SIP series Device failed to register with a SIP server Call has connected successfully Call has connected successfully
Enable SNMP Trap Deceptor P-Station Survet Weydration Survestuff Registration Failet Call Convector Call Convector		Information The device has started Device has successfully registered with a SIP server Device failed to register with a SIP server Call has connect successfully Call has failed to context Device failed to context
Enable SNMP Trap Decentor P-Sation Sarriet Registration Sarriet Call Convect Failed Call Convect Failed Call Convect Failed		Information The device has started Device has successfully registered with a SIP serier Device failed to register with a SIP server Call has connect successfully Call has failed to connect Call has disconnected DKK was pressed. Tog description will provide
Enable SNMP Trap Deceptor P-Sation Sarrest Registration Sarrest Call Connector Call Connect Failed Call Connect Failed Call Connect Failed		Information The device has started Device has successfully registered with a SIP server Device alled to register with a SIP server Call has connected successfully Call has failed to connect Call has failed to connect Call has device the device of the server DRA was present. Top description will provide DRA was present. Top description will provide
Enable SNMP Trap Deception IP-Station Startest Registration Startest Call Connected Call Connect Call Connect Call Universe Dat Researd.		Information The device has started Device has successfully registered with a SIP server Device failed to register with a SIP server Call has failed to nonsect Call has failed to nonsect Call has deviced for the device of the server DAT has deviced for the deviced of the server DAT has deviced for the server deviced for the serve
Enable SNMP Trap Decorton IP-Staton States Registration States Registration States Call Connector Call Connector Call Connector Call Connector Call Connector Call Connector Call Connector Call Connector Call Connector		Information The device has started Device has successfully registered with a SIP server Device failed to register with a SIP server Call has connected successfully Call has failed to consect Call has disconnected DK was pressed. Trap description will provide more the Jobur CBAK number. Bally was activate. Trap description will provide more in the Jobur CBAK number.
Enable SNMP Trap Deception P-Station Surrest: Registration Surresthilt Registration Failed: Call Convected Call Convected Call Convected Call Convected Dath Research Dath Research Relay Deactivated:		Information The device has started Device has successfully registered with a SIP server Device failed to register with a SIP server Call has connected successfully Call has failed to nonexted DMK was pressed. Trap description will provid more time about DMK number. Relay was settated. Trap description will provide more find about number. Relay was settated. Trap description will provide more find about number. Relay was settated. Trap description will provide more find about number.
Enable SNMP Trap Deceptor P-Station Surrect Registration Surrect Call Connected Call Connect Failed Call Disconnect Failed Call Disconnect Dak Presset Dak Rebused Reky Actuator Reky Actuator		Information     The device has started     Device has started     Device has successfully negistrand with a SIP     serier     Device failed to registrar with a SIP serier     Call has connected successfully     Call has failed to nonset     Call has failed to nonset     Call has failed to nonset     Call has reconsected     DMK was passed. Top description will provid     more into door number.     DMK was subseted. Top description will     provide more info door nervine number.     Relay was settinged. Top description will     provide more info door nervine number.     Relay was setting. Top description will     provide more info door nerving number.     Relay was been info door nerving number.     Ruton was purshed for more that 10 seconds     without bing released.
Enable SNMP Trap Pescepton P-Sation Sarrest Registration Fallot: Call Convert Fallot Call Convert Fallot	IS	Information The device has started Device has started Device has successfully registered with a SIP server Device Salad to register with a SIP server Call has characteristic successfully Call has failed to connect OH has disconnected OH was presed. Top description will provide more info about DMK number. Builty was serviced and DMK number. Relative as extinated. Top description will provide more info about DMK number. Relative as extinated. This description will provide more info about DMK number. Relative as serviced and the description will provide more info about traing runnber. Relative associated about traing number. Relative associated and about traing number. Builty was provided more than 10 accords without being metased Sound text tained
