DallasDelta
Corporation Pty. Ltd.

Australian Made
User Friendly
Low Cost Ownership
Vandal Resistant
Weather Resistant

VoIP CAM

User's Manual

Voice Over IP Doorstation & Camera

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INDEX

PRODUCT INFORMATION
HELP POINT TELEPHONE
Model DDC Cam VoIP

The Sentry IP Video Door Station is an Ethernet connected telephone which provides voice and video over internet protocol (VoIP) communication technology. Giving you the power of our PSTN base units in a IP format. Using a SIP protocol standard, it provides an easy connection to most VoIP based equipment. The Door Station is a loud speaker, wide angle camera hands-free unit. It is well suited for homes, hotels, hospitals and universities as well as many business and commercial settings and providing remote gate access guarding you against unwanted entry onto your property.

1.1 Basic Features

- Secured door access intercom for business, residential and multi-tenant premises.
- For use at places like hospitals, universities and hotels where several different dialling options may be available.
- Larger commercial premises (including government organisations) will benefit from the increased security the Sentry door station will deliver.
- Manufactured from stainless steel to ensure durability in environments with harsh weather conditions or where there may be a vandalism risk.
- Designed to be flush mounted into an existing wall / facia.
- Programming the unit is done via a web browser or an easy to use keypad on the rear of the unit. The keypad means a field technician can easily re-program the unit without the need for a laptop.
- Automatically dials a set number when the button is pushed.
- Automatically detects when the call has been terminated and will hang-up the unit (using SIP based call progress detection).
- Includes a DTMF activated relay which can be used as a remote gate / door release. This product can be used with up to 2 relays.
- Works with open standards based SIP based call processing systems.

1.2 Operation Modes

- Standard single button unit which will dial a pre-programmed number when the button is pressed. Can be programmed with up to 16 digits.
- Hotline mode.

1.3 Video Features

- Video codec: H.264,H.263,VP8 and MPEG-4
- Image codec: JPEG
- Video call format: VGA/QVGA/CIF/QCIF
- Frame rate selection: 10~30fps
- Adaptive bandwidth adjustment

WARNING

This telephone can not be used for emergency purposes during power failure unless a network connection is guaranteed with Power Over Ethernet (PoE).

1.4 Audio Features

- HD voice: HD codec, HD speaker
- Wideband codec: G.722,SPEEX,OPUS
- Narrowband codec: G.711(A/μ), GSM,AMR-NB,G729 and iLBC
- DTMF: Out-of-band(RFC 2833) and SIP INFO
- Full-duplex hands-free speakerphone
- Voice activity detection
- Acoustic echo cancelling
- Adaptive jitter buffers
- Packet loss concealment

1.5 Network and Security

- Session Initiation Protocol, RFC 3261
  - 8. General User Agent behaviour
  - 9. Cancelling a request
  - 10. Registrations
  - 12. Dialogs
  - 13. Initiating a session
  - 14. Modifying an existing session
  - 15. Terminating a session
  - 17. Transactions
  - 18. Transports
  - 22.4 The Digest Authentication Scheme
- Session Description Protocol, RFC 4566
- An Offer/Answer Model with the Session Description Protocol (SDP), RFC 3264
- An Extension to the Session Initiation Protocol (SIP), RFC 3581 (use of rport parameter)
- Session Initiation Protocol (SIP) INFO Method and Package Framework, RFC 6086
  - 4. The INFO Method
- The Session Initiation Protocol (SIP) Refer Method, RFC 3515
  - 2. The REFER Method
- Session Description Protocol (SDP) Security Descriptions for Media Streams, RFC 4568
- Session Initiation Protocol (SIP)-Specific Event Notification, RFC 3265
- A Presence Event Package for the Session Initiation Protocol (SIP), RFC 3856
- Session Initiation Protocol (SIP) Extension for Event State Publication, RFC 3903 (Creation of PUBLISH requests)
- SIP/UDP, SIP/TCP, SIP/TLS
- Session Traversal Utilities for NAT (STUN), RFC 5389 (Basic procedures)
- RTP: A Transport Protocol for Real-Time Applications, RFC 3550
  - 5. RTP Data Transfer Protocol
  - 6.4 Sender and Receiver Reports
  - 6.5 SDES: Source Description RTCP Packet
  - 6.6 BYE: Goodbye RTCP Packet
  - 6.7 APP: Application-Defined RTCP Packet

- RTP Profile for Audio and Video Conference with Minimal Control, RFC 3551
- Symmetric RTP / RTP Control Protocol (RTCP), RFC 4961
- Secure Real Time Transport Protocol (SRTP, RFC 3711)
- ZRTP, RFC 6189
- ICE, RFC 5245 & RFC 6336
- IP assignment: static/DHCP
- HTTP/HTTPS web server
- SysLog relayed on events & at set intervals
- Time and date synchronization
- QoS: 802.1 tagging (VLAN) , and DSCP

2.1 Configuration

The unit can be configured either through a web interface as described in the Web Interface subsection or by pushing a combination of the keypad buttons. Most of the settings are configured through an integrated administration web server.

2.1.1 Keypad Configuration

The keypad on the rear of the unit provides the installer access to:

- Listen to the current IP address
- Reset the unit to the factory default settings
- Reboot the unit.

2.1.1.1 Get current IP address

During standby, press '**C**' on the rear keypad to retrieve the current IP address.

2.2.1.2 Factory default settings

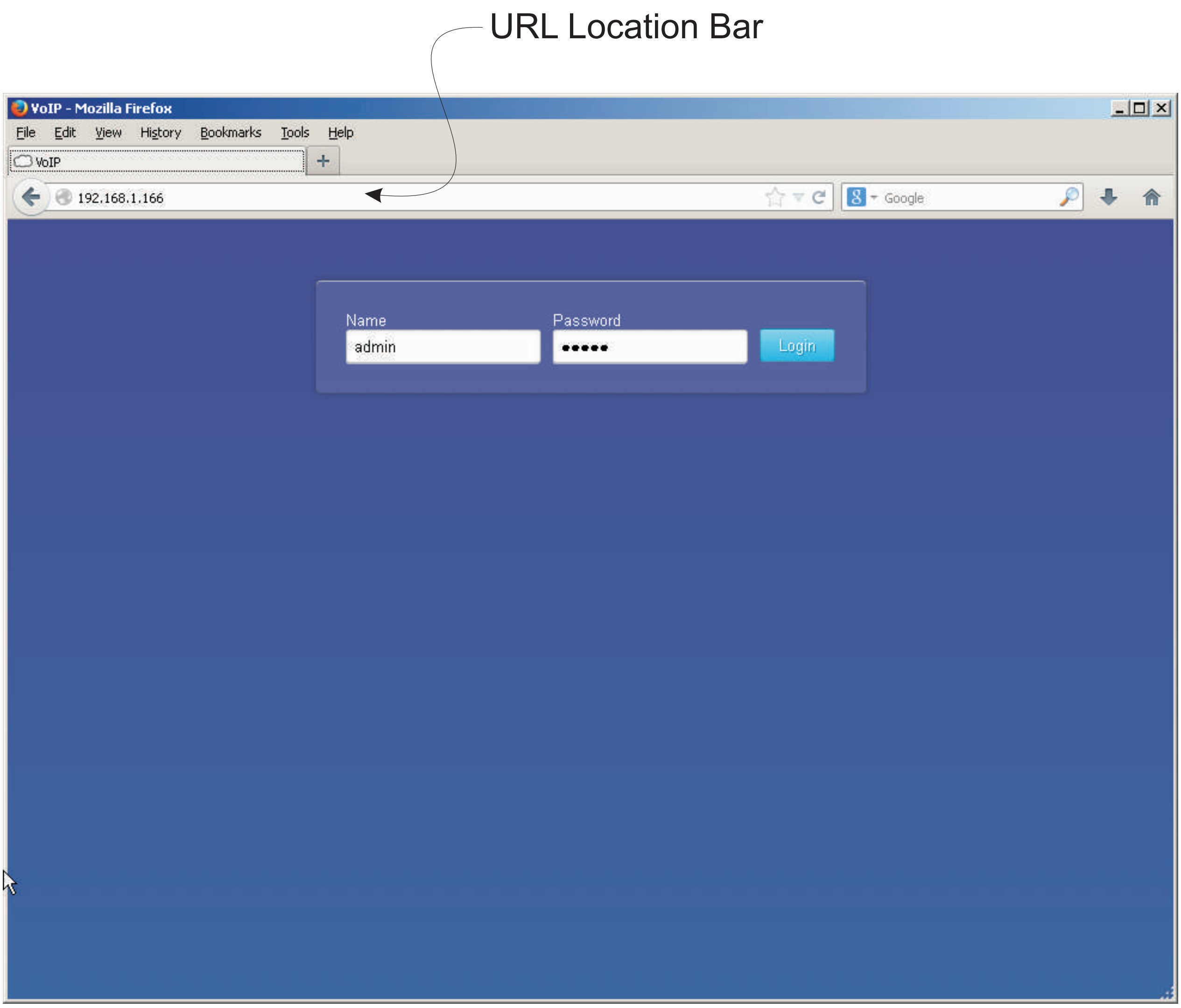
To default the unit to its factory settings, connect the power (via the power jack or Ethernet PoE lead). Wait about 20 seconds (You will hear a start-up tone).
Then press ***#*#7*2**, this will reboot the unit and will set address mode to Static, the IP address to **192.168.1.100**, the subnet mask to **255.255.255.0** and the administrator(**admin**) password to **admin**.

