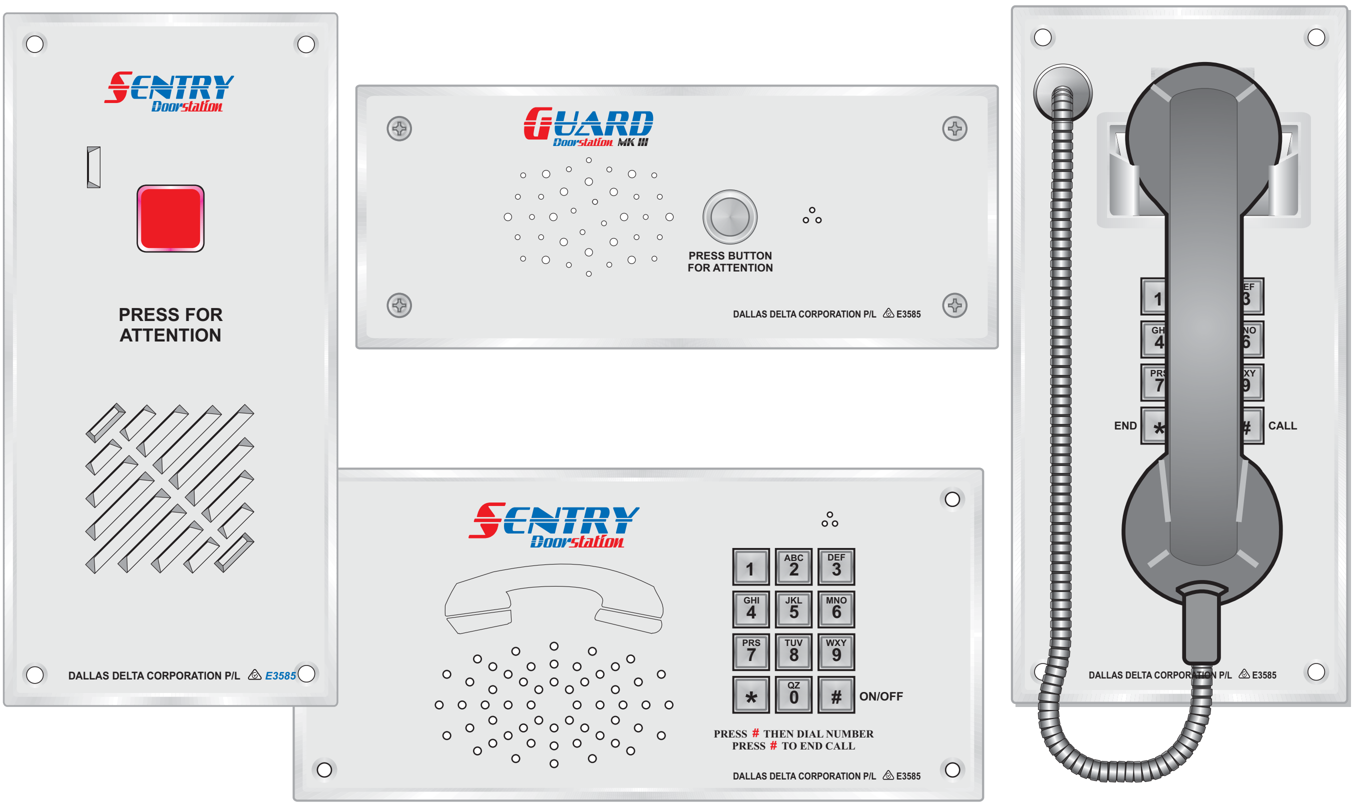
DDC VoIP V2

HANDS FREE or HANDSET
VOICE OVER IP
HELP POINT TELEPHONE



Dallas Delta Corporation Pty.Ltd.

102 Albert St. East Brunswick, 3057 Vic.
Tel: 613 93877388 Fax: 613 93873128
Email: sales@dallasdelta.com
www.dallasdelta.com

INDEX

PRODUCT INFORMATION
HELP POINT TELEPHONE
Model DDC VoIP

The **Sentry VoIP *Doorstation*** is an ethernet connected telephone and 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 (IAX2 available soon), 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.

DDC VoIP FEATURES

- Dedicated VoIP chip set
- Firmware up-gradable
- Only requiring a 10 Base-T ethernet
- SIP standard protocol
- Multi CODEC selection
- Up to 16 direct dial button inputs
- 2 on board relays with unique codes *(optional)*
- Either relay may be set for in-use function
- Relay on timer
- Hands-free with 1 Watt rms speaker output
- Handset model available
- Auto disconnect
- 2 modes of Power over Ethernet (PoE)
- Conversation timer
- All options set via HTTP
- Non volatile memory
- Asterisk PBX compatibility
- Optional dual line LCD with back light
- Remote relay activation *(door/gate release)*
- Vandal resistant
- Weather shielded *(optional)*

OPERATION

The DDC_VoIP telephone operates like a standard hands free telephone with many added features.

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 then 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.

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.
(refer to ***Relay Inuse*** on page 11 for more on this function).

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 11).

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.) This remote control requires RFC2833 dtmf protocol set throughout the system.

Call initiated relay control:

Either one of the relays may be configured to switch on after the call button is pressed. 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, unless the remote operator re-activates it by entering the code, then the relay will switch off after the ***Relay On timer*** period.
(again, see page 11 for more details).

WARNING

This telephone can not be used for emergency purposes during power failure unless fitted with a backup 12 volt battery!
and only if network connection is guaranteed.

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. Make test calls and control speaker level during the call.

Get current IP address;

During standby, press '**C**' on the rear keypad to retrieve the current IP address.
The address will continue to repeated, press 'D' to end the cycle and return the unit to standby mode.

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 set address mode to *Static* the IP address to 192.168.1.100, set the subnet mask to 255.255.0.0 and clear the Administration Password.
OR enter ***# *# 7*2,** this will set address mode to Dynamic IP and also clear the admin 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. IE, if a phone you wish to call is on IP *192.168.1.117:5060*, type
D 192*168*1*117*5060 #. (note not all phones can be called using this technique).
If the default port is 5060 for the remote phone, then it may not be required when calling.