Enable SNMP Trap Description P-Sauton Saurest Registration Saurest Call Converts Call Converts Call Convert Call Call Call Convert Call Call Call Call Call Call Call Call		Information     The device has started     Device has successfully registered with a SIP     server     Device failed to register with a SIP server     Call has failed to nonnect     Call has failed to nonnect     Call has failed to nonnect     DA vice server failed to register with a SIP server     Davice failed to register with a SIP server     Call has failed to nonnect     DA vice server failed to nonnect     Sund test failed     Automatically setting sound pressure level failed
Enable SNMP Trap Description IP-Scalon Startest Registration Startest Registration Failed Call Converter Call Converter		Information The device has started Device has started Device has successfully registered with a SIP server Device failed to register with a SIP server Call has connected successfully Call has failed to nonect Call has disconnected DK was pressed. Top description will provide more into about DK humbler. Beily was activated. Top description will provide more into about DK humbler. Relay was activated. Top description will provide more into about DK humbler. Relay was activated. Top description will provide more into about DK humbler. Relay was activated. Top description will provide more into about DK humbler. Relay was activated. Top description will provide more into about DK humbler. Relay was activated. Top description will provide more into about DK humbler. Relay was deviced from the has 10 seconds without being released Sound test has successful Sound test was successful
Enable SNMP Trap Perceptor Pristation Surret Registration Surret Call Connection Call Connection Call Connect Call Connect	S	Information The device has started Device has started Device has successfully registered with a SIP server Device failed to register with a SIP server Call has connected successfully Call has failed to consect Call has disconnected DK was pressed. Top description will provide more tim door tumber. UAK was released. Trap description will provide more in the door taky number. Reley was devised. Trap description will provide more in the door taky number. Reley was devised. Trap description will provide more in the door taky number. Reley was devised. Trap description will provide more in the door taky number. Reley was devised. Trap description will provide more in the door taky number. Reley was devised for more than 10 seconds without being released Sound test takas Automatically setting cound pressure level fail Sound test was successful Input betten was pressed. Top description will provide the was pressed. Top description will provide the was been setting
Enable SNMP Trap Deceptor P-Station Surrect Registration Surrect Call Connected Call Connect Failed Call Decented Call Decented	S	Information The device has started Device has storted Device has successfully registered with a SIP server Device Allar to register with a SIP server Call has connected successfully Call has failed to register with a SIP server Call has failed to register with a SIP server Call has failed to register with a SIP server Call has failed to accessfully Call has failed to ac
Enable SMMP Trap Percepton Percepton Registration Sarroschaft Call Connected Call Connect Failed Call Conn	S	Information     The device has scared     Device fails has scared     Device failed to register with a SIP server     Device failed to register with a SIP server     Call has connected uncreashify     Call has failed to connect     Data was presed. Tag description will provi     more into about DIAN number.     DAA was released. Tag description will provi     more info about DIAN number.     DAA was released. Tag description will provi     more info about DIAN number.     DAA was released. Tag description will provi     more info about Tally number.     Belly was activated. Tag description will provi     more info about Tally number.     Belly was activated. Tag description will provi     more info about Tally number.     Belly was activated. Tag description will     provide more info about tally number.     Buttors was presed. Tag description will     provide more info about pressure level fa     Sound test was successful     Input button was presed. Tag description will     provide more info about pressure level fa     Sound test maid     Input button was presed. Tag description will     provide more info about pressure level fa     Sound test maid     Input button was presed. Tag description will     provide more info about pressure level fa     Sound test maid