2.2.1.3 Reboot device

This function is used for device restarting in case of changes of some configuration parameters, namely network settings, administration web interface settings and some SIP configuration parameters. Any change of the settings will not be saved permanently on the device until the unit is rebooted.
To reboot the device press ***#*#7*1**.

3.1 Web Interface

The Sentry IP Video Door Station is configured over an integrated administration web server, which can currently only be controlled or changed using Google Chrome web browser.
The DDC VoIP CAM door station is programmed by logging into the phones web page.
Use Google Chrome, enter the phones IP address into the “URL Location Bar” at the top of the screen.
If you don’t know the IP of the phone, press the “C” key on the rear keypad of the unit. The unit’s voice prompt will tell you the current IP address of the unit. (The default factory IP address is 192.168.1.100)
To log in to the programming page type in the name and password and click the Login button, (units are generally delivered with the default username “admin” and password “admin”).



3.1.1 Login

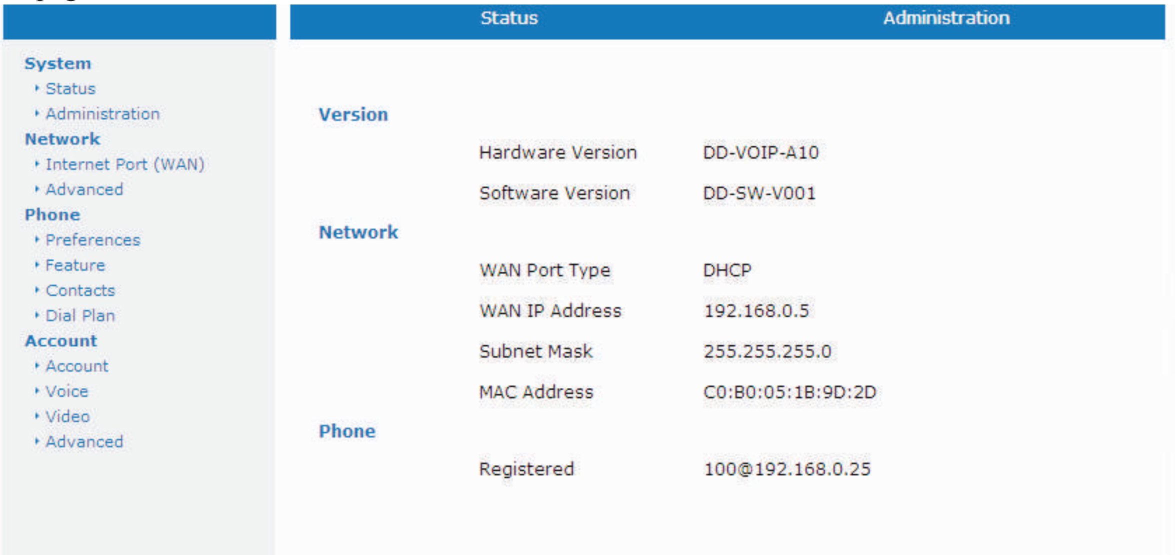
In the web browser enter the Sentry IP Video Door Station. Subsequently, a login screen will be displayed. The default login username and password are as follows:

Name: admin
Password: admin

Note: Currently the VoIP CAM only supports the three default codecs as listed in the Enabled CODECS.

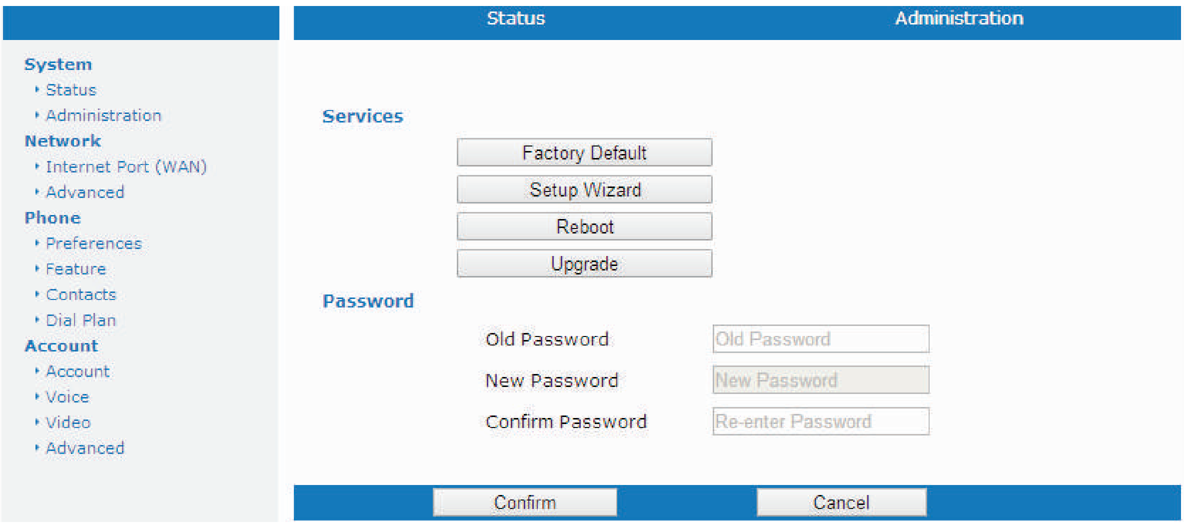
3.1.2 Status

In this subsection you will find the basic information on the respective Sentry system. The phone status is displayed on the first page of the web user interface.



Field	Description
Hardware Version	The Sentry hardware version
Software Version	The Sentry software version
WAN Port Type	DHCP status – displays the mode of obtaining the IP address : From the DHCP server or manual STATIC
WAN IP Address	The current IP address of Sentry IP.
Subnet Mask	The current subnet mask
MAC Address	The Ethernet interface address
Registered	Describe registration status the current Sentry IP -to-SIP proxy registration status: - In progress – registration in progress. - Registered – Sentry IP is registered to the SIP proxy. - Not registered – Sentry IP is not registered to the SIP proxy.

3.1.3 Administration

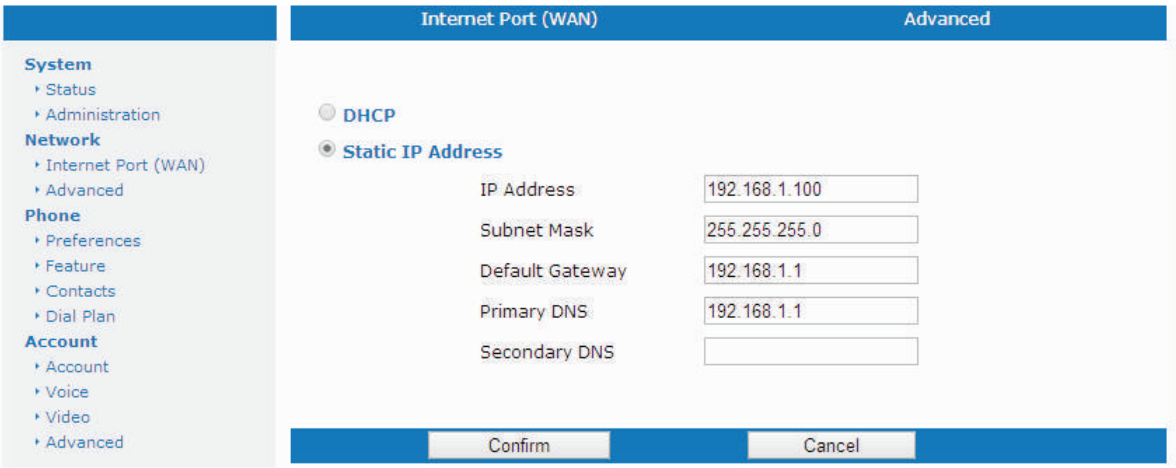


Field	Description
Factory Default	This will set the default factory settings and reboot the unit it will set address mode to Static, the IP address to 192.168.1.100, the subnet mask to 255.255.255.0 and the administrator (admin) password to admin.
Setup Wizard	N/A
Reboot	Reboot the unit and store the settings permanently.
Upgrade	N/A
Old Password	Enter the old password.
New Password	Enter the new password.
Confirm Password	Enter the new password again to confirm.

This page has intentionally been left blank.