Volume control 'B' & 'C'

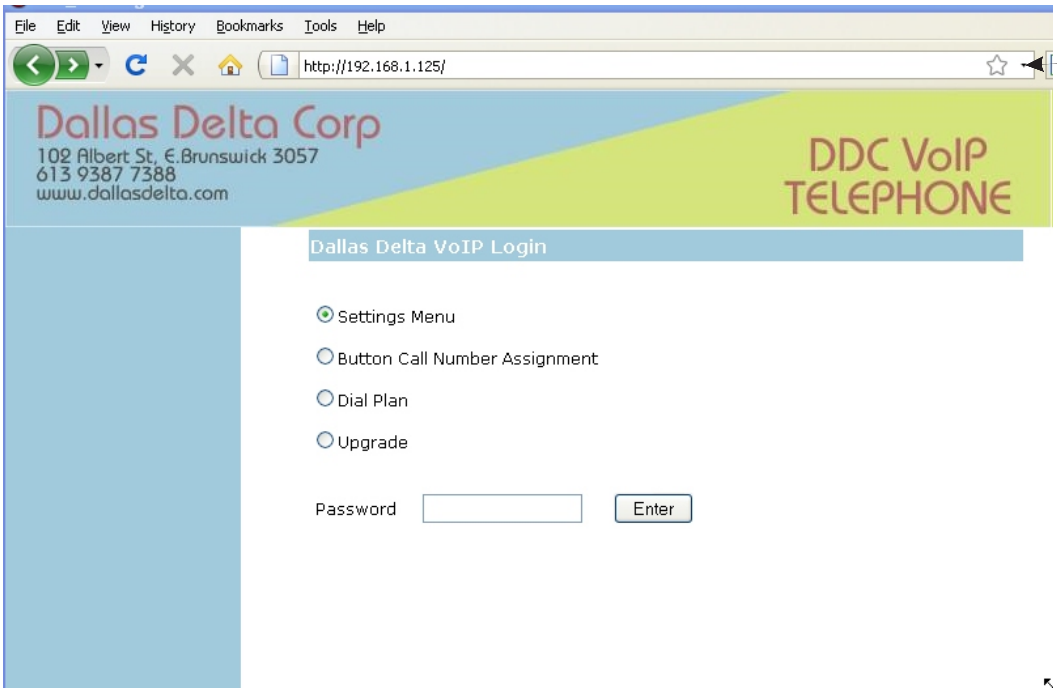
During conversation, the speaker volume may be increased by pressing **B** or decreased by pressing **C**. These adjustments only remain as set for the duration of the call and do not affect the default level.
To set the default level refer to ***System Settings*** on page 11.

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 5 on **'Get / Resetting the IP address'**.
At the login page, select one of the following;

- **'Settings Menu'** to set network and telephone parameter.
- **'Button Call Number Assignment'** to set the phones number for each button input.
- **'Dial Plan'** telephone number masks used for units fitted with front panel keypads.
- **'Upgrade'** option if you required to change the ring tone.

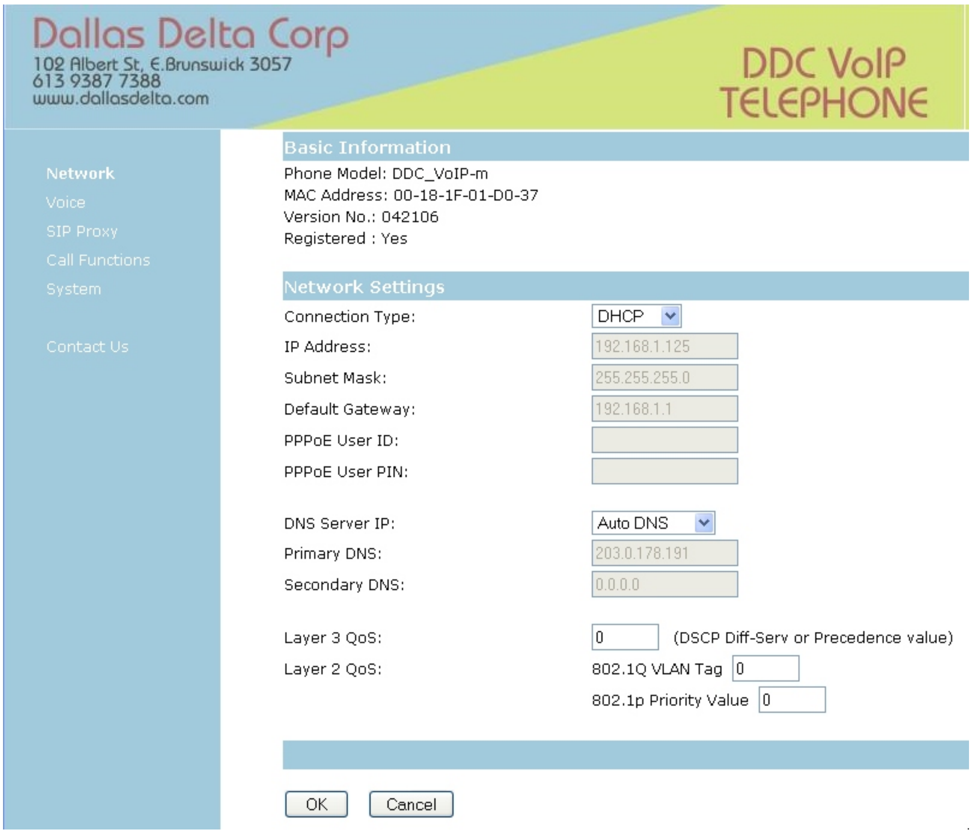
type in the password and click the Enter key, (units are generally delivered with no password).



URL Location Bar

SETTING MENU

On entering the Settings Menu, the current parameters of the telephone will be displayed. 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, DHCP or PPPoE* address.

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.

Alternatively, if connection is via a Internet Service Provider (ISP) then select PPPoE and set the user ID and PIN as supplied to you from your ISP.

Note that for some servers, the connection type may need to be set to DHCP.

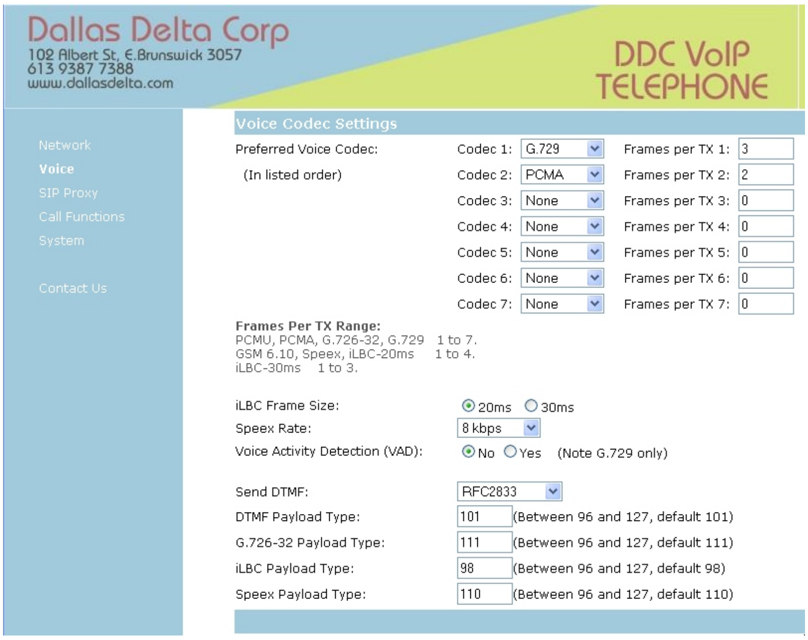
If you change the IP address you are then required to restart by logging in using the new IP address. Do not click on 'go back one page' of the browser (usually in the top left corner).

