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VoIP Doorstation
S-11 User's Manual

VoIP Doorstaion Phone S11

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Custom Communication Solutions

Dallas Delta Corporation Pty. Ltd. is an established and respected provider of specialist communications equipment.

We design and manufacture our products in Melbourne and have many thousands of satisfied customers within Australia and internationally.

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1.0 Introduction

Thanks you for choosing Dallas Delta products. We hope that our product will meet the demands of your application needs and it will give you many years of trouble free use.

The **VoIP Doorstation** is an ethernet connected telephone that provides voice 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 DoorStation is a loud speaker, 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.
This document outlines the units Features, Operation, Programming instructions and Installation procedures.

2.0 How to Use this Manual

This manual is set out with the intended use of both installers and end users in mind. It will give you the ease and quick turn-around time of installation required to get the job done efficiently and will also give you a very detailed understanding of the unit you are installing and putting into real-world use.

It is very important that installation of this unit is carried out by a qualified personnel to avoid damage of the unit and correct functionality of the product. It will also ensure that safely requirements are met for everyday use of this electrical product.

When installing this product for the first time, it is critical to follow the quick start guide to begin with. This will ensure that your new product is up and running in a very simple and easy manner with as minimal time as possible.

It is then recommended that the manual be read from cover to cover to allow you to fully understand the capabilities of your product and be able to obtain the most of this unit.

3.1 Getting Started - Basic Procedures

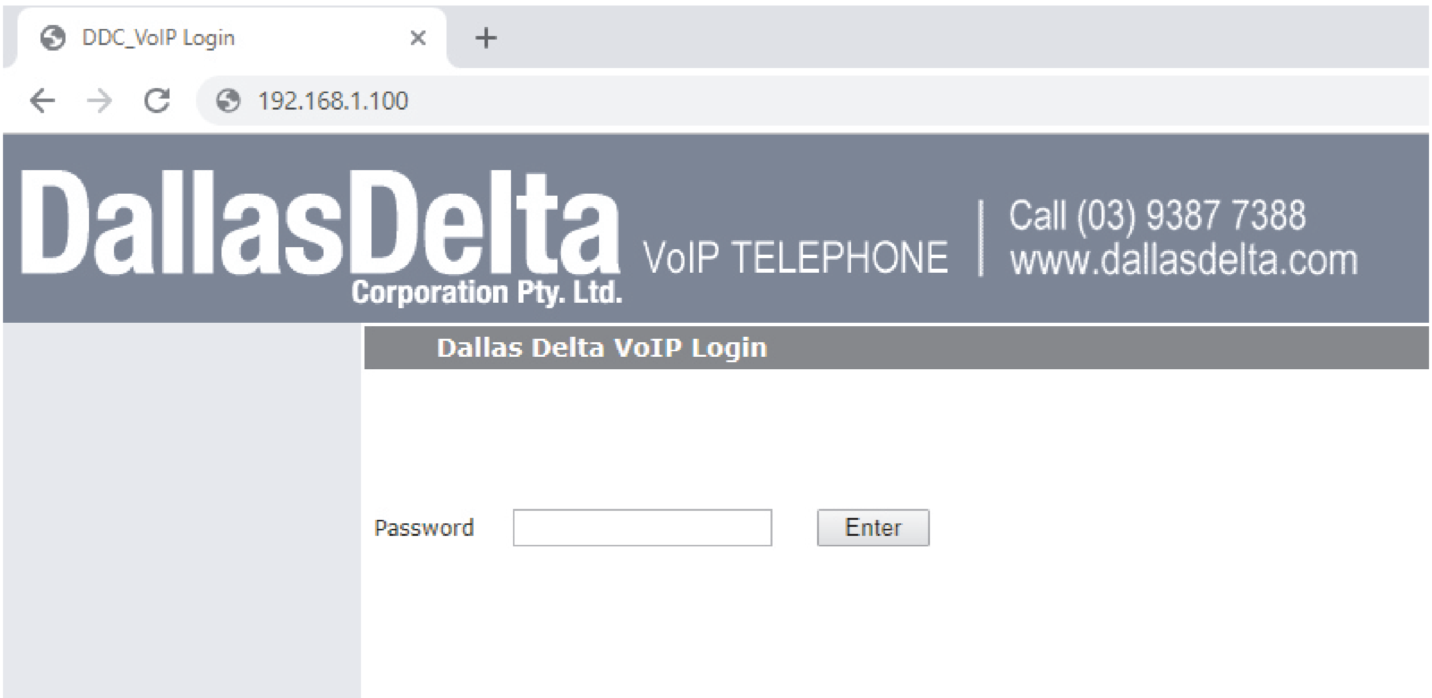
There are a few basic procedures that will need to be completed in order to get your VoIP Doorstation working immediately. Please follow these simple procedures in order before attempting to use your phone:

> Log on to your VoIP Doorstation
> Enter basic 'SIP Proxy' parameters
> Enter basic 'Systems' parameters
> Enter basic 'Network' parameters
> Program your call numbers for auto dialling

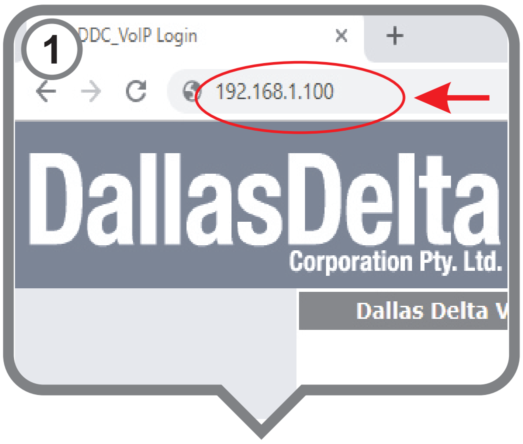
Below are very simple to follow step-by-step procedures to getting your VoIP Doorstation up and running:

3.2 Logging On to the Home Page

In order to begin setting up your VoIP Doorstation you must first log into the unit via a web browser (such as Google Chrome, Firefox, Internet Explorer, etc.). log into your VoIP doorstation by typing in the IP address of the phone in the address bar.



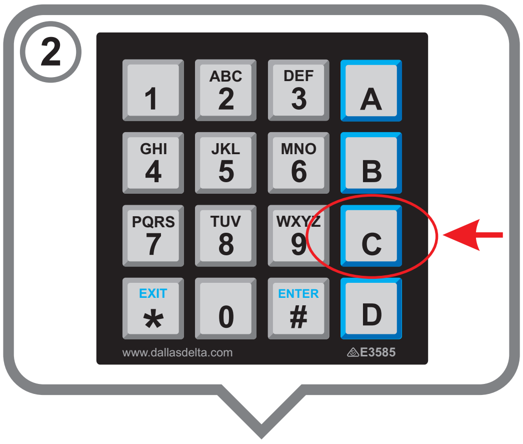
VoIP Doorstation Home Page



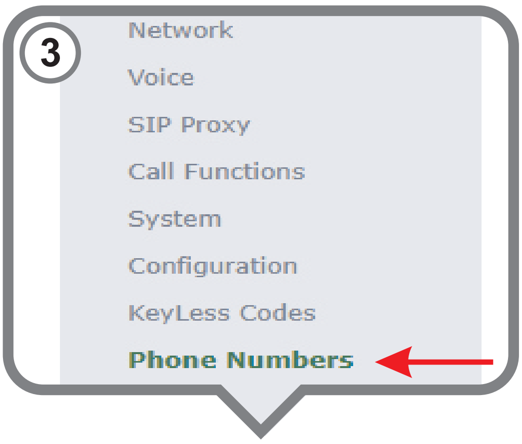
In the web browser's address bar, enter the IP address of the phone you wish to program.

The factory default address is 192.168.1.100

note: No password is programmed when dispatched from factory setup.



If you do not know the IP address of the phone, press the 'C' button on the programming keypad on the rear panel of the phone and it will play back a voice message with the current IP address.



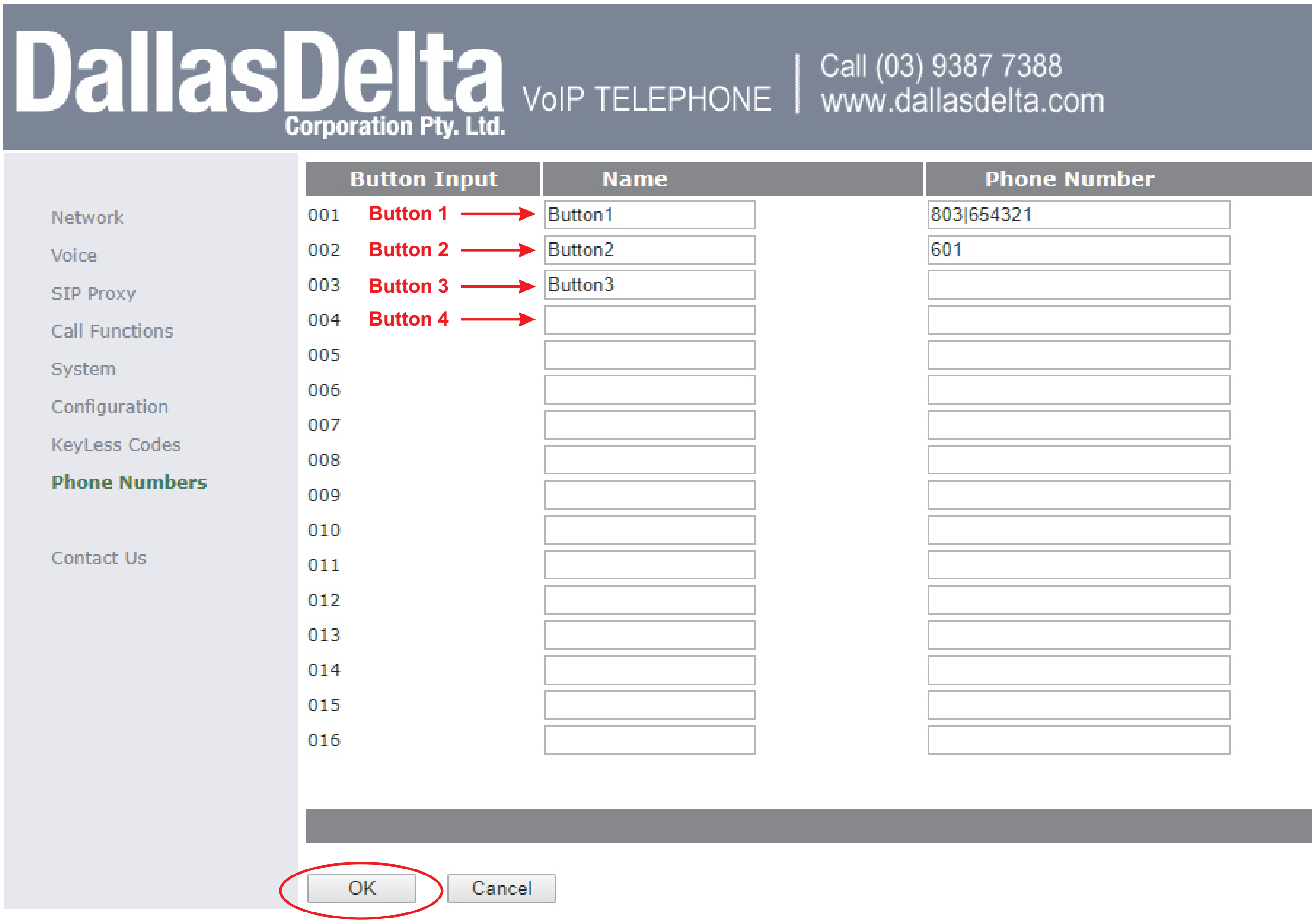
Once you have successfully logged on to the phone's home page, program numbers for the call buttons.