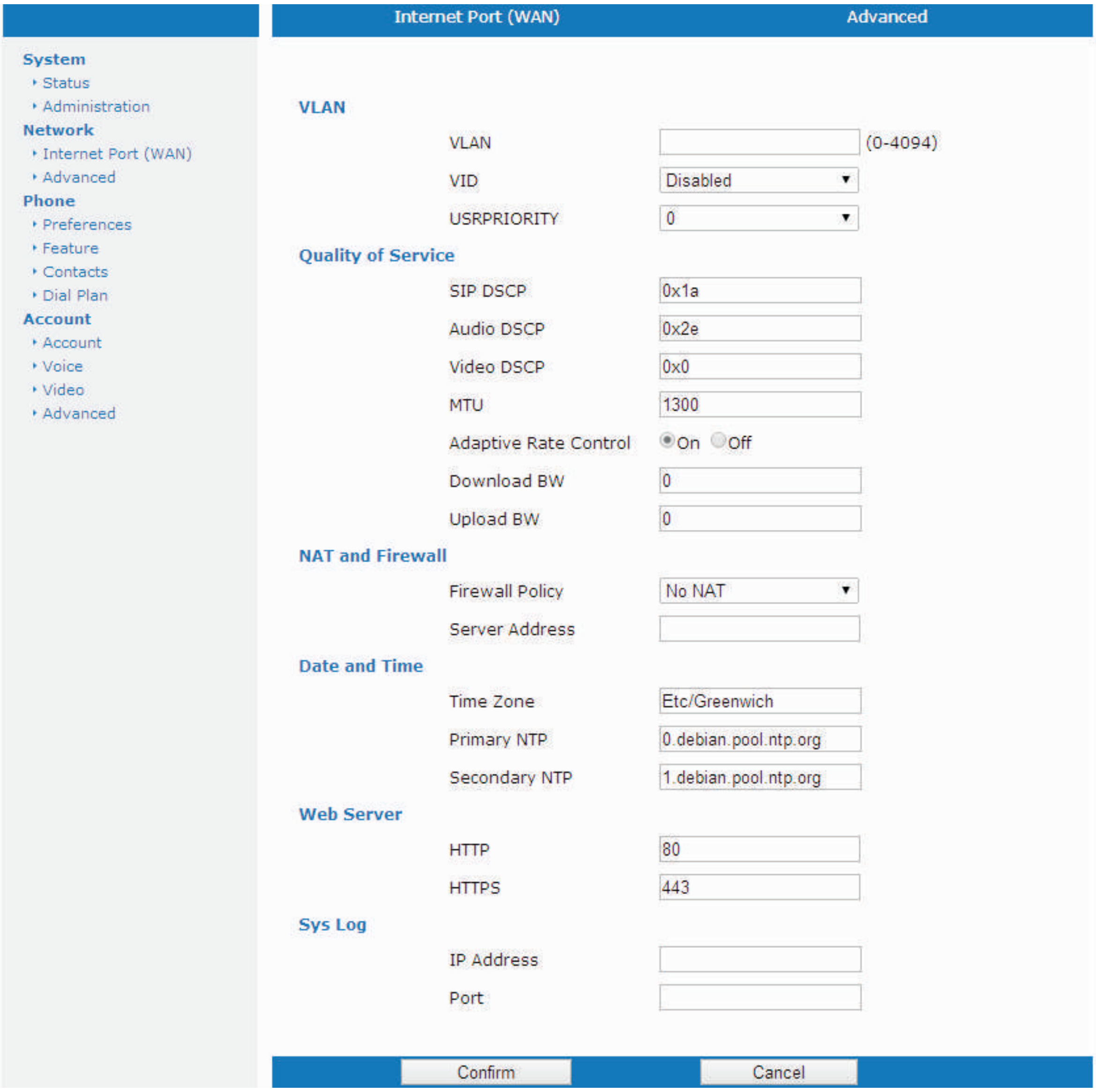
3.1.4 Internet Port (WAN)

This subsection is used for setting the Sentry IP network parameters. A change of any of these parameters will take effect immediately, and the web browser will be redirected to the new IP address however you need to restart Sentry IP for the configuration to be stored.



Field	Description
DHCP	Obtain network parameters through DHCP server
Static IP Address	Set the network parameters manually.
IP address	Set the IP address assigned by your LAN administrator
Network mask	Set the network mask
Default gateway	Set the default network gateway
Primary DNS	Set the primary Domain Name Server IP address for your LAN
Secondary DNS	Set the secondary Domain Name Server IP address for your LAN

3.1.5 Network Advanced

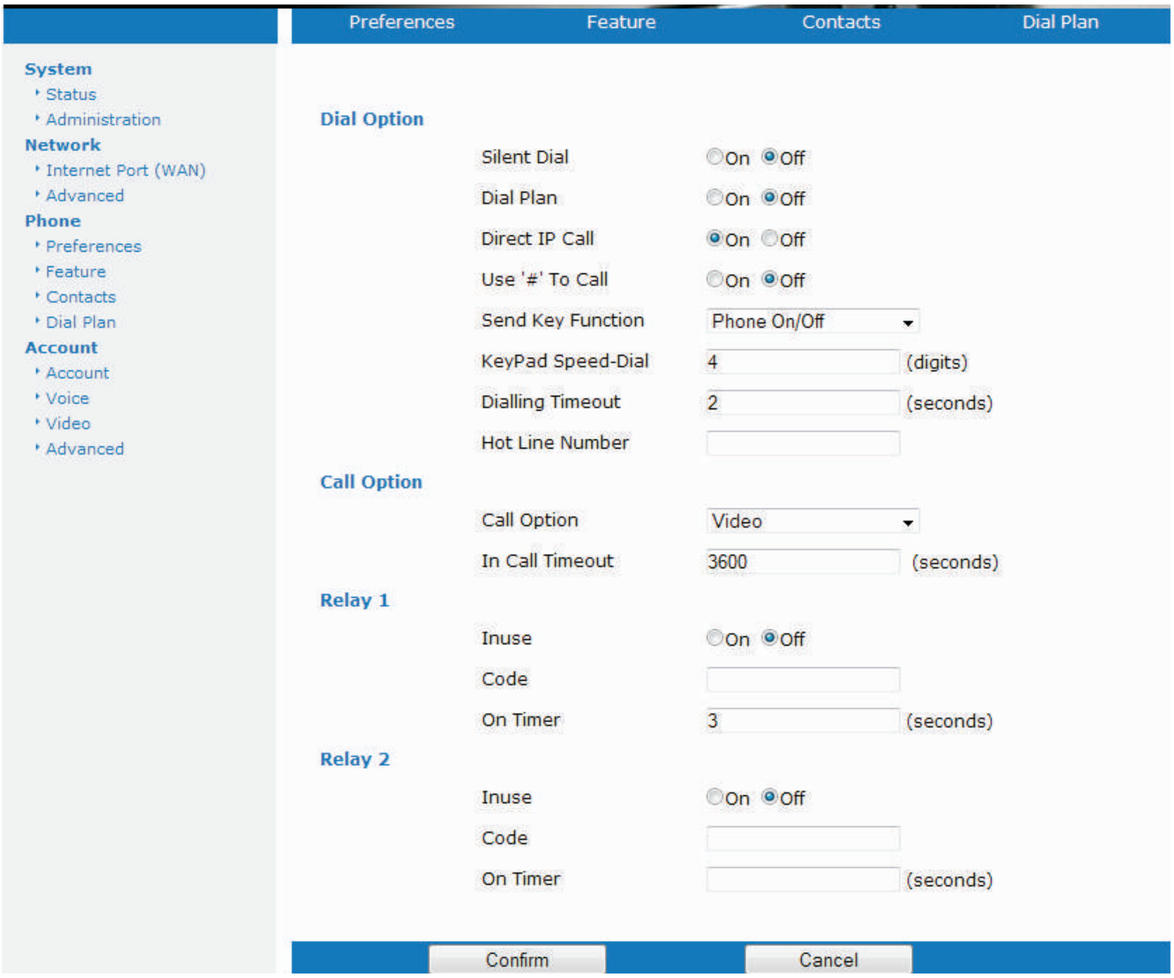


Field	Description	Values	Default
VLAN	Configures the VLAN ID that associates with the particular VLAN.	0-4094	
VID	Enable/Disable tag VLAN ID in the packets sent	Enabled Disabled	Disabled
USRPRIORITY	Specifies the priority used for transmitting VLAN packets.	0-7	0
SIP DSCP	Differentiated Services Code Point (DSCP) value in hexadecimal for the SIP Protocol	Number in hexadecimal	0x1a
Audio DSCP	Differentiated Services Code Point (DSCP) value in hexadecimal for the Audio RTP Stream	Number in hexadecimal	0x2e