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.

Voice

In this section, select the order and preferred CODEC's, plus the amount of frames per packet to sent for each.
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
GSM	13.2
iLBC 20ms	15.2
iLBC 30ms	13.3
Speex	2.15 - 24.6
G.729	8

If the iLBC or SPEEX codecs are used, then set the following for each,
a) **iLBC** codec, select the frame size your network requires to 20mS or 30mS.
b) **SPEEX** codec, select a suitable bit rate for best quality available, the range is from 2.15kBits/s to 24.6kBits/s.

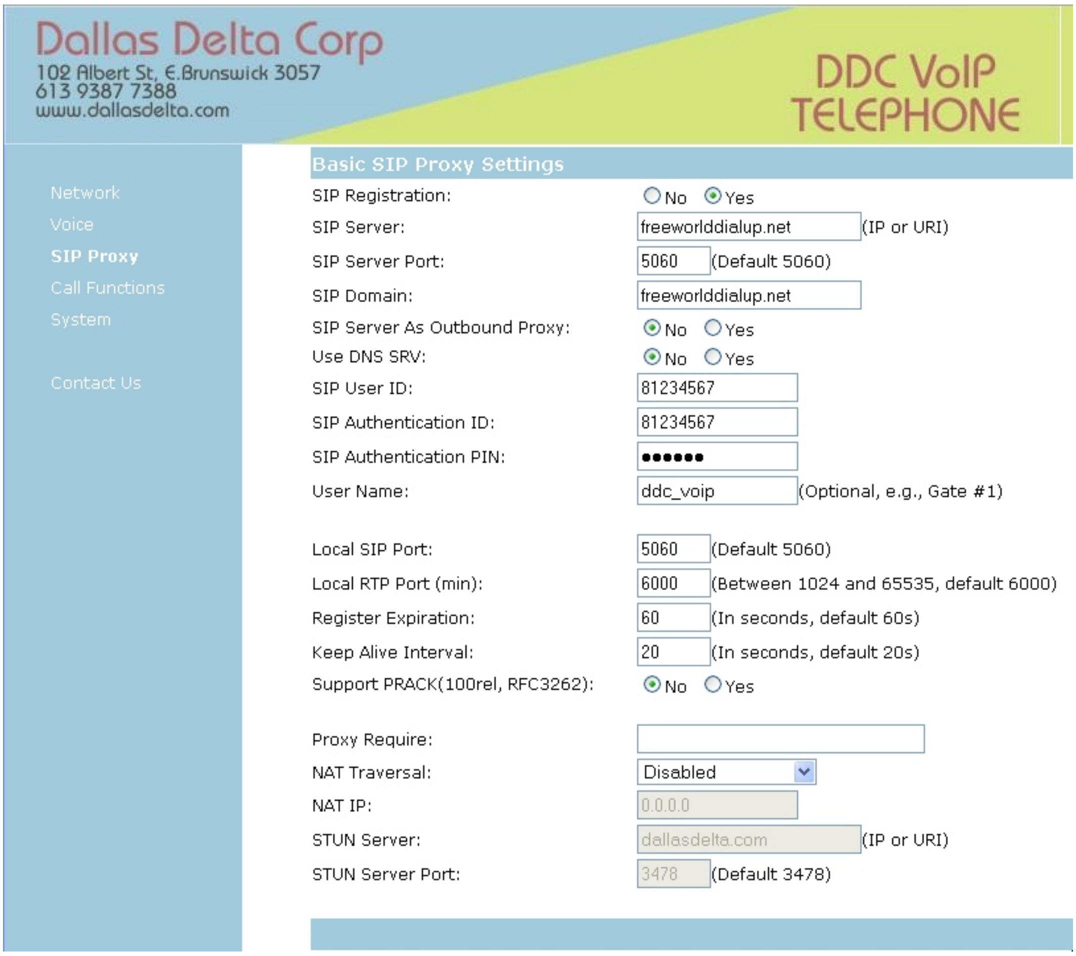
The ***Frames per TX*** selects how many blocks of data (10mS segments) are transmitted with each packet sent over the network. (note the higher the number, the more the delay), 2 - 4 is recommended.

The ***Send DTMF type*** selects how the unit will send *DTMF* digits over the network (only required for unit fitted with a front panel keypad and if digits are dialled during a conversation).

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 server that will be used for a call setup to and from the DCC_VoIP phone. Your ISP or network administrator will provide the necessary information to be used here.



If ***SIP registration*** is required with the server then set this switch to Y***es***. If the phone is configured to dial a direct IP location then this switch can be set to ***No***.

Type the Server address and the port (usually 5060 for SIP) into the next 2 input boxes. The ***SIP Domain*** is the address where an outgoing call information is sent to, and if the domain is via the internet then it will require the ***Outbound Proxy*** to be set to ***Yes***.

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 ***User Name*** is relayed to the connecting IP telephone as a caller ID and is optional.

Local 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, these interval are generally set to 60 and 20 seconds respectively.