Select 'Phone Numbers' and see the next page

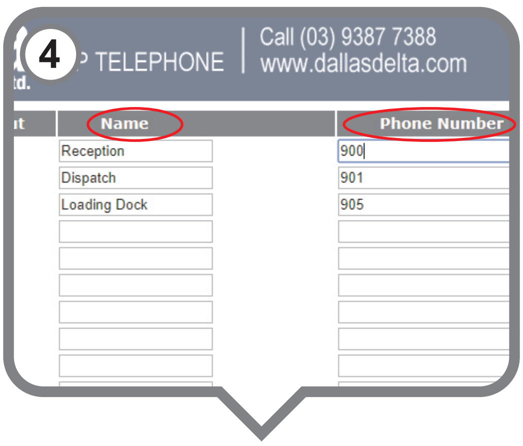
3.3 Button Phone Numbers

Here is where phone numbers are allocated to the Call Buttons on the front panel of the VoIP Doorstation. The 'Index' column is the allocated call button,

001 = button 1 The only location needed, If the unit you have only has one button.
002 = button 2
003 = button 3
004 = button 4 *(etc.)*



VoIP Doorstation Button Call Number Assignment



Enter a description in the 'Name' field of where the call will be directed.

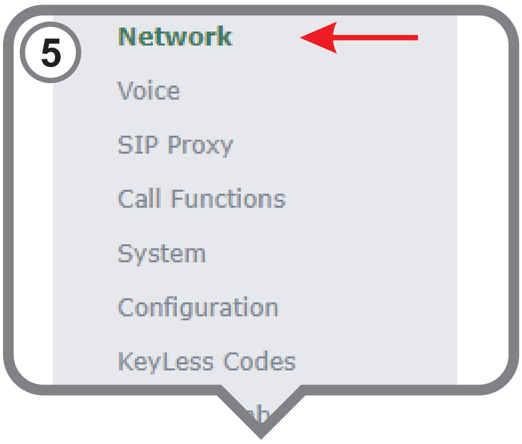
Enter the phone number you wish the button to call.

Press 'OK' to store your new entries.

3.4 Settings

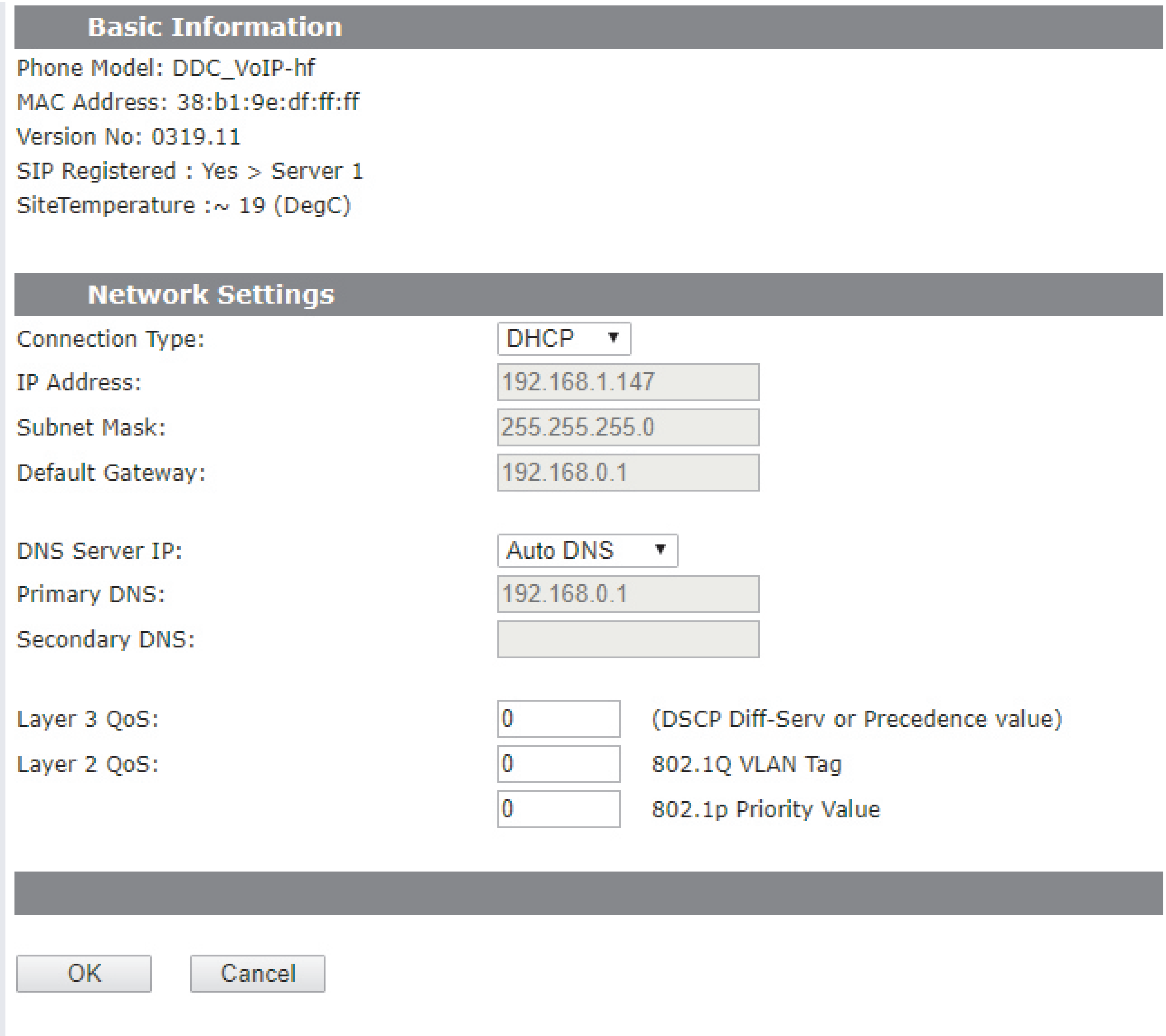
3.4.1 Network Settings

Note changing the network settings at this stage will required you to re-log back into the phone if the IP is changed. If so, follow the steps outlined in *3.2 Logging on to the Home Page*.

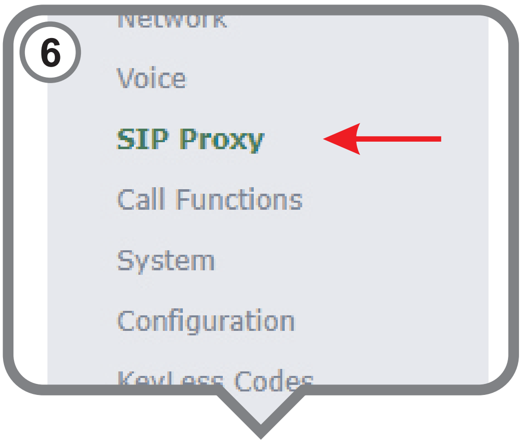


This page will show the current status of the registration, MAC ID, and version number of the F/W.

If you phone does require a static IP location, then verify with the IP department which IP can be used. Else leave the phone in DHCP mode to obtain an IP automatically.
If DHCP is selected, then most likely to leave the DNS to AUTO.
Care should be taken when setting the VLAN data, Refer to the IP department for these settings.



VoIP Doorstation Network Page



3.4.2 SIP Proxy Settings

Set up your SIP Proxy details here.

Select ‘SIP Proxy’ on the side menu to open up the SIP Proxy settings page.

To setup a basic SIP Registration you will need details setting from the site SIP PBX.
The PBX needs to be configured for a 3rd party SIP profile.
Then fill in the minimum required fields to complete the registration as follows:

SIP Server 1:

SIP Domain 1:

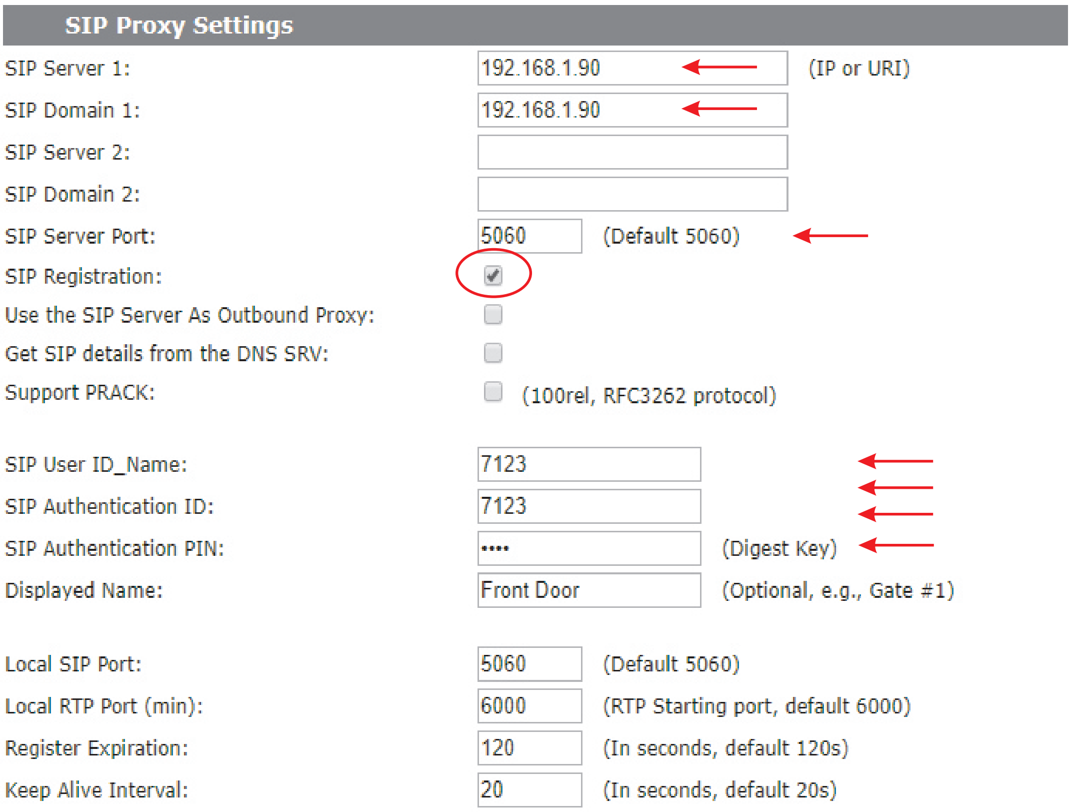
SIP User ID_Name:

SIP Authentication ID:

SIP Authentication PIN:
(This password is sometimes refer to as Digest KEY).