SAVE ALL

#### SNMP Settings

- Access Options are Read or Disable SNMP
- Enable SNMP v1
  - This enables reading of the MIB using SNMP version 1.
- Enable SNMP v2c
  - This enables reading of the MIB using SNMP version 2c.
- Community string ٠
  - Enter a text string used as a password for authentication.
- Allowed Network
  - This is used, together with the network mask, to determine the allowed network for reading the MIB on the station.
  - a network mask of 24, any station with an IP address in the range 10.5.2.0 to 10.5.2.255 · can access the MIB.
- Name
  - SNMP name string (sent as sysName OID in traps)
- Location
  - SNMP Location string
- Contact SNMP Contact string

### SNMP Trap and Informs Settings:

- Trap receiver
  - Enter the IP address of the server receiving SNMP traps. This is disabled if the field is left empty.
- Inform receiver •
  - Enter the IP address of the server receiving SNMP informs. This is disabled if the field is left empty.

#### **SNMP Custom Configuration File:**

- Upload the custom snmpd.conf file that will be used when running SNMP. The uploaded file will be preserved after the station has been upgraded to a newer firmware version.
- Enable SNMP Traps
- **IP-Station Started** •
  - If enabled, the station will send an SNMP trap when the station application is started.
- **Registration Successful** 
  - If enabled, the station will send an SNMP trap when successfully registered in the SIP domain.
- **Registration Failed** •
  - If enabled, the station will send an SNMP trap if registration in the SIP domain failed.
- Call Connected •
  - If enabled, the station will send an SNMP trap when a call is connected.

• The IP address is entered in regular dot notation, e.g. 10.5.2.100. For example with an allowed network 10.5.2.0 and

#### Call Disconnect

• If enabled, the station will send an SNMP trap when a call is disconnected.

#### Dak Pressed •

If enabled, the station will send an SNMP trap when a DAK is pressed.

Dak Released ٠

• If enabled, the station will send an SNMP trap when a DAK is released.

Relay Activated

· If enabled, the station will send an SNMP trap when a relay is activated.

- **Relay Deactivated** 
  - If enabled, the station will send an SNMP trap when a relay is deactivated.
- Button Hanging
  - If enabled, the station will send an SNMP trap when a button is hanging.
- Sound Test Failed If enabled, the station will send an SNMP trap when a sound test has failed.
- Sound Test Error • • If enabled, the station will send an SNMP trap when there is a sound test error.
- Sound Test Success • If enabled, the station will send an SNMP trap when a sound test is successful.
- Input Button Pressed
  - If enabled, the station will send an SNMP trap when an input is activated.
- Input Button Released . If enabled, the station will send an SNMP trap when an input is deactivated.
- Software Fault If enabled, the station will send an SNMP trap when software daemon has stopped working.

# 7.2 Network Access Control

IEEE 802.1X is an IEEE Standard for Port-based Network Access Control (PNAC). It provides an authentication mechanism to devices wishing to attach to a LAN, either establishing a point-to-point connection or preventing it if authentication fails.

802.1X Network Access Control is configured from the IP station web interface.

Select Advanced Network > 802.1X •

Station Adm	inistration Advanced ICX-AlphaCom	Advanced Network
мр	802.1X Settings	
.1X	Choose authentication method:	
wall	EAP-MOSTARY2 EAP-MD5 EAP-TTLS with PAP PEAP with MSCHAPV2 EAP-TLS	
	Description	Configuration
	802.1X Status:	DISABLED V
	Username:	Username
	Password:	
	Fast Re-Authentication:	8
	Get new DHCP on success:	8

Note: Copy paste all of the content in the certificate file(s). The certificates are not visible in the text area after they are uploaded. The certificates must be a X509 certificates and be in text format. Uploading a certificate for TTLS/PEAP is not needed if "Verify server with certificate" is off. The radio-button list lets the user choose the authentication method to configure.

The different authentication methods are:

- EAP-MSCHAPV2
- EAP-MD5
- EAP-TTLS with PAP
- PEAP with MSCHAPV2
- EAP-TLS
- EAP-MSCHAPV2 and EAP-MD5 will encrypt the password.
- EAP-TTLS with PAP and PEAP with MSCHAPV2 will encrypt both the Username and Password. The parameters to configure depend on the authentication method: • 802.1X status: Enable or disable 802.1X.
  - Username: The user name that identifies a station.
  - Password: The password associated with the user name.
  - Fake username: The fake user name sent outside of the encrypted tunnel with EAP-TTLS with PAP and PEAP with MSCHAPV2. The user name is encrypted.
- Verify server with certificate: [EAP-TTLS with PAP, PEAP with MSCHAPV2 and EAP-TLS only] Specifies that the client verifies that server certificates presented to the client have the correct signatures, have not expired, and were issued by a trusted root certification authority (CA). It's enabled by default.
- Fast Re-Authentication: Re-authenticates the clients connected to 802.1x-enabled interfaces.
- Get new DHCP on success: If 802.1X authentication is successful, the station will restart its DHCP client.
- Insert CA Certificate: [EAP-TTLS with PAP, PEAP with MSCHAPV2 and EAP-TLS only] Uploads certificate (public key) used by authentication server. This is not required if the Verify server with certificate option is disabled.
- Insert User Public Certificate: [EAP-TLS only] Uploads certificate (public key) used by authentication client (user).
- Insert User Private Key: [EAP-TLS only] Uploads the private key that is paired with the user public certificate.
- Click **SAVE** to save the current settings •
- Click REBOOT •

The new 802.1X settings will only come into effect after the reboot.