Video DSCP	Differentiated Services Code Point (DSCP) value in hexadecimal for the Video RTP Stream	Number in hexadecimal	0x0
MTU	Network's Maximum Transmission Unit (MTU). Use 0 to allow automatic discovery, otherwise set to a number of bytes.This parameter is only meaningful with video streams for which RTP packets are big	Number	1300
Adaptive rate control	Turn on/off Adaptive Rate Control	On Off	On
Upload BW	Estimated upload bandwidth in kbit/s	Number	0
Download BW	Estimated download bandwidth in kbit/s	Number	0
Firewall Policy	No NAT: assume there is no Network Address Translators (NAT). Gateway: use firewall address supplied in Server field below (discouraged). STUN: Use Session Traversal Utilities for NAT (STUN) server in Server field below to discover its own public IP address and ports. ICE: Interactive Connectivity Establishment (ICE) is a technique used in computer networking involving (NATs) in Internet applications of Voice over Internet Protocol	No NAT Gateway STUN ICE	No NAT
NAT Sever Address	Firewall address to use when in firewall policy=Gateway, or STUN server address to use when in firewall policy=STUN		
Time Zone	Set the Time zone		AUS/Melbourne
Primary NTP	Set the IP address of the Primary NTP server for time synchronisation	IP address/ Domain name	
Secondary NTP	Set the IP address of the Secondary NTP server for time synchronisation	IP address/ Domain name	
Web Server HTTP	Set the web server communication port		80
Web Server HTTPS	Set the Secure web server communication port		443
SysLog IP Address	Remote SysLog IP address	IP address/ Domain name	
SysLog Port	Remote SysLog Port	Number	

3.1.6 Preferences

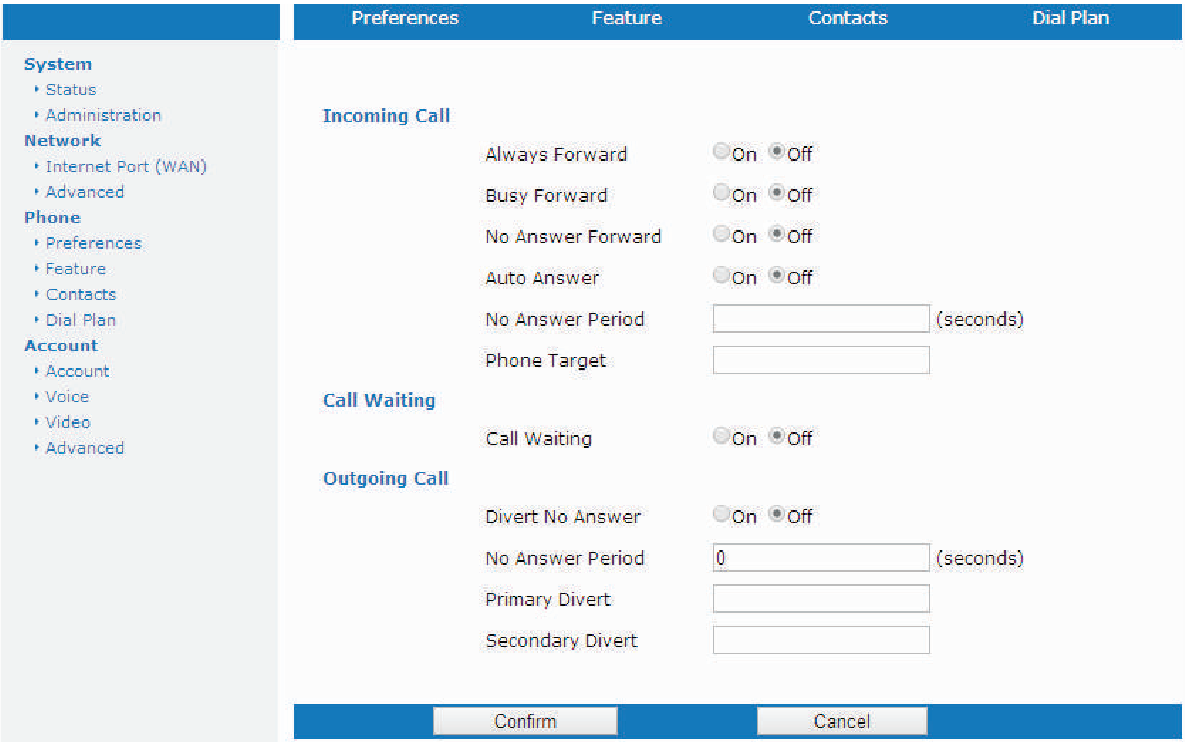
This subsection manages the dial process, the type and the duration of the call. The on-board relays may be activated remotely by dialling the digits that matches the fields for ***Relay 1*** & ***Relay 2*** code.



Field	Description	Values	Default
Silent Dial	Turn On/Off dial and keypad tones	On Off	Off
Dial Plan	A dial plan establishes the expected sequence of digits dialled, This option will turn On/Off this feature	On Off	Off
Direct IP Call	Enable/Disable the use of IP address digits dialling	On Off	On
Use '#' To Call	Enable/Disable the use of the HASH key to establish a call after dialling	On Off	
Send Key Function	The function of the (# key) Normal: Acts as a normal key. Phone on/off: Configured to turn the phone on/off (establish and terminate a call). Phone off: Configured to turn off the phone (terminate a call).	Normal ON/Off Off	Normal
Keypad Speed-Dial	Configure the keypad speed dial, dial memory location(phonebook) immidiatley after number of digits being pressed, if keys being pressed are less than number of digits it will dial after a timeout period,0 will disable this feature.	0 = disable Any integer value	Disable
Dialling Timeout	Time in seconds to establish a call, This will be ignored if the send key is configured to establish a dial	Time (seconds)	4
Hot Line Number		Number	
Relay Inuse	Enable/Disable the relay switch	On Off	Off
Relay Code	The code to turn on the relay	Number	
Relay On Timer	Time in seconds to keep the relay on for	Time (seconds)	

3.1.7 Feature

This section controls the way the unit makes and receive calls. Select options, as to what should happen when calls are made to the unit, whether to accept or reject it, during different modes of operation.



Field	Description	Values	Default
Always Forward	Enable/Disable Always Forward feature. If On all the incoming calls will be forwarded to the target number	On Off	Off
Busy Forward	Enable/Disable Busy Forward feature. If On and the phone in a busy state the call will be forwarded to the phone target	On Off	Off
No Answer Forward Forward	Enable/Disable No Answer Forward feature. If On and the has not being answered after a chosen timeout period the call will be forwarded to the target number	On Off	Off
Auto Answer	Enable/Disable Auto Answer feature. If On the call will be answered after a chosen timeout period	On Off	Off
No Answer Period	Timeout period in seconds. This is for the call forward and the auto answer features.	Time (seconds)	0
Phone Target	The phone number to forward the call	Number	
Call Waiting	Enable/Disable Call Waiting feature	Time (seconds)	4

Divert No Answer	Enable/Disable Divert No Answer feature	On Off	Off
No Answer Period	Time in seconds before an outgoing call can be diverted to another number	Time (seconds)	
Primary Divert	The primary phone number to divert if the outgoing call is not successful	Number	
Secondary Divert	The Secondary phone number to divert if the outgoing call from the primary number is not successful	Number	

3.1.8 Contacts

3.1.8.1 Phone Book

In this section the user can store the phone numbers into specific memory locations (Speed Dial), which will allow the user to make a call to a specific phone destination from the front keypad or call button.
To enable this feature set the 'KeyPad Speed-Dial' digits to zero (0) value in the Preferences section.

Note:
When entering values into fields you must press the enter/return key to register your new entry.

Phone Book

Location	Name	Phone Number	MIC
1	Office	900@192.168.1.190	INT
2	Workshop	500@192.168.1.200	INT
3	Reception	800@192.168.1.190	EXT
4	Sales	300@192.168.1.200	EXT

Page 1 of 1 10 View 1 - 4 of 4

Field	Description
Location	Speed dial memory location. Must be an integer.
Name	Any given name. Any characters.
Phone Number	Phone number, extension number, IP address or SIP URI ('ext'@'SIPaddress)
MIC (for using Master/Slave option)	INT selects internal MASTER microphone EXT selects external SLAVE microphone

Master/Slave feature:

When using this mode you can call extension numbers from different SIP URI accounts. This enables complete separation of two SIP URI accounts within the one unit.
The example above shows that the 'Office' can be called on extension '900' via the SIP URI account '192.168.1.190' using the Master unit's microphone for communication.
'Sales' can be called on extension '800' via the SIP URI account '192.168.1.200' using the Slave unit's microphone for communication.

Please refer to section 4.2 Wiring - Connection Details for Master/Slave setup.

3.1.8.2 Access Control

The Access control table allows the user to assign keyless entry codes and set the access time.
Currently the Access Control table is configured to use Relay1 timer (section...).