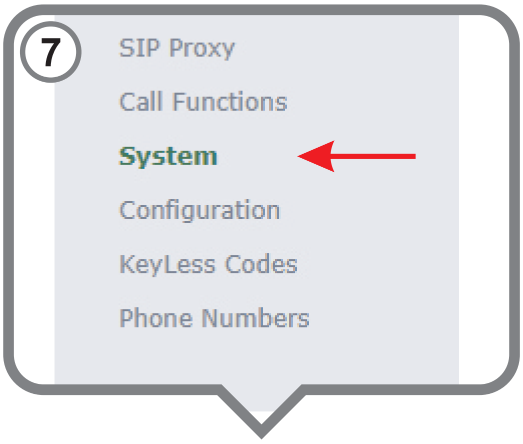
SIP Server Port:
(generally this is 5060)

All other settings are not critical at this point in time and can be set up at a later stage.



VoIP Doorstation SIP Proxy Settings Page

Press ‘OK’ to save your settings.



3.4.3 System Settings

To access the System Settings page, you do not need to return to the Home page.

Just simply select the ‘System’ link on the side menu.

To setup the System for basic operation, fill in the minimum required fields to complete the registration as follows:

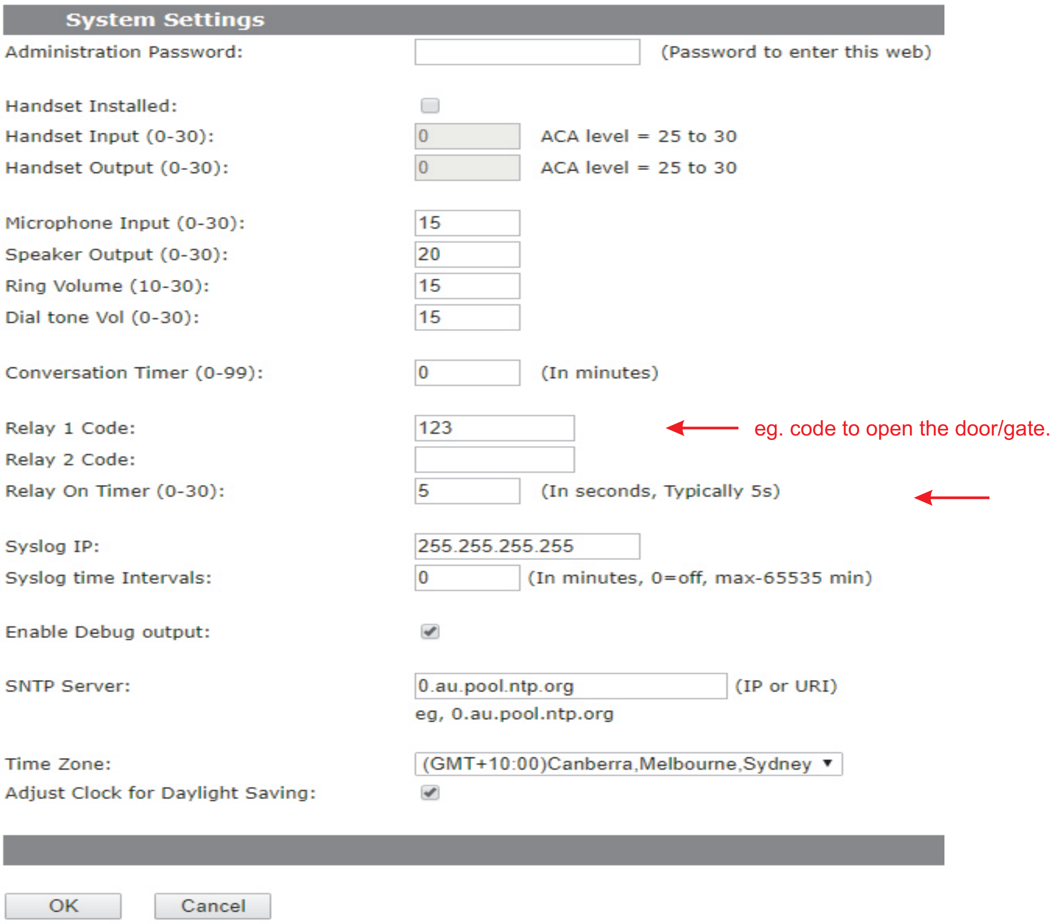
Relay Code 1:
This is the code used to trigger the onboard relay to open the door, gate, etc.

Note: The relay code is Factory default set to ‘123’ and the relay contact is Normally Open (N/O).

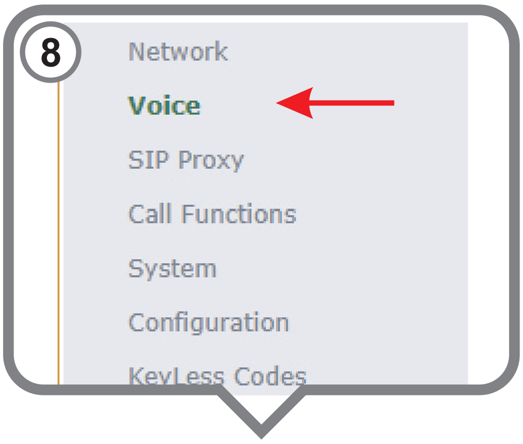
WARNING:
For the relays to operate in gate release mode, the ‘Relay Inuse’ option must be set to ‘None’. This can be found in the ‘Phone Configuration’ page. This is Factory default is ‘None’.

All other settings are not critical at this point in time and can be set up at a later stage.

Press ‘OK’ to save your settings.



VoIP Doorstation System Settings Page

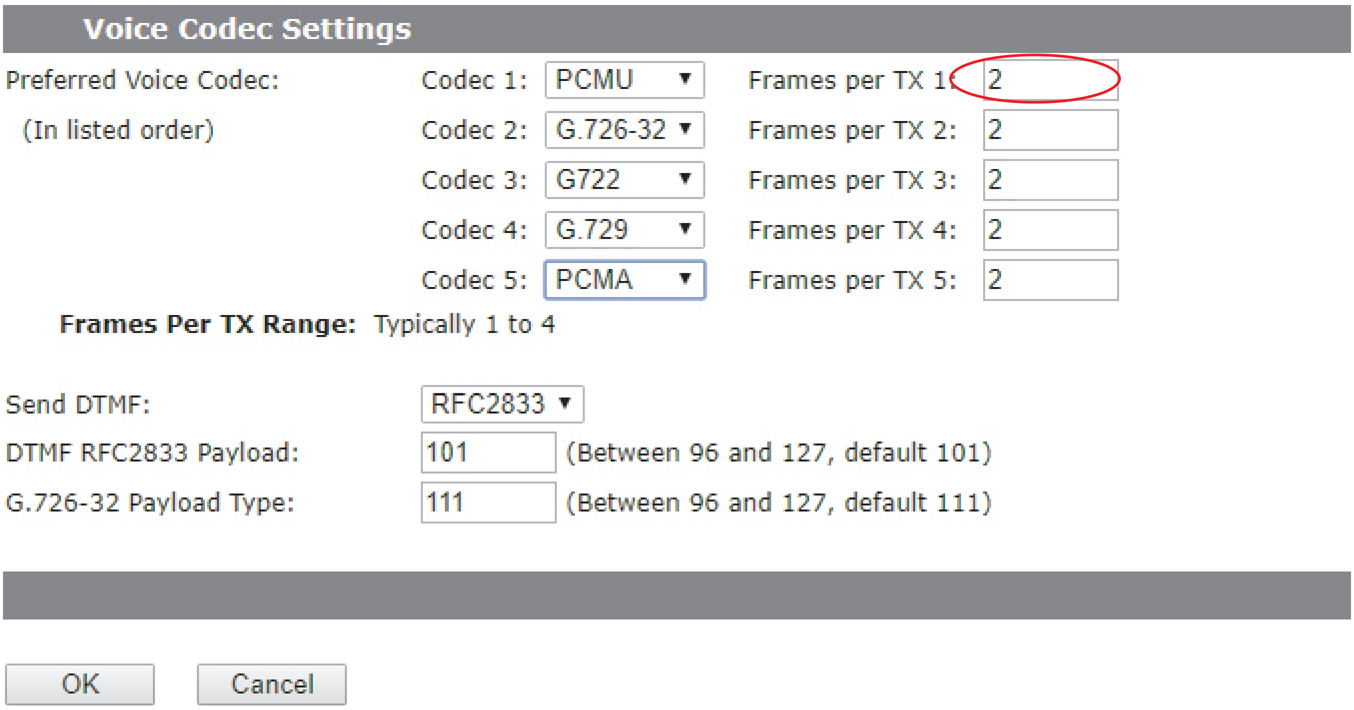


3.4.4 Voice Codec

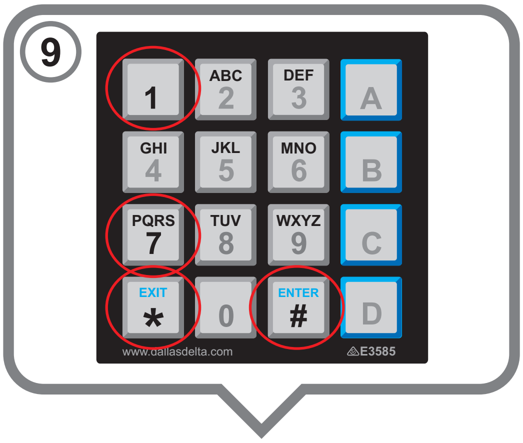
Most PBX phone systems today will work with the codec set to G711u (or PCMU).

This product does have other options, select preference and the order as required.
Note: The PBX can over-ride this settings to match the other party as part of the SIP protocol. If a match can not be made the call will terminate.

The RTP data is used to move the voice signal via the ethernet. Generally the voice frame is in 10mS segments and send with the network packet. How many frames are sent with each packet is set by the ‘Frames per Tx’ option. For most cases you would normally send 2-3 frames / packet.