# 7.3 Firewall Settings

Select Advanced Network > Firewall from the menu •

SNMP	Firewall Setting	S		
▶ 802.1X	Name	Protocol	Port	Enabled
<ul> <li>Firewall</li> </ul>	SSH	tcp	22	2
	нттр	tep	80	۲
	HTTPS	tcp	443	
	SIP	tcp	5060	8
	SIPS	tcp	5061	۲
	DIP	tcp	50001	8
	Demo	tep	50010	
	AudioData	udp	5035	
	TFTP Server	udp	69	8
	SNMP	udp	161	۲
	SIP	udp	5060	
	Edge	udp	5062	۲
	mDNS	udp	5353	2
	DIP Multicast	udp	50001	
	Discovery	udp	50002	۲
	ZAP	tcp	50004	
	ZapWeb	tcp	8080	
	VolP	udp	61000 :61250	
	RTSP	tco	554	1

- ٠ **SSH**: Enable or disable SSH on station, disabled by default.
- HTTP: Enable or disable HTTP on station, enabled by default. ٠
- HTTPS: Enable or disable HTTPS on station, enabled by default.
- SIP (tcp): Enable or disable SIP over tcp on station, enabled by default.
- SIPS: Enable or disable SIPS on station, enabled by default. ٠
- DIP: Enable or disable DIP on station, enabled by default.
- Demo: Enable or disable Demo on station, disabled by default.
- AudioData: Enable or disable AudioData on station, disabled by default.
- TFTP Server: Enable or disable TFTP Server on station, enabled by default. ٠
- SNMP: Enable or disable SNMP on station, enabled by default.
- SIP (udp): Enable or disable SIP over udp on station, enabled by default. ٠
- Edge: Enable or disable IC-EDGE on station, enabled by default. ٠
- mDNS: Enable or disable mDNS on station, enabled by default.
- DIP Multicast: Enable or disable DIP Multicast on station, enabled by default.
- Discovery: Enable or disable Discovery on station, enabled by default.
- Zap: Enable or disable ZAP on station, enabled by default.
- ZapWeb: Enable or disable ZapWeb on station, enabled by default.
- VoIP: Enable or disable VoIP on station, enabled by default. ٠
- RTSP: Enable or disable RTSP on station, enabled by default.

# 8 Station Software Upgrade

This sections describes the software upgrade procedure via the web interface of the station.

Other ways of upgrading the software on the IP station can be found on Zenitel Wiki:

- Uploading the software via AlphaWeb on the AlphaCom server, see wiki.zenitel.com/wiki/AlphaWeb#IP Station Upgrade
- Automatic Software Upgrade in SIP Mode, see wiki.zenitel.com/wiki/Automatic Software Upgrade
- Using the Zenitel Intercom Manager Tool (VS-IMT), see wiki.zenitel.com/wiki/VS-IMT Upgrade Stations

Note! Turbine Ex stations have a built-in Flowire component. Refer to A100K11958 FCDC3 Flowire Configuration Manual when upgrading software for Ex stations.

## **8.1 Prerequiseites**

The upgrade methods require that an TFTP Server is available and that the latest software image files have been downloaded from Zenitel Wiki. During the upgrade process, the IP station will connect to the TFTP Server and download the software. Install the TFTP Server program on your PC.

# 8.2 Upgrade via Station Web Interface

This section pertains to upgrading from version 5.1.x.x to a newer version. For information on upgrading from earlier versions, see wiki.zenitel.com/wiki/Turbine Software Upgrade.