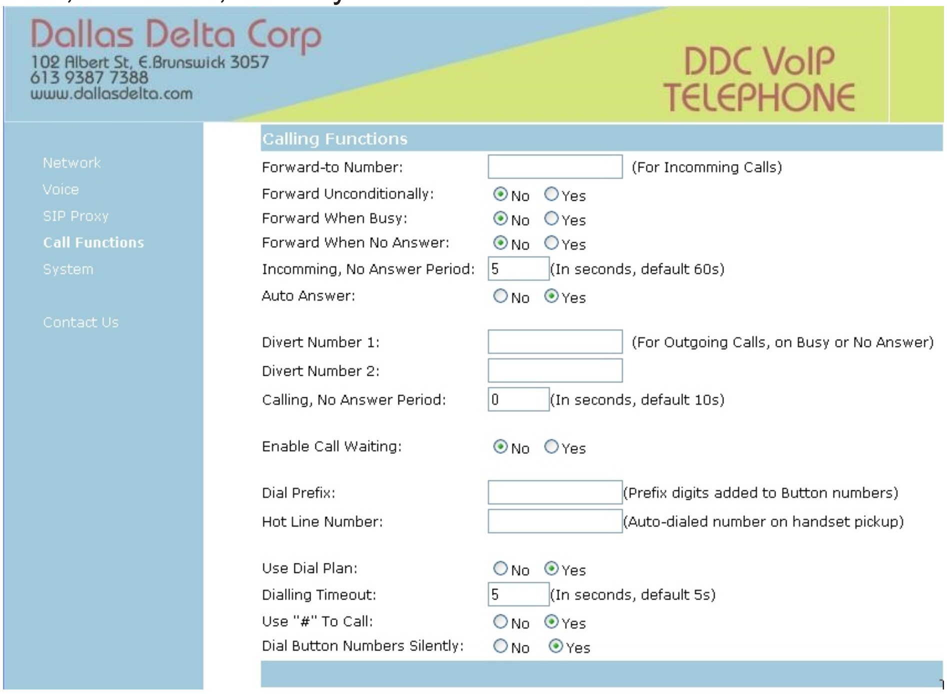
If a *Provisional Response Acknowledge PRACK* is required within your network, then enable the ***Support PRACK(100rel,RFC3262)*** option.

The SIP ***Proxy Required*** field may need to be set for Nortel MCS servers, if so, set this field to "*com.nortelnetworks.firewall*"

If the network server is behind a NAT, then 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.
Select options, as to what should happen when calls are made to the unit, whether to accept or redirect it, during the different modes of operation.
When making calls, numbers may be screened (via the dial plan) or prefix digits can be added.
Also, call divert, on busy or no answer.



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) if, 'When on busy' is set to yes, then any attempt to call the unit when the phone is in use will be redirected to the ***Forward to Number***.
3) Any un-answered call will also be redirected to this number when the ***Forward When No Answer*** switch is set to yes and the ***Incoming, No Answer Timeout*** period has elapsed.

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.

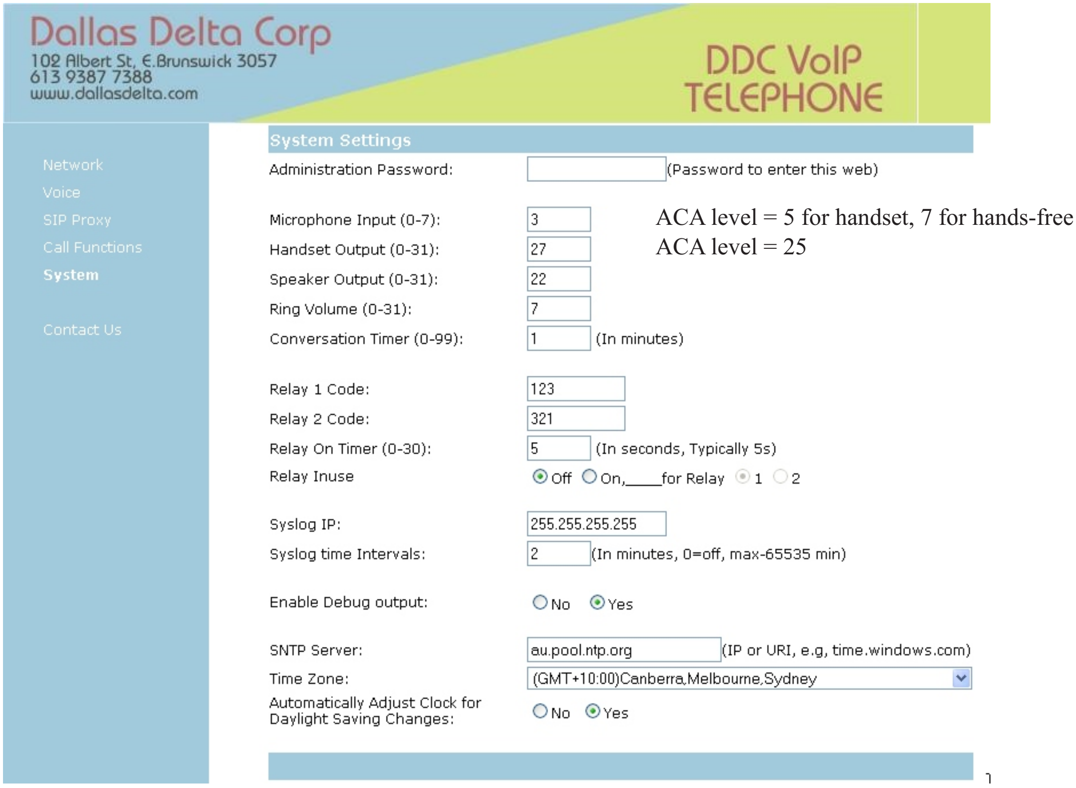
When the DDC_VoIP phone is making a call via a button press procedure, then it is possible to divert the call to 2 alternative numbers if the other party is busy or doesn't answer. To enable this function, enter the ***Divert Numbers*** and set the ***Outgoing, No Answer Period*** the a value in second the phone should wait for a pickup by the remote operator prior to diverting.
If call waiting, (audible tones transmitted to the speaker when an incoming call arrive during conversation) is required then click ***Yes*** to ***Enable Call Waiting.***

The number in the ***Dial Prefix*** are added to the number dialled via the keypad or the preprogrammed button numbers. (Also refer to page 13).
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***.
If the unit is supplied with a keypad, then use the **Dial Plan** option to restrict numbers that can be called. To enable this feature select yes for the ***Use Dial Plan*** option and set the dial plan fields, refer to page 14.
The ***Dialling Timeout*** period sets how long to wait after the last digit pressed before dialling the number. Note, If the dial plan option is on and the number type matches, then the number will be dialled without delay. The other option is to enable the ***Use # to Call*** to initiate dialling.

When a call button is pressed, the audible tones of each digit dialled may be muted. This option may be preferred if the number is long or for security reasons. If this feature is desired, then select the ***Yes*** option to ***Dial Button Numbers Silently***.

System Settings

Set the DDC_VoIP phone audio levels, relay options, Syslog details and time of day functions in this section. Audio levels for each phone should be recheck 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 these web pages. For added security set and document this password. The phone is supplied with this field cleared.

The microphone used by the handset (if fitted) and in the hands-free circuit, is adjusted to suit site background noise, when possible keep this level low, as un-necessary noise will be transmitted back to the remote operator.

The Handset and speaker phone levels are also set to suit back ground noise. Whilst in conversation, this level may be temporary altered by pressing the ‘B’ and ‘C’ key on the back of the unit to increase or decrease the level respectively.

Set the ***Ringer Volume*** level to suit site conditions.