VoIP Doorstation Settings Menu - Network Settings Page

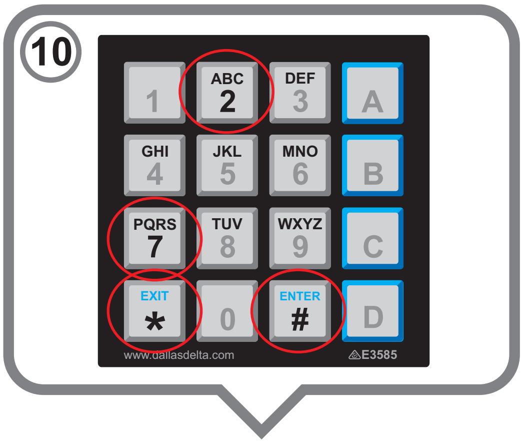


3.4.5 Reset Static IP Address (default)

The factory default address is 192.168.1.100
To set the phone to the default Static IP address, use the programming keypad on the back of the unit and enter the following key sequence:

*** # * # 7 * 1**

<u>**Note:**</u> Enter all digits in a continuous motion without pausing in between digit entries.

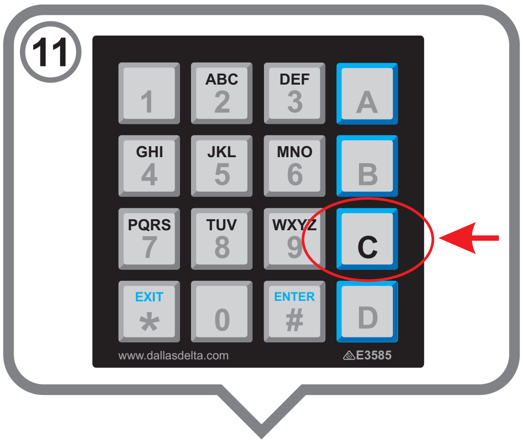


3.4.6 Reset DHCP IP Address Range

To set the phone's IP address to the DHCP range, use the programming keypad on the back of the unit and enter the following key sequence:

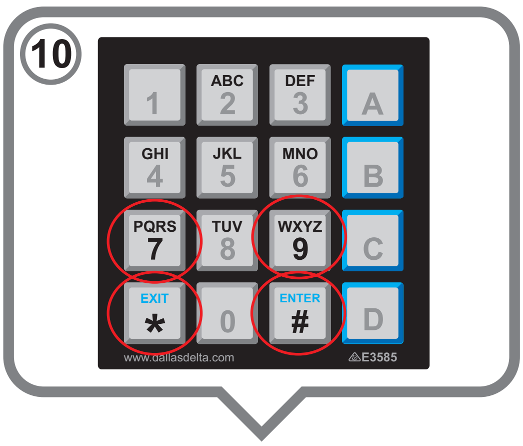
*** # * # 7 * 2**

<u>**Note:**</u> Enter all digits in a continuous motion without pausing in between digit entries.



3.4.7 Retrieving the IP Address

To check the IP address of the phone, press the 'C' button on the programming keypad during standby mode and it will play back a voice message with the current IP address.



3.4.8 Reset the phone back to factory defaults

To reset the phone's parameter back to a clean start (factory reset). Enter the following key sequence:

*** # * # 7 * 9**

<u>**Note:**</u> This will reset the IP address to static.
Clear the Admin PIN. and
Erase the SIP proxy details.

The DDC_VoIP telephone functionality operating in hands free mode.

When the call button is pressed, the unit will establish a call to a preset number.
Repressing the call button during a conversation period will terminate the call.
A call duration period may be set, if a limit is required.

The unit can be configured to auto-answer call or redirect to a forwarding number if not answered within the preset period.
During ring, the unit can be answered by pressing any for the buttons.

Handset models can be supplied with or without a keypad, and they can be setup to hotline (call a number) when the handset is pickup.

Calls made to a busy number or if the number if not answered can be re-direct to 2 other alternative numbers. Refer to the 'Call Function Section- ***Divert Numbers***'.

Relay Control

Two on board relays may be fitted to the door station, each may be switched on during conversation by the remote operator or either one can be configured to be activated when the call is initiated.

Remote relay control:

The DDC_VoIP phone is currently ship fitted with one relay as standard (the second relay on request).
The remote operator may activate the on board relay/s by entering the code set for that relay.
For example, if relay 1 code is set to 95, then remotely typing 95 will activate it, (refer to page 21).

A response beep will indicate that the code was correct and the relay has activated. No response tone indicates an error in the code and the process should be repeated after a minium 3 second delay.
(NB, as relay 2 is optionally fitted, the response will be the same whether the relay is installed or not.)
DTMF remote control uses RFC2833 or INFO protocol.

Call initiated relay control:

Either one of the relays may be configured to switch on from when the call button is pressed or from when it starts ringing.
This function may be used to control an external camera, light or to initiate an alarm, etc.
The relay will stay on for the duration of the call.
(refer to ***Relay Inuse*** in the Configuration Section).

WARNING

This telephone can not be used for emergency purposes during power failure unless fitted with a backup 12 volt battery AND network connection guaranteed.
To be installed and maintained by authorised service personnel only.

Keypad Functions

The keypad on the rear of the unit provides the installer access to:
Listen to the current IP address or reset it to a default IP and subnet MASK address.
Or make a test calls.

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

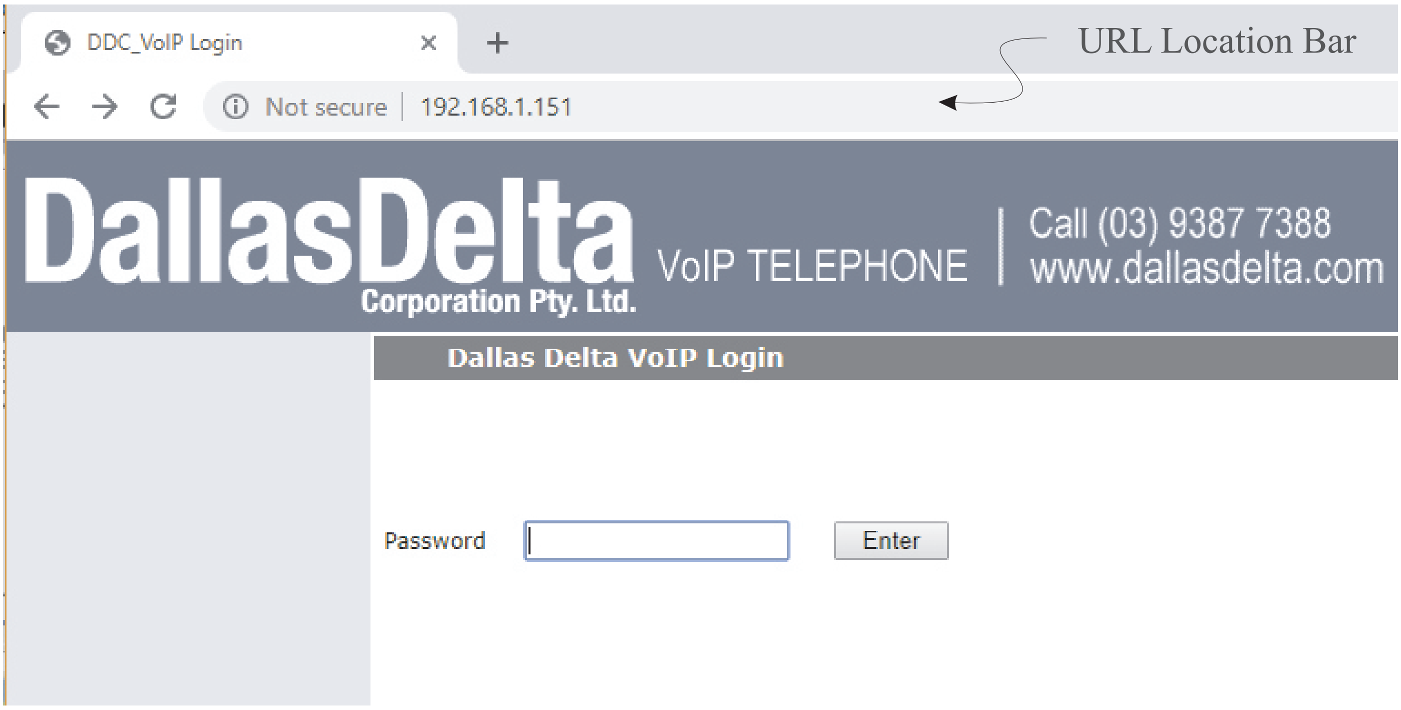
Resetting the IP address
To reset the address, connect the power (via the power jack or Ethernet PoE lead).
Wait about 10-20 seconds,
then enter ***# *# 7*1,** this will reset it to a *Static* IP address of 192.168.1.100, set the subnet mask to 255.255.255.0 and clear the Administration AND SIP Password.
OR enter ***# *# 7*2,** this will set the address mode to Dynamic IP and also clear the admin and SIP password.

Making a test call.
To test if the unit has been successfully registered, press '**D'** to loop the line, then dial the number required plus #. If all is well, a connection will be made.
Press **D** again to go off line. If the line is engaged or the other party hangs up the unit will automatically disconnect.
Calling a phone by its IP address can be done by entering a * between each group of digits and the port number if it's not the standard 5060.
IE, if a phone you wish to call is on IP *192.168.1.117:5080*, type
D 192*168*1*117*5080 #. (note not all phones can be called using this technique).
If the default port is 5060 for the remote phone, then it is not required when calling.

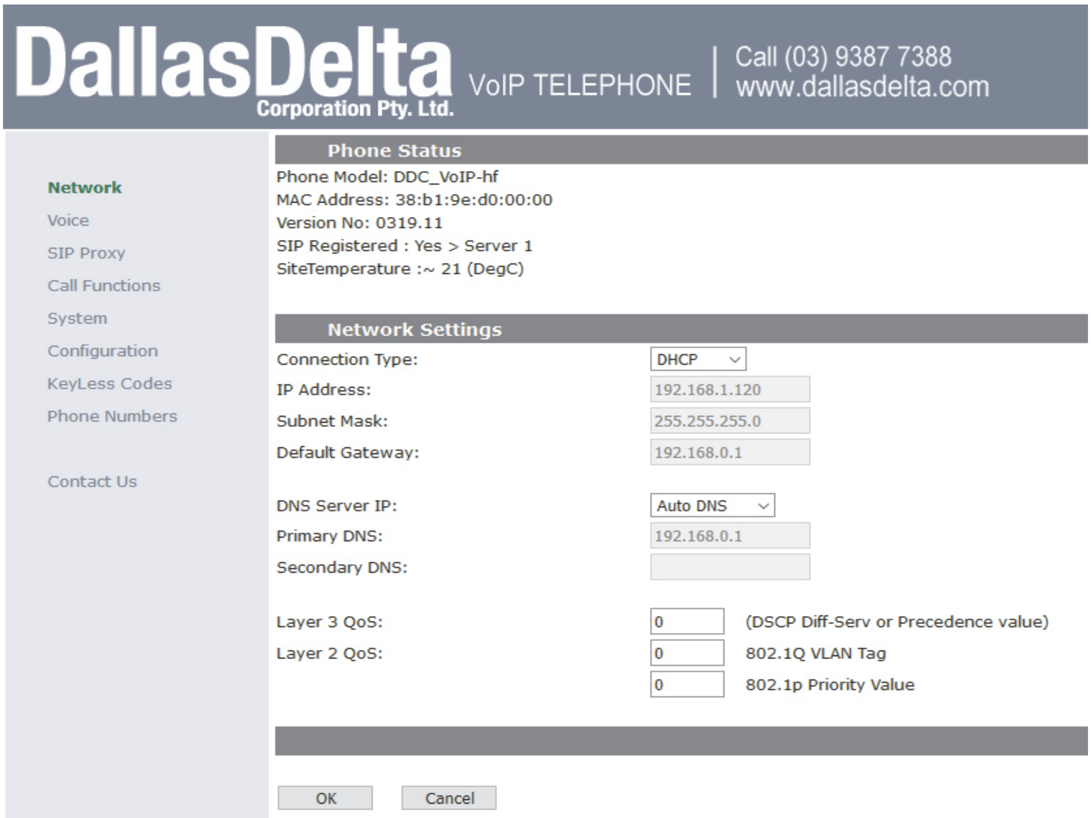
PROGRAMMING

Login page

The DDC_VoIP door station is programmed by logging into the phones web page.
Use a standard web browser (Firefox, Internet Explorer for example), 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, then refer to page 15 on **'Get / Resetting the IP address'**.
At the login page, type in the password and click the Enter key, (All units are delivered with no password).