Start the TFTP server program and click Browse to select the folder where the software image files are located

Tftpd64	by Ph. Jour	in				—		×
Current Directory C:\Users\hleong\Desktop\vsft-5.1.3.0					Browse			
Server interfac	Server interfaces 192.168.0.15 Intel(R) Wireless-AC 9260 160MHz 💌					Show Dir		
Tftp Server	Tftp Client	DHCP server S	yslog server	Log viewer				
peer		file	start time	progress	bytes	total	timeo	
Abo	ut		Se	ettings			Help	

#### Log on to the IP Station web interface 1.

### 2. Select Station Administration > Manual Upgrade

Main	Station Administr	ation Adva	nced ICX-AlphaCom	Advanced Network
* 1	Manual Upgrade	Upgrade	using TFTP (.zip	o images)
▹ Change Password		TFTP server:	192.168.0.15	
26		Image file:	vsft-5.1.3.0	
		Start Upgrad	de	

- 3. Enter the IP address of the TFTP server (your PC's IP address)
- Enter the prefix (e.g. vsft-5.1.x.x) to the software image files in the Image file field 4.
- 5. Click Start Upgrade

The station will now try to contact the TFTP server. The upgrade procedure takes about 3 minutes. The process can be monitored by clicking the Log viewer tab in the TFTP server program.

During an upgrade of the station, 802.1X will not be running. Thus if 802.1X reauthentication is enabled and ! is performed during the upgrade, the station may lose contact with the TFTP server (depending on the configuration when 802.1X authentication fails). If the station loses contact with the TFTP server, it will not be upgraded.

## **Station Indication LEDs** 9

# 9.1 LEDs on Front Plate - Compact Stations



#### 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

# 9.1.1 LEDs on Front Plate - ECPIR-3P, INDUSTRIAL & EX STATIONS

## Power LED - Green

Steady light: Power is OK



Steady light: Not used

Call LED - Red

# 9.2 Status LEDs on PCB

# Flashing 2 red + 1 green

# Flashing 1 red + 2 green exchange.

Compact

Flashing 3 green

# 9.3 Ethernet Activity and Speed LEDs

Compact

#### **Green LED**

### Yellow LED





Industrial

Flashing: Not registered/no connection to server

· Steady light: In a conversation/call

· Station has no connection to the AlphaCom server/exchange.

- Station connected but NOT registered in the AlphaCom server/

- Station connected and registered in the AlphaCom server/exchange.

 Steady light: Ethernet connection OK Flashing: Ethernet data traffic • No light: No Ethernet connection

 Steady light: 100 Mbit Ethernet connection No light: 10 Mbit Ethernet connection

# **10** Station Software Upgrade

A Turbine IP Station may have to be reset to its original factory default settings if, for instance, the password to the station web interface is forgotten. The defaults can either be set to Activated DHCP or Static IP.

# 10.1 Reset to Factory Default Settings with Activated DHCP

### To reset:

- While **pressing any button**, power up the station by 1. connecting to a PoE switch.
- 2. Hold the button until the station audio starts counting, and release the button on count 1.
- З. Press and hold the button on count 5 and release on count **0**. - If there is no 0 count, the procedure has failed and you have to start again
- 4. Press the call key, number keys or DAKs to make the station speak its IP address.
- Factory default values ٠
  - Station IP address: (determined by DHCP server)
  - Username: admin
  - Password: alphaadmin

# 10.2 Reset to Factory Default Settings with Static IP

#### To reset:

- While **pressing any button**, power up the station by 1. connecting to a PoE switch.
- 2. Hold the button until the station audio starts counting, and release the button on count 1.
- Press and hold the button on **count 3** and release on **count 0**. З. If there is no 0 count, the procedure has failed and you have to start again
- Press the call key, number keys or DAKs to make the station 4. speak its IP address.
- **Factory default values** ٠
  - Station IP address: 169.254.1.100
  - Username: admin
  - · Password: alphaadmin



1, 2, 3, 4, 5

123+

7890

 $\odot \odot \odot \odot$ 

0

(4) (5) (6)

# A: Compact Board Connectors

# A.1 PCB Front





- 24V EXT **OV FXT** COM NO NC
- Pin 3 COM relay Pin 4 - NO relay Pin 5 - NC relay

**Turbine Stations Technical Manual** 

Pin 1 - 10W Speaker amplifier + Pin 2 - 10W Speaker amplifier -Pin 3 - Electret Microphone + Pin 4 - Electret Microphone -



P1 - RJ45 PoE port for 10/100 Mbit Ethernet connection. The station can be powered from this port if the line supports Power over Ethernet (PoE).