The onboard relay may be activated remotely by dialling the digits that matches the fields for ***Relay 1*** & ***Relay 2 code***. Note that relay 2 is optionally fitted.

The period the relay will stay active is set in the ***Relay on timer*** field. Setting this field to 0, will cause the relay to remain on for the duration of the call.

Either one of the relays may be configured to switch on when the phone is in use. If this is required then set the ***Relay Inuse*** option to ***On*** and select one of the available relays. Note, that when this option is used the ***Relay on timer*** period is bypassed.

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.

NOTE: ACA maximum levels for:
handset models be set 5 for microphone and 25 for handset output.
hands-free microphone level to be set to 7.

System Settings continue.

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*. Although generally on a Linux and Unix 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* enabled server and the intervals into the next 2 fields.

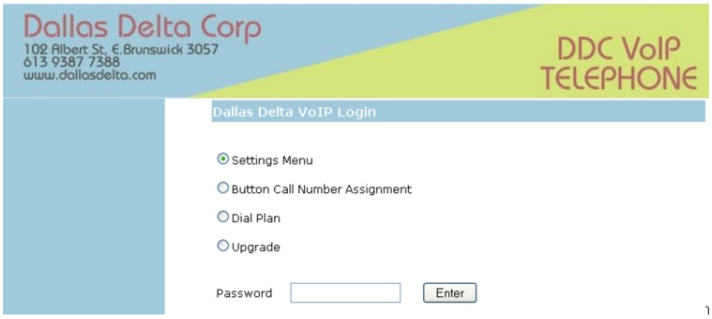
The DDC_VoIP phone outputs general debugging information on a continuous basis, If high traffic is a concern then set the ***Enable Debug output*** option to No.

The ***SNTP Server*** (simple network time protocol) provides 'time of day' that is uses by the unit for sysloging events. Most PC on the network with port '123' open on the firewall will do this or if

PROGRAMMING (Phone Number menu)

Button Call Number Assignment

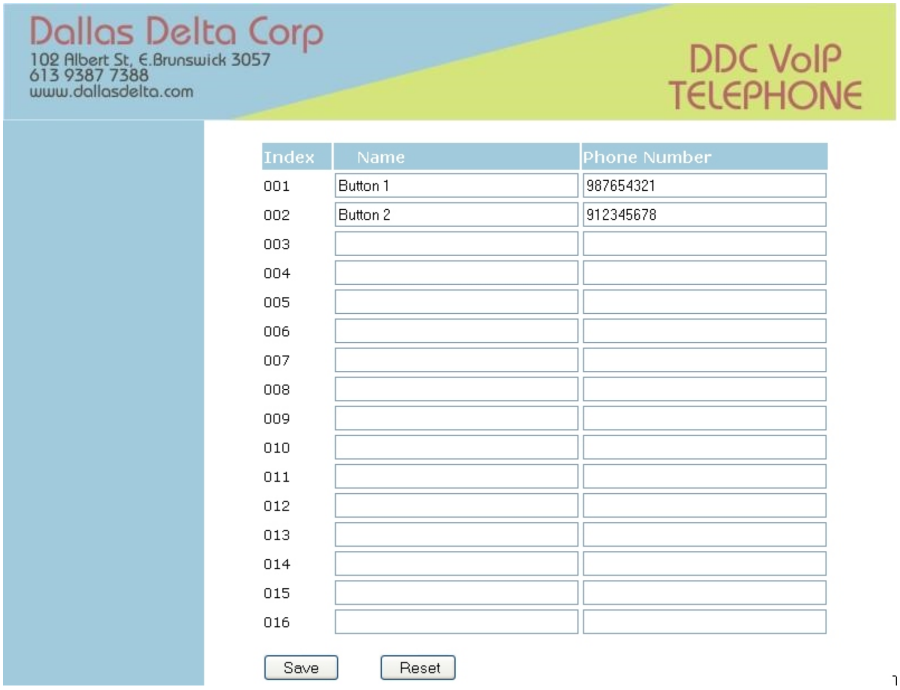
The DDC_VoIP phone can be configured with up to 16 buttons, each may be assigned to dial a different telephone number.
Click on the ***Button Call Number Assignment*** radio button and input the access code, then click 'Enter'.



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.

Sometimes, it may be required to dial this number without each DTMF tone played out of the speaker, especially so, for long phone numbers like an IP address, to enable silence dialling click ***Yes*** to the option ***Dial numbers silently*** in the ***System section***, see page 9.



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 DDC_VoIP phone then dials the PBX number.

To archive this, enter the first part of the number (the section that call the ATA) followed by a pipe character '|', and the PBX 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.

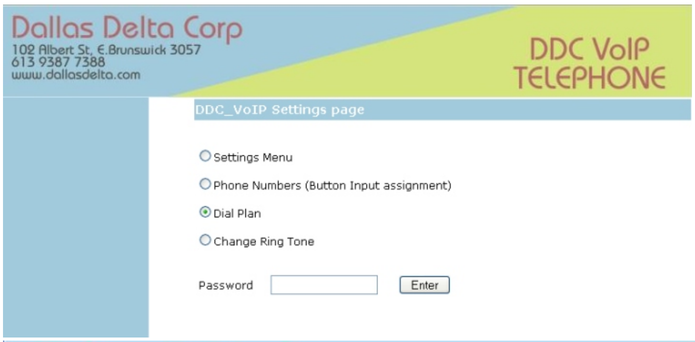
Note that the ATA number can be loaded in the ***Dial Prefix*** location, if all buttons calls are made via the ATA, then set the Dial Prefix to '**1234|**'.

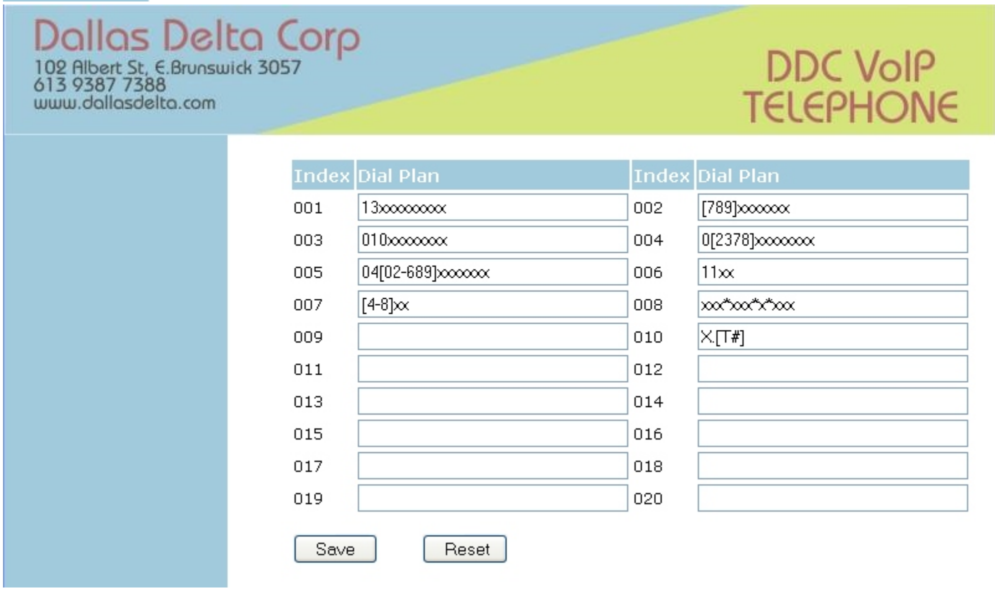
Note that the ATA number could well be an IP address.