On entering this section, the current status of the telephone will be displayed.
The SIP Register is updated every 8seconds.
Submenu’s are shown on the left of the WEB page, the first section is for ‘Network settings’ and will enable you to change IP address, subnet mask and gateway and option to set DNS server setting. The other sections are describe in the following pages of this manual.



Network Settings

The ***Connection Type*** enables the unit to be connected via a *STATIC IP or DHCP* address mode.

If the connection is DHCP then the IP address, Mask and Gateway are automatically assigned by the server.

When set to a Static IP, a suitable IP address, Subnet Mask and Gateway address will be required to suit the local area network (LAN).

If you change the IP address you are then required to restart by logging in using the new IP address.

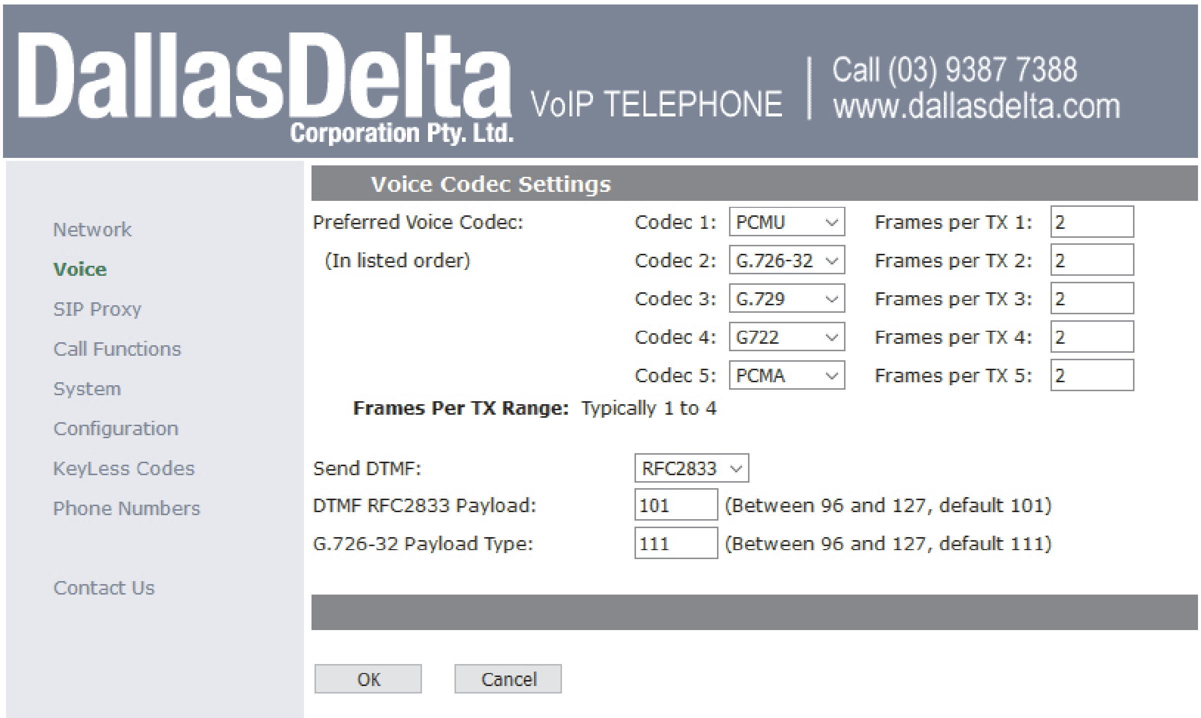
In most cases the DNS Server IP can remain set to ***Auto DNS***, in this mode the DHCP sever will assign the DNS automatically. If however, it is required, enable ***Manual DNS*** and set the Primary and Secondary DNS address.

If the network switch provides QoS traffic control, then set the ***Layer 2 & 3 QoS*** values to match.

Set the VLAN tag option for networks with VLAN option enabled, Take care here as once this is set then the PC/Laptop you are using may not have the VLAN option enabled. You will not be able to communicate with the phone.

Voice

In this section, select the order and preferred codecs, plus the amount of frames per packet to sent for each codec type.
Also, select the options that effect the different codecs used and DTMF sending method and payloads.



When selecting the codecs consider the data rate that will be required by your network. As a guild the list here show the data bit rate required for each codec, if the network is via a low speed ADSL then the PCMu/a may not be a suitable choice.

Table 1. Codec comparison

Codec	bit rate (kbit/s)
G.711(PCMu /a)	64
G726-32	32
G.729	8
G.722	32

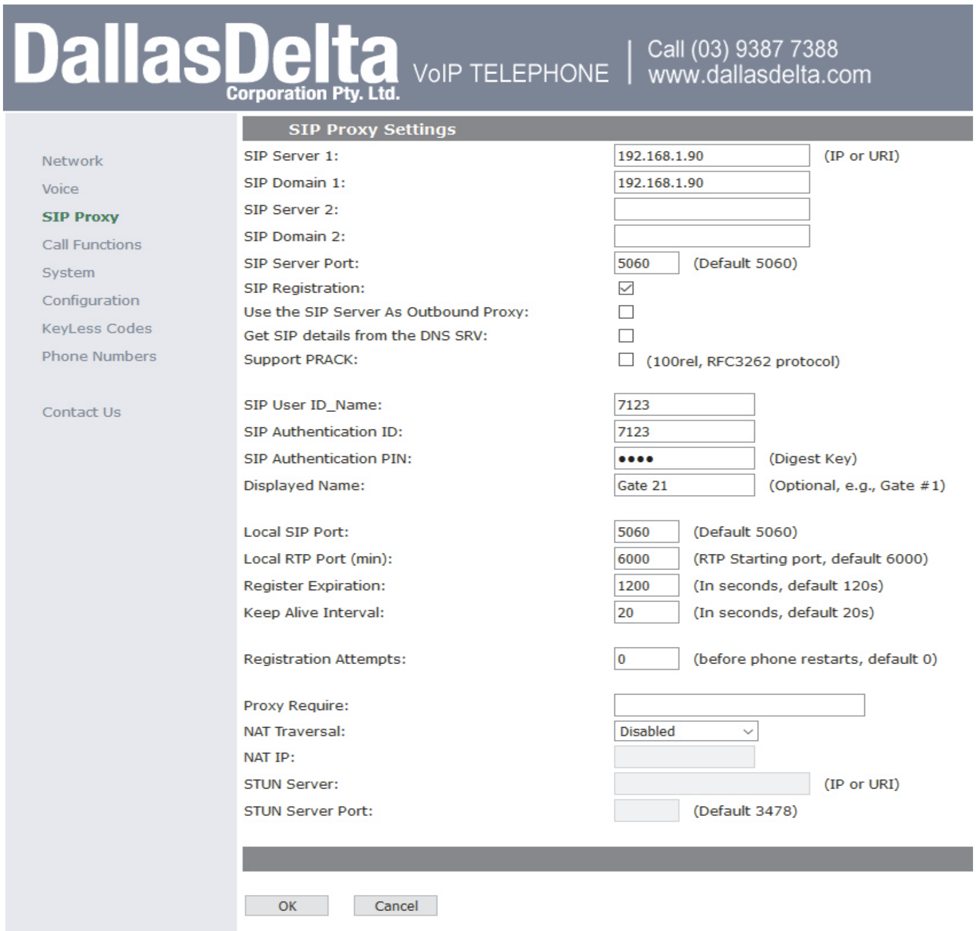
The ***Frames per TX*** selects how many blocks of voice data (10mS segments) are transmitted with each packet sent over the network, typically 2 - 4 is recommended.

The ***Send DTMF*** selects how the unit will send *DTMF* digits over the network.
There are 2 protocols available, RFC2833 and INFO.

The remaining options set the payload required for each protocol and it is recommend that the default values be used.

SIP Proxy

The parameters on this page will identify the SIP server and SIP authentication details.
Your ISP or network administrator will provide the necessary information to be used here.



SIP Server1 will be the main running location and the ***SIP Server2*** can be the fail over location if available. The User detail will be the same in both servers
When ***SIP registration*** is required with the server then enable this option. If the phone is configured to dial a direct IP location and no SIP server is used then leave this option disabled.

If the server is via an Outbound Proxy, then use the ***SIP Server*** field for this proxy and enable the option ***Use SIP Server as the Outbound Proxy***.

The user ID's, and PIN are supplied by the ISP/Network administrator, this information is case sensitive and should be typed in correctly. The ***Displayed Name*** is relayed to the connecting IP telephone as a caller ID and is optional. (Not all PBX relay this field).