P2 - 5-pin plug-on terminal for external connections. Pin 1/2 24 VDC for external secondary power if PoE is not used. Pin 1 is positive.

P4 Loudspeaker & Electret microphone.

Note! To enable the electret microphone input when using a TKIS-2, the Frontboard



# **A.2 Input Connectors**



- P3 10-pin plug-on terminal for external connections.
  - 5.3V Pin 1 Pin 2

Pin 3

Pin 4

Pin 5

Pin 6

Pin 7

Pin 8

Pin 9

- GND
- Button Input or LED Driver Button Input or LED Driver
- 600 ohm balanced line out +
- Pin 10 600 ohm balanced line out -

# A.3 Output Connectors + 1 Relay



The extra relay for any of the 6 I/O pins is connected as shown:



# A.4 Output Connectors + MRBD Relay Board

Outputs can be used together with the Multi Relay Board (MRBD) to provide up to 6 additional relay contacts.



A.5 PCB Rear



10 • • 9	
8••7	
6••5	
4••3	
2••1	

٠

Pin 1 - Left mic select (Vdd) Pin 2 - Right mic select (GND) Pin 3 - Data Pin 4 - Data Pin 5 - Vdd 3.3V Pin 6 - Vdd 3.3V Pin 7 - CLK Pin 8 - CLK Pin 9 - GND Pin 10 - GND

J6 - 10-pin terminal for digital MEMS Microphone

71

# A.6 Front Board - Front



**B: Extended Board Connectors** 

**B.1 PCB-Front** 



	5.3V	•	1	
	GND		2	
~	I/O 1		3	
/	I/O 2		4	
/	I/O 3		5	
	I/O 4		6	
/	I/O 5		7	
/	I/O 6		י 8	
	Line Out +		0 0	
	Line Out -		-9 10	
-			τu	

- Pin 1 5.3V Pin 2 - GND
- Pin 3 Button Input or LED Driver
- Pin 4 Button Input or LED Driver
- Pin 5 Button Input or LED Driver
- Pin 6 Button Input or LED Driver
- Pin 7 Button Input or LED Driver Pin 8 - Button Input or LED Driver Pin 9 - 600 ohm balanced line out + Pin 10 - 600 ohm balanced line out -

# The WEEE Directive does not legislate that Zenitel, as a 'producer', shall collect 'end of life' WEEE. This 'end of life' WEEE should be recycled appropriately by the owner who should use proper treatment and recycling measures. It should not be disposed to landfill.

Many electrical items that we throw away can be repaired or recycled. Recycling items helps to save our natural finite resources and also reduces the environmental and health risks associated with sending electrical goods to landfill.



Under the WEEE Regulations, all new electrical goods should now be marked with the crossed-out wheeled bin symbol shown.

Goods are marked with this symbol to show that they were produced after 13th August 2005, and should be disposed of separately from normal household waste so that they can be recycled.

### DOC NO. A100K11194

customer.service@zenitel.com

min

Zenitel and its subsidiaries assume no responsibility for any errors that may appear in this publication, or for damages arising from the information therein. Zenitel, Vingtor-Stentofon and Phontech products are developed and marketed by Zenitel. The company's Quality Assurance System is certified to meet the requirements in NS-EN ISO 9001. Zenitel reserves the right to modify designs and alter specifications without notice. ZENITEL PROPRIETARY. This document and its supplementing elements contain Zenitel or third-party information which is proprietary and confidential. Any disclosure, copying, distribution or use is prohibited, if not otherwise explicitly agreed in writing with Zenitel. Any authorized reproduction, in part or in whole, must include this legend. Zenitel - All rights reserved.