PROGRAMMING (Dial Plan menu)

Dial Plan

The DDC_VoIP phone can be supplied with a handset and/or a front panel keypad. With this configuration the user may dial any combination of digits, therefore you may wish to screen the phone numbers that can be dialled. This section describes the techniques to accomplish this. Click on the ***Dial Plan*** radio button, input the Password and click 'Enter'.





Each index dial plan shows a possible group of digits that are allowed, in all there are 20 groups.

To enable dial plan control, select ***Yes*** for ***Use Dial Plan***, in the ***Calling function*** section of the ***Settings Menu***, please refer to page 10 for more details.

When making a call, numbers that are accepted must match one of the groups above.

Digits **0** to **9** that are shown in each group will be tested, whereas the 'x' within the group indicates that any digit is accepted.

Digits within the square brackets [ ], indicate a range that are allowed at that location within the group. For instance, 03[02-6], means that the first two digits must be 03, the next digit can be 0 *or* any digits 2 to 6, therefore 1,7,8,9 are not allowed.

Another example, 037[189] indicates that only digits 1,8 and 9 are allowed after the 037 combination is entered.

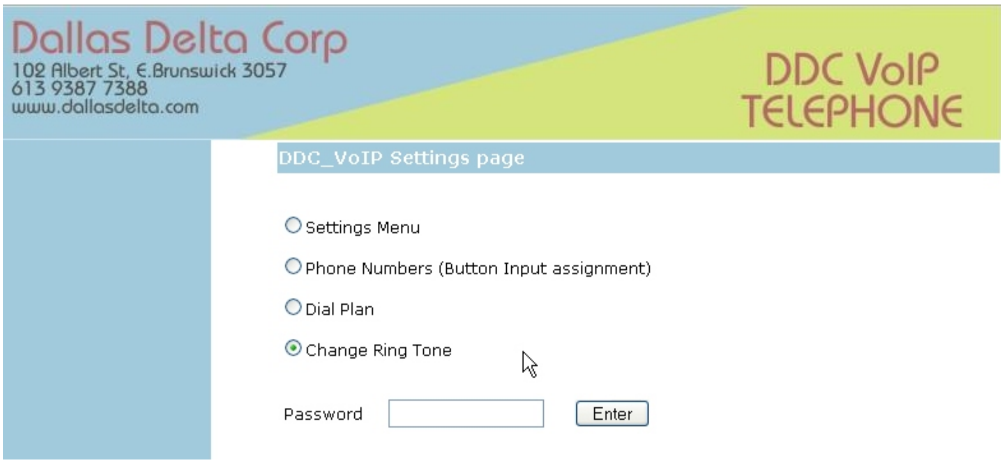
When the number dialled match one of the groups in the dial plan, the unit will make a call as soon a match has occurred.

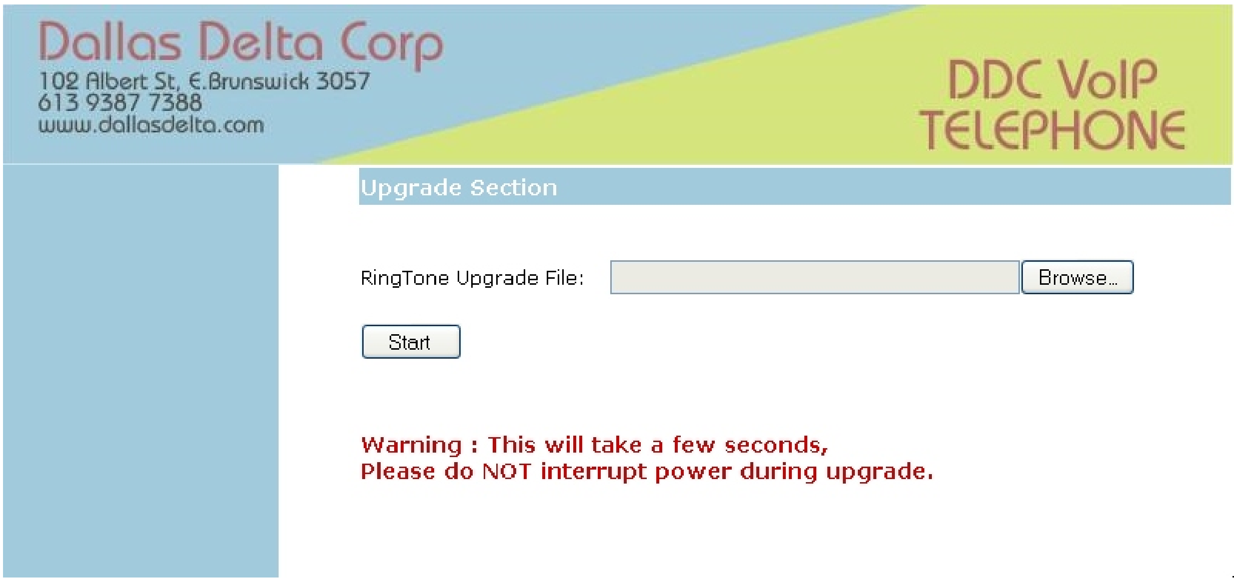
The ***Dialling Timeout*** option (see page 10) set a period to automatically dial the number, if a '#' is not entered to end the sequences of digits OR a match **did not** occur in the dial plan.

PROGRAMMING (Ring Tone)

Changing The Ring Tone

When a call is made to the DDC_VoIP telephone, the audible sound heard (ringing signal) may be changed to an alternative tone or perhaps a voice message or warning alarm.
To change this tone, select the ***Change Ring Tone*** option, enter the access Password and click ‘Enter’.





The ring tone sound file needs to be in the correct format. Wave and MPG-3 files etc. need to be converted before it is loaded into the unit. Conversion programs are available that can do this, CoolEdit pro is a good example.