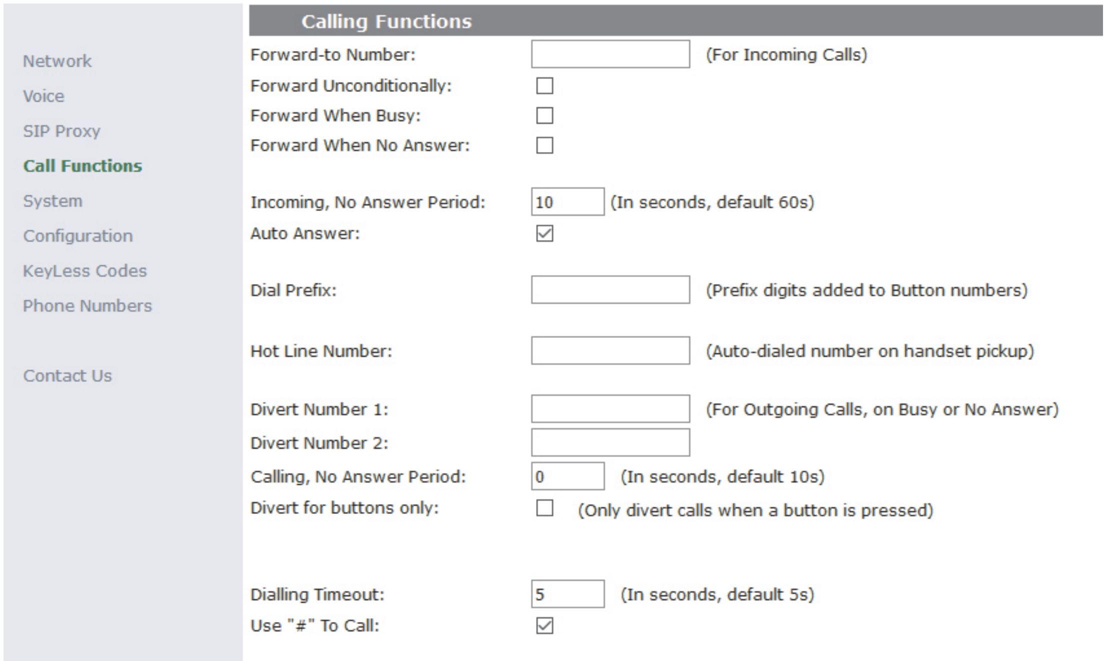
SIP and ***RTP*** port numbers are sent to the router/firewall and are use to provide paths for voice and control data to and from the phone.

The ***Register Expiration*** and the ***Keep Alive*** timers keep the SIP server mapping and NAT ports open. The ***Registration Attempts*** sets how many failed attempts before rebooting the phone, this maybe preferred to re-opbtain a new IP from the DHCP server and start from fresh.

If a *Provisional Response Acknowledge PRACK* is required within your network, then enable the ***Support PRACK(100rel,RFC3262)*** option.
When network server is behind a NAT, enable the ***NAT Traversal*** and set either the ***NAT IP*** or ***STUN server*** address and port.

Calling Functions

This section controls the way the unit makes and receives calls.



Forward to Number will set the telephone number that will be called when calls are redirected by one of three methods:
1) if the ***Forward unconditionally*** is set to yes (any calls made to the phone will be immediately redirected).
2) ***When on busy*** is set, then any attempt to call the unit when the phone is in use will be redirected.
3) Any un-answered call will also be redirected to this number when the ***Forward When No Answer*** switch is set and the ***Incoming, No Answer Timeout*** period has elapsed. (The Auto-answer option needs to be disable, for this to function).

The ***Incoming, No Answer Timeout*** period is also used the in conjunction with the ***Auto Answer*** switch. Set this option to 0 if the phone is not required to auto-answer.

The digits in the ***Dial Prefix*** are added to the any number dialled via the keypad or the preprogrammed button numbers.

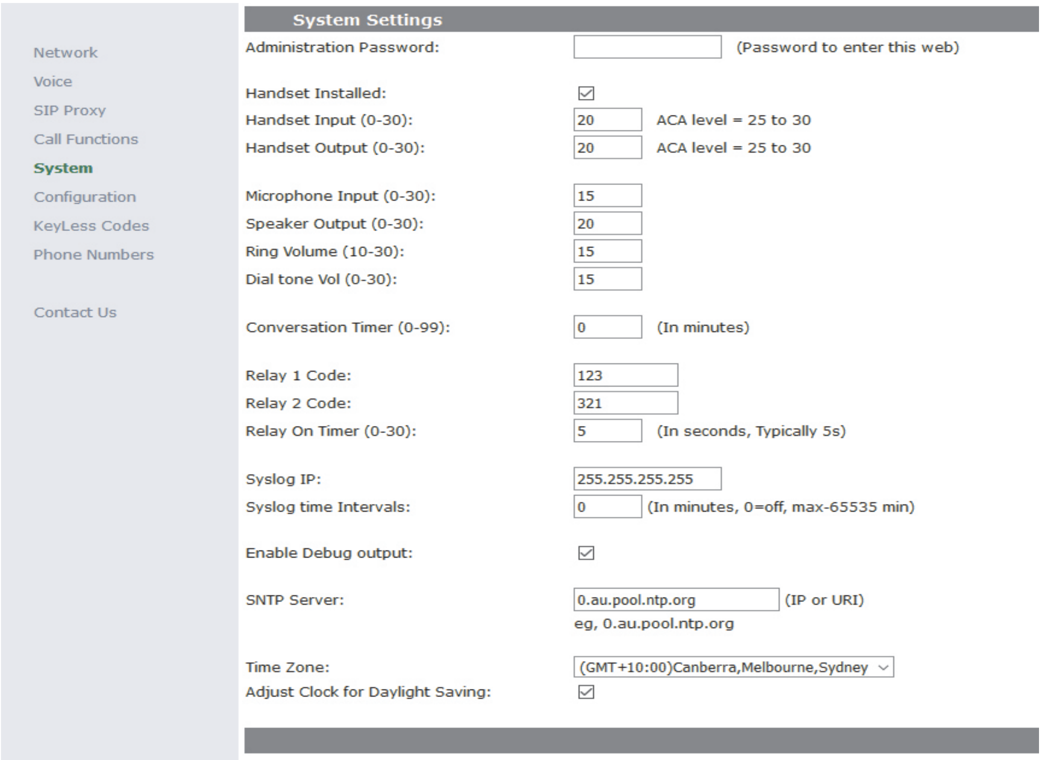
For DDC_VoIP phones that have a handset, a hotline number (dialled when the handset is pickup), maybe programmed in the field ***Hot Line Number***.

When the DDC_VoIP phone is making a call when the button is pressed, then it is possible to divert the call to 2 alternative numbers if the other party is busy or doesn't answer. To enable this function, enter the ***Divert Numbers*** and set the ***Outgoing, No Answer Period*** a value in second.
The **Divert for buttons only** is used if your unit has both buttons and a keypad installed.

The ***Dialling Timeout*** period sets how long to wait after the last digit pressed before dialling the number. The other option is to enable the ***Use # to Call*** to initiate dialling (only when dialling via a keypad).

System Settings

Set the DDC_VoIP phone audio levels, relay options, Syslog details and time of day functions in this section. Audio levels for the phone should be re-check on site.
The on board relay/s may be use for gate access control, back ground lighting / camera. They can be remotely activated or enable when the DDC_VoIP phone is in use.



The ***Administration Password*** is used to gain access to the web page. For added security set and document this password. The phone is supplied with no password set.

The microphone and the speaker levels for the handset (if fitted) and in the hands-free circuit, is adjusted to suit site background noise, when possible keep these level low, as un-necessary noise will be transmitted back to the remote operator.
If the phone does have a handset installed, then enable the option and set the audio levels as required for the handset.

Set the ***Ringer Volume, Dial Tone*** levels to suit site conditions.

Conversation period may be limited if desired, to set this, enter a period in minutes into the ***Conversation Timer*** field. If no timer is required then enter a 0 to disable this function.

The onboard relay may be activated remotely when the remote operator dials the digits that matches the fields for ***Relay 1*** & ***Relay 2 code***. Note that relay 2 is optionally fitted.
Use the ***Relay on timer*** field for the duration required. Or set it to 0, to remain on for the duration of the call.

Debugging can be enabled by Syslog server and/or UDP. If this is not required then disable these options. refer to the next page for more details on these option

NOTE: ACA maximum levels:
Do not exceed the handset maximum levels, for conformity and comfort keep handset levels to a minimum.

System Settings continue.

Debug Options:
For some installation, it may be required to record events that occur at the phone. These events may include;

- which button is pressed
- when a relay is activated remotely
- when the handset is on or off hook
- on ringing (incoming call)
- system reboot
- start of call and at auto answer
- at the end of a call
- and a status of it's current state, ie Logged-on or not at set intervals.

This information is stored at an allocated server and the protocol used is *SYSLOG on port 514*. Although generally on a Linux systems, *SYSLOG* programmes can be sought for the Windows system as well.

Each event is timed stamp with the phone ID and IP address. To enable this function set the IP of the *SYSLOG* server and the intervals.

The DDC_VoIP phone outputs general debugging information on a continuous basis on port 8225. If high traffic is a concern then disable the ***Enable Debug output*** option.

The 'time of day' used within the phone and for Syslog is set either from the SNTP server or the SIP server. If the LAN does not have access to a SNTP server then set this field to 0.0.0.0

The ***SNTP Server*** (simple network time protocol) provides 'time of day' for syslog events. SNTP is on port '123'. If the phone is connected to the internet then a site like 'time.windows.com' or '0.au.pool.ntp.org' may be used.

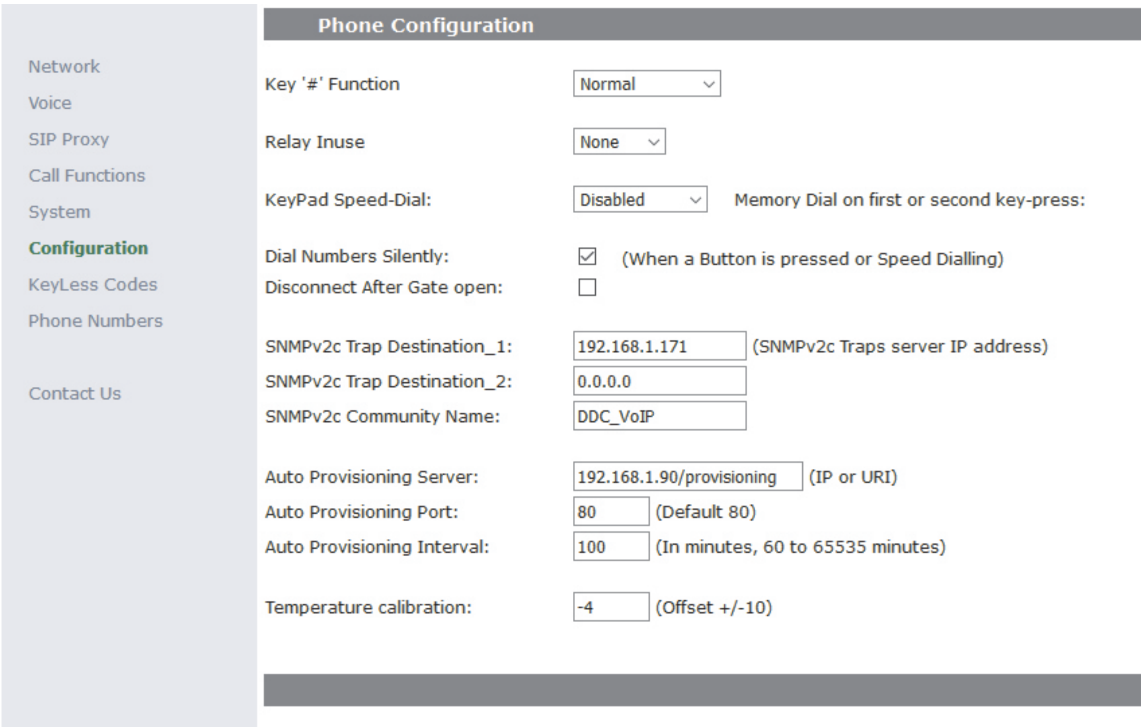
Select the State/ Country in the ***Time Zone*** option and whether to use '***DayLight saving***' time adjust is required in the next 2 fields

Configuration

This configuration section control some of the hardware and functions of the phone.
The first option ‘ ***Key # function***’ is only used for units that have a front user keypad installed. The ‘#’ key is used as the on/off key. When set to ‘on/off’ then pressing the # key will toggle the phone on-off hook. When set to ‘off only’, then it is used in conjunction with the ‘***Keypad Speed-dial***’ option (see below).