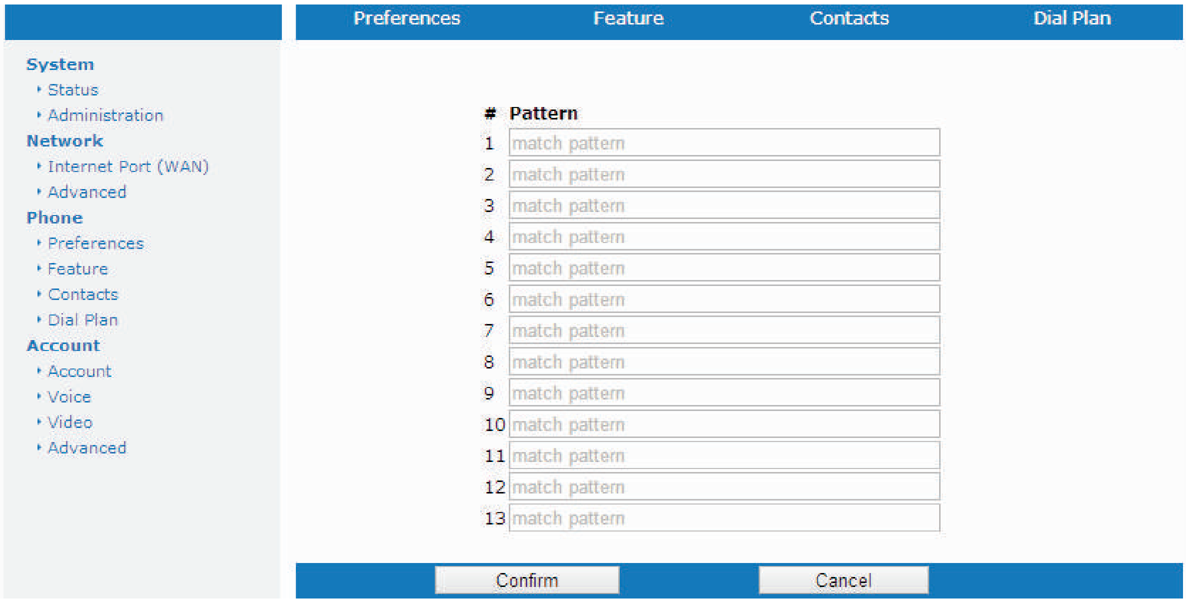
Key	Start Time	End Time
12345	08:00	20:15
9850385	15:35	19:45

Page 1 of 1 10 View 1 - 2 of 2

Field	Description
Key	Keyless entry code. Must be an integer. For keyless entry, press '*' then 'code'
Start Time	Start access time. 24 hour format. Eg: 00:00 - 23:59
End Time	End access time. 24 hour format. Eg: 00:00 - 23:59

3.1.9 Dial Plan

A dial plan establishes the expected sequence of digits dialled on the Sentry IP Video Door Station. To enable the dial plan select yes on the Preferences (***3.1.6)*** subsection. When making a call, numbers that are accepted must match one of the group patterns in the dial plan.



The following syntax used to identify a dial plan in a digit map is adapted from [RFC 2705].

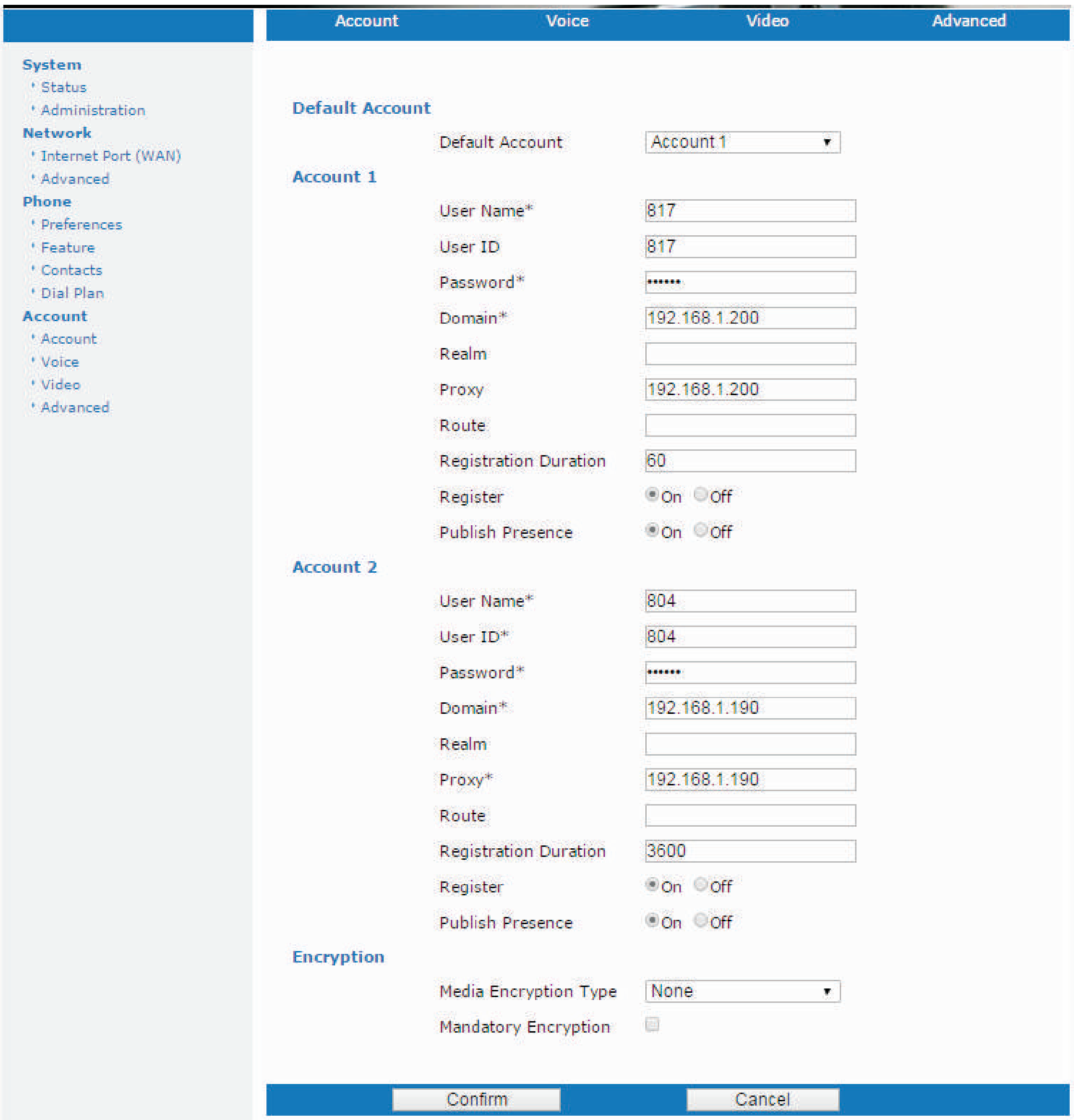
To specify a	Enter the following	Result
Digit	0 1 2 3 4 5 6 7 8 9 *	Identifies a specific digit (do not use #)
Range	[digit-digit]	Identifies any digit dialed that is included in the range
Range	[digit-digit, digit]	Specifies a range as a comma separated list
Wild Card	X	x matches any single digit that is dialled
Wild Card	**.**	**.** matches an arbitrary number of digits

Some *dial plan* examples using the above syntax look as follows:

For calls to	Dial plan
Internal Extension	xx
Mobile Number	04XXXXXXXX
interstate numbers	0[2-9]xxxxxxxx

3.1.10 Account

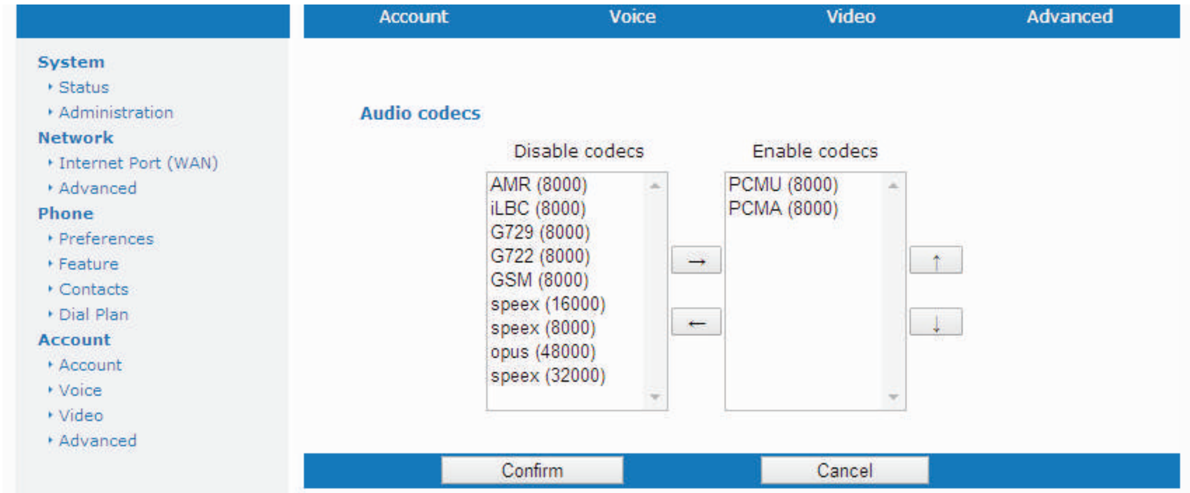
Two separate SIP accounts can be used within the one unit.
The example below shows that 'Account 1' is set as the default SIP account, so if an extension is called without the full SIP address making a call to extension 900 via account 2, (eg. 900@192.168.1.190) then the call will be made via the default SIP Account 1 (192.168.1.200).