The sound file used MUST comply with the following specifications:

- The file length MUST not exceed 390 kBytes AND a maximum length of 48 seconds.
- Must be 8000 bit/s sampling rate, Mono, PCM raw 8 bit format using muLaw codec.
- Save the file with a ‘.DAT’ extension.

Once the file has been created, select Browse, and pick the file to be loaded, then click on the Start button.

Within approximately 10 seconds the phone should response with a successfully update message.

WIRING

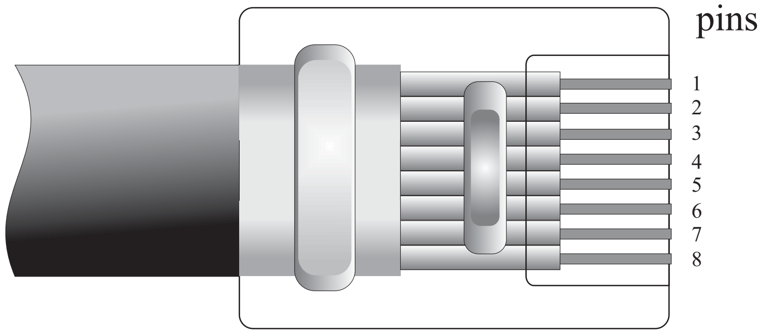
Power Over Ethernet (PoE)

The DDC_VoIP telephone may be powered from the 2 way connector marked ‘Power’ on the side of the unit, see page 19. 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 1 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

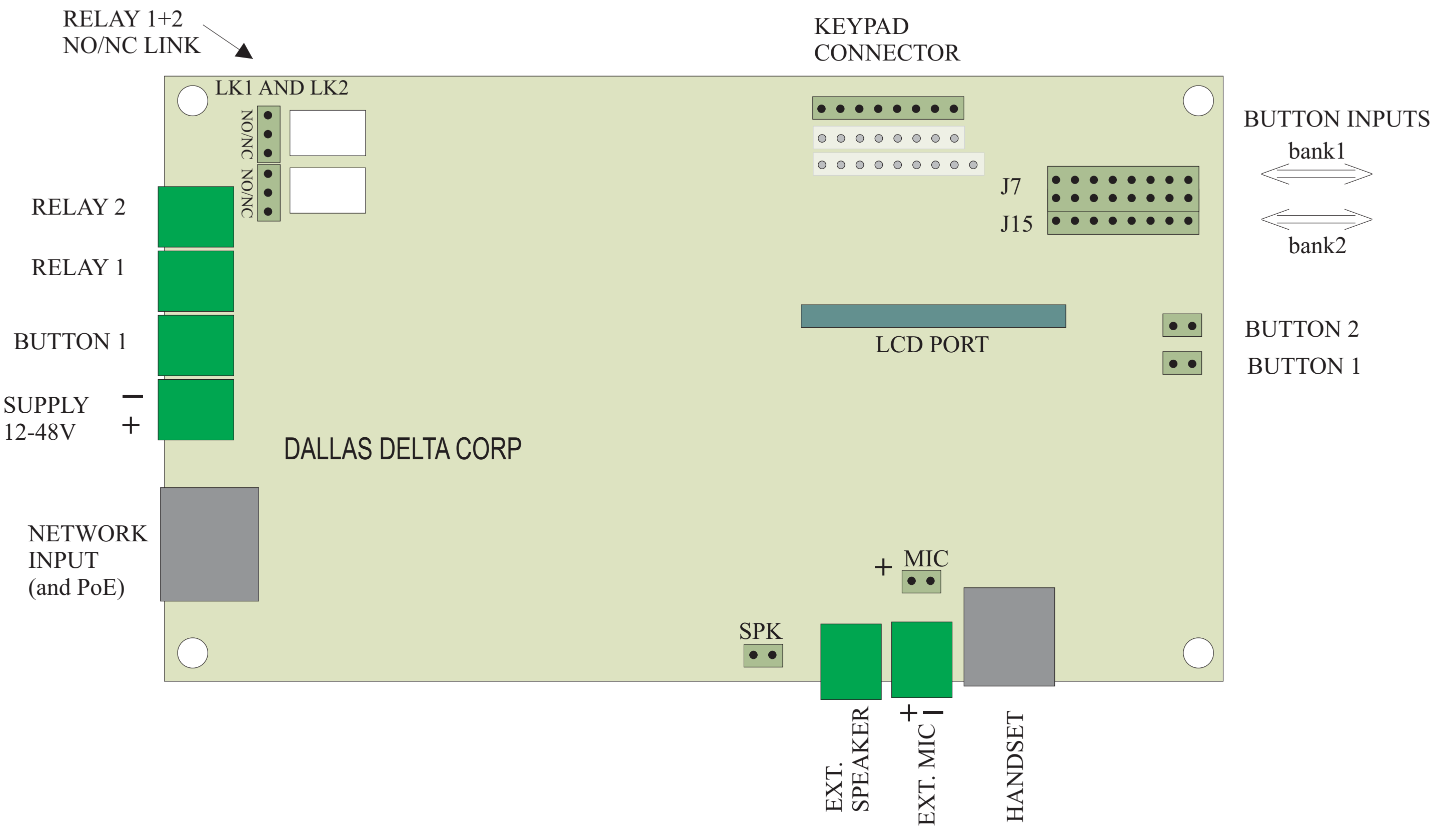
The figure above shows a typical RJ45 connector with the pins side facing up.
Note, In mode B, that the spare wires 4 and 5 are linked together within the PCB, the same for pins 7 and 8.

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 .

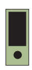
DDC_VoIP PCB Layout

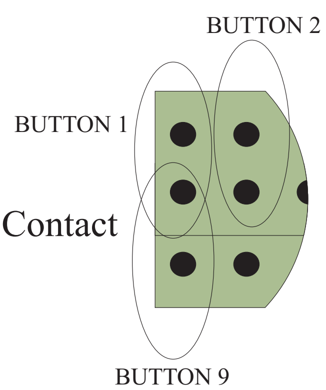
The DDC_VoIP PCB connection may be different from the diagram shown, The layout is of a generic configuration, some items may not be installed and/or not required.



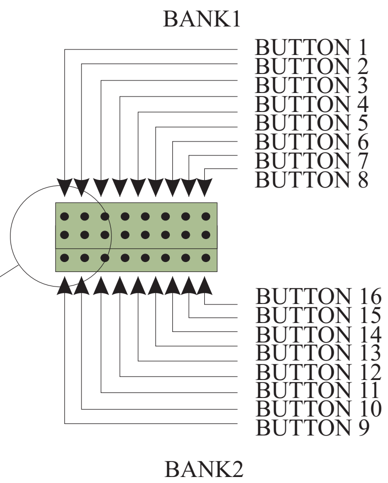
LK1 and LK2 relay links are used to select the normal state of the output contacts.