The ‘***Relay inuse***’ will allow either relay to switch on whenever the phone is being used. It will set the relay on when making and receiving a calls and stay on for the duration of the call.

‘***Keypad Speed-Dial***’ works as a memory dial command to dial the number as per the list in the ‘***Phone Number***’ section. If this is set to ‘***After 1 digit***’ then any key from 1-9 will dial memory location 1-9.
If set to ‘***After 2 digits***’ the location 01-16 can be called.
The phone will make the call to the location and the # key can be used to terminate the call.
If this option remains a ‘Normal’ then the keypad will only function after the handset is picked up and during a call to send digits.



When a call is made via pressing the button or via the Speed-Dial key, the tones of each digit may be muted, very useful for long phone number. If this suits, then enable the ‘***Dial Number Silently***’ option.

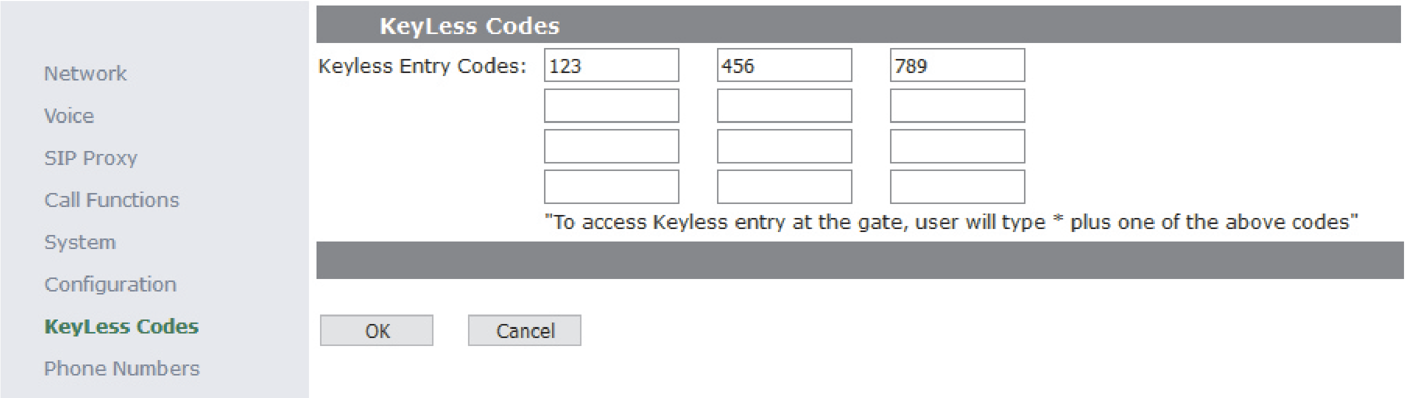
Set the option ‘***Disconnect after gate open***’ if needed.
Generally, after the relay has been activate remotely there is no further reason to continue in conversation, the phone can be set to terminate the call automatically by enabling this option.

In addition to the other debugging options, this product also can send SNMP traps to a server. It uses v2c protocol, If used then this protocol requires your community name as well as one or two servers IP addresses. It has an enterprises code of 1.3.6.1.4.1.45255.1.1.1.0 to .10.
A typical Get command would be ‘**snmpget -v2c -c DDC_VoIP 192.168.1.100 1.3.6.1.4.1.45255.1.1.1.0**’.

The ***Auto Provisioning Server*** maybe set up if the company has many phones on the network, allowing the admin officer to make changes to the functions of the phone as an alternative method to accessing via the web page. Generally, for few phones the web access is an easier method.

The Temperature output seen on the Status section can be offset by adding to the ‘***Temperature Calibration***’ value. Note that this value is only an approximation of +/-5 DegC, and only to be used as an indication of the site temperature.

Keyless Codes

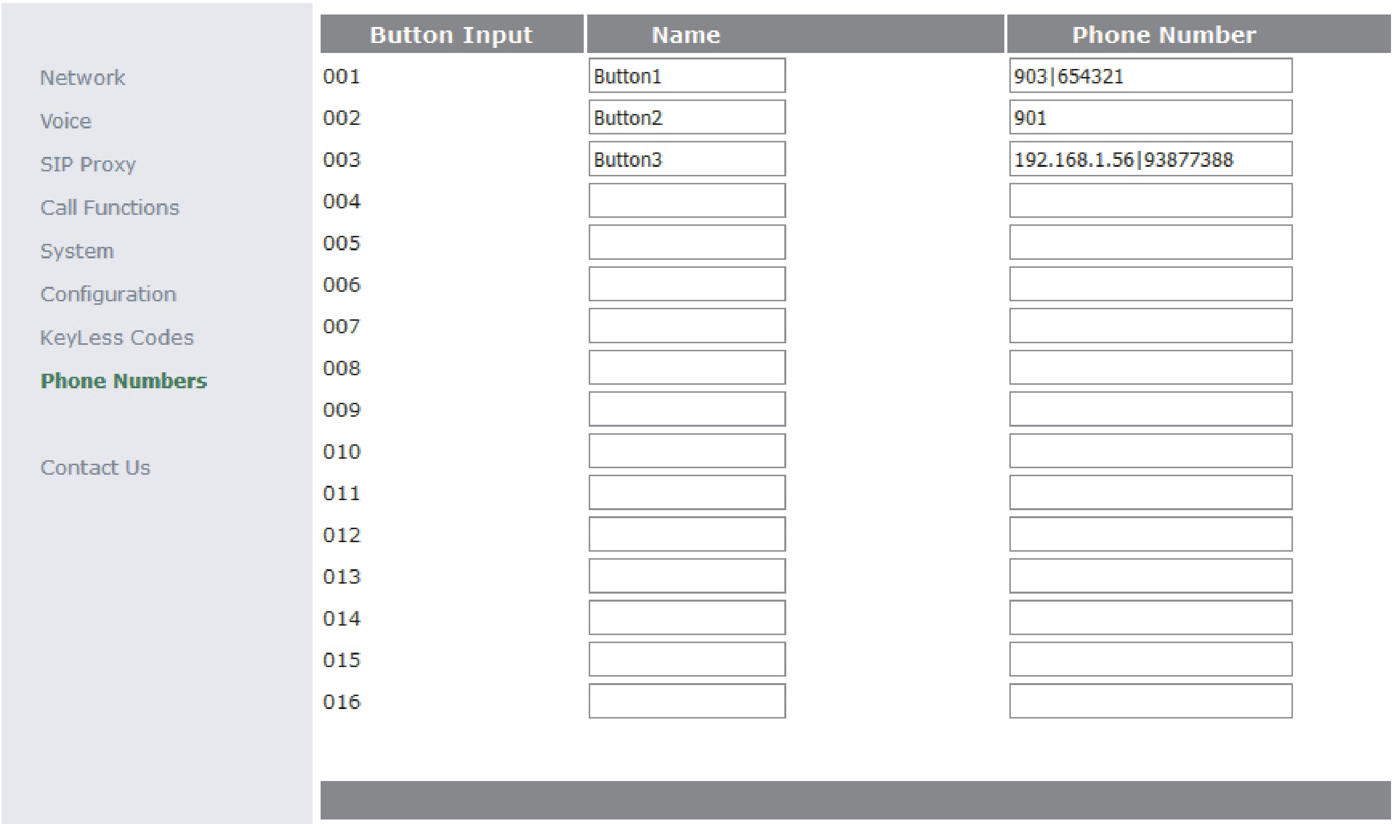


For DDC_VoIP phone supplied with a front panel keypad, the option of allowing gate access by entering a code maybe set here.
This will only activate relay1 of the phone. The period that the relay stays on for is the same as that set in the relay option section. (refer to page 21).
The user is required to enter * and the code to open the gate.
For sites that require an Emergency Services access, then it maybe advisable to use a code that is 8 to 10 digits long. As it may not be changed that easily once the code is supplied to the Emergency services personal.

Phone Numbers

The DDC_VoIP phone can be configured with up to 16 buttons,

Each Button, (if fitted) may be set to dial a different phone number. The Name field giving to each input is for internal reference only and is not required.
The phone number may be an IP address/port number, ie 192.168.1.123:5060. If the unit is not registered to a SIP server.
These locations are also used if the Speed-dial option is used when configured with a front panel keypad.
Each number maybe set to a URI location like '912345678@mysipproxy.org'



Special commands within the phone number,
The DDC_VoIP unit can be configured to connect to a PSTN/PBX via a ATA. In this scenario the unit calls the ATA and then, on answer, the phone then dials the PBX number.

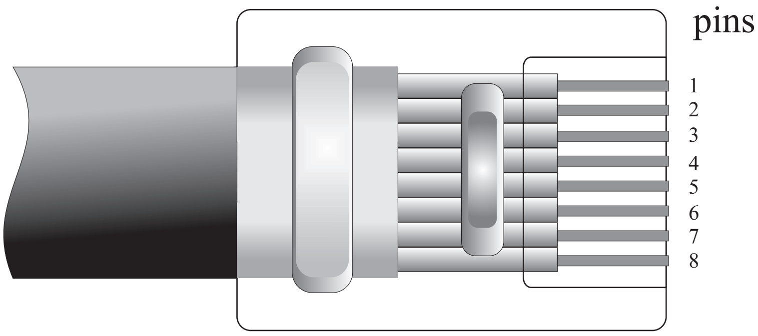
To do this, enter the number or the IP address of the ATA, followed by a pipe character '|', and then PSTN telephone number to call. (the PBX number may include a pause by adding a comma ','). Example, if the ATA number is 1234 and the PBX number is 9,91234567, then you would enter 1234|9,91234567.

Note, if all calls are made via the ATA, then the number can be loaded in the ***Dial Prefix*** location, In this case the Dial Prefix number would be '**1234|**'.

It may be required to dial this number without each DTMF tone played out over the speaker, especially so, for long phone numbers like an IP address, to enable silence dialling, enable the option ***Dial numbers silently.*** refer to page 23.