Field	Description	Values	Default
Default Account	Select the default VoIP account	No Account Account 1 Account 2	No Account
Display Name	Enter the authorisation ID to be used for authorisation		
Register Name	Set the user ID for registration		
Password	Enter the password to be used for authorisation during registration and calling		
Domain	The domain name or IP address of the server to be used for calling		
Outbound Proxy	Set SIP server address to send all outgoing SIP requests. It is usually left blank, otherwise it is commonly used to as outbound proxy		
Registration Duration	Expiration period of the registration in seconds	Time (seconds)	3600 (seconds)
Register	Set whether should register with the SIP proxy or not	ON Off	On
Publish Presence	Send a PUBLISH request to the proxy to notify about presence information	ON Off	Off
Media Encryption Type	-None no encryption -SRTP is Secure Real-time Transport Protocol. It is a security profile for RTP protocol -ZRTP is a cryptographic key-agreement protocol to negotiate the keys for encryption between two end points. It uses Diffie–Hellman key exchange and uses SRTP for encryption	None SRTP ZRTP	None
Mandatory Encryption	Enforce Encryption when using SRTP	ON Off	Off

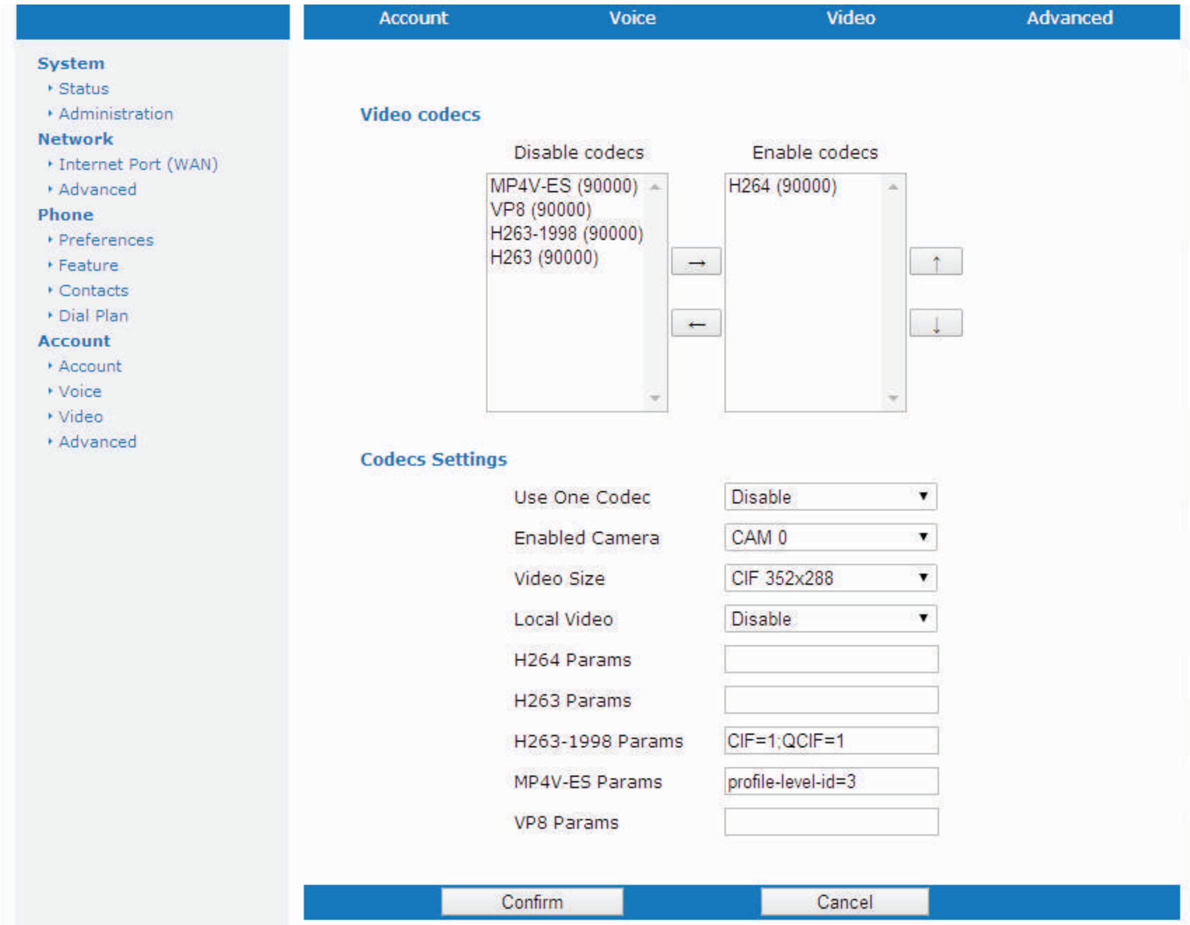
3.1.11 Audio

This subsection is used to enable or disable audio codecs.
The codec priorities are determined by the sequence: the top codec in the sequence has the highest priority.

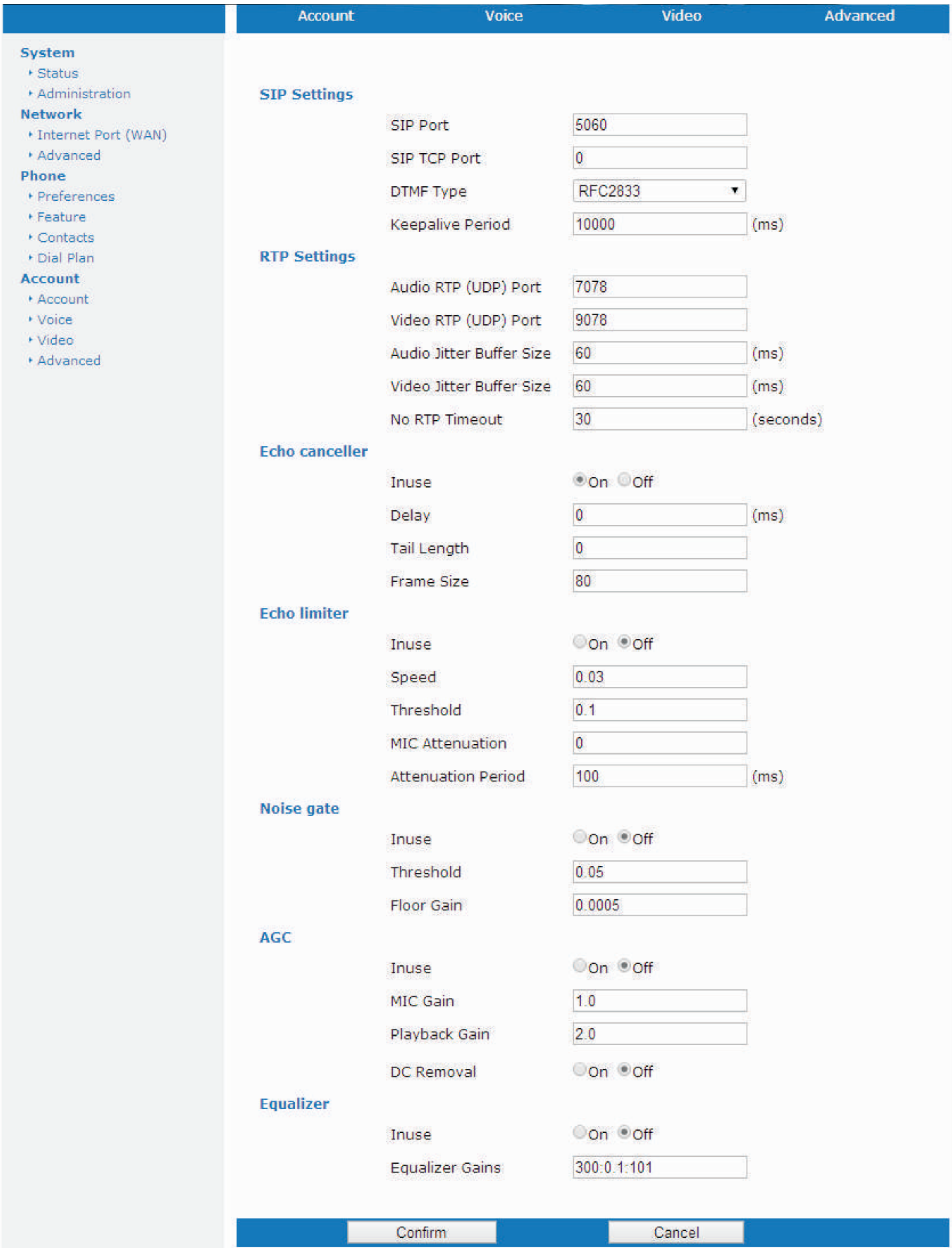


3.1.12 Video

This subsection is used to set the video camera resolution, the codec properties and parameters.
The codec priorities are determined by the sequence: the first codec in the sequence has the highest priority.
You can set the codecs parameters in such a manner that the video transmitted conforms to the needs of the opponent.
The codec format parameters are string to be sent in SDP for this codec, which normally corresponds usually to what we prefer to receive.