 Relay linked for normally **closed** contacts

 Relay linked for normally **open** contacts



Button Contact Details



J7 and J15 Button inputs

SPECIFICATIONS

power:	input voltage	9 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 CODECs	10 BASE-T SIP G711 (uLaw, aLaw), Speex, iLBC, G726-32 GSM, G.729		
		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 fitted with a backup 12 volt battery AND network connection guaranteed.
To be installed and maintained by authorised service personnel only.

ACA optimum handset levels

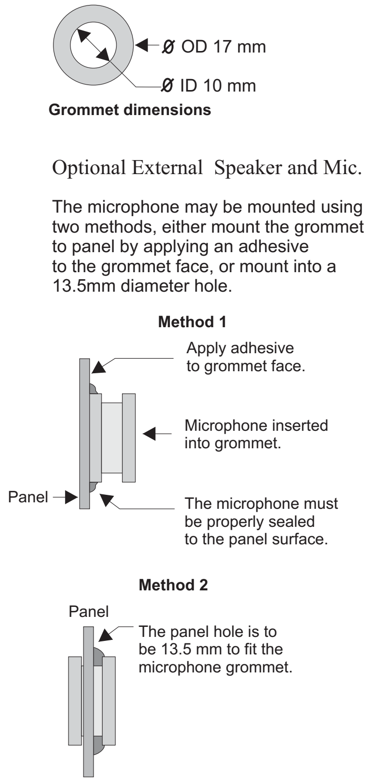
Microphone = 5, handset output = 25.
Microphone = 7 for hands-free.

CONNECTION DETAILS

ELECTRICAL WIRING

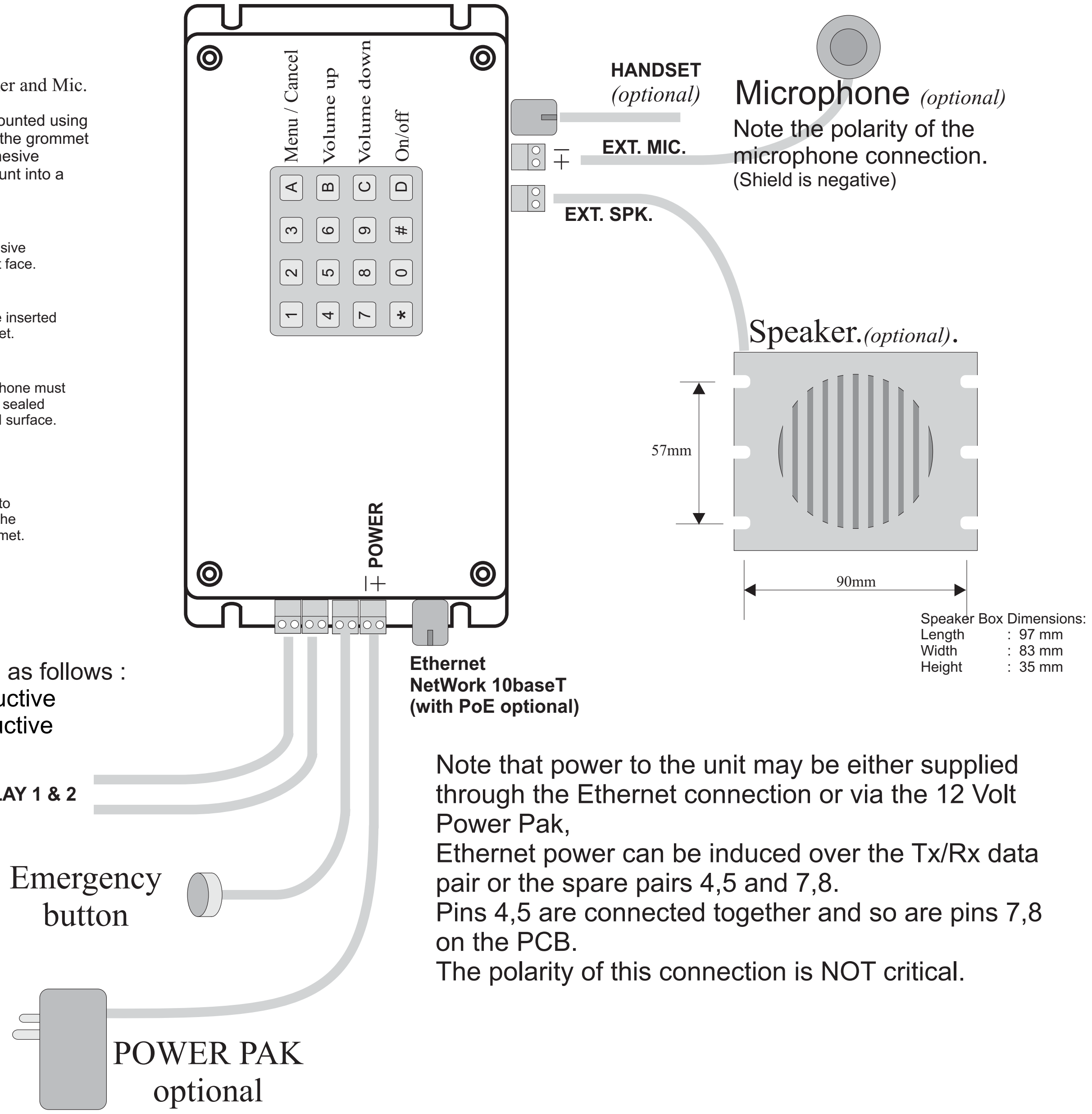
Note: On units fitted with external microphone and speaker, the Speaker and microphone are to be mounted no less than 100 mm apart and sealed well to the panel.

Microphone



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 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.

Dallas Delta Corporation Pty.Ltd.

102 Albert St. East Brunswick, 3057 Vic.
Tel: 613 93877388   Fax: 613 93873128

Email: sales@dallasdelta.com
www.dallasdelta.com

Manufacturers of:

Emergency Lift Telephones
Emergency Telephone Systems
Emergency Services Telephones
Industrial Telephones
Hygienic Environment Telephones
Freeway Telephones
Rugged Environment Telephones
Hotel Telephones

Prisoner Telephones
Prisoner Phone Monitor Systems
Security Door phones
Door phones
Loud Ringer Horns
High Voltage Line Isolators
Loudspeaking Telephones

and
they're
All

MADE IN AUSTRALIA