6.0 Specifications

power:	input voltage	9 Volts *(minimum)* - 50 Volts *(maximum)*		
PoE supply:	Class requirements	Class 0. (0.44 Watts to 12 Watts)		
current consumption:	-idle mode -on call	125mA @ 12Vdc (1.5 Watts) <250mA @ 12Vdc (3 Watts) <1300mA @ 12Vdc (15 Watts) *max 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		
communication:	Ethernet Connection protocol CODECs	100 BASE-T SIP G711 (uLaw, aLaw), G726-32, G722, G729		
		Sentry	Guard	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	approx. 0.9



RJ45 Ethernet connector

pins
1 = TXP
2 = TXN
3 = RXP
6 = RXN
mode A PoE

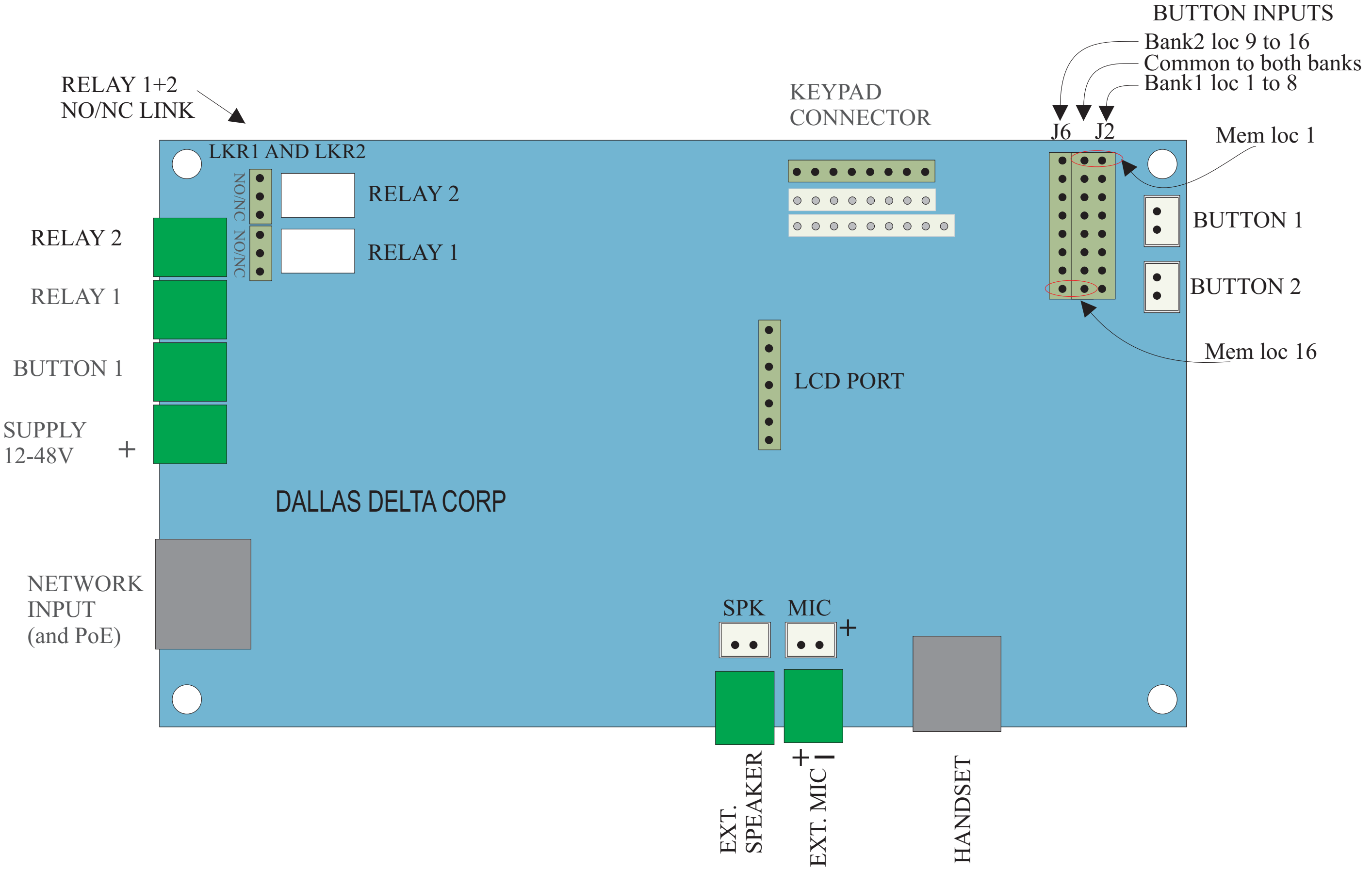
4,5 = Spare
7,8 = Spare
mode B PoE

WARNING

This telephone can not be used for emergency purposes during power failure unless fitted with a backup 12 volt battery AND network connection guaranteed.
To be installed and maintained by authorised service personnel only.

DDC_VoIP S_11 PCB Layout

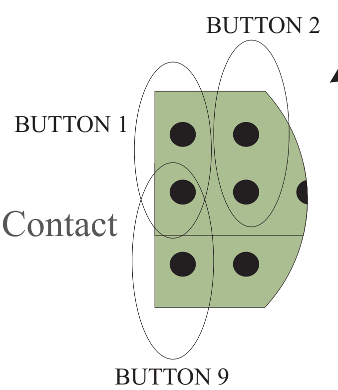
The DDC_VoIP PCB connection may be different from the diagram shown, The layout is of a generic configuration, some items may not be installed and/or not required.



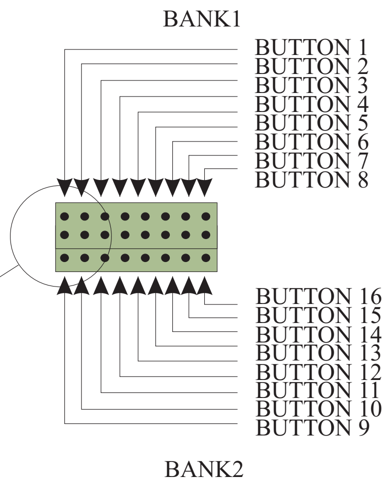
LKR1 and LKR2 relay links are used to select the normal state of the output contacts.

 Relay linked for normally **closed** contacts

 Relay linked for normally **open** contacts



Button Contact Details

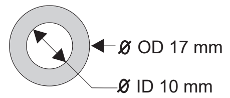


J6 and J2 Button inputs

ELECTRICAL WIRING

Note: On units fitted with external microphone and speaker, the Speaker and microphone are to be mounted no less than 100 mm apart and sealed well to the panel.

Microphone

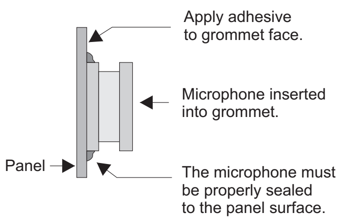


Grommet dimensions

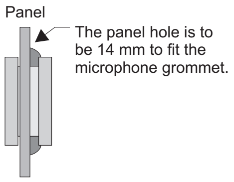
Optional External Speaker and Mic.

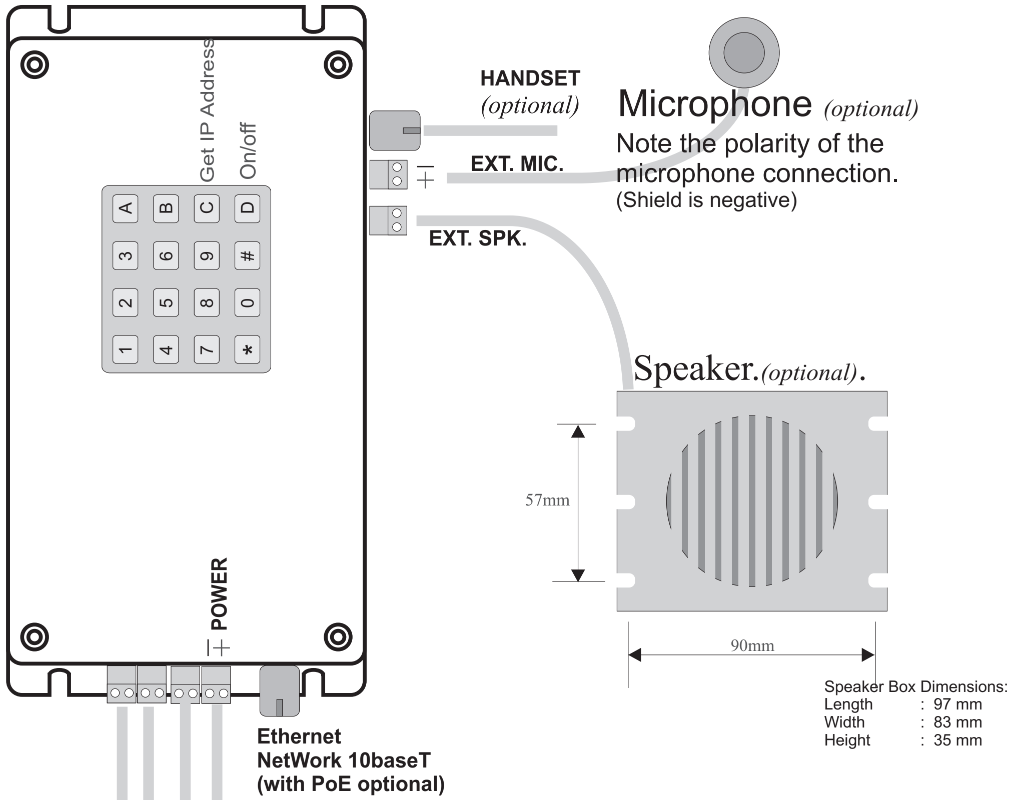
The microphone may be mounted using two methods, either mount the grommet to panel by applying an adhesive to the grommet face, or mount into a 14mm diameter hole.

Method 1



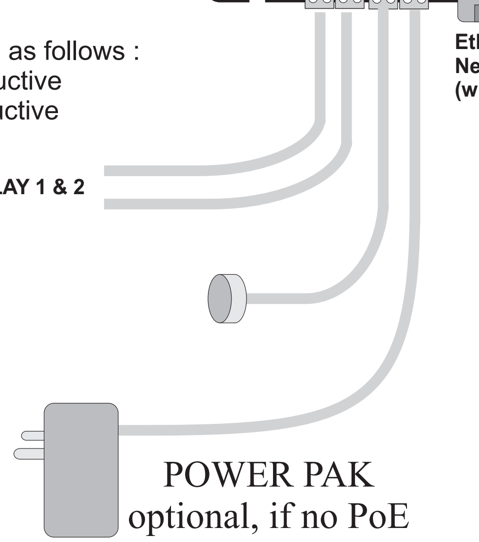
Method 2





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 LKR1 / LKR2



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 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 fitted with a backup 12 volt battery AND network connection guaranteed.

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

9.0 Release Notes

DATE:	VERSION:	REVISION NOTES:
1/5/19	PCB: S_11	Quick Start Guide and Manual

WARNING

This telephone can not be used for emergency purposes during power failure unless fitted with a backup 12 volt battery AND network connection guaranteed.
To be installed and maintained by authorised service personnel only.

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