3.1.13 Advanced Settings



Field	Description	Values	Default
SIP Port	SIP UDP Port to be used	Number	5060
SIP TCP Port	SIP TCP Port to be used	Number	0
DTMF Type	Defines how DTMF signalling should be received and sent	RFC2833 SIP INFO	RFC2833
Keepalive Period	Keepalive period in milliseconds for sending out SIP UDP keepalive to the proxies.	Time (ms)	10000
Audio RTP Port	Audio RTP (UDP) port	Number	7078
Video RTP Port	Video RTP (UDP) port	Number	9078
Audio Jitter Buffer Size	Nominal audio jitter buffer size in milliseconds	Time (ms)	60
Video Jitter Buffer Size	Nominal video jitter buffer size in milliseconds	Time (ms)	60
No RTP Timeout	Set the time limit for receiving audio stream RTP packets during a call. When the limit is exceeded, the call is terminated.	Time (Seconds)	30
Echo Canceller Inuse	Turn on/off echo cancellation	On Off	On
Delay	Expected delay of echo in milliseconds. This allows to reduce the tail length of the echo canceller, which speeds up convergence and reduces complexity of computations	Time (ms)	0
Tail Length	Tail length of echo canceller in milliseconds. Ideally it should be no more than the expected duration of the echo	Number	0
Frame Size	Frame size for AU-MDF echo canceller algorithm. This is a parameter internal to the echo canceller, recommended is too keep to its default value	Number	80
Echo limiter Inuse	Turn on/off the echo limiter. The echo limiter is an algorithm that consists in lowering the gain of the MIC input when the speaker is talking. Combined with the noise gate (see next section) it gives good results when the echo canceller no more works, because of non-linear distortion (saturation) of the echo path. Its drawback is that it turns the call in a kind of automatic half-duplex mode, which makes impossible to interrupt the person who is talking.	On Off	Off

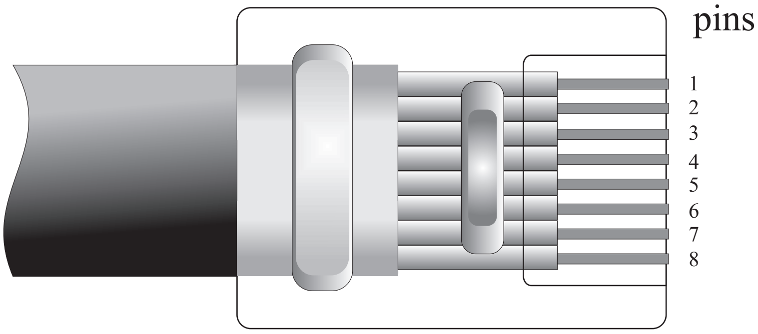
Speed	Gain changes are smoothed with this coefficient. It's a value between 0 and 1. 0.1 is already very fast, 0.001 is very low default value is 0.03. Recommendation is to keep it unchanged.	Number	0.03
Echo limiter Threshold	Threshold above which the system becomes active. It is a normalized power, between 0 and 1. Default value is 0.1 A smaller value can be better	Number	0.1
MIC Attenuation	The proportional coefficient controlling the MIC attenuation. Default value is 10	Number	0
Attenuation Period	Time in milliseconds for which the attenuation is kept unchanged after resuming from speech to silence on the network->speaker channel. The purpose of the parameter is to keep the MIC attenuated for some time until the echo of the audio buffers is finished. 100 ms is a reasonable value to start, can be higher	Time (ms)	100
Noise gate Inuse	Turn on/off the noise gate. The goal of the noise gate is to remove (or attenuate a lot) the background noise heard by the microphone. Noise and speech are distinguished using an energy threshold. The use of the noise gate can prevent feedback to produce between two devices running our VoIP system	On Off	Off
Noise gate Threshold	Noise gate threshold in linear power between 0 and 1: Above this threshold the noise gate becomes bypass.	Number	0.05
Floor Gain	Gain applied to the signal when its energy is below the threshold. It is expect to be low so that noise is attenuated.	Number	0.0005
AGC Inuse	Automatic gain control (of MIC input) - turns on or off	On Off	Off
MIC Gain	Static software gain (linear scale) to be applied to microphone signal	Number	0000
Playback gain	Static software gain (log scale) to be applied to signal sent to speaker	Number	0000
DC Removal	Enable or disable DC removal of the MIC input	On Off	Off
Equalizer Inuse	A parametric equalizer can be used to recover from speakers with bad spectral response. The frequency response of the equalizer is entirely configurable.	On Off	Off
Gains	Equalizer gains. It is a list of triplets ::	Triplets ::	300:01.10 1

4.1 Wiring - Power Over Ethernet (PoE)

The DDC_VoIP telephone may be powered from the 2 way connector marked ‘Power’ on the side of the unit, see page 26. The polarity to this connector must be correct for the unit to function. (Note that NO damage will occur if it is connected incorrectly).
The voltage input to this connection may be from 9 to 48 Volts.

The alternative way to power the DDC_VoIP telephone is via the Ethernet connector using a PoE switch.
The DDC_VoIP telephone is adapted to use a class 0 form of power source.

There are two methods that power can be supplied from a Ethernet Switch,
Mode A, supply power over the Tx/Rx pair, (TX pair pins 1&2 and RX pair pins 3&6).
Mode B, via the spare pairs 4&5 and 7&8.



RJ45 Ethernet connector

pins
1 = TXP
2 = TXN

3 = RXP
6 = RXN

mode A
PoE

4,5 = Spare
7,8 = Spare

mode B
PoE

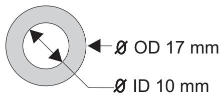
The figure above shows a typical RJ45 connector with the pins side facing up.
Note, In mode B, that the spare wires 4 and 5 are linked together within the PCB, the same for pins 7 and 8.

The polarity on pins 4,5 or 7,8 is not critical and is generally set within the Switch, (if used).

The DDC_VoIP telephone is configure to draw power from the PoE in class 1 (0.44-3.84 Watt) and should be connected to a IEEE 802.3af compatible PSE power unit .

4.2 Wiring - Connection Details

Microphone

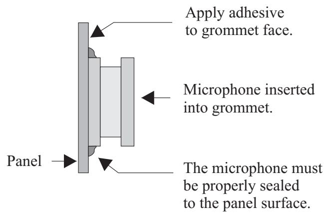


Grommet dimensions

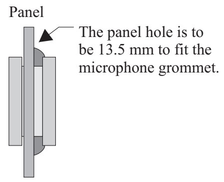
Optional External Speaker and Mic.

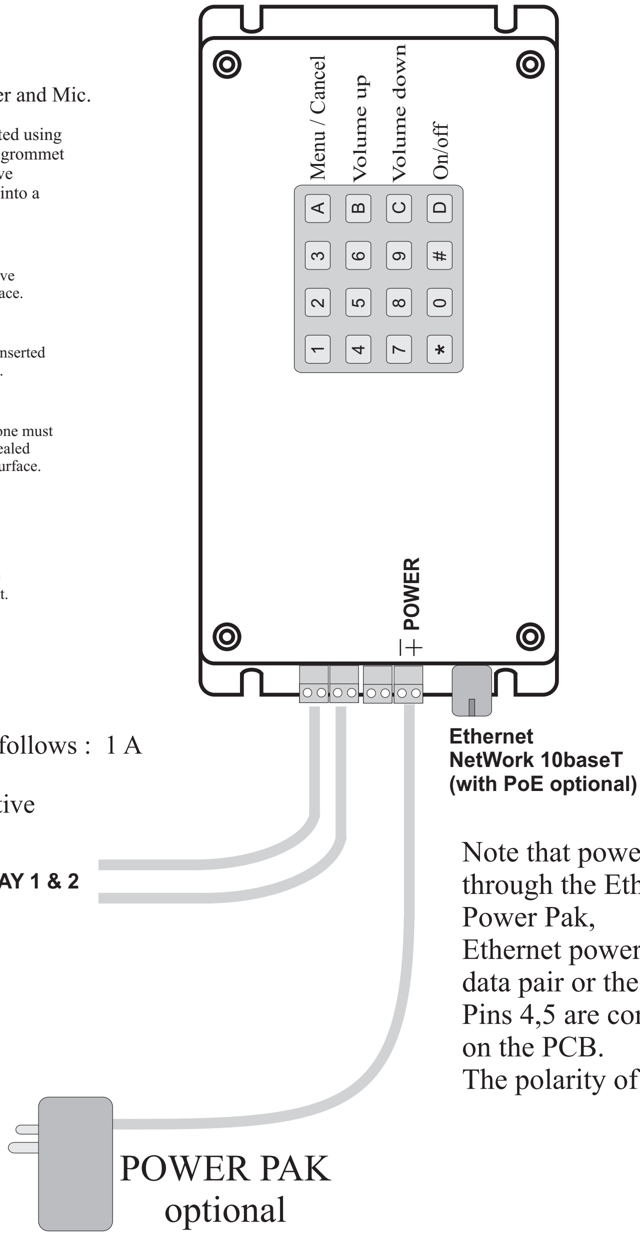
The microphone may be mounted using two methods, either mount the grommet to panel by applying an adhesive to the grommet face, or mount into a 13.5mm diameter hole.

Method 1



Method 2





Relay contacts are rated as follows : 1 A @ 50 V DC Non-inductive
1 A @ 30 V AC Non-inductive

N/O or N/C switch set by jumper LK1 / LK2

Note that power to the unit may be either supplied through the Ethernet connection or via the 12 Volt Power Pak,
Ethernet power can be induced over the Tx/Rx data pair or the spare pairs 4,5 and 7,8.
Pins 4,5 are connected together and so are pins 7,8 on the PCB.
The polarity of this connection is NOT critical.

WARNING

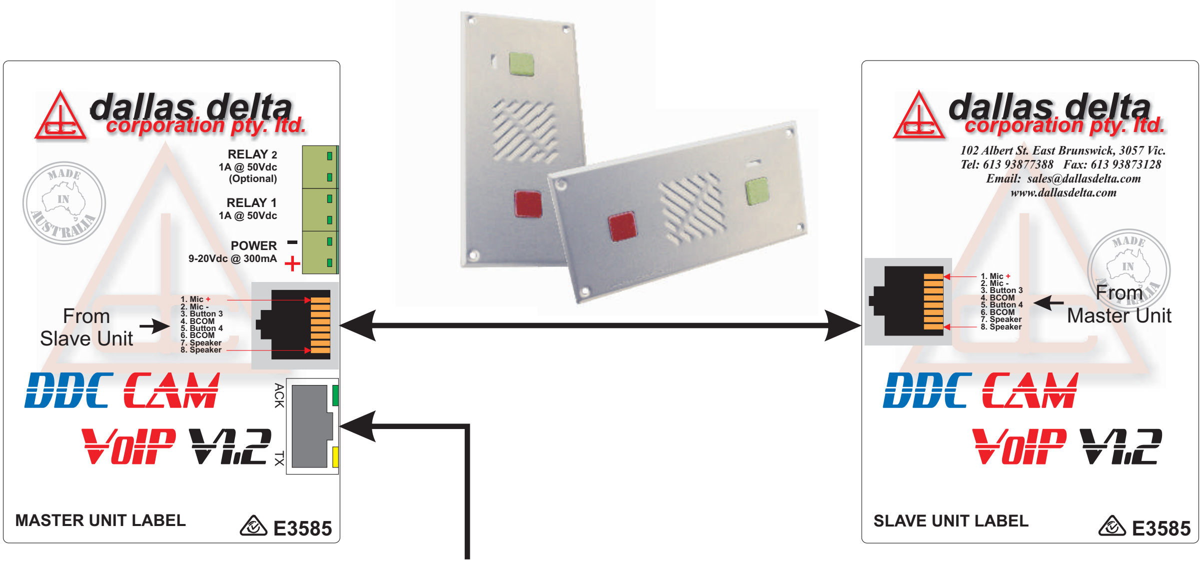
This telephone can not be used for emergency purposes during power failure unless a network connection is guaranteed with Power Over Ethernet (PoE).

Note: All wiring is to be routed away from high EMI radiating devices, such as transformers, fluorescent lighting etc.

4.2 Wiring - Connection Details continued...

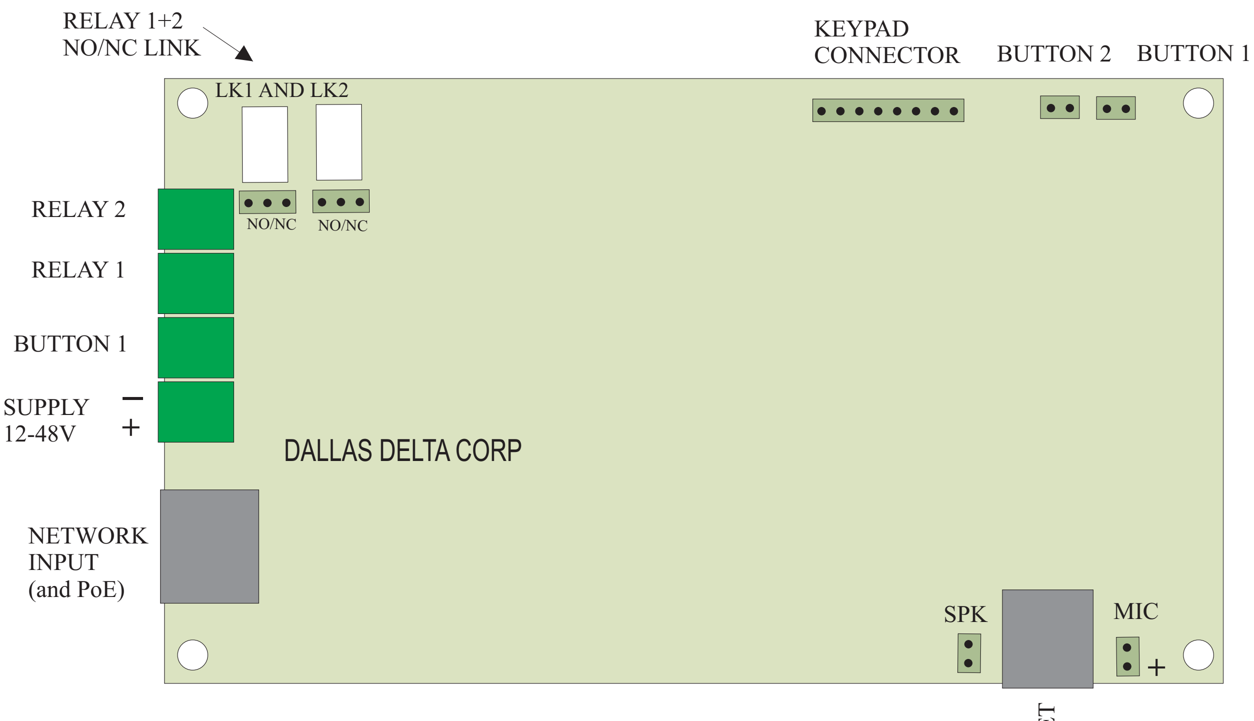
MASTER/SLAVE WIRING DIAGRAM

Below is how to wire up the VoIP units in Master/Slave configuration.



4.3 Wiring - PCB Layout

The DDC CamVoIP PCB connection may be different from the diagram shown, The layout is of a generic configuration, some items may not be installed and/or not required.



LK1 and LK2 relay links are used to select the normal state of the output contacts.

 Relay linked for normally **closed** contacts

 Relay linked for normally **open** contacts

5.0 Specifications

power:	input voltage	**12** Volts *(minimum)* - 50 Volts *(maximum)*
current consumption:	-idle mode -on call	60mA @ 13Vdc (0.8 Watts) 100mA @ 13Vdc (1.3 Watts) *normally* <300mA @ 13Vdc (3.9Watts) *(maximum volume into a 8ohms speaker)*
relay contacts:	switching maximum	1A @ 60Vdc / 40Vac SELV or TNV (non inductive load) voltage free outputs
temperature:	operating range	0ºC to +50ºC
SPL:	ringer output level	>80dBa @ 1 metre (32 steps)
communication:	Ethernet Connection protocol Audio CODECs Video CODECs	10 / 100 SIP G711 (uLaw, aLaw), Speex (narrow and wide band), G.722, Opus, GSM, G.729 VP8 (WebM) ,HH.263-1998, mpeg-4, H.264 with resolutions from QCIF (176x144) to VGA (640x480)

		Sentry	Guard(brick)	Guard(Vertical)
physical:	panel dimensions (mm)	270 x 130`	255 x 104	100 x 227 x 46
	wall cut-out (mm)	225 x 115 x 40	246 x 94 x 50	surface mount
	weight (kg)	approx. 1	approx. 1.02 (std brick cut out)	approx. 0.9

WARNING

This telephone can not be used for emergency purposes during power failure unless a network connection is guaranteed with Power Over Ethernet (PoE).

To be installed and maintained by authorised service personnel only.

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